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Electra **Elite** IPK II

**FEATURES AND
SPECIFICATIONS MANUAL**



Empowered by Imagination

NEC

**INT-1072 (IPK II)
DOCUMENT REVISION 4C
(VERSION 2100)**

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Technology Development

Preface

Before Reading this Manual

This manual provides detailed information for each of the system's features. If you are not familiar with the features, the Table of Contents provides a list of the features and where to find the feature within the manual.

GENERAL INFORMATION

Congratulations! You have purchased the NEC Electra Elite IPK II System.

The Electra Elite IPK II system is a feature-rich key system that provides many features including Automatic Call Distribution, Computer Telephony Integration, IP Station and IP Trunk support, ISDN compatibility, PBX compatibility, TAPI compatibility, Voice over Internet Protocol and Uniform Call Distribution.

The Electra Elite IPK II system meets the customer needs today, and as business expands, the system can be expanded to grow as well.

The Electra Elite IPK II system has a set of manuals that provide all the information necessary to install and support the system. This preface describes these manuals.

SUPPORTING DOCUMENTS

Electra Elite IPK II General Description Manual

This Manual provides general information about the system, its features, system configuration and standards. This manual provides an overview of the Electra Elite IPK II system and can be used to present information to potential customers.

Electra Elite IPK II System Hardware Manual

The System Hardware Manual is provided for the system installer. This manual has detailed instructions for installing the Electra Elite IPK II KSU, ETUs, Multiline Terminals, and optional equipment.

Electra Elite IPK II Programming Manual

This manual provides instructions for programming the Electra Elite IPK II system using a Multiline Terminal or PC.

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Introduction



SECTION 1 GENERAL INFORMATION

Electra Elite IPK II (DTH telephones), *D^{term}* Series i (DTR telephones), *D^{term}* IPK (ITH Telephones), Electra Elite (DTU telephones), and *D^{term}* Series E (DTP telephones) can be used with the Electra Elite IPK II system.

SECTION 2 MULTILINE TERMINALS USED WITH THE SYSTEM

Electra Elite IPK Terminals

The Electra Elite IPK II Terminals (DTH telephones) either with or without LCD display offer a variety of colors, and line sizes.

- Terminals are available in black or white.
- The large Liquid Crystal Display (LCD) on the display terminals provides call status data and programming information.
- Terminal line sizes include 8-line, 16-line, and 32-line.
- IP terminals are available in 4-line, 8-line and 16-line (with LCD).
- Speakerphone with full handsfree operation and headset jack is standard.
- All are compatible with the AD(A)-R, AP(A)-R, AP(R)-R, CT(A)-R Unit, CT(U)-R, or HF-R Unit adapter. The AP(R)-R Unit requires an AC-R Unit to supply AC power.
- The ADA-2R is compatible with ITH-2 IP terminals.
- The ADA-2R and PS(A)-R are compatible with ITH-3 IP terminals.
- An Attendant Add-On DCR-60-1 CONSOLE is available with 60 station and/or outside line assignments and 12 function keys.

Electra Elite IPK II Terminal Feature Access, Single On/Off, or One-Touch Keys

Keys are designated Feature Access, Single On/Off, or One-Touch throughout this manual. The keys operate much the same, but various limitations imposed on each type are described below.

Feature Access Keys

Depending on the type, a Multiline Terminal can have 2, 8, 16, or 24 line keys. These highly-flexible keys can be used for station DSS/BLF and Speed Dial.

Single On/Off Keys

Line keys may also be assigned as Single On/Off keys in System Programming to toggle a feature on/off. This assignment has no impact on the Feature Access keys, but the assigned features are very specific. Call Forward-All Call, Call Forward-Busy/No Answer, Scrolling (CID), headset, and DND are examples of features available for Single On/Off keys.

One-Touch Keys

One-Touch keys can perform the same function as Feature Access keys. A Multiline Terminal has a fixed number of these keys. No system assignment is necessary, and the number of keys ranges from none to 16 depending on the terminal type.

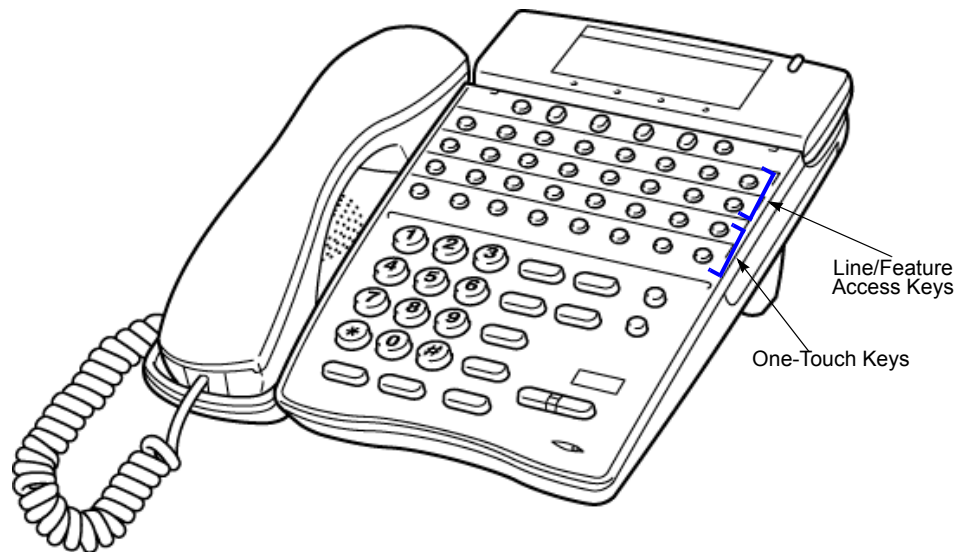


Figure 1-1 Key Assignment Example

***D*^{term} Series i Terminals**

The *D*^{term} Series i Terminals (DTR telephones) with or without LCD display offer a variety of colors and line sizes.

- Terminals are available in black or white.
- The large Liquid Crystal Display (LCD) on display terminals provides call status data and programming information.
- Line sizes include 8-line, 16-line, and 32-line.
- 2-line on the DTR-2DT-1.
- Speakerphone with full handsfree operation and headset jack is standard (except on the DTR-2DT-1).
- All but the DTR-2DT-1, DTR-1-1, DTR-1HM-1 and Cordless terminals are compatible with the AD(A)-R, AP(A)-R, AP(R)-R, CT(A)-R and HF-R Unit adapters. The AP(R)-R Unit requires an AC(A)-R Unit to supply AC power. For Attendant Positions, an Attendant Add-On DCR-60-1 CONSOLE is available with 60 station and/or outside line assignments and 12 function keys. The DTR-2DT-1 has an internal Analog Port without ringer.
- A two-line terminal with two Flexible Line keys (each with 2-color LED), nine function keys, built-in speakerphone, a large LED to indicate incoming calls or messages, and an outgoing only Analog SLT Port [AD(A)-R] is also available.
- The Electra Elite IPK II Single Line Terminals are offered in two variations (DTR-1-1 and DTR-1HM-1). Both terminals come in black or white. Both have DTMF and Pulse Dialing compatibility, and offer Flash and Redial key functionality. The Electra Elite IPK II Single Line Terminals come standard with a Message Waiting Indicator that also functions as an Incoming Call Indication. During a call, the receive audio level can be increased three levels and decreased two levels from the default setting (six volume level settings in all). The terminals offer four ring volume settings (Off, Soft, Medium, and Loud), and three ring patterns (Slow, Medium, and Fast). The DTR Single Line Terminals also have a Data Port that functions similar to that of an AP(R)-R optional adapter, and have a built-in wall mount adapter. The DTR-1HM-1 terminal has eight programmable speed dial buttons (maximum 21 digits each). The DTR-1HM-1 also has Hold and Monitor Function keys.

***D*^{term} IPK ITH Terminals**

The *D*^{term} IPK ITH Terminals come with LCD display and offer a variety of colors and line sizes:

- Terminals are available in black or white.
- The large Liquid Crystal Display (LCD) on display terminals provides call status data and programming information.

- Line sizes include 8-line and 16-line.
- Speakerphone with full-duplex handsfree operation and headset jack is standard.
- Terminals are compatible with AD(A)-2R and PS(A)-R.

Electra Elite and *D^{term}* Series E Terminals

The Electra Elite Terminals (DTU telephones) and *D^{term}* Series E terminals (DTP telephones) with or without LCD displays are available in a variety of colors and lines sizes.

- Terminals are available in black or white.
- The large Liquid Crystal Display (LCD) on the display terminals provides call status data and programming information.
- Line sizes include 8-line, 16-line, and 32-line.
- Speakerphone with full handsfree operation and headset jack is standard.
- The *D^{term}* Handset Cordless terminal is a 16-button telephone (display only).
- An Attendant Add-On DCU-60-1 CONSOLE is available for 60 station and/or outside line assignments and 12 function keys.
- A single line telephone Adapter can be used in place of a digital terminal for connecting Single Line Telephones, or similar devices.

Features



SECTION 1 ABOUT THIS CHAPTER

This chapter provides an alphabetical listing of the features that are available with the Electra Elite IPK II system.

Each feature provides the following information:

Description – briefly describes the feature and how it is use.

Conditions – provides special operating conditions (if any) that need to be considered with using the feature.

Default Settings – indicates the factory default setting (if any)

System Availability – describes multiline terminals that can be used with this feature and lists any additional equipment, such as adapters or ETUs, that must be installed for this feature to operate.

Programming – lists the memory blocks that support the feature.

Related Features – lists features that are associated with the feature being described (e.g., the Account Codes feature lists the Speed Dialing feature in the related features list because speed dialing bins can contain stored account code (if any).

Operation – provides step-by-step instructions for using the feature.

SECTION 2 **IMPORTANT NOTES**

Simplifying Multiline Terminal Operation with One-Touch Keys

A multiline terminal user can access many features Service Codes (e.g., Service Code 629 leaves a Message Waiting for a co-worker). To streamline the operation of their telephone, a multiline terminal user can store these codes under One-Touch Keys. This provides one-button operation for almost any feature. To find out more, turn to the One-Touch Calling and One-Touch Serial Operation features.

Programmable Keys

When reading an instruction using programmable keys, you will see a notation similar to (*PRG 15-07 or SC nnn*). This means that the key requires function code nnn, and you can program this code Program 15-07 or by dialing Service Code 751 or 752. Refer to the Programmable Function Keys feature for more information.

Using Handsfree

The manual assumes each extension has Automatic Handsfree. This lets a user just press a line key or Speaker key to answer or place a call. For extensions without Automatic Handsfree, the user must:

- Lift the handset or press **Speaker** for Intercom dial tone.
- Lift the handset or press **Speaker**, then press a line key for trunk dial tone.

Port Assignments

Port Calculation for Trunks:

The system detects the type of ETU (trunk or extension) and assigns the required extension or trunk port to the slot. The system uses the next available port number – it does not reserve any port.

SECTION 3 IPK TO IPK II FEATURE COMPARISON LIST

The following table provides a cross-reference between the Electra Elite IPK and the Electra Elite IPK II features.

IPK Feature Name	IPK II Feature Name
Account Code Entry	Account Code Entry
Account Code – Forced/Verified/Unverified	Account Code – Forced/Verified/Unverified
Add-On Conference	Conference
<i>Not Supported</i>	Alarm Reports
All Call Page	Paging, Internal
Alphanumeric Display	Alphanumeric Display
Analog Line Extender (<i>D^{term}</i> Analog EXT)	<i>Not Supported</i>
Ancillary Device Connection	Ancillary Device Connection
Answer Hold	Answer Hold
Answer Key	Answer Key
Assigned Night Answer (ANA)	Direct Inward Line (DIL)
Attendant Add-On Console	Direct Station Selection (DSS) Console
Attendant Camp-On	Call Waiting / Camp-On
Attendant Positions	Attendant Call Queuing
Attendant Station Outgoing Lockout	Code Restriction, Dial Block
Attendant Transfer	Transfer
Authorization Code	Code Restriction Override
Automated Attendant	Voice Response System (VRS)
Automatic Answer with Delay Message	Voice Response System (VRS)
Automatic Call Distribution	Automatic Call Distribution (ACD)
Automatic Callback	Callback
Automatic Day/Night Mode Switching	Night Service
Automatic Hold	Hold
Automatic Number Indication (ANI) on T1	T1 Trunking (with ANI/DNIS Compatibility)
Automatic Redial	Repeat Redial
Automatic Release	Automatic Release

IPK Feature Name	IPK II Feature Name
Automatic Route Selection (ARS)	Automatic Route Selection
Automatic Trunk-to-Trunk Transfer	Call Forwarding, Off-Premise
Background Music – Multiline Speaker	Background Music
Background Music Over External Speakers	Background Music
Barge-In	Barge-In
Battery Backup – System Memory	Battery Backup – System Memory
Battery Backup – System Power	Battery Backup – System Power
Busy Lamp Field on Multiline Terminals	One-Touch Calling
Call Alert Notification	Off-Hook Signaling
Call Appearance (CAP) Keys	Call Appearance (CAP) Keys
Call Arrival (CAR) Keys	Call Arrival (CAR) Keys
Call Forward – All Call	Call Forwarding
Call Forward – Busy/No Answer	Call Forwarding
Call Forward – Centrex	Call Forwarding – Centrex
Call Forward – Display	Call Forwarding
<i>Not Supported</i>	Call Forwarding/Do Not Disturb Override
Call Forward – Off-Premise	Call Forwarding, Off-Premise
<i>Not Supported</i>	Call Forwarding – Park and Page
<i>Not Supported</i>	Call Forwarding with Follow Me
Call Forward – Split	Call Forwarding
Call Monitoring	Call Monitoring
Call Park – System	Park
Call Pickup – Group	Group Call Pickup
Call Pickup Direct	Directed Call Pickup
<i>Not Supported</i>	Call Redirect
Callback Request	Message Waiting
Caller ID Call Return	Caller ID Call Return
Caller ID Indication (Analog Trunks)	Caller ID
Cascade CPU	<i>Not Supported</i>
Centralized Voice Mail (with E&M Tie Lines)	<i>Not Supported</i>

IPK Feature Name	IPK II Feature Name
<i>Not Supported</i>	Central Office Calls, Answering
<i>Not Supported</i>	Central Office Calls, Placing
Class of Service	Class of Service
Clock/Calendar Display	Clock/Calendar Display
CO Message Waiting Indication	CO Message Waiting Indication
CO/PBX, Tie Line Digit Restriction	Code Restriction
Code Restriction	Code Restriction
Computer Telephony Integration (CTI)	TAPI Compatibility
	Computer Telephony Integration (CTI) Applications
Consecutive Speed Dial	Speed Dial – System/Group/Station
<i>Not Supported</i>	Continued Dialing
Cordless Telephone Connection	Cordless Telephone Connection
Customized Message	Selectable Display Messaging
Data Line Security	Data Line Security
Delay Announcement	Voice Response System (VRS)
Delayed Ringing	Delayed Ringing
Dial 0 For Attendant	Operator
Dialed Number Indication Service (DNIS)	T1 Trunking (with ANI/DNIS Compatibility)
<i>Not Supported</i>	Dial Pad Confirmation Tone
<i>Not Supported</i>	Dial Tone Detection
Digit Insertion	Automatic Route Selection
Digital Line Extender (<i>D^{term}</i> ISDN EXTender Plus)	<i>Not Supported</i>
<i>Not Supported</i>	Digital Trunk Clocking
Digital Voice Mail	Digital Voice Mail
Direct Inward Dialing (DID)	Direct Inward Dialing (DID)
Direct Inward System Access (DISA)	Direct Inward System Access (DISA)
Direct Inward Termination (DIT)	Direct Inward Line (DIL)
Direct Paging Access	Paging, Internal
Direct Station Selection (DSS)	One-Touch Calling
Distinctive Ringing	Distinctive Ringing, Tones and Flash Patterns

IPK Feature Name	IPK II Feature Name
Do Not Disturb (DND)	Do Not Disturb
Door Lock Release Relays	Door Box
Door/Monitor Telephone	Door Box
DP to DTMF Switching	Pulse to Tone Conversion
Drop Key	Drop Key
<i>D^{term}</i> Analog Cordless Terminal	<i>Not Supported</i>
<i>D^{term}</i> Cordless II Terminal	<i>D^{term}</i> Cordless II Terminal
<i>D^{term}</i> Cordless Lite II Terminal	<i>D^{term}</i> Cordless Lite II Terminal
<i>D^{term}</i> Handset Cordless	<i>D^{term}</i> Handset Cordless
<i>D^{term}</i> IP Gateway System	<i>D^{term}</i> IP Gateway System
E&M Tie Lines (4-Wire)	Multiple Trunk Types
Elapsed Call Time	Call Duration Timer
Electra Elite IPK Terminals	Electra Elite IPK Terminals
Electra Elite Terminal Migration	Electra Elite Terminal Migration
Electra Professional Terminal Migration	<i>Not Supported</i>
Electronic Volume Control	Volume Controls
Elite ACD Plus	<i>Not Supported</i>
Elite CallAnalyst	Elite CallAnalyst
Elite Q-Master	<i>Not Supported</i>
EliteApps – Interactive Voice Response	EliteApps – Interactive Voice Response
EliteApps – PC Attendant	IPK II – PC Attendant
Emergency 911 – Cut	E911 Compatibility
Enhanced 911	E911 Compatibility
Equal Access Accommodation	Code Restriction
External Tone Ringer	Analog Communications Interface (ACI)
External Zone Paging (Meet-Me)	Paging, External
Facsimile CO Branch Connection	Facsimile CO Branch Connection
Feature Access – User Programmable	Programmable Function Keys
Flexible Line Assignment	Programmable Function Keys

IPK Feature Name	IPK II Feature Name
Flexible Numbering Plan	Flexible System Numbering
Flexible Ringing Assignment	Ring Groups
Flexible Timeouts	Flexible Timeouts
<i>Not Supported</i>	Forced Trunk Disconnect
Full Duplex Handsfree	Handsfree and Monitor
Full Handsfree Operation	Handsfree and Monitor
General Purpose Relays	Analog Communications Interface (ACI)
Ground Start Trunks	Multiple Trunk Types
Group Listening	Group Listen
Handset Mute	Handset Mute
Handsfree Answerback	Handsfree Answerback/Forced Intercom Ringing
Handsfree Dialing and Monitoring	Handsfree and Monitor
Headset Connection (Built-In)	Headset Operation
Hold With Recall (Exclusive and Non-Exclusive)	Hold
<i>Not Supported</i>	Hotel/Motel
Hot Key Pad	Hot Key-Pad
Hot Line	Hotline
Howler Tone Service	Howler Tone Service
I-Hold Indication	Distinctive Ringing, Tones and Flash Patterns
Incoming Call Identification	Caller ID
Incoming Trunk Name or Number Display	Name Storing
<i>Not Supported</i>	Intercom
Internal Hub	Internal Hub
Internal Voice/Tone Signaling	Handsfree Answerback/Forced Intercom Ringing
Internal Zone Paging (Meet-Me)	Paging, Internal
IP CPU & Media Gateway	<i>Not Supported</i>
<i>Not Supported</i>	IP Extenders/Mobile ConneX
IP Station (MEGACO)	IP Station (MEGACO) – IAD Integrated Access Device
<i>Not Supported</i>	IP Station (MEGACO) – MG 16
<i>Not Supported</i>	IP Station (SIP) – MG16

IPK Feature Name	IPK II Feature Name
<i>Not Supported</i>	IP Trunk – H.323 Protocol
<i>Not Supported</i>	IP Trunk – (SIP) Session Initiation Protocol
<i>Not Supported</i>	IP Trunk (SIP) – MG16
<i>Not Supported</i>	IPK II In-Mail
<i>Not Supported</i>	IPK II VoIP Management System
ISDN-BRI Trunk Connections	ISDN Compatibility
ISDN-PRI Trunk Connections	ISDN Compatibility
I-Use Indication	Distinctive Ringing, Tones and Flash Patterns
Key Function/Multifunction Registration	Multiple Trunk Types
<i>Not Supported</i>	K-CCIS – IP with IAD
<i>Not Supported</i>	K-CCIS – IP with PVA
Key-Common Channel Interoffice Signaling (K-CCIS)	K-CCIS – T1
Large LED Indication	Message Waiting
Last Number Redial	Last Number Redial
Least Cost Routing (LCR)	Automatic Route Selection
<i>Not Supported</i>	Licensing
Live Monitoring	Digital Voice Mail
<i>Not Supported</i>	Long Conversation Cutoff
Loop Start Trunks	Multiple Trunk Types
<i>Not Supported</i>	Meet Me Conference
<i>Not Supported</i>	Meet Me Paging
<i>Not Supported</i>	Meet Me Paging Transfer
Message Display Board	<i>Not Supported</i>
Message Waiting	Message Waiting
Microphone Control	Microphone Cutoff
Multiline Conference Bridge	Multiline Conference Bridge
Multilingual LCD Indication	Alphanumeric Display
Multimedia Conference Bridge	Multimedia Conference Bridge
Multiple Trunk Groups	Trunk Groups
Music on Hold	Music on Hold

IPK Feature Name	IPK II Feature Name
NEC Elite PC Assistant	IPK II – PC Assistant
Nesting Dial	<i>Not Supported</i>
Night Call Pickup	Night Service
Night Chime	Night Service
Night Transfer	Night Service
Off-Hook Ringing	Off-Hook Signaling
Off-Premise Extension	(OPX) Off-Premise Extension
<i>Not Supported</i>	One-Digit Dial Option
One-Touch Feature Access	Programmable Function Keys
<i>Not Supported</i>	PBX Compatibility
PC Programming	PC Programming
Pooled Line (Outgoing)	Trunk Group Routing
Power Failure Transfer	Power Failure Transfer
Preset Dialing	Dialing Number Preview
Prime Line Assignment	Prime Line Selection
Privacy on All Calls	Conference, Voice Call/Privacy Release
Privacy Release	Conference, Voice Call/Privacy Release
Private Lines	Private Line
Programming from Multiline Terminal	Programming from a Multiline Terminal
Pushbutton Dial – DTMF or DP	Single Line Telephones, Analog 500/2500 Sets
Quick Transfer to Voice Mail	Quick Transfer to Voice Mail
Recall Key	Flash
Recall With Station Identification	Transfer
Redial Key	Redial Key
Remote Programming	PC Programming
<i>Not Supported</i>	Remote (System) Upgrade
Resident System Program	Resident System Program
<i>Not Supported</i>	Reverse Voice Over
Restriction (Outgoing)	Code Restriction
Ring Tone Variation	Distinctive Ringing, Tones and Flash Patterns

IPK Feature Name	IPK II Feature Name
<i>Not Supported</i>	Ringdown Extension, Internal/External
Ringling Line Preference	Line Preference
<i>Not Supported</i>	Room Monitor
Route Advance Block	Trunk Group Routing
Save and Repeat	Save Number Dialed
Scrolling Directories	Directory Dialing
Secondary Incoming Extension	Secondary Incoming Extension
<i>Not Supported</i>	Secretary Call (Buzzer)
<i>Not Supported</i>	Secretary Call Pickup
Seized Trunk Name/Number Display	Name Storing
<i>Not Supported</i>	Selectable Ring Tones
<i>Not Supported</i>	Serial Call
Simplified Call Distribution	Department Calling
Single Line Telephone Access	Single Line Telephones, Analog 500/2500 Sets
SLT Adapter	SLT Adapter
SLT Timed Alarm	Alarm
<i>Not Supported</i>	SNMP Simple Network Management Protocol
Softkeys	Softkeys
Speed Dial – Station	Speed Dial – System/Group/Station
Speed Dial – System	Speed Dial – System/Group/Station
Speed Dial Stored Characters	Speed Dial – System/Group/Station
Station Add-On Console	Station Add-On Console
Station Camp-On	Call Waiting/Camp-On
Station Hunt	Station Hunt
Station Message Detail Recording (SMDR)	Station Message Detail Recording
Station Name Assignment User Programable	Station Name Assignment-User Programmable
Station Outgoing Lockout	Code Restriction, Dial Block
Station Relocation	Station Relocation
Station Transfer	Transfer
Step Call	Department Step Calling

IPK Feature Name	IPK II Feature Name
Store and Repeat	Memo Dial
Stored Hookflash	Speed Dial – System/Group/Station
Synchronous Ringing	Synchronous Ringing
System Data Up/Down Load	PC Programming
T1 Connection	T1 Trunking (with ANI/DNIS Compatibility)
<i>Not Supported</i>	Tandem Ringing
Tandem Switching of 4-Wire E&M Tie Lines	Multiple Trunk Types
Tenant Service	Night Service
Three-Minute Reminder	Warning Tone For Long Conversation
Tone Override	Tone Override
<i>Not Supported</i>	Traffic Reports
Trunk Queuing	Trunk Queuing/Camp-On
Trunk-to-Trunk Transfer	Tandem Trunking (Unsupervised Conference)
Two-Color LEDs	Distinctive Ringing, Tones and Flash Patterns
Unified Messaging	Unified Messaging
Unified Messaging – EliteMail CTI LX – Lite	Unified Messaging
Uniform Call Distribution (UCD)	Uniform Call Distribution (UCD)
Uniform Numbering Network	Uniform Numbering Network
Universal Slots	Universal Slots
Unsupervised Conference	Tandem Trunking (Unsupervised Conference)
User Programming Ability	User Programming Ability
<i>Not Supported</i>	Virtual Extensions
Voice Mail Integration (Analog)	Voice Mail Integration (Analog)
Voice Mail Message Indication on Line Keys	Voice Mail Message Indication on Line Keys
Voice Over Internet Protocol (VoIP)	Voice Over Internet Protocol (VoIP)
Voice Over Split	Voice Over
Voice Prompt	Voice Response System (VRS)
Wireless	<i>Not Supported</i>
Wireless – DECT	Wireless – DECT

SECTION 4 FEATURES

The remainder of this document provides the features for the Electra Elite IPK II system.

Account Code Entry

Description

Account Codes are user-dialed codes that help the system administrator categorize and/or restrict trunk calls. Optional Account Codes allow a user to enter an Account Code while placing a trunk call or anytime while on a call. The system does not require the user to enter the optional account code.

Account Codes for Incoming Calls

The system can control extension user ability to enter Account Codes for incoming calls. When this option is enabled, a user can dial * while on an incoming call, enter an Account Code, and then dial * to return to their caller. If the option is disabled, any digit the user dials after answering an incoming call outdials on the connected trunk.

Hiding Account Codes


Account Codes can be optionally hidden from a telephone display. This prevents, for example, an unauthorized co-worker from obtaining a Verified Account Code by watching the display and making note of the digits that dial out. When hidden, the Account Code digits show as * on the telephone display.

Account Code Capacity

Account Codes print along with the other call data on the SMDR record after the call completes. Account Codes can have 1~16 digits using 0~9 and #.

Redialed Numbers Do Not Contain Account Codes

When using the Last Number Redial, Save or Repeat Dial features, the system does not retain Account Code information. For any number redialed with these features, the user must enter an Account Code.

 *If a user enters *12345*203 926 5400*67890*, if the Last Number Redial feature is used, the system dials the number as 203 926 5400*67890*. The *67890* is not treated as an Account Code.*

A

Conditions

- If a user enters a code that exceeds 16 digits, the system ignores it.
- If the system has Account Codes disabled, the digits dialed (e.g., *1234*) appear on the SMDR report as part of the number dialed.
- Do not use * in a PBX access code when using Account Codes. Otherwise, after the *, the trunk stops sending digits to the central office.
- Account Codes appear on the SMDR report (even if they are hidden on the telephone display).
- To simplify Account Code Entry, store the Account Code (e.g., 1234) in a One-Touch Key, and press the key instead of dialing the code.
- Speed Dialing bins can contain stored Account Codes. Prevent them from being displayed using PRG 20-07-04.
- Automatic Route Selection (ARS) can force a user to enter an Authorization Code prior to using a certain route. The system verifies the ARS Authorization Code dialed against the ARS Authorization Code list (Program 44-03).

Default Settings

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Automatic Route Selection

One-Touch Calling

PBX Compatibility

Speed Dial – System/Group/Station

Station Message Detail Recording

Programming

- ➔ **14-01-11 Basic Trunk Data Setup – Account Code Required**
Enable (1) or disable (0) Account Codes for each trunk.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key as an Account Code key (code 50). Use this key instead of the dial pad to enter the * before and after the Account Code.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **21-01-05 System Options for Outgoing Calls – Dial Tone Detection Time**
Adjust the time the system waits for the Telco to return Dial Tone. By default, this option is set to 3 (Entries: 0~64800 seconds).
- ➔ **35-05-01 Account Code Setup – Account Code Mode**
For each Class of Service (1~15), select the Account Code Mode. The options are:
0 = Account Codes disabled. (Any codes you enter dial out as part of your initial call.)
1 = Account Codes optional (not required).
2 = Account Codes required (forced) but not verifiable.
3 = Account Codes required (forced) and verifiable.
- ➔ **35-05-03 Account Code Setup – Account Codes for Incoming Calls**
For each Class of Service (1~15), enter 1 in this option to enable Account Codes for incoming calls. Enter 0 to disable Account Codes for incoming calls. If disabled, any code you enter dialed out on the connected trunk.
- ➔ **35-05-04 Account Code Setup – Hiding Account Codes**
For each Class of Service (1~15), enter 1 to have the system hide Account Codes on an extension display as they are entered. Enter 0 to have the Account Codes displayed.

Operation

To enter an Account Code anytime while on a trunk call:


The outside caller cannot hear the Account Code digits you enter. Use this procedure if your system has Optional Account Codes enabled. You may also use this procedure for incoming calls. This procedure is not available for single line telephones.

1. Dial *.

- OR -

Press your Account Code key (PRG 15-07 or SC 751: code 50).

2. Dial your Account Code (1~16 digits, using 0~9 and #).

 *If Account Codes are hidden, each digit you dial shows an * on the telephone display.*

3. Dial *.

- OR -


Press your Account Code key (PRG 15-07 or SC 751: code 50).

To enter an Account Code before dialing the outside number:

If your system has Forced Account Codes, you must use this procedure. If it has Verified Account Codes, you can use this procedure instead of letting the system prompt you for your Account Code. You may also use this procedure if your system has Optional Account Codes.

If your system has Verified Account Codes enabled, be sure to choose a code programmed into your Verified Account Code list.

1. Access trunk for outside call.


 *Press a line key or dial a code (except 9) to access a trunk. Refer to [Central Office Calls, Placing on page 2-195](#) for more information.*

2. Dial *.

- OR -

Press your Account Code key (PRG 15-07 or SC 751: code 50).

3. Dial your Account Code (1~16 digits, using 0~9 and #).

 *If you make an incorrect entry, your system may automatically alert the operator. If Account Codes are hidden, each digit you dial shows an * on the telephone display.*

4. Dial *.

- OR -



Press your Account Code key (PRG 15-07 or SC 751: code 50).

5. Dial the number you want to call.

 *If you hear "stutter dial tone after dialing the number, ARS is requesting you to enter an Automatic Route Selection Authorization Code. Refer to [Automatic Route Selection on page 2-83](#) for more information.*



To enter an Account Code for an incoming call:

This procedure is not available for single line telephones.



1. Answer incoming call.
 -  *If Account Codes for Incoming Calls is disabled, the following steps dial digits out onto the connected trunk.*
2. Dial *.
3. Enter the Account Code.
 -  *You can enter any code of the proper length. Incoming Account Codes cannot be Forced or Verified.*
4. Dial *.

To enter an Account Code while placing a trunk call:

If your system has Forced Account Codes, you must follow this procedure.

1. Access trunk for outside call.
 -  *Press a line key or dial a code to access a trunk. Refer to [Central Office Calls, Placing on page 2-195](#) for more information.*
2. Dial *.
3. Dial your Account Code (1~16 digits, using 0~9 and #).
4. Dial *.
5. Dial number you want to call.
 -  *If you hear “stutter” dial tone after dialing the number, ARS is requesting you to enter an ARS Authorization Code. Refer to the [Automatic Route Selection on page 2-83](#) feature for more information on ARS Authorization Codes.*

To enter an Account Code at a single line telephone:

1. Access trunk for outside call.
 -  *You can access a trunk by dialing a code. Refer to [Central Office Calls, Placing](#) for more information.*
2. Dial *.
3. Enter Account Code (1~16 digits).
4. Dial *.
5. Dial number you want to call.
 -  *If you hear “stutter” dial tone after dialing the number, ARS is requesting you to enter an ARS Authorization Code. Refer to the [Automatic Route Selection on page 2-83](#) feature for more information on ARS Authorization Codes.*

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Account Code – Forced/Verified/Unverified

Description

Account Codes are user-dialed codes that help the system administrator categorize and/or restrict trunk calls. The system has two types of Forced Account Codes:

Forced Account Codes (Unverified)

Forced Account Codes **require** an extension user to enter an Account Code every time they place a trunk call. If the user does not enter the code, the system prevents the call. As with Account Codes, the extension user can elect to enter an Account Code for an incoming call. However, the system does not require it. **Forced Account Codes do not block emergency assistance (911) calls.**

Once set up in system programming, you can enable Forced Account Codes trunk-by-trunk. In addition, Forced Account Codes can apply to all outside calls or just long distance calls. Forced Account Codes for Toll Calls restricts calls according to the following chart:

Number of Digits Dialed	If first digit is not 1	If first digit is 1
1~3	Not allowed	Not allowed
4~7	Allowed – does not require Account Code	Allowed – requires Account Code
More than 7 ¹	Allowed – requires Account Code	Allowed – requires Account Code
800 and 888	Allowed – requires Account Code	Allowed – does not require Account Code
011 (International)	Allowed – requires Account Code	N/A
911	Allowed – does not require Account Code	N/A

¹ If you change the local call length in Toll Restriction, this value changes accordingly.

Verified Account Codes

With Verified Account Codes, the system compares the Account Code the user dials to a list of up to 2000 programmed codes. If the Account Code is in the list, the call goes through. If the code dialed is not in the list, the system prevents the call. Verified Account Codes can have 3~16 digits using the characters 0~9 and #. During programming, you can use “wild cards” to streamline entering codes into system memory. For example, the entry 123@ lets users dial Verified Account Codes from 1230 through 1239.

Operator Notification

To prevent Account Code abuse, the system can notify the operator each time an Account Code violation occurs (PRG: 20-13-20). This happens if the user fails to enter an Account Code (if Forced) or enters a Verified Account Code that is not in the list. The notification is an automatic Intercom call to the attendant and a *RESTRICT* message in the operator's display. (If the attendant fails to enter a valid Account Code, the system drops the call.)

Account Codes for Incoming Calls

The system can control extension user ability to enter Account Codes for incoming calls. When this option is enabled, a user can dial * while on an incoming call, enter an Account Code, and then dial * to return to their caller. If the option is disabled, any digit the user dials after answering an incoming call outdials on the connected trunk.

Hiding Account Codes


Account Codes can be optionally hidden from a telephone display. This prevents, for example, an unauthorized co-worker from obtaining a Verified Account Code by watching the display and making note of the digits that dial out. When hidden, the Account Code digits show as an * on the telephone display.

Account Code Capacity

Account Codes print along with the other call data on the SMDR record after the call completes. Account Codes can have 1~16 digits using 0~9 and #. Verified Account Codes can have from 3~16 digits.

Redialed Numbers Do Not Contain Account Codes

When using the Last Number Redial, Save or Repeat Dial features, the system does not retain Account Code information. If any number is redialed with these features, the user must enter an Account Code.

 *If a user enters *12345*203 926 5400*67890*, if the Last Number Redial feature is used, the system dials the number as 203 926 5400*67890*. The *67890* is not treated as an Account Code.*

Conditions

- If a user enters a code that exceeds 16 digits, the system ignores it.
- If the system has Account Codes disabled, the digits dialed (e.g., *1234*) appear on the SMDR report as part of the number dialed.
- If using Forced Account Code with single line telephone you need a VRS to get the prompts to enter the Forced Account Code.

- When you use Forced Account Code on only toll calls, and you dial a local call, you hear a beep.
- The Timer set in 21-01-14 is applied to toll calls and local calls.
- Speed Dial – System/Group/Station bins can contain stored Account Codes. They can be prevented from being displayed using 20-07-04.
- Automatic Route Selection (ARS) can force a user to enter an Authorization Code prior to using a certain route. The system verifies the ARS Authorization Code dialed against the ARS Authorization Code list (Program 44-03).
- To simplify Account Code Entry, store the Account Code (e.g., *1234*) in a One-Touch Key, and press the key instead of dialing the code.
- Account Codes appear on the SMDR report (even if they are hidden on the telephone display).
- When using a PBX/CTX trunk and Account Codes, do not use * as part of the PBX/CTX access code. Otherwise, after the *, the trunk stops sending digits to the central office.
- Account Codes for incoming calls not available for single line telephones.

Default Settings

Account Codes are disabled.

System Availability

Terminals

Any Station

Required Component(s)

VRS for Forced Account Codes for Single Line Telephones

Related Features

Automatic Route Selection

PBX Compatibility

Speed Dial – System/Group/Station

Station Message Detail Recording

Programming

- ➔ **14-01-11 Basic Trunk Data Setup – Account Code Required**
Enable (1) or disable (0) Forced Account Codes for each trunk.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key as an Account Code key (code 50). Use this key instead of the dial pad to enter the * before and after the Account Code.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-20 Class of Service Options (Supplementary Service) – Account Code/Toll Restriction Operator Alert (Restricted Operation Transfer)**
Turn Off (0) or On (1) the Operator Alert when a forced account code is incorrectly entered.
- ➔ **21-04-01 Toll Restriction Class for Extensions**
Assign a Toll Restriction Class (1~15) to an extension.
- ➔ **35-05-01 Account Code Setup – Account Code Mode**
For each Class of Service (1~15), select the Account Code Mode. The options are:
0 = Account Codes disabled. (Any codes you enter dial out as part of your initial call.)
1 = Account Codes optional (not required).
2 = Account Codes required (forced) but not verifiable.
3 = Account Codes required (forced) and verifiable.
- ➔ **35-05-02 Account Code Setup – Forced Account Code Toll Call Setup**
For each Class of Service (1~15), enter 1 to enable Forced Account Codes just for toll calls. With this option enabled, local calls have Optional Account Codes. Enter 0 to enable Forced Account Codes for local and toll calls. If a user needs to dial * as part of post-dialing digits, the * must be pressed three times (twice for the expected account code entry and the third as the post-dialing digit).
- ➔ **35-05-03 Account Code Setup – Account Codes for Incoming Calls**
For each Class of Service (1~15), enter 1 to enable Account Codes for incoming calls. Enter 0 to disable Account Codes for incoming calls. If disabled, any codes you enter dial out on the connected trunk.
- ➔ **35-05-04 Account Code Setup – Hiding Account Codes**
For each Class of Service (1~15), enter 1 to have the system hide Account Codes on an extension display as they are entered. Enter 0 to have the Account Codes displayed.
- ➔ **35-06-01 Verified Account Code Table – Verified Account Code**
Enter data in the Verified Account Code Table. You can enter up to 2000 codes with 3~16 digits. For a wild card @, press the LK 1.
- ➔ **40-10-01 Voice Announcement Service Option – VRS Fixed Message**
Use (1) or Don't Use (0) the Fixed VRS messages.

Operation

To enter an Account Code anytime while on a trunk call:


The outside caller cannot hear the Account Code digits you enter. You can use this procedure if your system has Optional Account Codes enabled. You may also use this procedure for incoming calls. This procedure is not available for single line telephones.

1. Dial *.

- OR -

Press your Account Code key (PRG 15-07-01 or SC 751: code 50).

2. Dial your Account Code (1~16 digits, using 0~9 and #).

 *If Account Codes are hidden, each digit you dial shows an * on the telephone display.*

3. Dial *.

- OR -


Press your Account Code key (PRG 15-07-01 or SC 751: code 50).

To enter a Forced Account Code before dialing the outside number:

If your system has Forced Account Codes, you must use this procedure. If it has Verified Account Codes, you can use this procedure instead of letting the system prompt you for your Account Code. You may also use this procedure if your system has Optional Account Codes.

If your system has Verified Account Codes enabled, be sure to choose a code programmed into your Verified Account Code list.

1. Access trunk for outside call.


 *Press a line key or dial a code to access a trunk. Refer to [Central Office Calls, Placing on page 2-195](#) for more information.*

2. Dial *.

- OR -

Press your Account Code key (PRG 15-07-01 or SC 751: code 50).

3. Dial your Account Code [1~16 digits, using 0~9 and # or (3~16 digits for Forced)].


 *If you make an incorrect entry, your system may automatically alert the operator. If Account Codes are hidden, each digit you dial shows * on the telephone display (depending on programming).*

4. Dial *.

- OR -


Press your Account Code key (PRG 15-07-01 or SC 751: code 50).

5. Dial the number you want to call.

 *If you hear “stutter” dial tone after dialing the number, ARS is requesting you to enter an Automatic Route Selection Authorization Code. Refer to [Automatic Route Selection on page 2-83](#) for more information.*

To dial an outside number and let your system tell you when a Forced Account Code is required:

1. Access a trunk and dial the number you want to call.

 *If you hear “stutter dial tone after dialing the number, ARS is requesting you to enter an ARS Authorization Code. Refer to [Automatic Route Selection on page 2-83](#) for more information.*

2. Wait for your call to go through.

- OR -

3. If you hear “Please enter an Account Code,” (depending on system programming) and your display shows *ENTER ACCOUNT CODE*.

Dial *.

- OR -

Press your Account Code key (PRG 15-07-01 or SC 751: code 50).

Dial your Account Code (3~16 digits, using 0~9 and #).

If Account Codes are hidden, each digit you dial shows * on the telephone display.

Dial *.


- OR -

Press your Account Code key (PRG 15-07-01 or SC 751: code 50).

To enter an Account Code for an incoming call:


This procedure is not available for single line telephones.

1. Answer incoming call.

 *If Account Codes for Incoming Calls is disabled, the following steps dial digits out onto the connected trunk.*




2. Dial *.

3. Enter the Account Code (1~16 digits).

 *You can enter any code of the proper length.*

4. Dial *.

To enter a Forced Account Code at a single line telephone:

1. Access trunk for outside call.
 -  *You can access a trunk by dialing a code. Refer to [Central Office Calls, Placing](#) for more information.*
 -  *With Forced Account Codes, you hear, “Please enter an Account Code.” (depending on programming).*
2. Dial *.
3. Enter Account Code (3~16 digits).
4. Dial *.
5. Dial number you want to call.
 -  *If you hear “stutter” dial tone after dialing the number, ARS is requesting you to enter an ARS Authorization Code. Refer to the [Automatic Route Selection on page 2-83](#) feature for more information on ARS Authorization Codes.*

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Alarm

Description

Alarm lets any station extension work like an Alarm clock. An extension user can have Alarm remind them of a meeting or an appointment. There are two types of Alarms:

- Alarm 1 (sounds only once at the preset time)
- Alarm 2 (sounds every day at the preset time)

Conditions

- Single line telephones ring and Music on Hold is heard when the alarm sounds.
- Only a Multiline Terminal can view what time the alarm is currently set for.

Default Settings

Alarm is enabled.

System Availability

Terminals

Any Station

Required Component(s)

None

Related Features



None

Programming


- ➔ **20-01-06 System Options – Alarm Duration**
Set the duration of the Alarm signal (0~64800 seconds).

Operation


To set the alarm:

1. At the multiline terminal, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **727**.
3. Dial alarm type (**1** or **2**).
 *Alarm 1 sounds only once. Alarm 2 sounds each day at the preset time.*
4. Dial the alarm time (24-hour clock).
 *For example, for 1:15 PM dial 1315.
A confirmation tone is heard if the alarm has been set. If the alarm was not set, an error tone is heard instead.*
5. At the multiline terminal, press **Speaker** to hang up.
- OR -
At the single line telephone, hang up.

To silence an alarm:

1. At multiline terminal, press **Exit**.
- OR -
At the single line telephone, lift the handset.
 *The single line set user hears Music on Hold when the handset is lifted.*

To check the programmed alarm time at a multiline terminal:

1. Press **Help**.
2. Dial **727**.
3. Dial alarm type (**1** or **2**).
 *The programmed time displays.*
4. Press **Exit**.

To cancel an alarm:

1. At the multiline terminal, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **727**.
3. Dial alarm type (**1** or **2**).
4. Dial **9999**.
5. At a multiline terminal, press **Speaker** to hang up.
- OR -
At the single line telephone, hang up.

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Alarm Reports

Enhancements

Version 1500 or higher is required to support this feature.
--

Description

The IPK II system logs various errors and reports information about the operation that can be used to determine the cause of a problem. The system can indicate several errors on the multiline telephone display, send the information to a printer immediately, or send data at a programmed time. The report data can also be sent via e-mail.

Alarm Report

The Alarm Reports indicate:

- System start-up/upgrade date and time
- ETU communication error with date and time and the restoration date and time
- Date and time an ETU was removed from the system
- Date and time an extension was disconnected from the system
- Date and time of any system data change

Table 2-1 Sample Alarm Report

<< Alarm Report

05/16/2006 14:30 PAGE 001

LVL	NO	STAT	DATE	TIME	ITEM	UNIT	SLT	PRT	PARAMETER
MIN	0002	REC	05/16/06	14:21	PKG Installation	PRT	02	00	
MAJ	0010	ERR	05/16/06	14:21	ISDN Link	PRT	02	12	
MAJ	0010	REC	05/16/06	14:21	ISDN Link	PRT	02	12	
MIN	0002	ERR	05/16/06	14:33	PKG Installation	PRT	02	00	
MIN	0002	ERR	05/16/06	14:33	PKG Installation	ESI	05	00	
MIN	0002	ERR	05/16/06	14:33	PKG Installation	SLIB	07	00	
MAJ	0050	WAR	05/16/06	14:33	System Start Up	none	00	00	
MIN	0002	REC	05/16/06	14:33	PKG Installation	PRT	02	00	
MAJ	0014	ERR	05/16/06	14:33	NTCPU-LAN Link	none	00	00	
MAJ	0014	REC	05/16/06	14:35	NTCPU-LAN Link	none	00	00	
MIN	0002	ERR	05/16/06	14:36	PKG Installation	CTP	08	00	
MIN	0002	REC	05/16/06	14:37	PKG Installation	VMS	08	00	
MIN	0002	ERR	05/16/06	14:38	PKG Installation	VMS	08	00	
MIN	0002	REC	05/16/06	14:40	PKG Installation	PRT	07	00	
MIN	0002	ERR	05/16/06	14:40	PKG Installation	PRT	07	00	
MAJ	0006	ERR	05/16/06	14:41	Blocking	ESIB	01	05	
MAJ	0006	REC	05/16/06	15:01	Blocking	ESIB	01	05	
MAJ	0006	ERR	05/16/06	15:05	Blocking	ESIB	01	07	
MAJ	0006	REC	05/16/06	15:07	Blocking	ESIB	01	07	

Table 2-2 Alarm Report Definitions

Alarm Report Heading	Definitions
LVL	Alarm Type (MAJ = Major, MIN = Minor)
NO	Number of Alarm (4-digit)
STAT	Status (REC = Recovered, ERR = Error, WAR = Warning)
DATE	Date the Alarm Occurred
TIME	Time the Alarm Occurred
ITEM	Name of the Alarm
UNIT	Name of the ETU
SLT	KSU Slot Number
PRT	KSU Port Number
PARAMETER	Related Information

Table 2-3 Alarm Report Item Definitions

Item Name	Definition
PKG Installation	ETU is removed or inserted.
ISDN Link	ISDN Line failure is detected.
CPU – LAN Link	CPU – Lan connection failure is detected.
Blocking	Terminal Failure may have occurred because terminal blocking is detected. Terminal is unplugged or wire is disconnected.
System Data Change	System Upgrade performed or Programming change.
System Start Up	System is reset.
SMDR Link	Connection failure has been detected between the CPU and SMDR printer device.

System Information

The system can print a report of the boards installed, the port assignments, and the port types. This information is sent to the extension defined in Program 90-13.

The System Information Reports indicate:

- Date and Time of the Report
- ETU names
- Slot condition (working, blocked)
- Port assignment
- Port classification

Table 2-4 Sample System Information Printout

System Information				05/18/2006 11:02	
slot	location	type	assign port	condition	note
1	1-1	ESIB	1-16	Running	***** ----- Connect: *
2	1-2	PRT	1-23	Running	
3	1-3	COID	25-28	Running	
4	1-4	none	none	Not Install	
5	1-5	ESI	33-40	Not Install	----- Connect: *
6	1-6	SLI	17-24	Running	
7	1-7	PRT	29-51	Not Install	
8	1-8	CTP	25-32	Running	
9	2-1	none	Not Install		
10	2-2	none	Not Install		
11	2-3	none	Not Install		
12	2-4	none	Not Install		
13	2-5	none	Not Install		
14	2-6	none	Not Install		

Table 2-4 Sample System Information Printout (Continued)

System Information				05/18/2006 11:02	
slot	location	type	assign port	condition	note
15	2-7	none	Not Install		
16	2-8	none	Not Install		
17	3-1	none	Not Install		
18	3-2	none	Not Install		
19	3-3	none	Not Install		
20	3-4	none	Not Install		
21	3-5	none	Not Install		
22	3-6	none	Not Install		
23	3-7	none	Not Install		
24	3-8	none	Not Install		

Conditions

- Alarm Reports and System Information Reports can be output to KSU COM Port 1 or a CTU/ CTA adapter.
- The IPK II supports the following Alarms to be output to the LCD of a multiline terminal:
 - SMDR Buffer Full
 - CPU-LAN link Error
- The IPK II does not support printouts of the following Alarms:
 - Power Failure
 - RAM Backup Battery Error
 - Networking Keep Alive Error
- Program 15-02-19 must be set to Non-Procedural Mode when using CTA/CTU adapters for printouts.
- Up to 12 System Alarm times can be scheduled to print on a Monthly, Daily, and Hourly time frame. The report indicates both Major and Minor Alarms.
- System Information Reports cannot be set to output at a scheduled time.

- When using the E-mail functionality of reports, the E-mail address in Program 90-11-10 (From Address) must be set for the E-mail feature to work.
- Output of DIM (debug) and Alarm Reports at the same time is not supported.
- Once output of new alarms is performed, these same alarms are not output a second time. New alarms must be generated before Program 90-12-04 can be performed a second time.
- Up to 100 System Alarm Reports can be stored. When the buffer fills, the oldest record is deleted to allow the new record to be saved.
- A CTU adapter USB connection should be safely stopped in Windows before the following procedures are performed:
 - CPUII Upgrade
 - ESIB/E Reset
 - CPUII Reset
- If the System is set up to E-mail the Alarm Reports and the Mail Server is down, the report is not sent.
- System Information Reports cannot be set for E-mail output.
- Scheduled Alarm Reports via E-mail prints all alarms. When the system detects New alarms, this information is output via E-mail individually.
- E-mail Alarm Reports can be sent whenever each New alarm occurs per event. If you want to receive complete Alarm Reports periodically, you must specify 12 individual dates and times in PRG 90-24-01 ~ PRG 90-24-04 per period.

Default Settings

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

CTA/CTU and available USB port

- OR -

Available RS-232 Serial Communications Port

Related Features

None

Programming

- ➔ **15-02-19 Multiline Telephone Basic Data Setup – CTA Data Communications Setup**
This Program must be set to Non-Procedural Mode (1) when using CTA/CTU adapters for printouts.
0 = CTI Mode
1 = Non-Procedural Mode (Non-SCS)
- ➔ **15-02-20 Multiline Telephone Basic Data Setup – Baud Rate for CTA Port**
0 = 4800bps
1 = 9600bps
2 = 19200bps (default)
- ➔ **90-10-01 System Alarm Setup – Alarm Type**
0 = Not Set (default)
1 = Major Alarm
2 = Minor Alarm
- ➔ **90-10-02 System Alarm Setup – Report**
0 = No Report (no autodial) (default)
1 = Report (autodial)
- ➔ **90-11-01 System Alarm Report – System Alarm Display Telephone**
Assign the display multiline extension number that should receive system alarms.
Extension Number (up to eight digits).
No default is assigned
- ➔ **90-12-05 System Alarm Output – Clear All Alarm Reports**
Clear all alarms.
All Clear?
0 = No
1 = Yes

-
-
- ➔ **90-12-06 System Alarm Output – Output Mode**
0 = Manual (default)
1 = Auto
 - ➔ **90-24-01 System Alarm Report Notification Time Setup – Month**
Set the month (00~12)
(0 = Disabled)
 - ➔ **90-24-02 System Alarm Report Notification Time Setup – Day**
Set the day (00~31)
(0 = Disabled)
 - ➔ **90-24-03 System Alarm Report Notification Time Setup – Hour**
Set the hour (00~23)
(0 = Disabled)
 - ➔ **90-24-04 System Alarm Report Notification Time Setup – Minute**
Set the minute (00~59)
(0 = Disabled)

For Printing Reports

- ➔ **90-12-01 System Alarm Output – Output Port Type**
Indicate the connection used for the System Alarms. The baud rate for the COM port should be set in Program 10-21-02.
0 = No Setting (default)
1 = COM Port (CPU1)
2 = -- Not Used --
3 = -- Reserve --
4 = CTA/CTU
- ➔ **90-12-02 System Alarm Output – Destination Extension Number**
If the output port type (PRG 90-12-01) is set to CTA, enter the extension number with the CTA connection.
Extension Number (up to eight digits)
No default is assigned
- ➔ **90-12-03 System Alarm Output – Output All Alarm Reports**
0 = Disable (Off) (default)
1 = Enable (On)
- ➔ **90-12-04 System Alarm Output – Printout New Alarm Reports**
Print New?
0 = No
1 = Yes

For Printing System Information Reports

- ➔ **90-13-01 System Information Output – Output Port Type**
Indicate the type of connection system information.
0 = No Setting (default)
1 = COM Port (CPUII)
2 = -- Not Used --
3 = LAN port (CPUII)
4 = CTA/CTU

- ➔ **90-13-02 System Information Output – Destination Extension Number**
If the output port type (PRG 90-13-01) is set to CTA, enter the extension number with the CTA connection.
Extension Number (up to eight digits)
No default is assigned

- ➔ **90-13-03 System Information Output – Output Command**
Dial 1 from this program to send the system report to the connected device.
0 = Disable (Off) (default)
1 = Enable (On)

For E-mailing Alarm Reports

- ➔ **10-12-01 CPUII Network Setup – IP Address**
1.0.0.1 ~ 126.255.255.254
128.1.0.1 ~ 191.254.255.254
192.0.1.1 ~ 223.255.254.254
Default = 172.16.0.10

- ➔ **10-12-02 CPUII Network Setup – Subnet Mask**
128.0.0.0
240.0.0.0
254.0.0.0
255.192.0.0
255.252.0.0
255.255.128.0
255.255.248.0
255.255.255.0
255.255.255.224
255.255.255.252
192.0.0.0
248.0.0.0
255.0.0.0
255.224.0.0
255.254.0.0
255.255.192.0
255.255.252.0
255.255.255.128
255.255.255.240
255.255.255.254
224.0.0.0
252.0.0.0

255.128.0.0
 255.248.0.0
 255.255.0.0 (Default)
 255.255.224.0
 255.255.254.0
 255.255.255.192
 255.255.255.248
 255.255.255.255

The setting of Subnet Mask is invalid when all Host Addresses are 0. If the network section is:
 0,

127
 128.0
 191.255
 192.0.0
 223.255.255

The setting of Subnet Mask is invalid.

➔ **10-12-03 CPUUI Network Setup – Default Gateway**

IP Address for Router
 1.0.0.1 ~ 126.255.255.254
 128.1.0.1 ~ 191.254.255.254
 192.0.1.1 ~ 223.255.254.254
 Default = 0.0.0.0

➔ **90-11-02 System Alarm Report – Report Method**

When Alarm Reports are to be e-mailed, set this option to 1. This program has higher priority than PRG 90-12-06

0 = No Report (default)
 1 = E-mail Address

➔ **90-11-06 System Alarm Report – SMTP Host Name**

When Alarm Reports are to be e-mailed, set the SMTP name (ex., smtp.yourisp.com). Contact your ISP (Internet Service Provider) for the correct entry if needed.

Up to 255 characters
 Default = No Setting

➔ **90-11-07 System Alarm Report – SMTP Host Port Number**

When Alarm Reports are to be e-mailed, set the SMTP host port number. Contact your ISP (internet service provider) for the correct entry if needed.

0 ~ 65535
 Default = 25

- ➔ **90-11-08 System Alarm Report – To Email Address**
When Alarm Reports are to be e-mailed, set this e-mail address to which the report should be sent.
Up to 255 characters
Default = No Setting
- ➔ **90-11-09 System Alarm Report – Reply Address**
When Alarm Reports are to be e-mailed, set the e-mail address where replies should be e-mailed.
Up to 255 characters
Default = No Setting
- ➔ **90-11-10 System Alarm Report – From Address**
When Alarm Reports are to be e-mailed, set this e-mail address for the station sending the report.
Up to 255 characters
Default = No Setting
- ➔ **90-11-11 System Alarm Report – DNS Primary Address**
When Alarm Reports are to be e-mailed, set the DNS primary address.
0.0.0.0 ~ 255.255.255.255
Default = 0.0.0.0
- ➔ **90-11-12 System Alarm Report – DNS Secondary Address**
When Alarm Reports are to be e-mailed, set the DNS secondary address.
0.0.0.0 ~ 255.255.255.255
Default = 0.0.0.0
- ➔ **90-11-13 System Alarm Report – Customer Name**
When Alarm Reports are to be e-mailed, enter a name to identify the particular system.
Up to 255 characters
Default = No Setting
- ➔ **90-25-01 System Alarm Report CC Mail Setup – CC Mail Address**
Up to 255 characters
Default = No Setting

Operation

To use this feature at any terminal:

The user must be logged in with an Installer (IN) level password defined in Program 90-02.

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Alphanumeric Display

Description

Multibutton display telephones have a 3-line, 24 character per line Alphanumeric Display that provides various feature status messages. These messages help the display telephone user process calls, identify callers and customize features.

The contrast is not adjustable when the telephone has background music enabled.

Conditions

None

Default Settings

Enabled for all display telephones.

System Availability

Terminals

All Display Multiline Terminals.

Required Component(s)

None

Related Features

Clock/Calendar Display

Selectable Display Messaging

Programming

- ➔ **11-11-13 Service Code Setup (for Setup/Entry Operation) – Display Language Selection for Multiline Terminal**
If needed, redefine the service code used to select the language for display multiline terminals (default: 678).

- ➔ **15-02-01 Multiline Telephone Basic Data Setup – Display Language Selection**
Select the language to be displayed on a multiline terminal display.

0 = Japanese	5 = Spanish	10 = Swedish
1 = English	6 = Dutch	11 = Turkish
2 = German	7 = Portuguese	12 = Latin American Spanish
3 = French	8 = Norwegian	
4 = Italian	9 = Danish	

- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

- ➔ **20-11-08 Class of Service Options (Hold/Transfer Service) – Transfer Information Display**
Enable (1)/Disable (0) an extension incoming transfer preanswer display.

Operation

Operation is automatic if enabled in programming.

Analog Communications Interface (ACI)

Description

The Analog Communications Interface (ACI) feature uses a PGD(2)-U10 ADP (Door Phone/Paging) adapter to provide two analog ports (with associated relays) for Music on Hold, External Paging, Door Boxes and auxiliary devices such as tape recorders and loud bells. The system allows up to 48 PGD(2)-U10 ADPs (when used for ACI ports) for a maximum of 96 analog ports. Each PGD(2)-U10 ADP requires an unused port on an ESIB/E-U20.

Music on Hold

You can connect up to two customer-provided Music on Hold music sources to a PGD(2)-U10 ADP. This lets you add additional music sources if the external source on the CPU II ETU or the internal source is not adequate. By using PGD(2)-U10 ADPs, you could even have a different music source for each trunk.

When the system switches the ACI analog port to a trunk on Hold, the PGD(2)-U10 ADP relay associated with the ACI analog port closes. Use this ability to switch on the music source, if desired.

Extension users can dial the ACI analog port extension number and listen to the connected music source. The PGD(2)-U10 ADP relay associated with the port closes when the call goes through.

For Music on Hold, connect the music source to the PGD(2)-U10 ADP module. Connect the music source control leads to the CTL (control relay) jack. Refer to the Electra Elite IPK II System Hardware Manual for additional details.

External Paging

An ACI analog port can also be an External Page output. When connected to customer-provided External Paging equipment, the ACI port provides External Paging. To use the External Paging, an extension user just dials the ACI analog port extension number and makes the announcement. The system broadcasts the announcement from the ACI analog port and simultaneously closes the associated PGD(2)-U10 ADP relay. You can use the relay closure to control the External Paging amplifier, if required. This external paging zone is not included in external all call paging or combination paging (internal and external).

For External Paging, connect the Paging amplifier to the PGD(2)-U10 ADP jack. Connect the amplifier control leads to the CTL (control relay) jack. Refer to the Electra Elite IPK II System Hardware Manual for additional details.

Auxiliary Device Control

The PGD(2)-U10 ADP can control a customer-provided tape recorder. When an extension user dials the ACI analog port extension number, they can automatically start the recorder and activate the record function. When the user hangs up, the recording stops and the tape recorder turns off. For tape recording, connect the tape recorder AUX input jack to the PGD(2)-U10 ADP jack. Connect the recorder control leads (if available) to the CTL (control relay) jack. Refer to the System Hardware Manual for additional details.

By using Department Calling, you can arrange multiple tape recorders into a pool. When an extension user dials the Department Group pilot number, they reach the first available tape recorder in the pool.

The relays in the PGD(2)-U10 ADP can optionally control customer-provided external ringers (loud bells) and buzzers. When an extension user dials the ACI analog port extension number, the associated PGD(2)-U10 ADP relay closes and activates the ringer. You could use this ability to control an emergency buzzer for a noisy machine shop floor, for example.

ACI Call Recording

ACI Call Recording allows you to use a recording device connected to a PGD(2)-U10 ADP to manually or automatically record calls. The recording device is typically a customer-provided tape recorder. You can set up ACI Call Recording to output to a single ACI port/recording device or to a pool of ACI ports/devices. With a single device, all calls are stored in a centralized location. With a pool of devices, be sure to have a port available for recording – even in peak traffic periods. You can set up recording per trunk or per extension.

When set up for manual recording, the user can press the ACI Conversation Record key (Service Code 751 + 69 + 0) to begin recording the call from that point. When set up for automatic recording, ACI Call Recording starts automatically as soon as the user places or answers their call. The system can be programmed to record all incoming trunk calls which ring an extension. This includes the following trunks:

- Central Office calls programmed to ring the extension
- Direct Inward Dialing (DID)
- Direct Inward Line (DIL)
- Direct Inward System Access (DISA)
- Tie lines

The system can also be programmed to record *outgoing* trunk calls, however, this is possible only using E&M Tie Lines, PRI or BRI trunks.

ACI Call Recording is not available for intercom calls, transferred calls, or calls placed on hold and answered by an extension with Call Recording enabled. To manually record any call (transferred, ICM, outgoing CO trunk, etc.), use the Voice Mail Conversation Record key (Service Code 751 + 78).

Physical Ports and Software Ports

Each PGD(2)-U10 ADP consists of a physical port for connection to the telephone system and two logical ports. For programming, the ports are also called software ports. The physical port connects to a station position on a ESI ETU. During installation, the first PGD(2)-U10 ADP you set up is physical port 1; the second PGD(2)-U10 ADP is physical port 2, etc. Each PGD(2)-U10 ADP has two software ports, which are numbered independently of the physical ports. Normally, the first PGD(2)-U10 ADP set up has software ports 1~2; the second PGD(2)-U10 ADP has software ports 3~4, etc. There are a total of 96 software ports (48 PGD(2)-U10 ADPs x 2 ports each). During programming, you assign ACI extension numbers and Department Group options to PGD(2)-U10 ADP software ports, not physical ports. During installation, you connect equipment to the jacks on the PGD(2)-U10 ADP that correspond to the software port. Refer to the Electra Elite IPK II System Hardware Manual for installation details.

Conditions

- ACD agents who are logged on can be recorded.
- ACI software ports cannot be Background Music music sources.
- An extension cannot have Hotline keys for ACI software ports. Music on Hold ACI software ports can be Music on Hold music sources.
- An extension can have One-Touch Keys for ACI software ports. The gives the extension user:
 - One-Touch access to external music
 - One-Touch External Paging
 - One-Touch loud ringer control
- ACI software ports can provide External Paging with control, independent of the External Paging circuits on the CPU II ETU. The PGD(2)-U10 ADP can be connected only to an ESIB(8)-U20 ETU. If the PGD(2)-U10 ADP is connected to an ESIE(8)-U20 ETU, the PGD(2)-U10 ADP is NOT recognized.
- When an ESIB(8)-U20 ETU has an ESIE(8)-U20 ETU installed [ESIB/E(16)], the PGD(2)-U10 ADP has both channels available for Paging, MOH, External Ringers, and Door Box according to strapping on ports 1~8. The corresponding port on the ESIE(8)-U20 ETU does not support a station or other hardware (it is used by the PGD(2)-U10 ADP for channel 2).
- When an ESIB(8)-U20 ETU is installed without an ESIE(8)-U20 ETU installed, a PGD(2)-U10 ADP installed on ports 1~8 is recognized, but only Channel 1 on the PGD(2)-U10 ADP is supported. Channel 2 is not available.
- ESIE(8)-U10 ETUs and ESIE(8)-U10 ETUs do not support the PGD(2)-U10 ADP.

The devices connected to the PGD(2)-U10 ADP must be compatible with the specifications below. Refer to the Electra Elite IPK II System Hardware Manual for installation details.

PGD(2)-U10 ADP/ACI Interface Specifications	
Relay Contacts	
Maximum Contact Ratings	30 V DC @ 60 mA
	90 V AC @ 10 mA
Minimum Application Load	1 V DC @ 1 mA
Audio/Music Input	
Input Impedance	47 K Ohms @ 1 K Hz
Maximum Input	0.4Vrms or 1.0Vp-p.
Audio/Paging Output	
Output Impedance	600 Ohms @ 1 K Hz
Maximum Output	+ 3 dBm

Default Settings

No PGD(2)-U10 ADPs programmed.

System Availability

Terminals

None

Required Component(s)

PGD(2)-U10 ADP

Related Features

Automatic Call Distribution (ACD)

Background Music

Hotline

One-Touch Calling

Paging, External

Programming

- **11-06-01 ACI Extension Numbering**
Assign extension numbers to ACI software ports. Select a number outside of the normal extension number range

ACI Ports: 1~96
- **11-08-01 ACI Group Pilot Number**
Assign pilot numbers to ACI groups. When a user dials the pilot number, they reach an available ACI software port within the group.

ACI Groups 1~16
- **14-09-01 Conversation Recording Destination for Trunks – ACI Recording Destination Extension Number**
Assign the ACI Call Recording destination on a per trunk basis. The destination can be an ACI port extension number (assigned in Program 11-06-01) or an ACI Department Group pilot number (assigned in Program 11-08-01). If destinations are assigned in Program 14-09 and Program 15-12, the destination in Program 15-12 is followed.
- **14-09-02 Conversation Recording Destination for Trunks – ACI Automatic Recording for Incoming Calls**
Determine whether or not a trunk should be automatically recorded when an incoming call is received.

0 = Off
1 = On
- **15-07-01 Programmable Function Keys**
If required, program an ACI Conversation Record Key (code 69 + 0). This key allows an extension user to press the key to manually record a call to the ACI.

-
-
- ➔ **15-12-01 Conversation Recording Destination for Extensions – ACI Recording Destination Extension Number**
Assign the ACI Call Recording destination per extension. The destination can be an ACI port extension number (assigned in Program 11-06) or an ACI Department Group pilot number (assigned in Program 11-08). If destinations are assigned in PRG 14-09 and PRG 15-12, the destination in PRG 15-12 is followed.

 - ➔ **15-12-02 Conversation Recording Destination for Extensions – ACI Automatic Recording for Incoming Calls**
Determine whether or not an extension should be automatically recorded when an incoming call is received.
0 = Off
1 = On

 - ➔ **33-01-01 ACI Port Type Setup – ACI Type**
Set each ACI software port for input (1) or input/output (2). Use input ports for Music on Hold sources. Use output ports for External Paging/ringer control.
ACI Ports: 1~96

 - ➔ **33-02-01 ACI Department Calling Group – Group Number**
Assign ACI software ports to ACI Department Groups. This lets ACI callers connect to ACI software ports by dialing the group's pilot number (set in Program 11-08).
ACI Ports: 1~96
ACI Groups: 1~16

Operation



To call an ACI software port:

1. Press idle **Speaker**.
2. Dial ACI software port extension number.
- OR -
Dial ACI Department Group extension number.
- OR -
Press the **One-Touch Key** for ACI extension or Department Group.

After you call an ACI software port:

- If the port is set for input (Program 33-01-01=1) and a music source is connected, you hear music.
- OR -
- If the port is set for output (Program 33-01-01=2) and External Paging is connected, you can page into the external zone.
- OR -
- If the port is set for output (Program 33-01-01=2) and a loud ringer is connected, you activate the loud ringer.

To manually record a call an ACI software port:

1. While on a call, press the **ACI Conversation Record** key (PRG 15-07 or SC 751: 69 + 0).
 *The key LED is on when recording.*
2. To stop recording, press the **ACI Conversation Record** key once again.
 *The key LED is off when recording has stopped.*

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Ancillary Device Connection

Enhancements

<p>Version 1500 uses the corresponding port on the ESIE(8)-U20 ETU as the port number used when using the APR on the second B-channel and not ports 193~256.</p>

Description

Ancillary Device Connection allows installation of selected peripheral (ancillary) devices to a multiline terminal. This feature enhances peripheral device objectives.

An Electra Elite IPK II multiline terminal user can accomplish this by using the AP(R)-R Unit (Analog Port Adapter with Ringer) or AP(A)-R Unit (Analog Port Adapter without Ringer) for analog telephone devices, or installing the AD(A)-R Unit to connect devices such as tape recorders.

The AP(A)-R/AP(R)-R Unit is the interface for installing a single line telephone, Modem, credit card reader, wireless headset, NEC Conference Max Conferencing unit or other compatible analog device.

The Elite IPK II *D^{term}* IP Gig Adaptor allows all compatible NEC IP multiline terminals to support a Gigabit connection.

System Availability

Terminals

All multiline terminals except DTR-2DT-1 TEL, DTP-2DT-1 TEL, and DTP-16HC-1 TEL.

The Elite IPK II *D^{term}* IP Gig Adaptor supports all compatible NEC IP multiline terminals.

Required Component(s)

APR-U, AP(R)-R, APA-U, AP(A)-R, or Dterm Gig Adaptor.

Conditions

- The optional device fits underneath the terminal.

- A single line telephone connected to an APR-U/AP(R)-R Unit or APA-U/AP(A)-R Unit cannot perform Trunk-to-Trunk Transfer and does not support a conference with itself and two outside parties.
- A single line telephone connected to an APR-U/AP(R)-R Unit or APA-U/AP(A)-R Unit does not support Message Waiting Indication or Caller ID Indication.
- An AP(R)-R Unit (analog port adapter with ringer) can be installed on a multiline terminal and function separately from the multiline terminal.
- When PRG 10-03-06 is assigned as APR you cannot manually assign a port number for an APR. The system uses ports 193~256 (starting with 256 and working down) for a total of 64 APR ports. APR 1 uses port 256, and APR 2 uses port 255, and so on (**Version 1100 or lower.**)
- The multiline terminal must be connected to an ESIB(8)-U20 ETU, and the ESIB(8)-U20 ETU must have an ESIE(8)-U20 ETU installed [ESIB/E(16)]. The corresponding port on the ESIE(8)-U20 ETU does not support a station or other hardware when used by the APR for channel 2.
- When PRG 10-03-06 is assigned as APR you cannot manually assign a port number for the APR. The system uses the corresponding port on the ESIE(8)-U20 ETU as the port number. For example, if a multiline terminal is connected to port 1 of the ESIB/E(16)-U20, the APR would use port 9 of the ESIB/E(16)-U20. (**Version 1500 or higher required.**)
- The D^{term} IP Gig Adaptor has three RJ45 connections on the back enabling a Gigabit connection from the LAN, a Gigabit connection to the PC, and a 10/100 connection to the phone.
- When using a gigabit connection, a straight cable category 5e or higher is required (maximum length 100 meters).
- DTP-2DT-1 and DTR-2DT-1 telephones have a built in APA adapter.
- Phones that have an APR/APA installed do not pass voice to a trunk until the interdigit timer expires (Program 21-01-03).

Default Settings

None

Related Features

Electra Elite IPK Terminals

Electra Elite Terminal Migration

Programming

- ➔ 15-03-01 Single Line Telephone Basic Data Setup – SLT Signaling Type
- ➔ 15-03-04 Single Line Telephone Basic Data Setup – Flashing

Operation

Depends on the connected ancillary device.

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Answer Hold

Description

Answer Hold allows a multiline terminal user to press the flashing Answer to answer an incoming ringing call or a Camp-On call. When the multiline terminal user is already answering a call, the first call is automatically placed on hold, depending on the user's setting in Program 15-02-06.

Conditions

- When multiple incoming calls activate the Answer Key LED, the LED continues to flash until all calls are answered.
- Use Program 15-02-06 (Normal Common, Exclusive Hold) to set the type of Hold to be used. (Default = Normal Common)
- For calls placed in a Park Group, the LED blinks fast (green).
- For calls placed in a Park Group by another user, the LED blinks slow (red).
- The Answer Hold Feature is available for Virtual Extensions.
- The Answer Hold feature does not function for incoming internal calls.
- CO/PBX incoming calls, not assigned to ring or assigned to another ring group, do not activate the Answer Hold feature.
- If the direct trunk appearance key is not assigned when all Call Appearance Keys are in use, the next incoming call cannot be answered.

Default Settings

Normal Hold

System Availability

Terminals

Any Multiline Terminal

Required Component(s)

Not Applicable

Related Features

Answer Key



Central Office Calls, Answering

Programming

- ➔ **15-02-06 Multiline Telephone Basic Data Setup – Hold Key Operating Mode**
0 = Normal Common
1 = Exclusive Hold
- ➔ **15-07 Programmable Function Keys**
*04 = Park Group Key
- ➔ **11-12-32 Service Code Setup (for Service Access) – Answer for Park Hold**
Default = *6
- ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M**
Default = COS 01~15 = 1

Operation

To answer a call on a different line key or CAP key with a call in progress:

1. Receive a CO/PBX, DID/DISA/DIL/E&M incoming ring.
 *Answer flashes.*
2. Press **Answer** and answer the new call.
 *The Answer LED goes out. The original call is put on hold.*
3. If additional calls are received, press **Answer** to place the current call on hold and connect to the next call as long as Call Appearance Keys and or CO line keys are available.

Answer Key

Description

Multiline Terminals have an Answer Key with an LED that flashes when the Multiline Terminal user receives an incoming CO/PBX, Tie/DID transfer, or CO/PBX transfer call. When multiple calls are received, Answer is used to pick up calls, and continues flashing until the last unanswered call is answered. Press Answer during a call to hold the current call and allow the next call to be answered.

Conditions

- The Answer LED functions for incoming CO/PBX calls, CO/PBX transfer/camp-on calls, and transfer/camp-on Tie/DID calls.
- Incoming calls are handled *first in-first out*.
- An Internal call, internal transfer/camp-on call, CAR/SIE/VE calls do not activate the Answer LED.

Default Settings

None

System Availability

Terminals:

All Multiline Terminals

Required Component(s)

None

Related Features

Answer Hold

Programming

- **22-02-01 Incoming Call Trunk Setup**
Assign the incoming trunk type for each trunk.
- **22-04-01 Incoming Extension Ring Group Assignment**
Assign extensions to Ring Groups.
- **22-05-01 Incoming Trunk Ring Group Assignment**
Assign trunks to incoming Ring Groups.
- **22-07-01 DIL Assignment**
Assign the destination extension or Department Calling Group for each DIL Incoming trunk.

Operation

To answer calls using the Answer Key:

1. Receive CO/PBX incoming ring.
2. Press **Answer**.
3. Talk with the CO/PBX incoming calling party.
4. When additional CO incoming calls are received, press Answer to place the current call on hold and connect the multiline terminal user to the next call.

Attendant Call Queuing

Description

Attendant extensions can have up to 32 incoming calls queued before additional callers hear busy tone. This helps minimize call congestion in systems that use the attendant as the overflow destination for unanswered calls. For example, you can program Direct Inward Lines and Voice Mail calls to route to the attendant when their primary destination is busy. With Attendant Call Queuing, these unanswered calls normally “stack up” for the attendant until they can be processed.

The 32 call queue total includes Intercom, DISA, DID, DIL, Tie Line and transferred calls. If the attendant does not have an appearance for the queued call, it waits in line to be answered. If the attendant has more than 32 calls queued, an extension can Transfer a call to the attendant only if they have Busy Transfer enabled.

Attendant Call Queuing is a permanent, non-programmable system feature.

Conditions

Forwarding when unanswered or busy can occur only at the attendant if more than 32 calls are in queue.

Default Settings

Enabled

System Availability

Terminals

Any Multiline Terminal assigned as an operator

Required Component(s)

None

Related Features

Call Forwarding

Programming

- **20-17-01 Operator Extension – Operator's Extension Number**
Assign the operator (attendant's) extension.

- **24-02-01 System Options for Transfer – Busy Transfer**
Prevent (0) or allow (1) extensions to transfer calls to an attendant that has more than 32 calls in queue.

Operation

None

Automatic Call Distribution (ACD)

Enhancements

In **Version 1100 or higher**, Delay Announcements using Analog Communications Interface (ACI) are supported.

Version 1600 or higher:

- ACD calls which overflow and are answered outside the original group are counted as overflowed calls in the MIS software. Prior to this release, an overflowed call was counted as an Abandoned call.
- A caller in ACD queue can dial out of a queue during a VRS delay announcement.
- A remote K-CCIS user can call, or transfer a call directly to an ACD Pilot number. Prior to this release, this type of call or transfer was not allowed.

Version 2000 or higher

- Delay Announcements using In-Mail are supported and 1-Key dial out options are supported when using VRS Delay Announcements.

Description

Automatic Call Distribution (ACD) uniformly distributes calls among agents of a programmed ACD Group. When a call rings into an ACD Group, the system automatically routes the call to the agent that has been idle the longest. Automatic Call Distribution is much more sophisticated and comprehensive than Department Calling and other group services – it can accurately judge the work load at each agent and distribute calls accordingly. The system allows up to 64 ACD Groups and 256 ACD agents.

You can put any agent in any group. In addition, an agent can be in more than one group only when using AICs. This allows, for example, a Technical Service representation to answer customer service calls at lunch time when many of the Customer Service representatives are unavailable.

The ACD Master Number is the extension number of the whole group. Calls directly ringing or transferred to the ACD Master number enter the group and are routed accordingly. Although the master number can be any valid extension number, you should choose a number that is out of the normal extension range.

Automatic Call Distribution operation is further enhanced by:

ACD Call Queuing

When all agents in an ACD Group are unavailable, an incoming call queues and causes the Queue Status Display to occur on the ACD Group Supervisor's display. The display helps the supervisor keep track of the traffic load in their group.

The Queue Status Displays shows:

- The number of calls queued for an available agent in the group.
- The trunk that has been waiting the longest, and how long it has been waiting.

For each ACD Group, you can set the following conditions:

- The number of trunks that can wait in queue before the Queue Status Display occurs.
- How often the time in queue portion of the display reoccurs.
- If the supervisor should hear a Queue Alarm when the time in queue portion reoccurs.
- This alarm is a single beep tone that reminds the supervisor to check the queue.
- With **Version 1600 or higher**, a remote K-CCIS user can call, or transfer to an ACD Pilot number. However, an incoming K-CCIS call to the ACD Pilot does NOT provide a Link Reconnect.

ACD Overflow (With Announcements)

ACD offers extensive overflow options for each ACD Group. For example, a caller ringing in when all agents are unavailable can hear an initial announcement (called the 1st Announcement). This announcement can be a general greeting like, "Thank you for calling. All of our agents are currently busy helping other customers. Please stay on the line and we will help you shortly." If the caller continues to wait, you can have them hear another announcement (called the 2nd Announcement) such as, "Your business is important to us. Your call will be automatically answered by the first available agent. Please stay on the line." If all the ACD Group's agents still are unavailable, the call can automatically overflow to another ACD Group or the Voice Mail. If all agents in the overflow ACD Group are busy, Lookback Routing automatically ensures that the waiting call rings into the first agent in either group that becomes free.

You can assign an ACD Group with any combination of 1st Announcement, 2nd Announcement and overflow methods. You can have, for example, a Technical Service group that plays only the 2nd Announcement to callers and then immediately overflows to Voice Mail. At the same time, you can have a Customer Service group that plays both announcements and does not overflow.

You can assign an ACD Group to play the Queue Depth only when using the VRS for message. The Queue Depth can be played after the 1st Announcement only, 2nd Announcement only, or after both Announcements.

Dial Out of Delay Announcements

When listening to a VRS delay announcement, the caller can press a 1-key option to transfer them out to another extension, Voice Mail, Ring Group, another ACD Group, and to a Speed Dial bin. The caller can press the digit during the message only or for X seconds after the message. This is a per Queue option, but it effects both the first and second delay announcement if set. (**Version 1600 or higher required.**)

VRS Delay Announcements Using In-Mail

With **Version 2000 or higher**, In-Mail can provide ACD Delay Announcements. Any of the 16 (1~16) In-Mail Master mailboxes (PRG 47-03-01) can be set to Announcement mailboxes and can be used as the message source for the 1st and 2nd Announcement Messages. This option is applicable only to ACD Overflow modes that are assigned ACD delayed messages and PRG 41-08-03 must be set to 2.

Agent Log In and Log Out Services

An ACD Agent can log in and log out of their ACD Group. While logged in, the agent is available to receive ACD Group calls. When logged out, the agent is excluded from the group calls. The programmable keys and Alphanumeric Display on an agent's telephone show at a glance when they are logged in or logged out.

Agent Identity Code (AIC)

An Agent Identity Code (AIC) allows ACD agents to log in any extension without setting Program 41-02-01. Using AIC, ACD agents can also log in to multiple ACD groups at the same time (up to 64 ACD Groups). The system also allows all extensions (up to the system maximum) to log in using the same AIC code. AIC and ACD groups for each work period (mode pattern number) can be set in Program 41-18-01 as shown in the following example.

Table #	AIC	Operation Group	Mode Pattern Number							
			1	2	3	4	5	6	7	8
1	789	1	1	1	-	-	-	-	-	-
2	789	1	2	1	-	-	-	-	-	-
3	789	1	16	1	-	-	-	-	-	-
4	567	10	10	10	10	10	10	10	10	10
5	678	2	2	2	2	2	2	2	2	2
6	678	2	3	3	3	3	3	3	3	3
7	678	2	5	5	5	5	5	5	5	5

EXAMPLE:

With this example, ACD works as follows:

Example 1: Log In with AIC 789

- During Mode Pattern 1, ACD agents belong to ACD groups 1, 2, and 16 at the same time.
- During Mode Pattern 2, ACD agents belong to only ACD group 1.

- During Mode Pattern 3~8, ACD agents do not belong to any ACD group and the ACD extensions work as normal extensions.

Example 2: Log In with AIC 567

- During Mode Patterns 1~8, ACD agents belong to only ACD group 10.

Example 3: Log In with AIC 678

- During Mode Patterns 1~8, ACD agents belong to ACD groups 2, 3 and 5 at the same time.

Multiple Agent Log In

ACD agents can log in any extension with multiple AICs (up to three). Using the example setup above, ACD works as follows:

EXAMPLE:

Example 1: Log In with AIC 789 and 567

- During Mode Pattern 1, ACD agents belong to ACD groups 1, 2, 10 and 16 at the same time.
- During Mode Pattern 2, ACD agents belong to ACD groups 1 and 10.
- During Mode Pattern 3~8, ACD agents belong to only ACD group 10.

Example 2: Log In with AIC 789, 567 and 678

- During Mode Pattern 1, ACD agents belong to ACD groups 1, 2, 3, 5, 10 and 16 at the same time.
- During Mode Pattern 2, ACD agents belong to ACD groups 1, 2, 3, 5 and 10.
- During Mode Pattern 3~8, ACD agents belong to only ACD groups 2, 3, 5 and 10.

Some conditions with Multiple Agent Log In:

- ACD agents cannot log in to the system supervisor or group supervisor's extension.
- To log in with AIC, the extension should be set to AIC Log In mode in Program 41-17-01.
- If the extension is set to AIC log in mode in Program 41-17-01, the system ignores the setting of Program 41-02-01 for the extension.
- Multiple extensions (up to the maximum capacity of the extension) can log-in with one AIC. For example, even if ACD agent A logs in extension 350 with AIC 789, ACD agent B can also log in to extension 351 with the same AIC 789 at the same time.
- A supervisor cannot log out an agent logged in by an AIC code.

Emergency Call

If an ACD Agent needs assistance with a caller, they can place an Emergency Call to their ACD Group Supervisor. Once the supervisor answers the Emergency Call, they automatically monitor both the ACD Agent and the caller. If the agent needs assistance, the supervisor can join in the conversation. Emergency Call can be a big help to inexperienced ACD Agents that need technical advice or assistance with a difficult caller. The supervisor can easily listen to the conversation and then “jump in” if the situation gets out of hand.

Enhanced DSS Operation

A programmed extension user can use their DSS Console to monitor the status of the ACD Agents within a group. The DSS Console is an essential tool for supervisors. The console key flash rates tell the supervisor at a glance which of the group agents are:

- Logged onto the group (i.e., in service).
- Logged out of the group (i.e., out of service).
- Busy on a call.
- Placing an Emergency Call to the supervisor.
- Not available or installed.

The ACD Supervisor can also use their console for placing and transferring calls – just like any other extension user.

The expanded capacity for DSS Consoles (two pages) which was added in **Version 2000** is not supported for DSS Consoles in the ACD Monitor mode.

Flexible Time Schedules

An ACD Work Schedule lets you divide a day into segments (called Work Periods) for scheduling the activity in your ACD Groups. You can set up four distinct Work Schedules, with up to eight Work Periods in each Work Schedule. Each day of the week has one Work Schedule, but different days can share the same schedule. For example, your Monday through Friday Work Schedule could consist of only two Work Periods. Work Period 1 could be from 8:00 AM to 5:00 PM – when your business is open. Work Period 2 could be from 5:00 PM to 8:00 AM – which covers those times when your business is closed.

Headset Operation (With Automatic Answer)

An ACD Agent or ACD Group Supervisor can use a customer-provided headset in place of the handset. The headset conveniently frees up the user’s hands for other work and provides privacy while on the call. In addition, an ACD Agent with a headset can have Automatic Answer. This allows an agent busy on a call to automatically connect to the next waiting call when they hang up.

Incoming Call Routing

Incoming trunk calls can automatically route to specific ACD Groups. These types of calls ring directly to the ACD Group without being transferred by a co-worker or the Automated Attendant.

Rest Mode

Rest Mode temporarily logs-out an ACD agent's telephone. There are two types of Rest Mode:

Manual Rest Mode:

An ACD Agent can enable Manual Rest Mode anytime they want to temporarily leave the ACD Group. They might want to do this if they go to a meeting or get called away from their work area. While in Rest Mode, calls to the ACD Group do not ring the agent's telephone.

Automatic Rest Mode:

When an ACD Group has Automatic Rest Mode, the system automatically puts an agent's telephone in Rest Mode if it is not answered. This ensures callers do not have to wait while ACD rings an extension that is not answered. For multiline terminals, the system enables Automatic Rest Mode for all telephones with Rest Mode keys. For single line telephones, you must set an option in programming to enable Automatic Rest Mode. If an agent's telephone is placed into Rest Mode because a call is not answered, the agent needs to manually cancel Rest Mode in order to log back into the ACD group.

With a Rest Mode key programmed on an ACD agent's telephone, when the agent is in rest mode, the key is on. If the Rest Mode key is pressed while an agent is on a call, the key flashes to indicate a pre-Rest Mode status. When the current call is finished, the agent's telephone is in rest mode. The agent can place intercom calls or receive direct incoming calls while in Rest Mode. The ability to receive incoming intercom calls is defined in system programming for each ACD group. Note that an ACD System Supervisor cannot be placed in Rest Mode.

Supervisor, ACD Group

You can designate an extension in an ACD Group to be the group's supervisor. Once assigned as an ACD Group Supervisor, the user can:

- Take the entire ACD Group out of service.
- Check the log out status of each agent after the group is taken down.
- Restore the ACD Group to service.

During programming, you can choose one of three modes of operation for each ACD Group supervisor:

- Supervisor's extension cannot receive calls to the ACD Group.
- Supervisor's extension can receive only ACD Group calls during overflow conditions.
- Supervisor's extension receives calls just like any other ACD Group agent.

An ACD Group can have only one supervisor. In addition, an extension can be a supervisor for only one ACD Group.

Supervisor, ACD System

You can designate an extension as an ACD System Supervisor. Once assigned as an ACD System Supervisor, the user can:

- Take the all the system ACD Groups out of service simultaneously.
- Check the log out status of each agent after the groups are taken down.
- Restore all the ACD Groups to service simultaneously.

The system can have only one ACD System Supervisor.

Work Time

Work Time temporarily busies-out an ACD agent's telephone so they can work at their desk uninterrupted. This gives the agent time to fill out important logs and records as soon as they are finished with their call. There are two types of Work Time:

Manual Work Time:

An ACD Agent can enable Manual Work Time anytime they need to work at their desk undisturbed. You might prefer this Work Time mode if an agent only occasionally has to fill out follow-up paper work after they complete their call. When the agent is through catching up with their work, they manually return themselves to the ACD Group.

Automatic Work Time:

The system implements Automatic Work Time for the agent as soon as they hang up their current call. This is helpful in applications (such as Tech Service groups) where follow-up paperwork is a requirement for every call. When the agent is done with their work, they manually return themselves to the ACD Group.

Hotline Key Shows Agent Status

An extension Hotline key provides the normal Busy Lamp Field (BLF) for co-workers and a unique BLF for ACD Agents. Similarly to the supervisor's DSS Console BLF, the unique BLF shows when the covered agent is in service, out of service or busy on a call. This enhanced BLF gives a department manager, for example, ACD Group monitoring abilities without having to become a supervisor with a DSS Console.

Hotline gives a multiline terminal user one-button calling and Transfer to another extension (the Hotline partner). Hotline helps co-workers that work closely together. The Hotline partners can call or Transfer calls to each other just by pressing a single key. Enhanced for ACD applications, Hotline provides a unique Busy Lamp Field for ACD agents as well as a BLF for co-workers that are not ACD agents. The charts below show both sets of BLF indications.

BLF For ACD Agents	
When the key is . . .	The ACD Agent is . . .
Off	Idle and is not an ACD Agent

BLF For ACD Agents	
On	Busy
Double Wink Off	Making an Emergency Call
Wink Off	Logged off or not installed
Double Wink On	Logged on

BLF For Co-Workers That Are Not ACD Agents	
When the key is . . .	Your co-worker is . . .
Off	Idle
On	Busy or ringing
Fast	Flash In Do Not Disturb – All calls (option 3) or Intercom calls (option 2)

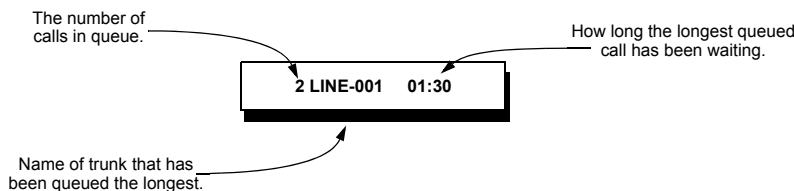
Enhanced Supervisor Options:

An ACD supervisor can individually assign extensions to ACD Groups, and set an agent’s status once assigned. This provides the supervisor with tremendous flexibility to reassign agents as work loads vary.

Queue Status Display with Scrolling:

When all agents in an ACD Group are unavailable, an incoming call queues and causes the Queue Status Display to occur on the ACD Group Supervisor and/or agent’s display (based on the Class of Service). The display helps the supervisor keep track of the traffic load in their group. In addition, any display multiline terminal can have a Queue Status Display Check programmable function key. The multiline terminal user can press this key anytime while idle, and using the VOL (▲) and VOL (▼), scroll through the Queue Status Displays of all the ACD Groups. The Queue Status Display shows (see the Queue Status Display illustration below):

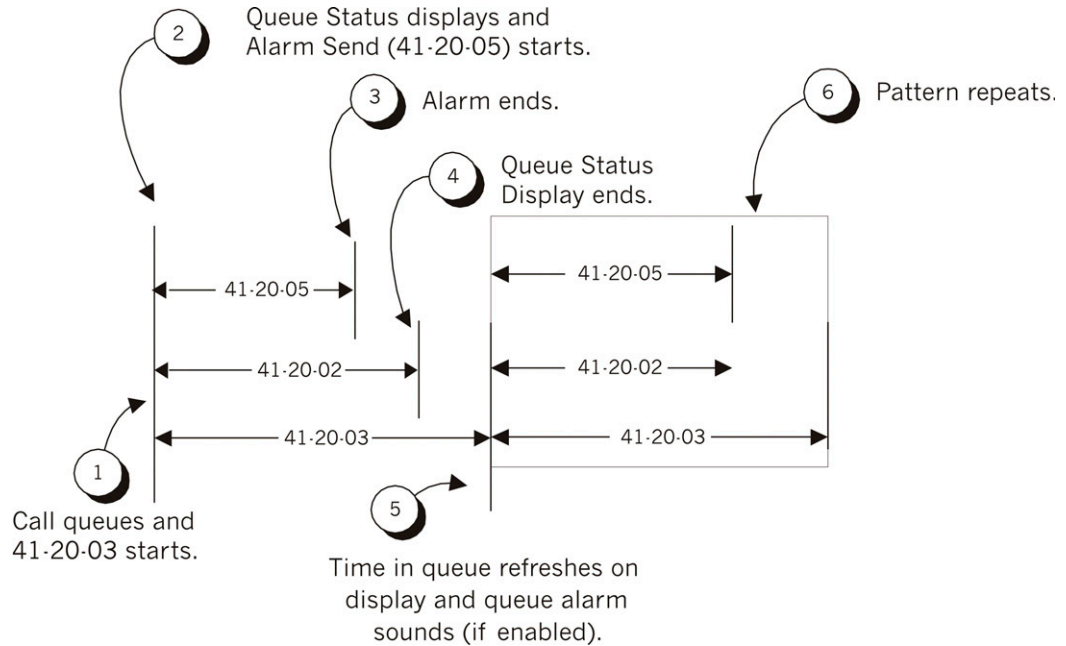
- The number of calls queued.
- The trunk that has been waiting the longest, and how long it has been waiting.



For each ACD Group, you can set the following conditions:

- The number of trunks that can wait in queue before the Queue Status Display occurs.

- How often the time in queue portion of the display reoccurs (see the Queue Status display Timing illustration below).
- Queue Status Display holding time.
- Queue Status Alarm enable/disable.
- Queue Status Alarm sending time.









When Logged Out of ACD Group:

When ACD agents are logged out and a call is placed into the ACD queue, the telephones of the logged out agents display the Queue Status and they hear the alarm according to the settings defined in system programming. Pressing the Queue Status Display Programmable Function key returns the telephone to idle until the time in Program 41-20-03 expires again.

Do not use both PRG 41-15-01~02 and PRG 41-20-01~05 to set the ACD queue alarm. Select either one or the other for the system to follow.

Feature	Available in Program 41-15-01~02	Available in Program 41-20-01~05
Queue Status Display	---	Yes
Queue Status Display Time	---	Yes
Alarm	Yes	Yes

Feature	Available in Program 41-15-01~02	Available in Program 41-20-01~05
Alarm Send Time	Program 41-15-02 determines the length/interval of the alarm.	Yes
Interval Time of Queue Status Display		Yes
Class of Service	---	Yes
Timing of alarm and display queue status	Alarm triggered after the number of calls in Program 41-15-01 is exceeded.	Alarm triggered after the number of calls in Program 41-20-01 is exceeded. Then follows Program 41-20-03 timing for displaying status.

-  *If a telephone is not idle, it cannot use the Queue Status Display Programmable Function key.*
-  *The Queue Status Display is not shown and the Queue Alarm is not heard by ACD agents those in Off-Duty mode.*
-  *To scroll through the ACD groups queue status, the Queue Status Display Programmable Function key must be used. You cannot scroll when the Queue Status Display appears due to an alarm condition.*
-  *If the Queue Status display and alarm are active and the queued called is answered/disconnected, the display and alarm continue until the times in Program 41-20-02 and Program 41-20-05 expire.*
-  *When an overflowed call is in queue, the call is included in its original ACD group queue and not in the group queue to which it overflowed.*
-  *The Queue Status is not displayed on a supervisor's telephone based on the settings in PRGs 41-20-xx. The supervisor must use the Queue Status Display Programmable Function key to view the queue.*

Programmable Wrap-up Timer

When an agent finishes their call, the system automatically starts a wrap-up timer and blocks any ACD calls to the agent. This gives them time to complete important logs and records before a new call comes in. When the time expires, the system returns the agent to the ACD Group to handle new callers.

MIS

The Elite IPK II ACD MIS is a series of Windows-based software programs designed to enhance the ACD features of the IPK II Telephone System. The software displays both real-time data and historical reports. Refer to the Electra Elite IPK II ACD MIS Supervisors Manual for more information.

ACD Group as Overflow Destination

The system can transfer an overflow call to a specific ACD Group, off-site via a speed dial bin, Ring Group or to voice mail using Program 41-09-01. When Program 41-08-02: ACD Overflow Destination has the ACD Overflow Destination set to 65, the system overflows the call to the ACD Group programmed in Program 41-09-01. (The system does not allow you to program an ACD group with that ACD group as the overflow.) If, while the call is ringing, the extension where the call was transferred becomes available, both the extension and the overflow ACD group ring.

Conditions

System:

- In-Mail can be used to play ACD Delay Announcements (**Version 2.00 or higher required.**)
- If all agents are logged out of an ACD Queue, a transferred call to the ACD Pilot number recalls immediately back to the transferring party.
- If all agents are logged out of an ACD Queue trunk call directly, ringing the Queue is placed in queue.
- If defined in Program 22-11-03, DID calls in queue display the trunk name with the Queue Status feature.
- When PRG 12-07-01 is customized, an agent's display does not indicate the WAIT ACD LOGIN status, however an agent may still log in.
- Conversation Recording is programmed system-wide – it is not ACD feature-specific.
- Refer to the Electra Elite IPK II ACD Manual for additional information.
- Up to 16 channels (speech paths) are available when using the DSP with VRS installed on the CPU11 for messages.
- When the PGD(2) is providing the 1st Delay Announcements, it continues to play until the call is answered, abandoned, or the timer in 41-10-04 expires and starts to play the 2nd Delay Announcement. The 2nd Delay Announcement continues to play until the call is answered, abandoned, or the timer in 41-10-05 expires and drops the call. This message does not start from the beginning because it is on a constant loop (**V1.10 or higher is required.**)
- The Dial Out of Queue feature is not supported during In-Mail Delay Announcements.
- Wireless – DECT is not supported with ACD.
- When all In-Mail talk paths (ports) are simultaneously being accessed by In-Mail Mailbox subscribers or Voice Mail Delay Announcements, or combination of the two, the next incoming call to the In-Mail Rings No Answer until an available talk path becomes idle (First Come – First Serve).

- When Voice Mail Delay Announcements are being played, In-Mail talk paths (ports) are used.
- In-Mail cannot be used for ACD Night Announcement.
- Program 41-08-03: ACD Overflow Options – Delay Announcement Source Type is not supported in **Version 1600**.
- The ACI port used for the ACD Delay Announcements program up like Music on Hold (MOH) ACI ports. Refer to the MOH [Programming on page 2-737](#).
- Door Boxes do not ring Logged In ACD agents.
- Transferred calls to an ACD Queue receive ringback and do not follow PRG 24-02-02.

MIS:

- The IPK II system does not buffer the ACD Statistics when the PC running the ACD Server application is not connected.
- If an ACD Queue call is transferred to another logged-in Agent, it shows that both agents took a call but the queue only shows it as one total call.
- The programming of the Agents and Queues in the IPK II system are not transferred to the PC running the ACD Server/MIS applications. The ACD Server/MIS applications are programmed separately.
- If the caller overflows out of the Queue to a Speed Dial Bin, Ring Group, or Voice Mail it reports as an abandoned call (**Version 1500 and lower.**)
- If the caller overflows out of the Queue to a Speed Dial Bin, Ring Group, or Voice Mail it reports as an overflow call (**Version 1600 or higher required.**)
- A supervisor assigned to not receive calls or take calls after the overflow timer is reached shows as idle in MIS when they are logged in and idle even when calls are queued up and not reaching the overflow timer.

Default Setting

Refer to the Electra Elite IPK II ACD Manual for more details.

System Availability**Terminals**

All Terminals

Required Component(s)

ACD Software License

DSP-U10 Unit

In-Mail (2 or 4) port flash drive (For Delay Announcements using In-Mail)

Required Software

Version 1100 or higher is required for Delay Announcements using ACI.

Version 2000 or higher is required for Delay Announcements using In-Mail.

Related Features

Direct Inward Dialing (DID)

Digital Voice Mail

IPK II In-Mail

Music on Hold

Night Service

Voice Mail Integration (Analog)

Programming

Refer to the Electra Elite IPK II ACD Manual for complete programming information.

- **41-14-13-1 ACD Options – ACD Queue 1-Digit Assignment (Version 1600 or higher required)**
For each ACD Queue (1~64) assign the One-Digit number (0~9, *, #) to be used for the One-Digit Dial Out Option.
- **41-14-13-2 ACD Options – Destination Number Type (Version 1600 or higher required)**
For each ACD Queue (1~64), assign the Destination Number Type.
 - 0 = None
 - 1 = Ext/VM
 - 2 = Ring Group
 - 3 = Speed Dial Bin
 - 4 = ACD Group

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-
- **41-14-13-3 ACD Options – Destination Number (Version 1600 or higher required)**
For each ACD Queue (1~64), assign the destination number for the assigned Destination Type. (Up to eight digits).
 - **41-14-14 ACD Options – DTMF Detection Assignment during Delay Announcement (Version 1600 or higher required)**
For each ACD Queue (1~64), assign if the One-Digit Dial Out option **can** (1 = Yes) or **cannot** (0 = No) be pressed during the Delay Announcements.
 - **41-14-15 ACD Options – DTMF Detect Time after Delay Announcement Message (Version 1600 or higher required)**
For each ACD Queue (1~64), assign the time (0~64800 seconds) after the Delay Announcement that the One-Digit Dial Out option works.

Operation

Using the Headset with Automatic Answer for ACD Agents:

1. With the multiline terminal idle, press **Feature**.
2. Press the **HEADSET** key (PRG 15-07-01 or SC 751: 05).
 - ✎ *The Headset key blinks when Automatic Headset is activated.*
 - ✎ *To cancel Automatic Headset, repeat these steps.*

Transferring Trunk Calls to the ACD Pilot Number:

1. While on an outside call, press **Transfer**.
2. Dial the ACD Pilot number.
3. Hang up.
 - ✎ *The call is transferred to the ACD group.*



A Supervisor can monitor an ACD call:

1. When an ACD agent is on an outside call, the supervisor presses the **MONITOR** key (PRG 15-07-01 or SC 752: *15).
 - ✎ *The supervisor can hear but cannot participate in the call. If participation is required, use the Barge-In feature instead.*
2. To cancel the call monitoring, press the **MONITOR** key again.

AIC Agent Log In



To log in:

Multiline Terminal


1. Press the **ACD LOG IN/LOG OUT** key (Program 15-07-01 or SC 752: *10).
- OR -
Press **Speaker** and dial the AIC Log In service code (Program 11-13-08).
2. Dial the log in code (up to 20 digits).
 *This step is not required if the ID code is disabled in PRG 41-01-02.*
3. Dial the Agent Identity Code (AIC) (up to four digits).
 *The **ACD LOG IN/LOG OUT** key lights.*

To log out (for single or multiple agent log ins):

Multiline Terminal

-  *All AIC log ins become logged out.*
1. Press the **ACD LOG IN/LOG OUT** key (Program 15-07-01 or SC 752: *10).
 2. Dial **1** to accept.
- OR -
Press **Speaker** and dial the AIC Log In service code (Program 11-13-08).
 *The **ACD LOG IN/LOG OUT** key goes out.*

Single Line Telephone



-  *All AIC log ins become logged out.*
1. Lift the handset.
 2. Dial the AIC Log Out service code (Program 11-13-08).
- OR -
 1. To log out of an ACD group without using AIC: Lift the handset.
 2. Dial the ACD Log Out service code **655** (Program 11-13-02).

Multiple Agent Log In






To log in:

Multiline Terminal

After already being logged in:




1. Press the **ACD LOG IN/LOG OUT** key (Program 15-07-01 or SC 752: *10).
 2. Dial **0** to cancel the log out option.
 3. Dial the Agent Identity Code (AIC) (up to four digits).
 -  *The **ACD LOG IN/LOG OUT** key lights.*
- OR -**
- Press **Speaker** and dial the AIC Log In service code (Program 11-13-08).
4. Dial the Agent Identity Code (AIC) (up to four digits).
 -  *The **ACD Log In/Log Out** key lights.*

Single Line Telephone

-  *Follow Steps 1~3 to log in with additional AICs (up to three) anytime.*
1. Lift the handset and dial the AIC Log In service code (Program 11-13-08).
 2. Dial the log in code (up to 20 digits).
 -  *This step is not required if the ID code is disabled in Program 41-01-02.*
 3. Dial the first Agent Identity Code (AIC) (up to four digits).
 -  *You hear a confirmation tone when immediately logging in with additional AICs.*
 4. For second agent log: Dial the second Agent Identity Code (AIC) (up to four digits).
 -  *You hear a confirmation tone.*
 5. For third agent log: Dial the third Agent Identity Code (AIC) (up to four digits).
 -  *You hear a confirmation tone.*

Queue Status Display

When Logged Into ACD Group

1. With an idle multiline terminal, press the Queue Status Display Programmable Function Key (Code: *19).
 -  *The display indicates the number of calls in queue, the trunk name, and the time the call has been waiting.*
 -  *When the Queue Status Display key is pressed, the queue status of the extension group is displayed. When the extension is not in an ACD group, the Queue Status of group 1 is displayed instead.*
 -  *When an agent logs in using an AIC code, the Queue Status of the default ACD group defined in PRG 41-18-02 is displayed.*
2. Press **VOL UP** and **VOL DOWN** to scroll through the Queue Status Displays of all the ACD Groups.
3. Press the **CLEAR** key to return the telephone to idle.

When Logged Out of ACD Group



When ACD agents are logged out and a call is placed into the ACD queue, the telephones of the logged out agents display the Queue Status and they hear the alarm according to the settings defined in system programming.

Pressing the Queue Status Display Programmable Function key returns the telephone to idle until the time in Program 41-20-03 expires again.



Rest Mode

To set the manual Rest Mode:

Multiline Terminal


1. With the multiline terminal in an idle state, press the **ACD Rest Mode** key (PRG 15-07-01 or SC 752: *13).
 -  *The ACD Rest Mode key lights. If the Rest Mode key is pressed while the agent is on an active call, the key flashes until the agent hangs up.*
 -  *This operation is not available for the System Supervisor.*

Single Line Telephone


1. Lift the handset and dial **658**.
 -  *A fast busy is heard.*
 -  *To set Pre-Rest Mode (while on a call), press the hookflash and then dial 658. Press the Hookflash again to return to the outside party. Rest Mode begins once the call is completed.*
2. Hang up.

To cancel the manual Rest Mode:

Multiline Terminal

1. Press the **ACD Rest Mode** key (PRG 15-07-01 or SC 752: *13).
 -  *The ACD Rest Mode key light goes off.*

Single Line Telephone

1. Lift the handset.
 -  *A fast busy is heard.*
2. Dial **659**.
3. Hang up.

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Automatic Release

Description

Automatic Release drops the line circuit when an outside party abandons the call. For this feature to work with Loop Start Trunks, the CO/PBX providing the outside line must provide a timed disconnect signal. Automatic Release is normally provided on Ground Start, DID, ISDN, and Tie Line trunks.

Conditions

- Automatic Release on ISDN trunks is provided by the protocol.
- When an outside line is accessed using a dedicated line key, the LED associated with the line key goes off when Automatic Release occurs.
- This feature functions while a call is in progress, on hold, or in a conference.
- This feature applies to all ICM type calls in progress, holding or parked.
- When Automatic Release occurs and the telephone is in handsfree mode, **Speaker** automatically turns off. If using the handset, the station is set to idle when the handset goes on-hook.

Default Settings

None

System Availability

Terminals

Not applicable

Required Component(s)

None

Related Features

Central Office Calls, Answering

Central Office Calls, Placing

Programming

- ➔ **14-02-09 Analog Trunk Data Setup – Busy Tone Detection**
0 = Disable (Default)
1 = Enable

- ➔ **14-02-14 Analog Trunk Data Setup – Loop Start/Ground Start**
0 = Loop Start (Default)
1 = Ground Start

Operation

None

Automatic Route Selection

Description

Automatic Route Selection (ARS) provides call routing and call restriction based on the digits a user dials. ARS gives the system the most cost-effective use of the connected long distance carriers.

ARS is an on-line call routing program that you can customize (like other system options) from a display telephone. ARS accommodates 400 call routing choices – without a custom-ordered rate structure database. With ARS, you can modify the system routing choices quickly and easily. This is often necessary in the telecommunication world of today where the cost structure and service choices frequently change.

The ARS feature can add or delete digits and route calls according to predetermined levels. When Electra Elite IPK II systems are networked together by Tie lines or K-CCIS, the networked systems can be called by a system number and a user's extension number, just an extension number, or by using a trunk access code.

ARS Feature Summary

ARS provides:

Call Routing

ARS can apply up to 24-digit analysis to every number dialed. For programming, ARS provides separate 8-digit and 24-digit tables. Each table can have up to 250 numbers.

Dialing Translation (Special Dialing Instructions)

ARS can automatically execute stored dialing instructions (called Dial Treatments) when it chooses a route for a call. The system allows up to 15 Dial Treatments. The Dial Treatments can:

- Insert or delete an area code (NPA)
- Add digits (such as a dial-up OCC number), pauses and waits to the dialing sequence
- Require the user to enter an authorization code when placing a call (refer to PRG 44-03)

Time of Day Selection

For routing, ARS provides 10 different day selections (called Time Schedule Patterns). Each Time Schedule Pattern can provide up to 20 time intervals which are assigned to one of the eight day/night modes. The Time Schedule Patterns are then assigned to a day of the week (Monday~Friday, Saturday, Sunday or Holiday).

Hierarchical Class of Service Control

ARS allows or denies call route choices based on an extension ARS Class of Service. This allows lower Classes of Service (e.g., 1) to access routes unavailable to higher Classes of Service (e.g., 16). The system provides up to 16 (0=unrestricted, 1~16) ARS Classes of Service.

 Separate Routing for Selected Call Types

To provide unique control, you can program separate routing instructions for:

- Directory assistance (411, 1411 and 555) calls
- Emergency (911) calls

 Separate Routing for Equal Access (1010XXX) Calls

Choose different routing for directly-dialed (1010XXX + 1) and operator-assisted (1010XXX + 0) Equal Access calls.

Basic ARS Operation

When a user places an outside call, ARS analyzes the digits dialed and assigns one of 400 Selection Numbers to the call. The Selection Number chosen depends on which digits the user dialed. ARS then checks the time of day, the day of week and the extension ARS Class of Service. Based on these call routing options, ARS selects a trunk group for the call and imposes the Dial Treatment instructions (if any).

Class of Service Option Allows Outgoing Calls to Not Follow Access Map

Using this option allows a Class of Service to be set so that ARS does not follow the trunk access map settings (Program 14-07-01 and Program 15-06-01). The feature allows an extension user to have CO line keys on their telephone which allow incoming access only. The user would only have outgoing access on the CO lines when using ARS to place a call.

Class of Service Matching

With the ARS Class of Service Match Access feature, you can determine whether the system should allow a call based on the COS assigned to the Dial Analysis Table (Program 26-02). This change can create a tenant-like application. It then uses the trunk group defined in the Additional Entry in Program 26-02-03 to place the outgoing call.

When this feature is enabled, the calls are routed in sequential order, and forward provided the Class of Service for the trunk groups match.

For this feature, **Program 26-01-06: Automatic Route Selection Service, COS Match Access** is used.

The examples below use the following system programming:

Program 26-02 for Dial Analysis Table for ARS set as:

Table No.	Program 26-02-01 Dial	Program 26-02-02 Service Type	Program 26-02-03 Add Data	Program 26-02-04 ARS COS
1	203@@@@@@@	1:Route to trunk group	3 (Group 3)	5
2	214@@@@@@@	1:Route to trunk group	1 (Group 1)	4
197	@@@@@@@@@@	1:Route to trunk group	2 (Group 2)	4
198	@@@@@@@@@@	1:Route to trunk group	3 (Group 3)	3
199	@@@@@@@@@@	1:Route to trunk group	2 (Group 2)	2
200	@@@@@@@@@@	1:Route to trunk group	1 (Group 1)	1

Program 12-02 for Automatic Night Service Patterns as:

Time Pattern No.	Program 12-02-01 Start Time	Program 12-02-02 End Time	Program 12-02-03 Operation Mode
1	00:00	08:30	2 (Night)
2	08:30	17:00	1 (Day)
3	17:00	00:00	2 (Night)

Program 12-02 for Automatic Night Service Patterns as:

Mode	Ext. 301	Ext. 302	Ext. 401	Ext. 402
Mode 1 (Day)	1	2	3	3
Mode 2 (Night)	1	4	3	5

Program 26-01-03 for ARS Misdialed Number Handling as: 1 (Warning Tone)**With Program 26-01-06: ARS COS Match Access disabled (set to 0):**

- If at 9:00 AM, each extension dialed 9+(203)926-5400
All Extension would use Trunk Group 3
- If at 9:00 AM, each extension dialed 9+(214)262-2000
All Extension would use Trunk Group 1
- If at 6:00 PM, each extension dialed 9+(203)926-5400
All Extension would use Trunk Group 3
- If at 6:00 PM, each extension dialed 9+(214)262-2000
Extension 301, 302 and 401 would use Trunk Group 1
Extension 402 would not be able to dial out as the COS is lower

With Program 26-01-06: ARS COS Match Access enabled (set to 1):

- If at 9:00 AM, each extension dialed 9+(203)926-5400
Extension 301 would use Trunk Group 1
Extension 302 would use Trunk Group 2
Extension 401, 402 would use Trunk Group 3
- If at 9:00 AM, each extension dialed 9+(214)262-2000
Extension 301 would use Trunk Group 1
Extension 302 would use Trunk Group 2
Extension 401, 402 would use Trunk Group 3
- If at 6:00 PM, each extension dialed 9+(203)926-5400
Extension 301 would use Trunk Group 1
Extension 302 would use Trunk Group 2
Extension 401, 402 would use Trunk Group 3
- If at 6:00 PM, each extension dialed 9+(214)262-2000
Extension 301, 302 would use Trunk Group 1
Extension 401 would use Trunk Group 3
Extension 402 would not be able to dial out as the COS does not match

Conditions

- Do not use ARS behind a Centrex/PBX.
- ARS is intended for areas that use the North American Number Plan (NANP).
- Line keys, Call Appearance (CAP) Keys, outgoing trunk group keys, dialing 704 + trunk group, dialing +trunk number, and speed dial numbers assigned to a certain trunk group can all be used to by-pass ARS.
- If no PBX access code is entered in the Dial Treatment, the system can still dial 911.
- Toll Restriction overrides ARS.

- A system with Automatic Route Selection cannot also have Trunk Group Routing.
- With ARS installed, Trunk Queuing automatically queues for the least costly route. The system automatically redials the queued call when the extension user lifts the handset.
- Speed Dialing may bypass ARS routing.
- Set up other options for outgoing calls (e.g., unassign line keys, adjust gains, ARS access key, Call Appearance (CAP) Keys, etc.).
- Refer to the Dial Tone Detection feature for the specifics on how the system handles Dial Tone Detection.
- ARS does not permit 0 and 011+ calls to be routed out separate trunk groups. The IPK II only supports direct trunk selection for dial 0 (Operator) type calls.
- If an entry of 911 is programmed in ARS, but ARS is turned off, 911 calls still attempt to route using ARS.

Default Setting

ARS is off (disabled) at default.

System Availability

Terminals

None

Required Component(s)

None

Related Features

Central Office Calls, Placing

Code Restriction

Dial Tone Detection

E-911 Compatibility

Speed Dial – System/Group/Station

Trunk Group Routing

Trunk Queuing/Camp On

Programming

- ➔ **11-01-01 System Numbering**
Set the system internal (Intercom) numbering plan. The numbering plan assigns the first and second digits dialed and affects the digits an extension user must dial to access other extensions and features, such as service codes and trunk codes. If the default numbering plan does not meet the site requirements, use this program to tailor the system numbering to the site.
- ➔ **11-09-01 Trunk Access Code**
Specify the single digit code used to access ARS (normally 9).
- ➔ **14-05-01 Trunk Group – Trunk Group Number**
Program trunks of the same carrier type into the same trunk group.
- ➔ **14-07-01 Trunk Access Map Setup**
Set up the Trunk Access Maps. This sets the access options for trunks

Trunk Access Maps: 1~200
- ➔ **15-06-01 Trunk Access Map for Extensions**
Assign Trunk Access Maps to extensions.

Trunk Access Maps: 1~200
- ➔ **20-03-04 System Options for Single Line Telephones – Dial Sending Start Time for SLT or ARS**
When ARS or an analog extension user accesses a trunk and dials an outside call, the system waits this interval before outdialing the first digit (0~64800 seconds).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-08-17 Class of Service Options (Outgoing Call Service) – ARS Override of Trunk Access Map**
Enable (1)/Disable (0) an extension user ability to override the trunk access map programming (Program 14-07-01 and Program 15-06-01) for outgoing calls.
- ➔ **26-01-01 Automatic Route Selection Service – ARS Service**
Enable (1)/Disable (0) ARS.
- ➔ **26-01-02 Automatic Route Selection Service – Network Outgoing Inter-Digit ARS Timer**
With Networking, this timer (0~64800 msec.) replaces PRG 20-03-04 when determining if all network protocol digits are received. If ARS is enabled at Site B, this timer can be programmed for 5 (500 msec) at Site A. If ARS is disabled and Site B is using F-Route for outbound dialing, this timer should be programmed for 30 (3 seconds) at Site A.

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- ➔ **26-01-03 Automatic Route Selection Service – ARS Misdialed Number Handling**
If a user dials a number not programmed in ARS, this option determines if the system should route over trunk group 1 (0) or play error tone (1).

 - ➔ **26-01-06 Automatic Route Selection Service – Class of Service Match Access**
With the ARS Class of Service Match Access feature, you can determine whether or not the system should allow a call based on the COS assigned to the Dial Analysis Table (Program 26-02). This change can be used to create a tenant-like application. It then uses the trunk group set in the Additional Entry in Program 26-02-03 to place the out-going call. When this feature is enabled, the calls are routed in sequential order, and forward – provided the Class of Service for the trunk groups match.

0 = Disabled (COS not compared)
1 = Enabled (COS Match Access)

 - ➔ **26-02-01 Dial Analysis Table for ARS/LCR – Dial**
Enter the digits (16 digits maximum: 1~9, 0 * #, @; 400 separate entries) for the Dial Analysis Table which is analyzed by ARS/LCR. This table is checked after any programmed F-Route operation is completed. The system then refers to PRG 26-02-02 and PRG 26-02-03 to determine the routing for the call. To enter a wild card/don't care digit, press Line Key 1 to enter an @. Remember that the system checks the table numbers in numerical order. This means that entries for specific numbers should be entered first (such as your local area codes), then enter the items containing wild card digits. If the system sees an entry of 2@@@ any table entry which follows is ignored.
For example, if 268, 269, and 270 are local exchanges, these would be the first three table entries which route according to the settings made in Program 26-02-02 and Program 26-02-03 for each of the table entries. If the next entry is 2@@@, the system checks no further in this program and routes all other 2xx numbers according to the entries made in PRG 26-02-02 and PRG 26-02-03 for this table entry.

 - ➔ **26-02-02 Dial Analysis Table for ARS – ARS Service Type**
For each Dial Analysis Table (1~200), select 0 for no ARS, 1 for *Service Type 1 – Route to Trunk Group Number* to have the number route to a trunk group [Refer to Program 26-02-03] or 2 for *Service Type 2 – F-Route Selected* to have the dialed number controlled by the F-Route table. If Service Type 2 is selected and F-Route operation is on, the F-Route table used is determined by Program 44-04. If F-Route operation is off, the routing is determined by Program 44-05.

 - ➔ **26-02-03 Dial Analysis Table for ARS – Additional Data/Service Number**
For each Dial Analysis Table (1~200), if Service Type 1 was selected in Program 26-02-02, enter the trunk group number (0~100, 0 = No Route).

 - ➔ **26-02-04 Dial Analysis Table for ARS – ARS Class of Service**
For each Dial Analysis Table (1~200), set the Automatic Route Selection (ARS) Class of Service (0~16).

 - ➔ **26-02-05 Dial Analysis Table for ARS – Dial Treatment for ARS**
For each Dial Analysis Table (1~200), set the Automatic Route Selection (ARS) Dial Treatment (0~15) to be used.
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- ➔ **26-03-01 ARS Dial Treatments – Treatment Code**
Assign the Dial Treatments (1~15) for automatic ARS dialing translation. Assign Dial Treatments to Service Numbers (Trunk Groups) in Program 26-02. The ARS Dial Treatment options are:
- **3** - Delete the NPA if dialed as part of the initial call. This requires at least eight digits in the ARS table (Program 26-02-01).
 - **2** - Delete the leading digit if dialed as part of the initial call. This requires at least eight digits in the ARS table (Program 26-02-01).
 - **1** - Add a leading 1 if not dialed as part of the initial call. This requires at least eight digits in the ARS table (Program 26-02-01).
 - **INPA** - Insert the NPA specified by NPA.
 - **DNN** - Outdial the NN number of digits or execute the code that follows. For example, D041234 out-dials 124. Valid entries are 0~9, #, *, Wnn (wait nn seconds) and P (pause). Each digit's code counts as a digit. So for example, if a P was added for a pause, the entry would look like: D05P1234. This Dial Treatment can only be added from telephone programming.
 - **Wnn** - Wait nn seconds.
 - **P** - Pause in analog trunk.
 - **R** - Redial the initially dialed number, including any modifications.
 - **E** - End of Dial Treatment. All Dial Treatments must end with the E code.
 - **X** - When ARS is enabled, X must be entered in the Dial Treatment for the system to output the extension number of the call's originator to the black box for the E911 feature.
- ➔ **26-04-01 ARS Class of Service**
Set an extension ARS Class of Service (0~16). Automatic Route Selection uses ARS Class of Service when determining how to route an extension call.
- ➔ **26-11-01 Transit Network ID Table – Transmit Network ID (Carrier ID)**
Enter the Transit Network Selection information element to be added to an ARS call using an ISDN trunk. This information element identifies a requested transit network.
- ➔ **44-01-01 System Options for ARS/F-Route – ARS/F-Route Time Schedule**
Select whether the ARS/F-Route feature should use the time schedule (0=not used, 1=used). If this option is set to 0, the F-Route table selected is determined only by the digits dialed without any relation to the day or time of the call. If this option is set to 1, the system first refers to Program 44-10. If there is a match, the pattern defined in that program is used. If not, the F-Route pattern in Program 44-09 and time setting in Program 44-08 are used.
- ➔ **44-02-01 Dial Analysis Table for ARS/F-Route Access – Dial**
Set the Dial digits for the Pre-Transaction Table for selecting ARS/F-Route (eight digits max: 1~9, 0 * #, @). To enter a wild card/don't care digit, press Line Key 1 to enter @.
- ➔ **44-02-02 Dial Analysis Table for ARS/F-Route Access – Service Type**
Set the Service Type (0~3) for the Pre-Transaction Table for selecting ARS/F-Route.
- 0 = No Setting
1 = Extension Call
2 = ARS/F-Route Table
3 = Dial Extension Analyze Table
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- ➔ **44-02-03 Dial Analysis Table for ARS/F-Route Access – Additional Data**
If a Service Type is selected in PRG 44-02-02, set the additional data, if required, for the Pre-Transaction Table for selecting ARS/F-Route (24 digits max: 1~9, 0 * #, @). To enter a wild card/don't care digit, press Line Key 1 to enter @.
 - ➔ **44-02-04 Dial Analysis Table for ARS/F-Route Access – Dial Tone Simulation**
Determine if the Dial Tone Simulation is on (1) or off (0) for the Pre-Transaction Table for selecting ARS/F-Route. If enabled, this option sends dial tone to the calling party once routing is determined. This may be required if the central office at the destination does not send dial tone.
 - ➔ **44-03-01 Dial Analysis Extension Table – Dial**
Set the Dial digits (24 digits max: 1~9, 0 * #, @) to be used for the Dial Extension Analysis Table. When Program 44-02-02 is set to type 3, this program sets the dial extension analysis table. These tables are used when the analyzed digits must be more than eight digits. To enter a wild card/don't care digit, press Line Key 1 to enter @.
 - ➔ **44-03-02 Dial Analysis Extension Table – ARS/F-Route Select Table Number (1~250)**
When dialed digits match the setting in Program 44-03-01, select the ARS/R-Route table number (0~500) to be used for the Dial Extension Analysis Table.
 - ➔ **44-03-03 Dial Analysis Extension Table – ARS/F-Route Select Table Number (251)**
If the received digits are not identified in tables 1~250, the F-Route selection table number (0~500) defined in table 251 is used.
 - ➔ **44-03-04 Dial Analysis Extension Table – Next Table Area Number (252)**
If the received digits do not match the digits set in tables 1~250, table number 252 is used refer to the next Extension Table Area (1~4) to be searched.
 - ➔ **44-04-01 ARS/F-Route Selection for Time Schedule**
Assign each ARS/F-Route Selection number (1~500) to an ARS/F-Route table number for each ARS/F-Route time mode. There are eight time modes for ARS/F-Route Access.
 - ➔ **44-05-01 ARS/F-Route Table – Trunk Group Number**
For each ARS/F-Route table (1~500) assign a priority number (1~4) and enter the trunk number to be used.

0 = No Setting
101~150 = Networking
255 = Extension Call
 - ➔ **44-05-02 ARS/F-Route Table – Delete Digits**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Enter the number of digits to be deleted (0~255) from the dialed number.
 - ➔ **44-05-03 ARS/F-Route Table – Additional Dial Number Table**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Enter the table number (defined in PRG 44-06) for additional digits to be dialed (0~1000).
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- ➔ **44-05-04 ARS/F-Route Table – Beep Tone**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select whether or not a beep is heard if a lower priority trunk group is used.

0 = No Beep
1 = Beep
 - ➔ **44-05-05 ARS/F-Route Table – Gain Table Number for Internal Call**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the gain table number used for internal calls (0~500).
 - ➔ **44-05-06 ARS/F-Route Table – Gain Table Number for Tandem Connections**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the gain table number used for tandem connection (0~500).
 - ➔ **44-05-07 ARS/F-Route Table – ARS Class of Service**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the Class of Service used for ARS (0~16). An extension ARS COS is determined in Program 26-04-01.
 - ➔ **44-05-08 ARS/F-Route Table – Dial Treatment**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the Dial Treatment used (0~15). Dial Treatments are defined in Program 26-03-01.
 - ➔ **44-05-09 ARS/F-Route – Maximum Digit**
For each ARS/F-Route table (1~500) assign the priority (1~4). Set the maximum digits to be dialed (1~99) for each ARS/F-Route table.
 - ➔ **44-06-01 Additional Dial Table**
If an Additional Dial Number Table is entered in Program 44-05-03, define the additional dial table (1~1000) to add digits in front of the dialed ARS/F-Route number (24 digits max: 1-9, 0 * #, Pause). To enter a wild card/don't care digit, press Line Key 1 to enter P (pause).
 - ➔ **44-07-01 Gain Table for ARS/F-Route Access – Incoming Transmit**
 - 44-07-02 Gain Table for ARS/F-Route Access – Incoming Receive**
 - 44-07-03 Gain Table for ARS/F-Route Access – Outgoing Transmit**
 - 44-07-04 Gain Table for ARS/F-Route Access – Outgoing Receive**
Set the gain table to be used (1~500). If an extension dials ARS/F-Route number;
 - The Extension Dial Gain Table is activated, which is assigned in Program 44-05.
 - The Extension Dial Gain Table follows Outgoing transmit and Outgoing receive settings.
 If the incoming call is transferred to another line using ARS/F-Route;
 - The Tandem Gain Table is activated, which is assigned in Program 44-05.
 - The Tandem Gain Table follows the Incoming transmit and Incoming receive settings for incoming line, and Outgoing transmit and Outgoing receive settings for the outgoing line.
 For ARS/F-Route calls, the CODEC gains defined in PRG 14-01-02 and PRG 14-01-03 are not activated.
 - ➔ **44-08-01 Time Schedule for ARS/F-Route**
Define the daily pattern of the ARS/F-Route feature. ARS/F-Route has 10 time patterns. These patterns are used in PRG 44-09 and PRG 44-10. The daily pattern consists of 20 time settings.
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- ➔ **44-09-01 Weekly Schedule for ARS/F-Route**
Define a weekly schedule for using ARS/F-Route day numbers 1~7 (1 = Sun, 7 = Sat), pattern numbers (1~10). The pattern number is defined in Program 44-08-01.
- ➔ **44-10-01 Holiday Schedule for ARS/F-Route**
Define a yearly schedule for ARS/F-Route. This schedule is used for setting special days such as national holidays (pattern numbers 1~10). The pattern number is defined in Program 44-08-01.
- ➔ **80-03-01 DTMF Tone Receiver Setup – Detect Level**
Use Items 11~32 to set the criteria for dial tone detection for outgoing ARS calls.

Operation

To place a call using ARS:

1. At the multiline terminal, press idle **Speaker**.
- OR -
At the single line telephone, lift the handset.
 *You hear normal Intercom dial tone.*
2. Dial **9**.
 *You hear a second, “stutter” dial tone.*
3. Dial the outside number.
 *If you hear another “stutter” dial tone, you must enter your extension ARS Authorization Code.*

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Background Music

Description

Background Music (BGM) sends music from a customer-provided music source to the speakers of the Multiline Telephone when the station is idle.

Conditions

- An ACI [PGD(2)-U10 ADP] port must be used as an alternate External Music on Hold or Background Music source when different External MOH and BGM sources are required.
- Background Music stops while the Multiline Terminal is in use.
- Originating a call, answering a voice announcement, a ringing call, or internal paging interrupts Background Music.
- Background Music is not available on single line telephones.
- ESIB(8)-U10 ETUs and ESIE(8)-U10 ETUs do not support the PGD(2)-U10 ADP.
- When a ESIB(8)-U20 ETU is installed without an ESIE(8)-U20 ETU installed, a PGD(2)-U10 ADP installed on ports 1~8 is recognized, but only channel 1 on the PGD(2)-U10 ADP is supported. Channel 2 is not available.
- When a ESIB(8)-U20 ETU has an ESIE(8)-U20 ETU installed [ESIB/E(16)], the PGD(2)-U10 ADP has both channels available for Paging, Music on Hold, External Ringers and Door Box according to strapping on ports 1~8. The corresponding port on the ESIE(8)-U20 ETU does not support a station or other hardware (it is used by the PGD(2)-U10 ADP for channel 2).
- The PGD(2)-U10 ADP can be connected only to an ESIB(8)-U20. If the PGD(2)-U10 ADP is connected to an ESIE(8)-U20 ETU, the PGD(2)-U10 ADP is NOT recognized.

Default Settings

Background Music (BGM) is allowed

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System Availability

Terminals

All Multiline Terminals

Required Component(s)

- Externally provided Music Source.
- PGD(2)-U10 ADP if different external MOH and BGM sources are required.

Related Features

Music on Hold

Programming

- ➔ **10-38-01 BGM Resource Setup – BGM Resource Type**
Configure the Background Music Source input for CPU11 (0) or ACI Port (1).
- ➔ **10-38-02 BGM Resource Setup – ACI Port Number for BGM Source (only used if PRG 10-38-01 is set to 1)**
Program the ACI Port to be used for BGM (0~96).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-30 Class of Service Options (Supplementary Service) – Background Music**
For extension Class of Service, allow (1) or prevent (0) an extension user from turning Background Music on and off.

Operation

To turn Background Music on or off:

1. Press idle **Speaker**.
2. Dial **725**.
3. Press **Speaker** to hang up.

Barge-In

Description

Barge-In permits an extension user to break into another extension user's established call, including Conference calls. This sets up a Conference-type conversation between the intruding extension and the parties on the initial call. With Barge-In, an extension user can get a message through to a busy co-worker right away.

There are two Barge-In modes: Monitor Mode (Silent Monitor) and Speech Mode. With Monitor Mode, the caller Barging In can listen to another user's conversation but cannot participate. With Speech Mode, the caller Barging In can listen and join another user's conversation.

Caution

The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversation or other sound activities, whether or not contemporaneous with transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to a telephone conversation, such as using a beep tone or other notification methods or requiring the consent of all parties to the telephone conversation, prior to monitoring or recording the telephone conversation. Some of these laws incorporate strict penalties.

Conditions

- An extension user can Barge-In on a conference.
- An extension user cannot Barge-In on an Intercom call if one of the intercom callers is using Handsfree Answerback. Both Intercom parties must lift the handset or press Speaker.
- With Program 20-13-10 set to 0, a call which is barged into can be placed on hold by the originator of the outside call. Both the outside caller and the extension that barged into the call are placed on hold.
- With Program 20-13-10 set to 1, a call which is barged into can be placed on Park by the originator of the outside call, but only the outside caller is placed in Park. The extension which barged into the call is dropped.
- Privacy blocks Barge-In attempts.
- Function keys simplify Barge-In.
- When Silent Monitor Mode is used, MIC or Feature + 1 can be used to activate speech path to the internal and external parties.

Default Setting

Disabled

System Availability

Terminals

Multiline and Single Line Terminals

Required Component(s)

None

Related Features

Call Monitoring

Conference

Hold

Intercom

Off-Hook Signaling

Park

Programmable Function Keys


Programming

- **11-12-08 Service Code Setup (for Service Access) – Barge-In**
Determine what the service code should be for an internal party to use Barge-In (default = **710**).




- ➔ **11-16-02 One-Digit Service Code Setup – Barge-In**
Set up Item 02 for single digit Barge-In. For example, you can assign Item 02 to use digit 5 for Barge-In. This allows you to program a function key with an extension number plus the Barge-In code (i.e., 5). This allows one-touch access to Barge-In for an extension.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Barge-In (code **34**).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-10 Class of Service Options (Supplementary Service) – Barge-In Monitor**
For extension Class of Service, enable the Barge-In Speech Mode (0) or Monitor Mode (1) at the initiating extension (i.e., Barge-In initiator).
- ➔ **20-13-15 Class of Service Options (Supplementary Service) – Barge-In, Initiate**
Enable (1)/Disable (0) Barge-In at the initiating extension (i.e., Barge-In initiator).
- ➔ **20-13-16 Class of Service Options (Supplementary Service) – Barge-In, Receive**
Enable (1)/Disable (0) Barge-In at the receiving extension (i.e., Barge-In receive).
- ➔ **20-13-17 Class of Service Options (Supplementary Service) – Barge-In Tone/Display (Intrusion Tone)**
Enable (1)/Disable (0) the Barge-In (Intrusion) Tone. If disabled, this also turns off the Barge-In display at the called extension.
- ➔ **20-13-32 Class of Service Options (Supplementary Service) – Multiple Barge-Ins**
Enable (1)/Disable (0) the extension user ability to have multiple users Barge-In to their conversation.
- ➔ **20-14-11 Class of Service Options for DISA/E&M – DISA/Tie Trunk Barge-In**
Enable (1)/Disable (0) a DISA or tie trunk user from using the Barge-In feature with the service code defined in Program 11-12-08.
- ➔ **20-18-07 Service Tone Timers – Intrusion Tone Repeat Time**
After a user barges in, the system repeats the Barge-In tone after this time.
- ➔ **21-01-03 System Options for Outgoing Calls – Trunk Interdigit Time (External)**
Program the time an extension must wait before the Barge-In feature can be used on a call (this time must expire before a call is put in a talk state). This time also affects Voice Over.

Operation

To Barge-In after calling a busy extension:

-  *The time in PRG 21-01-03 must expire before you can Barge-In.*
1. Call a busy extension.
 2. Press Barge-In key (PRG 15-07-01 or SC 751: 34).





To Barge-In without first calling the busy extension:

1. Pick up the handset or press **Speaker**.
2. Dial **710**.
- OR -
Press Barge-In key (PRG 15-07-01 or SC 751: 34).
3. Dial busy extension.
 -  *The extension user hears a warning tone.*
 -  *The DISA user is rerouted to the defined ring group.*
 -  *The Tie Line user hears a busy tone.***- OR -**

The following steps are not available for DISA or Tie Line trunks:

1. Dial the extension number of the busy internal party.
2. Dial the single digit service code or service code **710**.

To Barge-In to a Conference Call:

1. Pick up the handset or press **Speaker** and dial the service code (default = **710**).
 -  *If the telephone does not have the proper COS, a warning tone is sent. After the user hangs up, the system automatically places a callback to the extension.*
2. Dial the extension number or press a DSS key of a telephone in a conference call.
When a new call is added to the conference, an intrusion tone is heard by all parties in the Conference, depending on system programming, and all display multiline terminals show the joined party. If a Conference is not possible:
 -  *The extension user hears a warning tone.*
 -  *The DISA user is rerouted to the defined ring group.*
 -  *The Tie Line user hears a busy tone.*

Not available for DISA or Tie Line trunks.


- OR -**
1. Dial the extension number of the internal party.
 2. Dial the single digit service code or service code **710**.

Battery Backup – System Memory

Description

The battery on the CPUII()-U10 ETU retains the Clock/Calendar and Last Number Redial (LNR) buffers for each station when the CPUII()-U10 ETU encounters a power loss. With a fully charged battery, the settings are retained for approximately three years.

The system programmed memory (Customer Database) is stored in Non-Volatile Memory and can be erased only by performing a First Initialization.

 *For additional storage time, the database and Caller ID History can be copied to the Compact Flash card on the CPUII()-U10 ETU.*

Conditions

- The battery on the CPUII()-U10 ETU should be removed during long term storage but must be installed (protection against loss of power) just before ETU installation to provide battery backup for System Memory.
- When fully charged, the battery retains System Memory for approximately three years.
- You should replace the CPUII()-U10 ETU battery every three years.
- During normal operation, the battery is continually recharged using a built-in charging circuit from the CPUII()-U10 ETU.
- To prevent losing Caller ID History, save the database before storing the CPUII()-U10 ETU.
- Battery backup on the CPUII()-U10 ETU does not protect the following:
 - ❑ Callback
 - ❑ Off-line Status (for programming system or station assignments)
 - ❑ Repeat Redial
 - ❑ Trunk Queuing/Camp-On
 - ❑ Caller ID History

Default Settings

None

 *The battery must be installed on the CPUII()-U10 ETU prior to programming a customer database.*

System Availability

Terminals

Not applicable

Required Component(s)

None

Related Features

Battery Backup – System Power

Programming

- **90-03-01 Save Data**
1 = Save Data
Use Service code **##*9** (Default PRG 11-15-03) to save the Caller ID History from working memory to SRAM.

Operation

None

Battery Backup – System Power

Description

A built-in battery provides complete system operating power for approximately 30 minutes during commercial power outages. When optional (locally provided) batteries are connected and fully charged, full system operation can be maintained for an extended time. Actual time depends on system configuration, traffic conditions, and the battery capacity.

Conditions

- During normal operation, the batteries are continually recharged by a built-in charging circuit.
- The B64-U() KSU has batteries for system battery backup.
- External Battery Pack can be connected to the system to provide extended time during commercial power outage. Refer to the Electra Elite IPK II System Hardware Manual for further details.

Default Settings

None

System Availability

Terminals

Not applicable

Required Component(s)

B64-U() KSU

Related Features

Battery Backup – System Memory

Programming

None

Operation

None

Call Appearance (CAP) Keys

Description

This feature automatically places an outside call on a Call Appearance key when the system is operated as a hybrid (Multifunction) system. These keys can be assigned on any Multiline Terminal or the same key can appear on multiple terminals. This feature allows efficient call handling when numerous CO calls are received and a limited number of CO line key appearances are available.

After a Call Appearance (CAP) Key call is set up, the user can handle it like any other trunk call. For example, the user can place the call on hold, transfer it to a co-worker or send it to a park orbit. An incoming call is answered on the first available CAP key, beginning with the lowest numbered key. If keys 1~3 are Call Appearance (CAP) Keys, for example, the first incoming call is answered on key 1. If key 1 is busy, the next call is answered on key 2. If keys 1 and 2 are busy, the next call is answered on key 3. If all three keys are busy, additional incoming calls queue for the first available key.

Conditions

- A trunk call originated or answered at a multiline terminal must appear on a line key. The line key can be assigned as the Trunk Key, or as a Call Appearance Key. A CAP is dynamic because it is used for any trunk call. An 8-button multiline terminal can have eight CAP keys that allow the telephone to process all trunks, eight trunks at a time.
- Multiline terminals can be assigned to the same CAP Key. Trunk calls that appear on the same CAP Key at multiple stations have the same visual appearance of the call (Busy or Hold).
- Any held call left on a CAP key for more than the programmed time recalls to the multiline terminal where the call was originally put on hold.
- When a multiline terminal (other than the one that originally initiated or received a call) is used to retrieve a held call, the SMDR records a transfer to the multiline terminal where the call was retrieved.
- Only outside lines use a CAP key.
- A multiline terminal can have multiple CAP keys assigned to it.
- Outside lines reside on the CAP key in the order of lowest to highest line key number on the station. For instance, when line keys 1, 2 and 3 are CAP keys, the first call resides on line key 1, the second call resides on line key 2 and third call resides on line key 3.

- All Flexible Line keys on a multiline terminal can be assigned as CAP keys in System Programming.
- A conference call involving two outside lines cannot reside on one Call Appearance key.
- For Call Appearance (CAP) Keys, trunks must be assigned to trunk group 1 or higher (Program 14-05-01). Trunk Group 0 indicates KF (Key Function) mode.
- CAP Keys can be programmed from 0001~9999. 0000 assigns the next available CAP Key.
- Trunk Group (*02), Virtual Extension (*03) and Call Appearance (CAP) Key (*08), codes cannot be programmed on a DSS Console as the system does not allow entry of the additional data required.
- If you have a trunk line key and Call Appearance (CAP) Key for the same trunk, the line key has priority. An incoming call rings the trunk line key and when answered, the trunk line keys lights, not the CAP Key. When you access the trunk for an outgoing call, the Trunk line key lights, not the Call Appearance (CAP) Key.

Default Setting

Disabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Automatic Route Selection

Call Arrival (CAR) Keys

Central Office Calls, Answering

Central Office Calls, Placing

Off-Hook Signaling

Programmable Function Keys

Secondary Incoming Extension

User Programming Ability

Virtual Extensions

Programming

- ➔ **14-05-01 Trunk Group – Trunk Group Number**
Assign trunks to trunk groups (1~100).
- ➔ **15-07-01 Programmable Function Keys**
Assign a key as a CAP key (code *8) plus a CAP orbit number (01~64).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to extensions.
- ➔ **20-11-19 Class of Service Options (Hold/Transfer Service) – Hold/Extended Park**
Determine whether an extension Class of Service should allow normal (0) for PRG 24-01-06, or extended Park (1) for PRG 24-01-07).
- ➔ **24-01-02 System Options for Hold – Hold Recall Callback Time**
A call parked longer than the programmed time recalls the extension where it was initially parked.
- ➔ **24-01-06 System Options for Hold – Park Hold Time - Normal**
Set the Park Hold Time (0~64800 seconds). A call parked longer than the programmed time recalls the extension where it was initially parked.
Refer to [Flexible System Numbering on page 2-447](#) for setting Flexible Timeouts for Class of Service.
- ➔ **24-01-07 System Options for Hold – Park Hold Time - Extended (Recall)**
Set the Extended Park Hold Time (0~64800 seconds). A call parked longer than the programmed time recalls the extension where it was initially parked.

Operation

To place an outgoing call on hold and retrieve it using a multiline terminal:

1. Use the handset to go off-hook, and wait for internal dial tone.
- OR -
Press **Speaker** and wait for internal dial tone.
2. Dial the Trunk Access Code (default: **9**).
3. Dial the outside party (the Call Appearance key lights). Begin your conversation.
4. Press **Hold** (the Call Appearance key flashes).
5. Press the flashing **Call Appearance** key to retrieve the call.

To receive an incoming call, put it on hold and then retrieve it using a Multiline Terminal:

1. Receive CO/PBX incoming ring.
2. Use the handset to go off-hook, or press **Speaker** (the Call Appearance key lights). Talk with outside party.
3. Press **Hold** (the Call Appearance key flashes).
4. Press the flashing **Call Appearance** key to retrieve the call.

Call Arrival (CAR) Keys

Description

Call Arrival (CAR) Keys are software extensions available on the Basic and Expanded Port Packages. A Call Arrival Extension assigned to a line key, can appear and ring on an individual station or multiple stations. Call Arrival Keys are busy only in the ringing state and are not used during the talking state.

Call Arrival Keys are shared with the Virtual Extensions (VE). In virtual extension mode, the key acts as a secondary extension. Up to 256 CAR/VE keys are provided.

Conditions

- There are 256 available ports/extensions shared between CAR keys and virtual extensions.
- The 256 available ports/extensions are assigned per extension for CAR key mode or virtual extension (VE) key mode.
- More than one extension can share a CAR key.
- An extension can have more than one CAR key assigned.
- Up to 32 incoming calls can be queued to a busy CAR key.
- If multiple CAR/SIE/VE keys are ringing on a station at the same time, the CAR/SIE/VE key on the lowest Line Key is answered first.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Virtual Extensions

Programming

- **11-04-01 Virtual Extension Numbering**
Virtual Extension Ports (V-Ports) = 1~256
Virtual Extension Number = Extension Number
- **15-01-01 Basic Extension Data Setup – Extension Name**
Assign the extension name.
- **15-01-05 Basic Extension Data Setup – Restriction for Outgoing Disable on Incoming Line**
Enable (1)/Disable (0) supervised dial detection for an extension.
- **15-02-07 Multiline Telephone Basic Data setup – Automatic Hold for CO Lines**
0 = Hold
1 = Disconnect
- **15-02-21 Multiline Telephone Basic Data Setup – Virtual Extension Access Mode (when idle Virtual Extension key pressed)**
Virtual Extension Key Mode 0 = DSS (default)
1 = OTG (Outgoing)
2 = Ignore
- **15-07-01 Programmable Function Keys**
KYxx = Key Number 01~32
KY01 = *03 (*01 by default)
- **15-08-01 Incoming Virtual Extension Ring Tone Setup**
ICM Tone Pattern,
0 = Pattern 1 (default)
1 = Pattern 2
2 = Pattern 3
3 = Pattern 4
- **15-09-01 Virtual Extension Ring Assignment**
Mode1: 0 = No (default)
1 = Yes
- **15-10-01 Incoming Virtual Extension Ring Tone Order Setup**
Order 1 Pattern 0 = Pattern 1 (default)
Order 2 Pattern 1 = Pattern 2 (default)
Order 3 Pattern 2 = Pattern 3 (default)
Order 4 Pattern 3 = Pattern 4 (default)

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- ➔ **15-11-01 Virtual Extension Delayed Ring Assignment**
KY01 Mode1: 1 = Yes
0 = No

 - ➔ **15-18-01 Virtual Extension Key Enhanced Options – Virtual Extension Key Operation Mode**
0 = Release
1 = Land On

 - ➔ **15-18-02 Virtual Extension Key Enhanced Options – Display Mode When Placing a Call on Virtual Extension Key**
0 = Virtual Extension
1 = Real Extension

 - ➔ **20-04-03 System Options for Virtual Extensions – CAR/SIE/Virtual Extension Delay Interval**
Delay = 10 sec. (default)

 - ➔ **20-06-01 Class of Service for Extensions**
Mode1: Class No. = 1 (default)

 - ➔ **20-07-10 Class of Service Options (Administrator Level) – Programmable Function Key Programming (Appearance Level)**
1 = On (default)
0 = Off

 - ➔ **20-10-08 Class of Service Options (Answer Service) – Virtual Extension Off-Hook Answer**
0 = Off
1 = On (default)

 - ➔ **20-13-27 Class of Service Options (Supplementary Service) – Busy on Seizing Virtual Extension**
1 = On (default)
0 = Off

 - ➔ **21-01-15 System Options for Outgoing Calls – Outgoing Disable on Incoming Line (Toll Restriction)**
0 = Off (default)
1 = On


 - ➔ **23-04-01 Ringing Line Preference for Virtual Extensions**
Order 1 Stg Group = 0~64 (0 by default)
Order 2 Stg Group = 0~64 (0 by default)
Order 3 Stg Group = 0~64 (0 by default)
Order 4 Stg Group = 0~64 (0 by default)

Operation


To answer a call ringing a Call Arrival (CAR) Key:

1. Press the flashing Call Arrival (CAR) Key.

To place a call to a Call Arrival (CAR) Key:

1. Lift the handset, or press **Speaker**.
2. Dial the CAR key extension, or press the Call Arrival (CAR) Key.
 *The operation depends on the setting in PRG 15-02-21.*

To program a Call Arrival (CAR) Key on a telephone:

1. Press **Speaker**.
2. Dial **752**.
3. Press the key you want to program.
4. Dial ***03**.
5. Dial the number of the extension you want to appear on the key.
6. Press **Hold** once for Immediate Ring
 *To set for Delayed Ring, skip to Step 8.*
7. Dial the Mode number in which the key rings.

1 = Day 1	5 = Day 2
2 = Night 1	6 = Night 2
3 = Midnight 1	7 = Midnight 2
4 = Rest 1	8 = Rest 2

8. Press **Hold** to set up Delayed Ring.

- OR -

Skip to Step 10.

9. Dial the mode number in which the key delay rings.

1 = Day 1	5 = Day 2
2 = Night 1	6 = Night 2
3 = Midnight 1	7 = Midnight 2
4 = Rest 1	8 = Rest 2

10. Press **Speaker**.

Call Duration Timer

Description

Call Duration Timer lets a multiline terminal with an LCD time their trunk calls on the telephone display. This helps users that must keep track of their time on the telephone. For incoming trunk calls, the Call Time begins as soon as the user answers the call.

Conditions

- The Call Timer starts over each time the call is retrieved from Hold or Park.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals with an LCD

Required Component(s)

None

Related Features


Alphanumeric Display

Programming

- ➔ **20-13-36 Class of Service Options (Supplementary Service) – Call Duration Timer Display**
Enable (1)/Disable (0) an extension Call Timer.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-09-06 Class of Service Options (Incoming Call Service) – Incoming Time Display**
If this option is set to 1, the Incoming Call Time is displayed on the multiline terminal LCD while the telephone is ringing.
- ➔ **21-01-03 System Options for Outgoing calls – Trunk Interdigit Time (External)**
The system waits for this time to expire before starting the Call Timer.

Operation

To time your trunk calls:

1. Place a trunk call.
 -  *The timer starts automatically.*

Call Forwarding

Enhancements

Version 1500 or higher supports:

- Call Forwarding when DND is set.
- Call Forwarding when an IP telephone loses connection.
- Call Forwarding Function keys (Busy, No Answer, Busy/No Answer, and Immediate) light solid instead of flash when the key forward type is set.
- DSS/BLF displays when a telephone is off-hook (Busy) when Call Forward Immediate is set.

Description

Call Forwarding permits an extension user to redirect their calls to another extension or an off-premise number. Call Forwarding ensures that the user's calls are covered when they are away from their work area. The types of Call Forwarding are:

- Call Forwarding when Busy or Unanswered**
Calls to the extension forward when busy or unanswered.
- Call Forwarding – Centrex**
When using PBX/Centrex trunks, calls to the extension perform a Centrex transfer using Immediate, Busy and No Answer Forwarding. (**Version 1500 or higher** is required to support this operation.)
- Call Forwarding Immediate**
All calls forward immediately to the destination, and only the destination rings.
- Call Forwarding with both Ringing**
All calls forward immediately to the destination, and both the destination and the forwarded extension ring (not for Voice Mail).
- Call Forwarding when Unanswered**
Calls forward only if they are unanswered (Ring No Answer).
- Call Forwarding Follow Me**
Refer to [Call Forwarding with Follow Me on page 2-149](#) for more information.

❑ Personal Answering Machine Emulation

Allows the extension to emulate an answering machine. Refer to Voice Mail for more information.

Call Forwarding reroutes calls ringing an extension, including calls transferred from another extension. Call Forwarding can also be split, allowing internal and external calls to forward to different destinations. The extension user can enable Call Forwarding from their telephone. An extension user can also set the forwarding for another extension by using Call Forward for any Extension to Destination. To redirect calls while a user is at another telephone, use Call Forwarding with Follow Me. A periodic VRS announcement can remind users that their calls are forwarded.

Conditions

- Virtual Extensions can be set to Call Forward. PRG 15-02-21 must be set to 1, to allow the Virtual Extension to place outgoing calls.
- If an extension in a call forward chain has Call Forward with Both Ring or Call Forward with Follow Me set, calls do not continue routing to other extensions in the chain.
- Call Forwards can be chained allowing calls to forward from one extension to the next. Up to 32 extensions can be linked in a call forward chain.
- Periodic reminder message requires a DSP daughter board for Voice Response System (VRS).
- Call Forwarding an extension in a Department Group prevents that extension from receiving Department Pilot Calls.
- Ring Groups do not follow Call Forwarding.
- Call Forward Split does not allow Call Forward with Follow Me.
- If Call Forwarding off premise, a trunk access code must be included in the forwarding number.
- Call Forward with Follow Me allows a single station to set follow me for multiple stations. When canceling Call Forward with Follow Me, the user must specify the station to cancel or cancel all.
- The telephone must be idle to enable call forwarding with a Programmable Function Key, or receive dial tone to enable call forwarding with a service code.
- Call Forward for any Extension to Destination cannot be set or canceled from a Virtual Extension.
- Call Forwarding/Do Not Disturb Override allows Overriding a Call Forwarding or DND setting at another extension.
- When a call is transferred because of Call Forwarding No Answer, Call Forwarding Busy, or DND, the Reason for Transfer option is displayed to the transferred extension.
- An extension user can forward their calls to a Department number.

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- A DSS key indicates a Call Forwarding indication for extensions.
 - When DND and Call Forward are set on the same telephone, call forwarding does not work. (**Version 1100 or lower.**)
 - When DND and Call Forward are set on the same telephone, call forwarding works. If Busy and No Answer Forwarding are set to different locations, it follows the Busy forwarding. (**Version 1500 or higher required.**)
 - When DND and any Call Forwarding is set, the call forwards immediately. (**Version 1500 or higher required.**)
 - Function keys simplify Call Forwarding operation.
 - If an extension Class of Service denies Call Forwarding (PRG 20-11-01~PRG 20-11-05, off), the extension user can still dial the service code to Set/Cancel Call Forwarding, but cannot set any data.
 - Call Forward Both Ring Split does not work to an off-premise destination.
 - If an IP telephone has forwarding set and then loses connection, it follows the forwarding. (**Version 1500 or higher required.**)
 - If an IP phone has Busy and No Answer Forwarding set to different locations and it loses connection, it follows No Answer forwarding. (**Version 1500 or higher required.**)
 - If you do the following in sequence,
 - Set Call Forwarding Busy/No Answer on extension.
 - Set Call Forwarding Immediate on extension.
 - Cancel Call Forwarding Immediate on extension.then,
Call Forwarding Busy/No Answer is set back on extension
 - If you do the following in sequence,
 - Set Call Forwarding No answer on extension.
 - Set Call Forwarding Immediate on extension.
 - Cancel Call Forwarding Immediate on extension.then,
All Call Forwarding is canceled
 - ✎ Settings in Programs 24-09-04 and 24-09-05 are copied to Programs 24-09-02 and 24-09-03 and changed to Call Forwarding Busy/No Answer.

- If you do the following in sequence,
 - Set Call Forwarding Busy on extension.
 - Set Call Forwarding Immediate on extension.
 - Cancel Call Forwarding Immediate on extension.
- then,
Call Forwarding Busy/No Answer is set back on extension

Default Setting

Enabled

System Availability

Terminals

Any Station and Virtual Extensions

Required Component(s)

None



Related Features

Call Forwarding, Off-Premise

Call Forwarding with Follow Me

Call Forwarding/Do Not Disturb Override

Central Office Calls, Answering

Department Calling

Direct Station Selection (DSS) Console

Do Not Disturb

Programmable Function Keys

Voice Response System (VRS)

Programming

Program Number	Name	Defaults
11-10-06	Setting the Automatic Transfer for Each Trunk Line	733
11-10-07	Canceling the Automatic Transfer for Each Trunk Line	734
11-10-08	Setting the Destination for Automatic Trunk Transfer	735
11-10-18	Off-Premise Call Forward by Door Box	722
11-11-01	Call Forward – All	741
11-11-02	Call Forward – Busy	742
11-11-03	Call Forward – No Answer	743
11-11-04	Call Forward – Busy/No Answer	744
11-11-05	Call Forward – Both Ring	745
11-11-07	Call Forwarding – Follow Me	746
11-11-08	Do Not Disturb	747
11-11-45	Set/Cancel Call Forwarding All Split	No Setting
11-11-46	Set/Cancel Call Forwarding Busy Split	No Setting
11-11-47	Set/Cancel Call Forwarding No Answer Split	No Setting
11-11-48	Set/Cancel Call Forwarding Busy/No Answer Split	No Setting
11-11-49	Set/Cancel Call Forwarding Both Ring Split	No Setting
11-11-52	Set/Cancel Call Forward Immediate for any Extension to Destination	790
11-11-53	Set/Cancel Call Forward Busy for any Extension to Destination	791
11-11-54	Set/Cancel Call Forward No Answer for any Extension to Destination	792
11-11-55	Call Forward Busy No Answer Destination (No Split)	793
11-11-58	Call forward with personal greeting	713
11-12-01	Bypass Call	707
11-16-06	DND/Call Forward Override Bypass	No Setting

- ➔ **15-07-01 Programmable Function Keys**
Assign a Function Key for Call Forwarding setup code (code 10~15).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

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- ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
Turn off or on an extension ability to receive a second call from a DID, DISA, DIL, or tie line caller.
 - ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Turn off or on an extension user ability to have calls queued if a call rings the extension when it is busy.
 - ➔ **20-11-01 Class of Service Options (Hold/Transfer Service) – Call Forward All**
Enable (1)/Disable (0) an extension user ability to set Call Forward Immediate.
 - ➔ **20-11-02 Class of Service Options (Hold/Transfer Service) – Call Forward When Busy**
Enable (1)/Disable (0) an extension user ability to set Call Forwarding when Busy.
 - ➔ **20-11-03 Class of Service Options (Hold/Transfer Service) – Call Forwarding When Answered**
Enable (1)/Disable (0) an extension user ability to set Call Forwarding when Unanswered.
 - ➔ **20-11-04 Class of Service Options (Hold/Transfer Service) – Call Forwarding (Both Ringing)**
Enable (1)/Disable (0) an extension user ability to set Call Forwarding with Both Ringing.
 - ➔ **20-11-05 Class of Service Options (Hold/Transfer Service) – Call Forwarding with Follow Me**
Enable (1)/Disable (0) an extension user ability to set Call Forward with Follow Me.
 - ➔ **20-11-23 Class of Service Options (Hold/Transfer Service) – CAR/VE Call Forward Set/Cancel**
Enable (1)/Disable (0) an extension user ability to set and cancel Call Forwarding for a CAR or Virtual Extension.
 - ➔ **20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
Turn off or on an extension ability to receive Off-Hook signals.
 - ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension to manually (0) or automatically (1) receive Off-Hook signals.
 - ➔ **24-02-03 System Options for Transfer – Delayed Call Forwarding Time**
Set the Delayed Call Forwarding time. For an unanswered call, Call Forward No Answer occurs after this time.
 - ➔ **24-09-01 through 24-09-07 Call Forward Split Settings**
Assign Call Forwarding type and the destination numbers for each extension/virtual extension.

Operation



To set Call Forward – Immediate at a forwarding station:

1. Pick up the handset or press **Speaker**.

2. Dial the Call Forward – Immediate Service Code (default: 741).

- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** Keys.
(PRG 15-07-01, 10 or SC 751, Key Code 10)


3. Dial **1** (Set).
4. Dial the destination extension or off-premise number.
5. Press **Speaker** or hang up.
 -  Refer to [Call Forwarding, Off-Premise on page 2-133](#).
 -  The Call Forwarding Programmable Function Key lights.

To cancel Call Forward – Immediate at a forwarding station:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward – Immediate Service Code (default: 741).

- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** keys.
(PRG 15-07-01, 10 or SC 751, Key Code 10)

3. Dial **0** (Cancel).
4. Press **Speaker** or hang up.
 -  The Call Forwarding Programmable Function Key turns off.

To set Call Forward – Busy/No Answer at a forwarding station:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward – Busy/No Answer Service Code (default: 744).


- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** keys.
(PRG 15-07-01, 13 or SC 751, Key Code 13)

3. Dial **1** (Set).
4. Dial the destination extension or off-premise number.

5. Press **Speaker** or hang up.

 Refer to *Call Forwarding, Off-Premise on page 2-133*.

 *The Call Forwarding Programmable Function Key turns on.*


To cancel Call Forward – Busy/No Answer at a forwarding station:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward – Busy/No Answer Service Code (default: 744).

- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** keys.
(PRG 15-07-01, 13 or SC 751, Key Code 13)

3. Dial **0** (Cancel).
4. Press **Speaker** or hang up.

 *The Call Forwarding Programmable Function Key turns off.*


To set Call Forward – Both Ring at a forwarding station:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward – Both Ring Service Code (default: 745).

- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** keys.
(PRG 15-07-01, 14 or SC 751, Key Code 14)

3. Dial **1** (Set).
4. Dial the destination extension number.
5. Press **Speaker** or hang up.

 *The Call Forwarding Programmable Function Key turns on.*


To cancel Call Forward – Both Ring at a forwarding station:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward – Both Ring Service Code (default: 745).

- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** keys.
(PRG 15-07-01, 14 or SC 751 Key Code 14)

3. Dial **0** (Cancel).
4. Press **Speaker** or hang up.

 *The Call Forwarding Programmable Function Key turns off.*


To set Call Forward – Follow Me from the destination station:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward – Follow Me Service Code (default: 746).

- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** keys.
(PRG 15-07-01, 10 or SC 751, Key Code 15)

3. Dial **1** (Set).
4. Dial the station number to be forwarded and then the destination number.
5. Press **Speaker** or hang up.

 *The Call Forwarding Programmable Function Key comes on.*


To cancel Call Forward – Follow Me from the destination station:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward – Follow Me Service Code (default: 746).

- OR -

At the multiline terminal only, press the **Call Forwarding Programmable Function** keys.
(PRG 15-07-01, 10 or SC 751, Key Code 15)

3. Dial **0** (Cancel).
4. Dial the station number, which is forwarded, or **0** to cancel all extensions.
5. Press **Speaker** or hang up.

 *The Call Forwarding Programmable Function Key goes off.*

To set Call Forward Immediate for any Extension to Destination:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward Immediate for any Extension to Destination Service Code (Default: 790).
3. Dial **1** (Set).
4. Dial the extension number to be forwarded and then the destination number.
5. Press **Speaker** or hang up.

To cancel Call Forward Immediate for any Extension:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward Immediate for any Extension to Destination Service Code (default: 790).

3. Dial **0** (Cancel).
4. Dial the station number which is forwarded.
5. Press **Speaker** or hang up.

To set Call Forward Busy/No Answer for any Extension to Destination:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward Busy/No Answer for any Extension to Destination Service Code (default: 793).
3. Dial **1** (Set).
4. Dial the extension number to be forwarded and then the destination number.
5. Press **Speaker** or hang up.

To cancel Call Forward Busy/No Answer for any Extension to Destination:

1. Pick up the handset or press **Speaker**.
2. Dial the Call Forward Busy/No Answer for any Extension to Destination Service Code (default: 793).
3. Dial **0** (Cancel).
4. Dial the station number, which is forwarded.
5. Press **Speaker** or hang up.

To set Call Forward – Immediate using a Virtual Extension:

1. Press the idle **Virtual Extension** key.
2. Dial the Call Forward – Immediate Service Code (default: 741).
3. Dial **1** (Set).
4. Dial the destination extension or off-premise number.
5. Press **Speaker** or hang up.

 Refer to *Call Forwarding, Off-Premise* on page 2-133.

To cancel Call Forward – Immediate at a forwarding station:

1. Press the idle **Virtual Extension** key.
2. Dial the Call Forward – Immediate Service Code (default: 741).
3. Dial **0** (Cancel).

4. Press **Speaker** or hang up.

To set Call Forward – Busy/No Answer using a Virtual Extension:

1. Press the idle **Virtual Extension** key.
2. Dial the Call Forward – Busy/No Answer Service Code (Default: 744).
3. Dial **1** (Set).
4. Dial the destination extension or off-premise number.
5. Press **Speaker** or hang up.

 Refer to [Call Forwarding, Off-Premise on page 2-133](#).

To cancel Call Forward – Busy/No Answer using a Virtual Extension:

1. Press the idle **Virtual Extension** key.
2. Dial the Call Forward – Busy/No Answer Service Code (default: 744).
3. Dial **0** (Cancel).
4. Press **Speaker** or hang up.

Call Forwarding – Centrex

Enhancements

This feature added with Version 1500 .

Description

The Call Forwarding – Centrex feature allows a station to forward an incoming PBX/Centrex CO call to an outside location using the same PBX/Centrex CO line to free the line for additional use.

Call Forwarding – Centrex supports the following:

- Call Forward – Immediate
- Call Forward – Busy
- Call Forward – No Answer
- Call Forward – Busy/No Answer

Conditions

- Call Forwarding – Centrex calls transferred from another station are forwarded when the transferred Trunk is assigned as PBX in Program 14-04-01.
- The following incoming calls follow Call Forwarding – Centrex when the incoming trunk is a PBX/Centrex trunk:
 - DIT/ANA
 - Station Transfer
 - Automated Attendant Transfer
 - DISA Calls
- Call Forwarding – Centrex is not supported for Call Forward Both Ring Split.
- A maximum of 24 digits can be assigned in the destination for Call Forwarding – Centrex.
- When a trunk is set to CTX/PBX, and is set for Call Forwarding – Centrex to an incorrect number, the call recalls and follows CO incoming ringing (i.e., DIL, Normal Ring Group Programming).
- When Call Forwarding – Centrex is set and all trunks are changed in PRG 14-04-01 from PBX to Trunk, Call Forward is cleared from memory.

- When DND and any Call Forwarding – Centrex is set, the call forwards immediately. **(Version 1500 or higher required.)**
- Call Forwarding – Centrex does not follow the Code Restriction of the stations.

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

PBX/Centrex CO line

Related Features

Call Forwarding

Call Arrival (CAR) Keys

Code Restriction

Direct Inward Dialing (DID)

Do Not Disturb

Door Box

PBX Compatibility

Virtual Extensions

Voice Response System (VRS)

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**
Allocate the circuits on the CPUII ETUs for either DTMF receiving or dial tone detection. PRG 14-01-13 Basic Trunk Data Setup – Loop Supervision. Enable (1) loop supervision for each trunk that should be able to use Call Forwarding – Centrex.
- ➔ **11-11-45 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward All (Split)**
Assign the Call Forward All Split Service Code (default not assigned).
- ➔ **11-11-46 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward Busy (Split)**
Assign the Call Forward Busy Split Service Code (default not assigned).
- ➔ **11-11-47 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward No Answer (Split)**
Assign the Call Forward No Answer Split Service Code (default not assigned).
- ➔ **11-11-48 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward Busy/No Answer (Split)**
Assign the Call Forward Busy No Answer Split Service Code (default not assigned).
- ➔ **14-04-01 Behind PBX Setup – Type of Connection**
Indicate if the trunk is installed behind a PBX (1) or not (0). There is one item for each Night Service Mode.
- ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
Turn off or on an extension ability to receive a second call from a DID, DISA, DIL, or tie line caller.
- ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Turn off or on an extension user ability to have calls queued if a call rings the extension when it is busy.
- ➔ **20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
Turn off or on an extension ability to receive Off-Hook signals.
- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension user ability to manually (0) or automatically (1) receive Off-Hook signals.
- ➔ **24-02-13 System Options for Transfer – Hook Flash Sending Timer When the System Answers Automatically (Version 1600 and 2100 or higher required)**
Set the time the system waits before sending the hookflash for the Centrex Transfer after answering the call.
0~64800 (sec) (default: 2)
- ➔ **24-09-06 Call Forward Split Settings – Call Forwarding Destination for CTX/PBX for All Call, No Answer (Version 1500 or higher required)**
0~9, #, *, P, R, @ (up to 24 digits)

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- ➔ 24-09-07 **Call Forward Split Settings – Call Forwarding Destination for CTX/PBX for Busy (Version 1500 or higher required)**
0~9, #, *, P, R, @ (up to 24 digits)

Operation

To activate Call Forwarding – Centrex:

1. At a multiline terminal, press **Speaker**.
- OR -
At a single line telephone, lift the handset.
2. Dial the Call Forwarding Split Service Code (default not assigned).
3. Dial **3** (CTX/PBX).
4. Dial **1** (Set).
5. Dial number to Centrex Forward to.
6. Hang up.

To cancel Call Forwarding – Centrex:

1. At a multiline terminal, press **Speaker**.
- OR -
At a single line telephone, lift the handset.
2. Dial the Call Forward Split Service Code (default not assigned).
3. Dial **0** (Cancel).
4. Dial **3** (CTX/PBX) or **0** (All).

Call Forwarding/Do Not Disturb Override

Description

An extension user can override Call Forwarding or Do Not Disturb at another extension. This is helpful, for example, to dispatchers and office managers that always need to get through.

Conditions

None

Default Setting

Disabled

System Availability

Terminals

Any Station

Required Component(s)

None

Related Features

Programmable Function Keys

Programming

- ➔ 11-12-01 **Service Code Setup (for Service Access) – Bypass Call**
Customize the Service Code which is to be used for Call Forwarding/DND Override (default: 707).

- ➔ **11-16-06 Single Digit Service Code Setup – DND/Call Forward Override Bypass**
Customize the 1-digit Service Code used for DND/Call Forward Override.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Call Forwarding/Do Not Disturb Override (code 37).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-04 Class of Service Options (Supplementary Service) – Call Forward/DND Override (Bypass Call)**
Enable (1)/Disable (0) an extension user ability to initiate Call Forwarding/DND Override.

Operation

To override an extension Call Forwarding or Do Not Disturb:

1. Call the forwarded or DND extension.
2. Press the Override key (PRG 15-07 or SC 751: 37).

Call Forwarding, Off-Premise

Description

Off-Premise Call Forwarding allows an extension user to forward their calls to an off-site location. By enabling Call Forward, Off-Premise, the user can stay in touch by having the system forward their calls while they are away from the office. The forwarding destination can be any telephone number the user enters, such as a mobile phone, home office, hotel or meeting room. Off-Premise Call Forwarding can route the off-site telephone number over a specific trunk or through a trunk group, Automatic Route Selection or Trunk Group Routing.

Off-Premise Call Forwarding reroutes the following types of incoming calls:

- Ringing intercom calls from co-worker's extensions
- Calls routed from the VRS or Voice Mail ¹
- Direct Inward Lines ¹
- DISA, DID and Tie Line calls to the forwarded extension ¹
- Transferred calls ¹

Off-Premise Call Forwarding does not reroute Call Arrival (CAR) Keys, Call Arrival (CAR) Keys, or Ring Group calls (i.e., trunk ringing according to Ring Group assignments made in PRG 22-04 and PRG 22-05).

Conditions

- If a call that forwards Off-Premise goes out on a trunk assigned as TIE or DID, and the called party does not answer before the timer in PRG 34-07-05, the call recalls to the station that performed the transfer.
- Call Forwarding Off-Premise requires either loop start trunks with disconnect supervision or ground start trunks.
- The trunk access code and the outside telephone number combined cannot exceed 24 digits.
- Call Forwarding an extension in a Department Group prevents that extension from receiving Department Pilot Calls.
- If a Programmable Function key is not defined for Call Forwarding (10~17), the DND key flashes to indicate that the extension is call forwarded.

1. Off-Premise Call Forwarding can reroute an incoming trunk call only if the outgoing trunk selected has disconnect supervision enabled (refer to the Programming section).

- DID calls to an extension with Off-Premise Call Forwarding set do not recall if there is no answer.
- Calls to extensions with DND active do not follow Call Forwarding programming. DIL calls ring an idle Department Group member, then follow PRG 22-08 programming then PRG 22-05 programming.
- Door Boxes must be programmed for the calls to be transferred Off-Premise.
- The outside number Call Forwarding dials must be a number normally allowed by the forwarded extension Toll Restriction.
- In systems with a DSP daughter board for VRS, callers to an extension forwarded off-premise hear, "Please hold on, your call is being rerouted." This option can be disabled in PRG 40-10-01.
- When DND and any Call Forwarding, Off-Premise is set, the call forwards immediately. **(Version 1500 or higher required.)**

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Call Arrival (CAR) Keys

Call Forwarding

Code Restriction

Direct Inward Dialing (DID)

Do Not Disturb

Door Box

Virtual Extensions

Voice Response System (VRS)

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**
Allocate the circuits on the CPU11 ETUs for either DTMF receiving or dial tone detection. PRG 14-01-13 Basic Trunk Data Setup – Loop Supervision Enable (1) loop supervision for each trunk that should be able to use Call Forwarding Off-Premise.
- ➔ **11-11-01 Service Code Setup (for Setup/Entry Operation) – Call Forward – All**
Assign the Call Forward – All Service Code (default: 741).
- ➔ **11-11-02 Service Code Setup (for Setup/Entry Operation) – Call Forward – Busy**
Assign the Call Forward – Busy Service Code (default: 742).
- ➔ **11-11-03 Service Code Setup (for Setup/Entry Operation) – Call Forward – No Answer**
Assign the Call Forward – No Answer Service Code (default: 743).
- ➔ **11-11-04 Service Code Setup (for Setup/Entry Operation) – Call Forward – Busy/No Answer**
Assign the Call Forward – Busy No Answer Service Code (default: 744).
- ➔ **11-11-05 Service Code Setup (for Setup/Entry Operation) – Call Forward – Both Ring**
Assign the Call Forward – Both Ring Service Code (default: 745).
- ➔ **11-11-45 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward All (Split)**
Assign the Call Forward All (Split) Service Code (default not assigned).
- ➔ **11-11-46 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward Busy (Split)**
Assign the Call Forward Busy (Split) Service Code (default not assigned).

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- ➔ **11-11-47 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward No Answer (Split)**
Assign the Call Forward No Answer (Split) Service Code (default not assigned).
 - ➔ **11-11-48 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward Busy/No Answer (Split)**
Assign the Call Forward Busy No Answer (Split) Service Code (default not assigned).
 - ➔ **11-11-49 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward Both Ring (Split)**
Assign the Call Forward Both Ring (Split) Service Code (default not assigned).
 - ➔ **11-11-52 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward All Destination (No Split)**
Assign the Call Forward All for any Extension Service Code (default: 790).
 - ➔ **11-11-53 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward Busy Destination (No Split)**
Assign the Call Forward Busy for any Extension Service Code (default: 791).
 - ➔ **11-11-54 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward No Answer Destination (No Split)**
Assign the Call Forward No Answer for any Extension Service Code (default: 792).
 - ➔ **11-11-55 Service Code Setup (for Setup/Entry Operation) – Set/Cancel Call Forward Busy/No Answer Destination (No Split)**
Assign the Call Forward Busy No Answer for any Extension Service Code (default: 793).
 - ➔ **15-07-01 Programmable Function Keys**
 - ➔ **20-06-01 Class of Service for Extensions**
Assign Class of Service (1~15) to an extension.
 - ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
Turn off or on the extension ability to receive a second call from a DID, DISA, DIL, or tie line caller.
 - ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Turn off or on an extension user ability to have calls queued if a call rings the extension when it is busy.
 - ➔ **20-11-12 Class of Service Options (Hold/Transfer Service) – Call Forwarding Off-Premise (External Call Forwarding)**
Enable (1)/Disable (0) setting up Call Forwarding Off-Premise at the extension.
 - ➔ **20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
Turn off or on an extension ability to receive Off-Hook signals.
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- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension to manually (0) or automatically (1) receive Off-Hook signals.
 - ➔ **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
Timer starts when a trunk begins talking with another trunk (for example: Trunk-to-Trunk Transfer, outgoing from trunk, Tandem Trunking) (0~64800 seconds, default: 1800). When this time expires, a warning tone is heard.
 - ➔ **24-09-01 through 24-09-07 Call Forward Split Settings**
Assign Call Forwarding type and the destination numbers for each extension/virtual extension.
 - ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**
Determine the time a DISA caller or any Trunk-to-Trunk (such as Tandem Trunking) conversation can talk before the Long Conversation tone is heard (0~64800 seconds). (default: 3600)
 - ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
Determine the time the system waits before disconnecting a DISA or any trunk-to-trunk (such as Tandem Trunking) call after the Long Conversation tone is heard (0~64800 seconds). (default: 10)


Trunk-to-Trunk Forwarding – Normal (0) Trunks

- ➔ **11-10-06 Service Code Setup (for System Administrator) – Setting the Automatic Transfer for Each Trunk Line**
Customize the service code used to set the Automatic Trunk Forwarding feature (default: 733).
- ➔ **11-10-07 Service Code Setup (for System Administrator) – Canceling the Automatic Transfer for Each Trunk Line**
Customize the service code used to cancel the Automatic Trunk Forwarding feature (default: 734).
- ➔ **11-10-08 Service Code Setup (for System Administrator) – Setting the Destination for Automatic Trunk Transfer**
Customize the service code used to set the destination for the Automatic Trunk Forwarding feature (default: 735).
- ➔ **13-01-01 Speed Dialing Option Setup – Speed Dialing Auto Outgoing Call Mode**
Determine if dialing a Speed Dialing number dials an outside number (seizing a trunk as assigned in Program 13-05) or an Intercom number (0 = Trunk Dialing Mode, 1 = Extension Dialing Mode).
- ➔ **13-04-01 Speed Dialing Number and Name – Speed Dialing Data**
Enter the Common and Group Speed Dialing numbers and names which are to be used for Trunk-to-Trunk Forwarding.
- ➔ **14-01-26 Basic Trunk Data Setup – Automatic Trunk-to-Trunk Transfer Mode**
Enable (1)/Disable (0) trunk ability to use Step Transfer.

- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-07-05 Class of Service Options (Administrator Level) – Automatic Trunk-to-Trunk Transfer**
Enable (1)/Disable (0) an extension user ability to use the Trunk-to-Trunk Forwarding service codes.
- ➔ **22-02-01 Incoming Call Trunk Setup – Incoming Type**
Assign the incoming type for each trunk. There is one item for each Mode. When using Trunk-to-Trunk Forwarding the trunks must be set for Normal (0).
- ➔ **24-02-11 System Options for Transfer – No Answer Step Transfer**
Assign the time each transfer destination rings before step transfer is performed.
- ➔ **24-02-12 System Options for Transfer –No Answer Trunk-to-Trunk Transfer**
Define the elapsed time before the automatic Trunk-to-Trunk Transfer is performed.
- ➔ **24-04-01 Automatic Trunk-to-Trunk Transfer Target Setup**
Assign the Speed Dialing number bin (0~1999) to a trunk and the mode which should be used as the destination of the Automatic Trunk-to-Trunk Forwarding.

Trunks: 001~200
Time Modes: 1~8


Trunk-to-Trunk Forwarding – DID (3) Trunk Forwarding by Department Groups

 Refer to *Department Calling* for additional Department Group programming.

- ➔ **11-07-01 Department Group Pilot Numbers – Dial**
Assign pilot numbers to the Extension (Department) Groups you set up in Program 16-02.
- ➔ **11-11-25 Service Code Setup (for Setup/Entry Operation) – Automatic Transfer for Each Extension Group**
Customize the service code used to set the Automatic Trunk Forwarding feature for a Department Group (default: 602).
- ➔ **11-11-26 Service Code Setup (for Setup/Entry Operation) – Automatic Transfer Cancellation for Each Extension Group**
Customize the service code used to cancel the Automatic Trunk Forwarding feature for a Department Group (default: 603).
- ➔ **11-11-27 Service Code Setup (for Setup/Entry Operation) – Destination of Automatic Transfer Each Extension Group**
Customize the service code used to set the destination for the Automatic Trunk Forwarding feature for a Department Group (default: 604).
- ➔ **13-01-01 Speed Dialing Option Setup – Speed Dialing Auto Outgoing Call Mode**
Determine if dialing an Speed Dialing number dials an outside number (seizing a trunk as assigned in PRG 13-05) or an Intercom number (0 = Trunk Dialing Mode, 1 = Extension Dialing Mode).

- ➔ **13-04-01 Speed Dialing Number and Name – Speed Dialing Data**
Enter the Common and Group Speed Dialing numbers and names which are used for Trunk-to-Trunk Forwarding.
- ➔ **14-01-04 Basic Trunk Data Setup – Transmit Gain Level for Conference and Transfer Calls**
Use CODEC gain set at 0 dB [PRG 14-01-04 = 32 (CODEC Gain Type 2)] to set the CODEC gain level for multiline Conference or transferred calls.
- ➔ **14-01-05 Basic Trunk Data Setup – Receive Gain Level for Conference and Transfer Calls**
Use CODEC gain set at 0 dB [PRG 14-01-04 = 32 (CODEC Gain Type 2)] to set the CODEC gain level for multiline Conference or transferred calls.
- ➔ **14-06-01 Trunk Group Routing**
Set up an outbound routing table for the trunk groups you assign in Program 14-05-01. When users dial 9, the system routes their calls in the order (priority) you specify.
- ➔ **15-07-01 Programmable Function Keys**
Assign an Automatic Forwarding at Department Group key (58) or a Delayed Forwarding at Department Group key (59) for an extension user.
- ➔ **16-02-01 Department Group Assignment for Extensions**
Assign extensions to Department Groups (1~64) and set the priority assignment.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-11-17 Class of Service Options (Hold/Transfer Service) – Department Group Trunk-to-Trunk Transfer (Each Telephone Group Transfer)**
Enable (1)/Disable (0) an extension in a Department Group ability to use the Trunk-to-Trunk Forwarding service codes.
- ➔ **24-05-01 Department Group Transfer Target Setup**
Assign the Speed Dialing number bin (0~1999) which should be used as the destination of the Department Group Trunk-to-Trunk Forwarding.


Trunk-to-Trunk Forwarding – DID (3) Trunk Forwarding Using DID Translation Table

 Refer to Direct Inward Dialing (DID) for additional DID programming.

- ➔ **22-11-05 DID Translation Table Number Conversion – Transfer Destination Number 1**
- 22-11-06 DID Translation Table Number Conversion – Transfer Destination Number 2**
For each DID Translation Table entry (1-2000), specify the first and second Transfer Destinations if the callers receives a busy or no answer (action defined in PRG 22-11-04).

Destination Options:
 0 = No Setting
 1~100 = Incoming Group
 102 = In-Skin/External Voice Mail
 103 = Centralized Voice Mail
 01~264 = Department Group
 400 = DID
 401 = DISA

501~548 = DISA/VRS Message
1000~1999 = Speed Dial Bin Number (000~999)

 *If the Transfer Destinations are busy or receive no answer, those calls are transferred to the final transfer destination (Program 22-10).*

Operation

To activate Call Forwarding Off-Premise non-split:

1. At a multiline terminal, press **Speaker**.

- OR -

At a single line telephone, lift the handset.


2. Dial the Call Forwarding Service Code.

- OR -

At a multiline terminal only, press the Call Forwarding Programmable Function keys (PRG 15-07-01, PRG 15-07-10 ~PRG 15-07-15 or SC 751 Key Code 10~15).

3. Dial **1** (Set).

4. Dial the Trunk Access Code (default: 9) + Number (9+2142622000).

 *Trunk access codes are 9 (ARS/Trunk Group Routing), 704 + Line Group (1~9, 01~99 or 001~100) or #9 + Line number (e.g., 05 or 005 for line 5).*

 *Your DND or Call Forwarding (Device) Programmable Function key flashes.*

To cancel Call Forwarding Off-Premise non-split:

1. At a multiline terminal, press **Speaker**.

- OR -

At a single line telephone, lift the handset.

2. Dial the Call Forward Access Code (default not assigned).

3. Dial **0** (Cancel).

To activate Call Forwarding Off-Premise Split:



1. At a multiline terminal, press **Speaker**.

- OR -



At a single line telephone, lift the handset.

2. Dial the Call Forwarding Service Code.


3. Dial **1** (Set).

4. Dial **1** (Internal) or **0** (External).
5. Dial Trunk Access Code (default: 9) + number (9 + 2142622000).
 -  *Trunk access codes are 9 (ARS/Trunk Group Routing), 704 + Line Group (1~9, 01~99 or 001~100) or #9 + Line number (e.g., 05 or 005 for line 5).*
 -  *Your DND or Call Forwarding (Device) Programmable Function key flashes.*

To cancel Call Forwarding Off-Premise Split:

1. At the multiline terminal, press **Speaker**.
- OR -
2. At a single line telephone, lift the handset.
3. Dial the Call Forward Access Code (default not assigned).
4. Dial **0** (Cancel).
 -  *If Internal and External are set both are canceled.*
 -  *Your DND or Call Forwarding (Device) Programmable Function key flashes.*

Off-Premise Call Forwarding for Door Boxes

 *These operations are performed at the Door Box Ringing Extension only.*

To activate Call Forwarding Off-Premise for a Door Box:


 *This option only works for ISDN PRI or BRI Trunks.*

1. At the multiline terminal, press **Speaker** + dial **SC 722**.
- OR -
At the multiline terminal only, press Call Forward (Device) key (PRG 15-07-01 or SC 751, code 54).
- OR -
At the single line telephone, lift the handset + dial **722**.
2. Dial the Door Box number (**1~4**).
3. Dial the Speed Dialing number where the calls should be forwarded.
4. Press **Speaker** (or hang up at the single line telephone) to hang up.

To cancel Call Forwarding Off-Premise for a Door Box:

1. At the multiline terminal, press **Speaker** + dial SC **722**.
- OR -
At the multiline terminal only, press Call Forward (Device) key (PRG 15-07-01 or SC 751, code 54).
- OR -
At the single line telephone, lift the handset + dial **722**.
2. Dial **0** (Cancel).


Trunk-to-Trunk Forwarding**Set the Destination and Forward the Line:**

1. Lift the handset.
2. Dial **735**.
3. Dial trunk port number (**001~200**) to be defined.
4. Select the mode (**1~8**) to be defined.
5. Enter the telephone number, which is the destination of the forwarded trunk.
 *The number is stored in the Speed Dial bin number assigned in Program 24-04-01. This entry overwrites any existing number defined in the bin.*
6. Press **Hold** to accept the entry.
7. Repeat from Step 3 to define another mode entry or press **Speaker** to hang up.

Cancel the Line Forwarding:

1. Lift the handset.
2. Dial **735**.
3. Dial trunk port number (**7 001~200**) to be defined.
4. Select the mode (**1~8**) to be defined.
5. Press the **Exit** key.
6. Press **Speaker** to hang up.

Automatic Trunk-to-Trunk Transfer (Step Transfer) (follows the predefined destination in Program 24-04-01) Set Automatic Trunk Forwarding:


-  *The Speed Dial bin must be defined in PRG 13-04-01 for the line to forward.*
1. Lift the handset.
 2. Dial **733**.

3. Dial trunk port number to be used (**001~200**).
4. Press **Speaker** to hang up.

Cancel Automatic Trunk Forwarding:

1. Lift the handset.
2. Dial **734**.
3. Dial trunk port number to be used (**001~200**).
4. Press **Speaker** to hang up.


Department Group Line Forwarding**Method 1****Set the Destination and Forward the Line:**

1. Lift the handset.
2. Dial **604**.
3. Dial the Department Group number (**01~64**) to be defined.
4. Select the time mode (**1~8**) to be defined.
5. Enter the telephone number, which is the destination of the forwarded trunk.
 *The number is stored in the Speed Dial bin number assigned in Program 24-04-01. This entry overwrites any existing number defined in the bin.*
6. Press **Hold** to accept the entry.
7. Repeat from Step 3 to define another time mode entry or press **Speaker** to hang up.

Cancel the Line Forwarding:

1. Lift the handset.
2. Dial **604**.
3. Dial the Department Group number (**01~64**) to be defined.
4. Select the time mode (**1~8**) to be defined.
5. Press the **Exit** key.
6. Press **Speaker** to hang up.

Method 2 (follows the predefined destination in Program 24-05-01)**Set Automatic Trunk Forwarding:**

 *The Speed Dial bin must be defined in PRG 13-04-01 for the line to forward.*

1. Lift the handset.
2. Dial **602**.
3. Dial the Department Group number (**01~64**) to be defined.
4. Press **Speaker** to hang up.

Cancel Automatic Trunk Forwarding:

1. Lift the handset.
2. Dial **603**.
3. Dial the Department Group number (**01~64**) to be defined.
4. Press **Speaker** to hang up.

Call Forwarding – Park and Page

Description

When an extension user is away from their phone, Park and Page can let them know when they have a call waiting to be answered. The Personal Greeting and Park & Page options can have up to 200 messages total (note that the Park & Page feature uses two messages). To enable Park and Page, the user records a Personal Greeting along with an additional Paging announcement. Park and Page then answers an incoming call and plays the Personal Greeting to the caller. The caller then listens to Music on Hold (if available) while the system broadcasts the recorded Paging announcement. When the extension user hears the Page, they can go to any telephone and use Directed Call Pickup to intercept the call.

For example, John Smith could record a Personal Greeting that says:

“Hello, this is John Smith. I am away from my phone right now but please hold on while I am automatically paged.”

The recorded Paging announcement could say:

“John Smith, you have a call waiting on your line.”

The incoming caller hears the first message and listens to Music on Hold while the system broadcasts the second message. John Smith could then walk to any phone and pick up his call. If John doesn't pick up the call, the Page periodically repeats.

Park and Page follows the rules for Personal Greeting for All Calls, immediately rerouted. This means that Park and Page activates for ringing Intercom calls, DID calls and DISA calls. It also activates for calls transferred from the Automated Attendant. Additionally, calls from the Automated Attendant follow Automatic Overflow routing if not picked up. Park and Page activates for transferred outside calls but does not play the Personal Greeting to the caller. If a call comes in when the specified Page zone is busy, the system broadcasts the announcement when the zone becomes free.

Conditions

- Park and Page announcements only repeat once.
- Voice Announcement (VAU) recording time is fixed at two minutes and cannot be changed.
- While Park and Page is enabled, only one DID call can be processed at a time. Subsequent calls receive a busy tone.

Default Setting

- Park and Page is available at default for internal paging access code 701, zone 1.
- Use access code of 713. See feature Operation. Set PRG 40-10-01 for VRS guidance message.

System Availability

Terminals

None

Required Component(s)

Voice Response System (VRS) card and DSPII-U10 Unit

Related Features

Analog Communication Interface (ACI)

Music on Hold

Programming

- ➔ **11-11-58 Service Code Setup (for Setup/Entry Operation) – Call Forward with Personal Greeting**
Call forward with Personal greeting VRS. Service code setup. Default set to 713.
- ➔ **11-12-19 Service Code Setup (for Service Access) – Internal Group Paging**
Service code setup. Default set to 701.
- ➔ **11-12-20 Service Code Setup (for Service Access) – External Paging**
External paging access code. Service code setup. Default set to 703.
- ➔ **11-12-24 Service Code Setup (for Service Access) – Combined Paging**
Combined paging, internal/external access code. Service code setup. Default set to *1.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **31-02-01 Internal Paging Group Assignment – Internal Paging Group Number**
Assign extensions to Internal Paging Groups (i.e., Page Zones).

- ➔ **31-03-01 Internal Paging Group Settings – Internal Paging Group Name**
Assign names to Internal Paging Groups (i.e., Page Zones). The system shows the names you program on the telephone displays.
- ➔ **31-04-01 External Paging Zone Group – Paging Group Number**
Assign each External Paging Speaker to an External Paging Zone.
- ➔ **31-06-01 External Speaker Control – Broadcast Splash Tone Before Paging (Paging Start Tone)**
Enable (1)/Disable (0) splash tone before Paging over an external zone. If enabled, the system broadcasts a splash tone before the External Paging announcement.
The paging chime overrides the first four seconds of the announcement. Allow a delay in announcement recording for chime time.
- ➔ **31-06-02 External Speaker Control – Broadcast Splash Tone After Paging (Paging End Time)**
Enable (1)/Disable (0) splash tone after Paging over an external zone. If enabled, the system broadcasts a splash tone at the end of an External Paging announcement.
- ➔ **40-10-01 Voice Announcement Service Option – VRS Fixed Message**
Enable (1)/Disable (0) the system ability to play the fixed VRS messages (such as “You have a message.”).
- ➔ **40-10-05 Voice Announcement Service Option – Park and Page Repeat Timer (VRS Msg Resend)**
If a Park and Page is not picked up in this time, the Paging announcement repeats.

Operation

To have the system page you when you have a call:

1. Press **Speaker** (or lift the handset at the single line telephone) and dial **713**.
2. When you hear, “Please start recording,” record your Personal Greeting.
If you already have Park and Page or Personal Greeting set up, you can dial:

 - 3 to erase (the optionally HOLD to cancel the erase)*
 - 5 to listen (then # again to listen again)*
 - 7 to re-record*
3. Dial **#7**.
4. When you hear, “Please start recording,” record your page and dial **#** when the announcement is complete.
A paging chime overrides the first four seconds of an announcement. Allow a delay in announcement recording for chime time.
5. Dial the Page Zone that should broadcast your announcement.
*For example, for Internal Zone 1 dial 701 + 1, or for Combined Paging Zone, 1 dial *1 + 1.*

6. Dial the Park and Page type:
2 = All Calls
3 = Outside Calls Only
7. Press **Speaker** to hang up (or go on-hook at the single line telephone).

To pick up your Park and Page:

1. Press **Speaker** (or lift the handset at the single line telephone).
2. Dial ****** + your extension number.

To cancel your Park and Page:

1. Press **Speaker** (or lift the handset at the single line telephone).
2. Dial **713 + 3**.
3. Press **Speaker** to hang up (or go on-hook at the single line telephone).

Call Forwarding with Follow Me

Description

While at a co-worker's desk, a user can have Call Forwarding with Follow Me redirect their calls to the co-worker's extension. This helps an employee who gets detained at a co-worker's desk longer than expected. To prevent losing important calls, the employee can activate Call Forwarding with Follow Me from the co-worker's telephone.

Call Forwarding with Follow Me reroutes calls from the destination extension. To reroute calls from the initiating (forwarding) extension, use Call Forwarding.

Conditions

- Call Forwarding an extension in a Department Group prevents that extension from receiving Department Pilot Calls.
- If a Programmable Function key is not defined for Call Forwarding (10~17), the DND key flashes to indicate that the extension is call forwarded.
- Multiple Stations can set Call Forward Follow Me to one station.
- Calls to extensions with DND active do not follow Call Forwarding programming. DIL calls ring an idle Department Group member, then follow PRG 22-08 programming then PRG 22-05 programming.

Default Setting

Enabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Do Not Disturb

Programmable Function Keys


Programming

- ➔ **11-11-07 Service Code Setup (for Setup/Entry Operation) – Call Forwarding – Follow Me**
Assign the service code of Call Forward Follow Me (Default: 746).
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for one-touch access to the Call Forwarding (Station) setup code (code 10~17).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-11-05 Class of Service Options (Hold/Transfer Service) – Call Forwarding with Follow Me**
Allow (1)/Prevent (0) the setting of Call Forwarding with Follow Me.

Operation

To activate Call Forward Follow Me from a multiline terminal:

1. At a multiline terminal, other than your own, press **Speaker** and dial Service Code (**746**, PRG 11-11-07).
- OR -
Press the Call Forward Follow Me key (PRG 15-07-01 or SC 751: Code 15).
2. Dial **1** to set.
3. Dial the Extension to Follow.

 *The multiline terminal with display indicates on the display of the telephone which Call Forward Follow Me is set to. Also, the Programmed Follow Me Flexible Line Key flashes (if assigned) when Follow Me is set.*

To cancel Call Forward Follow Me from your own Multiline Terminal:

1. At your multiline terminal, press **Speaker** and dial Service Code (**746**, PRG 11-11-07).
- OR -
Press the Call Forward Follow Me key (PRG 15-07-01 or SC 751: Code 15).
2. Dial **0** to cancel.

3. Dial **0** (Cancel All Forward Follow Me).

- OR -

Dial the extension number with Follow Me set.

To activate Call Forward Follow Me from a single line telephone:

1. At a single line telephone, other than your own, lift the handset and dial the Service Code (**746** PRG 11-11-07).
2. Dial **1** to set.
3. Dial the extension to follow.

To cancel Call Forward Follow Me from your own single line telephone:

1. At your single line telephone, lift the handset and dial Service Code (**746**, PRG 11-11-07).
2. Dial **0** to cancel.
3. Dial **0** (Cancel All Forward Follow Me).

- OR -

Dial the extension number with Follow Me set.

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Call Monitoring

Enhancements

Version 1500 or higher supports Call Monitoring with coaching.

Description

Call Monitoring allows selected Multiline Terminal Users to monitor another user's conversation without the ability to participate. A programmable audible alert tone can be sent to that station user. Without the audible alert (silent monitor), no indication is provided to either the monitored station or the outside party.

Caution

The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversation or other sound activities, whether or not contemporaneous with transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to a telephone conversation, such as using a beep tone or other notification methods or requiring the consent of all parties to the telephone conversation, prior to monitoring or recording the telephone conversation. Some of these laws incorporate strict penalties.

Call Monitoring with Coaching Ability

Call Monitoring with Coaching Ability allows the transmit path to be opened to the monitored station only, to provide the Coaching ability for the person performing the Call Monitoring. Press **MIC**, or dial **Feature + 1** to toggle the Coaching ability on and off.

Conditions

- Call Monitoring is allowed for internal calls.
- An extension user cannot Monitor an Intercom call if one of the Intercom callers is using Hands-free Answerback. Both Intercom parties must lift the handset or press **Speaker**.
- An extension user cannot monitor a conference, however an extension programmed for Call Monitor can Barge-In to a conference.

- With Program 20-13-10 set to 0, a call, which is barged into, can be placed on hold by the originator of the outside call. Both the outside caller and the extension, which is monitoring the call, are placed on hold.
- The handset and microphone are muted during Call Monitoring.
- Live Record does not work for Call Monitor calls.
- While being monitored, an extension cannot receive Voice Over.
- When a monitored extension places a call on hold, Call Monitor is automatically finished.
- With Program 20-13-10 set to 1, a call which is being Monitored can be placed on park by the originator of the outside call, but only the outside caller is placed in park. The extension which is monitoring the call is dropped.
- When Program 20-13-10 is set to 0 (Off), coaching is not permitted. When Program 20-13-10 is set to 1 (On), Program 20-13-45 takes effect.
- When Silent Monitor Mode is used, MIC or Feature + 1 can be used to activate speech path to the internal and external parties.

Default Setting

Disabled.

System Availability

Terminals

All terminals

Required Component(s)

None

Related Features

Barge-In

Conference

Hold

Intercom

Park

Privacy (Data)

Programmable Function Keys

Voice Mail

Programming

- ➔ **11-12-08 Service Code Setup (for Service Access) – Barge-In**
Determine what the service code should be for an internal party to use Barge-In (default = 710).
- ➔ **11-16-02 Single Digit Service Code Setup – Barge-In**
Set up Item 02 for single digit Barge-In. For example, you can assign Item 02 to use digit 5 for Barge-In. This allows you to program a function key with an extension number plus the Barge-In code (i.e., 305 5). This allows one-touch access to Barge-In for extension 305. If you undefine a service code, be careful when you change this item that you don't inadvertently disable any essential dialing function (such as Voice Mail or Message Waiting).
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Barge-In (code 34).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-10 Class of Service Options (Supplementary Service) – Barge-In Monitor**
Enable the Barge-In Speech Mode (0) or Monitor Mode (1) at the initiating extension (i.e., Barge-In initiator).
- ➔ **20-13-15 Class of Service Options (Supplementary Service) – Barge-In, Initiate**
Enable (1)/Disable (0) Barge-In at the initiating extension (i.e., Barge-In initiator).
- ➔ **20-13-16 Class of Service Options (Supplementary Service) – Barge-In, Receive**
Enable (1)/Disable (0) Barge-In at the receiving extension (i.e., Barge-In receive).
- ➔ **20-13-17 Class of Service Options (Supplementary Service) – Barge-In Tone/Display (Intrusion Tone)**
Enable (1)/Disable (0) the Barge-In Tone. If disabled, this also turns off the Barge-In display at the called extension.
- ➔ **20-13-32 Class of Service Options (Supplementary Service) – Deny Multiple Barge-Ins**
Enable (1)/Disable (0) the extension user ability to have multiple users Barge-In to their conversation.

- ➔ **20-13-45 Class of Service Options (Supplementary Service) – MIC Key Mode While Call Monitoring (Call Monitor Mode) (Version 1500 or higher required)**
Use Disable (0) for Barge-In with Alert Tone or Enable (1) to allow coaching.
- ➔ **20-14-11 Class of Service Options for DISA/E&M – DISA/Tie Trunk Barge-In**
Enable (1)/Disable (0) a DISA or tie trunk user from using the Barge-In feature with the service code undefined in Program 11-12-08.
- ➔ **20-18-07 Service Tone Timers – Intrusion Tone Repeat Time**
After a user Barges In, the system repeats the Barge-In tone after this time. Normally, you should enter 0 to disable this time. (This time also affects any other call interruption feature, such as Voice Mail Conversation Recording, Voice Over, etc.)
- ➔ **21-01-03 System Options for Outgoing Calls – Trunk Interdigit Time (External)**
Program the time an extension must wait before the Barge-In feature can be used on a call (this time expires before a call is put in a talk state). This time also affects Voice Over.

Operation

The call must be set up for about 10 seconds before it can be Monitored. Listen for busy/ring or busy tone.

To Call Monitor after calling a busy extension:

1. Call a busy extension.
2. Press the Barge-In key (PRG 15-07 or SC 751: 34).

- OR -


The following steps are not available for DISA or Tie Line trunks.

1. Dial the extension number of the busy internal party.
2. Dial the single digit service code or the service code **710** (the access code will change).

To Call Monitor without first calling the busy extension:

1. Press **Speaker** or lift handset.
2. Dial **710** or press the Barge-In key (PRG 15-07 or SC 751: 34) (the access code will change).

3. Dial a busy extension.

 *If Monitoring is not possible:*

- the extension user hears a warning tone.*
- the DISA user is rerouted to the defined ring group.*
- the Tie Line user hears a busy tone.*

To Call Monitor using Coaching Ability

1. Call a busy extension.
2. Press the **Barge-In** Key (PRG 15-07 or SC 751:34).
3. Press **MIC** or **Feature + 1** to toggle Coaching Ability on and off to the monitored station.
- OR -
1. Dial the extension number of the busy party.
2. Dial the single digit service code or the service code **710** (the access code changes).
3. Press **MIC** or **Feature + 1** to toggle Coaching Ability on and off.

To Call Monitor using Coaching Ability without first calling the busy extension

1. Press **Speaker** or lift the handset.
2. Dial **710** or press the **Barge-In** key (PRG 15-07 or SC 751:34). The access code changes.
3. Dial a busy extension.
4. Press **MIC** or **Feature + 1** to toggle Coaching Ability on and off to the monitored station.

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Call Redirect

Description

Call Redirect allows a multiline terminal user to transfer a call to a predefined destination (such as an operator, voice mail, or another extension) without answering the call. This can be useful if you are on a call and another rings in to your extension. You can press the Call Redirect key, to transfer the call, allowing you to continue with your current call.

This feature works with the following calls:

- Normal trunk call
- DID
- DISA
- DIL
- E&M
- ICM

The following calls *cannot* be redirected with the feature:

- ACD
- Transferred
- Department Group (all ring mode)
- Door Box
- Virtual Extension

Conditions

- After you press the Call Redirect key, the call does not recall to the extension.
- The predefined destination must be an extension number or voice mail pilot number.
- When a call is Redirected to another phone it does not follow the forwarding on that phone.

Default Setting

Enabled

System Availability

Terminals

Any Multiline Terminal

Required Component(s)

None

Related Features

None

Programming

- **15-07-01 Programmable Function Keys**
Assign a function key for Call Redirect (code 49 + destination extension number).
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- **20-11-16 Class of Service Options (Hold/Transfer Service) – Call Redirect**
Enable (1)/Disable (0) an extension user ability to transfer a call to a predefined destination (such as an operator, voice mail, or another extension) without answering the call.

Operation

To redirect a ringing call:

With an incoming call ringing your extension, press the Call Redirect key (Program 15-07 or SC 751: 49 + Destination Extension Number) without lifting the handset.

Call Waiting/Camp-On

Description

With Call Waiting, an extension user may call a busy extension and wait in line (Camp-On) without hanging up. When the user Camps-On, the system signals the busy extension with two beeps indicating the waiting call. The call goes through when the busy extension becomes free. Call Waiting helps busy extension users know when they have additional waiting calls. It also lets callers wait in queue for a busy extension without being forgotten.

Conditions

- Call Arrival (CAR) Key (virtual extension) keys do not support Call Waiting/Camp-On Programmable Function keys (code 35).
- If an extension user Camps-On and then hangs up, the system converts the Camp-On to a callback.
- If an extension busy on a call has Off-Hook Signaling, an incoming Intercom calls rings the idle second line appearance.
- Off-Hook Signaling allows an extension user to block a caller from dialing 704 to Camp-On and/or DID callers from automatically camping on.
- Function keys simplify Call Waiting/Camp-On operation.
- An extension user can Transfer a call to a busy extension.
- Trunk Queuing lets an extension user camp-on to a trunk.
- Call Queuing must also be disabled to disable Call Waiting.

Default Setting

Enabled

System Availability

Terminals

Multiline Terminal and Single Line Telephone

Required Component(s)

None

Related Features

Callback

Off-Hook Signaling

Programmable Function Keys

Transfer

Trunk Queuing/Camp On

Programming

- **11-11-23 Service Code Setup (for Setup/Entry Operation) – Second Call for DID/DISA/DIL**
Enable Second Call to each extension when PRG 20-09-01 (Second Call) is set to 0 (disable).
(default: 679)
- **11-12-04 Service Code Setup (for Service Access) – Set Camp-On**
Customize the Service Code used for setting Camp-On.
(default: 750)
- **11-12-05 Service Code Setup (for Service Access) – Cancel Camp-On**
Customize the Service Code used for cancelling Camp-On.
(default: 770)
- **11-12-47 Service Code Setup (for Service Access) – Call Waiting Answer/Split Answer**
If required, change the code users dial to Split while on a call. This code is normally 794.
- **11-16-05 Single Digit Service Code Setup – Camp-On**
Customize the 1-digit Service Code used for setting Camp-On.
(default: #)
- **15-02-06 Multiline Telephone Basic Data Setup – Hold Key Operating Mode**
Set the function of the extension Hold key.

0 = Normal Common (default)
1 = Exclusive Hold

-
-
- ➔ **15-02-12 Multiline Telephone Basic Data Setup – Off-Hook Ringing**
For each extension, set the Off-Hook Ringing type.
0 = Muted (default)
1 = None
3 = Beep in Speaker
4 = Beep in Handset
5 = Speaker & Handset Beep

 - ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Camp-On (code 35). This key is also the Callback key.

 - ➔ **20-01-08 System Options – Trunk Queuing Callback Time**
Set the Trunk Queuing callback time (default: 15). A Trunk Queuing Callback rings an extension for this time.

 - ➔ **20-01-09 System Options – Callback/Trunk Queuing Cancel Time**
The system cancels an extension Callback or Trunk Queueing request after this time (default: 64800 seconds).

 - ➔ **20-03-01 System Options for Single Line Telephones**
The system cancels an extension Callback or Trunk Queueing request after this time (default: 64800 seconds).

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service to an extension (1~15).

 - ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DDI/DIL/E&M**
Enable (1)/Disable (0) an extension ability to receive a second call from a DID, DISA, DIL, or Tie line caller.

 - ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Enable (1)/Disable (0).

 - ➔ **20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook signaling**
Enable (1)/Disable (0) the extension ability to receive Off-Hook Signaling.


 - ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension user to manually (0) or automatically (1) receive off-hook signals.

 - ➔ **20-13-35 Class of Service Options (Supplementary Service) – Block Camp-On**
Enable (1)/Disable (0) an extension user ability to block callers from dialing 2 to camp-on.

 - ➔ **20-18-06 Service Tone Timers – Interval of Call Waiting Tone**
Set the time between call waiting tones. This also sets the time between Off-Hook signaling alerts.

Operation

To Camp-On a busy extension:

1. Call the busy extension.
2. Dial # or press the Camp-On key (PRG 15-07 or SC 751: 35).
3. Do not hang up.
 -  To camp-on to a trunk, refer to [Trunk Queuing/Camp-On on page 2-999](#).

To cancel a Camp-On request:


1. Hang up.
2. At a multiline terminal, press **Speaker** and dial **770**.
 - OR -
 - At a multiline terminal, press the Camp-On key (PRG 15-07 or SC 751: 35).
 - OR -
 - At the single line telephone, lift the handset and dial **770**.



To Split (answer a waiting call) at a single line telephone:

Listen for Camp-On beep.

1. Hookflash or hookflash and dial **794** to repeatedly split between the two calls.

To Split (answer a waiting call) at a single line telephone:

 *Listen for Camp-On beep.*

1. Hookflash and dial **794** to repeatedly split between the two calls.
 -  *The operation depends on the setting in PRG 11-12-47.*
 -  *This operation is valid only before the caller performs the camp-on operation (refer To Camp-On a busy extension – step 2).*

Callback

Description

When an extension user calls a co-worker that does not answer or is busy, they can leave a Callback request for a return call. The user does not have to repeatedly call the unanswered extension back, hoping to find it idle.

The system processes Callback requests as follows:

1. Caller at extension A leaves a Callback at extension B.
✎ Caller can place or answer additional calls in the meantime.
2. When extension B becomes idle, the system rings extension A. This is the Callback ring.
3. Once caller A answers the Callback ring, the system rings (formerly busy or unanswered) extension B.
✎ If caller A does not answer the Callback ring, the system cancels the Callback.
4. As soon as caller B answers, the system sets up an Intercom call between A and B.

Callback Automatic Answer determines how an extension user answers the Callback ring. When Callback Automatic Answer is enabled, a user answers the Callback ring when they lift the handset. When Callback Automatic Answer is disabled, the user must press the ringing line appearance to answer the Callback ring.

Conditions

- An extension can leave only one Callback request at a time.
- Call Arrival (CAR) Key (virtual extension) keys do not support Call Waiting/Camp-On Programmable Function keys (code 35).
- If an extension user initiates a Callback but does not hang up, their extension Camps-On to the busy extension.
- Function Keys simplify Callback operation.

Default Setting

Enabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Call Waiting/Camp-On)



Programmable Function Keys

Programming

- ➔ **11-12-05 Service Code Setup (for Service Access) – Cancel Camp-On**
If required, redefine the service code used cancel Camp-On (default: 770).
- ➔ **11-12-44 Service Code Setup (for Service Access) – Callback Test for SLT**
If required, redefine the service code used for SLT Callback Test (default: 799).
- ➔ **11-16-05 Single Digit Service Code Setup – Camp-On**
If required, redefine the service code used to set Camp-On (default: #).
- ➔ **15-02-11 Multiline Telephone Basic Data Setup – Callback Automatic Answer**
Enable (1)/Disable (0) Callback Automatic Answer.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Callback (code 35). This key is also the Camp-On key.
- ➔ **20-01-07 System Options – Callback Ring Duration Time**
Set the duration of the Callback ring (0~64800 seconds).
- ➔ **20-01-09 System Options – Callback/Trunk Queuing Cancel Time**
The system cancels Callback and Trunk Queuing requests after this time (0~64800 seconds).

Operation


To place a Callback:

1. Call unavailable (busy or unanswered) extension.
2. Dial # or press the Callback key (PRG 15-07 or SC 751: 35).
3. Hang up.
4. Lift the handset when busy extension calls you back.
 -  *If the unavailable extension was unanswered (not busy), the Callback goes through after your co-worker uses their telephone for the first time.*
 -  *If you have Callback Automatic Answer, you automatically place a call to the formerly busy extension when you lift the handset. If you do not have Callback Automatic Answer, you must press the ringing line appearance to place the call.*

To cancel a Callback:

1. At the multiline terminal, press **Speaker** and Dial **770**.
 - OR -
 - At the multiline terminal, press Camp-On key (PRG 15-07 or SC 751: 35).
 - OR -
 - At the single line telephone, lift the handset and dial **770**.

To test Callback at a single line telephone:

1. Lift the handset.
2. Dial **799**.
3. Hang up.
4. When the telephone rings, lift the handset.
 -  *You hear the Hold tone.*
5. Hang up.

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Caller ID


Enhancements

Version 1600 and Version 2100 or higher:

- Supports adding the local area code in front of the seven digit stored Caller ID information.
- When the user makes an outgoing call with Caller ID History, the system adds the digits programmed in PRG 15-02-40 in front of the stored Caller ID Number.

Description

Caller ID allows a display terminal to show an incoming caller's telephone number (called the Directory Number or DN) and optional name. The Caller ID information is available as pre-answer display. With the pre-answer display, the user previews the caller's number before picking up the ringing line.

-  On the CPUII for Caller ID (also used for DTMF receivers and Call Progress Tone Detection) 32 resources are available. The DSPII-U10 Unit provides an additional 32 resources.

Second Call Display

While busy on a call, the telephone display can show the ID of an incoming trunk or Intercom call. For incoming trunk calls, the display shows the Caller ID or ANI data or the trunk name if Caller ID or ANI are not installed. (Refer to [T1 Trunking \(with ANI/DNIS Compatibility\) on page 2-941](#) for more information on ANI compatibility.) For incoming Intercom calls, the display shows the calling extension name.

Caller ID supports the Telco Called Number Identification (CNI) and Called Number Delivery (CND) service, when available. These services provide the Caller ID information (i.e., messages) between the first and second ring burst of an incoming call. Two Caller ID message formats are currently available: Single Message Format and Multiple Message Format. With Single Message Format, the Telco sends only the caller's telephone number (DN). The DN has either seven or 10 digits. In Multiple Message Format, the Telco sends the DN and the caller's name. The DN for this format has seven or 10 digits, and the name provided consists of up to 15 ASCII characters.

The telephone display can show up to 12 Caller ID digits (for non-ACD calls).

Once installed and programmed, Caller ID is enabled for all trunk calls, including:

- Ring Group calls
- Calls transferred from another extension
- Calls transferred from the VRS

- Calls transferred from Voice Mail (unscreened)
- Direct Inward Lines (DILs)

Caller ID temporarily stores 50 calls (total of abandoned and answered/unanswered). New calls replace old calls when the buffer fills.

Temporary Memory

An unanswered call causes the Call History key (PRG 15-07 or SC 751: 08) to flash, indicating a new call was placed in the temporary memory. If enabled in programming, the telephone display shows CHECK LIST.

This Caller ID data from the temporary memory can be saved in either Speed Dial bins or in One-Touch keys making them available for placing future calls.

Outputting Caller ID Data

The system includes the Caller ID data on the SMDR report. The report provides the incoming call DN in the DIALED NUMBER field. The CLASS field shows PIN (just like all other incoming calls).

Caller ID data can output also to a PC or other type of computer through a CTA Module. This allows off-line database lookups. In a customer service department, for example, the computer could search for a caller's records and display their account status even before a customer service representative picked up the telephone.

Caller ID Digits to Voice Mail

A Caller ID/ANI trunk can send Remote Log-On Protocol with Caller ID digits to the voice mail. When a trunk 001 receives the Caller ID as 12345, the protocol becomes *****60001*12345***.

Display Reason for No Caller ID Information

With Caller ID enabled, the system provides information for analog calls that do not detect the Caller ID information. If the Caller ID information is restricted, the telephone display shows PRIVATE. If the system cannot provide Caller ID information because Telco information is not detected, the display shows NO CALLER INFO.

Calling Party Number Information

When using the Electra Elite IPK II Wireless telephone, the system can provide the Caller ID information for an external call if it is provided by the Telco.

Option to Enable Caller ID Name for SLT

System programming provides an option for single line telephones to display Caller ID.

Add Trunk Access Code to Caller ID with Electra Elite IPK II Wireless – Phones

Electra Elite IPK II Wireless – DECT Phones on the Electra Elite IPK II can hold incoming call history. This history is created based on the Caller ID information element contained in the call Setup message which is transmitted from the Electra Elite IPK II. This information allows users to return calls dialing the number stored.

The stored number, however, does not contain the trunk access code. Without this code, the system may not seize an outside line to complete the call.

With this feature, when an Electra Elite IPK II Wireless user receives an incoming trunk call, the trunk access code defined in programming can be added to the Caller ID. This allows the system to seize an outside line, and then dial the stored number.

- This function is applied only to incoming ISDN calls. It does not apply to incoming extension calls.
- Caller ID must be available for this feature to work.
- The maximum number of Caller ID digits is 20. If the total number of digits [trunk access code (PRG 10-02-05) and Caller ID] is over 20, the remaining Caller ID digits are not dialed.

For example:

Trunk Access Code (Program 10-02-05): 123456##* (eight digits)

Incoming Caller ID: 12345678901234567890 (20 digits)

Electra Elite IPK II Wireless Dials: 123456##*123456789012

- An additional digit (such as 1) may be required to complete the call (Program 10-02-04).

For example:

Incoming Caller ID shows: 2125551212.

If your area code is NOT 212, then define a 1 in 10-02-04. When callback is executed, the system prefixes a 1 on the digits dialed string.

Caller ID Sender Queuing Added

The Electra Elite IPK II system can provide Caller ID (calling party number) to a single line telephone with a display.

The system can queue incoming calls to the single line telephone if the system Caller ID sender resources are busy. Refer to Program 20-19-05 in the Electra Elite IPK II Programming Manual.

If the single line telephone user lifts their handset while an incoming call is waiting in queue, they hear silence (no dial tone) and cannot dial out. When the single line telephone user goes back on-hook, the system immediately sends the queued call to the single line telephone without Caller ID.

Option Available for FSK or DTMF Type for Single Line Telephone

An option (Program 15-03-11) is available for the Caller ID which allows you to select either FSK or DTMF as the Caller ID type to be received by a single line telephone.

Option Available for FSK or DTMF Type from Analog Trunk

An option (Program 14-02-16) is available for the Caller ID which allows you to select the type of Caller ID signal from an analog trunk – FSK or DTMF.

Conditions

- To have pre-answer Caller ID from the voice mail, the call must be an unscreened transfer.
- Caller ID is provided by the CPUII. The DSPII daughter board, which plugs onto the CPUII, can provide additional resources for Caller ID if needed.
- Caller ID Name can display up to 12 characters.
- Caller ID Number can display up to 11 characters.
- A Caller ID Number with more than 12 digits follows PRG 20-19-01 (first 10 or the last 10 digits).
- Caller ID information can be stored in Speed Dialing or One-Touch bins.
- Caller ID can be displayed for incoming calls and transferred calls.
- ARS can block outgoing Caller ID information call-by-call. To do this, insert the Caller ID block code (e.g., .67) in the ARS Dial Treatments.
- Trunks with Privacy Release enabled display Caller ID until the call is answered. To view it after the call is picked up, press the line key, which sets the call to private mode. To keep the call on Privacy Release, press the Help + Exit keys.
- An extension user can display the Caller ID information for a call in Park if Automatic Handsfree in PRG 15-02-08 is set to 0 (Preselect).
- An extension user can display the Caller ID information for multiple incoming calls without answering the call by pressing the line key if Automatic Handsfree in Program 15-02-08 is set to 0 (Preselect).
- Caller ID information outputs on the SMDR report.
- ANI/DNIS can use the Caller ID tables for routing. Refer to [T1 Trunking \(with ANI/DNIS Compatibility\) on page 2-941](#) for more information.
- The system can send Caller ID digits to the voice mail if allowed in PRG 14-02-10.
- When more than 20 characters are set in PRG 20-20 : Message Setup for Non-Caller ID Data, either the first or last character is missing (based on the entry in PRG 20-19-01).
- If Program 20-09-06 : Class of Service Options (Incoming Call Service) : Incoming Time Display is set to 1 (On), the first line displays the time and date.
- When you shut down the system, incoming history data is cleared. But you can press **Speaker + # * # 9** to back up the history data.

Default Setting

Disabled

System Availability

Terminals

All Multiline Terminals with a display and Single Line Telephones equipped to receive Caller ID

Required Component(s)

- COIB(4)-U() ETU (in COID mode)
- COIB(8)-U() ETU (in COID mode)
- BRT(4)-U(20) ETU with **Firmware 3.00 or higher**
- DTI-U40 ETU with **Firmware 5.0 or higher**
- SLIB/E-U() (in COID mode)

Related Features

Automatic Route Selection

Call Arrival (CAR) Keys

Caller ID Call Return

Conference, Voice Call/Privacy Release

Cordless II/Cordless Lite II Telephones

One-Touch Calling

Park

Speed Dial – System/Group/Station

Station Message Detail Recording

T1 Trunking (with ANI/DNIS Compatibility)

Voice Mail Message Indication on Line Keys

Programming

- ➔ **10-02-04 Location Setup – Area Code (Version 1600 and 2100 or higher required)**
Up to 6 digits (0~9, *, #) (default not assigned).
- ➔ **10-02-05 Location Setup – Trunk Access Code**
For Electra Elite IPK II Wireless users, enter the trunk access code digits required to place an outgoing call (eight digits maximum: 0~9, *, #). This code is added to the Caller ID information for incoming ISDN trunk calls to allow the call to dial out.
- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**
Allocate the circuits (1~16 or 1~64) on the CPUII ETU for either DTMF receiving or dial tone detection (0 = Common Use, 1 = Extension Only, 2 = Trunk Only).
- ➔ **11-15-03 Service Code Setup, Administrative (for Special Access) – Backup Data Save**
Back up the programmed data on the SRAM and Call History to the Flash ROM. While saving the database, it may cause system lock up.
- ➔ **14-01-20 Basic Trunk Data Setup – Block Outgoing Caller ID**
Allow (1)/Prevent (0) the system from automatically blocking outgoing Caller ID information when a user places a call. If allowed (i.e., block enabled), the system automatically inserts the Caller ID block code *67 (defined in Program 14-01-21) before the user dialed digits (this requires Program 14-02-10 to be enabled). If prevented (i.e., block disabled), the system outdials the call just as it was dialed by the user.

Trunks: 1~200
- ➔ **14-01-21 Basic Trunk Data Setup – Caller ID Block Code**
Enter the code, up to eight digits, that should be used as the Caller ID Block Code. This code is automatically inserted before dialed digits if Program 14-01-20 is set to 1.

Trunks: 1~200
- ➔ **14-01-22 Basic Trunk Data Setup – Caller ID to Voice Mail**
Enable (1)/Disable (0) the system ability to send the Caller ID digits to voice mail.

Trunks: 1~200
- ➔ **14-02-10 Analog Trunk Data Setup – Caller ID**
Enable (1)/Disable (0) a trunk ability to receive Caller ID name/number information. If this option is set to 1, the system suppresses ringing and lighting for trunk calls for the first ring to receive the Caller ID information.

Trunks: 1~200

✎ When the destination is Voice Mail, Single Line Telephone with Caller ID or ACD, the system suppresses ringing and lighting to receive the Caller ID. When the incoming call destination is a Multiline Terminal or Normal Single Line Telephone (SLT with no Caller ID) the system does not suppress ringing and lighting.

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- ➔ **14-02-16 Analog Trunk Data Setup – Caller ID Type**
Select the type of Caller ID signal from an analog trunk – FSK (0) or DTMF (1). By default, this option is set to 0.

Trunks: 1~200
 - ➔ **15-02-08 Multiline Telephone Basic Data Setup – Automatic Handsfree**
Set whether pressing a key accesses a One-Touch Key (1) or preselects the key (0).
 - ➔ **15-02-40 Multiline Telephone Basic Data Setup – Additional Dial for Caller ID Call Return (Version 1600 and 2100 or higher required)**
Up to eight digits (0, 1~9, #, *) (default not assigned).
 - ➔ **15-03-09 Single Line Telephone Basic Data Setup – Caller ID Function – For External Module**
Enable (1)/Disable (0) the Caller ID FSK signal for an external Caller ID module or a 3rd party vendor telephone with Caller ID display. If voice mail is used, this setting must be disabled or the system integration codes for disconnect are incorrect. For Caller ID Sender Queuing, set this option to 1.
 - ➔ **15-03-10 Single Line Telephone Basic Data Setup – Caller ID Name**
Determine whether or not a single line telephone extension user telephone should display the Caller ID name (0 = Disable, 1 = Enable).
 - ➔ **15-03-11 Single Line Telephone Basic Data Setup – Caller ID Type**
Select the type of Caller ID signal sent from an SLT extension port – FSK (0) or DTMF (1). By default, this option is set to 0.
 - ➔ **15-07-01 Programmable Function Keys**
Assign an Outgoing Call Without Caller ID (ISDN) function key (63) or a Caller ID Log History key (08) if needed.
 - ➔ **20-02-08 System Options for Multiline Telephones – LCD Display Holding Time**
Determine the time a user display shows Caller ID for a second incoming call (0~64800 seconds).
 - ➔ **20-06-01 Class of Service for Extensions**
Assign Class of Service (1~15) to extensions.
 - ➔ **20-08-15 Class of Service Options (Outgoing Call Service) – Block Outgoing Caller ID**
Allow (1)/Prevent (0) a user Class of Service from automatically blocking outgoing Caller ID information when a call is placed. If block is enabled, the system automatically inserts the Caller ID block code *67 (defined in Program 14-01-21) before the user dialed digits (this requires PRG 14-02-10 to be enabled). If block is disabled, the system outdials the call just as it was dialed by the user.
 - ➔ **20-09-02 Class of Service Options (Incoming Call Service) – Caller ID Display**
Enable (1)/Disable (0) the extension ability to display incoming Caller ID name/number information.

 *This is disabled at default.*
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- ➔ **20-09-04 Class of Service Options (Incoming Call Service) – Notification for Incoming Call List Existence**
Determine whether or not the CHECK LIST message is displayed to indicate a missed call (0 = Off, 1 = On).
 - ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension user to manually (0) or automatically (1) receive off-hook signals.
 - ➔ **20-19-01 System Options for Caller ID – Caller ID Displaying Format**
Determine whether the first 10 digits (0) or last 10 digits (1) should be displayed when Caller ID exceeds 12 digits.
 - ➔ **20-19-02 System Options for Caller ID – Caller ID Wait Timer**
Determine the time an incoming CO call should wait for Caller ID information from Telco before the system connects the call.
 - ➔ **20-19-05 System ID Options for Caller ID – Caller ID Sender Queuing Time (Sender Wait)**
With the Caller ID Sender Queuing option, determine the time an incoming call waits in queue for a DSP resource to become available (0~64800 seconds). If a resource becomes available during this time, the call immediately rings the single line telephone with Caller ID. If the time expires before a resource becomes available, the system rings the single line telephone without Caller ID (until the queuing time expires, the single line telephone does not ring). If the queuing timer is set to 0, the system does not queue the incoming call.
 - ➔ **20-20-01 Message Setup for Non-Caller ID Data – Private Call**
Enter the text (up to 24 alphanumeric characters) to be displayed for Caller ID when a user receives a call which is classified as a private call.
 - ➔ **20-20-02 Message Setup for Non-Caller ID Data – Call from Out of Service Area**
Enter the text (up to 24 alphanumeric characters) to be displayed for Caller ID when a user receives a call which is classified as an out-of-service area call.
 - ➔ **20-20-03 Message Setup for Non-Caller ID Data – Call Information with Error**
Enter the text (up to 24 alphanumeric characters) to be displayed for Caller ID when a user receives a call which is classified as a call with a CID error.
 - ➔ **90-03-01 Save Data**
Save the programmed data on the SRAM and Flash ROM to the Removable Compact Flash Memory.
 - ➔ **90-04-01 Load Data**
Load the data from inserted Compact Flash Memory to SRAM and Flash ROM.

Operation

Storing a Number

To store a Caller ID number in an Speed Dial bin:

1. With an idle multiline terminal the display shows:

```

1-01 FRI 09:00AM
301          STA 301
LIST DIR ICM PROG

```

2. Press the **LIST** Softkey. The display shows:

```

LIST MENU

Redial CID

```

3. Press the **CID** Softkey (Caller ID). The display shows:

```

##:          xxxxxxxxxxxxxx
           mm-dd hh:mm
↑   ↓      Store  DEL

```

= List Number

xx = Caller ID number

mm-dd hh:mm = incoming date and time

↑ = Preview List

↓ = Next List

Store = Store in List

DEL = Delete from List

4. Press the **STORE** Softkey. The display shows:

```

##:          xxxxxxxxxxxxxx
           mm-dd hh:mm
STA SYS

```

= List Number

xx = Caller ID number

mm-dd hh:mm = incoming date and time

STA = Store in Station Speed Dial bin.

SYS - Store in System Speed Dial bin.

5. Press the **STA** or **SYS** Softkey. The display shows:


```

Store to SYS: COMMON
ENTER BIN

```

6. Dial the Speed Dial bin in which the number is to be stored. If you press **Hold**, the next available Speed Dial bin is used. The display shows:

<p>SYS XXXX: XXXXXXXXXXXX</p>

 If all Speed Dial bins are used, the display shows *TABLE IS FULL*.

7. Press **HOLD**. The display shows:

<p>SYS XXXX -</p>

8. Enter the name to be associated with the stored number.

Table 2-5 Keys for Entering Names

Use this keypad digit . . .	When you want to . . .
1	Enter characters: 1 @ [¥] ^ _ ` { } Æ " Á À Â Ã Ç É Ê ì ó
2	Enter characters: A-C, a-c, 2.
3	Enter characters: D-F, d-f, 3.
4	Enter characters: G-I, g-i, 4.
5	Enter characters: J-L, j-l, 5.
6	Enter characters: M-O, m-o, 6.
7	Enter characters: P-S, p-s, 7.
8	Enter characters: T-V, t-v, 8.
9	Enter characters: W-Z, w-z, 9.
0	Enter characters: 0 ! “ # \$ % & ’ () ô Õ ú ä ö ü α ε θ
*	Enter characters: * + , - . / : ; < = > ? π Σ σ Ω ∞ € £
#	# = Accepts an entry (only required if two letters on the same key are needed - ex: TOM). Pressing # again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.)
CONF	Clear the character entry one character at a time.
HOLD	Clear all the entries from the point of the flashing cursor and to the right.


9. Press **Transfer**. The display shows:

```

SET SYS

```

10. Press **Speaker**.

 *The telephone returns to idle.*

To store a Caller ID number in a One-Touch key:

1. With an idle telephone the display shows:

```

1-01 FRI 09:00AM
301          STA 301
LIST DIR ICM PROG

```

2. Press the **LIST** Softkey. The display shows:

```

LISTiMENU
Redial CID

```

3. Press the **CID** Softkey (Caller ID). The display shows:

```

##:          xxxxxxxxxxxxxx
            mm-dd hh:mm
↑   ↓       Store  DEL

```

= List Number

xx = Caller ID number

mm-dd hh:mm = incoming date and time

↑ = Preview List

↓ = Next List

Store = Store in List

DEL = Delete from List

4. Press the **STORE** Softkey. The display shows:

```

##:          xxxxxxxxxxxxxx
            mm-dd hh:mm
STA  SYS

```

= List Number

xx = Caller ID number

mm-dd hh:mm = incoming date and time

STA = Store in Station Speed Dial bin.

SYS = Store in System Speed Dial bin.

5. Press the **STA** Softkey. The display shows:


```

Store to ONE-TOUCH
ENTER BIN

```

6. Press the **One-Touch** key in which the number is to be stored or dial **1~9, 0**. If you press **Hold**, the next available One-Touch key is used. The display shows:

<p>Key ##: XXXXXXXXXXXX</p>

 *If all One-Touch keys are used, the display shows TABLE IS FULL.*

7. Press **Hold**. The display shows:


<p>KEY ## -</p>

8. Enter the name to be associated with the stored number. Refer to [Table 2-5 Keys for Entering Names on page 2-178](#).

9. Press **Hold**. The display shows:

<p>KEY PROG ONE TOUCH</p>


10. Press **Speaker**.

 *The telephone returns to idle.*

Temporary Memory

An unanswered call causes the Call History key (PRG 15-07 or SC 751: 08) to flash, indicating a new call has been placed in the temporary memory. If enabled in programming, the telephone display shows CHECK LIST.

1. Press the **Call History** key (PRG 15-07 or SC 751: 08) or press the **LIST** Softkey and CID.


 *The last addition to the list is displayed.*

2. Press the **ARROW DOWN** Softkey to scroll through the list of numbers in memory.


3. Press the **DEL** Softkey to delete the entry and scroll to the next entry.

4. The **Call History** key remains on as long as entries remain in memory.

5. To place a call back to a number in the temporary memory list, with the number to be dialed displayed, press a line key or **Speaker**. (Refer to [Table 2-5 Keys for Entering Names on page 2-178](#).)

 *The outgoing call is placed.*

To display Caller ID for a call in Park:

 *Program 15-02-08 is set to 0 (preselect) for this feature.*

1. *With Program 15-02-08 set to 0 (preselect) and a call in park, press the **PARK** key. (PRG 15-07 or SC 752: *04).*


- OR -


*With Program 15-02-08 set to 1 (One-Touch) and a call in park, press **RECALL** then the **PARK** key (PRG 15-07 or SC 752 *04).*

Checking your Answered/Unanswered Caller ID Calls**To review the last 50 outside calls your extension received:**

1. At a display multiline terminal, press the **LIST** Softkey.

2. Press **CID**.

 *The first row of your display shows the Caller ID number. If an * is next to the call record number in the left-hand corner, this indicates that it is a call you missed (unanswered). The second row shows the date and time of the call.*

 *Press the up and down softkeys to see the list of calls available in the buffer.*

3. If the Caller ID includes a name, you can press the **HELP** key to view the number of the caller.
4. To call the displayed number, press a **line/Call Appearance (CAP)** Key.

- OR -

To erase the displayed number without returning the call, press the **DEL** Softkey.

5. Press **Speaker** to hang up.

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Caller ID Call Return

Description

The Caller ID Call Return feature allows the voice mail system to use Caller ID information captured with the message to call and connect the person that left the message with the voice mail user that is checking messages. After the call is ended by either party, the voice mail user returns to checking messages.

FMS Voice Mail System Software Q revision 05931 database version 6.68 or higher is required.

VMS Voice Mail System Software Q revision 00931 database version 6.68 or higher is required.

Conditions

- A caller using a telephone without Softkeys, calling from outside the system, or from a remote system is prompted to hear Caller ID information and return a call.
- Return Call is available for subscriber messages and public messages.
- Return Call is accessible to a subscriber during and after message playback.
- Return Call is available for new and old messages.
- Return Call is accessible to a subscriber using Softkeys in Softkey mode or using DTMF in voice conversation Mode.
- One minute before disconnecting the original caller, voice mail plays a warning prompt and immediately before disconnecting plays a prompt to indicate that it is returning to the subscriber mailbox.
- When a subscriber listens to a message from a Softkey equipped telephone, and Caller ID information is unavailable, the voice mail system leaves the second line of the LCD blank. When Caller ID is disabled on the system, voice mail displays the message count.
- From the subscriber options Softkey menu, a subscriber can access a Softkey menu that allows selection of name or number to be displayed on the LCD during message playback. The default is name. Voice mail uses this setting to determine the initial display on the LCD during playback.
- Voice mail continues to display Caller ID on the LCD while the post-message playback menu is still displayed on a telephone equipped with Softkeys.
- During return call, the voice mail port is in conference with the box owner and messages.
- VMS revision Q008XX v 6.68 and FMS revision Q058XX v 6.68 or higher are also required.

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- When Centralized Voice Mail is used, the remote voice mail user gets only Caller ID number when voice mail answers incoming CO calls and performs an Await-Answer transfer to the remote user. A Call that forwards to voice mail from the remote system does not have Caller ID information.
 - Live Record is not available when using Return Call.
 - A Telephone used as an ACD Plus agent or supervisor station should not have mailboxes that support Softkeys. Softkeys can be disabled per mailbox in Access Codes Options on Easymade Application Screen, Page 5 of 6. Place * in front of Hands Free Play (Item N) to disable Softkeys for a particular station.
 - The Return Call feature is enabled per mailbox in Expanded Access Codes and can be enabled for internal numbers only or for both internal and external numbers.
 - To use this feature for long distance calls, LCR or ARS must be programmed for the voice mail ports set to dial out. Refer to the Electra Elite IPK II Programming Manual for detailed programming instructions.
 - The enhanced Caller ID is enabled by setting CID=0,0 on the Integration Options line of Easymade Switch Setup Screen, Page 1. When Caller ID is already enabled on a system that is updated, the CID = X,Y setting must be changed to CID = 0,0. Only one CID setting is allowed on this line.
 - The Return Call parameter must be entered on the Integration Options line of Easymade Switch Setup Screen, Page 1 to enable this voice mail feature. Default is RCV = 6,10 where 6 is the number of rings voice mail tries when returning a call, and 10 is the number of minutes a returned call can last.
 - A trunk access code must be entered on Easymade Switch Setup Screen, Page 1, line 9 so the Return Call feature can access a trunk to return the call. When this is not entered, the mailbox user is not prompted to return the call even when Caller ID information is available.
 - The Rings to answer field on Easymade Applications Screen, Page 2, line 15 must be set to 2 or greater because Caller ID information is provided from Telco between the first and second ring.
 - Use Program 14-01-22 Caller ID to Voice Mail to enable or disable the system ability to send the Caller ID digits to voice mail.
 - When 5-, 6-, or 7-digit station numbers are used, this feature is not available.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

- VMS(2)/(4)/(8)-U() ETU
- FMS(2)/(4)/(8)-U() ETU
- VMP(2)/(4)/(8)-U()

Related Features

Voice Mail Message Indication on Line Keys

Programming

None

Operation

None

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Central Office Calls, Answering

Description

The system provides flexible routing of incoming CO (trunks) calls to meet the exact site requirements. This lets trunk calls ring and be answered at any combination of system extensions. A maximum of 200 trunks are available. For additional information on making trunks ring, refer to [Ring Groups on page 2-845](#).

Delayed Ringing

Extensions in a Ring Group can have delayed ringing for trunks. If the trunk is not answered at its original destination, it rings the DIL No Answer Ring Group (this ring group applies to DIL or non-DIL trunks). This could help a secretary that covers calls for their boss. If the boss does not answer the call, it rings the secretary's telephone after a programmable interval.

Universal Answer

Universal Answer allows an employee to answer a call by going to any multiline terminal and dialing a unique Universal Answer code. The employee does not have to know the trunk number or dial any other codes to pick up the ringing trunk. You normally set up Universal Answer along with Universal Night Answer (refer to [Night Service on page 2-745](#)). When a Universal Night Answer call rings the External Paging, an employee can answer the call from the first available telephone. You might also want to use Universal Answer in a noisy warehouse or machine shop where the volume of normal telephone ringing is not adequate. After hearing the ringing over the Paging, an employee can then easily pick up the call from a shop telephone.

The Automatic Off-Hook Answer of Universal Answer Call options (Program 20-10-07) determines whether or not the extension has the Auto Answer feature for ringing calls. This option allows a user to lift the handset to answer a ringing call; dialing the service code is unnecessary.

Additional Trunk Ring Tones

Various ring tone patterns and melodies for incoming calls are available (Program 22-03-11); Ring Tone Patterns 1~4 and Melodies 1~5.

Sidetone Volume Setup

This allows system programming for the multiline terminal side tone volume. There are two levels, based on whether the connected trunk is a digital trunk or analog trunk.

Conditions

- The incoming ring group assignment programmed in PRG 41-03-01 overrides the setting in PRG 22-05-01.
- Ringing calls can be picked up regardless of access map programming.
- An extension user can lift the handset to answer an outside call.
- Long Conversation Cutoff can disconnect incoming and outgoing CO calls after a set time. Using the Warning Tone for Long Conversation feature allows users on outgoing calls to hear a warning tone prior to the call disconnecting.
- Line keys and Call Appearance (CAP) Keys simplify answering outside calls.
- If the Absent text message was set by the originating extension, the destination extension displays the assigned text message instead of the Reason for Transfer message.
- If an extension is assigned to a Trunk Access Map that has no access for a trunk, the extension can still retrieve parked calls on that trunk. The extension can also Group Call Pickup and Direct Call Pickup calls ringing another extension on that trunk.

Default Setting

Enabled

System Availability

Terminals

All Terminals

Required Component(s)

Any Trunk ETU (i.e., COI(4/8), COID(4/8), DID or DTI)

Related Features

Automatic Call Distribution (ACD)

Call Forwarding

Directed Call Pickup

Direct Inward Dialing (DID)

Direct Inward Line (DIL)

Direct Inward System Access (DISA)

Do Not Disturb

Group Call Pickup

ISDN Compatibility

Line Preference

Long Conversation Cutoff

Night Service

Programmable Function Keys

Selectable Display Messaging

Warning Tone for Long Conversation

Programming

- **10-03-03 ETU Setup – (for SLI PKG Setup) Transmit Gain Level (S-Level)**
- **10-03-04 ETU Setup – (for SLI PKG Setup) Receive Gain Level (R-Level)**
Customize the transmit and receive levels of the CODEC Gain Types for 500/2500 type single line telephones.
- **11-11-13 Service Code Setup (for Setup/Entry Operation) – Display Language Selection for Multiline Terminal**
Select the service code which can be used at an extension to change the displayed language on a multiline terminal display (default: 678).

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- **11-12-30 Service Code Setup (for Service Access) – Specified Trunk Answer**
If required, redefine the service code used to answer a specific trunk which is either ringing or on hold (default: 672).

 - **14-01-02 Basic Trunk Data Setup – Transmit Level**
14-01-03 Basic Trunk Data Setup – Receive Level
Customize the transmit and receive levels of the CODEC Gain Types for each trunk.

Trunks: 1~200

 - **14-02-02 Analog Trunk Data Setup – Ring Detect Type**
For each trunk, enable delayed ringing (0) or immediate ringing (1).

Trunks: 1~200

 - **14-05-01 Trunk Group – Trunk Group Number**
Assign trunks to trunk groups then go to Program 14-06-01 below to set up Trunk Group Routing.

Trunk Groups: 1~100

 - **14-06-01 Trunk Group Routing**
Set up trunk routes, and then go to Program 23-03-01 below to assign the trunk routes to extensions. An extension user can use Universal Answer to answer a call if the ringing trunk is in its assigned route.

Trunk Group Routes: 1~100

 - **14-07-01 Trunk Access Map Setup**
Set up the Trunk Access Maps. This sets the access options for trunks. Ring Group programming overrides Access Map programming.

Trunk Access Maps: 1~200

 - **15-01-05 Basic Extension Data Setup – Restriction for Outgoing Disable on Incoming Line**
Enable (1)/Disable (0) supervised dial detection for an extension.

 - **15-02-01 Multiline Telephone Basic Data Setup – Display Language Selection**
Select the language to use on a telephone display:

0 = Japanese	5 = Spanish	10 = Swedish
1 = English	6 = Dutch	11 = Turkish
2 = German	7 = Portuguese	12 = Latin American Spanish
3 = French	8 = Norwegian	
4 = Italian	9 = Danish	

 - **15-02-02 Multiline Telephone Basic Data Setup – Trunk Ring Tone**
Trunks ring extensions according to the Ring Tone Range selected in Program 22-03-01 and the settings made with either Service Code 720 or Program 15-02-02.
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- ➔ **15-02-22 Multiline Telephone Basic Data Setup – Multiple Incoming From Intercom and Trunk**
When this option is set to 0, incoming calls to an extension indicate on any Hotline key for that extension as solid (busy). When this option is set to 1, lighting is determined by the setting of Program 22-01-01. When this program is set to 1, the Hotline key lights solid when a trunk call rings in. If set to 0, the Hotline key does not light for incoming trunk calls, but lights solid for intercom calls.

 - ➔ **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**
Enter 1 to allow a single line port to receive DTMF tones after the initial call setup. Enter 0 to have the port ignore DTMF tones after the initial call setup. For Voice Mail, always enter 1 (e.g., receive DTMF tones).

 - ➔ **15-06-01 Trunk Access Map for Extensions**
Assign Trunk Access Maps to extensions.

Trunk Access Maps: 1~200

 - ➔ **15-07-01 Programmable Function Keys**
To have outside calls ring specific keys, assign trunks to line keys (code *01 + trunk). You can also program function keys as trunk group (*02) or Call Appearance (CAP) Keys (*08 + bin). Use the Call Redirect key (49) to allow a user to transfer a call to another extension or voice mail without answering the call.

Trunks: 1~100

 - ➔ **20-02-09 System Options for Multiline Telephones – Disconnect Supervision**
Enable (1)/Disable (0) disconnect supervision for the system trunks.

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

 - ➔ **20-10-07 Class of Service Options (Answer Service) – Automatic Off-Hook Answer**
Enable (1)/Disable (0) the extension user ability to use Auto-Answer to pick up Universal Answer calls. This allows a user to pick up the handset instead of dialing a service code.

 - ➔ **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**
Enable (1)/Disable (0) an extension user ability to use Continued Dialing, which allows DTMF signal sending while talking on extension.

 - ➔ **20-13-23 Class of Service Options (Supplementary Service) – Display the Reason for Transfer**
Enable (1)/Disable (0) an extension ability to display the Reason for Transfer message.

 - ➔ **21-01-15 System Options for Outgoing Calls – Outgoing Disable on Incoming Line (Toll Restriction)**
Enable (1)/Disable (0) the Outgoing Disable on Incoming Line feature.

 - ➔ **21-01-16 System Options for Outgoing Calls – Supervise Dial Detection Timer**
With the Outgoing Disable on Incoming Line feature, if dial tone is not detected after the extension answers an incoming line, the system determines the call cannot complete and releases the DTMF receiver (0~64800 seconds).
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- ➔ **21-01-17 System Options for Outgoing Calls – Restriction Digit in Outgoing Disable on Incoming Line**
With the Outgoing Disable on Incoming Line feature, determine the number of digits (0~9) to be dialed before the call should be disconnected.
 - ➔ **22-01-01 System Options for Incoming Calls – Incoming Call Priority**
Determine if intercom calls (0) or trunk calls (1) have answer priority when both are ringing simultaneously. Refer to Program 15-02-22 for lighting conditions.
 - ➔ **22-01-02 System Options for Incoming Calls – Incoming Call Ring No Answer Alarm**
Enable (1)/Disable (0) the Incoming Call RNA Alarm. If enabled, the ring cadence changes for a call that rings longer than the time set in PRG 22-01-03.
 - ➔ **22-01-03 System Options for Incoming Calls – Ring No Answer Alarm Time**
Set the Ring No Answer Alarm interval (0~64800 seconds). If a trunk rings a multiline terminal longer than this time, the system changes the ring cadence.
 - ➔ **22-01-04 System Options for Incoming Calls – DIL No Answer Recall Time**
If an incoming trunk call rings longer than this time, it reroutes to the Ring Group set in Program 22-08-01 (default = 0 seconds).
 - ➔ **22-02-01 Incoming Call Trunk Setup**
Select the trunk type (0~7) for each trunk at each time mode (1~8): (default = Normal).

0 = Normal	4 = DIL
1 = VRS	5 = E&M Tie Line
2 = DISA	6 = Delayed DID
3 = DID	7 = ANI/DNIS

Trunks: 1~200
 - ➔ **22-03-01 Trunk Ring Tone Range – Ring Tone Pattern**
Assign Ring Tone Ranges to trunks. Trunks ring extensions according to the Ring Tone Range selected in PRG 22-03-0 and the settings made with either Service Code 720 or PRG 15-02-02.
 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign extensions to ring groups.
 - ➔ **22-05-01 Incoming Trunk Ring Group Assignment**
To have trunks ring extensions, assign trunks to a Ring Group. The incoming ring group assignment programmed in PRG 41-03-01 overrides the setting in this program.
 - ➔ **22-06-01 Normal Incoming Ring Mode**
Indicate whether the trunks in the Ring Group assigned in PRG 22-04-01 should ring (1) or not ring (0).
 - ➔ **22-07-01 DIL Assignment**
Assign the destination extension for each DIL incoming trunk (001~200).
☞ For this selection to work, set Program 22-02-01 to 4 = DIL.
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- ➔ **22-08-01 DIL/IRG No Answer Destination**

If an incoming trunk call rings longer than the DIL No Answer Time (Program 22-01-04), it routes to the destination you specify in this option. Determine if the destination should be a Ring Group, In-Skin/External Voice Mail, or Central Voice Mail.

Ring Groups: 1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
 - ➔ **23-03-01 Universal Answer/Auto Answer**

For Universal Answer, assign trunk routes set in Program 14-06-01 to extensions. An extension user can use Universal Answer to answer a call if the ringing trunk is in its assigned route.

Trunk Group Routes: 1~100
 - ➔ **82-08-01 Sidetone Volume Setup**

Adjust the multiline terminal side tone volume. The two levels depend on whether the connected trunk is a digital trunk or analog trunk.

Operation

To answer an incoming trunk call:

1. Lift the handset.

To use Universal Answer to answer a call ringing over the Paging system:

1. Go off-hook.
 - 📎 *Depending on system programming, this may answer the call and you can skip Step 2.*
2. Dial **#0**.
 - 📎 *If you hear error tone, your extension Class of Service prevents Universal Answer.*

To listen to the incoming trunk ring choices:

1. Press **Speaker**.
2. Dial **711 + 2**.
3. Select the ringing (**1~8**) and tone range (**1~4**) you want to check.
4. Go back to step 3 to listen to additional choices or press **Speaker** to hang up.

To change the ringing of your incoming trunk:

1. Press **Speaker**.
2. Dial **720 + 2**.

3. Select the ringing (**1~8**).
4. Press **Speaker** to hang up.

Central Office Calls, Placing

Description

The system provides flexibility in the way each extension user can place outgoing trunk calls. This lets you customize the call placing options to meet site requirements and each individual's needs. A user can do one of the following to place a call:

- Press a Line Key
- Press a Trunk Group Key
- Press a Trunk Group Routing (dial 9) Key
- Dial a code for a specific trunk (#9 + the trunk number)
- Dial a code for a Trunk Group (704 + group number)
- Dial a code for Trunk Group Routing or ARS (9)
- Dial an Alternate Trunk Route Access Code (which you must define)
- Press or Use a Speed Dial bin

There are 200 available trunks.

Trunk Port Disable

The system provides a service code (default: 645) which can be used by an extension user to block a trunk for outgoing calls. The user which busied out the trunk still has access to it. All other users are blocked from seizing it to place an outgoing call. The trunk, however, can still be answered by any user programmed with trunk access.

Sidetone Volume Setup

Allows the system programming for the multiline terminal side tone volume. There are two levels, based on whether the connected trunk is a digital trunk or analog trunk.

Conditions

- If the trunk name seize display is enabled in programming, the Call Timer starts automatically after the user places a trunk call. Disabling the trunk name seize display also disables the Call Timer.
- The system can automatically select the correct type of line to use based on the number dialed and the time.

- With Automatic Handsfree, an extension user can press a line key to place a trunk call without having to first lift the handset or press Speaker. Users without Automatic Handsfree can preselect a line key before they lift the handset or press Speaker.
- Long Conversation Cutoff can disconnect incoming and outgoing CO calls after a set time. Using the Warning Tone for Long Conversation feature allows a user on outgoing calls to hear a warning tone prior to the call disconnecting.
- An extension Toll Class of Service may prevent them from dialing certain numbers.
- Dialing 9 or any other trunk access code after dialing an extension results in termination of the intercom call and a trunk is seized.
- Phones that have an APR/APA installed do not pass voice to a trunk until the interdigit timer expires (Program 21-01-03).

Default Setting

Enabled

System Availability

Terminals

All Multiline terminals

Required Component(s)

Any Trunk ETU (i.e. COI(4/8), COID(4/8), DID or DTI)

Related Features

Alphanumeric Display/Call Timer

Automatic Route Selection

Call Appearance (CAP) Keys

Code Restriction

Dial Tone Detection

Handsfree

Long Conversation Cutoff

Microphone Cutoff

Programmable Function Keys

Trunk Group Routing

Trunk Groups

Programming

- ➔ **10-03-03 ETU Setup (for SLI PKG Setup) – Transmit Gain Level (S-Level)**
- ➔ **10-03-04 ETU Setup (for SLI PKG Setup) – Receive Gain Level (R-Level)**
For for SLI package, Customize the transmit and receive level of the CODEC Gain Type for 500/2500 type single line telephones.
- ➔ **11-01-01 System Numbering – Service Code**
Setup number to use for Service codes to access trunks.
- ➔ **11-09-01 Trunk Access Code**
If required, change the single-digit Trunk Access Code (normally 9). If you change this code, you must also review the settings in PRG 11-01-01 for the new code selected.
- ➔ **11-09-02 Trunk Access Code – 2nd Trunk Route Access Code**
Assign the Service Code set up in PRG 11-01-01 for Alternate Trunk Route Access.
- ➔ **11-10-27 Service Code Setup (for System Administrator) – Trunk Port Disable for Outgoing Calls**
Define the service code which should be used by an extension user to block a trunk from being used for outgoing calls (Default: 645).
- ➔ **11-11-13 Service Code Setup (for Setup/Entry Operation) – Display Language Selection for Multiline Terminal**
Select the service code used at an extension to change the displayed language on a multiline terminal display (default: 678).
- ➔ **11-12-01 Service Code Setup (for Service Access) – Bypass Call**
Define the service code used by an extension user to select outgoing Trunk Group (code 704).
- ➔ **14-01-01 Basic Trunk Data Setup – Trunk Name**
To make identifying calls easier, assign a name to each trunk.
- ➔ **14-01-02 Basic Trunk Data Setup – Transmit Level**
- ➔ **14-01-03 Basic Trunk Data Setup – Receive Level**
Assign a CODEC Gain Type to each trunk. This sets the gain (amplification) for the selected trunk.

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- ➔ **14-01-07 Basic Trunk Data Setup – Outgoing Calls**
For each trunk, allow (1) or prevent (0) outgoing calls.

 - ➔ **14-01-10 Basic Trunk Data Setup – DTMF Tones for Outgoing Calls**
For each trunk, Enable (1)/Disable (0) an extension user ability to hear the DTMF of the digits dialed when placing the outgoing call.

 - ➔ **14-02-05 Analog Trunk Data Setup – Dial Tone Detection for Manual Accessed Trunks**
Enable (1)/Disable (0) dial tone detection for directly accessed trunks. If disabled, the system outdials on the trunks without monitoring for dial tone.

 - ➔ **14-02-11 Analog Trunk Data Setup – Next Trunk in Rotary if No Dial Tone**
Enable (1)/Disable (0) the system ability to skip over a trunk if dial tone is not detected. This option pertains to calls placed using Call Appearance (CAP) Keys, Speed Dial, Automatic Route Selection (ARS), Last Number Redial or Save Number dialed. It does not pertain to line key or Direct Trunk Access calls.

 - ➔ **14-02-14 Analog Trunk Data Setup – Loop Start/Ground Start**
Select loop start (0) or ground start (1) for the trunk.

 - ➔ **14-05-01 Trunk Group – Trunk Group Number**
Assign trunks to Trunk Groups. You can also assign the outbound priority for trunks in the group. When a user dials the trunk group, they seize the trunks in the order specified in the outbound priority entry.

Trunks: 1~200
Trunk Groups: 1~100
📌 At default, all group are assigned to Trunk Group 1.

 - ➔ **14-06-01 Trunk Group Routing**
Set up an outbound routing table for the trunk groups you assigned in Program 14-05-01. When a user dials 9, the system routes the calls in the order (priority) specified.

Trunk Routing Table: 1~100

 - ➔ **14-07-01 Trunk Access Map Setup**
Set up the Trunk Access Maps. This sets the access options for trunks.

Trunk Access Maps: 1~200
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-

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- ➔ **15-02-01 Multiline Telephone Basic Data Setup – Display Language Selection**
Select the language used on a multiline terminal display:

0 = Japanese	5 = Spanish	10 = Swedish
1 = English	6 = Dutch	11 = Turkish
2 = German	7 = Portuguese	12 = Latin American Spanish
3 = French	8 = Norwegian	
4 = Italian	9 = Danish	

 - ➔ **15-02-08 Multiline Telephone Basic Data Setup – Automatic Handsfree**
Determine if when a One Touch/DSS key is pressed on the extension phone it automatically dials (1), or is preselected (0) and you have to go off-hook to dial.

 - ➔ **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**
Enter 1 for this option to allow a single line port to receive DTMF tones after the initial call setup. Enter 0 to have the port ignore DTMF tones after the initial call setup. For Voice Mail, always enter 1 (e.g., receive DTMF tones).

 - ➔ **15-06-01 Trunk Access Map for Extensions**
Assign Trunk Access Maps to extensions.

Trunk Access Maps: 1~200

 - ➔ **15-07-01 Programmable Function Keys**
To simplify placing calls, assign function keys for placing trunk calls: Line keys (code *01 + trunk), Trunk Group keys (code *02 + group), Call Appearance (CAP) Keys (code *08 + key #).

Trunks: 1~200

 - ➔ **20-02-06 System Options for Multiline Telephones – Preselection Time**
Set the preselection time (0~64800 seconds). When a multiline terminal user preselects a line key, the system remembers the preselection for this time.

 - ➔ **20-02-09 System Options for Multiline Telephones – Disconnect Supervision**
Enable (1)/Disable (0) disconnect supervision for the system trunks.

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to extensions.

 - ➔ **20-07-12 Class of Service Options (Administrator Level) – Trunk Port Disable**
For each Class of Service (1~15), enable (1)/disable (0) an extension user ability to use the Trunk Port Disable feature.

 - ➔ **20-08-02 Class of Service Options (Outgoing Call Service) – Trunk Outgoing Calls**
Enable (1)/Disable (0) trunk calling.

 - ➔ **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**
Enable (1)/Disable (0) an extension user ability to use Continued Dialing which allows DTMF signal sending while talking on extension.
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- ➔ **21-01-15 System Options for Outgoing Calls – Outgoing Disable on Incoming Line (Toll Restriction)**
Enable (1)/Disable (0) the Outgoing Disable on Incoming Line feature.
 - ➔ **21-01-16 System Options for Outgoing Calls – Supervise Dial Detection Timer**
With the Outgoing Disable on Incoming Line feature, if dial tone is not detected after the extension answers an incoming line, the system determines the call is unable to complete and releases the DTMF receiver (0~64800 seconds).
 - ➔ **21-01-17 System Options for Outgoing Calls – Restriction Digit in Outgoing Disable on Incoming Line**
With the Outgoing Disable on Incoming Line feature, determine the number of digits (0~9) to be dialed before the call should be disconnected.
 - ➔ **21-02-01 Trunk Group Routing for Extensions**
Assign Program 14-06 routes to extensions. (Trunk Group Routes: 1~100)
 - ➔ **21-15-01 Individual Trunk Group Routing for Extensions**
Designate the trunk route accessed when a user dials the Alternate Trunk Route Access Code. Refer to Trunk Group Routing to set up outbound routing. (Trunk Group Routes: 1~100)
 - ➔ **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
This timer starts when a trunk begins talking with another trunk (e.g., Trunk-to-Trunk Transfer, outgoing from trunk, Tandem Trunking) (0~64800 seconds). Assign the time before a Trunk -to -Trunk call is disconnected.
 - ➔ **82-08-01 Sidetone Volume Setup**
Adjust the multiline terminal side tone volume. There are two levels, based on whether the connected trunk is a digital trunk or analog trunk.

Operation

To place a call over a trunk group:

1. Go off-hook.
 2. Dial **704**.
 3. Dial trunk group number (**001~100**).
 4. Dial the number.
- OR -
1. At the multiline terminal, press the **trunk group** key (PRG 15-07-01 or SC 751: ***02** + group).
 2. Dial the number.


To place a call using Trunk Group Routing:

1. Go off-hook.
2. Dial **9**.
 -  *If your system has an Alternate Trunk Route Access code, you may dial that instead.*
3. Dial the number.
 - OR -
 - 1. At the multiline terminal, press the **Trunk Group Routing** key (PRG 15-07-01 or SC 752: *05).
 - 2. Dial the number.

To place a call over a specific trunk:

1. Dial **#9**.
2. Dial the line number (e.g., 005 for line 5).
3. Dial the number.
 - OR -
 - 1. At the multiline terminal, press line key (PRG 15-07-01 or SC 752: 001 to 200).
 - 2. Dial the number.

To busy out a trunk from outbound usage:

1. Press **Speaker + 645 + Trunk Number (001~200) + 1**.
 -  *The user which busied out the trunk still has access to it. All other users are blocked from seizing it to place an outgoing call. The trunk, however, can still be answered by any users programmed with the trunk access.*

To Remove a Trunk from a Busied Out State:

1. Press **Speaker + 645 + Trunk Number (001~200) + 0**.

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Class of Service

Enhancements

<p>Version 1600 or higher, when a Ring Group call rings a station, a BLF Indication for this station shows idle. Prior to this release, the BLF Indication showed busy.</p>
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Description

Class of Service (COS) sets various features and dialing options (called items) for extensions. The system allows any number of extensions to share the same Class of Service. An extension can have a different Class of Service for each Night Service mode. This lets you program a different set of dialing options for daytime operation, nighttime operation and even during lunch breaks. An extension Class of Service can be changed in system programming or via a Service Code (normally 677). There are 15 available Classes of Service.

Conditions

- Before assigning a new COS, make sure the new COS matches the old COS or you may enable options, which the extension should not have or remove options, which it should have.
- An extension can have a different Class of Service for each Service mode. At default, the Mode names are assigned as follows:
 - Mode 1 = No setting
 - Mode 2 = Night
 - Mode 3 = Midnight
 - Mode 4 = Rest
 - Mode 5 = Day2
 - Mode 6 = Night2
 - Mode 7 = Midnight2
 - Mode 8 = Rest2
- Program 20-11-27: Class of Service Options (Hold/Transfer Service) – Call Park Automatically Searched is not supported in **Version 1600**.

Default Setting

- The attendant (extension 101) has Class of Service 15 in all Night Service modes. All other extensions have Class of Service 1 in all Night Service modes.
- Refer to [Table 2-6](#) through [Table 2-13](#) in this section for the default settings of the individual options.

If changing Class of Service via Service Code:

- An extension can use Service Code 677 to change another extension Class of Service (Program 20-13-28 = 1).
- An extension can automatically block another extension attempt to change their Class of Service via Service Code 677 (Program 20-13-28 = 0).
- The default Service Code for this option is 677 (Program 11-11-24 = 677).

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Night Service

Programming

- ➔ **11-11-24 Service Code Setup (for Setup/Entry Operation) – Change Station Class of Service**
If required, change the Service Code a user dials to change an extension Class of Service.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service to extensions (1~15). Any Class of Service assignment you change using Service Code 677 (see below) automatically updates this program.

- ➔ **20-07 Class of Service Options (Administrator Level)**
Refer to [Table 2-6 Class of Service Options \(Administrator Level\)](#), Program 20-07 on page 2-205.
- ➔ **20-08 Class of Service Options (Outgoing Call Service)**
Refer to [Table 2-7 Class of Service Options \(Outgoing Call Service\)](#), Program 20-08 on page 2-207.
- ➔ **20-09 Class of Service Options (Incoming Call Service)**
Refer to [Table 2-8 Class of Service Options \(Incoming Call Service\)](#), Program 20-09 on page 2-209.
- ➔ **20-10 Class of Service Options (Answer Service)**
Refer to [Table 2-9 Class of Service Options \(Answer Service\)](#), Program 20-10 on page 2-210.
- ➔ **20-11 Class of Service Options (Hold/Transfer Service)**
Refer to [Table 2-10 Class of Service Options \(Hold/Transfer Service\)](#), Program 20-11 on page 2-211.
- ➔ **20-12 Class of Service Options (Charging Cost Service)**
Refer to [Table 2-11 Class of Service Options \(Charging Cost Service\)](#), Program 20-12 on page 2-214.
- ➔ **20-13 Class of Service Options (Supplementary Service)**
Refer to [Table 2-12 Class of Service Options \(Supplementary Service\)](#), Program 20-13 on page 2-215.
- ➔ **20-14 Class of Service Options for DISA/E&M**
Refer to [Table 2-13 Class of Service Options for DISA/E&M Service](#), Program 20-14 on page 2-219.

If changing Class of Service via Service Code:

- ➔ **20-13-28 Class of Service Options (Supplementary Service) – Allow Station COS to be Changed**
Permit (1)/Block (0) an extension user ability to change this extension Class of Service via Service Code 677.

Table 2-6 Class of Service Options (Administrator Level), Program 20-07

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
01	Manual Night Service Enabled Turn off or on an extension user ability to use manual Night Service Switching.	0 = Off 1 = On	0	1	11-10-01
02	Changing the Music on Hold Tone Turn off or on an extension user ability to change the Music on Hold tone.	0 = Off 1 = On	0	1	11-10-02

Table 2-6 Class of Service Options (Administrator Level), Program 20-07 (Continued)

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
03	Time Setting Turn off or on an extension user ability to set the Time via Service Code 728.	0 = Off 1 = On	1	1	11-10-03
04	Storing Speed Dialing Entries Turn off or on an extension user ability to store System or Group Speed Dialing numbers.	0 = Off 1 = On	1	1	11-10-04
05	Set/Cancel Automatic Trunk-to-Trunk Transfer Turn off or on an extension user ability to use the trunk-to-trunk transfer service codes.	0 = Off 1 = On	0	0	11-10-06 11-10-07 11-10-08
10	Programmable Function Key Programming (Appearance Level) Turn off or on an extension user ability to program their Appearance function keys using Service Code 752 (by default).	0 = Off 1 = On	1	1	20-13-18
11	Forced Trunk Disconnect (analog trunk only) Turn off or on an extension user ability to use Forced Trunk Disconnect.	0 = Off 1 = On	0	1	
12	Trunk Port Disable Turn off or on an extension user ability to busy out a trunk. using Service Code 645. The user which busied out the trunk still has access to the trunk for placing outgoing calls. All other users are blocked from seizing the trunk to place an outgoing call. The trunk, however, can still be answered by any user programmed with the trunk access.	0 = Off 1 = On	0	1	11-10-27
13	VRS Record (VRS Msg Operation) Turn off or on an extension user ability to dial service code 616 and record, erase and listen to VRS messages.	0 = Off 1 = On	0	1	11-10-20
14	VRS General Message Play Turn off or on an extension user ability to dial 4 or Service Code 611 and listen to the General Message.	0 = Off 1 = On	0	1	11-10-21
15	VRS General Message Record/Delete Turn off or on an extension user ability to dial Service Code 612 and record, listen to or erase the General Message.	0 = Off 1 = On	0	1	11-10-22
18	SMDR Printout Accumulated Extension Data	0 = Off 1 = On	0	1	11-10-23

Table 2-6 Class of Service Options (Administrator Level), Program 20-07 (Continued)

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
19	SMDR Printout Department Group (STG) Data	0 = Off 1 = On	0	1	11-10-24
20	SMDR Printout Accumulated Account Code Data	0 = Off 1 = On	0	1	11-10-25
21	Register/Delete DECT Turn off or on an extension user ability to register or unregister an Electra Elite IPK II Wireless – DECT handset using the service codes defined in PRG 11-10-30 and PRG 11-10-31.	0 = Off 1 = On	0	0	11-10-30 11-10-31
23	CO Message Waiting Indication Callback Number Programming	0 = Off 1 = On	0	0	
24	Set/Cancel Private Call Refuse	0 = Off 1 = On	0	0	
25	Set/Cancel Caller ID Refuse	0 = Off 1 = On	0	0	
26	Dial-In Mode Switch	0 = Off 1 = On	0	0	
27	Do-Not-Call Administrator	0 = Off 1 = On	0	0	

Table 2-7 Class of Service Options (Outgoing Call Service), Program 20-08

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
01	Intercom Calls Turn off or on Intercom calling for the extension.	0 = Off 1 = On	1	1	
02	Trunk Outgoing Calls Turn off or on outgoing trunk calling for the extension.	0 = Off 1 = On	1	1	
03	System Speed Dialing	0 = Off 1 = On	1	1	11-12-10
04	Group Speed Dialing	0 = Off 1 = On	1	1	11-12-11

Table 2-7 Class of Service Options (Outgoing Call Service), Program 20-08 (Continued)

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
05	Dial Number Preview (Preset Dial) Turn off or on an extension user ability to use Dial Number Preview.	0 = Off 1 = On	1	1	
06	Toll Restriction Override Turn off or on Toll Restricting Override (Service Code 663).	0 = Off 1 = On	0	0	21-01-07, 21-07 11-11-36
07	Repeat Redial Turn off or on an extension user ability to use Repeat Redial.	0 = Off 1 = On	1	1	
08	Toll Restriction Dial Block Turn off or on an extension user ability to use Dial Block.	0 = Off 1 = On	0	0	
09	Hotline/Extension Ringdown Turn off or on Ringdown Extension for extensions with this COS.	0 = Off 1 = On	0	0	
10	Signal/Voice Call Turn off or on an extension user ability to force Handsfree Answerback or Forced Intercom Ringing for outgoing Intercom calls.	0 = Off 1 = On	1	1	
11	Protect for the Call Mode Switching from Caller When an extension is set to ring mode for ICM calls, enable this option to prevent a caller from changing the call to voice announce mode.	0 = Off 1 = On	0	0	
12	Department Group Step Calling Turn off or on an extension user ability to use Department Group Step Calling.	0 = Off 1 = On	1	1	
13	ISDN CLIP Determine if the ISDN calling line identity presentation and screening indicators are allowed.	0 = Off 1 = On	0	0	
14	Call Address Information	0 = Off 1 = On	0	0	
15	Block Outgoing Caller ID Turn off or on the system ability to automatically block outgoing Caller ID information when a user places a call. If this option is on, the system automatically inserts the Caller ID block code *67 (defined in PRG 14-01-21) before the user dialed digits.	0 = Off 1 = On	0	0	14-01-20 14-01-21

Table 2-7 Class of Service Options (Outgoing Call Service), Program 20-08 (Continued)

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
16	Display E911 Dialed Extension Name and Number Turn off or on an extension ability to display the name and number of the extension that dialed 911.	0 = Off 1 = On	0	0	
17	ARS Override of Trunk Access Map Turn off or on an extension ability to override the trunk access map programming for outgoing calls.	0 = Off 1 = On	0	0	
19	Hotline for SPK Turn off or on an extension user ability to activate hotline or ringdown when pressing Speaker.	0 = Off 1 = On	0	0	
20	Hot Key Pad (Version 1500 or higher required)	0 = Off 1 = On	0	0	
21	Automatic Trunk Seizing by Pressing Speaker Key (Version 1500 or higher required)	0 = Off 1 = On	0	0	
22	Voice Over to busy Virtual Extension (Version 1600 or higher required)	0 = Off 1 = On	0	0	

Table 2-8 Class of Service Options (Incoming Call Service), Program 20-09

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
01	Second Call for DID/DISA/DIL/E&M Override Turn off or on the extension ability to receive a second call from a DID, DISA, DIL, or Tie Line caller. <i>With this option set to 1, the destination extension must be busy for a second DNIS caller to ring through. If the destination extension does not have a line or Call Appearance (CAP) Keys available for the second call and a previous call is ringing the extension but has not yet been answered, the second caller hears busy regardless of this program setting.</i>	0 = Off 1 = On	1	1	
02	Caller ID Display Turn off or on the Caller ID display at an extension.	0 = Off 1 = On	0	0	
03	Sub Address Identification Define whether or not an extension displays the Caller Sub-Address.	0 = Off 1 = On	0	0	

Table 2-8 Class of Service Options (Incoming Call Service), Program 20-09 (Continued)

Item No.	Item	Input Data	Default		Related Program
			COS 01~14	COS 15	
04	Notification for Incoming Call List Existence Determine whether or not an extension displays Check List when an incoming call is missed by a user.	0 = Off 1 = On	0	0	20-09-02
05	Signal/Voice Call Turn off or on an extension user ability to enable Handsfree Answerback or Forced Intercom Ringing for their incoming Intercom calls.	0 = Off 1 = On	1	1	11-11-15 11-11-16
06	Incoming Time Display If this is set to 1, the Incoming Call Time is displayed on the multiline terminal LCD while the telephone is ringing.	0 = Off 1 = On	0	0	
07	Call Queuing Turn off or on and extension user ability to have calls queued if a call rings the extension when it is busy.	0 = Off 1 = On	1	1	20-13-06
08	Calling Party Information Turn off or on and extension user ability to display calling party information on CCIS calls.	0 = Off 1 = On	1	1	50-02-05

Table 2-9 Class of Service Options (Answer Service), Program 20-10

Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
01	Group Call Pickup (Within Group) Turn off or on Group Call Pickup for calls ringing an extension Pickup Group and ring group calls (Service Code *#).	0 = Off 1 = On	1	1
02	Group Call Pickup (Another Group) Turn off or on Group Call Pickup for calls ringing outside a group (Service Code 769).	0 = Off 1 = On	1	1
03	Group Call Pickup for Specific Group Turn off or on Group Call Pickup for a specific group using service code 768.	0 = Off 1 = On	1	1
04	Telephone Call Pickup Turn off or on an extension user ability to pick up a call ringing into a Pickup Group (Service Codes *# and 756).	0 = Off 1 = On	1	1

Table 2-9 Class of Service Options (Answer Service), Program 20-10 (Continued)

Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
05	Directed Call Pickup for Own Group Turn off or on Directed Call Pickup for calls ringing an extension Pickup Group (Service Code 756).	0 = Off 1 = On	1	1
06	Meet-Me Conference and Paging Turn off or on an extension user ability to use Meet Me Conference and Paging.	0 = Off 1 = On	1	1
07	Automatic Off-Hook Answer Turn off or on an extension user ability to use Universal Auto Answer (no service code required).	0 = Off 1 = On	0	0
08	Virtual Extension Off-Hook Answer Turn off or on an extension user ability to answer an incoming call on a Call Arrival (CAR) Key by lifting the handset.	0 = Off 1 = On	0	0
09	Call Pickup Callback Turn off or on an extension user ability to use Call Pickup to Pick up Callback calls.	0 = Off 1 = On	1	0
10	Answer Preset	0 = Off 1 = On	0	0

Table 2-10 Class of Service Options (Hold/Transfer Service), Program 20-11

Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
01	Call Forward All Turn off or on an extension user ability to initiate Call Forwarding Immediate.	0 = Off 1 = On	1	1
02	Call Forward When Busy Turn off or on an extension user ability to use Call Forward When Busy.	0 = Off 1 = On	1	1
03	Call Forwarding When Unanswered Turn off or on an extension user ability to use Call Forward When Unanswered.	0 = Off 1 = On	1	1
04	Call Forwarding (Both Ringing) Turn off or on an extension user ability to activate Call Forwarding with Both Ringing.	0 = Off 1 = On	1	1

Table 2-10 Class of Service Options (Hold/Transfer Service), Program 20-11 (Continued)

Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
05	Call Forwarding With Follow Me Turn off or on an extension user ability to initiate Call Forwarding with Follow Me.	0 = Off 1 = On	1	1
06	Unscreened Transfer (Ring Inward Transfer) Turn off or on an extension user ability to use Unscreened Transfer.	0 = Off 1 = On	1	1
07	Transfer Without Holding Turn off or on an extension user ability to use Transfer Without Holding.	0 = Off 1 = On	0	0
08	Transfer Information Display Turn off or on an extension incoming Transfer pre-answer display.	0 = Off 1 = On	1	1
09	Group Hold Initiate Turn off or on an extension user ability to initiate a Group Hold.	0 = Off 1 = On	1	1
10	Group Hold Answer Turn off or on an extension user ability to pick up a call on Group Hold.	0 = Off 1 = On	1	1
11	Automatic On-Hook Transfer Turn off or on an extension user ability to use Automatic On-Hook Transfer.	0 = Off 1 = On	0	0
12	Call Forwarding Off-Premise (External Call Forwarding) Turn off or on an extension user ability to set up Call Forwarding Off-Premise for their telephone.	0 = Off 1 = On	0	0
13	Operator Transfer After Hold Callback Turn off or on an extension user ability to have a call which recalls from hold transfer to the operator.	0 = Off 1 = On	0	0
14	Trunk- to-Trunk Transfer Restriction Turn off or on the Trunk-to-Trunk Transfer Restriction option. If enabled, Trunk-to-Trunk transfer is not possible.	0 = Off 1 = On	0	0
15	VRS Personal Greeting (Message Greeting) Turn off or on an extension user ability to dial Service Code *4 7 to record, listen to or erase the Personal Greeting Message.	0 = Off 1 = On	1	1
16	Call Redirect Turn off or on an extension user ability to transfer a call to a predefined destination (such as an operator, voice mail, or another extension) without answering the call.	0 = Off 1 = On	1	1

Table 2-10 Class of Service Options (Hold/Transfer Service), Program 20-11 (Continued)

Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
17	Department Group Trunk-to-Trunk Transfer (Each Telephone Group Transfer) Turn off or on an extension user ability to set Trunk-to-Trunk Forwarding for a Department Group.	0 = Off 1 = On	1	1
18	No Recall Allow/Deny answered Transferred calls from recalling the originating extension.	0 = Off (Allow) 1 = On (Deny)	0	0
19	Hold/Extended Park Determine if an extension Class of Service should allow either a normal or extended Park.	0 = Off (Normal) 1 = On (Extended)	0	0
20	No Callback Turn off or on an extension ability to receive Callbacks.	0 = Off 1 = On	0	0
21	Restriction for Tandem Trunking on Hang Up Allow (0)/Deny (1) an extension user ability to set up a tandem/conference call automatically when they hang up.	0 = Off 1 = On	0	0
22	Restrict Unsupervised Conference Allow/Deny an extension user ability to initiate a Trunk-to-Trunk Transfer (Tandem Trunking).	0 = Off 1 = On	0	0
23	CAR/VE Call Forward Set/Cancel Allow/Deny an extension user ability to set/cancel Call Forwarding for a virtual extension.	0 = Off 1 = On	1	1
24	Trunk Park Hold Mode Define if an extensions Park & Hold mode is Exclusive or Non-Exclusive.	0 = Off (Non-Exclusive) 1 = On (Exclusive)	1	1
25	Transfer Park Call Turn on or off an extension user ability to transfer a parked call.	0 = Off 1 = On	1	1
27	Call Park Automatically Search (Version 2000 or higher required)	0 = Off 1 = On	1	1

Table 2-11 Class of Service Options (Charging Cost Service), Program 20-12

Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
01	--- Not Used ---			
02	Advice of Charge Enable/Disable a DISA or tie trunk caller ability to dial 9 for Trunk Group Routing or Automatic Route Selection (ARS).	0 = Off 1 = On	0	0
03	Cost Display ISDN-AOC	0 = Off 1 = On	1	1

Table 2-12 Class of Service Options (Supplementary Service), Program 20-13

Item No.	Item	Input Data	Default	
			COS 01-14	COS 15
01	Long Conversation Alarm Turn off or on the Warning Tone for Long Conversation (not for single line telephones).	0 = Off 1 = On	0	0
02	Long Conversation Cutoff (Incoming) Turn off or on an extension user ability to use Long Conversation Cutoff for incoming calls.	0 = Off 1 = On	0	0
03	Long Conversation Cutoff (Outgoing) Turn off or on an extension user ability to use Long Conversation Cutoff for outgoing calls.	0 = Off 1 = On	0	0
04	Call Forwarding/DND Override (Bypass Call) Turn off or on an extension user ability to use Call Forwarding/DND Override.	0 = Off 1 = On	1	1
05	Intercom Off-Hook Signaling Turn off or on an extension ability to receive off-hook signals.	0 = Off 1 = On	1	1
06	Automatic Off-Hook Signaling (Automatic Override) Allow a busy extension to manually or automatically receive off-hook signals.	0 = Off (Manually) 1 = On (Automatically)	1	1
07	Message Waiting Turn off or on an extension user ability to leave Message Waiting.	0 = Off 1 = On	1	1
08	Conference Turn off or on an extension user ability to initiate a conference or Meet Me Conference.	0 = Off 1 = On	1	1
09	Privacy Release Turn off or on an extension user ability to initiate a Voice Call Conference.	0 = Off 1 = On	1	1
10	Barge-In Monitor Enable the extension Barge-In to be speech or Monitor mode.	0 = Off (Speech) 1 = On (Monitor)	0	0
11	Room Monitor, Initiating Extension Turn off or on an extension user ability to initiate Room Monitor.	0 = Off 1 = On	0	0
12	Room Monitor, Extension Being Monitored Turn off or on an extension ability to be monitored.	0 = Off 1 = On	0	0

Table 2-12 Class of Service Options (Supplementary Service), Program 20-13 (Continued)

Item No.	Item	Input Data	Default	
			COS 01-14	COS 15
13	Continued Dialing (DTMF) Signal on ICM Calls Turn off or on an extension user ability to use Continued Dialing which allows DTMF signal sending while talking on extension.	0 = Off 1 = On	1	1
14	Department Calling (PLT No Called Extension) Turn off or on an extension user ability to call a Department Group.	0 = Off 1 = On	1	1
15	Barge-In, Initiate Turn off or on Barge-In at the initiating extension.	0 = Off 1 = On	0	0
16	Barge-In, Receive Turn off or on Barge-In at the receiving extension.	0 = Off 1 = On	0	0
17	Barge-In Tone/Display (Intrusion Tone) Turn off or on the Barge-In tone. If on, callers hear an alert tone and their display indicates the Barge-In when another extension barges into their conversation. If off, there is no alert tone or display indication.	0 = Off 1 = On	1	1
18	Programmable Function Key Programming (General Level) Turn off or on an extension user ability to program their General function keys using Service Code 751 (by default).	0 = Off 1 = On	1	1
19	Selectable Display Messaging (Text Messaging) Turn off or on an extension user ability to use Selectable Display Messaging.	0 = Off 1 = On	1	1
20	Account Code/Toll Restriction Operator Alert (Restricted Operation Transfer) Turn off or on operator alert when an extension improperly enters an Account Code or violates Toll Restriction.	0 = Off 1 = On	0	0
21	Extension Name Turn off or on an extension user ability to program its name.	0 = Off 1 = On	1	1
22	Busy Status Display (Called Party Status) Turn off or on the ability to display the detail state of called party.	0 = Off 1 = On	0	0
23	Display the Reason for Transfer Select whether or not an extension should display the reason a call is being transferred to their extension (Call Forward Busy, Call Forward No Answer, DND).	0 = Off 1 = On	0	0

Table 2-12 Class of Service Options (Supplementary Service), Program 20-13 (Continued)

Item No.	Item	Input Data	Default	
			COS 01-14	COS 15
24	Privacy Release by Pressing Line Key Turn off or on an extension user ability to press a line key to barge into an outside call. The Barge-In feature must be enabled if this option is used.	0 = Off 1 = On	0	0
26	Group Listen Turn off or on an extension user ability to use Group Listen.	0 = Off 1 = On	1	1
27	Busy on Seizing Virtual Extension If set to 1, you can call a busy extension which is talking on a virtual extension key. PRG 20-13-06 (Call Waiting) must be set to off for this option to work.	0 = Off 1 = On	1	1
28	Allow Class of Service to be Changed Turn off or on the ability of an extension COS to be changed via Service Code 177.	0 = Off 1 = On	0	0
29	Paging Display Turn off or on an extension user ability to display paging information.	0 = Off 1 = On	1	1
30	Background Music Turn off or on an extension user ability to turn Background Music on and off.	0 = Off 1 = On	1	1
31	Connected Line Identification (COLP)	0 = Off 1 = On	0	0
32	Deny Multiple Barge-Ins Turn off or on the extension ability to have multiple users Barge-In to their conversation.	0 = Off 1 = On	0	0
33	ACD Supervisor's Position Enhancement This option must be on for the operator to use service codes in PRG 11-13-10 through PRG 11-13-13.	0 = Off 1 = On	0	0
34	Block Manual Off-Hook Signaling Turn off or on an extension user ability to block off-hook signals manually sent from a co-worker.	0 = Off 1 = On	0	0
35	Block Camp-On Turn off or on an extension user ability to block callers from dialing 2 to Camp-On.	0 = Off 1 = On	0	0
36	Call Duration Timer Display Turn off or on an extension Call Timer. The system waits until the interdigit timer (PRG 21-01-01) expires before starting this timer.	0 = Off 1 = On	1	1

Table 2-12 Class of Service Options (Supplementary Service), Program 20-13 (Continued)

Item No.	Item	Input Data	Default	
			COS 01-14	COS 15
38	Headset Ringing Turn off or on an extension user ability to use the Headset ringing.	0 = Off 1 = On	0	0
39	ACD Queue Status Display Turn off or on the ACD Queue Status Display for an extension Class of Service. Any extension user with this option enabled also hears the queue alarm.	0 = Off 1 = On	0	0
40	Do Not Disturb Allow/Prevent an extension user ability to use the Do Not Disturb feature.	0 = Off 1 = On	1	1
41	Voice Mail Message Indication on DSS Turn off or on the ability for an DSS BLF key of an extension to show VM indication.	0 = Off 1 = On	0	0
42	Extension Data Swap Enabling Turn off or on an extension user ability to use Station Relocation.	0 = Off 1 = On	1	1
44	Live Monitor Enabling Turn off or on an extension user ability to use Live Monitor.	0 = Off 1 = On	1	1
45	MIC Key Mode while Call Monitoring (Call Monitor Mode) (Version 1500 or higher required)	0 = Off 1 = On	1	1
47	Station Number Display (Version 1600 or higher required)	0 = Off 1 = On	1	1
48	Station Name Display (Version 1600 or higher required)	0 = Off 1 = On	1	1
49	BLF Indication on CO Incoming State (Version 1600 or higher required)	0 = Off 1 = On	0	0
50	AIC Agent display which call is from (Version 1600 or higher required)	0 = Off 1 = On	1	1
51	Number and Name appear in the directory (Version 1600 or higher required)	0 = Off 1 = On	1	1

Table 2-13 Class of Service Options for DISA/E&M Service, Program 20-14








Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
01	First Digit Absorption (Delete First Digit Dialed) For Tie Lines, Enable/Disable the ability to absorb (ignore) the first incoming digit. Use this to make the tie trunk compatible with 3- and 4-digit Tie Line service. This option does not apply to DISA.	0 = Off 1 = On	0	0
02	Trunk Group Routing/ARS Access Enable/Disable a DISA or tie trunk caller ability to dial 9 for Trunk Group Routing or Automatic Route Selection (ARS).	0 = Off 1 = On	1	1
03	Trunk Group Access Enable/Disable a DISA or tie trunk caller ability to access trunk groups for outside calls (Service Code 704).	0 = Off 1 = On	1	1
04	Outgoing System Speed Dialing Enable/Disable a DISA or tie trunk caller ability to use System Speed Dialing.	0 = Off 1 = On	0	0
05	Operator Calling Enable/Disable a DISA or tie trunk caller ability to dial 0 for the telephone system operator.	0 = Off 1 = On	1	1
06	Internal Paging Enable/Disable a DISA or tie trunk caller ability to use the telephone system Internal Paging.	0 = Off 1 = On	1	1
07	External Paging Enable/Disable a DISA or tie trunk caller ability to use the telephone system External Paging.	0 = Off 1 = On	1	1
08	Direct Trunk Access Enable/Disable a DISA or tie trunk caller ability to use Direct Trunk Access (Service Code 715).	0 = Off 1 = On	0	0
09	Forced Trunk Disconnect <Not for ISDN T-point> Enable/Disable a tie trunk caller ability to use Forced Trunk Disconnect (Service Code *26). This option is not available to DISA callers.	0 = Off 1 = On	0	0
10	Call Forward Setting by Remote via DISA Enable/Disable a DISA callers ability to use the Call Forward service codes (PRG 11-11-01 through PRG 11-11-05).	0 = Off 1 = On	0	0
11	DISA/Tie Trunk Barge-In Enable/Disable a DISA or tie trunk caller ability to use Barge-In.	0 = Off 1 = On	0	0

Table 2-13 Class of Service Options for DISA/E&M Service, Program 20-14 (Continued)


Item No.	Item	Input Data	Default	
			COS 01~14	COS 15
12	Retrieve Park Hold Turn off or on the ability for a DISA caller to retrieve parked or held calls.	0 = Off 1 = On	1	1

Operation


To change an extension Class of Service (via Service Code 677):

1. Press **Speaker**.
2. Dial **677**.
3. Dial the extension number you want to change.
 -  *You see: MODE1:nn*
Press **Hold** to leave the current value unchanged.
The extension you dial may be set to block your attempt to change their Class of Service.
4. Enter the Day 1 Mode Class of Service for the extension you selected in step 3 and press **Hold**.
 -  *You see: MODE2:nn*
Press **Hold** to leave the current value unchanged.
5. Enter the Night 1 Mode Class of Service for the extension you selected in step 3 and press **Hold**.
 -  *You see: MODE3:nn*
Press **Hold** to leave the current value unchanged.
6. Enter the Midnight 1 Mode Class of Service for the extension you selected in step 3 and press **Hold**.
 -  *You see: MODE4:nn*
Press **Hold** to leave the current value unchanged.
7. Enter the Rest 1 Mode Class of Service for the extension you selected in step 3 and press **Hold**.
 -  *You see: MODE5:nn*
Press **Hold** to leave the current value unchanged.
8. Enter the Day 2 Mode Class of Service for the extension you selected in step 3 and press **Hold**.
 -  *You see: MODE6:nn*
Press **Hold** to leave the current value unchanged.
9. Enter the Night 2 Mode Class of Service for the extension you selected in step 3 and press **Hold**.
 -  *You see: MODE7:nn*
Press **Hold** to leave the current value unchanged.

10. Enter the Midnight 2 Mode Class of Service for the extension you selected in step 3 and press **Hold**.

 *You see: MODE8:nn*
*Press **Hold** to leave the current value unchanged.*

11. Enter the Rest 2 Mode Class of Service for the extension you selected in step 3 and press **Hold**.

 *You see: Enter Station#*

12. Go to step 3 and enter another extension number.

- OR -

Press **Speaker** to hang up.

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Clock/Calendar Display

Enhancements

Version 1600 and Version 2100 or higher:

- The default settings in Program 10-24 are updated to support the new Daylight Savings requirement.

Description

The system uses Clock/Calendar Display for:

- Central Office Calls (Access Maps)
- Class of Service (Class)
- Direct Inward Lines
- Display Telephones
- Night Service (Automatic)
- Programmable Trunk Parameters
- Ring Groups
- Station Message Detail Recording
- System Reports
- Toll Restriction (Class)
- Trunk Group Routing
- Voice Mail
- Voice Response System

Using the Daylight Savings Setup program, you can determine whether the system should automatically adjust the system time for daylight savings time/standard time changes.

Clock Adjustment

The system can be programmed to automatically adjust the system clock nightly. This feature allows you to make adjustments should the system cabinet regularly lose or gain time.

Conditions

- The system retains the Clock/Calendar Display after a power failure or system reset.
- Changing the time may change the current Class of Service (COS) service depending on the COS mode setup.
- You can program the system to automatically switch modes.
- Single line telephone users cannot set the time and date.

- Changing the system time automatically changes the IPK II In-Mail time.
- The default settings in Program 10-24 are updated in **Versions 1600 and 2100** to support the new Daylight Savings requirement.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals with a display

Required Component(s)

None

Related Features

Class of Service

Digital Voice Mail

IPK II In-Mail

Night Service

Single Line Telephones, Analog 500/2500 Sets

Voice Mail Integration (Analog)

Programming

- **10-01-01 Time and Date – Year**
Enter two digits (00~99) for the year.
- **10-01-02 Time and Date – Month**
Enter two digits (01~12) for the month.

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- ➔ **10-01-03 Time and Date – Day**
Enter two digits (01~31) for the day.
 - ➔ **10-01-04 Time and Date – Week**
Enter the digit (1 = Sunday, 7 = Saturday) to indicate the day of the week.
 - ➔ **10-01-05 Time and Date – Hour**
Enter two digits (00~23) for the hour.
 - ➔ **10-01-06 Time and Date – Minute**
Enter two digits (00~59) for the minutes.
 - ➔ **10-01-07 Time and Date – Second**
Enter two digits (00~59) for the seconds.
 - ➔ **10-24-01 Daylight Savings Setup – Daylight Savings Mode**
Enable (1)/Disable (0) the system ability to adjust the time for daylight savings/standard time.
 - ➔ **10-24-02 Daylight Savings Setup – Time for Daylight Savings**
Enter the time of day the system should adjust for daylight savings time (0000~2359).
 - ➔ **10-24-03 Daylight Savings Setup – Start of Month (Summer Time)**
Enter the month of system should adjust the time for daylight savings time (01~12).
With Version 1500 and below and Version 2000 the default = 4
With Version 1600 and Version 2100 and higher the default = 3
 - ➔ **10-24-04 Daylight Savings Setup – Start of Week**
Enter the week of the month the system should adjust the time for daylight savings time (0 = last week of the month or 1~5).
With Version 1500 and below and Version 2000 the default = 1
With Version 1600 and Version 2100 and higher the default = 2
 - ➔ **10-24-05 Daylight Savings Setup – Start of Week Day**
Enter the day of the week the system should adjust the time for daylight savings time (01 = Sunday, 02 = Monday, etc.).
 - ➔ **10-24-06 Daylight Savings Setup – End of Month**
Enter the month of system should adjust the time for standard time (01~12).
With Version 1500 and below and Version 2000 the default = 10
With Version 1600 and Version 2100 and higher the default = 11
 - ➔ **10-24-07 Daylight Savings Setup – End of Week**
Enter the week of the month the system should adjust the time for standard time (0 = Last Week of the Month or 1~5).
With Version 1500 and below and Version 2000 the default = 0
With Version 1600 and Version 2100 and higher the default = 1
 - ➔ **10-24-08 Daylight Savings Setup – End of Week Day**
Enter the day of the week the system should adjust the time for daylight savings time (01 = Sunday, 02 = Monday, etc.).

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- ➔ **20-02-07 System Options for Multiline Telephones – Time and Date Display Mode**
Select the display mode (type 1~8) for Time and Date (i.e., time and date format).
 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
 - ➔ **20-07-03 Class of Service Options (Administrator Level) – Time Setting**
Enable (1)/Disable (0) an extension user ability to set the Time.
 - ➔ **99-01-16 System Clock Adjustment**
If for any reason the system clock loses or gains time, the system can automatically adjust the system clock every midnight using this program.
 - 🔑 *This program is available only through telephone programming.*
 - 🔑 *No other programs in 99-01-xx should be adjusted. Leave all other entries at their default setting. This program is only available through telephone programming.*

0 = No adjustment (default)
1~127 = seconds to be added to clock
128~255 = seconds to be backed up on the clock (-128 to -1 seconds)

Operation

The date must be set in system programming (10-01).

To set the system Time:

1. Press **Speaker**.
2. Dial **728**.
3. Dial two digits for the hour (24 hour clock, 13 = 1:00 PM).
4. Dial two digits for the minutes (00~59).
5. Press **Speaker** to hang up.

CO Message Waiting Indication

Description

This feature provides a Message Waiting indication when Voice Mail from the Central Office is used. The CO provides this feature using Visual Message Waiting Indication (VMWI) standards. Visual Message Waiting Indication visually indicates to a user that a message is present in their voice mail box. When VMWI is provided, the Electra Elite IPK II provides a flashing LED on a line key assigned with the trunk appearance.

The VMWI standard supported by the Electra Elite IPK II includes:

- Type 1 Caller ID, FSK without power ringing using the MDMF protocol
- Type 1 Caller ID, FSK without power ringing using the SDMF protocol

Conditions

- When a new message is stored in the CO Voice Mail system, the LED flashes green (0.5 sec ON, 0.5 sec OFF) on the Direct Trunk Appearance line key at stations assigned for this feature.
- When the Direct Trunk Appearance line key is used by other ports during green blink (flutter), the line key is in use and LED is turned on red.
- When the station is using a DTH-16LD-() telephone, <> flashes on the LCD of a Direct Trunk appearance line key to indicate a new message is stored in the CO voice mail system.
- A local Voice Mail system and this feature can be supported in the same system.
- When power outage or some other reason causes the Central Office – Message Waiting Indication (CO-MWI) to be out of synchronization with the system, an Attendant Position can clear the CO-MWI per CO line.
- The CO-MWI Callback Speed Dial number uses System Speed Dial Area.
- This feature is supported at Multiline Terminals and DSS Consoles assigned with a direct line key appearance of the CO/PBX line key supporting this feature and with proper Class of Service assignment.
- When additional digits (e.g., for password) are included in the CO Message Waiting Indication System Speed Dial buffer, they must be separated by pauses to allow connection to the CO Voice Mail system.
- A Single Line Telephone or Wireless – DECT Handset cannot indicate the CO-MWI.
- The Message Display Board does not support the CO-MWI.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

COIB(4)/(8)-U30 ETU

Related Features

Battery Backup – System Power

Digital Voice Mail

Message Waiting

Speed Dial – System/Group/Station

Voice Mail Integration (Analog)

Programming

- **15-17-01 CO Message Waiting Indication – LED Flash Assignment**
Assign the Station and Trunk used for CO Message Waiting Indication (default = Not Assigned), Off = 0, On = 1.

- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) for the appropriate extensions using the CO Message waiting.

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-
- ➔ **20-07-23 Class of Service Option (Administrator Level) – CO Message Waiting Indication Callback Number Programming**
Waiting Indication Call Back Number Programming: 0 = Off (default), 1 = On.
Allow the following operations:
 - Forced LED Clear (Feature + 20)
 - MWI Callback number assignment (Feature + 28)
 - Edit Callback number digits (Feature + 29)


 - ➔ **21-22-01 CO Message Waiting Indication – Call Back Settings – CO MWI Call Back Enabling**
MWI Call Back Enabling
Enable (1)/Disable (0) VMWI Service).

 - ➔ **21-22-02 CO Message Waiting Indication – Call Back Settings – CO-MWI Call Back Number Area Setting**
Set the Trunk port number ### to the System Speed Dial Bin No [0000-1999] (default: 1999).


 - ➔ **20-02-08 System Options for Multiline Telephones – LCD Display Holding Time**
Determine the time a user display shows Caller ID for a second incoming call (default: 5 sec).

Operation


To program the CO Message Waiting Callback Speed Dial Bin from an Attendant Position:

1. Press **Feature**.
2. Dial **28**.
3. Press the **CO line**.
4. Dial Speed Dial bin (default = 1999).
 *The valid range is 0000~1999 and depends on system programming.*
5. Press **Feature**.

To program the Central Office – Message Waiting Indication callback number from an Attendant Position:

1. Press **Feature**.
2. Dial **29**.
3. Press the **CO line**.
4. Dial the Central Office – Message Waiting Indication callback number.
 *Use the Exit key to clear all digits.*


5. Press Feature.

-  *This operation updates data in PRG 13-04-01, a user can also edit the dial digits in PRG 13-04-01 from handset-programming or PCPro/WebPro.*

To retrieve a Central Office – Message Waiting Indication:**1. Press Feature.****2. Dial 27.****3. Press the CO line key.**

-  *The LCD indicates ERROR if the CO Line is not flashing for a CO Message Waiting.*

4. Listen to the message.

-  *The operation for deletion is based on the remote voice mail system.*

5. Hang up.**To clear the Central Office – Message Waiting Indication from an Attendant Position:****1. Press Feature.****2. Dial 20.****3. Press the CO line key.**

Code Restriction

Description

Code Restriction limits the numbers an extension user may dial. By allowing extensions to place only certain types of calls, you can better control long distance costs. The system applies Code Restriction according to the Code Restriction Class. The system allows for up to 15 Code Restriction Classes and 416 extensions.

Conditions

- If a Code Restriction Class has the same entries in both a permit and restriction table, the system does not restrict the call.
- Code Call Digit counting may prevent users from taking advantage of long distance automated services like ACD and automated Technical Service.
- Code Restriction is applied when accessing ARS.
- If Program 21-01-10 is set to other than 0, a call cannot have a talk path unless the user dials at least the number of digits entered in this option when placing an out going call. This means an entry of 4 or higher in this program causes a problem when dialing 911. Since it is only a 3-digit number, the call does not have a talk path, preventing the emergency dispatcher from hearing the caller. This option should be kept at its default setting of 0 to prevent any problem with dialing 911.
- Common Permit Code Table

Use the Common Permit Code Table when you have numbers you want all Code Restriction Classes to dial. To let all users dial 911, for example, put 911 in the Common Permit Code Table. The Common Permit Code Table overrides the Restrict Code and Common Restrict Code Tables. The system provides 10 tables, with 10 entries in each table. Each code is four digits maximum, using 0~9, #, * and Recall (as a wild card).

- Common Restrict Code Table

The Common Restrict Code Table lets you globally restrict certain numbers for all Code Restriction Classes. To prevent all users from dialing directory assistance (411), for example, put 411 in the Common Restrict Code Table. Be sure you do not allow the codes you want to restrict in the Permit Code Table or the Common Permit Code Table. The system provides 10 tables, with 10 entries in each table. Each code is four digits maximum, using 0~9, #, * and Recall (as a wild card).

- Restrict Code Table

When you want Code Restriction to allow most calls and restrict only selected calls, use the Restrict Code Table. To block only 1-900 calls, for example, enter 1900 in the Restrict Code Table. (If the same Code Restriction Class has both Permit and Restrict Code Tables, the system restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The system provides four tables, with 60 entries (restricted codes) in each table. A restricted code is 12 digits maximum, using 0~9, #, * and Recall (as a wild card).

- Permit Code Table

The Permit Code Table lets you set up Code Restriction so that users can dial only selected (permitted) telephone numbers. Use this table when you want to restrict most calls. To allow all users to dial only area code 203, for example, enter 1203 in the Permit Code Table. 1 + 203 + NNX + nnnn are the only numbers users can dial. (If the same Code Restriction Class has both Permit and Restrict Code Tables, the system restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The system provides four tables, with 200 entries (permitted codes) in each table. A permitted code has 12 digits maximum, using 0~9, #, * and Recall (as a wild card).

- International Call Restriction

International Call Restriction lets you limit the international calls an extension user may dial. You can build a restrict table to prevent only certain calls, or you can build a permit table to allow only certain calls. To allow most international calls, use the *International Call Restrict Table*. To prevent most international calls, use the *International Call Allow Table*. The system provides 10 International Call Restrict tables with up to four digits in each table entry and 20 International Call Allow tables, with up to six digits in each table entry. Valid entries are 0~9, #, * and Recall (for a wild card).

- Code Restriction for Speed Dialing

Speed Dialing can bypass or follow Code Restriction. If you allow many users to program Speed Dialing, consider code restricting the numbers they dial. If only administrators can program Speed Dialing, Code Restriction may not be necessary. You can separately restrict Group and Common Speed Dialing.

- Toll Digit Counting

Use Call Digit Counting to limit the number of digits local callers can dial. Use this to prevent users from accessing local dial-up services. For example, set the Maximum Number of Digits in Local Calls to seven to limit local callers to dialing the exchange code (NNX) and local address (nnnn) only. The system provides four tables in which you can make entries for this option. The range is 4~30 digits.

Code Call Digit Counting

With Code Call Digit Counting, you can limit the number of digits long distance callers can dial. This lets you prevent callers from dialing extensively into long distance dial-up services. You can make four entries (4~30 digits).

Toll Free Trunks

Certain trunks can be completely unrestricted, such as the company president's Private Line. Users can place calls on Code Free Trunks anytime – to anywhere, without inadvertently being Code restricted.

PBX Call Restriction

Code Restriction programming lets you enable/disable PBX Call Restriction and enter PBX access codes. You only need to do this if your system is behind a PBX and you have trunks programmed for behind PBX operation. Refer to [PBX Compatibility on page 2-787](#) feature for the specifics.

Additional Default Entries For Common Permit Code Table

Additional entries were added to the default Common Permit Code Table. The default setting is as follows:

- | | | |
|-------------------------------------|-------------------------------------|-------------------------------------|
| <input type="radio"/> Table 1: 911 | <input type="radio"/> Table 4: 1822 | <input type="radio"/> Table 7: 1855 |
| <input type="radio"/> Table 2: 1800 | <input type="radio"/> Table 5: 1833 | <input type="radio"/> Table 8: 1866 |
| <input type="radio"/> Table 3: 1888 | <input type="radio"/> Table 6: 1844 | <input type="radio"/> Table 9: 1877 |

Tie Line Code Restriction Enhanced

If PRG 34-01-05 : E&M Tie Line Basic Setup – System Code Restriction is set to 0, the system follows the setting in PRG 21-05-13 : Code Restriction Class – Restriction of Tie Line Calls to determine whether or not the Code restriction setting in PRG 34-08 is to be followed. If this option is set to 1, the system follows the system Code restriction settings defined in PRG 21-05-01 through PRG 21-05-13.

- A user can temporarily override an extension Code Restriction.
- The system allows or denies outgoing access to trunks depending on Code Restriction.
- If the system detects the call is answered by detecting reversal in an analog trunk this restores both-way voice paths immediately.
- When using DISA or Tie Lines, additional programming is required for Code Restriction (DISA, refer to Program 25-10; Tie Lines, refer to Program 34-04).
- A user can temporarily block their extension Code Restriction access, preventing unwanted calls from being placed on their telephone while they are away from their desk.

- A phone and a trunk have a Restriction Class. The higher class applies for outgoing calls.

For example:

- When trunk class is 01 and station class 02, Toll Restriction Class 02 is applied.
- When trunk class is 15 and station class 03, Toll Restriction Class 15 is applied.

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Central Office Calls, Placing

Code Restriction, Dial Block

Code Restriction Override

Direct Inward System Access (DISA)

PBX Compatibility

Tie Lines

Programming

- ➔ **14-01-08 Basic Trunk Data Setup – Toll Restriction**
For each trunk, enter 1 to enable Toll Restriction; enter 0 to disable Code Restriction.

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- ➔ **15-02-30 Multiline Telephone Basic Data Setup – Toll Restriction Class**
 - ➔ **20-06-01 Class of Service for Extensions**
For each extension, assign a Class of Service (1~15).
 - ➔ **20-08-02 Class of Service Options (Outgoing Call Service) – Trunk Outgoing Calls**
For each Class of Service, Enable (1)/Disable (0) an extension user ability to place an outgoing trunk call.
 - ➔ **20-13-20 Class of Service Options (Supplementary Service) – Account Code/Toll Restriction Operator Alert (Restricted Operation Transfer)**
When toll restriction is violated, determine if the operator should be notified.
Operator Alert:
0 = Disable
1 = Enable
 - ➔ **21-01-10 System Options for Outgoing Calls – Dial Digits for Toll Restriction Path**
If this option is set an entry other than 0, a call does not have a talk path unless the user dials at least the number of digits entered in this option when placing an outgoing call. This means an entry of 4 or higher in this program causes a problem when dialing 911. Since it is only a 3-digit number, the call does not have a talk path, preventing the emergency dispatcher from hearing the caller. This option should be kept at its default setting of 0 to prevent any problems with dialing 911. If the system detects the call is answered, by detecting Reversal in analog trunks, this restores both-way voice paths immediately.
 - ➔ **21-01-15 System Options for Outgoing Calls – Outgoing Disable on Incoming Line (Toll Restriction)**
Enable (1)/Disable (0) the Outgoing Disable on Incoming Line feature.
 - ➔ **21-01-16 System Options for Outgoing Calls – Supervise Dial Detection Timer**
With the Outgoing Disable on Incoming Line feature, if dial tone is not detected after the extension answers an incoming line, the system determines the call cannot complete and releases the DTMF receiver (0~64800 seconds).
 - ➔ **21-01-17 System Options for Outgoing Calls – Restriction Digit in Outgoing Disable on Incoming Line**
With the Outgoing Disable on Incoming Line feature, determine the number of digits (0~9) to be dialed before the call should be disconnected.
 - ➔ **21-04-01 Toll Restriction Class for Extensions**
For each extension, assign a Toll Restriction Class for each Night Service mode.
 - ➔ **21-05-01 Toll Restriction Class – International Call Restriction Table**
For the Toll Restriction Class you select, Enable (1)/Disable (0) the International Call Restrict Table (Program 21-06-01).
 - ➔ **21-05-02 Toll Restriction Class – International Call Permit Code Table**
For the Toll Restriction Class you select, Enable (1)/Disable (0) the International Call Permit Table (Program 21-06-02).

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-
- ➔ **21-05-04 Toll Restriction Class – Maximum Number of Digits Table Assignment**
For the Class you select, Enable (1)/Disable (0) the dialing limit for calls. When enabling, select from entries 1~4 in Program 21-06-03.
 - ➔ **21-05-05 Toll Restriction Class – Common Permit Code Table**
For the Class you select, Enable (1)/Disable (0) the Common Permit Code Table (Program 21-06-04).
 - ➔ **21-05-06 Toll Restriction Class – Common Restriction Table**
For the Class you select, Enable (1)/Disable (0) the Common Restrict Code Table (Program 21-06-05).
 - ➔ **21-05-07 Toll Restriction Class – Permit Code Table**
For the Class you select, Enable (1)/Disable (0) the Permit Code Table. When enabling, select from tables 1~4 in Program 21-06-06.
 - ➔ **21-05-08 Toll Restriction Class – Restriction Table**
For the Class you select, Enable (1)/Disable (0)) the Restrict Code Table. When enabling, select from tables 1~4 in Program 21-06-07.
 - ➔ **21-05-09 Toll Restriction Class – Restriction for Common Speed Dials**
For the Class you select, Enable (1)/Disable (0) Code Restriction for Common Speed Dialing numbers.
 - ➔ **21-05-10 Toll Restriction Class – Restriction for Group Speed Dials**
For the Class you select, Enable (1)/Disable (0) Code Restriction for Group Speed Dialing numbers.
 - ➔ **21-05-11 Toll Restriction Class – Intercom Call Restriction**
For the Class you select, Enable (1)/Disable (0) Intercom Call Restriction. If enabled, extensions cannot place or receive Intercom calls.
 - ➔ **21-05-12 Toll Restriction Class – PBX Call Restriction**
For the Class you select, Enable (1)/Disable (0)) PBX Call Restriction.
 - ➔ **21-05-13 Toll Restriction Class – Restriction of Tie Line Calls**
Enable (1)/Disable (0) the Toll Restriction set up in PRG 34-08-01 .
 - ➔ **21-06-01 Toll Restriction Table Data Setup – International Call Restriction Table**
Enter the international dialing codes you want to restrict.
 - ➔ **21-06-02 Toll Restriction Table Data Setup – International Call Permit Code Table**
Enter the international dialing codes you want to permit.
 - ➔ **21-06-03 Toll Restriction Table Data Setup – Maximum Number of Digits Table Assignment**
Select the maximum number of digits allowed in outgoing calls for each table (4~30).
 - ➔ **21-06-04 Toll Restriction Table Data Setup – Common Permit Code Table**
Program codes in the Common Permit Code Table.
 - ➔ **21-06-05 Toll Restriction Table Data Setup – Common Restriction Table**
Program codes in the Common Restrict Code Table.
-
-

-
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- ➔ **21-06-06 Toll Restriction Table Data Setup – Permit Code Table**
Program codes in the Permit Code Tables.
 - ➔ **21-06-07 Toll Restriction Table Data Setup – Deny Restriction Table**
Program codes into the Restrict Code Tables (200 codes per table maximum).
 - ➔ **21-06-08 Toll Restriction Table Data Setup – PBX Access Code**
The system allows up to four tables for PBX access codes. PBX Access Codes can have up to two digits, using 0~9, #, * and LINE KEY 1 (don't care). Refer to the PBX Compatibility feature for the specifics.
 - ➔ **21-21-01 Toll Restriction Class for Trunks (Seized Trunk Basis Setting) – Restriction Class**
Toll restriction class for trunks. For each trunk, assign a Class restriction Class for each Day/Night Service Mode.
 - ➔ **34-01-05 E&M Tie Line Basic Setup – System Toll Restriction**
Enable (1)/Disable (0) the ability for Tie Line calls to follow the system Toll Restriction entries in PRG 21-05-01 through PRG 21-05-13. If disabled, Tie Line Code Restriction is determined by PRG 21-05-13.
 - ➔ **34-08-01 Toll Restriction Data for E&M Tie Lines**
Define the Toll Restriction data for E&M Tie Lines. This data must be defined if Tie Line Code Restriction is enabled in Program 21-05-13.

Operation

To place a trunk call if your system is Code Restricted:

1. Place call normally.
 - ✎ *If your Code Restriction Class does not allow the number you dial, the call is cut off.*

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Code Restriction, Dial Block

Description

Code Restriction-Dial Block lets a user temporarily block an extension Code Restriction. This helps a user block his or her phone from being used by another person while they are away from their desk. A user needs to enter a 4-digit personal code to enable/disable this feature.

Dial Block can be set also by the supervisor's access code. If Dial Block was already set by an extension user, the supervisor cannot release it. If Dial Block is set by the supervisor's code, the extension user cannot release it.

Important: This function works by password and Class of Service control (the supervisor is not an assigned extension). If Dial Block is available for all Classes of Service, everyone may become a supervisor if they know the Dial Block password.

Conditions

- If the system is reset by a first initialization, the Dial Block feature is cleared.
- This feature is not available for ISDN S-Bus extensions.
- Both PRGs 21-09-01 (Code Restrict Class) and PRG 21-10 (Dial Block Restriction Class per Extension) can be set at the same time.
- The system gives priority to the setting in Program 20-10.
- Dial Block can temporarily block an extension Code Restriction setting by changing to a predefined table that has more restrictions.

Default Settings

Disabled

System Availability

Terminals:

None

Required Component(s)

None

Related Features


None

Programming


- ➔ **11-10-17 Service Code Setup (for System Administration) – Dial Block by Supervisor**
Assign a service code (601 by default) used by the supervisor to set Dial Block for another extension.
- ➔ **11-11-33 Service Code Setup (for Setup/Entry Operation) – Dial Block**
Assign a service code (600 by default) used for Dial Block.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a COS to each extension (1~15).
- ➔ **20-08-08 Class of Service Options (Outgoing Call Service – Toll Restriction Dial Block**
Enable (1)/Disable (0) an extension user ability to use Dial Block.
- ➔ **21-09-01 Dial Block Setup – Toll Restriction Class with Dial Block**
Assign a Code Restriction COS (1~15) when the Dial Block feature is used.
- ➔ **21-09-02 Dial Block Setup – Supervisor Password**
Assign a 4-digit password to be used by the supervisor to enable/disable Dial Block for other extensions.
- ➔ **21-10-01 Dial Block restriction Class per Extension**
Assign the Code Restriction COS (1~15) used by an extension when the Dial Block feature is enabled. If this data is 0, Code Restriction COS follows Program 21-09-01.
- ➔ **90-19-01 Dial Block Release**
Enter the extension number to be released from the Dial Block Restriction. This program can be used when a password is forgotten by the user.

Operation


To set Dial Block:

1. At the multiline terminal, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **600** (default).
3. Dial the 4-digit Dial Block Code (user defined).
4. Dial **1**.
 *Confirmation tone is heard.*
5. Press **Speaker** or replace the handset to hang up.

To release Dial Block:

1. At the multiline terminal, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **600**.
3. Dial the 4-digit Dial Block code.
4. Dial **0**.
 *Confirmation tone is heard.*
5. Press **Speaker** or replace the handset to hang up.

To set Dial Block from another extension:

1. At the multiline terminal, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **601** (default).
3. Dial the 4-digit Dial Block code (as set in programming).
4. Dial the extension number to blocked.
5. Dial **1**.
 *Confirmation tone is heard.*
6. Press **Speaker** or replace the handset to hang up.

To release Dial Block from another extension:

1. At the multiline terminal, press **Speaker**.
 - OR -
 At the single line telephone, lift the handset.
2. Dial **601**.
3. Dial the 4-digit Dial Block code.
4. Dial the extension number to be released from Dial Block.
5. Dial **0**.
 📞 *Confirmation tone is heard.*
6. Press **Speaker** or replace the handset to hang up.

Code Restriction Override

Description

Code Restriction Override lets a user temporarily bypass the Code Restriction for an extension. This helps a user that must place an important call that Code Restriction normally prevents. For example, you could set up Code Restriction to block 900 calls and then provide a Code Restriction Override code to your attendant and executives. When the attendant or executive needs to place a 900 call, they just:

- Press **Speaker**, dial a service code and enter their override code.
- Press **Speaker** and dial a trunk access code (e.g., 9 or #9 002).
- Place the 900 call without restriction.

You can assign a different Code Restriction Override code to each extension. Or, extensions can share the same override code.

Code Restriction Override overrides *all* Code Restriction programming. Walking Code Restriction allows you to assign a Code Restriction level for each user. When a call is placed using Walking Code Restriction, the restriction for the call is based on the Code Restriction level defined in Programs 21-05-xx and Programs 21-06-xx.

Conditions

- Off-Premise notification and external extensions require access to outside lines.
- In the Class heading in the SMDR report, POTA indicates that the call was placed using Code Restriction Override.
- Code Restriction Override and Walking Code Restriction temporarily overrides an extension Code Restriction.
- If the system has VRS, users hear, "Your call cannot go through. Please call the operator" when they dial a number that Code Restriction prevents.

Default Setting

Disabled

Related Features

Central Office Calls, Placing

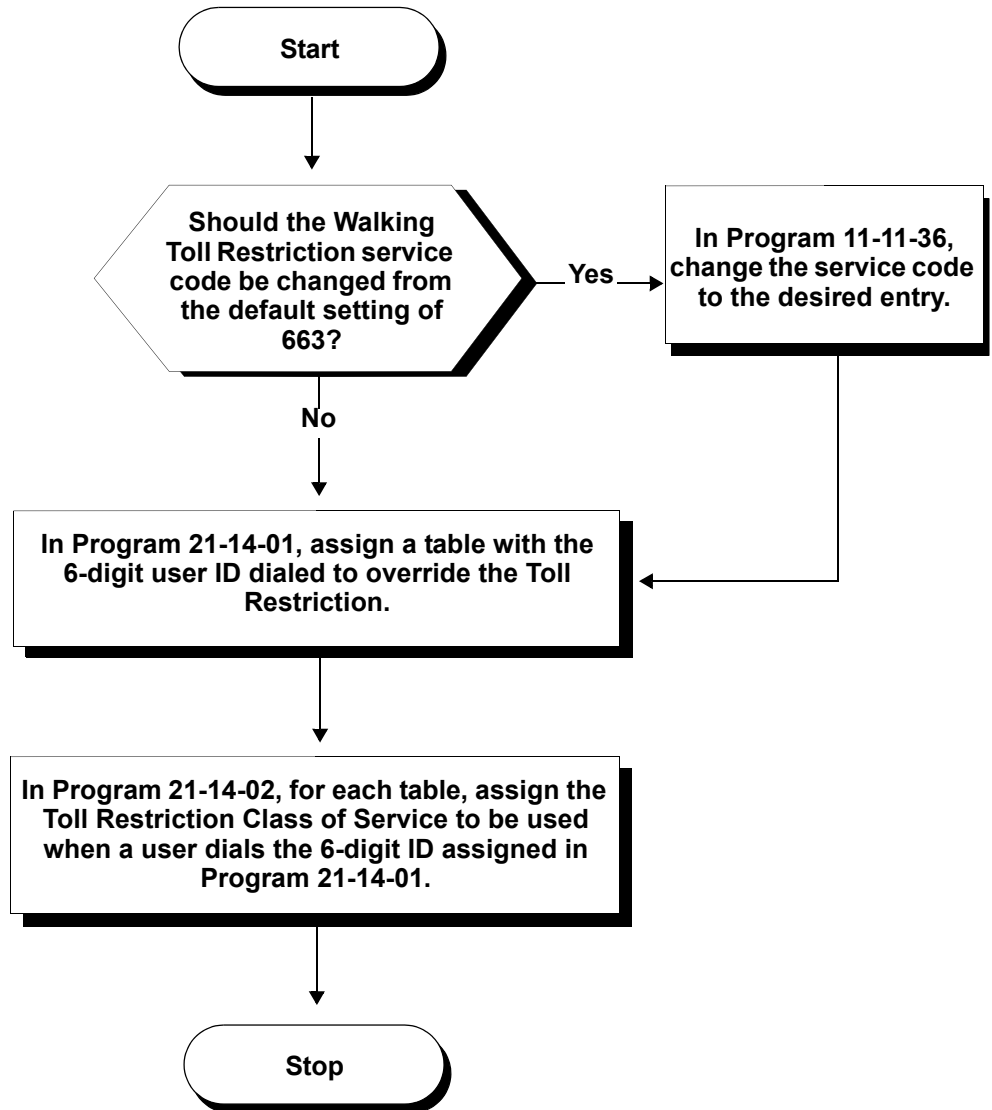
Code Restriction

Station Message Detail Recording

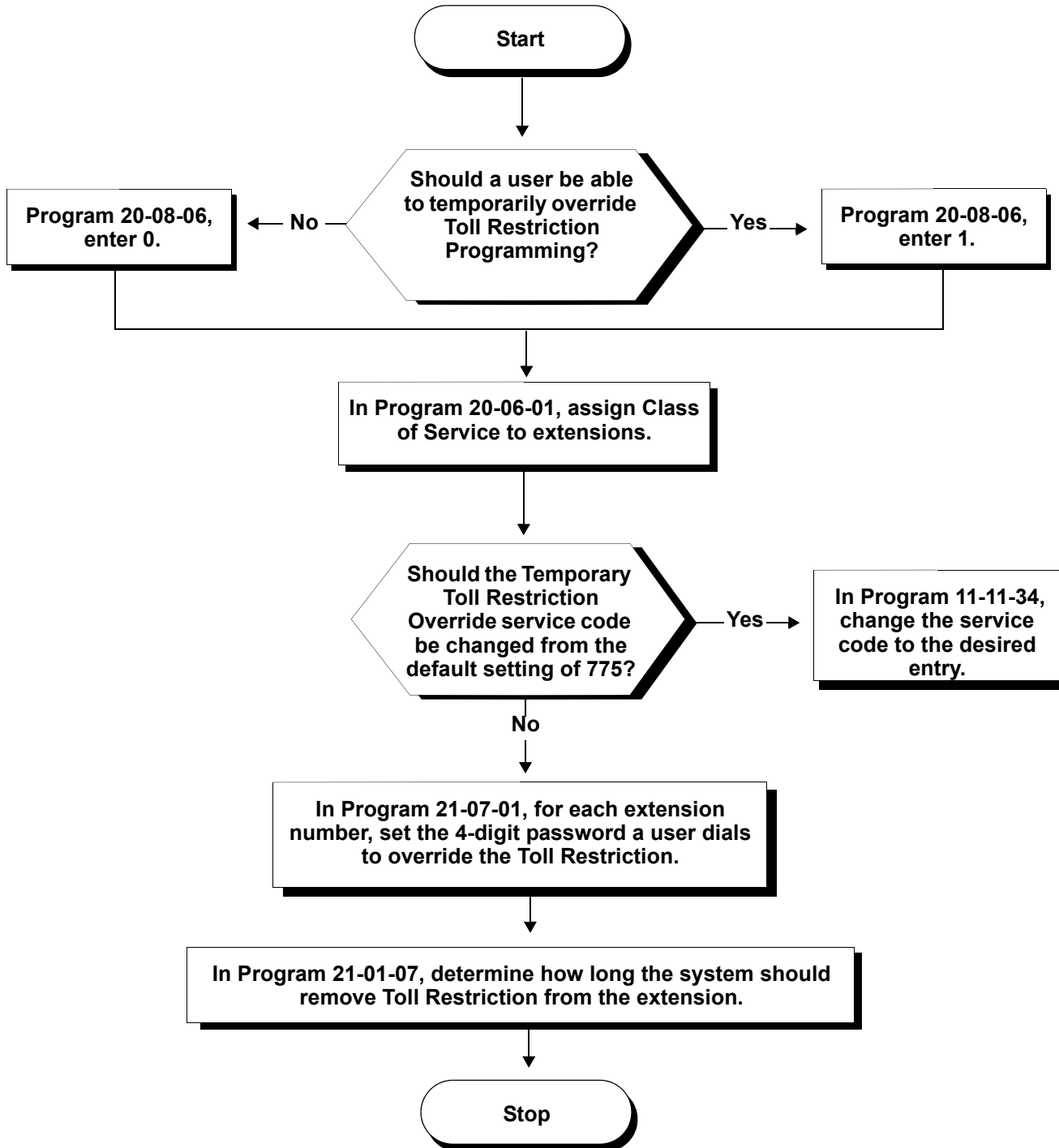
Voice Response System (VRS)

Programming

Walking Code Restriction



Temporary Code Restriction Override



Programming


- ➔ **11-11-34 Service Code Setup (for Setup/Entry Operation) – Temporary Toll Restriction Override**
If required, change the service code (775) for Temporary Toll Restriction Override.
- ➔ **11-11-36 Service Code Setup (for Setup/Entry Operation) – Toll Restriction Override**
If required, change the service code (663) for Toll Restriction Override.
- ➔ **20-08-06 Class of Service Options (Outgoing Call Service) – Toll Restriction Override**
Enable (1)/Disable (0) an extension user ability to use Temporary Toll Restriction Override.
- ➔ **21-01-07 System Options for Outgoing Calls – Toll Restriction Override Time**
Set the Toll Restriction Override Time (0~64800 seconds). After dialing the Toll Restriction Override code, the system removes Toll Restriction for this time.
- ➔ **21-07-01 Toll Restriction Override Password Setup**
Assign Toll Restriction Override codes to extensions. Each code must have four digits, using any combination of 0~9, # and *. Each extension can have a separate code, or many extensions can share the same override code.
- ➔ **21-14-01 Walking Toll Restriction Password Setup – User ID**
Enter the Walking Toll Restriction Override User ID codes (six digits) into tables. Up to 500 different override codes can be entered.
- ➔ **21-14-02 Walking Toll Restriction Password Setup – Walking Toll Restriction Class Number**
Enter the Walking Toll Restriction Class of Service (1~15) to be used for each table number assigned in PRG 21-14-01.
- ➔ **35-02-01 through 35-02-18 SMDR Output Options**
Assign the output condition for the printer or PC. If the data is set to 1 (output), the symbol of the call prints as POTA and the password table number is printed as W/@@@ (@@@ is the three digit table number) in the Account Code area.


Operation

To temporarily override a restricted extension Code Restriction:

 *You can override restriction for only one call at a time.*

1. At the multiline terminal, press **Speaker**.
- OR -
At single line telephone, lift the handset.
2. Dial **775**.
3. Dial the 4-digit Code Restriction Override code.

 *If you wait too long before going to the next step, you may have to repeat the procedure. After dialing the service code, the display indicates the override codes as they are being entered. As the last digit is entered, the display is cleared and ICM dial tone is heard.*

 *You hear error tone if you dial your code incorrectly.*

4. Press idle line key or dial trunk access code.
5. Dial the number without any restriction.

To use your Walking Code Restriction level at an extension:


 *You can override restriction for only one call at a time.*


1. At the multiline terminal, press **Speaker**.

- OR -


At the single line telephone, lift the handset.

2. Dial **663** and dial the 6-digit Walking Toll Restriction Class of Service code.

 *After dialing the service code, the display indicates the override codes as they are being entered. As the last digit is entered, the display is cleared and ICM dial tone is heard.*

 *You hear error tone if you dial your code incorrectly.*

3. Press idle line key or dial trunk access code.
4. Dial the number.

 *The call is allowed or denied based on the user Toll Restriction Class of Service level.*

Computer Telephony Integration (CTI) Applications

Description

Computer Telephony Integration (CTI) applications automate your office with TAPI compatibility and external PC control. CTI puts your telephone system on the cutting edge of modern office productivity with:

Personal Computer Interface (PCI)/PC Dialing

Use a CTA or CTU Adapter installed in your multiline terminal as a Personal Computer Interface. Installing the TAPI software driver and TAPI compatible software in your personal computer allows your PC to operate your telephone. The TAPI software driver provides all TAPI Basic Services and a host of TAPI Supplemental Services. Refer to [TAPI Compatibility on page 2-967](#) for more information.

Conditions

- The CT(U)-R is supported only on the ESIB(8)-U20 or ESIE(8)-U20.

Default Setting

None

System Availability

Terminals

None

Required Component(s)

ESIB(8)-U20/ESIE(8)-U20 [CT(U)-R only]

Related Features

TAPI Compatibility

See this feature for the details on TAPI services provided by the system and the TAPI driver.

Programming

Refer to the Electra Elite IPK II System Hardware Manual for details on connecting the CTA and CTU Adapters.

Operation

None

Conference

Enhancements

Conference Call functionality is available for Virtual Extensions with **Version 1600 or higher**.

Description

Conference lets an extension user add additional inside and outside callers to their conversation. With Conference, a user may set up a multiple-party telephone meeting without leaving the office. The CPUII provides 64 Conference ports, allowing any number of internal or external parties to be conferenced together up to a limit of 32. This means that one extension can conference up to 31 internal and/or external parties together (the originator would be the 32nd party reaching the maximum of 32). While this Conference call is active, another user can use initiate a separate Conference also up to the limit of 32 parties, or any number a conferences can be initiated with any number of parties (up to 32) until all 64 Conference ports are all busy.

Conditions

- An ADA module is required for speech recording.
- Call Arrival (CAR) Keys (virtual extensions) do not support Conference Programmable Function keys nor can they be used to add a call to a Conference.
- Split allows a user to alternate (i.e., switch) between their callers in Conference. This allows a dispatcher, for example, to control a telephone meeting between themselves, a customer and a service technician. The dispatcher can meet together with all parties, privately set up a service strategy with the technician and then meet again to set the schedule.
- Split cycles through the Conference in the same order in which the Conference was initially set up. If a user places an outside call, conferences extension 101 followed by extension 102, Split cycles from the trunk, to 101 and finally to 102. The Split cycle then repeats.
- If a user's extension has Barge-In enabled, they can also Barge-In on an established Conference. This permits, for example, an attendant or supervisor to join a Conference in an emergency. It also allows a co-worker to leave a conference – and then rejoin the telephone meeting when it is convenient to do so.
- A Class of Service option is available which allows/denies an extension user from automatically setting up a Conference/Tandem Trunking call when hanging up the telephone.

-
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- An extension with Barge-In can Transfer a call into an existing Conference. This allows, for example, an attendant to locate co-workers and then Transfer them into an existing telephone meeting. There is no need for the attendant to locate all the parties at the same time and sequentially add them into the Conference. Transfer Call Into Conference Code (624).
 - An available option allows an extension Conf key to be programmed for Conference or for Transfer. When set for Transfer, the user places a call on hold, dials the extension to which it should be transferred, then presses Conf. The call is then transferred. When set for Conference, with an active call, the user presses Conf, places a second call, then presses Conf twice. All the calls are then connected.
 - Users can Barge-In on a Conference call if allowed in programming.
 - Define the outgoing call options for each trunk and user.
 - Set up a Conference with a co-worker in your immediate work area.
 - DISA and Tie Line users may use Barge-In on a Conference call if they know the service code and are permitted in their DISA/Tie Line Class of Service.
 - Meet Me Conference lets an extension user set up a Conference via Paging.
 - Meet Me Paging lets an extension user set up a two-party meeting via Paging.
 - Call Arrival (CAR) Keys (virtual extensions) do not support Conference Programmable Function keys nor can they be used to add a call to a conference.
 - For a Multiline Terminal to have Conference, it must have a Conference function key.
 - A user can set up an Unsupervised Trunk-to-Trunk Conference and then drop out of the call, to allow the remaining parties to continue the conversation. Establish two trunk calls, press Hold, and dial #8.
 - You can optionally program Conf (Transfer) for Transfer. In this case, the Multiline Terminal must have a Conference function key. The system also allows a call to be transferred to a Conference call.
 - When the Conference Originator hangs up with a conference on Hold, or when trying to add another caller, all internal calls are dropped.
 - If a conference call with internal parties is placed on hold, and the station goes on-hook (idle) before the conference call is taken off hold, the internal parties are dropped.
 - If internal parties are placed on hold by using the conference key and the initiating station goes on-hook (idle), the internal parties on hold are dropped.
 - Conferencing when talking on a Virtual Extension (**Version 1600 or higher required**):
 - ❑ While talking on a Virtual Extension, if the station has an internal call on Hold, a conference call cannot be established.
 - ❑ While talking on a Virtual Extension, if the station receives an intercom call (call to its actual station number), a conference call cannot be established.

- While talking on a Virtual Extension, if the station has a call on Hold, a conference call cannot be established.
- Conferences can range from two groups of 32 parties, or up to 21 groups of three parties or any combination between until the maximum of 64 conference ports are used for the conferencing feature.

Default Setting

Enabled

System Availability

Terminals

Multiline and Single Line Terminals

Required Component(s)

None

Related Features

None

Programming

- ➔ 11-12-08 Service Code Setup (for Service Access) – Barge-In**
If required, change the code a user dials to Barge-In to a Conference call. This code is normally 710.
- ➔ 11-12-47 Service Code Setup (for Service Access) – Call Waiting Answer/Split Answer**
If required, change the code a user dials to Split while on a Conference call. This code is normally 794.
- ➔ 11-12-57 Service Code Setup (for Service Access) – Tandem Trunking**
With two trunks in Conference, press Hold, dial #8, and the Conference/Tandem happens.
- ➔ 11-12-58 Service Code Setup (for Service Access) – Transfer Into Conference**
If required, change the service code used to transfer a call to a Conference call (default: 624).

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- **11-16-02 Single Digit Service Code Setup – Barge-In**
If required, define the single digit service code a user dials to Barge-In to a Conference call.
 - **14-01-04 Basic Trunk Data Setup – Transmit Gain Level for Conference and Transfer Calls**
CODEC gain set at 0 dB (PRG 14-01-04 = 32) can be used to set the CODEC gain type for multiline Conference calls.
 - **15-02-24 Multiline Telephone Basic Data Setup – Conference Key Mode**
Program an extension Conf key for Conference or for Transfer. When set for Transfer (1), the user can place a call on hold, dial the extension to which it should be transferred, then press Conf. The call is then transferred. When set for Conference (0), with an active call, the user can press Conf, place a second call, then press Conf twice. All calls are then connected.
 - **15-07-01 Programmable Function Keys**
Assign a function key for Conference (code 07) and Barge-In Code (34).
 - **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
 - **20-11-21 Class of Service Options (Hold/Transfer Service) – Restriction for Tandem Trunking on Hang Up**
Allow (0)/Deny (1) an extension user ability to set up a tandem call automatically when they hang up. By default, this option is set to 0.
 - **20-13-08 Class of Service Options (Supplementary Service) – Conference**
Enable (1)/Disable (0) an extension user ability to initiate a Conference.
 - **20-13-10 Barge-In Mode per COS**
 - **20-13-15 Class of Service Options (Supplementary Service) – Barge-In Initiate**
Enable (1)/Disable (0) an extension user ability to Barge-In to a call.
 - **20-13-16 Class of Service Options (Supplementary Service) – Barge-In Receive**
Enable (1)/Disable (0) an extension user ability to receive a Barge-In on a call.
 - **20-14-11 Class of Service Options For DISA/E&M – DISA/Tie Trunk Barge-In**
Determine if a DISA or tie trunk user should be allowed to dial a service code to Barge-In to a Conference call (0 = Disable, 1 = Enable).

Operation

To establish a Conference:

Multiline Terminal


1. Establish intercom or trunk call.
2. Press **Conf** or **Conf** softkey (PRG 15-07 or SC 751: 07).
3. Dial the extension you want to add.

- OR -


Access outside call.


- OR -

Retrieve call from Park orbit.


 *To get the outside call, you can either press a line key or press the Speaker key and dial 9, the Trunk Access Code + the trunk number (default #9). You can optionally go back to step 2 to add more parties to your Conference.*


4. When called party answers, press the **Conf** softkey or **Conf** twice.

 *If you cannot add additional parties to your Conference, you have exceeded the system's Conference limit.*

 *If the call being added is busy/unanswered:*

With an outside call, press the line or Call Appearance (CAP) key for a call previously added to the Conference. The unanswered call will drop and the initiator will be back into the Conference call.

 *Adding an Intercom call to an outside Conference call: Press the **Conf** softkey on the Multiline Terminal display or **Conf** twice to re-establish the Conference. If using a non-display telephone, press **Conf** twice.*

 *With only Intercom calls in the Conference: Press **Conf** twice to re-establish the Conference. If the voice mail answers, there is no way to drop that extension out. You need to drop the Conference call.*

5. Repeat steps 2~4 to add more parties.

Single Line Telephone

1. Establish Intercom or trunk call.

2. Hookflash and dial #1.

3. Dial extension you want to add.

- OR -

Access trunk call.

- OR -

Retrieve call from Park orbit.

4. Hookflash and repeat step 3 to add more parties.

- OR -


Hookflash twice to set up the Conference.

To Split (alternate) between the parties in Conference:


Multiline_Terminal

1. Press **Conf (Transfer)** or **Conference** key (PRG 15-07 or SC 751: 07).


2. Dial Split service code (794).

-  Repeat this procedure to alternate between the remaining parties in the Conference. Press the **Conf** softkey or press **Conf** twice to set up the Conference again.


Single Line Telephone**1. Hookflash and dial 794.**

-  Repeat this procedure to alternate between the remaining parties in the Conference. Hookflash twice to set up the Conference again.

To drop an outside call from the Conference:

1. Press **Hold** to place the conference call on hold.
2. Hang up.
 -  The lines involved in the Conference ring back separately to the telephone.
3. Answer and disconnect the unwanted outside call.
4. To re-establish the Conference, press **Conf** to answer the remaining call after each call is answered. Press **Conf** twice when all calls are answered.

To exit a Conference without affecting the other parties:**With internal and outside conference members**Multiline Terminal

1. Hang up.
 -  If you press **Hold** while on a call with two outside callers, the outside callers hear what is programmed in PRG 10-04-01.

Single Line Telephone

1. Hang up.
 -  If you are not permitted to use Tandem Trunking, outside callers may hear Music on Hold.



When all conference members are outside partiesMultiline Terminal

1. Press Hold.
2. Dial # 8.
3. Hang up.


Single Line Telephone

1. Hookflash, and dial # 8.
2. Hang up.






To Barge-In to Conference Call:

1. Pick up the handset or press **Speaker**, and dial the service code (default = **710**).
 -  *If the telephone does not have the proper COS, a warning tone is sent. After the user hangs up, the system automatically places a Callback to the extension.*
2. Dial the extension number or press a **DSS** key of a telephone in a Conference call.
 -  *When a new call is added to the conference, an intrusion tone is heard by all parties in the Conference, depending on system programming, and all display multiline terminals show the joined party. If a Conference is not possible:*
 - the extension user will hear a warning tone*
 - the DISA user is rerouted to the defined ring group*
 - **OR** -
 - the Tie Line user hears a busy tone.*
 - **OR** -

The following steps are not available for DISA or Tie Line trunks:


1. Dial the extension number of the internal party.
2. Dial the single digit service code, if programmed.
 -  *Instead of the single digit service code, the service code 710 can be dialed also at this point.*

To Transfer a Call into a Conference:

1. While on a call, press **Hold**.
2. Dial the Transfer to Conference service code (default = **624**).
 -  *If the telephone does not have the proper COS, a warning tone is sent. After the user hangs up, the system automatically places a Callback to the extension.*
 -  *The display shows the line Number, Number/Name and Extension Name/Number.*
3. Dial the extension number or press a **DSS** key of a telephone within a Conference call.
 -  *If an error tone is heard, Barge-In is not enabled for the extension and the call cannot go through. Retrieve the call by pressing the flashing line or Call Appearance (CAP) Key or hang up and the call recalls the extension.*
 -  *When the call is transferred into the Conference, an intrusion tone is heard by all parties in the conference, depending on the entries in Program 20-13-17 and Program 80-01, and all display multiline terminals show the joined party.*
 -  *To cancel the transfer, press the flashing line or Call Appearance (CAP) Key to retrieve the call.*
4. Hang up.

Break Up the Conference But Retain the Trunk Calls:

1. While on an active Conference call, press **Hold**.
2. Select the line or Call Appearance (CAP) Key of the caller you wish to talk to.

 *Any intercom calls which are involved in the Conference are disconnected.*

Dropping a Party From the Conference:

1. While on an active Conference call, press the line/Call Appearance (CAP) Key of the party to retain. This action retains any intercom calls involved in the Conference.

Conference, Voice Call/Privacy Release

Description

Voice Call Conference lets extension users in the same work area join in a trunk Conference. To initiate a Voice Call Conference, an extension user just presses the Meet-Me Conference key and tells their co-workers to join the call. The system releases the privacy on the trunk, and other users can press the trunk line key to join the call. Line keys assigned for the trunk blink indicating that privacy is released, and others can join the current call.

Voice Call Conference does not use the telephone system features to announce the call. The person initiating the Voice Call Conference announces it verbally. A tone, indicating others have joined the conference, can be provided.

The CPU11 provides 64 Conference circuits, to allow any number of internal or external parties to be conferenced together up to a limit of 32.

Privacy Mode Toggle Option

This option allows an extension user to quickly change an outside call from non-private mode to private mode. If the outside call is on a line key, the user can press the line key to switch from non-private mode to private mode. For systems using the Privacy Mode Toggle option, trunks initially have the privacy released. The remainder of the call is private. If the call is on a Call Appearance (CAP) Key, the user can press their Meet-Me Conference function key instead. Unlike pressing the line key, pressing the Meet-Me Conference key toggles back and forth between private and non-private mode for the call.

Conditions

- Call Arrival (CAR) Keys and Virtual Extensions do not support Voice Call Conference Programmable Function keys.
- Voice Call Conference requires a Meet-Me Conference function key and trunk line keys.
- This feature is not available on single line telephones.
- With Caller ID enabled, a call with Privacy Release shows the Caller ID until the call is answered. To view it again, press the line key, though this sets the call to Private mode. To keep the call on Privacy Release, press the Help + Exit keys.

Default Setting

Disabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Caller ID

Conference

Programmable Function Keys

Programming

- ➔ **14-01-19 Basic Trunk Data Setup – Privacy Mode Toggle Option**
Determine if a trunk should be toggled to a private/non-private line (0 = Disabled, 1 = Enabled). This option is not required for Voice Call Conference.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for the users to set up Voice Call Conference (code 32) and line keys (*01 + trunk number) on the telephones of the invited co-workers.

(Trunks: 1~200)
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-09 Class of Service Options (Supplementary Service) – Privacy Release**
Enable (1)/Disable (0) an extension user ability to use Privacy Release allowing for a Voice Call Conference. The trunk must be programmed in Program 14-01-19 to allow this feature.

- **20-13-17 Class of Service Options (Supplementary Service) – Barge-In Tone/Display (Intrusion Tone)**
Enable (1)/Disable (0) the tone the user hears when the Voice Call Conference feature is used. People on the call do not hear the barge-in tone if disabled (the person intruding hears a beep when barging in). If enabled, both extension users and the outside caller hear the tone. (This option is also used for Barge-In.)

- **31-01-04 System Options for Internal/External Paging – Privacy Release Time**
Set the time users have to join a Voice Call Conference after it is announced. (This interval is also used for Meet Me Conference.)

Operation

To join a Voice Call Conference (if invited):

1. After Conference request, press indicated line key.
 - ✎ *Conf* is displayed on both telephones.
 - ✎ *A trunk with privacy release or Voice Call Conference blinks.*

To exit a Voice Call Conference without affecting the other parties:

1. Press **Speaker** to hang up.

To toggle between Private and Non-Private mode:

1. Press the Meet-Me Conference key (PRG 15-07-01, SC 751: 32).

- OR -

Press the Trunk Line Key. (This toggles from Non-Private to Private. To go back to Non-Private, the Meet-Me Conference Key above must be pressed.)

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Continued Dialing

Description

Continued Dialing allows an extension user to dial a call, wait for the called party to answer and then dial additional digits. This helps users that need services like Voice Mail, automatic banking and Other Common Carriers (OCCs).

There are two types of Continued Dialing:

Continued Dialing for Intercom Calls

Depending on an extension Class of Service, a multiline terminal user may dial additional digits after their Intercom call connects. In systems with Voice Mail, for example, Continued Dialing lets extension users dial the different options after the Voice Mail answers. Without Continued Dialing, extension users cannot access these Voice Mail options.

Continued Dialing for Trunk Calls

Continued Dialing gives a user access to outside services like automatic banking, an outside Automated Attendant, bulletin boards and Other Common Carriers (OCCs). After the outside service answers, the user can dial digits for whatever options the services allow. Without Continued Dialing, the system Toll Restriction cuts off the call after a specific number of dialed digits. See Programming below for additional information.

NOTICE

Continued Dialing may make the system more susceptible to toll fraud.

Conditions

- Using Continued Dialing on trunk calls is set by Toll Restriction programming.
- Continued Dialing for intercom calls only applies to calls made to analog type devices.
- With Pulse to Tone Conversion, a user can place a call to services over Dial Pulse trunks - and then dial DTMF digits after the service answers.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Pulse to Tone Conversion


Programming

- **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**
Enter 1 for this option to allow a single line port to receive DTMF tones after the initial call setup. Enter 0 to have the port ignore DTMF tones after the initial call setup. For Voice Mail, always enter 1 (e.g., receive DTMF tones).
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**
Enable (1)/Disable (0) an extension user ability to use Continued Dialing for Intercom calls to 2500 type devices.
- **21-04-01 Toll Restriction Class for Extensions**
Assign a Toll Restriction Class (1~15) to an extension.
- **21-05-04 Toll Restriction Class – Maximum Number of Digits Table Assignment**
Assign a Maximum Number of Digits Table to each Toll Restriction Class.
- **21-06-03 Toll Restriction Table Data Setup – Maximum Number of Digits Table Assignment**
Set the Maximum Number of digits dialed for each table.

Operation

To use Continued Dialing:

1. Place an intercom or trunk call.
2. Continue dialing after the call connects.

 *Toll Restriction and Class of Service programming may limit Continued Dialing.*

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Cordless Telephone Connection

Description

Using an AP(R)/AP(A)-R Unit for a DTH/DTR telephone, or an APR/APA-U Unit for a DTU/DTP telephone, a cordless telephone (2500-type) can be connected to a multiline terminal

The SLI(4)/(8)-U() ETU and the SLTII(1)-U() ADP also supports cordless telephones, but this feature refers to multiline terminal cordless connection.

Conditions

- A voice announced internal call to the multiline terminal does not ring the cordless telephone.
- Only one cordless single line telephone can be connected to an APR-U, APA-U, AP(R)-R, or AP(A)-R Unit.
- When CO Prime Line is assigned to the associated multiline terminal, internal dial tone cannot be transferred to the cordless telephone.
- The cordless telephone requires a PBR circuit while dialing. When all PBR circuits are busy, a busy tone is heard when the phone goes off-hook.
- Depending on your environment, the maximum number of cordless devices used without interference varies.
- This feature works with 2500-type cordless single line telephones.
- The multiline terminal user and the associated cordless telephone user cannot talk to each other.
- An AP(A)-R, AP(R)-R, APA-U, or APR-U Unit with hookflash enabled follows the same operating procedures as a single line terminal connected to an SLI(8)-U() ETU.
- The multiline terminal LCD displays normal information for multiline terminal when a cordless terminal is used.
- When the Multiline Terminal user goes off-hook before the cordless single line telephone user, a PBR circuit is not connected for the cordless single line telephone.
- The ring pattern for the cordless telephone can be selected by a jumper switch in the AP(R)-R or APR-U Units. The patterns are:
 - 1 second ON/2 seconds OFF
 - 2 seconds ON/4 seconds OFF
- The cordless telephone must be installed within 10 feet of the AP(R)-R Unit or APR-U Unit.

Default Setting

None

System Availability

Terminals

- Any DTH/DTR terminal with an AP(A)-R or AP(R)-R Unit except the *DTR-2DT-1 TEL*
- Any DTU/DTP terminal with an APA-U or APR-U Unit except the *DTP-2DT-1 TEL*

Required Component(s)

2500-type cordless Single Line Telephone

Related Features

Ancillary Device Connection

Programming

- ➔ 20-07 **Class of Service Options (Administrator Level)**
Refer to [Table 2-6 Class of Service Options \(Administrator Level\)](#), Program 20-07 on page 2-205.
- ➔ 20-08 **Class of Service Options (Outgoing Call Service)**
Refer to [Table 2-7 Class of Service Options \(Outgoing Call Service\)](#), Program 20-08 on page 2-207.
- ➔ 20-09 **Class of Service Options (Incoming Call Service)**
Refer to [Table 2-8 Class of Service Options \(Incoming Call Service\)](#), Program 20-09 on page 2-209.
- ➔ 20-10 **Class of Service Options (Answer Service)**
Refer to [Table 2-9 Class of Service Options \(Answer Service\)](#), Program 20-10 on page 2-210.
- ➔ 20-11 **Class of Service Options (Hold/Transfer Service)**
Refer to [Table 2-10 Class of Service Options \(Hold/Transfer Service\)](#), Program 20-11 on page 2-211.

- ➔ **20-12 Class of Service Options (Charging Cost Service)**
Refer to [Table 2-11 Class of Service Options \(Charging Cost Service\)](#), Program 20-12 on page 2-214.
- ➔ **20-13 Class of Service Options (Supplementary Service)**
Refer to [Table 2-12 Class of Service Options \(Supplementary Service\)](#), Program 20-13 on page 2-215.
- ➔ **20-14 Class of Service Options for DISA/E&M**
Refer to [Table 2-13 Class of Service Options for DISA/E&M Service](#), Program 20-14 on page 2-219.

Operation

To make a call using a cordless single line telephone:

1. Go off-hook.
2. Dial the station number or dial the Trunk Access Code and telephone number.

To answer a call using a cordless single line telephone:

When the Multiline Terminal is ringing, the incoming call can be answered by the cordless single line telephone user by going off-hook, when ringing line preference is assigned for the multiline terminal.

To transfer a call from a cordless single line telephone to its associated multiline terminal:

1. The multiline terminal user goes off-hook.
2. The single line telephone user goes on-hook (at this time, the call is automatically connected to the multiline terminal).

To transfer a call from a multiline terminal to its associated cordless single line telephone:

1. The single line telephone user goes off-hook (at this time, the call is automatically connected to the single line telephone).
2. The multiline terminal user goes on-hook.

To use Hookflash:

Refer to the operation for Single Line Telephone Access.

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Data Line Security

Description

Data Line Security protects any station port from receiving audible tones (such as Camp-On or Override) and denies a station Barge-In while busy to prevent disruption of data transmission when using a modem or facsimile machine.

Conditions

- When a multiline terminal and a single line telephone are assigned for Data Line Security, Tone Override/Voice Override and Call Alert notification tone are not heard over the handset speaker.
- Data Line Security protects a station from Barge-in, even when Barge-In is allowed in Class of Service.
- When any multiline terminal or single line telephone calls a station with Data Line Security, a constant busy tone is heard.

Default Setting

None

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

None

Programming

- ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Turn off or on an extension ability to have calls queued if a call rings the extension when it is busy.

0 = Off
1 = On (default)
- ➔ **20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
Turn off or on an extension ability to receive Off-Hook Signaling.

0 = Off
1 = On (default)
- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension user to manually or automatically receive Off-Hook Signaling.

0 = Manually
1 = Automatically (default)
- ➔ **20-13-35 Class of Service Options (Supplementary Service) – Block Camp On**
Turn off or on an extension COS ability to block caller from dialing # to camp on.

0 = Off (default)
1 = On

Operation

None

Delayed Ringing

Description

Delayed Ringing allows programmed secondary answering positions to ring on incoming calls after a programmed time. This feature applies to CO/PBX lines, Secondary Incoming Extensions, Virtual Extensions, and Call Arrival Keys.

Conditions

- An extension user can answer an outside call by lifting the handset (depending on programming).
- Terminals must have a CAP or CO line appearance for a trunk call to be answered on the telephone.

Default Settings

None

System Availability

Terminals:

All Terminals

Required Component(s)

None

Related Features

Call Arrival (CAR) Keys

Central Office Calls, Answering

Secondary Incoming Extension

Virtual Extensions

Programming

- ➔ **14-07-01 Trunk Access Map Setup**
Set up the Trunk Access Maps. This sets the access options for trunks. Ring Group programming overrides Access Map programming.


Trunk Access Maps: 1~200
- ➔ **15-06-01 Trunk Access Map for Extensions**
Assign Trunk Access Maps to extensions.

Trunk Access Maps: 1~200 (default: 01)
- ➔ **15-07-01 Programmable Function Keys**
Assign CAR/SIE/VE function keys (code *03 + extension number) or CO function keys (Code *01 + trunk port) on multiline terminals.
- ➔ **15-09-01 Virtual Extension Ring Assignment**
Individually program an extension Virtual Extension key(s) to either ring (1) or not ring (0).
- ➔ **15-11-01 Virtual Extension Delayed Ring Assignment**
Individually program an extension Virtual Extension key(s) for Delayed Ringing (1) or Immediate Ringing (0).
- ➔ **20-04-03 System Options for Virtual Extensions – CAR/SIE/Virtual Extensions Delay Interval**
CAR/SIE/VE set for Delayed Ringing (Program 15-11-01) rings the covering extension after this interval. (0~64800 seconds). (default: 10).
- ➔ **22-01-04 System Options for Incoming Calls – DIL No Answer Recall Time**
If an incoming trunk call rings longer than this time, it reroutes to the Ring Group set in Program 22-08-01. (0~64800 seconds). (Default: 0).
- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign the incoming trunk type for each trunk.
- ➔ **22-08-01 DIL/IRG No Answer Destination**
If an incoming trunk call rings longer than the DIL No Answer Time (Program 22-01-04), it routes to the destination you specify in this option. Determine if the destination should be a Ring Group, In-Skin/External Voice Mail, or Central Voice Mail.

Ring Groups: 1~100 (default: 1)
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail

Operation

To answer Delay Ringing calls:

1. Go off-hook.
- OR -
Press **Answer**.
- OR -
Press the flashing key.
 *Either Trunk key or CAR/SIE/VE key.*

To program a CAR/SIE/VE key on a phone:

1. Press **Speaker** .
2. Dial **752**.
3. Press the key you want to program.
4. Dial ***03**.
5. Dial the number of the extension you want to appear on the key.
6. Press **Hold** once for Immediate Ring (skip to step 8 for Delayed Ring).
7. Dial the mode number in which the key will ring.
1 = Day 1
2 = Night 1
3 = Midnight 1
4 = Rest 1
5 = Day 2
6 = Night 2
7 = Midnight 2
8 = Rest 2
8. Press **Hold** for a second time for Delayed Ring, or Skip to step 10.

9. Dial the mode number in which the key delays ring.

1 = Day 1

2 = Night 1

3 = Midnight 1

4 = Rest 1

5 = Day 2

6 = Night 2

7 = Midnight 2

8 = Rest 2

10. Press **Speaker**.

Department Calling

Description

With Department Calling, an extension user can call an idle extension in a programmed Department Group (64 Department Groups available) by dialing the group pilot number. For example, this lets a caller dial the Sales department by knowing the Sales department pilot number. The caller does not have to know any Sales department extension number.

Two types of routing are available with Department Calling: Priority Routing and Circular Routing. With Priority Routing, an incoming call routes to the highest priority extensions first. Lower priority extensions ring only if all higher priority extensions are busy. With Circular Routing, each call rings a new extension.

Overflow Routing

Department Calling also provides overflow routing for extensions in the group. If a user directly dials a busy extension in a Department Group, the system can optionally route the call to the first available group member. The system follows Program 22-15-01 ~ 22-15-07 for playing the periodic VRS message.

Department Calling also allows for each Department group to transfer calls to a predefined Speed dial bin (PRG 24-05-01) immediately or after a Delayed time (PRG 24-02-08). Internal and transferred calls are not supported for Delayed transfer.

DID's and Overflow Routing

The three types of Overflow supported for DID calls are shown below:

Immediate Transfer:

Enable (1)/Disable (0) this feature using a (58) key programmed in PRG 15-07. It can be done also by using the service codes in Program 11-11-25 (set) and Program 11-11-26 (cancel). When this feature is activated, any DID call pointed directly to the Pilot Number goes immediately to the transfer destination and does not ring anyone in the group. To set up the destination use Program 24-05 and Program 13-04. Once these programs are set, the access code assigned in Program 11-11-27 can be used to change the destination as needed.

Delay:

Enable (1)/Disable (0) this feature using a (59) key programmed in PRG 15-07. It can be done also by using service codes assigned in Program 11-11-28 (set) and Program 11-11-29 (cancel). When this feature is activated, any DID call pointed directly to the Pilot follows one of the two patterns:

- If all available members are busy or logged out, the call goes immediately to the transfer destination.
- If agents are logged in and not busy, the call comes in and hunts through the idle members until the timer in Program 24-02-08 expires. Once this time expires, the call is then routed to the transfer destination assigned in Program 24-05 and Program 13-04. Once these programs are assigned, the access code assigned in Program 11-11-27 can be used to change the destination as needed.

 DND:

This feature can be enabled using a (60) key programmed in PRG 15-07 or by using service codes assigned in Program 11-11-30 (set) and Program 11-11-31 (cancel). When this feature is activated any DID pointed directly to the Pilot gets a busy tone, and the call does not route.

User Log Out/Log In

An extension user can log out and log in to a Department Calling Group. By logging out, the user removes their extension from the group. Once logged out, Department Calling bypasses their extension. When they log back in, Department Calling routes to their extension normally. All users can dial a code to log in or log out of their Department Calling Group. A multiline terminal can optionally have a function key programmed to login/logout.

Enhanced Hunting

Department Calling is enhanced with expanded hunting abilities. Hunting sets the conditions under which calls to a Department Group pilot number cycles through the members of the group. The hunting choices are:

 Busy

A call to the pilot number hunts past only a busy group member to the first available extension.

 Not Answered

A call to the pilot number cycles through the idle members of a Department Calling group. The call continues to cycle until it is answered or the calling party hangs up. If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call does not hunt to the next available extension.

 Busy or Not Answered

A call to the pilot number cycles through the idle members of a Department Calling group. The call continues to cycle until it is answered or the calling party hangs up.

If all members of the Department Group are busy, an incoming or transferred call to the group pilot number queues for an available member. Each group has a queue that can hold any number of waiting calls. If a display telephone is waiting in queue, the user sees: *WAITING (group name)*. If a transferred call in queue is an outside call, and the system has DSP daughter board installed with the VRS, the queued caller hears, *"Please hold on. All lines are busy. Your call will be answered when a line becomes free."*

The VRS can also transfer calls to Department Groups. Refer to [Voice Response System \(VRS\) on page 2-1089](#) for information on setting up the VRS.

The system prevents hunting to a Department Group extension if it is:

- Busy on a call
- In Do Not Disturb
- Call Forwarded
- Logged Out

Conditions

- When a DIL rings to a Department Group, the DIL may follow overflow programming (PRG 22-01-04 and PRG 22-08-01).
- If all agents are logged out and an intercom call to the Department Group is made you get a busy signal.
- Extensions in a Department Group which have Call Forwarding enabled are not included in the call hunt. The extension to which the user is forwarded does not receive the hunted calls. When you use the automatic Department Step calling (PRG 16-01-03) it hunts only to members with the same or lower priority.
- Easily step call to an idle Department Group member if the member called is busy.
- A virtual extension can be programmed to receive multiple calls which can camp-on to the extension – no analog port is required.
- An extension user can Transfer a call to a Department Group Pilot number. If unanswered, the call recalls (depending on programming) the transferring extension after the Transfer Recall Time (Program 24-02-04).
- Voice mail uses one Department Group for voice mail.
- When PRG 16-01-05 is set to (1) Automatic, all telephones in the Department group Ring for ICM calls & DID calls Directed to the Department Group Pilot Number only.
- The Overflow feature is supported only for DID calls pointed directly to the Pilot Number. POTS lines and transferred DIDs ignore the Overflow settings.

Default Setting

Disabled

Priority Routing

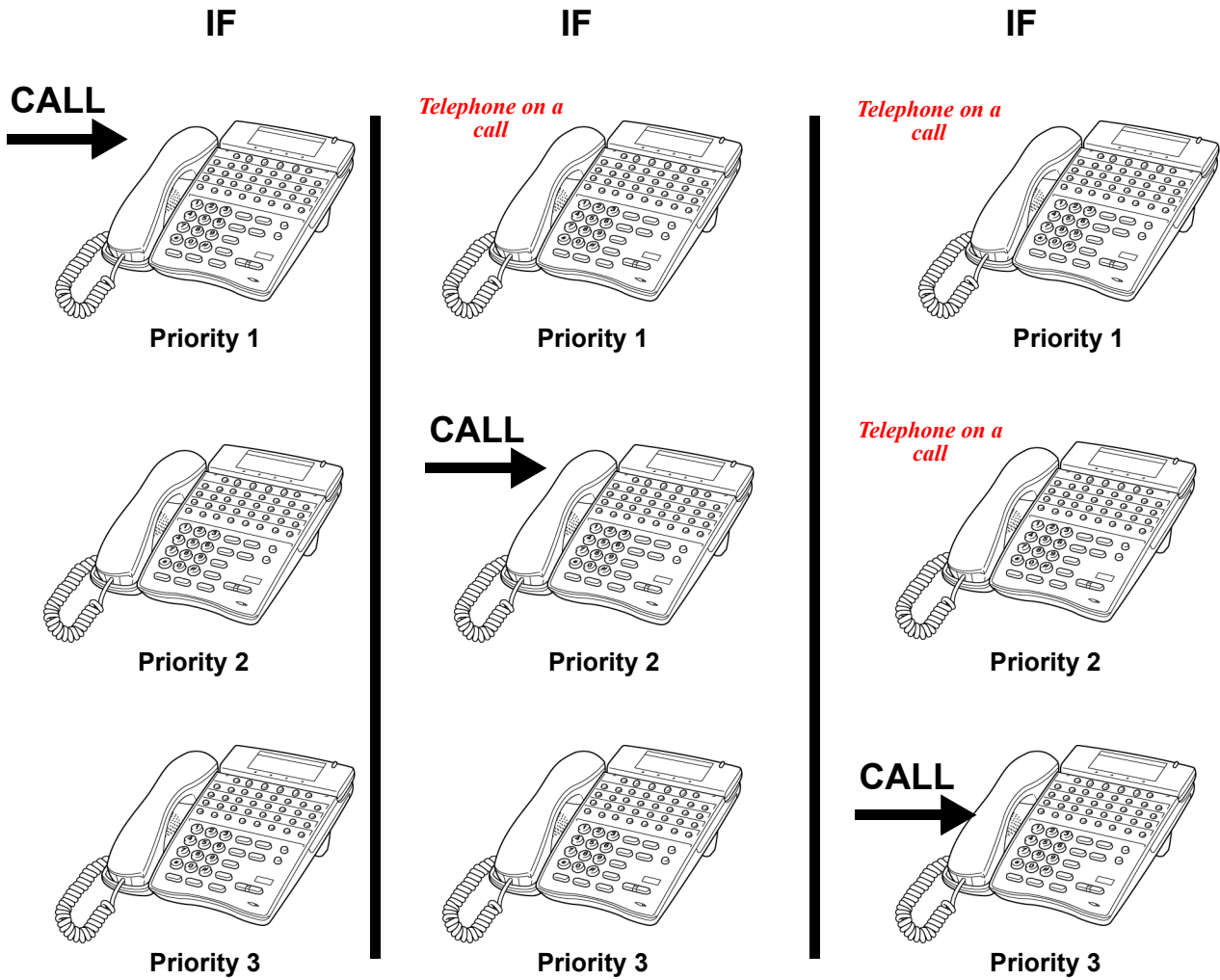


Figure 2-1 Department Calling Priority Call Routing

Circular Routing

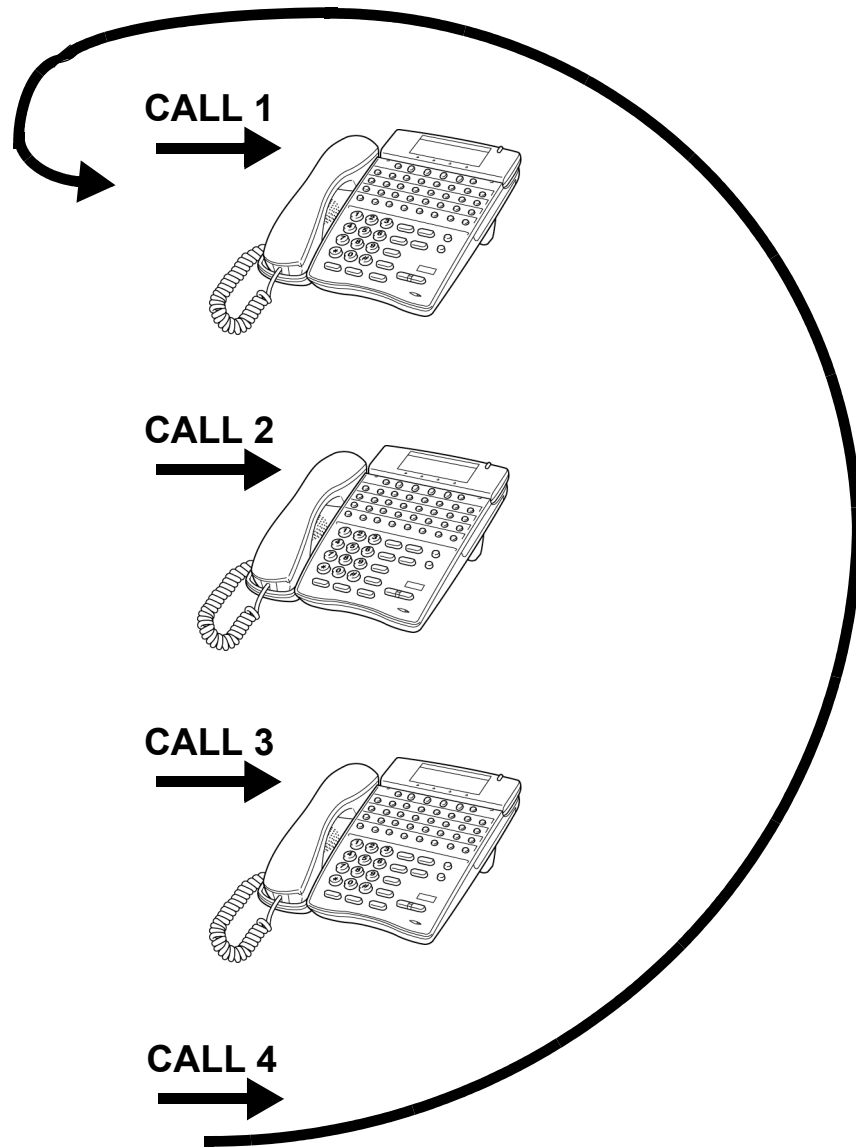


Figure 2-2 Department Calling Circular Routing

System Availability

Terminals

All Terminals

Required Component(s)

VRS for Messaging

Related Features

Call Arrival (CAR) Keys

Call Forwarding

Department Step Calling

Transfer

Voice Mail Message Indication on Line Keys

Voice Response System (VRS)

Programming

- ➔ **11-07-01 Department Group Pilot Numbers – Dial**
Assign pilot numbers to the Extension (Department) Groups you set up in Program 16-02-01~Program 16-02-10.
- ➔ **11-11-25 Service Code Setup (for Setup/Entry Operation) – Automatic Transfer Setup for Each Extension Group**
Set the service code (default = 602) to activate immediate automatic transfer for ICM and transferred calls to Department Groups.
- ➔ **11-11-26 Service Code Setup (for Setup/Entry Operation) – Automatic Transfer Cancellation for Each Extension Group**
Set the service code (default = 603) to deactivate immediate automatic transfer for ICM and transferred calls to Department Groups.

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-
- **11-11-27 Service Code Setup (for Setup/Entry Operation) – Destination of Automatic Transfer for Each Department Group**
Set the service code (default: 604) for setting the destination for immediate automatic transfer for ICM and transferred calls to Department Groups.
 - **11-11-28 Service Code Setup (for Setup/Entry Operation) – Delayed Transfer for Every Extension Group**
Use this service code (default: 605) to set the delayed transfer destination Department Group.
 - **11-11-29 Service Code Setup (for Setup/Entry Operation) – Delayed Transfer Cancellation for Each Extension Group**
Use this service code (default: 606) to cancel the delayed transfer destination Department Group.
 - **11-12-09 Service Code Setup (for Service Access) – Change to STG (Department Group) All Ring**
Set the service code for ringing all members of a Department Group.
 - **11-16-10 Single Digit Service Code Setup – (Department) STG All Ring Mode**
Assign the Single Digit (post-dialing) Service Code for All Member Ring.
 - **15-07-01 Programmable Function Keys**
Assign a Department Calling key (46) so extension users can install or remove themselves from the Department Calling Group. Additional keys can also be assigned for Department Group features immediate calling destination (58), delayed calling destination (59) and DND destination (60).
 - **16-01-01 Department Group Basic Data Setup – Department Name**
Assign a name to the Extension (Department) Groups.
 - **16-01-02 Department Group Basic Data Setup – Department Calling Cycle**
Set the routing cycle for calls to a department (i.e., when a user dials the department pilot number). The system can ring the highest priority extension available, Priority Routing (0) or cycle in circular order to a new idle extension for each new call, Circular Routing (1).
 - **16-01-03 Department Group Basic Data Setup – Department Routing When Busy (Auto Step Call)**
Set how the system routes an Intercom call to a busy Department Group member. The caller can hear busy tone (0) or overflow to the first available Department Group member (1). This option is for Intercom calls to an extension, not a pilot number.
 - **16-01-04 Department Group Basic Data Setup – Hunting Mode**
Set if an unanswered call should hunt once stopping at the last member tried (0) or continually hunt through the idle members (1).
 - **16-01-05 Department Group Basic Data Setup – Extension Group All Ring Mode Operation**
Set if all members of the group should ring automatically (1) or through using the service code (0) defined in Program 11-12-09. Selecting automatic overrides the settings of Programs 16-01-03 and 16-01-04.
 - **16-01-06 Department Group Basic Data Setup – STG Withdraw Mode**
0 = Disable (Camp On) (default)
1 = Enable (Overflow Mode)
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- ➔ **16-01-07 Department Group Basic Data Setup – Call Recall Restriction for STG**
Set if a transferred call to the pilot number should recall (0) or camp on (1) to the busy or idle Department Group.

 - ➔ **16-01-09 Department Group Basic Data Setup – Department Hunting No Answer Time**
Set the time a call rings a Department Group extension before hunting occurs.
(0~64800 seconds). (default: 15).

 - ➔ **16-01-10 Department Group Basic Data Setup – Enhanced Hunt Type**
Set the type of hunting for each Department Group.

0 = No Queuing
1 = Hunting When Busy
2 = Hunting When Not Answered
3 = Hunting When Busy or No Answer

 - ➔ **16-02-01 Department Group Assignment for Extensions**
Assign extensions to Department Groups and set the priority assignment.

Department Groups: 1~64

 - ➔ **16-03-01 Secondary Department Group**
Assign extensions to multiple Department Groups and set the priority assignment. Each Secondary Department Group can have up to 16 extensions assigned.

Department Groups: 1~64

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

 - ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Turn off (0) or on (1) an extension user ability to have calls queued if a call rings the extension when it is busy.

 - ➔ **20-11-17 Class of Service Options (Hold/Transfer Service) – Department Group Trunk-to-Trunk Transfer (Each Telephone Group Transfer)**
Allow (1)/Prevent (0) an extension from using the immediate automatic transfer of ICM and transferred calls to the Department Groups feature.

 - ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension user to manually (0) or automatically (1) receive off-hook signals.

 - ➔ **20-13-14 Class of Service Options (Supplementary Service) – Department Calling (PLT No Called Extension)**
Allow (1)/Prevent (0) extensions from receiving Department Calls when the pilot number is dialed.

 - ➔ **22-02-01 Incoming Call Trunk Setup**
If you want a trunk to be a DIL to a Department Group, assign Service Type 4 for each Night Service Mode. Refer to PGM 22-07-01.
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- ➔ **22-07-01 DIL Assignment**
For each trunk assigned Service Type 4 in PRG 22-02-01 above, assign the DIL destination as the Department Group pilot number (as assigned in Program 11-07-01).

Department: Groups 1~64
 - ➔ **24-02-05 System Options for Transfer – Message Wait Ring Interval Time**
For SLTs without Message waiting lamps, this is the time between intermittent ringing. If the time is set to 0, the system rings once (0~64800 seconds). (default: 30).
 - ➔ **24-02-08 System Options for Transfer – Delayed Transfer Timer for All Department Groups**
Determine the time a call should ring a Department Group before transferring the call (0~64800 seconds). (default: 10).
 - ➔ **24-05-01 Department Group Transfer Target Setup**
Assign the Speed Dial bin to each Department Group to hold the destination for the immediate automatic transfer of ICM and transferred calls to the Department Group feature.

Operation

To call a department group:

1. Go off-hook.
2. Dial department's extension number.
 - ✎ *The system routes the call to the first free telephone in the department group.*
3. **Optional:** To manually ring all members of the group, dial the single digit service code assigned for All Member Ring (Program 11-16-10).

To log out of your Department Calling Group:

✎ *While you are logged out, Department Calling cannot route calls to your extension.*

1. Press **Speaker**.
2. Dial **650 + 1**.

- OR -

Press **Department Calling Log In** key (PRG 15-07-01 or SC 751: 46).

✎ *The key lights while you are logged out.*

To log back in to your Department Calling Group:


✎ *When you log back in, Department Calling will route calls to your extension.*

1. Press **Speaker**.

2. Dial **650 + 0**.

- OR -

Press Department Calling Log In key (PRG 15-07-01 or SC 751: 46).

 *The key goes out when you log back in.*

To change the Department Group Overflow Destination:

1. Press **Speaker**.
2. Dial **604 + Department Group** (01 ~ 64).
3. Dial **01 ~ 08** (Refer to PRG 24-05).
4. Dial the destination the calls **route to**.
5. Press **Hold**.

Department Step Calling

Description

After calling a busy Department Calling Group member, an extension user can have Department Step Calling quickly call another member in the group. The caller does not have to hang up and place another Intercom call if the first extension called is unavailable. Department Step Calling also allows an extension user to cycle through the members of a Department Group.

Conditions

- If required, use this option to change the Department Step Calling Single Digit Service Code (default code = 2).
- A function key for Department Step Calling can be assigned (code 36).
- In PRG 20-08-12, Enable (1)/Disable (0) an extension user ability to use Department Step Calling.

Default Setting

Enabled

System Availability

Terminals

All Stations

Required Component(s)

None

Programming

- ➔ **11-12-07 Service Code Setup (for Service Access) – Step Call**
If required, customize the Step Call service code used by an extension user (default: 708).
- ➔ **11-16-01 Single Digit Service Code Setup – Step Call**
If required, change the Department Step Calling Single Digit Service Code (default: 2).
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key Department Step Calling (code 36).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-08-12 Class of Service Options (Outgoing Call Service) – Department Group Step Calling**
Turn Off (0) or On (1) the ability to use Department Step Calling.


Related Features

Department Calling

Programmable Function Keys

Operation

To make a Step Call:

1. Place a call to a busy Department Group member.
- OR -
Place a call to a Department Group pilot number.
2. Dial Department Step Code (2) to call the next available Department Group member.
3. Repeat step 2 to call other Department Group members.
 *Step through Department Groups set in Program 16-02-01.*

Dialing Number Preview

Description

Dialing Number Preview lets a display multiline terminal user dial and review a number before the system dials it. Dialing Number Preview helps the user avoid dialing errors.

Conditions

- An extension user cannot edit the displayed number.
- To place an outgoing call, an extension user must have outgoing access to a line, CAP or trunk group key.
- If the system has VRS installed, you must first press * to preview a number.

Default Setting

Enabled

System Availability

Terminals

All Display Multiline Terminals

Required Component(s)

None

Related Features

Central Office Calls, Placing

Voice Response System (VRS)

Programming

- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-08-05 Class of Service Options (Outgoing Call Service) – Dial Number Preview (Preset Dial)**
Turn Off (0) or On (1) the ability to use Dial Number Preview.

Operation

To use Dial Number Preview to place a call (multiline terminal only):

1. Do not lift the handset or press **Speaker**.
2. To preview *any number*, dial the number you wish to call.
 - ✎ *With VRS installed, you must first press * to preview the number.*
 - To preview a Speed Dial – System/Group number, press **Redial** and dial the Speed Dial – System/Group bin number you want to call.
 - ✎ *The number displays.*
3. To dial out the displayed trunk number, press a Line/CAP/Trunk Group key.
 - ✎ *If the previewed number as a trunk access code (e.g., 9), you can press **Speaker** instead.*
 - OR -
 - To dial an Intercom number, press **Speaker**.
 - OR -
 - To cancel the number without dialing it out, press **Hold**.

Dial Pad Confirmation Tone

Description

For an extension with Dial Pad Confirmation Tone enabled, the user hears a beep each time they press a key. This is helpful for Intercom calls and Dial Pulse trunk calls, since these calls provide no Call Progress tones.

Conditions

- Dial Pad Confirmation Tone does not apply to single line telephones or Wireless – DECT Terminals.
- Dial Pad Confirmation Tone is not canceled when dialing in handset mode.
- Dial Pad Confirmation Tone is canceled when dialing in handsfree mode, but only for internal calls. The tone is still heard for external dialing.

Default Setting

Disabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

None

Programming

- ➔ 11-11-19 **Service Code Setup (for Setup/Entry Operation) – Key Touch Tone On/Off**
If required, change the service code to enable/disable the Key Touch Tone (default: 724).

Operation

To enable/disable Dial Pad Confirmation Tone:

1. Pick up the handset or press **Speaker**.
2. Dial **724**.

Dial Tone Detection

Description

If a trunk has Dial Tone Detection enabled, the system monitors for dial tone from the Telco or PBX when a user places a call on that trunk. If the user accesses the trunk directly (by pressing a line key or dialing # 9 and the trunk number), the system drops the trunk if dial tone does not occur. If the user accesses the trunk via a Trunk Group (by dialing a trunk group code or automatically using a feature like Last Number Redial), the system can drop the trunk or optionally skip to the next trunk in the group. Refer to the chart under Programming for more information.

Conditions

None

Default Setting

Disabled for manually dialed calls; enabled for automatically dialed calls.

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Automatic Route Selection

Call Appearance (CAP) Keys (Outbound)

Central Office Calls, Placing

Last Number Redial**Save Number Dialed****Speed Dial – System/Group/Station****T1 Trunking (with ANI/DNIS Compatibility)****Trunk Group Routing****Trunk Groups**

Programming**Table 2-14 Dial Tone Detection Program Interaction**

Method	14-02-05	14-02-11	Result if dial tone not present . . .
Press a line key - or - Dial #9+ Trunk number	0	0	Trunk hangs (does not disconnect)
	0	1	Trunk hangs (does not disconnect)
	1	0	Trunk drops
	1	1	Trunk drops
Dial a Trunk Group code - or - Automatically through a feature	0	0	Trunk hangs (does not disconnect)
	0	1	Trunk reroutes after time-out
	1	0	Trunk drops
	1	1	Trunk reroutes after time-out

- **10-09-01 DTMF and Dial Tone Circuit Setup**
If dial tone detection is enabled, allocate at least one circuit for dial tone detection [ICM/Trunk (0) or Trunk (2)].
- **14-02-05 Analog Trunk Data Setup – Dial Tone Detection for Manual Accessed Trunks**
Enable(1)/Disable (0) dial tone detection for manually dialed trunk calls. This option only pertains to calls placed using line keys or Direct Trunk Access.
- **14-02-11 Analog Trunk Data Setup – Next Trunk in Rotary if No Dial Tone**
If enabled, the system skips over a trunk if dial tone is not detected. This option pertains to calls placed using Call Appearance (CAP) Keys, Speed Dial, ARS, Last Number Redial or Save Number Dialed. It does not pertain to line key or Direct Trunk Access calls.

- **21-01-05 System Options for Outgoing Calls – Dial Tone Detection Time**
If dial tone detection is enabled, the system waits this time for the Telco to return dial tone. When this time expires, the system assumes dial tone is not present. To disable this time (and have the system wait continuously), enter 0.

- **21-01-06 System Options for Outgoing Calls – Dial Pause at First Digit**
If Dial Tone Detection is disabled, the system waits this time before sending dialed digits. If using Dial Tone Detection, this time should be set longer than the time set in PRG 21-01-05, otherwise, if this time is set shorter than PRG 21-01-05, Dial Tone Detection is satisfied and PRG 21-01-05 is disregarded (not used).

Operation

Dial Tone Detection is automatic if enabled in programming.

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Digital Trunk Clocking

Enhancements

This feature was added to the documentation with Version 2000 . This feature is available with version 1000 or higher .

Description

The IPK II CPUII has a built-in clock source for all digital trunk cards. Digital trunk cards are connected via an internal PLO (Phase Locked Oscillator) to derive Primary Clock from the network in priority order. If priority is set up incorrectly, or if two primary clocks are coming in, slips may occur causing improper data synchronization. The Phase Locked Oscillator (PLO) equipped with the IPK II CPUII ETU is the timing source for all digital trunk ETUs in the system. The PLO synchronizes the system and clock signals from another office. When the IPK II is a clock receiver office, the PLO generates the clock signal according to the source clock signals received from the source office in the network. The source clock signals are extracted from digital trunk ETUs and supplied to the PLO.

The PLO synchronization source priorities are as follows:

1. PRI
2. DTI (External)
3. BRI
4. CPU

Conditions

- If multiple PRIs exist, the system chooses the first one that synchronizes with the carrier.
- If there are multiple PRIs, and the one being used for the source goes down, the system begins to count forward in slot numbers looking for the next available PRI.
- If multiple BRIs exist and no PRI or DTI (External) exists, the IPK II CPU chooses the first BRI that synchronized with the carrier.
- If there is one PRI, and the one being used for the source goes down, the IPK II CPU looks to see if any BRIs are installed in the system. If there are no BRIs, the IPK II CPU becomes the new synchronization source. The reason for this is when a PRI is installed in the system, all T1s must be assigned as (INTERNAL). T1 (INTERNAL) is not a clocking priority.

Default Setting

None

System Availability

Terminals

N/A

Required Component(s)

CPUII()-U10 ETU

BRT(4)-U() ETU

- OR -

DTI-U40 ETU with **firmware V5.00 or higher**

Related Features

ISDN Compatibility

K-CCIS – T1

T1 Trunking (with ANI/DNIS Compatibility)

Digital Trunk Clocking Examples:

If multiple PRIs exist, the first one that synchronized with the carrier is chosen. In this example, the PRI in S1 was the first to synchronize with the carrier; therefore, it is the PLO synchronization source.

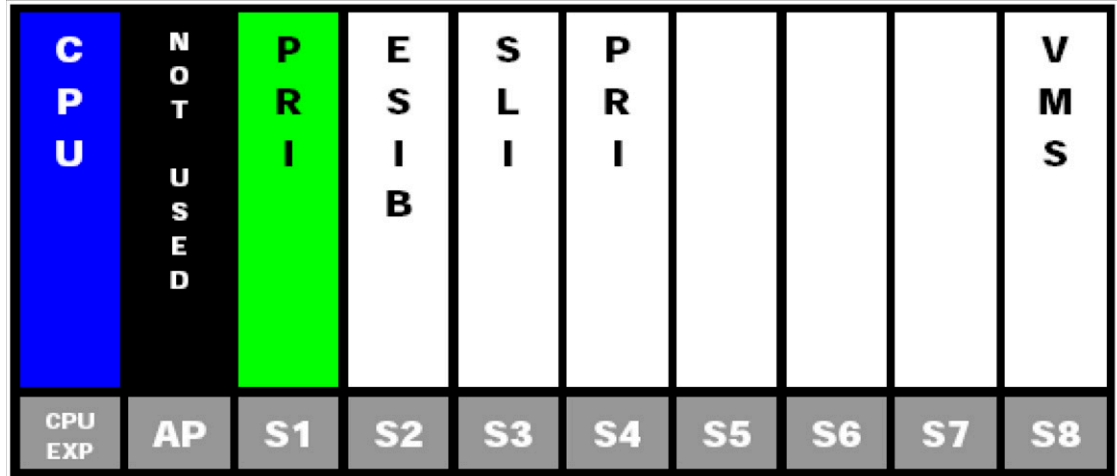


Figure 2-3 Digital Trunk Clocking Example 1

If multiple PRIs exist and the one being used for the source goes down, the system begins to count forward in slot numbers looking for the next available PRI. In this example, the PRI in S1 went down, so the system now begins looking forward in slot numbers for the next PRI to use as the clock source.

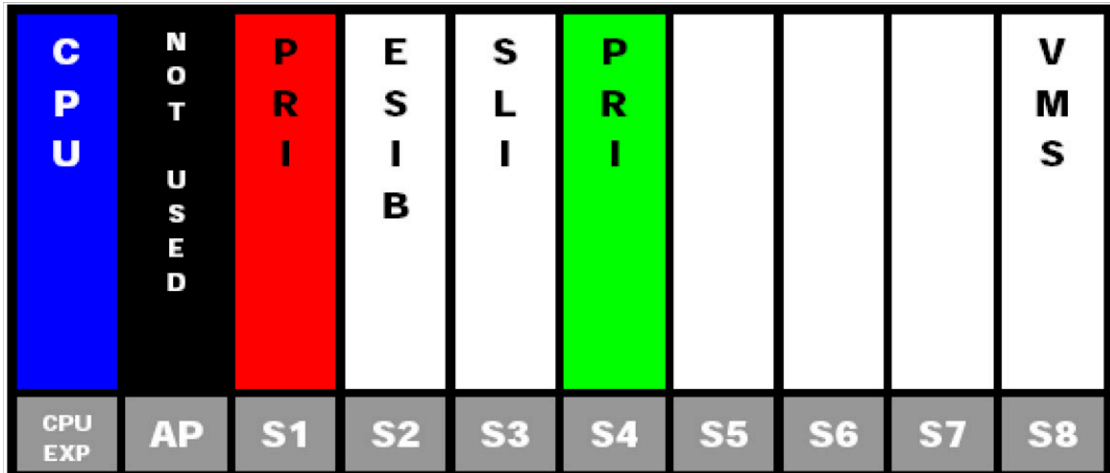


Figure 2-4 Digital Trunk Clocking Example 2

In this example, the PRI in S4 was the first to synchronize with the carrier and became the PLO synchronization source. The PRI in S4 then went down and the system began looking forward in slot numbers to find the next PLO source. In this case, the PRI in S1 was the next source because after it looks through the rest of the slots in the system, it starts over with S1.

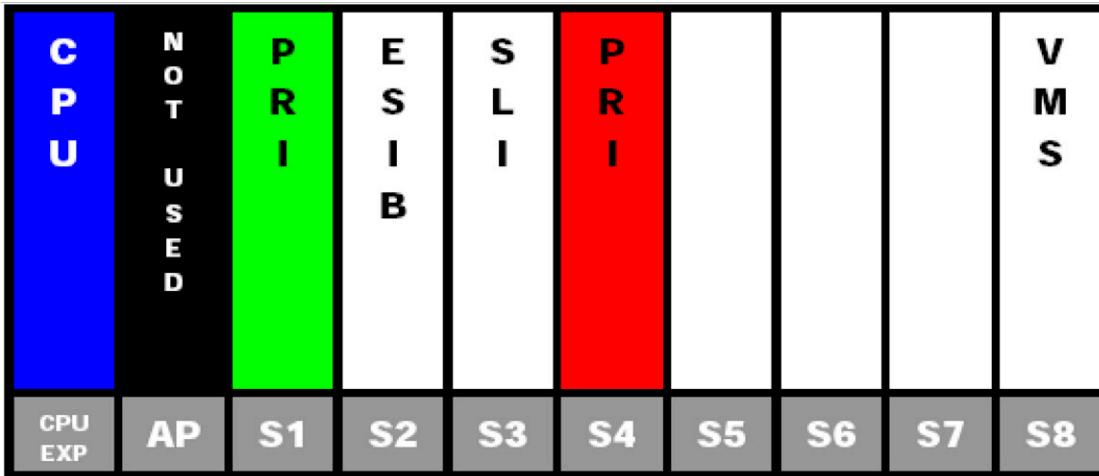


Figure 2-5 Digital Trunk Clocking Example 3

In this example, there are multiple T1 circuits in the system. There can only be one T1 circuit assigned as EXTERNAL in the system, so the T1 assigned as EXTERNAL is the PLO synchronization source.

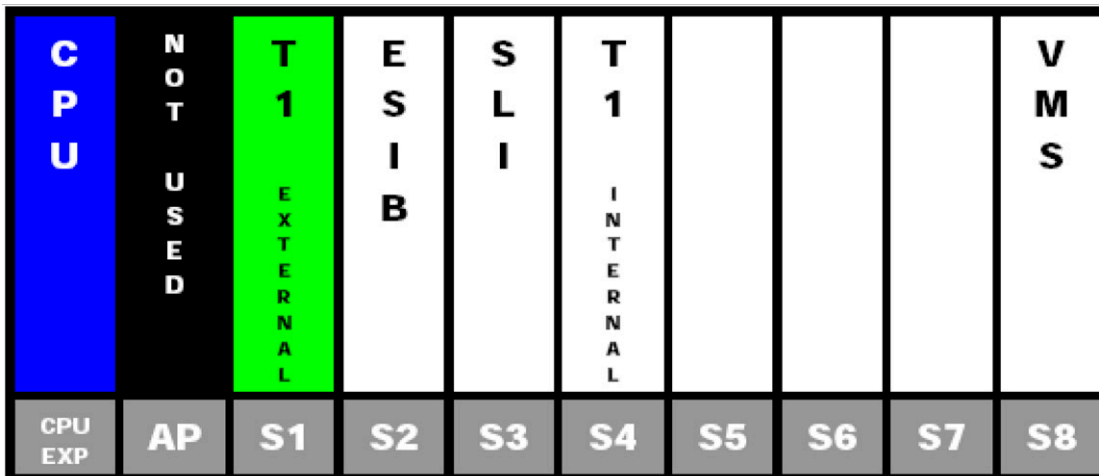


Figure 2-6 Digital Trunk Clocking Example 4

In this example, there are multiple T1 circuits and a BRI circuit. Since the T1 assigned as EXTERNAL has higher priority than a BRI, the T1 EXTERNAL is the PLO synchronization source.

C P U	N O T U S E D	T 1 E X T E R N A L	E S I B	S L I	T 1 I N T E R N A L		B R I		V M S
CPU EXP	AP	S1	S2	S3	S4	S5	S6	S7	S8

Figure 2-7 Digital Trunk Clocking Example 5

In this example, there is a PRI, multiple T1s, and a BRI. The PRI was the PLO synchronization source until it went down. The BRI then becomes the PLO synchronization source because when a PRI is in the system, T1s cannot be assigned as EXTERNAL, which are not in the PLO Synchronization Source priority list.

C P U	N O T U S E D	P R I	E S I B	S L I	T 1 I N T E R N A L	T 1 I N T E R N A L	B R I		V M S
CPU EXP	AP	S1	S2	S3	S4	S5	S6	S7	S8

Figure 2-8 Digital Trunk Clocking Example 6

If multiple BRIs exist and no PRI or T-1 EXTERNAL exists, the system chooses the first BRI that synchronized with the carrier. In this example, the BRI in S4 synchronized with the carrier first and became the PLO synchronization source.

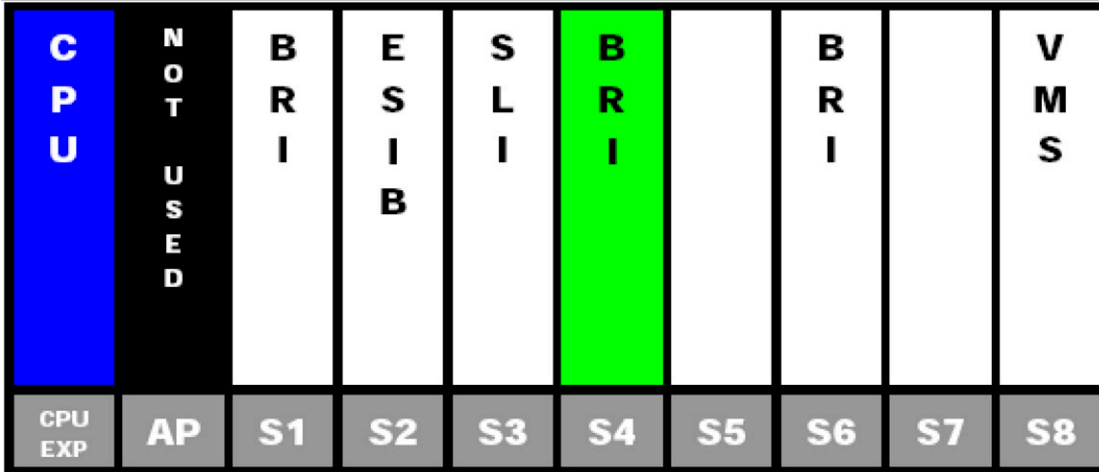


Figure 2-9 Digital Trunk Clocking Example 7

In this scenario, the PRI was the clocking source until it went down. There are no other PRIs, T1 (Externals), or BRIs in the system. The IPK II CPU now becomes the PLO synchronization source.

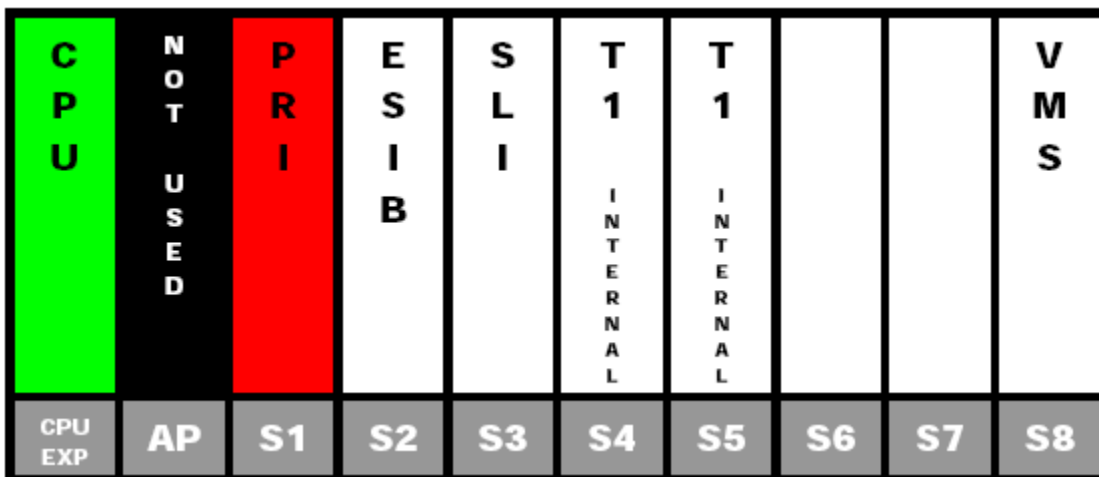


Figure 2-10 Digital Trunk Clocking Example 8

Programming

Refer to the related features for programming.

Operation

Refer to the related features for details.

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Digital Voice Mail

Enhancements

Version 1600 or higher:

- Supports configurable Voice Mail Message Waiting lamp color and Voice Mail Message Waiting lamp flash pattern.
- Trunk mapping for trunk numbers 001~200. This is supported in a future Linux 2.0 voice mail software build. Prior to this release, Trunk mapping was supported only for trunks 01~99.

Description

The system is fully compatible with NEC digital voice mail systems. These systems provide telephone users with comprehensive Voice Mail and Automated Attendant features. Voice Mail ends the frustration and cost of missed calls, inaccurate written messages and telephone tag. This frees busy receptionists and secretaries for more productive work.

Automated Attendant automatically answers the system incoming calls. After listening to a customized message, an outside caller can dial a system extension or use Voice Mail.

Integrated Voice Mail enhances the telephone system with the following features:

Call Forwarding to Voice Mail

An extension user can forward their calls to Voice Mail. Once forwarded, calls to the extension connect to that extension mailbox. The caller can leave a message in the mailbox instead of calling back later. Forwarding can occur for all calls immediately, for unanswered calls or only when the extension is busy. When a user transfers a call to an extension forwarded to Voice Mail, the call waits for the Delayed Call Forwarding time before routing to the called extension mailbox. This gives the transferring party the option of retrieving the call instead of having it go directly to the mailbox.

Leaving a Message

Voice Mail lets a multiline terminal extension user easily leave a message at an extension that is unanswered, busy or in Do Not Disturb. The caller just presses their Voice Mail key to leave a message in the called extension mailbox. There is no need to call back later.

Transferring to Voice Mail

By using Transfer to Voice Mail, a multiline terminal extension user can Transfer a call to their own or a co-worker's mailbox. After the Transfer goes through, the caller can leave a message in the mailbox.

❑ Live Record

While on a call, an extension user can have Voice Mail record the conversation. The multiline terminal user just presses the Voice Mail Record key; the single line telephone user dials a code. Once recorded, the Voice Messaging System stores the conversation as a new message in the user's mailbox. After calling their mailbox, a user can save, edit or delete the recorded conversation.

❑ Live Monitor

A multiline terminal user can have their idle extension emulate a personal answering machine. This lets Voice Mail screen their calls, just like their answering machine at home. If activated, the extension incoming calls route to the user's subscriber mailbox. Once the mailbox answers, the user hears the caller's incoming message. The multiline terminal user can then:

- Let the call go through to their mailbox.
- Intercept the call before it goes to their mailbox.

❑ Voice Mail Overflow

If Voice Mail automatically answers trunks, Voice Mail Overflow can reroute those trunks to other extensions when all Voice Mail ports do not answer or, with certain software, are busy. During periods of high traffic, this prevents the outside calls from ringing Voice Mail for an inordinate time. There are two types of Voice Mail Overflow: Immediate and Delayed. With immediate overflow, calls immediately reroute to other extensions when all Voice Mail ports do not answer or, with certain software, are busy. With delayed overflow, calls reroute after a preset interval. Without overflow, the outside calls ring Voice Mail until a port becomes available or the outside caller hangs up.

❑ Voice Mail Caller ID

The Voice Mail can use ANI/DNIS information to identify the outside caller that left a message in a user's mailbox. When the message recipient dials 0 or presses the CID softkey while listening to a message, they hear the outside telephone number of the message sender.

The message recipient can also return the call from their mailbox if allowed by system programming by pressing the CALL softkey or #,0. Refer to [Caller ID Call Return on page 2-183](#).

Voice Mail Queuing

When accessing the voice mail, the system provides a voice mail queue. If all the voice mail ports are busy, any call trying to get to the voice mail is placed in queue. As the voice mail ports become available, the calls are connected to the voice mail in the order in which they were received.

As the Voice Mail Queue follows Department Hunting programming, the queue can hold a maximum of 10 calls. If the queue is full or if the voice mail ports are not assigned to a Department Group, the calls are handled as though there were no voice mail queuing feature enabled. The calls either access voice mail if a port is available or they receive a busy signal.

The Voice Mail Queuing feature does not work with the Conversation Record feature.

Message Key Operates as Voice Mail Key

The system enhances a telephone Message key function when connected to a system which has voice mail installed. When an extension receives a voice mail, the Message key can be used to check the number of messages in voice mail, or call the voice mail to listen to the messages. If no Voice Mail Programmable Function Key is defined (Program 15-07-01, code 77), the telephone Message Waiting LED flashes to indicate new messages.

This option is not available with a networked voice mail – the voice mail must be local.

Directory Dialing

Directory Dialing allows an Automated Attendant caller to reach an extension by dialing the first few letters in the extension user's name. With Directory Dialing, the caller does not have to remember the extension number of the person they wish to reach – just their name. Here is how Directory Dialing works:


1. When the Automated Attendant answers, it sends the call to a Directory Dialing Mailbox. (Optionally, the caller may be asked to dial a digit to access Directory Dialing.)
2. The Directory Dialing Mailbox plays the Directory Dialing Message which asks the caller to dial letters for the name of the person they want to reach.
3. The caller dials the first three letters for the person's name. They can dial by first name or last name, depending on how the Directory Dialing Message was recorded and the Directory Dialing Mailbox was set up.
4. Voice Mail searches the list of programmed extension names for a match of the caller-entered letters.
5. The caller dials the digit for the extension they wish to reach, and Voice Mail sends the call to that extension. The call is sent as a Screened or Unscreened transfer, depending on programming.

For callers to use Directory Dialing, the system must have a name programmed for each extension. Each extension should also have a name recorded in their Subscriber Mailbox. In addition, each extension used by Directory Dialing must be installed.

Conditions

- In-Mail and VMS/FMS/VMP/CTI/CTP cannot be used at the same time in the same system.
- Only one of the following voice mails can be installed in the system at one time: VMS/FMS/VMP/CTI/VP/CTP.
- Ring Group calls do not follow extension call forwarding to voice mail.
- Analog ports on the CPUII do not support Message Waiting lighting.
- With KSU software **Version 1500 or lower**, trunk mapping is supported for trunk numbers 1~99 only. Trunk numbers higher than 99 cannot be trunk mapped.

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- Conversation Record does not work for monitored calls.
 - Caller ID information is passed from voice mail to an extension for pre-answer display on an unscreened transfer from voice mail.
 - Off-premise notification and external extensions require access to outside lines.
 - To have the Voice Mail Automated Attendant answer a trunk, program the trunk as a DIL to a Voice Mail port.
 - When the voice mail places a call on hold, it uses Group Hold. Any line appearances for the trunk shows the hold flash rate, however, users cannot pick up these calls (a busy signal is heard).
 - Centrex Transfer from voice mail ports requires KSU software **Version 1.10 or higher**.
 - The following voice mail features require system tones be changed in PRG 80-01-02 to work. Refer to the Programming section for details.
 - Call Holding
 - Call Screening
 - Busy Greeting
 - Await Answer Transfer
 - If the following programs are changed while phone is online, a reset of the phone or reset of the feature is required before the setting takes effect:
 - PRG 15-02-35 Message Waiting Lamp Cycle for Calling Extension
 - PRG 15-02-36 Message Waiting Lamp Cycle for Called Extension
 - PRG 15-02-37 Voice Mail Message Wait Lamp Color
 - PRG 15-02-38 Voice Mail Message Wait Lamp Cycle

 If this LED is used also for Message Waiting Indication, and there are both voice mail messages and Message Wait indications, the color set for Message Wait overrides the color used for voice mail indications (red).
 - With KSU software **Version 2000 or higher**, trunk mapping is supported for trunk numbers 001~200. This also requires Linux 2.0 voice mail software **Version 2.0.3.x or higher**.
 - During a Conversation Record session, DTMF digits are not transmitted. If the End softkey is used to stop the Conversation Record, DTMF to the outside party is restored. If you press the Conversation Record button to end the recording DTMF is not restored.
 - Stutter Dial Tone is supported to Single Line Telephones (SLTs) for Voice Mail Message Waiting.

Default Setting

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

- VMS(2)/(4)/(8)-U() ETU
- FMS(2)/(4)/(8)-U() ETU
- VMP(2)/(4)/(8)-U() ETU
- VMP LX(2)/(4)/(8)-U() ETU
- CTP LX(4)/(8)/(12)/(16)-U() ETU

Related Features

Barge-In

Call Forwarding

Caller ID

Caller ID Call Return

Central Office Calls, Placing

Direct Inward Line (DIL)

Hold

Message Waiting

One-Touch Calling

Programmable Function Keys

Quick Transfer to Voice Mail

Transfer

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**

Assign at least one circuit for DTMF reception (type 0 or 1).
Use the following as a guide when allocating DTMF receivers:

 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.

- ➔ **11-07-01 Department Group Pilot Numbers – Dial**

Assign a Department Group pilot number for the Voice Mail (eight digits maximum). The extensions are assigned to the group in Program 16-02-01.

- ➔ **15-02-26 Multiline Telephone Basic Data Setup – MSG Key Operation Mode**

Determine whether an extension MSG key should function as a Message key (0: default) or Voice Mail key (1). If set as a Message key, the user can press the key to call the voice mail only when they have new messages. If set as a Voice Mail key, it functions as a normal Voice Mail key (it is not active if Centralized Voice Mail is used).

- ➔ **15-02-35 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Calling Extension (Version 1600 or higher required)**
 - 1 = Cycle 1
 - 2 = Cycle 2
 - 3 = Cycle 3 (default)
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7

- ➔ **15-02-36 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Called Extension (Version 1600 or higher required)**
 - 1 = Cycle 1
 - 2 = Cycle 2 (default)
 - 3 = Cycle 3
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7

- ➔ **15-02-37 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Color (Version 1600 or higher required)**
 - 0 = Green
 - 1 = Red (default)

-
-
- ➔ **15-02-38 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Cycle (Version 1600 or higher required)**
 - 1 = Cycle 1
 - 2 = Cycle 2 (default)
 - 3 = Cycle 3
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7

 - ➔ **15-03-01 Single Line Telephone Basic Data Setup – SLT Signaling Type**

For each Electra Elite IPK II voice mail extension, this option *must* be set to 0.

 - ➔ **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**

Set all SLIU ports used for Voice Mail as type 1.

 - ➔ **15-03-09 Single Line Telephone Basic Data Setup – Caller ID Function - for External Module**

This option *must* be set to 0 when voice mail is used or the integration code for the disconnect function is incorrect.

 - ➔ **15-07-01 Programmable Function Keys**

Assign a Voice Mail key to an extension. You must enter the Voice Mail key code (code 77) followed by:

 - Your own extension number if you are setting up your own Voice Mail key.
 - A virtual extension number if you are setting up a Message Center key for a virtual extension.
 - A co-worker's extension number if you are setting up a Message Center key for an installed extension.
 - An uninstalled extension number if you are setting up a Message Center key for an uninstalled extension.

(Optional) Assign a Voice Mail Record key to an extension (code 78).
(Optional) Assign a Personal Answering Machine Emulation key (code 91).
(Optional) Use a Call Redirect key (49) to allow a user to transfer a call to another extension or voice mail without answering the call.

 - ➔ **16-02-01 Department Group Assignment for Extensions**

Put all the Voice Mail extensions in a unique Department Group (do not use Department Group 1). No other extensions should be in this group. This allows DILs to Voice Mail to ring other Voice Mail extensions when the DIL assigned port is busy. Only one Voice Mail group is allowed per system. Be sure to set up the Department Group in order. (For example, the first extension should be order 1, the second order 2, etc.)

 - ➔ **20-02-09 System Options for Multiline Telephones – Disconnect Supervision**

Enable (1) disconnect supervision for the system.

 - ➔ **20-06-01 Class of Service for Extensions**

Assign a Class of Service (1~15) to the voice mail extensions. Using COS 14 is recommended for all time modes.

 - ➔ **20-09-02 Class of Service Options (Incoming Call Service) – Caller ID Display**

Set these options to 1 for the voice mail COS.
-
-

-
-
- ➔ **20-11-01 through 20-11-05**
 - 20-11-12 Class of Service Options (Hold/Transfer Service)**
Set these options to 0 for the voice mail COS.

 - ➔ **20-13-01 Class of Service Options (Supplementary Service) – Long Conversation Alarm**
 - 20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
 - 20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
 - 20-13-07 Class of Service Options (Supplementary Service) – Message Waiting**
 - 20-13-15 Class of Service Options (Supplementary Service) – Barge-In, Initiate**
 - 20-13-16 Class of Service Options (Supplementary Service) – Barge-In, Receive**
 - 20-13-28 Class of Service Options (Supplementary Service) – Allow Class of Service to be Changed**
 - 20-13-35 Class of Service Options (Supplementary Service) – Block Camp On**
Set these options to 0 for the voice mail COS.

 - ➔ **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**
Enable Continued Dialing (1) for all extensions that dial Voice Mail features.

 - ➔ **22-01-04 System Options for Incoming Calls – DIL No Answer Recall Time**
If setting up Delayed Voice Mail Overflow, enter a time greater than 0. Overflow occurs after this time (provided the other related programming is correct). If setting up Immediate Voice Mail Overflow, enter 0. (0~64800 seconds). (default: 0).

 - ➔ **24-02-02 System Options for Transfer – MOH or Ringback on Transferred Calls**
Enable (1)/Disable (0) MOH on Transfer. If enabled, a transferred caller hears Music on Hold while their call rings the destination extension. If disabled, a transferred caller hears ringback while their call rings the destination extension. For this option to work with voice mail, the transferred call must be an unscreened transfer.

 - ➔ **45-01-18 Voice Mail Integration Options – Trunk Number Mapping (Version 1600 or higher required)**
2~3 (default: 2).

Assign Trunks As Automated Attendant Trunks – Method 1

- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign Service Type 4 to each trunk you want to ring into Voice Mail as a Direct Inward Line (DIL).

- ➔ **22-07-01 DIL Assignment**
Assign the master/pilot number of the voice mail group from Program 11-07-01 as the DIL destination. If all Voice Mail ports are in the same unique Extension (Department) Group (see Program 16-02 above), the DIL rings another Voice Mail port if its assigned port is busy.

Assign Trunks As Automated Attendant Trunks – Method 2

- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign Service Type 0 to each trunk you want to ring into Voice Mail as a normal line.
- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign Ring Group 102 for an In-Skin/External Voice Mail, or 103 for a Central Voice Mail as the destination.

For Either Method:

- ➔ **11-12-42 Service Code Setup (for Service Access) – Flash on Trunk Lines**
Program the dial access code to use for sending a hook flash to Telco. This code is used for Centrex Transfer using Digital Voice Mail ports. If this code starts with #, PRG 45-01-05 must be set to 0.
- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
To enable Voice Mail Overflow, assign selected extensions to a Ring Group that rings for unanswered DILs to Voice Mail ports. In Program 22-06, enter 1 to enable overflow ringing.
- ➔ **22-08-01 DIL/IRG No Answer Destination**
For Voice Mail Overflow, enter the Ring Group that unanswered DILs to Voice Mail rings after the DIL Call Waiting time (Program 22-01-04).
- ➔ **24-02-03 System Options for Transfer – Delayed Call Forwarding Time**
Set the time a transferred call waits at a forwarded extension before routing to the called extension mailbox.
- ➔ **45-01-01 Voice Mail Integration Options – Voice Mail Department Group Number**
Assign which Extension (Department) Group number is to be assigned as the voice mail group. This program defines the Quick Transfer to Voice Mail destination. If set to 0, no voice mail is installed.
Department Groups: 0, 1~64
- ➔ **45-01-02 Voice Mail Integration Options – Voice Mail Master Name**
Enter the Voice Mail master name up to 12 characters.
- ➔ **45-01-05 Voice Mail Integration Options – Message Wait**
When using Centrex Transfer From a voice mail port, the following items must be considered:
 - If the Feature Access Code starts with #, set PRG 45-01-05 to Off.
 - When assigning the dial string in voice mail, one or more Pauses may be needed too, depending on what Telco needs.
- ➔ **45-01-06 Voice Mail Integration Options – Record Alert Tone Interval Time**
Set the time (0~64800 seconds) between Voice Mail Conversation Record alerts.

➡ 45-01-18 Voice Mail Integration Options – Trunk Number Mapping (Version 2000 or higher required)

Set if the system sends 2 digit trunk numbers only or 2 and 3 digit trunk numbers to voice mail.

Options

2 = Trunks 01~99

3 = Trunks 001~200

(default: 2)

➡ 80-01-02 Service Tone Setup – Basic Tone Number

The following features require that the system tones listed below be changed to match the table. After changing these settings the KSU must be reset for the changes to take effect.

- Call Screening
- Call Holding
- Busy Greeting
- Await Answer Transfer

Tone 06			
Unit	Basic Tone	Duration	Gain Level
1	11 ~ 480/620Hz -13/-13dB	300ms	32
2	0 - No Tone	300ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

Tone 14			
Unit	Basic Tone	Duration	Gain Level
1	10 ~ 440/480Hz -13/-13dB	1000ms	32
2	0 - No Tone	2100ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	

Tone 14 (Continued)			
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

Tone 39			
Unit	Basic Tone	Duration	Gain Level
1	12 ~ 440/620Hz -16dB	500ms	32
2	0 - No Tone	500ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

➔ 80-03-01
through
80-03-08 **DTMF Tone Receiver Setup**

➔ 80-04-01
through
80-04-09 **Call Progress Tone Detector Setup**
Use these programs to set the criteria for DTMF dial, ringback and busy tones.

Operation

Calling Your Mailbox

To call your mailbox:

With a multiline terminal, our Voice Mail key and your Message Center keys flash red when they have messages waiting. If you do not have a Voice Mail key, your MW LED flashes instead.

Multiline Terminal


1. Press the **Voice Mail** key (PRG 15-07 or SC 751: 77).

- OR -

Press **VM Msg** softkey.

- OR -

Press **Speaker** and dial the Voice Mail Master Number. After Voice Mail Answers, dial your mailbox number.


 *Your mailbox number is normally the same as your extension number. You may optionally dial a co-worker's mailbox – or use this procedure to call your mailbox from a co-worker's telephone.*


- OR -

Press **Speaker** and dial ***8**.

2. To return a displayed Message Waiting, press **Speaker** or lift the handset.


To listen to the voice messages, with "Voice Message" displayed, press **Speaker** or lift the handset.

 *The voice mail is called.*

 *The voice mail is called only if there are new messages. If the display indicates Check Messages No Messages, press the EXIT key to return the telephone to idle.*

Single Line Telephone


1. Lift the handset and dial ***8**.


 *If you are at a co-worker's telephone, you can dial the Voice Mail master number and your mailbox number instead. You can also use this procedure from your own telephone to call a co-worker's mailbox.*

2. If requested by Voice Mail, enter your security code.

Checking Messages**If PRG 15-02-26 = 0 (Message Key):**


1. Press the **Message** key once.


 *The user can use the VOL UP and VOL DOWN keys to view the new messages. If there are both voice mail messages and Message Waiting calls, the display indicates the number of new voice mail messages and then each Message Waiting call is shown.*

 *When there are new messages, the MW LED on the telephone flashes red.*




2. **To return a displayed Message Waiting**, press the **Message** key a second time, press **Speaker** or lift the handset.

To listen to the voice messages, with VOICE MESSAGE displayed, press either the **Message** key a second time, press **Speaker** or lift the handset.

 *The voice mail is called.*


 *The voice mail is called only if there are new messages. If the display indicates Check Messages No Messages, press the Exit key to return the telephone to idle.*

If PRG 15-02-26 = 1 (Voice Mail Key):

1. Press the **Message** key once.
 -  *The voice mail is called.*
 -  *When there are new messages, the MW LED on the telephone flashes red.*
 -  *With this option set, the MSG key can be used as a Voice Mail key for any function calling voice mail or transferring a call to voice mail (Hold + MSG + Extension Number).*


Leaving A Message (Multiline Terminal Only)

To leave a message in the mailbox of an unanswered extension: (the extension you call can be busy, in DND or unanswered)

1. Press the **Voice Mail** key (PRG 15-07 or SC 751: code 77)
 - OR -
 - Dial **8**.
 -  *The Voice Mail system prompts you to leave a message.*

Forwarding Calls to your Mailbox

To activate or cancel Call Forwarding:

1. Press **Speaker** (or lift the handset at the single line telephone) and choose from the following dial access codes:
 - 741 = Call Forward – Immediate
 - 742 = Call Forward – Busy
 - 743 = Call Forward – No Answer
 - 744 = Call Forward – Busy/No Answer
 - 745 = Call Forward – Both Ring
 - 746 = Call Forwarding – Follow Me
2. Dial the Voice Mail master number.
3. Press **Speaker** to hang up (or hang up handset at the single line telephone).
 -  *Your DND or Call Forwarding (Station) key flashes when Call Forwarding is activated.*

Transferring Calls to a Mailbox

To Transfer your active call to a mailbox:

Multiline Terminal


1. Press **Hold**.

2. Press the **Voice Mail** key (PRG 15-07 or SC 751: code 77 + VM Pilot).

- OR -


Press the **Message** key.

3. Dial number of mailbox to receive the transfer.


 *This can be your mailbox number or a co-worker's mailbox number.*

- OR -

Press DSS Console or One-Touch key for the extension mailbox that receives the Transfer.

 *If the Transfer destination is an extension forwarded to Voice Mail, the call waits before routing to the called user's mailbox. This gives you the option of retrieving the call instead of having it picked up by Voice Mail.*

4. Hang up.

 *Voice Mail prompts your caller to leave a message in the mailbox you selected.*

- OR -


1. Dial extension number or press a DSS Console key for the extension mailbox that will receive the Transfer.

2. Press the **Voice Mail** key (PRG 15-07 or SC 751: code 77 + VM Pilot)

- OR -

Press the **Message** key.


3. Hang up.

 *Voice Mail prompts your caller to leave a message in the mailbox you selected.*

Single Line Telephone

1. Hookflash.

Dial Voice Mail master number followed by destination mailbox.





 *If the Transfer destination is an extension forwarded to Voice Mail, the call waits before routing to the called user's mailbox. This gives you the option of retrieving the call instead of having it picked up by Voice Mail.*

2. Hang up.



Recording Your Call

To record your active call in your mailbox:

Multiline Terminal

1. Press the **Voice Mail Record** key (PRG 15-07 or SC 751: code 78)
 -  *You hear two beeps and your Record key flashes. The beeps periodically repeat to remind you that the call is recording.*
 -  *To stop recording, press the Voice Mail Record key again or press the End softkey. To pause recording press the Pause softkey. You can restart and stop recording as required.*
- OR -**
1. Press **Hold** to place the call on hold.
 2. Dial **654** or press flashing line key where the call resides.
 -  *The system automatically reconnects you to your call.*
 -  *To stop recording, press the Voice Mail Record key again or press the End softkey. To pause recording, press the Pause softkey. You can restart and stop recording as required.*

Single Line Telephone

1. Hookflash.
2. Dial **654**.
 -  *The system automatically reconnects you to your call.*
 -  *The recording continues until the caller hangs up.*

Live Monitor, Answering Machine Emulation (Multiline Terminal Only)



To enable or cancel Personal Answering Machine Emulation:

1. Press the **Live Monitor** key (PRG 15-07 or SC 751: code 91).


When Live Monitor Broadcasts your Caller's Message, you can:

With voice mail set to Automatic Live Monitor:

Your telephone must be idle. After voice mail provides record beep to caller, the phone starts playing audio on speaker. The telephone display softkeys change to offer Cancel selection.

1. Listen to the message as it is being recorded.
 -  *The message is automatically being recorded in your mailbox.*
- OR -**
- To stop audio press the **Cancel** softkey.
-  *The message is automatically being recorded in your mailbox.*


- (Optional) Lift the handset to take call.

 *You connect to the caller. The system records the first part of the message in your mailbox.*

With voice mail set to Manual Live Monitor:


Your telephone must be idle. After voice mail provides record beep to the caller, the telephone display softkeys change offering Start and Cancel selections.

- To listen to the message as it is being recorded, press the **Start** softkey.


 *You do not connect to the call and the audio is heard on the speaker. The message is still being recorded in your mailbox.*

- OR -

To decline Live Monitor session press the **Cancel** softkey.

 *You do not connect to the call and the message is automatically recorded in your mailbox.*


- (Optional) While listening to caller leave message, lift the handset to take the call.

 *You connect to the caller. The system records the first part of the message in your mailbox.*

Checking Your Messages (Multiline Terminal Only)

To check your messages:

- Press the **Message** key.

 *You can have any combination of the messages in the table below on your telephone.*

If you see. . .	You have. . .
VOICE MESSAGE n MESSAGES	New messages in your Voice Mail mailbox
CHECK MESSAGE VRS GENERAL MESSAGE	Not listened to the current General Message
CHECK MESSAGE (name)	Message Waiting requests left at your telephone by your co-workers

- Press VOL ▲ or VOL ▼ to scroll through your display.
- When you find the message you want to answer, press **Speaker**. You either:
 - Go to your Voice Mail mailbox.
 - Automatically call the extension that left you a Message Waiting.

Direct Inward Dialing (DID)

Enhancements

Version 1500 or higher is required for the DID Call By Time Schedule feature.
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Description

Direct Inward Dialing (DID) lets outside callers directly dial system extensions. DID saves time for callers who know the extension number they wish to reach. To place a DID call, the outside caller dials the local exchange (NNX) and additional digits to ring the telephone system extension. For example, DID number 926-5400 can directly dial extension 400. The caller does not have to rely on attendant or secretary call screening to complete the call.

 *Direct Inward Dialing requires DID service from Telco.*

In addition to direct dialing of system extensions, DID provides:

- DID Dialed Number Translation
- Flexible DID Service Compatibility
- DID Intercept
- DID Camp-On

There are 20 DID Translation tables that can be divided between 2000 entries.

DID Dialed Number Translation

DID allows different tables for DID number translation. This gives you more flexibility when buying DID service from Telco. If you cannot buy the exact block of numbers you need (e.g., 301~556), use the translation tables to convert the digits received. For example, a translation table could convert digits 501~756 to extension numbers 301~556.

The Electra Elite IPK II system has 2000 DID Translation Table entries that you can allocate among the 20 DID Translation Tables. One translation is made in each entry. For a simple installation, you can put all 2000 entries in the same table. For more flexibility, you can optionally distribute the 2000 entries among the 20 tables.

In addition to number conversion, each DID Translation Table entry can have a name assigned to it. When the DID call rings the destination extension, the programmed name displays.

Flexible DID Service Compatibility

With three-digit service, the Telco sends three digits to the system for translation. Be sure to program your system for compatibility with the provided Telco service. For example, if the Telco sends four digits, make sure you set up the translation tables to accept four digits.

The system is compatible with Dial Pulse (DP) and DTMF DID signaling. DID trunks can be either wink start or immediate start.

DID Camp-On

DID Camp-On sets what happens to DID calls to busy extensions when you have Busy Intercept disabled. With DID Camp-On enabled, a call to a busy extension camps-on for the DID Ring No Answer Time. It then diverts to the programmed DID Intercept extension ring group or Voice Mail. Without DID Camp-On, the caller to the busy extension hears only busy tone.

DID Routing Through the VRS

DID calls can optionally route through the VRS. The DID caller hears an initial Automated Attendant Greeting explaining their dialing options. If the caller misdials, they can hear a second greeting with additional instructions. For example, the first Automated Attendant Greeting can be, "Thank you for calling. Please dial the extension number you wish to reach or dial 0 for the operator." If the caller inadvertently dials an extension that does not exist, they could hear, "The extension you dialed is unavailable. Please dial 0 for assistance or dial # to leave a message so we can call you back."

You assign Automated Attendant greetings (i.e., VRS Messages) to the numbers in each Translation Table. This provides you with extensive flexibility when determining which greetings the system should play for which dialed numbers. You could, for example, set up 926 5401 through 926 5449 to route to extensions 401~449, and have 926 5450 route to the automated attendant.

The system allows an extension to be defined as a 1-digit number that can be dialed by the outside caller on a DID/DISA trunk using the VRS. The outside caller can access the desired extension/department group by dialing only one digit after the system answers the call. If the same number is used as the first digit of an extension number and the 1-digit access code for DID/DISA, the outside caller cannot access the extension.

EXAMPLE:

If 2 is defined as a 1-digit access code to department group 300, outside callers cannot access extensions 200~299 directly.

SMDR Includes Dialed Number

The SMDR report can optionally print the trunk name (entered in system programming) or the number the incoming caller dialed (i.e., the dialed DID digits). This allows you to analyze the SMDR report based on the number your callers dial. (This option also applies to an ISDN trunk.)

DID Intercept

DID Intercept automatically reroutes DID calls under certain conditions. There are three DID Intercepts:

Vacant Number Intercept

If a caller dials an extension that does not exist or misdials, Vacant Number Intercept can reroute the call to the programmed DID Intercept extension ring group or Voice Mail. Without Vacant Number Intercept, the caller hears error tone after mis-dialing.

Busy Intercept

Busy Intercept determines DID routing when a DID caller dials a busy extension. If Busy Intercept is enabled, the call immediately routes to the programmed DID Intercept extension ring group or Voice Mail. If Busy Intercept is disabled, the call follows DID Camp-On programming.

Ring-No-Answer Intercept

Ring-No-Answer Intercept sets the routing options for DID calls that ring unanswered at the destination extension. With Ring-No-Answer Intercept enabled, the unanswered call reroutes to the DID Intercept extension ring group or Voice Mail after the DID Ring-No-Answer Time. If Ring-No-Answer Intercept is disabled, the unanswered call rings the destination until the outside caller hangs up.

Delayed DID

Delayed DID allows a user a programmed time to answer a call. If the call is not answered within this time, the system automatically answers the call. An outside party hears a voice message, music, or dial tone according to the following conditions:

- If a VRS is installed, the system sends a prerecorded message from the VRS.
- If a customer-provided audio system (example: tape recorder) is connected, an error message or music can be played for the caller.
- If equipment is not connected for an announcement, the system sends a unique dial tone to the outside caller.

This feature is not available for the normal incoming call on ISDN trunks.

DID Intercept Destination for Each DID Number

With this feature the system allows you to program a DID Intercept destination for a DID number which receives no answer or busy call. The system can be programmed to use a trunk ring group, the VRS or the voice mail as the programmed destination. Each vacant number intercept for a DID number can have two destinations. The first destination is for an invalid DID number, busy or no answer extension. The second destination is for a no answer trunk ring group.


For busy or no answer intercept calls, a third destination can be defined in Program 22-12. If the first and third destinations are programmed but the second destination is not, the incoming call goes to the third destination after the first destination. If the first and second destinations are not programmed, but the third destination is, the call goes directly to the third destination.

This feature works for DID trunks with a trunk service type 1 in Program 22-02. Other trunks may use the DID table, but the DID intercept feature is not yet supported.

With the DID Intercept for each DID number feature, when the primary destination (PRG 22-11-05) is set to Voice Mail, the Voice Mail protocol is:

1. Busy Intercept = Forward Busy
2. Ring-No-Answer Intercept = Forward RNA

When the secondary destination (Program 22-11-06) is set to Voice Mail, the Voice Mail protocol is based on the first destination routing. When the incoming call is forwarded to the first destination by a busy intercept, the Voice Mail protocol forwards busy calls. When the incoming call is routed to the first destination by a ring-no-answer intercept, the protocol forwards ring-no-answer. The Voice Mail transfers the calls to the mailbox number defined in Program 22-11-02.

 *Any valid DID number must be entered in the DID table (Program 22-11 or Program 22-17-01). If a valid DID number is not entered, there is no ring destination for any incoming calls to that number (the calls do not ring any extension in the system).*

Calls Can Follow Ring Group Programming for Transferring Calls

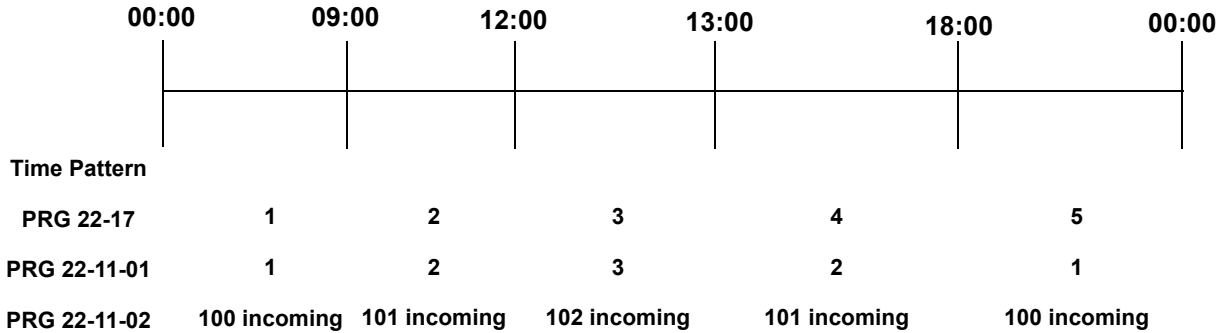
An option has been added to Program 22-11 which allows you to determine if the DID routing should use the programmed ring group entry in Program 22-12-01 when transferring calls from a busy or no answer number.

If DID digits match with the conversion table but there is no extension, no Voice Mail, or Voice Mail did not boot up, use Program 22-11-11 to decide what to do with the incoming call. Go to normal ring (1: default) or caller hears a Busy Tone (0).

DID Call by Time Schedule

DID Call by Time Schedule allows for 100 programmed DID Conversion table entries (**PRG 22-17-01**) that can be routed based on Time Patterns. Each DID Conversion table has a maximum of eight programmable Time Patterns and each Time Pattern can reference any of the 2000 different Dial-In Conversion table entries in **PRG 22-11-01**.

Example 1 (Automatic Change)



PRG 22-11-01 and PRG 22-11-02		
Table No.	Receive Dial	Transfer Dial
1	No setting	100
2	No setting	101
3	No setting	102

PRG 22-17					
Table No.	Receive Dial	Time Pattern	Start Time	End Time	PRG 22-11
1~100	1111	1	00:00	09:00	1
		2	09:00	12:00	2
		3	12:00	13:00	3
		4	13:00	18:00	2
		5	18:00	00:00	1
		6	00:00	00:00	0
		7	00:00	00:00	0
		8	00:00	00:00	0

Table 2-15 Example 2 (Manual Change)

PRG 22-17					
Table No.	Receive Dial	Time Pattern	Start Time	End Time	PRG 22-11
1~100	1111	1	00:00	00:00	1
		2	00:00	00:00	2
		3	00:00	00:00	3
		4	00:00	00:00	0
		5	00:00	00:00	0
		6	00:00	00:00	0
		7	00:00	00:00	0
		8	00:00	00:00	0

Federal Communications Commission DID Requirements

Allowing this equipment to operate in a manner that does not provide proper answer supervision signaling is in violation of Part 68 rules.

This equipment returns answer supervision to the Public Switched Telephone Network when the DID trunk is:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user
- Routed to a dial prompt

This equipment returns answer supervision on all DID calls forwarded back to the Public Switched Telephone Network. Permissible exceptions are when:

- A call is unanswered.
- A busy tone is received.
- A reorder tone is received.

When ordering DID service, provide the Telco with the following information:

Electra Elite IPK II	KF = US:NIFKF07B
	MF = US:NIFMF07B
	PF = US:NIFPF07B
DID Facility Interface Code	02RV2-T
DID Service Order Code	9.0F
DID Answer Supervision Code	A S.2
DID USOC Jack Type	RJ21X

Conditions

- Analog DID requires installation of a DID(4)-U10 ETU (provides four DID ports). Depending on programming, the system may assign both trunk and extension ports (if OPX is selected in Program 10-03-01) when this ETU is installed.
- DID service must be purchased from your local telephone company.
- DID Intercept for each DID number works for DID trunks with a trunk service type 3 in Program 22-02. Other trunks may use the DID table, but the DID intercept feature for each DID number is not yet supported.
- T1 or PRI DID requires a DTI-U40 ETU.
- When a call is transferred because of Call Forwarding No Answer, Call Forwarding Busy, or DND, the reason for Transfer option can display to the transferred extension when the call is ringing to their telephone.
- Direct Inward Lines (DILs) also allow outside callers to dial a system extension, virtual extension, or Department Group directly.
- DISA also allows outside callers to dial a system extension directly.
- The Off-Hook Signaling provide DID calls with signaling options. Refer to Off-Hook Signaling for specific details.
- DID trunks do not ring external page speakers. Only trunks defined as normal in Program 22-02-01 ring external page speakers.
- To simplify answering DID calls, assign function keys as line keys for the DID trunks.
- SMDR can print trunk port names or received dialed number for ANI/DNIS or DID trunks. If enabled, DNIS digits can be printed on the SMDR reports instead of the trunk name.
- Transferred calls on DISA, DID, DIL, ISDN trunks, or from the VRS can display the reason a call is being transferred (Call Forward, Busy, No Answer, or DND).
- When defining trunks as DID or DID Mode in PRG 22-02-01, DID translation (PRG 22-11 or PRG 22-17) must be used, even if the incoming digits match the extension number.

- When using DID Call by Time Schedule and breaking out the Time Patterns, set the start time to 00:00 and end time to 00:00 for this feature to operate correctly. Refer to [DID Call by Time Schedule on page 2-325](#) for more details.
- DID Call by Time Schedule Priority is given to the pattern that is set **manually**. However, when a time pattern changes with the time schedules set in PRG 22-17, the pattern applied by the Manual change is canceled and the Time Pattern is given priority.
- When Transfer Operation Mode is set to busy, call queuing must be turned off for it to work.

Default Setting

Disabled

Related Features

Central Office Calls, Answering

Direct Inward Line (DIL)

Direct Inward System Access (DISA)

Off-Hook Signaling

Paging, External

Programmable Function Keys

Station Message Detail Recording

Transfer

Programming

- ➔ **10-03-01 ETU Setup**
Set the function of the DID(4)-U() ETU port for DID (0).

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-
- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**

If the system has DTMF DID trunks, be sure to reserve at least one circuit for analog trunk DTMF reception (type 0 or 2). Each DTMF DID trunk must have an available receiver. Use the following as a guide when allocating DTMF receivers:

 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.
 - ➔ **14-05-01 Trunk Group – Trunk group Number**

Put DID trunks in the same trunk group (other than group 1). If you have several types of DID trunks, put each type in a separate trunk group.
 - ➔ **15-07-01 Programmable Function Keys**

Assign line or Call Appearance (CAP) Keys for DID trunks (Trunks: 1~200).
 - ➔ **20-06-01 Class of Service for Extensions**

Assign a Class of Service (1~15) to an extension.
 - ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**

Turn Off (0) or On (1) an extension ability to receive a second call from a DID, DISA, DIL, or Tie Line caller. (default: 1).
 - ➔ **20-13-23 Class of Service Options (Supplementary Service) – Display the Reason for Transfer**

Enable (1)/Disable (0) an extension ability to display the reason (Call Forward, Busy, No Answer or DND) a VRS, DID, DISA, or ISDN call is being transferred to their extension. (default: 0).
 - ➔ **21-01-02 System Options for Outgoing Calls – Intercom Interdigit Time**

Set the time-out time for DID callers that do not dial. After this time, the DID call routes according to Vacant Number Intercept programming. (default: 10).
 - ➔ **22-01-06 System Options for Incoming Calls – DID Ring-No-Answer Time**

Set the DID Ring No Answer (RNA) Intercept time (0~64800 seconds). In systems with RNA Intercept, the DID call rings the destination extension for this time, and then rings Intercept Ring Group. (default: 20).
 - ➔ **22-02-01 Incoming Call Trunk Setup**

For each Night Service Mode, enter service type 3 when the trunk should be a DID trunk.
 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**

Assign extensions to Ring Groups. Calls ring the extensions according to programming in PRG 22-06.
 - ➔ **22-09-01 DID Basic Data Setup – Expected Number of Digits**

For each DID Translation Table (1~20), enter the number of digits the table expects to receive from the CO (eight maximum). For example, for a table used with 3-digit DID service, enter 3.
 - ➔ **22-09-02 DID Basic Data Setup – Received Vacant Number Operation**

Selectively Enable (1)/Disable (0) Vacant Number Intercept based on each DID Translation Database (Program 22-11-01 and Program 22-11-02).
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- ➔ **22-10-01 DID Translation Table Setup**
Assign the start and end range of DID Translation Table entries (1~2000) to each DID Translation Table (1~20).
 - ➔ **22-11-01 DID Translation Number Conversion – Received Number**
For each DID Translation Table entry (1~2000), specify the digits received by the system (eight maximum).
 - ➔ **22-11-02 DID Translation Number Conversion – Target Number**
For each DID Translation Table entry (1~2000), specify the extension the system dials after translation (24 digits maximum).
 - ➔ **22-11-03 DID Translation Number Conversion – DID Name**
For each DID Translation Table entry (1~2000), specify the name that should show on the dialed extension display when it rings (12 characters maximum).
 - ➔ **22-11-04 DID Translation Number Conversion – Transfer Operation Mode**
For each DID Translation Table entry (1~2000), specify the condition required to transfer the call to the destination defined in PRG 22-11-05 and PRG 22-11-06.
 - ➔ **22-11-05 DID Translation Number Conversion – Transfer Destination Number 1**
 - ➔ **22-11-06 DID Translation Number Conversion – Transfer Destination Number 2**
For each DID Translation Table entry (1~2000), specify the first and second Transfer Destinations if the caller receives a busy or no answer (defined in PRG 22-11-04).

Destination Options:
0 = No Setting
1~100 = Incoming Group
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
201~264 = Extension Group
400 = Valid Extension Number/DID termination to VRS
401 = DISA
501~548 = DISA/VRS Message
1000~1999 = Speed Dial Bin Number (000~999)

☞ If the Transfer Destinations are busy or receive no answer, those calls are transferred to the final transfer destination (Program 22-10).
 - ➔ **22-11-07 DID Translation Number Conversion – Call Waiting**
For each DID Translation Table entry (1~2000), specify whether or not Call Waiting should be allowed (Enable (1)/Disable (0)).
 - ➔ **22-11-08 DID Translation Number Conversion – Maximum Number of DID Calls**
For each DID Translation Table entry (1~2000), specify the maximum number of DID calls.

0 = Unlimited (default)
1~200
-
-

-
-
- ➔ **22-11-09 DID Translation Number Conversion – Music On Hold Source**
For each DID Translation Table entry (1~2000), specify the source of music to be used for DID trunks.

0 = IC/MOH Port (default)
1 = BGM Port
2 = ACI Port
 - ➔ **22-11-10 DID Translation Number Conversion – ACI Music Source Port**
For each DID Translation Table entry (1~2000), if item 2 is selected in PRG 22-11-09, specify the port to be used for the source of music heard on DID trunks. (default: 0).
 - ➔ **22-11-11 DID Translation Number Conversion – Ring Group Transfer**
Enable (1)/Disable (0) each conversion table to follow the Ring Group programming defined in Program 22-12-01 : DID Intercept Ring Group. If Program 22-11-05 : DID Translation Number Conversion, Transfer Destination Number 1 and Program 22-11-06 : DID Translation Number Conversion, Transfer Destination Number 2 are set, the priority of transferring is in this order: Program 22-11-05 then Program 22-11-06 then if Program 22-11-11 is enabled, Program 22-12-01.
 - ➔ **22-12-01 DID Intercept Ring Group**
For each DID Translation Table, program the DID Intercept destination. The destination can be a Ring Group, In-Skin/External Voice Mail, or Centralized Voice Mail. This program is used when no destination is programmed in Program 22-11-05. It is unrelated to Program 22-11-06 and PRG 22-11-07.

Ring Groups:
1~100 (default: 1)
102 = In-Skin/External Voice Mail
 - ➔ **22-13-01 DID Trunk Group to Translation Table Assignment**
Assign the DID trunk groups to translation tables. If all the DID trunks use the same type of DID service, you may have only one DID trunk group and one DID Translation Table (with many entries).

Trunk Groups: 1~100
 - ➔ **25-01-01 VRS/DISA Line Basic Data Setup – VRS/DISA Dial-In Mode**
Determine whether the system should use Extension number/Service code specify (0)) or Dial conversion table (1) for calls.
 - ➔ **25-01-03 VRS/DISA Line Basic Data Setup – VRS/DISA Transfer Alarm**
Determine whether the system should use Normal (0) or Alarm (1) for calls.

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- ➔ **25-02-01 VRS/DISA VRS Message**
For each trunk port and each night mode, select the message source (0 = No Message, 1 = VRS, 2 = ACI, 3 = S LT), assign the VRS message number used as the Automated Attendant Message for each trunk, which is assigned as VRS/DISA [with VRS = 01~48 (VRS message number), with ACI = 1~4 or 01~16 (ACI group number), with SLT = 1~8 or 01~64 (Department Group number)].
- Trunk Ports:
001~200
Night Time Service Modes:
1~8
- ➔ **25-03-01 VRS/DISA Transfer Ring Group With Incorrect Dialing**
For each trunk port, set what happens to a call when the DISA or Automated Attendant caller dials incorrectly or waits too long to dial. The call can either disconnect (0) or Transfer (1) to an alternate destination (a ring group, In-Skin/External, Centralized). When setting the DISA and DID Operating Mode, make an entry for each Night Service mode.
- Ring Groups:
1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
Trunk Ports:
001~200
Night Service Time Modes:
1~8
- ➔ **25-04-01 VRS/DISA Transfer Ring Group With No Answer/Busy**
For each trunk port (001~200), set the operating mode of each DISA trunk. This sets what happens to the call when the DISA or Automated Attendant caller calls a busy or unanswered extension. The call can either disconnect (0) or Transfer (1) to an alternate destination (a ring group, In-Skin/External, Centralized). When setting the DISA and DID Operating Mode, you make an entry for each Night Service mode.
- Ring Groups:
1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
Trunk Ports:
001~200
Night Service Time Modes:
1~8
- ➔ **25-05-01 VRS/DISA Error Message Assignment**
For each trunk that is answered by the VRS, enter the VRS message (1~48) the outside caller hears if they dial incorrectly after answer. If you enter 0, the call reroutes according to Program 25-03 and Program 25-04. Make one entry for each Night Service mode.

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- ➔ **25-06-02 VRS/DISA One-Digit Code Attendant Setup – Destination Number**

Set up single digit dialing for Automated Attendant callers. For each VRS Message programmed to answer outside calls, specify:

 - The digit the Automated Attendant caller dials (1~12, where 10 = 0, 11 =* and 12 = #).
(Remember that if you assign destinations to digits three and four, outside callers cannot dial system extensions that begin with that digit.)
 - The destination reached (four digits maximum) when the caller dials the single digit code.

 - ➔ **25-07-01 System Timers for VRS/DISA – VRS/DISA Dial Tone Time**

After answering a VRS/DISA trunk, the system waits this time for the caller to dial the first digit of the DISA password. If the caller fails to dial within this time, the system drops the call (0~64800 seconds).

 - ➔ **25-07-02 System Timers for VRS/DISA – VRS/DISA No Answer Time**

A VRS/DISA caller can ring an extension for this time before the system sets the call as a Ring No Answer. After this time expires, the call follows the programmed Ring No Answer routing (set in Program 25-03 and PRG 25-04) (0~64800 seconds).

 - ➔ **25-07-03 System Timers for VRS/DISA – Disconnect after VRS/DISA retransfer to IRG**

(0~64800 seconds). (default: 60).

 - ➔ **25-07-04 System Timers for VRS/DISA – Calling Time to Automatic Answering Telephone Set**

Set the answering waiting time of the automatic answering extension when an incoming DID trunk call is received (0~64800 seconds).

 - ➔ **25-07-05 System Timers for VRS/DISA – Duration Time for Guidance Message by Automatic Answering Telephone Set**

Set the announcement time of the automatic answering extension before an incoming DID trunk caller is disconnected (0~64800 seconds).

 - ➔ **25-07-06 System Timers for VRS/DISA – Duration Time for Guidance Message by ACI**

Set the announcement time by the ACI before an incoming DID trunk caller is disconnected (0~64800 seconds).

 - ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**

Determine the time a DISA caller or any trunk-to-trunk (such as Tandem Trunking) conversation can talk before the Long Conversation tone is heard. (0~64800 seconds). (default: 3600).

 - ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect Time**

Determine how long the system waits before disconnecting a DISA caller or any trunk-to-trunk (such as Tandem Trunking) conversation call after the Long Conversation tone is heard. (0~64800 seconds). (default: 10).

 - ➔ **25-07-09 System Timers for VRS/DISA – DISA Internal Paging Time**

Set the maximum length of an Internal Page placed by a DISA caller. If the Page continues longer than this time, the system terminates the DISA call. (0~64800 seconds). (default: 30).

 - ➔ **25-07-10 System Timers for VRS/DISA – DISA External Paging Time**

Set the maximum length of an External Page placed by a DISA caller. If the Page continues longer than this time, the system terminates the DISA call. (0~64800 seconds). (default: 30).
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- ➔ **25-07-11 System Timers for VRS/DISA – VRS/DISA Answer Delay Time**
Set the time the system waits after receiving an incoming VRS/DISA call before the system automatically answers the call (0~64800 seconds).
- ➔ **25-07-13 System Timers for VRS/DISA – VRS/DISA Busy Tone Interval**
If a DISA caller dials a busy extension (and Program 25-04-01 = 0), the system plays busy tone for this time before disconnecting (0~64800 seconds).
- ➔ **25-07-14 System Timers for VRS/DISA – Delayed VRS Answer Time**
Assign the delay time from switching from a normal incoming status to DID mode. If this time is set to 0, the call switches to DID mode immediately (0~64800 seconds).
- ➔ **34-01-01 E&M Tie Line Basic Setup – DID/E&M Start Signaling**
Set the start signaling mode for DID and tie trunks. DID and tie trunks can use either immediate start or wink start signaling.

0 = 2nd Dial Tone
1 = Wink (default)
2 = Immediate
3 = Delay

DID Call by Time Schedule (in addition to the above programming)

- ➔ **11-10-35 Service Code Setup (for Administrator) – Dial-In Mode Switching**
Assign the service code Dial-In Mode Switching (Default: Not Set)
- ➔ **15-07-01 Programming Function Keys**
Assign a function key for one-touch access to the Dial-In Mode Switching setup code (Code 88).
- ➔ **20-07-26 Class of Service Options (Administrator Level) – Dial-In Mode Switch**
Enable (1)/Disable (0) an extension user ability to manually change Dial-In Modes.
- ➔ **22-02-01 Incoming Call Trunk Setup**
For each Night Service Mode, enter service type 8 when the trunk should be a DID (DDI) Mode Switching trunk.
- ➔ **22-11-02 DID Translation Table Number Conversion – Target Number**
For each DID Translation Table entry (1~2000), specify the extension the system dials after Translation in PRG 22-17 (24 digits maximum).
☞ Do not assign Received Digits in PRG 22-11-01 when using DID Call by Time Schedule.
- ➔ **22-17-01 through 22-17-04 Dial-In Conversion Table Area Setup for Time Pattern**
Define the Time Zone and Dial-In Conversion Table (PRG 22-11) for each Time Pattern.
- ➔ **30-03-01 DSS Console Key Assignment**
Customize DSS Console keys to function as Dial-In Mode Switching keys. Use the key [when defined as a Dial-In Mode Switching key (code 88)] to manually switch between Dial-In Modes.

Operation

DID calls ring extensions like normal trunk calls.

To Activate DID Call by Time Schedule:

1. At any display multiline terminal, press **Speaker**.
2. Dial the Dial-In Mode Switching Service Code (Default = Not assigned).
 - OR -
 Press the Dial-In Mode Switching Programmable Function key (PRG 15-07-01, 88, or SC 751 Key Code 88).
3. Dial **1~100** (table number).
4. Dial the Time Pattern **1~8**.

Table 2-16 LED Flash Patterns

Time Pattern	LED Appearance
Pattern 1	Off
Pattern 2	On
Pattern 3	Slow Flash
Pattern 4	Fast Flash
Patterns 5~8	Off

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Direct Inward Line (DIL)

Description

A Direct Inward Line (DIL) is a trunk that rings an extension, virtual extension or Department Group directly. Since DILs only ring one extension or group (i.e., the DIL destination), employees always know which calls are for them. For example, a company operator can have a Direct Inward Line for International Sales Information. When outside callers dial the DIL telephone number, the call rings the operator on the International Sales line key. The DIL does not ring other extensions.

There are 200 available trunks, 64 Department Groups, 256 extensions and 256 virtual extensions.

DIL Delayed Ringing

Extensions in a Ring Group can have delayed ringing for another extension DIL. If the DIL is not answered at its original destination, it rings the DIL No Answer Ring Group. This could help a Technical Service department, for example, that covers calls for an Inside Sales department. If the Inside Sales calls are not answered, they ring into the Technical Service department.

Conditions

- If unanswered, a DIL without delayed ringing rings an extension until the outside party hangs up.
- If a DIL rings a Department Group and all agents are busy, the system routes the call as follows:
 1. The trunk rings the overflow destination assigned in PRG 22-08.
 2. If there is no 22-08 assignment, the call rings according to the Ring Group assignments in PRG 22-04 and PRG 22-05.
 3. If none of the destinations in steps 1~2 above are available, the call continues to ring until a destination becomes free.
- The DIL follows call forwarding programming, even to voice mail.
- When a call is transferred by Call Forwarding – No Answer, Call Forwarding – Busy, or DND, the Reason for Transfer can display at the transferred extension.
- You can place DILs in trunk groups to make outgoing DIL calls easier.
- A DIL can have an Extension (Department) Group as its destination.
- If a DIL destination extension is in DND, an incoming call rings according to Ring Group programming (PRG 22-08 then PRG 22-05).
 - *If a user puts the telephone in Do Not Disturb, calls routed to the telephone in DND **do not** follow call forwarding.*

- A user can activate Group Call Pickup to intercept a DIL ringing another extension.
- Program a name for a DIL in Program 14-01-01. This makes it easier to identify the incoming call.
- If a Multiline Terminal is busy, a second incoming DIL call provides Call Alert Notification, depending on KSU programming. The second DIL call waits in line for the user to answer the call. The outside caller hears ringback tone while this occurs.
- If an extension has a line key for a DIL, the call rings the key. If not, the call rings an available line appearance. For other extensions, the DIL indicates as busy.
- A DIL rings its assigned extension without Ring Group programming. A DIL only rings its assigned extension. It does not ring other extensions in a Ring Group.
- Transferred calls on DISA, DID, DIL, ISDN trunks, or from the VRS can display the reason a call is being transferred (Call Forward, Busy, No Answer, or DND).

Default Setting

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Call Forwarding

Central Office Calls, Answering

Central Office Calls, Placing

Department Calling

Do Not Disturb

Group Call Pickup**Name Storing****Off-Hook Signaling****Paging, External****Programmable Function Keys****Ring Groups****Transfer****Programming**

- ➔ **15-07-01 Programmable Function Keys**
To have the DIL ring a key, program a line key for the DIL trunk.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
Enable (1)/Disable (0) the ability to receive a second call from a DID, DISA, DIL, or Tie Line caller.
- ➔ **20-13-23 Class of Service Options (Supplementary Service) – Display the Reason for Transfer**
Enable (1)/Disable (0) the display of the reason (Call Forward, Busy, No Answer or DND) a VRS, DID, DISA, or ISDN call is being transferred.
- ➔ **22-01-04 System Options for Incoming Calls – DIL No Answer Recall Time**
Set the DIL No Answer Time (0~64800 seconds). If DIL Delayed Ringing is set in Program 22-08, this option sets the DIL Delayed Ring time.
- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign each DIL Service Type 4. Make an entry for each Night Service mode.
- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign the extensions that should receive the overflow to the ring group programmed in PRG 22-08. Set the ringing in Program 22-06.

0 = Trunks Do Not Ring, but flash line keys.
1 = Trunks Ring

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- ➔ **22-07-01 DIL Assignment**
Set the destination extension number for each DIL – for each Night Service mode. The destination can be an extension port, virtual extension number, or Department Group pilot number (as assigned in PRG 11-07-01).

 - ➔ **22-08-01 DIL/IRG No Answer Destination**
For each DIL with delayed ringing, enter the DIL No Answer Ring Group. An unanswered DIL rings this group after the DIL No Answer Time. Make an entry for each Night Service mode.

Ring Groups: 1~100

Operation

To answer a call on your Direct Inward Line:

1. Lift the handset.
2. Press the flashing line key for DIL on the multiline terminal.
 - Press the flashing Answer Key to put the first call on hold and answer the second incoming call. This can be repeated until all incoming calls are answered.
 - If you have Ringing Line Preference, lift the handset to answer the call.
 - If you do not answer the call, it may ring other extensions (i.e., the DIL No Answer Ring Group).

To place a call on your Direct Inward Line:

1. Lift the handset.
2. At the multiline terminal, press the line key for DIL.
 - OR -
 - Dial # **9** and the DIL trunk number (e.g., 005).
 - OR -
 - Dial **704** and the DIL trunk group number (e.g., 05).
 - OR -
 - Dial **9** for Trunk Group Access.
3. Dial the number.

Direct Inward System Access (DISA)

Description

DISA permits outside callers to directly dial system extensions, trunks and selected features. This could help an employee away from the office that wants to directly dial co-workers or use the company's trunks for long distance calls. To use DISA, the employee:

- Dials the telephone number that rings the DISA trunk
- Waits for the DISA trunk to automatically answer with a unique dial tone
- Dials the 6-digit DISA password (access code)
- Waits for a second unique dial tone
- Accesses a system trunk, uses a selected feature or dials a system extension

DISA calls ring system extensions like other outside calls. If an extension has a line key for the DISA trunk, the call rings that key. If the extension does not have a line key, the extension must have a Call Appearance (CAP) key to answer the call.

You can set DISA operation differently for each Night Service mode. For example, a trunk can be a normal trunk during the day and a DISA trunk at night. You can also set the routing for DISA trunks when the caller dials a busy or unanswered extension, dials incorrectly or forgets to dial.

DISA allows 15 users, 15 DISA Classes of Service and 200 trunks.

DISA Class of Service

DISA Class of Service provides features and dialing restrictions for DISA callers. This allows you to control the abilities of the DISA callers dialing into your system. When a DISA caller first accesses the system, they can be prompted to enter a DISA password before proceeding. The system associates the password entered with a specific user number, which in turn has a Class of Service. If the Class of Service allows the action (such as making outgoing trunk calls), the call goes through. If the DISA Class of Service does not allow the action, the system prevents the call. The DISA Class of Service options are:

- Trunk Group Routing/ARS Access

When a DISA caller dials into the system, they can dial 9 and place outside calls. Any toll charge is incurred by the system. The call follows the system Trunk Group Access or Automatic Route Selection – whichever is enabled.

Trunk Group Access

DISA callers may access a specific trunk group for outgoing calls through the system. To access a Trunk Group, the user dials Service Code 704 followed by the Trunk Group number (Trunk Groups 1~100). This allows the DISA caller to place an outgoing call over the selected group. Trunk Group Access bypasses the system Trunk Group Routing/ARS/Trunk Access Maps. As with dial 9 access, any toll charge is incurred by the system.

 Speed Dial – System/Group/Station

The System Speed Dial dialing bins are available to DISA callers. This could save the DISA caller time when dialing. To access the System Speed Dialing bins, the caller dials Service Code #2 and the System Speed Dial Bin number.

 Operator Calling

A DISA caller may dial 0 for the system operator.

 Paging

Internal and External Paging is available to DISA callers. This allows co-workers in adjacent facilities, for example, to broadcast announcements to each other.

 Direct Trunk Access

DISA callers may select a specific trunk for outgoing calls through the system. To directly access a trunk, the user dials Service Code #9 followed by the trunk number (e.g., 001). This allows the DISA caller to place an outgoing call over the selected trunk. Direct Trunk Access bypasses the system Trunk Group Routing/ARS/Trunk Access Maps. As with dial 9 access, any toll charge is incurred by the system.

 Call Forward

DISA callers may set Call Forwarding to redirect an extension calls to another extension. Call Forwarding ensures that the user's calls are covered when they are away from their work area.

 DISA/Tie Trunk Barge-In

The DISA/Tie Trunk Barge-In option allows a DISA/Tie Line caller to break into another extension user's established call. This sets up a three-way conversation between the intruding party and the two parties on the initial call.

DISA Toll Restriction

The digits a DISA caller dials for an outgoing call may be subject to the system Toll Restriction. For example, Toll Restriction can prevent users from dialing a 1-900 service. When an incoming DISA caller tries to use system trunks to dial 1-900, Toll Restriction denies the call.

DISA Operating Modes

The DISA Operating Modes determine what happens when a DISA caller forgets to dial, calls a busy or unanswered extension or dials incorrectly. The system can either drop the call or send it to a preset Ring Group (called a the DISA Transfer Destination).

Department Calling with Overflow Message

If a DISA caller dials a busy Department Calling Group, the system can periodically play the voice prompt, *"Please hold on. All lines are busy. Your call will be answered when a line becomes free."* while the caller waits. The time between the voice prompts is the VRS Waiting Message Interval Time. When an extension in the Department Group becomes available, the call automatically goes through. If the Department Calling Group remains busy past the DISA No Answer Time, the DISA call routes to the overflow destination or disconnects. (What happens to the unanswered call is set by the DISA Operating Mode). The Overflow Message requires a VRS.

Warning Tone for Long DISA Calls

You can set up the system to provide a warning tone to DISA callers that have been on a call too long. The warning tone can be just a reminder (which the caller can ignore) or can be followed by a forced disconnect of the call. When the DISA caller hears the warning tone, they have the option of dialing a code to continue the conversation or disconnect.

Trunk Continue/Disconnect Codes

Users have the option to use a Continue or Disconnect service code. The Continue service code extends the conversation for a programmed time. If the user enters the Disconnect service code, the call is immediately disconnected.

EXAMPLE:

The following example indicates how a call is handled with the system programmed as follows:

- Program 14-01-25: **1**
 - Program 20-28-01: **#**
 - Program 20-28-02: **No Setting**
 - Program 20-28-03: **180**
 - Program 24-02-07: **600** (Used only with manually transferred Tandem Trunk calls)
 - Program 24-02-10: **30** (Used only with manually transferred Tandem Trunk calls)
 - Program 25-07-07: **600** (Used only with automatically transferred Tandem Trunk calls or DISA calls)
 - Program 25-07-08: **30** (Used only with automatically transferred Tandem Trunk calls or DISA calls)
1. An external call connects to an external number (either by transferring with Tandem Trunking or by DISA caller).

2. After 10 minutes (Tandem Trunking = Program 24-02-07 or DISA = Program 25-07-07), a warning tone is heard and the user dials # (Program 20-28-01) to extend the conversation.
3. After three minutes (Program 20-28-03), the warning tone is heard again. After 30 seconds (Tandem Trunking = Program 24-02-10 or DISA = Program 25-07-08), the call is disconnected.

Conditions

- The DISA caller must use an analog (DTMF) telephone. DISA is compatible with calling devices that meet the DTMF signaling requirements of EIA Specification RS-464. DISA trunks must be ground start or supervised loop start.
- The Continue/Disconnect code must be DTMF.
- With an analog trunk, the Continue/Disconnect code may work using DTMF sounds from the opposite side trunk. With an ISDN trunk, Program 14-01-25 must be enabled to detect the Continue/Disconnect code.
- The Continue/Disconnect code is not accepted while dialing a trunk.
- Continue/Disconnect codes do not work if all DTMF receivers are busy.
- When used with the Networking feature, both systems must be programmed the same.
- In a system with ARS enabled:
When a DISA caller dials 9 for an outside call (if allowed), the system routes the call via ARS.
- In a system with ARS disabled:
When a DISA caller dials 9 for an outside call (if allowed), the system uses the routes programmed for Trunk Group Routing.
- Transferred calls on DISA, DID, DIL, ISDN trunks, or from the VRS can display the reason a call is being transferred (Call Forward, Busy, No Answer or DND).
- Long conversation cutoff is controlled separately for manually transferred Tandem Trunk calls, automatically transferred Tandem Trunk calls, and DISA calls.
- Tandem Trunking also uses the Continue/Disconnect codes DISA uses.
- Department Calling with Overflow Message requires a DSP daughter board for VRS.
- DISA can be set only to call forward to another extension. Call Forward Off-Premise is not supported.

Default Setting

Disabled

System Availability

Terminals

Remote Analog DTMF (2500 type) telephones

Required Component(s)

DSPII-U10 Unit and VRS Compact Flash (for Announcements)

Related Features

Automatic Route Selection

Central Office Calls, Answering

Direct Inward Dialing (DID)

Direct Inward Line (DIL)

Long Conversation Cutoff

Tandem Trunking (Unsupervised Conference)

Transfer

Voice Response System (VRS)

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**
Reserve at least one circuit for DTMF reception (entry 0 or 2).
Use the following as a guide when allocating DTMF receivers:
 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.

- ➔ **11-01-01 System Numbering**
Set up a Service Code for Alternate Trunk Route Access.

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- **11-09-02 Trunk Access Code – 2nd Trunk Route Access Code**
Assign the Service Code set up in PRG 11-01 for Alternate Trunk Route Access.
 - **14-01-02 Basic Trunk Data Setup – Transmit Level**
 - **14-01-03 Basic Trunk Data Setup – Receive Level**
Customize the CODEC gains (transmit and receive levels 1~63) for DISA trunks.
 - **14-01-13 Basic Trunk Data Setup – Trunk-to-Trunk Transfer**
If DISA caller can place outgoing calls through the system (refer to Program 20-14 in the Electra Elite IPK II Programming Manual), enable loop supervision (1) for the DISA trunk. If DISA caller cannot use the system trunks for outgoing calls, enter 0 to disable loop supervision.
 - **20-01-05 System Options – DTMF Receive Active Time**
After answering the call, the system attaches a DTMF receiver to the DISA trunk for this time (0~64800 seconds).
 - **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
 - **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
Turn Off (0) or On (1) an extension ability to receive a second call from a DID, DISA, DIL, or Tie Line caller.
 - **20-13-23 Class of Service Options (Supplementary Service) – Display the Reason for Transfer**
Enable (1)/Disable (0) an extension user ability to display the reason (Call Forward, Busy, No Answer or DND) a VRS, DID, DISA, or ISDN call is being transferred to their extension.
 - **20-14-01 through 20-14-11 Class of Service Options for DISA/E&M**
Enable (1)/Disable (0) the following options for each DISA Class of Service (1~15).
 - First Digit Absorption (Delete First Digit Dialed) (Item 1)
 - Trunk Group Routing/ARS Access (Item 2)
 - Trunk Group Access (Item 3)
 - Outgoing System Speed Dialing (Item 4)
 - Operator Calling (Item 5)
 - Internal Paging (Item 6)
 - External Paging (Item 7)
 - Direct Trunk Access (Item 8)
 - Forced Trunk Disconnect <Not for ISDN T-point> (Item 9)
 - Call Forward Setting by Remote Via DISA (Item 10)
 - DISA/Tie Trunk Barge-In (Item 11)
 - **21-15-01 Individual Trunk Group Routing for Extensions**
Designate the trunk route accessed when a user dials the Alternate Trunk Route Access Code. Refer to Trunk Group Routing to set up outbound routing.
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- ➔ **22-01-11 System Options for Incoming Calls – VRS Waiting Message Interval Time**
Set up the duration time between announcing the VRS Waiting Message for Auto – Attendant & Queuing. The message is repeatedly sent out during the specified time (0~64800 seconds).
 - ➔ **22-02-01 Incoming Call Trunk Setup**
For DISA operation, set the trunk service type to 2. You can have a different service type for each Night Service mode.
 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign the extensions that should receive the overflow. Set the ringing in Program 22-06.

0 = Trunks Do Not Ring (just flash line keys)
1 = Trunks Ring
 - ➔ **25-01-01 VRS/DISA Line Basic Data Setup – VRS/DISA Dial-In Mode**
Select whether the DISA trunk uses the Extension number/Service code specify (0) or the Dial Conversion Table (1). Default is 0.
 - ➔ **25-01-02 VRS/DISA Line Basic Data Setup – DISA User ID**
Select whether or not the DISA User ID is used.

0 = Off
1 = On (default)
 - ➔ **25-01-03 VRS/DISA Line Basic Data Setup – VRS/DISA Transfer Alarm**
Select whether or not the DISA transfer alarm is used.

0 = Normal (default)
1 = Alarm
 - ➔ **25-02-01 VRS/DISA VRS Message**
Assign the source and VRS message number used as the Automated Attendant Message for each trunk (001~200) which is assigned as a VRS/DISA:

0 = No Message (default)
1 = VRS (VRS Message Number = 01~48)
2 = ACI (ACI Group Number = 01~16)
3 = SLT (Department Group Number = 01~64)
 - ➔ **25-03-01 VRS/DISA Transfer Ring Group With Incorrect Dialing**
Set the operating mode of each DISA trunk. This sets what happens to the call when the DISA caller dials incorrectly. The call can either disconnect (0), transfer to an alternate ring group destination, or transfer to In-Skin/External Voice Mail, or Centralized Voice Mail. Default is 0.

Ring Groups:
31~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
-
-

-
-
- ➔ **25-04-01 VRS/DISA Transfer Ring Group With No Answer/Busy**
Set the operating mode of each DISA trunk. This sets what happens to the call when the DISA caller calls a busy or unanswered extension. The call can either disconnect (0), or transfer to an alternate ring group destination, In-Skin/External Voice Mail, or Centralized Voice Mail. Default is 0.
- Ring Groups:
1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
- ➔ **25-05-01 VRS/DISA Error Message Assignment**
Assign the VRS message number to be used as the Automated Attendant error message. For each VRS/DISA trunk that the VRS answers, enter the VRS message (1~48) the outside caller hears if they dial incorrectly. If you enter 0 (i.e., no error message), the call reroutes according to PRG 25-03 and PRG 25-04. For each trunk, you make a separate entry for each Night Service mode. Default is 0.
- ➔ **25-06-01 VRS/DISA One-Digit Code Attendant Setup – Next Attendant Message Number**
Set up single digit dialing through the VRS. This gives VRS callers single-key access to extensions, the company operator, Department Calling Groups and Voice Mail. For each VRS message set to answer outside calls (see PRG 25-02 and PRG 25-05), you specify:
- The digit the VRS caller dials (0~9, *, #). (Keep in mind that if you assign destinations to digits, outside callers cannot dial system extensions, starting with that digit.
 - The destination reached (eight digits maximum) when the caller dials the specified digit.
- The destination can be an extension, a Department Calling pilot number or the Voice Mail master number. A one-digit code can be assigned for each Automated Attendant message. Default is 0 (No Setting)
- ➔ **25-07-01 System Timers for VRS/DISA – VRS/DISA Dial Tone Time**
After answering the DISA trunk, the system waits this time (0~64800 seconds) for the caller to dial the first digit of the password. If the caller fails to dial within this time, the system drops the call. Default is 10.
- ➔ **25-07-02 System Timers for VRS/DISA – VRS/DISA No Answer Time**
A DISA caller can ring an extension for this time (0~64800 seconds) before the system sets the call as a Ring No Answer. After this time expires, the call follows the programmed Ring No Answer routing. (Refer to Program 25-04 in the Electra Elite IPK II Programming Manual.) Default is 0.
- ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**
Determine the time a DISA caller or any automatically transferred trunk-to-trunk (such as Tandem Trunking) conversation can talk before the Long Conversation tone is heard (0~64800 seconds). Default is 3600.
- ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
This determines the time the system waits before disconnecting a a DISA or any automatically transferred trunk-to-trunk (such as Tandem Trunking) call after the Long Conversation tone is heard (0~64800 seconds). Default is 10.
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-
-
- ➔ **25-07-09 System Timers for VRS/DISA – DISA Internal Paging Time**
This is the maximum time for an Internal Page placed by a DISA caller. If the Page continues longer than this time (0~64800 seconds), the system terminates the DISA call. Default is 30.
 - ➔ **25-07-10 System Timers for VRS/DISA – DISA External Paging Time**
This is the maximum time an External Page is placed by a DISA caller. If the Page continues longer than this time (0~64800 seconds), the system terminates the DISA call. Default is 30.
 - ➔ **25-07-13 System Timers for VRS/DISA – VRS/DISA Busy Tone Interval**
If a DISA caller dials a busy extension (and Program 25-04 = 0), the system plays busy tone for this time (0~64800 seconds) before disconnecting. Default is 5.
 - ➔ **25-08-01 DISA User ID Setup – Password**
For each DISA user, set the 6-digit password. Default is No Setting.
 - ➔ **25-09-01 Class of Service for DISA Users**
Assign a DISA Class of Service for each user. **The DISA Class of Service cannot be 0.** You cannot use the PRG 20-XX-XX and PRG 42-XX-XX programs to assign Class of Service to DISA trunks (15 users, with one password and DISA Class of Service 1~15 for each user). Default is 1.
 - ➔ **25-10-01 Trunk Group Routing for DISA**
Assign the Trunk Group Route chosen when a user places a DISA call into the system and dials 9. Set Trunk Group Routing in Program 14-06. If the system has ARS, dialing 9 accesses ARS. The route chosen is based on the DISA Class of Service, which is determined by the password the caller dials. Default is 1.

Trunk Group Routes: 1~100
 - ➔ **25-11-01 DISA Toll Restriction Class**
If the system uses Toll Restriction, enter a Toll Restriction Class (1~15) for each DISA user (1~15). The system uses the Toll Restriction Class you enter in Program 21-05. The Toll Restriction Class assigned to a DISA call is based on the DISA Class of Service, which is determined by the password the caller dials. You cannot use Program 21-04 to assign Toll Restriction to DISA trunks. Default is 2.
 - ➔ **25-12-01 Alternate Trunk Group Routing for DISA**
Assign the trunk route that DISA Callers access if they dial the Alternate Trunk Route Access Code. Refer to [Central Office Calls, Placing on page 2-195](#) for more information on setting up Alternate Trunk Route Access. Default is 1.

Route Table Number: 0~100 (0 = No Setting)

 *The Continue and Disconnect codes are set in PRG 20-28 and apply to both DISA and Tandem Trunk calls.*

Trunk Continue/Disconnect Codes

- ➔ **14-01-25 Basic Trunk Data Setup – Continued/Discontinued Trunk-to-Trunk Conversation**
When Program 24-02-10 is set to disconnect a trunk after the defined time, determine whether or not a user should be able to use the continue/disconnect code.

0 = Disable
1 = Enable
- ➔ **20-28-01 Trunk to Trunk Conversation – Conversation Continue Code**
When Program 14-01-25 is enabled, determine the 1-digit code the user should dial (0~9, *, #) to extend the conversation for the time defined in PRG 20-28-03. If the Continue and Disconnect codes are programmed the same (e.g., #), the system follows the Continue operation. Using the Continue code before the warning tone is heard has no action.
- ➔ **20-28-02 Trunk to Trunk Conversation – Conversation Disconnect Code**
When Program 14-01-25 is enabled, determine the 1-digit code the user should dial (0~9, *, #) to immediately disconnect their call. Using the Disconnect code before the warning tone is heard disconnects the call.
- ➔ **20-28-03 Trunk to Trunk Conversation – Conversation Continue Time**
When Program 14-01-25 is enabled, determine the time a call is extended (0~64800 seconds) when the user dials the Continue code (defined in PRG 20-28-01).
- ➔ **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
This timer starts when a trunk begins talking with another trunk (for example: Trunk-to-Trunk Transfer/Tandem Trunking). When this time expires, a warning tone is heard. If Program 24-02-10 is set, the conversation disconnects after that time expires. This timer is set again when the external digit time expires. One of the trunks used must be an analog trunk (or leased line). Entries: (0~64800 seconds).

☞ This applies to manually transferred Tandem Trunk and DISA calls.
- ➔ **24-02-10 System Options for Transfer – Disconnect Trunk-to-Trunk**
Determine the time (0~64800 seconds) a conversation continues after the time in Program 24-02-07 expires. If this option is set to 0, the conversation is disconnected immediately. This program has no affect if Program 24-02-07 is set to 0. One of the trunks used must be an analog trunk (or leased line).

☞ This applies to manually transferred Tandem Trunk and DISA calls.
- ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**
Determine the time a DISA caller can talk before the Long Conversation tone is heard (0~64800 seconds). If Program 25-07-08 is set to 0, the call is disconnected after the time expires. This timer is set again when the external digit time expires. Default is 3600.

☞ This applies to automatically transferred Tandem Trunk and DISA calls.
- ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect Time**
Determine the time the system waits (0~64800 seconds) before disconnecting a DISA call after the Long Conversation tone is heard. This program has no affect if Program 25-07-07 is set to 0. Default is 10.

☞ This applies to automatically transferred Tandem Trunk and DISA calls.

Operation

To place a DISA call into the system (from any 2500 type telephone):

1. Dial the telephone number that rings the DISA trunk.
2. Wait for the DISA trunk to automatically answer with a unique dial tone.
3. Dial the 6-digit DISA password (access code).
4. Wait for a second unique dial tone.
5. Dial an extension.

- OR -

Dial **9** for Trunk Group Routing or ARS.

- OR -

Dial Alternate Trunk Route Access Code (if enabled).

- OR -

Dial **704** + a trunk group number (**1~100**) for an outside call.

- OR -

Dial **#9** + a trunk number (**1~200**) for an outside call.

- OR -

Dial **#2** + System Speed Dialing bin number.

- OR -

Dial **0** for the operator.

- OR -

Dial **701** + an Internal Paging Zone number (**0, 1~9, 00, 01~64**).

- OR -

Dial **703** + an External Paging Zone number (**1~8** or **0** for All Call).

- OR -

Dial **710** + a busy extension number to barge in to a call.

To forward an extension calls using a DISA call into the system (from any 2500 type telephone):

1. Dial the telephone number that rings the DISA trunk.
2. Wait for the DISA trunk to automatically answer with a unique dial tone.
3. Dial the 6-digit DISA password (access code).
4. Wait for a second unique dial tone.
5. Dial the Call Forward service code (as defined in PRG 11-11-01 through PRG 11-11-05).

6. Dial the number of the extension to be forwarded.
7. Dial **1** to set Call Forwarding or 0 to cancel Call Forwarding.
8. Dial the extension number where the calls are forwarded.

To use the Continue code to extend a DISA call:

1. An external call connects to an external number (either by transferring with Tandem Trunking or by DISA caller).
2. After the programmed time (Program 25-07-07), a warning tone is heard and the user dials the Continue code (Program 20-28-01) to extend the conversation.
3. After the programmed time (Program 20-28-03), the warning tone is heard again. After the programmed time (Program 25-07-08), the call is disconnected if the Continue code is not dialed again.

Direct Station Selection (DSS) Console

Enhancements


In **Version 1500 or higher**, DSS/One-Touch keys can be used to one-touch transfer without using Transfer.

In **Version 1600 or higher**, when a Ring Group call rings a station, a BLF indication for this station shows idle or busy based on a new Class of Service option (20-13-49). Prior to this release, the BLF Indication showed busy.

Version 2000 or higher:

- ❑ A Direct Station Selection (DSS) Console can have a shift key to expand the console capacity. Two pages are available. In Page 1 mode, keys 01~54 are available for DSS/BLF function, feature access, etc. In Page 2 mode, an additional 54 keys (01~54) are available for DSS/BLF function, feature access, etc. Keys 55~60 do not shift. A total of 113 keys are available when using the Page key ($54 + 54 + 6 - 1 = 113$). The Page key (shift key) must be assigned on key 55~60.

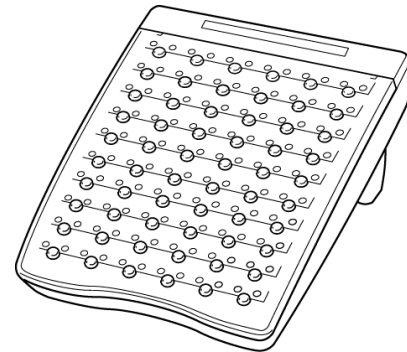
Table 2-17 DSS/BLF Priority Chart

Priority	V1000/V1100	V1500 and higher
Highest  Lowest	VM MSG Waiting	VM MSG Waiting
	DND/Call Forward	Busy
	VM Auto Attendant (Immediate)	DND/Call Forward
	Busy	VM Auto Attendant (Immediate)
	Idle	Idle

Description

The DSS Console gives a multiline terminal user a Busy Lamp Field (BLF) and one-button access to extensions, trunks, and system features. This saves time for users that do a lot of call processing (e.g., attendants, operators, or dispatchers). The DSS Console simplifies:

- Calling extensions and door boxes
- Placing, answering and transferring outside calls
- Making an External or Internal Page
- Switching the Night Service mode
- Activating DSS Console Alternate Answer



The DSS Console also provides DSS Console Alternate Answer. This lets a multiline terminal user with a DSS Console quickly reroute their calls to a co-worker. Transferred and dial 0 calls ring both DSS Consoles and, if the VRS is installed, the main operator hears the message, "Your calls have been forwarded". Central office calls ring both consoles and no message is heard by the operator.

You can also program the DSS Console keys to store Service Codes (up to 29 digits). This provides the DSS Console user with many of the features available on One-Touch and Programmable Feature Keys. The DSS Console keys can optionally store additional associated digits after the Service Code. For example, storing 70401 under a DSS Console key accesses Trunk Group 1 when the console user presses the key.

The maximum number of consoles allowed per system is 32.

DSS Lamp Table Changed to Apply to DSS/Hotline Keys for Multiline Terminals

Using Programs 30-05-02~30-05-21 DSS Console Lamp Table, you can assign LED flash patterns for DSS and Hotline keys on multiline terminals and DSS Consoles.

ACD/Non-ACD Agent DSS Lighting Available

With the Electra Elite IPK II system, Programs 30-05-02~30-05-21 allow a non-ACD DSS console to light indicating the status of both non-ACD agents and ACD agents.

Conditions

- Changing flash patterns for DSS Consoles also changes them for Hotline keys.

- When installing a DSS, the system must auto-detect the console for the LEDs to function correctly. When connecting the DSS to an extension previously defined with another circuit type, undefine the circuit type (enter 00 in Program 10-03-01 for the extension number), then connect the DSS Console.
- Programmable Function Keys for ACD codes (*10, *12, *13, *14, *15, *16, *17, *18, *19) cannot be programmed on a DSS Console.
- Programmable Function keys for Trunk Group (*02), Virtual Extension (*03), and Call Appearance (CAP) Key (*08) cannot be programmed on a DSS Console as the system does not allow entry of the additional data required for these keys.
- A user can use the One-Touch Programmable Function Key (code 01) to have DSS Console keys for Personal Speed Dial and common and group Speed Dial.
- Lighting status for ACD agents and non-ACD agents does not appear on the same console type. For ACD agent's lighting status, a DSS Console must be programmed as an ACD console in Program 30-01-01. For non-ACD agents, the console must be programmed as a business console.
- A DSS key indicates only a Call Forwarding indication for extensions forwarded with Immediate Call Forwarding.
- A DSS Console can have line keys for placing and answering calls.
- The DSS Console provides one-touch calling and a Busy Lamp Field for Door Boxes. Refer to [Door Box on page 2-379](#) when programming Door Boxes.
- The DSS Console provides one-touch Night Service switching. Refer to [Night Service on page 2-745](#) when programming Night Service options.
- Like a One-Touch Key, a user can have DSS Console keys for Direct Station Selection, Trunk Calling, Personal Speed Dial, Speed Dialing, and Service Code access.
- The DSS Console provides one-touch External and Internal Page zone access. Refer to [Paging, External on page 2-769](#) and [Paging, Internal on page 2-775](#).
- You can program the DSS Console keys with service codes to provide the functions of many of the Programmable Function keys. The stored service code can have up to three digits, but it can have additional option codes added (e.g. to set Immediate Call Forward for all calls. Trunk Group (*02), Virtual Extension (*03), and Call Appearance (CAP) Key (*08) codes cannot be programmed on a DSS Console as the system does not allow entry of the additional data required.
- A DSS one-touch button cannot have a pause stored in the dial string.
- With **Version 2000 or higher**, the capacity of a console can be expanded by assigning a Page key (shift key). The Page key (shift key) must be assigned on keys 55~60.
- The expanded capacity for DSS Consoles (two pages) which was added in **Version 2000**, is not supported for DSS Consoles in the ACD Monitor Mode.

- When a multiline terminal user is on a call, they can press a DSS key for that station to transfer to another station. It is not necessary to press Transfer to transfer to another station using a DSS key.
 - ✎ When a multiline terminal user is on a call, they must press Transfer to transfer a call off site with a DSS key.
- Program 30-03-01: DSS Console Key Assignment, Key Number 001~114, Function Number 95: Page Switching is not supported in **Version 1600**.
- Pauses can be entered in the dial string of a DSS/One Touch button. The pause is entered as P in the dial string and causes the system to wait three seconds before sending the rest of the digits that follow the P (pause). Multiple pauses can be entered.
- The @ can be entered in the dial string of a DSS/One Touch button. The @ only applies to ISDN and Intercom calls. When using the @, the system waits for the destination to answer (answer supervision), and then sends the rest of the digits.
- Entering a P (pause) in a DSS/One Touch dial string can be used for CO calls, Intercom calls, or after the @ for ISDN calls.

Default Setting

- No DSS Consoles assigned (in Program 30-02-01).
- All DSS Console key ranges are ports 1~200.
- Once a DSS Console is enabled, the console keys are DSS keys (Program 30-03-01).

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Automatic Call Distribution (ACD)

Call Forwarding

Central Office Calls, Answering**Central Office Calls, Placing****Door Box****Night Service****One-Touch Calling****Paging, External****Paging, Internal****Programmable Function Keys****Speed Dial – System/Group/Station**



Programming

- **10-03-01 ETU Setup – ESI Terminal Type (B1)**
The system automatically assigns the terminal type (10) for the port which has a DSS console installed.
- **15-02-08 Multiline Telephone Basic Data Setup – Automatic Handsfree**
Set to 1 for a DSS Console to have one-touch operation. If set to 0, the user needs to lift the handset before pressing a DSS key for the call to complete.
- **20-06-01 Class of Service for Extensions**
Assign Class of Service (1~15) to extensions.
- **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension user to manually (0) or automatically (1) receive off-hook signals.
- **20-13-49 Class of Service Options (Supplementary Service) – BLF Indication on CO Incoming State (Version 1600 or higher required)**
Turn Off (0) or On (1) the BLF Indication when a Ring Group call rings a station. This assignment does not apply to single line telephones.
- **20-17-01 Operator Extension – Operator’s Extension Number**
Define the extension numbers used as operators.
- **30-01-01 DSS Console Operating Mode**
Set the system DSS Consoles mode. The available options are Business Mode (0), Hotel Mode (1), and ACD Monitor Mode (2). Default is 0.

- ➔ **30-02-01 DSS Console Extension Assignment – Extension Number**
Designate the DSS Console installations (i.e., the extensions that have DSS Consoles connected to them).
- ➔ **30-03-01 DSS Console Key Assignment**
Customize DSS Console keys to function as DSS keys, Service Code keys, Programmable Function Keys, and One-Touch Calling keys. The key [when defined as a DSS/One-Touch key (code 01)] can have any function up to four digits (e.g., extension number or Service Code). The function information (such as extension number or Service Code) would then be entered as the additional data.
- ➔ **30-05-02 through 30-05-21 DSS Console Lamp Table**
If required, customize the system DSS Console flash rates. This table also affects DSS/Hotline key lighting for multiline terminals.

Operation

Calling an extension from your DSS Console:

1. Press the **DSS Console** key.
 -  *If the call voice-announces, you can dial 1 to make it ring.*
 -  *If you do not have Handsfree, you must lift the handset to speak.*

Extension Busy Lamp Field	
When the DSS key is...	The assigned extension is...
On	Busy on a call
Off	Idle
Flashing Fast	In Do Not Disturb


Answering a trunk call from your DSS Console:

1. Press the flashing **DSS Console** key assigned to the trunk.
 -  *If you do not have Handsfree, you must lift the handset to speak.*

Transferring a call using your DSS Console:


1. Place or answer the call.
2. Press **Transfer**.
3. Press the DSS key for the extension that receives the transfer.


- (Optional) Announce the call.

 *If called party does not want the call, press the flashing line key to retrieve it.*

Making an External Page using your DSS Console:

- Press the **DSS Console External Page** zone key (1~8).


 *If the zone you want is busy, try again later.*

 *If you do not have Handsfree, lift the handset to make your announcement.*

External Page Busy Lamp Field	
When the DSS key is...	The External Page zone is...
On	Busy
Off	Idle

Making an Internal Page using your DSS Console:

- Press the **DSS Console Internal Page** zone key (Group key 1~64).

 *If the zone you want is busy, try again later.*

 *If you do not have Handsfree, lift the handset to make your announcement.*



Internal Page Busy Lamp Field	
When the DSS key is...	The Internal Page zone is...
On	Busy
Off	Idle

Switching the Night Service mode from your DSS Console:

- Press the Night Service key.

Night Service Busy Lamp Field	
When this key is ON...	The system is in the...
DAY	Day 1 Mode
NIGHT	Night 1 Mode
BREAK	Break 1 Mode
NIGHT 2	Night 2 Mode

Using a DSS Console key as a One-Touch or Programmable Function Key:

-  A user can have DSS Console keys programmed as One-Touch Keys. These keys can be used for Direct Station Selection, Trunk Calling, Personal Speed Dial, Speed Dialing, and Service Code access. The stored service code cannot be longer than three digits.
1. Press the **DSS Console** key for function.
 -  For example, you can press **DSS** key + 1 + destination to forward your calls. Your DSS key must be programmed for the Call Forward feature.

Directed Call Pickup

Description

Directed Call Pickup permits an extension user to intercept a call ringing another extension. This allows a user to conveniently answer a call for a co-worker from their own telephone. With Directed Call Pickup, an extension user can pick up:

- Trunk calls (i.e., Ring Group calls)
- Direct Inward Lines
- Transferred trunk calls
- Transferred Intercom calls
- Ringing and voice-announced Intercom calls

Conditions

- Calls which were on hold or transferred which recall the extension can be answered using Directed Call Pickup.
- Personal Park also uses the Directed Call Pickup code.
- Voice Mail Park and Page also uses the Directed Call Pickup code.

Default Setting

Enabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Call Arrival (CAR) Keys

Department Calling

Group Call Pickup

Hold

Hotline

Park

Secretary Call Pickup

Secondary Incoming Extension


Transfer

Virtual Extensions

Voice Mail Message Indication on Line Keys


Programming

- 11-12-25 **Service Code Setup (for Service Access) – Direct Call Pickup - Own Group**
(default: 756)
- 11-12-26 **Service Code Setup (for Service Access) – Call Pickup for Specified Group**
(default: 768)
- 11-12-27 **Service Code Setup (for Service Access) – Call Pickup**
(default: * #)
- 11-12-28 **Service Code Setup (for Service Access) – Call Pickup for Another Group**
(default: 769)
- 11-12-29 **Service Code Setup (for Service Access) – Direct Extension Call Pickup**
(default: * *)
- 11-12-30 **Service Code Setup (for Service Access) – Specified Trunk Answer**
(default: 672)
- 20-06-01 **Class of Service for Extensions**
Assign a Class of Service to extensions (1~15).

-  **20-10-05 Class of Service Options (Answer Service) – Directed Call Pickup for Own Group**
Enable (1)/Disable (0) an extension user ability to use Directed Call Pickup for calls ringing their own group.

Operation

To use Directed Call Pickup to intercept a call to a co-worker's extension:

1. Pick up the handset or press **Speaker**.
2. Dial * *.
3. Dial number of extension whose call you want to intercept.
 -  *If more than one call is coming in, the system sets the priority for which call it answers first.*

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Directory Dialing

Enhancements

With **Version 1500 or higher** telephones that are not connected or uninstalled do not show up in the Extension Directory.

With **Version 1600 or higher**, extensions (including Virtual Extensions) can be removed from the Extension Directory list based on a new Class of Service option (PRG 20-13-51). Prior to this release, installed extensions and Virtual Extensions with a Name Assigned always displayed in the Extension Directory list.

Description

Directory Dialing allows a Multiline Terminal user to select a co-worker or outside caller from a list of names, rather than dialing the telephone number. There are four types of Directory Dialing:

- SYS → Company (Common) System Speed Dials
- SPDg → Department (Group) Speed Dials
- STA → Personal Speed Dials
- EXT → Co-worker's Extensions

Conditions

- Directory Dialing sorts and searches directory names in alphabetical order (based on all characters of the name entered) when the system starts up or reboots. The system resorts extension names when:
 - You change Program 15-01-01 (Extension Numbers and Names).
 - Any user dials 700 and changes their extension name.
- Directory Dialing follows all the programmed options and conditions for Speed Dial - System/Group/Station, Intercom Calling and One-Touch Calling.
- Extension Directory only shows a telephone/VE that is connected and has a name assigned in PRG 15-01-01. (**Version 1500 or higher is required.**)

Default Setting

Enabled

System Availability

Terminals

All Display Multiline Terminals with Softkeys

Required Component(s)

None

Related Features

Last Number Redial

Name Storing

Speed Dial – System/Group/Station

Softkeys

Programming

- ➔ **20-13-51 Class of Service Options (Supplementary Service) – Number and Name Appear in the Directory (Version 1600 or higher.)**
Turn Off (0) or On (1) the extension to be displayed in the Extension Directory list (including Virtual Extensions).






- ➔ **21-01-02 System Options for Outgoing Calls – Intercom Interdigit Time**
If a user waits longer than this time (0~64800 seconds) between Directory Dialing steps, Directory Dialing automatically cancels. Default is 10.

Operation

To use Directory Dialing from a multiline terminal with an LCD:

1. Press the **Dir** softkey.

2. Press the **softkey** for the Directory Dialing type:
 - SYS → Company (Common) System Speed Dials
 - SPDg → Department (Group) Speed Dials
 - STA → Personal Speed Dials
 - EXT → Co-worker's Extensions

 *Directory Dialing follows any feature restrictions that your system may have enabled. For example, if your extension cannot normally use Speed Dial – System/Group/Station, Directory Dialing can not access it either.*
3. Dial letter/number range for the party you want to call (e.g., dial 2 for A, B, C or 2).
 -  *You can enter several letters to help narrow the search.*
 -  *Press # to enter additional letters on the same key (ex: TOM = 8666#6).*
4. Press the **Down Arrow** softkey to jump to that section.
5. Press the Volume **▲** or **▼** key to scroll through the list.
 -  *If you wait too long between your selections, Directory Dialing automatically cancels.*
6. Lift the handset or press the **DIAL** softkey, or press **Speaker** to place the selected call.
 -  *If you selected an outside call, it routes according to your system Trunk Group Routing/ARS setup.*

To cancel Directory Dialing:

Press the **Exit** key.

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Distinctive Ringing, Tones and Flash Patterns

Enhancements

Version 1600 or higher:

- When a Ring Group call rings a station, a BLF indication for this station shows idle or busy based on a new Class of Service option (PRG 20-13-49). Prior to this release, the BLF indication would show busy.
- Allows the flash rate and color of the Message Waiting LED to be configured for the following conditions:
 - Message Waiting Lamp Cycle for Calling Extension (PRG 15-02-35)
 - Message Waiting Lamp Cycle for Called Extension (PRG 15-02-36)
 - Voice Mail Message Wait Lamp Color (PRG 15-02-37)
 - Voice Mail Message Wait Lamp Cycle (PRG 15-02-38)

This provides more distinction between incoming calls, Message Waiting (Set/Received) and a VM Message Waiting indication.

Description

Distinctive Ringing, Tones and Flash Patterns provide extension users with audible and visual call status signals. This lets users tell the types of calls by listening to the ringing/tones and watching the keys. It also helps users monitor the progress of their calls. In addition, Distinctive Ringing lets multiline terminal users customize their Intercom and trunk call ringing. This is helpful for users that work together closely. For example, if several co-workers set their multiline terminals to ring at different pitches, each co-worker can always tell which call is for them. You can also customize the tones the system uses for splash tone, confirmation tone, trunk ring tone, Intercom ring tone and Alarm ring tone. Refer to the chart below and the Electra Elite IPK II Programming Manual for more details.

Table 2-18 Distinctive Ringing: Tones and Flash Patterns

Program	Description
80-01-01~04 Service Tone Setup	Set the frequency of the system splash tone. This is the tone the system uses, for example, to alert the user of an incoming voice-announced Intercom call.
30-05-02~21 DSS Console Lamp Table	Set the DSS and Hotline key flash rates for busy, idle, DND, ACD Agent status, and hotel options.

Conditions

- Single line telephone users cannot listen to or hear the pitch of their telephone incoming ring.

- DTU/DTP Style telephones only follow high, mid range and low ring tone settings. They do not follow Melodies.
- If PRG 22-03-01 is set to 0~3 and PRG 15-02-02 is set to 1~3, trunk calls follow the ring pattern in PRG 22-03-01 and the pitch in PRG 15-02-02.
- If PRG 22-03-01 is set to 4~8 and PRG 15-02-02 is set to 1~3, trunk calls follow the ring pattern in PRG 22-03-01.
- If PRG 22-03-01 is set to 0~8 and PRG 15-02-02 is set to 4~8, trunk calls follow the ring pattern in PRG 15-02-02.
- The following voice mail features require system tones be changed in PRG 80-01-02 to work. Refer to the Programming section of the [Digital Voice Mail](#) feature for details.
 - Call Holding
 - Busy Greeting
 - Call Screening
 - Await Answer Transfer
- When a ring group call rings a Single Line Station, the BLF indication shows busy. (**Version 2000 or higher required.**)
- The priority of the Large LED is as follows:
 1. CO Call Ringing
 2. Message Waiting Received
 3. VM Message Waiting
 4. Message Waiting Set

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Call Arrival (CAR) Keys

Single Line Telephones, Analog 500/2500 Sets

Programming

- ➔ **15-02-02 Multiline Telephone Basic Data Setup – Trunk Ring Tone**
From the range specified in Program 22-03-01, select the multiline terminal extension trunk ring tone.
- ➔ **15-02-03 Multiline Telephone Basic Data Setup – Extension Ring Tone**
Select the extension intercom ring tone.
- ➔ **15-02-35 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Calling Extension (Version 1600 or higher required)**
 - 1 = Cycle 1
 - 2 = Cycle 2
 - 3 = Cycle 3 (default)
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7
- ➔ **15-02-36 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Called Extension (Version 1600 or higher required)**
 - 1 = Cycle 1
 - 2 = Cycle 2 (default)
 - 3 = Cycle 3
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7
- ➔ **15-02-37 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Color (Version 1600 or higher required)**
 - 0 = Green
 - 1 = Red (default)

- ➔ **15-02-38 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Cycle (Version 1600 or higher required)**
 1 = Cycle 1
 2 = Cycle 2 (default)
 3 = Cycle 3
 4 = Cycle 4
 5 = Cycle 5
 6 = Cycle 6
 7 = Cycle 7

- ➔ **20-13-49 Class of Service Options (Supplementary Service) – BLF Indication on CO Incoming State (Version 1600 or higher required)**
 Turn Off (0) or On (1) the BLF Indication when a Ring Group call rings a station. This assignment does not apply to single line telephones.

- ➔ **20-15-01 through 20-15-11 Ring Cycle Setup**
 Define the ringing cycles for each ring tone, when Synchronous Ringing is disabled.

- ➔ **22-03-01 Trunk Ring Tone Range**
 Set the ring tone range (1~9) for each trunk.

- ➔ **80-01-01 Service Tone Setup – Repeat Count**
 Customize the system basic tones and system service tones. You must reset the system for the changes to take affect.

- ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
 The following features require that the system tones listed below be changed to match the table. After changing these settings the KSU must be reset for the changes to take effect.
 - Call Screening
 - Call Holding
 - Busy Greeting
 - Await Answer Transfer

Tone 06			
Unit	Basic Tone	Duration	Gain Level
1	11 ~ 480/620Hz -13/-13dB	300ms	32
2	0 - No Tone	300ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	

Tone 06			
8	0 - No Tone	0ms	

Tone 14			
Unit	Basic Tone	Duration	Gain Level
1	10 ~ 440/480Hz -13/-13dB	1000ms	32
2	0 - No Tone	2100ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

Tone 39			
Unit	Basic Tone	Duration	Gain Level
1	12 ~ 440/620Hz -16dB	500ms	32
2	0 - No Tone	500ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

Operation

To listen to the incoming ring choices:

1. Press **Speaker**.
2. Dial **711**.

3. Dial **1** to check ringing for intercom calls.
- OR -
Dial **2** to check ringing for trunk calls.
4. For Intercom calls, select the pitch you want to check (1~8).
- OR -
For trunk calls, select the pitch (1~8) and the tone (1~4) you want to check.
5. Go back to step 4 to listen to additional choices or press **Speaker** to hang up.

To change the pitch of your incoming ring (multiline terminal only):

1. Press **Speaker**.
2. Dial **720**.
3. Dial **1** to change ringing for Intercom calls.
- OR -
Dial **2** to change ringing for trunk calls.
4. Select the pitch (1~8).
5. Press **Speaker** to hang up.

Do Not Disturb

Enhancements

Version 1500 or higher supports:

- Call Forwarding when Do Not Disturb (DND) is set.
- DSS/BLF lights solid instead of flashing when Do Not Disturb (DND) is set.
- DSS/BLF displays when a telephone is off-hook (Busy) when Do Not Disturb (DND) is set.

Description

Do Not Disturb blocks incoming calls and Paging announcements. DND permits an extension user to work by the telephone undisturbed by incoming calls and announcements. The user can activate DND while their telephone is idle or they are on a call. Once activated, incoming trunk calls still flash the line keys. The user may use the telephone in the normal manner for placing and processing calls.

Five Do Not Disturb options are available at each extension:

- 1 = Incoming trunk calls blocked.
- 2 = Paging, incoming Intercom, Call Forwards and transferred trunk calls blocked.
- 3 = All calls blocked.
- 4 = Incoming Call Forwards blocked.
- 0 = Do Not Disturb canceled.

Conditions

- If there is no Call Forwarding key (Program 15-07: 10~17), the DND key blinks when the extension is forwarded.
- Call Arrival (CAR) Key/Virtual Extension (VE) do not support DND Programmable Function keys.
- Multiline Terminal users can activate or deactivate Do Not Disturb while on a call. This option is not available for single line telephones.
- When DND and Call Forward are set on the same telephone, call forwarding does not work. (**Version 1100 or lower.**)

- When DND and Call Forward are set on the same telephone, call forwarding works. If Busy and No Answer Forwarding are set to different locations, it follows the Busy forwarding. **(Version 1500 or higher required.)**
- If an extension already receiving forwarded calls activates DND option 4, callers to the forwarded extension hear DND tone.
- If an extension activates DND option 4, other extensions can still forward calls to it, but the callers hear DND tone.
- An extension user can override Call Forwarding or Do Not Disturb at another extension using any of the following methods:
 1. Program 11-12-01 Service Code Setup (for Service Access) – Bypass Call (default: 707)
 2. Program 11-16-06 Single Digit Service Code Setup – DND/Call Forward Override Bypass (default: No Setting)
 3. OVRD Softkey
- When a call is transferred because of Call Forwarding No Answer, Call Forwarding Busy, or DND, the Reason for Transfer option can display to the transferred extension while the call is ringing to the user telephone.
- DND modes 1~3 causes calls to follow PRG 22-08 programming, then PRG 22-05 programming even if the extension is forwarded.
- When Selectable Display Messaging is set as DND All, all other DND modes are canceled when Selectable Display Messaging is canceled.
- When DND and any Call Forwarding is set, the call forwards immediately. **(Version 1500 or higher required.)**

Default Settings

Enabled for all extensions.

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Call Forwarding

Call Forwarding/Do Not Disturb Override

Central Office Calls, Answering

Direct Inward Line (DIL)

Distinctive Ringing, Tones and Flash Patterns

Selectable Display Messaging

Programming

- ➔ **11-11-08 Service Code Setup (for Setup/Entry Operation) – Do Not Disturb**
Assign Service Code for DND (Default 747).
- ➔ **11-12-01 Service Code Setup (for Service Access) – Bypass Call**
Assign Service Code for DND (Default 707).
- ➔ **11-16-06 Single Digit Service Code Setup – DND/Call Forward Override Bypass**
If a single digit service code is to be used, assign an available code number.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-04 Class of Service Options (Supplementary Service) – Call Forward/DND Override (Bypass Call)**
Turn Off or On an extension user ability to use Call Forwarding/DND Override.

0 = Off
1 = On
- ➔ **20-13-40 Class of Service Options (Supplementary Service) – Do Not Disturb**
Turn Off or On and extension user ability to set or cancel Do Not Disturb.

0 = Off
1 = On

Operation

To activate or deactivate Do Not Disturb while your extension is idle:

Multiline Terminal

1. Do not lift the handset.
2. Press the **DND** key.
- OR -
Press **Speaker**, and dial **747**.
3. Dial the DND option code.
0 = Cancel DND
1 = Incoming Trunk Calls Blocked
2 = Paging, incoming Intercom, Call Forwards and Transferred Trunk Calls Blocked
3 = All Calls Blocked
4 = Call Forwards Blocked

Single Line Telephone

1. Lift the handset.
2. Dial **747**.
3. Dial the DND option code.
0 = Cancel DND
1 = Incoming Trunk Calls Blocked
2 = Paging, Incoming Intercom, Call Forwards and Transferred Trunk Calls Blocked
3 = All Calls Blocked
4 = Call Forwards Blocked

Door Box

Description

The Door Box is a self-contained Intercom unit typically used to monitor an entrance door. A visitor at the door can press the Door Box call button (like a door bell). The Door Box then sends chime tones to all extensions programmed to receive chimes. To answer the chime, the called extension user lifts the handset. This lets the extension user talk to the visitor at the Door Box. The Door Box is convenient to have at a delivery entrance, for example. It is not necessary to have company personnel monitor the delivery entrance; they answer the Door Box chimes instead. Any number of system extensions can receive Door Box chime tones.

Each Door Box has a pair of normally open relay contacts that can be connected to an electric door strike. Use these contacts to remotely control the entrance door. After answering the Door Box chimes, a multiline terminal user can press the Recall key to activate the Door Box contacts. This in turn releases the electric strike on the entrance door. The device connected to the Door Box contacts cannot exceed the contact ratings shown in the following table:

Door Box Specifications	
Contact Configuration	Normally Open
Maximum Load	60mA @30 VDC
	10mA @90 VDC
Maximum Initial Contact Resistance	50m Ohms

The system can have up to eight Door Boxes. Six chime tones are available.

Conditions

- The Door Box Feature Requires a PGD(2)-U10 ADP. A maximum of 56 PGD(2)-U10 ADPs can be installed in an IPK II system. Refer to the Electra Elite IPK II System Hardware Manual for more information.
- If a PGD(2)-U10 ADP circuit has a Door Box (doorphone) connected, you cannot use that circuit for External Paging.
- Door Boxes can ring multiline, single line, or wireless telephones.
- External Call forward by Doorphone can forward Doorphone calls Off-Premise while a user is away. This feature only works for ISDN lines.
- Off-hook signaling is available for Door Boxes. If an extension user is on the telephone, the Large LED flashes indicating the Door Box ringing, and the display shows a call from the door box.

- Each channel in the PGD(2)-U10 ADP has a jumper which must be set for Door Box operation. Refer to the Electra Elite IPK II System Hardware Manual for additional details.
- The PGD(2)-U10 ADP can be connected only to an ESIB(8)-U20 ETU. If the PGD(2)-U10 ADP is connected to an ESIE(8)-U20 ETU, the PGD(2)-U10 ADP **is not** recognized.
- When a ESIB(8)-U20 ETU has an ESIE(8)-U20 ETU installed [ESIB/E(16)], the PGD(2)-U10 ADP has both channels available for Paging, MOH, External Ringers and Door Box according to strapping on ports 1~8. The corresponding port on the ESIE(8)-U20 ETU does not support a station or other hardware (it is used by the PGD(2)-U10 ADP for channel 2).
- When a ESIB(8)-U20 ETU is installed without an ESIE(8)-U20 ETU installed, a PGD(2)-U10 ADP installed on ports 1~8 is recognized, but only channel 1 on the PGD(2)-U10 ADP is supported. Channel 2 is not available.
- ESIB(8)-U10 ETUs and ESIE(8)-U10 ETUs do not support the PGD(2)-U10 ADP.
- A Single Line Telephone (SLT), connected to an APR does not ring when the Door Phone rings the multiline telephone.
- *D^{term}* cordless telephones are not supported with the Door Box feature.
- Door Boxes do not ring Logged In ACD Agents.

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

PGD(2)-U10 ADP

Programming

- ➔ **10-03-01 ETU Setup (For ESI PKG) – Terminal Type (B1)**
- 10-03-06 ETU Setup (For ESI PKG) – Terminal Type (B2)**
For ESI package support, set the terminal type to 8 [PGD (for Door Box)]. First set 10-03-01 to 0 with no device plugged into that port, then plug the device in and the system should recognize it as a door box, and set PRG 10-03-06.

-
-
- ➔ **10-05-01 General Purpose Relay Setup**
Define which Relay circuits (5~8) on a PGD(2)-U10 ADP Adapter are used for General Purpose Relay.
 - ➔ **11-12-36 Service Code Access (for Service Access) – Door Box Access**
If the service code for Doorphone Access (702 by default) is not acceptable, change it here.
 - ➔ **15-07-01 Programmable Function Keys**
Assign a function key for External Call Forward by Doorphone (Code 54).
 - ➔ **32-01-01 Door Box Timers – Door Box Answer Time**
Set the time (0~64800 seconds) a user has to answer the Door Box chimes. Default is 30.
 - ➔ **32-01-02 Door Box Timers – Door Lock Cancel Time**
Set the time (0~64800 seconds) the Door Box strike stays open when the single line telephone user hookflashes or a multiline terminal user presses the Recall key. Default is 10.
 - ➔ **32-02-01 Door Box Ring Assignments**
Determine which Door Boxes should ring which extensions by entering the extension number. Each Door Box can be programmed to ring up to 32 extensions and an extension can be programmed to ring for multiple Door Boxes.

Door Boxes: 1~8
 - ➔ **32-03-01 Door Box Basic Setup – Chime Pattern**
Set the chime pattern (0~6) for each Door Box.
 - ➔ **32-03-02 Door Box Basic Setup – CODEC Transmit Gain Setup**
 - ➔ **32-03-03 Door Box Basic Setup – CODEC Receive Gain Setup**
Select the CODEC gain type (1~63) for each Door Box.

Door Boxes: 1~8
Gain Type: 1~63 (default: 32)

Related Features

Call Forwarding, Off-Premise

Paging, External

Single Line Telephones, Analog 500/2500 Sets

Wireless – DECT

Operation

To call a Door Box:

Multiline Terminal

1. Press **Speaker**.
2. Dial **702**.
3. Dial Door Box Number (**1~8**).

Single Line Telephone

1. Lift the handset.
2. Dial **702**.
3. Dial Door Box Number (**1~8**).

To activate the Door Box strike:

Multiline Terminal

1. While talking to the Door Box, press the Recall key.

Single Line 500/2500 Telephone

1. While talking to the Door Box, hookflash.

To answer a Door Box chime:


1. Lift the handset or press **Speaker**.

To Answer a Door Box call while busy on another call:

Multiline Terminal

If you are busy on a call, the display shows the incoming Door Box call and the large LED flashes.

1. Press **Hold** to place your active call on hold.
2. When you hear dial tone, dial the door box access code (**702** by default) plus the door box number (**1~8**) to answer the Door Box call.

 *To retrieve the original call, hang up with the door box, and press Conf.*

Single Line Telephone

If you are busy on a call, an off-hook signal is heard indicating the incoming Door Box call.

1. Press the **Flash** key or hookflash to place your active call on hold.
2. Dial the door box access code (**702** by default) plus the door box number (**1~8**) to answer the Door Box call.

 *To retrieve the original call, hang up. The original call rings the single line telephone.*

To activate Call Forwarding Off-Premise for a Door Box:

 *This option only works for ISDN PRI or BRI Trunks.*

1. At the multiline terminal, press **Speaker** + dial SC 722.

- OR -

At the multiline terminal only, press the External Forward by Doorphone key (PRG 15-07-01 or SC 751, code 54).

- OR -

At the single line telephone, lift the handset + dial 722.

2. Dial the Door Box number (**1~4**).
3. Dial the Speed Dialing number where the calls should be forwarded.
4. Press **Speaker** (or hang up at the single line telephone) to hang up.

To cancel Call Forwarding Off-Premise for a Door Box:

1. At the multiline terminal, press **Speaker** + dial SC 722.

- OR -

At the multiline terminal only, press External Forward by Doorphone key (PRG 15-07-01 or SC 751, code 54).

- OR -

At the single line telephone, lift the handset + dial 722.

2. Dial **0** for Cancel.

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Drop Key

Description

The Drop Key abandons a call while retaining the PBX/Centrex line to originate another call. The Drop Key is provided by programming a Programmable Function Key. This feature allows the Recall key to be used to provide a hookflash to the PBX or Central Office. A single line telephone user can use the Drop Key function by an access code.

Conditions

- The Drop key provides a timed disconnect signal on CO/PBX lines.
- The Drop key cannot be used for internal, DID, or Tie line calls.

Default Setting

None

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Flash

PBX Compatibility

Programming

- ➔ **11-12-42 Service Code Setup (for Service Access) – Flash on Trunk lines**
Default is #3.
- ➔ **11-12-59 Service Code Setup (for Service Access) – Trunk Drop Operation for SLT**
Default is Not Assigned.
- ➔ **14-02-03 Analog Trunk Data Setup – Flash Type**
Select the flash type.

0 = Open Loop Flash
1 = Ground

Always set this option to 0.
- ➔ **14-02-04 Analog Trunk Data Setup – Hooking Type**
For each trunk, indicate if Flash is for one of the following:

0 = Timed Flash
1 = Disconnect
- ➔ **15-02-05 Multiline Telephone Basic Data Setup – Transfer Key Operation Mode**
If the Conf key should access Flash, enter 2. Otherwise, enter 0 or 1.
- ➔ **15-03-04 Single Line Telephone Basic Data Setup – Flashing**
In a single line (500/2500 type) telephone Class of Service, Enable (1)/Disable (0) an extension user ability to hookflash for system feature access.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Drop Key (code 84) if required.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **81-10-07 COI Initial Data Setup – Hookflash Time Selection 1**
Set the Flash duration (20 ms~5.0 sec) for analog trunk [COI()-U() ETU] circuits.
- ➔ **81-10-08 COI Initial Data Setup – Hookflash Time Selection 2**
Set the open loop disconnect duration (20 ms~5.0 sec) for analog trunk [COI()-U() ETU] circuits.


Operation

To use the Drop key from a Multiline Terminal with a CO/PBX call in progress:

1. Press the **Function** key programmed as a Drop key.

2. Receive the new CO/PBX dial tone.
3. Dial the desired number.

To use the Feature Key plus Recall key from a Multiline Terminal with a CO/PBX call in progress:

1. Press **Feature**.
2. Press **Recall**.
 *Receive the new CO/PBX dial tone.*
3. Dial the desired number.

To use the Drop key feature from a single line telephone with a CO/PBX call in progress:

1. Hookflash.
2. Receive internal dial tone.
3. Dial the Service Code (PRG 11-12-59, Default Not Assigned).
4. Receive the new CO/PBX dial tone.
5. Dial the desired number.

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D^{term} Cordless II Terminal

Description


The NEC *D^{term}* Cordless II Terminal may be used with the Electra Elite IPK II KTS. The DTR-4R-1 TEL uses 900 MHz Digital Spread Spectrum (DSS) Technology and must be connected in tandem to a multiline terminal.

Press the applicable key on the Base Unit to Switch between cordless operation and multiline terminal operation.

Feature	<i>D^{term}</i> Cordless II (DTR-4R-1)
Digital Technology	900 MHz Spread Spectrum
LCD	2-line, 16-digit LCD Display
Silent Alarm	Yes
Dedicated Keys	TALK, TRANSFER, HOLD, CONF, CHAN, REDIAL, MUTE, R/VOL
Programmable Line Keys	4
Operational Range *	50~350 feet
Message Waiting Indication	☒ Icon
Headset Connection	Yes
Channels	10

* Determined by environmental conditions

Conditions

- When a message is received the ☒ icon is displayed.
- When the RING OFF/ON switch on the right side is down the 🔔 OFF icon is displayed.
 - ☐ When the battery is low, the  icon is displayed.
- When there is no transmission between the Base Unit and the handset for about five minutes, there are no LCD or LED indications on the cordless terminal handset. Ringing off-hook or pressing keys resumes LCD and LED indications.
- The ring pattern for the cordless terminal can be selected by system programming and multiline terminals.
- When a cordless terminal is ringing, press the flashing function key programmed for DSS to answer the call.

- Synchronous Ringing does not apply to the cordless terminals.
- A beep indicates when the cordless terminal receives off-hook ringing.
- Depending on your environment, the maximum number of cordless devices used without interference varies.
- Multiple Base Units and handset units should not be closer than 20 feet anytime.
- Radio interference causes interruptions in conversation. When this happens, your unit is not defective. When noise continues, move to a different location while you talk. (You might even need to move the base unit.) When the situation persists, contact NEC Unified Solutions, Inc., National Technical Assistance Center (NTAC).
- Environments with many metal parts, metal shelves, or metal buildings have been found to reduce telephone performance.
- Dterm cordless phones are not supported with the Door Box feature.

Default Setting

None

System Availability

Terminals

DTR-4R-1 TEL

Required Component(s)

None

Related Features

None


Programming


None


Operation

To program the *D^{term}* Cordless II:


1. Press and hold down * and #, then press **TALK**. The F1 LED flashes red.
2. Press **R/VOL** repeatedly to scroll through the line key (LK) and feature options for function key *F1*.
3. Press **MUTE** to select the displayed line key or feature.
4. When a Line Key is assigned, press MUTE once to enter the Off-Hook Ringing ON or OFF Mode. Press R/VOL to toggle between /TALK for ON or /NO TALK for OFF.
5. Press **MUTE** to advance to the next function key (*F2 ~ F4*).
6. After programming *F4*, press **MUTE** to advance to Global Off-Hook Ringing Assignment.
7. Press **R/VOL** to turn Global Off-Hook Ringing ON/OFF (LCD indicates ON or OFF).
8. Press **TALK** to exit.

 *Function keys F1 ~ F4 can be programmed as Line Keys 1~16, Redial (LNR/SPD), Answer (ANS), Feature (FNC), or Recall. When assigned, these keys operate the same as on an NEC Multiline Terminal.*


 *When initially installed, function keys F1~F4 default to Line Keys 1~4 respectively and Off-Hook Ringing defaults to ON.*

 *Global Off-Hook Ringing must be ON (default) for any Function key to work with Off-Hook Ringing.*



To place an internal call:

1. Press **TALK**.  icon is displayed.
2. Dial the Station Number.
- OR -
Press (**F1~F4**), programmed for Direct Station Selection (DSS).
3. Announce the call after tone burst or wait for call to be answered.

To place an outside call:


1. Press **TALK**.  icon is displayed.
2. Dial the Station Number.
- OR -
Press (**F1~F4**) programmed for Direct Station Selection (DSS).
3. Announce the call after the tone burst or wait for call to be answered.

To answer an Incoming Ringing Call:

1. When the handset is in the charger, lift it.  icon is displayed.
2. When the handset is out of the Base Unit, press **TALK** if ringing line preference is assigned.  icon is displayed.
3. Talk.

To place a call on Hold (internal or outside):

With a call in process, press **HOLD**.

 *To retrieve a held call, press flashing F1~F4.*

 *After a programmed time, the held call recalls to the originating terminal.*

To redial a number:

1. Press **REDIAL**.
2. The previously dialed number is called.

To transfer a call:

1. With a call in process, press **TRANSFER**.
2. Dial Station Number.
- OR -
Press (**F1~F4**) programmed for Direct Station Selection (DSS).
3. Announce call (optional).
4. Press **TALK** to complete transfer.

To place a conference call:

1. With a call in progress, press **CONF**.
2. Place second internal or external call.
3. Announce conference.
4. Press **CONF** again. A 3-party conference is established. When any party hangs up, the conference still includes the remaining parties.

To adjust ring volume:

While the telephone is not being used, press **R/VOL** (upper key on left side) repeatedly to select desired setting display:

- Ring Type A High
- Ring Type A Low
- Ring Type B High
- Ring Type B Low
- Ring Type C High
- Ring Type C Low
- Ring Off (Vibration)

To adjust receive volume level:


1. With a call in progress, press **R/VOL** to decrease volume.
2. Press **R/VOL** to restore normal volume.

To Mute the microphone:

1. Press and hold **MUTE** (lower key on left side).
2. Release **MUTE** to activate microphone.

To use the Charging Unit:





1. Place handset and/or spare battery in the charging slots.
2. The Charge 1 LED is on red during and after charging the handset.
3. The Charge 2 LED is on red while charging the spare battery and turns off after charging is complete.

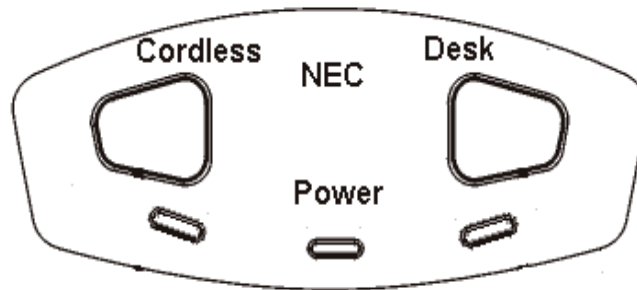
 *If the handset is placed in the charger without an installed battery, the Charge 1 LED flashes.*

To Switch from Cordless to Desk Using Base Unit:

1. When Cordless II is associated with a terminal, use the Base Unit to switch between Cordless II and the Multiline Terminal. Refer to Cordless II Base Unit Controls illustration on the next page.
2. Either press Cordless to select D^{term} Cordless II.


3. Press **Desk** to select multiline terminal.

-  *Switching must be accomplished while both terminals are idle.*
-  *A call in progress cannot be switched. When tried, it is dropped.*
-  *Switching held calls is not recommended because LED indications are not provided.*
-  *The Power LED and the applicable position LED are on to indicate selection.*



Cordless II Base Unit Controls

To switch from Desk to Cordless using the Redial Key:

1. Verify that the Base Unit is in Desk mode.
2. Remove the battery.
3. Press and hold * and # on the handset.
4. Install the battery while holding these keys.
5. After the handset beeps one time, release the keys.
6. The handset is in Setup mode.
7. Press **HOLD** to display the current mode (Digital/Redial).
8. Press **F2** and then **HOLD**.
9. The handset displays Digital (F8), and changes **REDIAL** program to Desk/Cordless Softkey Switching mode.
10. Press **TALK** to restore standby mode.
11. Press **REDIAL** to test this feature.
12. The Base Unit automatically changes from Desk to Cordless.
 -  *This is a one time operation, but one of the F1~F4 keys can be programmed for the redial function.*

13. To restore the normal redial function, enter setup mode, and press **F3** and **HOLD** to display Digital (Redial).
14. Press **TALK** to restore standby mode.

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D^{term} Cordless Lite II Terminal

Description

The NEC *D^{term}* Cordless II Lite Terminal may be used with the Electra Elite IPK II KTS. The DTH-4R-1 TEL uses 900 MHz FM with ADPCM (digital) Technology and is connected in tandem to a multiline terminal.

Press the applicable key on the Base Unit to Switch between Cordless operation and multiline terminal operation.

Feature	<i>D^{term}</i> Cordless Lite II (DTH-4R-1)
Digital Technology	900 MHz FM with ADPCM (digital)
LCD	2-line, 16-digit LCD Display
Silent Alarm	Yes
Dedicated Keys	TALK, TRANSFER, HOLD, CONF, CHAN, REDIAL, MUTE, R/VOL
Programmable Line Keys	4
Operational Range *	50~150 feet
Message Waiting Indication	Yes (Icon)
Headset Connection	Yes
Channels	30

* Determined by environmental conditions. These are cordless RF devices and, therefore, some interference may take place when operating in the same environment as other wireless devices which operate within the same frequency spectrum.

Conditions

- The *D^{term}* Cordless Lite II can be used in conjunction with the Electra Elite, Electra Elite IPK II, and *D^{term}* Series E and Series i Digital Multiline Telephones.
- Battery Capacity is 700 mAh, 3.6V with a Talk Mode of six hours (typical) and a Standby Mode of five days (typical).
- The battery can be hot swapped while on a call. The battery must be replaced with another charged battery pack within 20 seconds, otherwise the connection is lost.
- The handset has visual and audible indicators to warn of a low battery condition.
- When a message is received, the message icon is displayed.

- Synchronous Ringing does not apply to the cordless terminals.
- A beep indicates when the cordless terminal receives off-hook ringing.
- A spare battery is available as an Optional Available Part. A second battery is not shipped with the product.
- The battery can be charged only when it is installed in the handset and the handset is in the charger. A stand-alone battery charger is not available.
- Environments with many metal parts, metal shelves, or metal buildings are known to reduce telephone performance.
- When multiple cordless telephones are used in your office, they must operate on different channels and be at least 20 feet apart (including the base unit and the telephones).
- Dterm cordless phones are not supported with the Door Box feature.

Default Setting

None

System Availability

Terminals

DTH-4R-1 TEL

Required Component(s)

ESI(8)-U() ETU

-OR-

ESIB(8)-U() ETU

-OR-

ESIE(8)-U() ETU

Related Features

Cordless Telephone Connection

Programming

- ➔ **10-03-01 ETU Setup**
Terminal Type (B1) for ESI
- ➔ **15-07 Programmable Function Keys**
- ➔ **20-07 Class of Service Options (Administrator Level)**
Refer to [Table 2-6 Class of Service Options \(Administrator Level\)](#), Program 20-07 on page 2-205.
- ➔ **20-08 Class of Service Options (Outgoing Call Service)**
Refer to [Table 2-7 Class of Service Options \(Outgoing Call Service\)](#), Program 20-08 on page 2-207.
- ➔ **20-09 Class of Service Options (Incoming Call Service)**
Refer to [Table 2-8 Class of Service Options \(Incoming Call Service\)](#), Program 20-09 on page 2-209.
- ➔ **20-10 Class of Service Options (Answer Service)**
Refer to [Table 2-9 Class of Service Options \(Answer Service\)](#), Program 20-10 on page 2-210.
- ➔ **20-11 Class of Service Options (Hold/Transfer Service)**
Refer to [Table 2-10 Class of Service Options \(Hold/Transfer Service\)](#), Program 20-11 on page 2-211.
- ➔ **20-12 Class of Service Options (Charging Cost Service)**
Refer to [Table 2-11 Class of Service Options \(Charging Cost Service\)](#), Program 20-12 on page 2-214.
- ➔ **20-13 Class of Service Options (Supplementary Service)**
Refer to [Table 2-12 Class of Service Options \(Supplementary Service\)](#), Program 20-13 on page 2-215.
- ➔ **20-14 Class of Service Options for DISA/E&M**
Refer to [Table 2-13 Class of Service Options for DISA/E&M Service](#), Program 20-14 on page 2-219.

Operation

To set up and program the *D^{term}* Cordless Lite II (DTH-4R-1):

1. Press and hold down * and #, then press **TALK**. The F1 LED flashes red and F1=LK01 is displayed on the LCD.
2. Press **Ring/Vol** repeatedly to scroll through the line key (LK) and feature options for function key **F1**.

3. Press **On/Off MUTE** to select the displayed line key or feature.
4. When a Line key is assigned, press **MUTE** once to enter the Off-Hook Ringing ON or OFF Mode. Press **Ring/Vol** to toggle between TALK for On or NO TALK for Off.
 - ✎ *TALK is selected when the F1~F4 function keys are programmed for CO or Call Appearance Keys. NO TALK is selected when F1~F4 function keys are programmed for functions not requiring an off-hook state (e.g., Log On/Off or DND.)*
5. Press **On/Off MUTE** to advance to the next function key (F2 ~ F4).
6. After programming F4, press **On/Off MUTE** to advance to Global Off-Hook Ringing Assignment.
7. Press **Ring/Vol** to turn Global Off-Hook Ringing On or Off (LCD indicates ON or OFF as appropriate).
8. Press **TALK** to exit.
 - ✎ *Function keys F1 ~ F4 can be programmed as Line Keys 1~16, Redial (LNR/SPD), Answer (ANS), Feature (FNC), or Recall. When assigned, these keys operate the same as on an NEC Multiline Terminal.*
 - ✎ *When initially installed, function keys F1~F4 default to Line keys 1~4 respectively and Off-Hook Ringing defaults to ON.*
 - ✎ *Global Off-Hook Ringing must be ON (default) for any function key to operate with off-hook ringing.*

Switching Between the Desktop Multiline Telephone and the D^{term} Cordless Lite Telephone Using the Base Unit:

When the D^{term} Cordless Lite II is associated with a Multiline Telephone the following is applicable:

- Switching between the cordless mode and desk mode must be done while both telephones are idle.
- A call in progress cannot be switched between the D^{term} Cordless Lite II and the associated Multiline Telephone.
- Switching held calls between the D^{term} Cordless Lite II Telephone and the associated Multiline Telephone is not recommended because line key LED indications are not provided.

Switching from Multiline Telephone and D^{term} Cordless Lite Telephone:

1. Press the Cordless button on the base unit.

- OR -

Press **REDIAL** key from the D^{term} Cordless Lite II handset. *

* This applies only if the **REDIAL** key is programmed to perform desk to cordless switching.

To switch from D^{term} Cordless Lite II Telephone to Multiline Telephone:


1. Press the **DESK** button on the base unit.
For additional Operating Procedures, refer to the D^{term} Cordless Lite II (DTH-4R-1) Owner's Guide.

D^{term} *Handset Cordless*

Description

The *D^{term}* Handset Cordless Terminal is a stand-alone telephone with a direct connection to one digital port on the ESI(8)-U() ETU.

The *D^{term}* Handset Cordless Terminal has the following features:

- 40 separate Channels for Base unit communication
- 3-Channel semi-auto scan (MCA)
- 900 MHz Analog FM spectrum with Voice Scramble
- MW Lamp for incoming call and voice mail message notification
- 30~100 foot operating range between Handset and Base unit without obstructions or other environmental factors
- AutoTalk™ Feature
 -  *AutoTalk is a trademark of Uniden America Corp.*
- Any Key Answer
- LED Low Battery Warning
- Talk (Talk), Transfer (Trf), Mute (Mute), Channel (Ch), Volume (Vol), and Ringer On/Off keys
- 4-hour Talk Time
- 40-hour Standby Time

Conditions

- The Handset Cordless terminal provides 40 different channels for communication with the Base Unit.
- When communication is stopped for about 30 seconds during handset conversation, the Handset Cordless terminal automatically searches for a clear channel. An error tone rings the handset when it is not on the Base Unit.
- When reception becomes garbled or static occurs during conversation, channels can be changed manually. Press the **Ch** key to activate the 3-channel automatic scan (MCA). When channel change fails, an error tone is generated and the channel does not change. While the channel is changing, the Talk/Batt.low LED flashes (On 100 msec., Off 100 msec.).

-
-
- When a communication path is established between the Handset Cordless terminal and the Base Unit, the Talk/Batt.low LED is on and dial tone is generated.
 - When a communication path cannot be established in five seconds, an error tone is generated and the Handset returns to standby.
 - A pencil eraser or contact cleaner should be used monthly to clean the battery contacts on the Handset and Base Unit to maintain stable charge.
 - When the battery is low during standby, the Talk/Batt.low LED blinks (On 50 msec., Off 50 msec.).
 - When the battery is low during a call, an alarm tone is generated, and the Talk/Batt.low LED blinks (On 50 msec., Off 50 msec.). The Handset must be returned to the Base Unit.
 - The battery is charged by the line voltage.
 - Only one Handset Cordless terminal can be connected per ESI port.
 - A provided RF line cord should be used for best performance.
 - The Handset Cordless is a stand-alone terminal.
 - Base units of the Handset Cordless terminals should be installed at least 17 feet apart for optimum performance. The distance range between the Base Unit and the handset is 30~100 feet without obstructions, depending on the environment.
 - Distance of DTP-16HC-1 from the switch is 60 Meters (197 feet). Condition is less than 10 ohms loop resistance. Maximum distance when ACA-U Unit is connected, is 330 meters (1,083 Feet). Condition is less than 57 ohms loop resistance.
 - The Handset does not ring when the battery is low.
 - The standard charged nickel-cadmium battery may last four hours for conversation and 40 hours for standby. Initially the battery must be charged without interruption for 17.5 hours. The charge LED (below the Base Unit contacts) is red during charging and remains on after charging is completed.
 - After continuous charging, the Handset battery may fail to charge or discharge quickly. In either case, operate the Handset until the low battery alarm sounds, auto shutoff occurs, and the battery is fully discharged. Charge the battery without interruption for 17.5 hours. When the battery no longer charges, replace it.
 - The ACA-U Unit is the only adapter that can be used for the Handset Cordless Terminal.
 - ✎ Using the ACA-U Unit minimizes normal display blanking when the cordless handset is returned to the Base Unit.
 - The D^{term} Handset Cordless is not wall mountable.
 - Depending on your environment, the maximum number of cordless devices used without interference varies.
 - Multiple base and handset units should not be closer than 20 feet anytime.

- Radio interference causes interruptions in conversation. When this happens, your unit is not defective. When noise continues, move to a different location while you talk. (You might even need to move the Base Unit.) When the situation persists, contact NEC Unified Solutions, Inc., National Technical Assistance Center (NTAC).

Default Setting

None

System Availability

Terminals

- DTP-16HC-1 TEL
- AC(A)-U Power Adapter

Required Component(s)

None

Related Features

None

Programming

None

Operation


All Operating Procedures assume these features are assigned to the DTP-16HC-1 terminal:

- Prime Line Pickup to the Primary Extension
- Ringing Line Preference

To change the Auto Talk mode:

1. Hold down **Trf** key while changing the Ringer On/Off switch from Off to On.
2. When the mode changes, a confirmation tone is generated:
Auto Talk Off: 3 Beeps
Auto Talk On: 2 Beeps

Answering calls:

1. When the Handset is on the Base Unit, only the Base Unit rings and the Large LED flashes as usual.
2. When in Standby mode, the handset and Base Unit both ring and the message waiting LED flashes rapidly in conjunction with the Base Unit Talk/Batt.low LED.
 *When Ringer On/Off switch is in Off mode, the Handset does not ring, but the message waiting LED flashes.*


With Auto Talk off:

1. When the Handset is on the Base Unit, lift it and press any key.
2. The Talk/Batt.low LED flashes then lights continuously to indicate established communication path between the Handset and the Base Unit.
3. Talk to the caller. The Message waiting LED is Off.
4. When finished, press the **Talk** key or place Handset back in the Base Unit to disconnect the call.

With Auto Talk On:


1. When the Handset is on the Base Unit, lift it.
2. The Talk/Batt.low LED flashes then lights continuously to indicate established communication path between the Handset and the Base station.
3. Talk to the caller. The Message waiting LED is Off.
4. When finished, press the **Talk** key or place Handset back in the Base Unit to disconnect the call.
5. In Auto Standby, the Talk key does not have to be pressed to disconnect. Place the Handset in the Base Unit to disconnect the call.
6. After call is disconnected the Message Waiting LED becomes active again when messages are waiting.

Setting handset ringer On/Off Volume in Standby Mode:

1. Press **Vol** key to adjust volume normal or high.
2. While **Vol** is pressed, the ringer starts to ring based on the current setting.
 *Ringer Volume can be adjusted with Ringer On/Off key in Off.*


To Adjust Receiver Volume with a call in progress:

Press the **Vol** key repeatedly to adjust from Medium † High † Low † Normal in this order.

 *When call is disconnected, default volume returns to Normal.*

To Mute Handset to disable the microphone:

1. Press **Mute** key.
2. The Talk/Batt.low Led Blinks 500 msec On, 500msec Off during Mute.
3. Press **Mute** again to reactivate the microphone.

 *Voice is not transmitted during Mute.*

To make a call with Auto Talk Off:

1. Lift the Handset from the Base Unit.
2. Press **Talk** key.
3. The Talk/Batt.low LED flashes then lights continuously to indicate an established communication path between the Handset and the Base Unit and dial tone is generated.
4. Dial the number.
5. Talk to the party.
6. When finished, press the **Talk** key or place Handset back in the Base Unit to disconnect the call.

To make a call with Auto Talk On:


1. Lift the Handset from the Base Unit.
2. The Talk/Batt.low LED flashes then lights continuously to indicate established communication path between the Handset and the Base Unit and dial tone is generated.
3. Dial the number.
4. Talk to the party.
5. When finished, press the **Talk** key or place Handset back in the Base Unit to disconnect the call.

To Hold a call in progress at the Base Unit:

1. Press **Hold** and return the Handset to the Base Unit.
2. To return to the held party press the flashing green line key and press **Talk** or pick up the Handset.

 *After a programmed time, the held call recalls to the originating terminal.*


To Hold a call in progress at the handset:

1. Press **Trf**.
2. An Internal Dial tone is generated.
3. To return to the held party press the flashing green line key or conference key.
 *After a programmed time the held call recalls to the originating terminal.*

To establish a conference call at the Base Unit:

1. With a call in progress, press **Conf** on the Base Unit.
2. Dial another station number.
3. Announce Conference (Optional).
4. Press **Conf** on the Base Unit to establish the conference.

To Install the battery:

1. Slide the battery compartment cover on the lower Handset back down to remove it.
2. Remove the IMPORTANT charging NOTE from the battery compartment.
3. Align the guides on the bottom of the battery plug and insert it in the connector.
4. Put the battery in the compartment with the description visible.
5. Replace and push up the cover to snap it in place.
 *Ensure that the wires are inside the cover.*

To install the belt clip:

1. Position the clip to the back side of the Handset unit with the clip spring pointed up.
2. Align the four prongs of the clip to the four slots on the sides of the Handset.
3. Gently press the belt clip to snap the prongs into place.

D^{term} *IP Gateway System*

Description

The *D^{term}* IP Gateway system converts traditional voice traffic and its accompanying signaling for call setup and networking to IP for transport across a managed IP network. The system allows users at branch offices or telecommuters to take advantage of the rich feature set of the company central site Key Telephone System as though they were connected locally without loss of functionality. Traditional voice traffic (plus call setup and networking signaling) travel from the KTS to the Gateway where they are converted to IP packets that are shipped to a 10 Mb (Gateway) or 10 Base-T/100 Base-TX (Gateway II) Ethernet LAN port. They are then picked up by the enterprise router and sent to the corporate WAN.

At the branch office, the local router receives the WAN signals and feeds them to the local 10 Base-T/100 Base-TX LAN. From there they reach the *D^{term}* IP Branch unit that converts the IP packets back to standard (TDM) voice signals for distribution to attached *D^{term}* Series E/Electra Elite telephones. The *D^{term}* IP Branch supports up to 12 voice circuits. When the remote site is a telecommuter at home or only a few users, the remote IP/TDM conversion is performed by a *D^{term}* IP Adapter that fits easily on the bottom of a standard *D^{term}* Series E telephone.

This system:

- supports circuit-/packet-based Networks.
- allows mix and match IP-capable EXTender clients up to eight or 12 users.
- connects to the digital-line side of the Electra Elite IPK II KTS.
- allows virtual configuration, management, and troubleshooting of EXTender clients from a central location.
- allows synchronous transmission.
- supports an asynchronous Terminal Adapter (TA).
- allows encrypted user name and password on each port.
- supports call suspend mode on ISDN line with Asynchronous TA.
- supports IP Precedence and DiffServ QoS mechanisms.
- allows choice of network topologies and variable compression rates.

SOFTWARE SUPPORT

- TCP/IP protocol supports IP traffic and management.
- RVPoIP uses UDP/IP protocol for voice transmission.
- Serial RVP uses HDLC encapsulation for voice transmission.
- Management and utilities include Telnet, inband RVP, RS-232 console interface (Gateway and Branch), SNMP, and HTML.
- Software upgradable, using FTP, between the Gateway and the branch, RS-232 serial port to PC, or inband connection.

Conditions

- Refer to *D^{term}* IP Gateway System Administrator's Guide or *D^{term}* IP Gateway II System Administrator's Guide for more information.
- TAPI is not supported.
- When you dial 911 on your *D^{term}* telephone, the 911 facility reached is the one that serves the Gateway facility. To reach the correct 911 service for your area, use a telephone connected locally. The *D^{term}* IP Branch unit provides an analog port for local dialing.
- Synchronous ringing does not work on the remote telephone. Incoming trunk calls automatically ring two seconds on/two seconds off.
- Analog Fax machines and modems cannot operate with a *D^{term}* Multiline Terminal that is connected to an APR-U Unit or AP(A)-R Unit optional adapter.
- The CT(A)-R Unit, CTA-U Unit, or CTU(S)-U Unit cannot be installed to a KTS remote Multiline Terminal to provide TAPI functionality with the KTS.
- The DTP-32DE-1 Multiline Terminal is the only one certified for use in a home environment. None of the other *D^{term}* Series E or Electra Elite Multiline Terminals are certified for home use because of FCC restrictions, but they are approved for use in an office environment.
- The HFU-U/HF-R Unit handsfree adapter cannot operate with a KTS remote Multiline Terminal for home use because of FCC restrictions, but it is approved for use in an office environment.
- An applicable network terminating device must be installed and functioning at both the corporate facility and the branch office.
- The data connection between the corporate facility and the branch office must be configured properly and operational.
- Network termination equipment must support an RS-232, V.35 or RS-530 synchronous interface or support IP devices over Ethernet.
- The system operates from 100-240 VAC, 50-60 Hz. Power must not be applied to the Gateway unit until specified in the installation procedures.

- The Gateway power supply and cables should be installed away from high power/high RF noise devices such as computers, fans, fluorescent ballast, or power supplies.
- Good wiring practices are required. Do not run wires across devices such as fluorescent lights, computers, or air conditioners to avoid introducing noise in the modems.
- The distance between the PBX/KTS and the Gateway unit must not exceed 500 feet.
- When 5-, 6-, or 7-digit station numbers are used, this feature is not available.

Default Setting

None

System Availability

Terminals

- DTU/DTP/DTR Multiline Terminals
- DTP-32DE-1 for home environment

Required Component(s)

Components include the following:

- Electra Elite IPK II KTS with one ESI port for each Branch port and IP Adapter
- Gateway unit
- Branch unit and/or IP Adapters with total ports not exceeding the number of ports supported by the Gateway unit
- Power Cord
- RS-530 cable
- 2 RJ-21 50-pin cables to connect Gateway to DTS and Branch to punch block – connectors required include a female connector to attach to D^{term} IP, and a male connector to attach to the KTS.
- Punch blocks that can cross-connect the D^{term} IP Gateway to the KTS and D^{term} telephones to the branch locations.
- Installation hardware for Gateway and Branch units
- System Administrator's Guide
- Quick Installation Guide

- Application Guide
- PC for configuration
- T1, Fractional T1, ISDN, Frame Relay or ATM circuit that can connect the Branch unit to the Gateway location.

A networking terminating device that can interface with the Gateway and Branch/Adapter units and support synchronous serial protocol using RS-232, V.35, or RS-530 interface, or 10 Base-T Ethernet for an IP connection.

Network Requirements

- The Gateway must be installed on an existing LAN or WAN network.
- Each unit requires a network device that supports a synchronous serial interface or an Ethernet connection to a TCP/IP network.
- The network must be operational and active to complete installation of the Gateway.
- The network device must support one of the following:
 - ❑ Synchronous serial signaling on its data port using an RS-232, V.35, or RS-530 interface
 - ❑ 10 Base-T Ethernet connections

Related Features

None

Programming

None

Operation

Operates the same as any Electra Elite Multiline Terminal

E911 Compatibility

Description

IMPORTANT - PLEASE NOTE THE FOLLOWING!	
1.	All local trunks or the trunk connected to external 911 equipment must be put in the E911 route. Placing all the local trunks into the E911 route assures that any user may make a call to 911.
2.	When ARS is NOT enabled and the system allows trunk access by dialing 9, single line telephones disregard Program 20-03-03 – System Options for Single Lines Telephones - SLT DTMF Dial to Trunk Lines. This prevents the system from connecting to a trunk until all the digits are dialed. This can be avoided by using either 8 or 9x (but not 91) as the trunk access code. Be aware that this change requires additional programming changes.
3.	When using external E911 equipment, do not allow analog single line telephones to directly access trunks. When an analog SLT directly accesses a trunk (#9 xxx) and dials 911, the system does not follow the 911 routing. If your system is connected to external E911 equipment, the system does not route the call to that equipment.
4.	Do not use * within a PBX access code if the Account Code feature is used. With the Account Code feature enabled, if * is used in the access code, the trunk stops sending digits to the central office after the * is sent.
5.	Finally, but most importantly, TEST - TEST - TEST! Due to the nature of the E911 feature, it is imperative that when programming this, or any other feature, to be aware of the consequences. Make sure to test the extensions with the E911 feature to confirm that other features do not prevent the call from being completed. When using external equipment, make sure the dial treatment tables are working properly.

E911 Compatibility ensures that emergency calls always get through. If an emergency occurs, a user simply goes to any telephone, lifts the handset and dials 911. The system built-in E911 compatibility places the emergency call even if the user forgets to dial an access code or press a line key. The E911 abilities include:

Attendant Notification

The attendant receives a notification each time a co-worker dials an emergency 911 call. This notification is the co-worker's name and number display optionally accompanied by an audible alarm. Notification occurs regardless of whether the attendant is idle or busy on a call. You can optionally extend this ability to other supervisory extensions as well.

 Emergency Routing

When an extension user dials 911, the system can automatically find a trunk for the call. The system can choose a route to which the user normally does not have access. If all normal routes are busy, the system can even disconnect an active call and place the emergency call. E911 Compatibility uses the flexibility of the Automatic Route Selection Call Route Options to route 911 emergency calls (even in systems in which ARS is not enabled).

 E911 Outgoing Dialing

The E911 calls follow the trunk group route programming. You can use the flexibility of the Automatic Route Selection Call Route Options for additional routing options.

 Forced Disconnect Follows Timer to Disconnect Call

When all lines in the programmed route are busy and the system must drop a call to place a 911 call, the system waits the time set in Program 81-01 before disconnecting the call.

 Compatibility with Customer Provided E911 Equipment

The system can automatically send a 911 call to customer-provided E911 equipment (such as the Proctor 911 ANI-LINK System II). The E911 equipment dials emergency service and provides the caller's extension number to the emergency personnel. When using this type of equipment, the system must be programmed to send E911 calls to the trunk connected to the E911 box. This is done by assigning the trunk to the E911 trunk group and using ARS to route all 911 calls to that port. The Dial Treatment must then be set to send the extension number and 911 to that port. This is usually accomplished using XRE in the Dial Treatment, but check the requirements of your E911 box.

 Calling Party Number (CPN) Presentation from Station

With ISDN installed, the system can provide the Calling Party Number (CPN) Presentation from Station. No additional customer-provided 911 equipment is required.

Uninstalled Trunks in Trunk Group Prevent Call from Dialing Out

By system default, all trunks in Program 14-05-01 Trunk Group are in group 1. When placing a 911 call, the system tries to access the trunks defined in the group. If the trunks do not exist, the call does not dial out. For E911 to function correctly, remove any uninstalled trunks from the trunk group.

If Program 21-01-12 : System Options for Outgoing Calls – Dial 911 Routing Without Trunk Access is set to 0 (trunk access code required), when using the Dial Number Preview feature and dialing 9+911, if all trunks are busy, the user hears a busy signal and the call does not dial out.

If option PRG 21-01-12 is set to 1 (trunk access code not required) and using Dial Number Preview, 911 is dialed, the system disconnects a trunk and dials the call.

Dial Number Preview is when a telephone number is first dialed (previewing the number in the display) then SPEAKER or the line key is pressed to place the call.

Conditions

- If Program 21-01-10 is set to an entry other than 0, a call does not have a talk path unless the user dials at least the number of digits entered in this option when placing an outgoing call. An entry of 4 digits or higher causes a problem when dialing 911. Since it is only a 3-digit number, the call does not have a talk path, preventing the emergency dispatcher from hearing the caller. This option should be kept at its default setting of 0 to prevent any problem with dialing 911.
- CAMA Trunks are not supported.
- If a 911 call originates from a Virtual Extension, the system sends the Virtual Calling Party Number (CPN) Presentation from Station.

Default Setting

Disabled

Programming

- ➔ **11-12-56 Service Code Setup (for Service Access) – E911 Alarm Shut Off**
Select the Service Code (normally 786) an extension user dials to shut off the E911 Alarm Ring.
- ➔ **14-05-01 Trunk Group – Trunk Group Number**
Assign the outbound trunks you want to use for E911 service to the same Trunk Group.
Trunk Groups: 1~100
- ➔ **14-06-01 Trunk Group Routing**
Set up an outbound routing table for trunk groups assigned in Program 14-05. This program and Program 21-02 are the minimum required if E911 must seize a line to dial.
Trunk Group Routes: 1~100
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service to an extension (1~15).
- ➔ **20-08-16 Class of Service Options (Outgoing Call Service – Display E911 Dialed Extension Name and Number)**
Enable (1)/Disable (0) the ability to display the name and number of the extension that activated E911 service. If disabled (0), the E911 Alarm (set in Program 21-01-13) never occurs.
- ➔ **21-01-12 System Options for Outgoing Calls – Dial E911 Routing Without Trunk Access**
If enabled (1), an extension user can dial 911 without first dialing a trunk access code or pressing a line key. If disabled (0), an extension user must dial a trunk access code (e.g., 9) or press a line key before dialing 911. If enabled, dialing 9+911 still dials out.
- ➔ **21-01-13 : System Options for Outgoing Calls – Alarm Ring Timer (E911)**
Set the duration of the E911 Alarm Ring Time (0~64800 seconds). If set to 0 (default), the E911 Alarm does not ring.

-
-
- ➔ **21-02-01 Trunk Group Routing for Extensions**
Assign the routes set in Program 14-06 to extensions. This program and Program 14-06 are the minimum required if E911 must seize a line to dial.

Trunk Groups: 1~100

System Availability

Terminals

None

Required Component(s)

None

Related Features

Automatic Route Selection

Central Office Calls, Placing

T1 Trunking (with ANI/DNIS Compatibility)

ISDN Compatibility

Operation

To place an emergency 911 call:

When Dial 911 Routing Without Trunk Access is enabled


1. Go to any telephone.
2. Lift the handset or press **Speaker**.
3. Dial **911**.

When Dial 911 Routing Without Trunk Access is disabled

1. Go to any telephone.
2. Lift the handset or press **Speaker**.

3. Dial a trunk access code (e.g., 9) or press a line key.
4. Dial **911**.

To turn off the E911 Alarm at your telephone:

1. Lift the handset or press **Speaker**.
2. Dial **786**.
 *The alarm goes off.*
- **OR** - (if you have a display telephone)
 1. Press the **Exit** key once to turn off the alarm.
 2. Press the **Exit** key again to clear the alarm display.

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Electra Elite IPK Terminals

Description

The Electra Elite IPK Terminals provide ergonomic form and user-friendly functions. With advanced digital circuitry, the IPK Terminals consists of distinct models to meet diverse user telephone terminal needs.

The Electra Elite IPK II system allows a maximum of 240 Electra Elite IPK terminals to be attached to the system.

Conditions

- The Electra Elite IPK II Terminals, with an adjustable display, offer softkey operation. The LCD panel has three lines of display, each with 24 characters. Standard features include headset jacks, wall mount units, and adjustable-base units.
- The Electra Elite IPK II Terminals support dedicated function keys to provide easy one-touch access to the most common telephone operations. These keys include: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, and Message.
- The dial pad is detachable allowing for easy customization for a foreign language (French or Spanish), or for Automatic Call Distribution (ACD) applications.
- All Electra Elite IPK II Terminals are Class B devices and comply with U.S. FCC regulations for office and residential use. They also comply with requirements of the Canadian Interference-Causing Equipment Regulations.
- With the DTH-16LD-1 TEL, the 16-Line Keys are labeled by the LCD by assignment in system data. The LCD also supports the LED status for trunks, Call Appearance (CAP) Keys, DSS/BLF keys, and select Feature keys/Feature Access keys. PRG 15-02-01 will not support language conversion entered in PRG 15-20.
- Use PRG 15-20 (LCD Line Key Name Assignment) to assign a name to each LCD Line Key of the DTH/DTR-16LD-1/2 Telephone. Up to eight characters can be assigned.
- The LCD of the Electra Elite IPK II Terminal provides a volume bar indication, while adjusting the following volume levels or controls:
 - Speaker Volume
 - Handset/Headset Volume
- BGM Volume
- Ring Volume/Off-Hook Ring Volume

- LCD Contrast
 - ✎ Only English displays are provided (SPEAKER, HANDSET, RING, LCD).
- MIC controls the built-in microphone during speakerphone mode and controls the handset mute feature during handset/headset operation.
- The DIR Key performs the same function as the Caller ID Scroll Key.
- The MSG Key acts as a VM access key to call the VM pilot number (Feature selection PRG 15-02-26).
- The distance from the KSU for IPK Multiline Terminals (DTH type) can be extended when local AC power is provided. When the IPK Terminal is powered by a local AC-R Unit (AC Adapter), a built-in Long Line Adapter allows these telephones to be connected up to a distance of 2000 feet by Twisted 1-Pair Cable at 24 AWG.
- For compatibility of Adapter Units and Terminals, refer to the following table:

Adapter Unit	Terminal						
	DTH-16LD-1	DTH-32D-1	DTH-16D-1	DTH-8D-1	DTH-8-1	ITH-16D-2	ITH-8D-2
AD(A)-R	X	X	X	X	X	–	–
AP(A)-R	X	X	X	X	X	–	–
AP(R)-R	X	X	X	X	X	–	–
CT(A)-R	X	X	X	X	X	–	–
CT(U)-R	X	X	X	X	X	–	–
HF-R	X	X	X	X	X	–	–
IP-R(IPK)	X	X	X	X	–	–	–
Button Units							
BS(F)-R() UNIT	X	X	X	X	X	X	X
BS(S)-R() UNIT	X	X	X	X	X	X	X
Other							
WM-R UNIT	X	X	X	X	X	X	X
AC-R UNIT	X	X	X	X	X	X	X
IP Telephone Adapters							
AD(A)-2R UNIT	–	–	–	–	–	X	X
ILPA-R	–	–	–	–	–	X	X

X = Compatible

– = Not comparable

- With non-IP Electra Elite IPK II Terminals, up to two adapters can be installed in a telephone. For compatibility of multiple adapter units, refer to the following table:

	AD(A)-R	AP(A)-R	AP(R)-R	CT(A)-R	CT(U)-R	HF-R	IP-R(IPK)
AD(A)-R		X	X	X	X	X	–
AP(A)-R	X		–	X	X*	X	–
AP(R)-R	X	–		X	X*	X	–
CT(A)-R	X	X	X		–	X	–
CT(U)-R	X	X*	X*	–		X	–
HF-R	X	X	X	X	X		–
IP-R(IPK)	–	–	–	–	–	–	

X = Compatible

– = Not compatible

* = The Voice Application of CT(U) cannot be used when installed with AP(A/R)-R Unit.

- An AC-2R Unit/AC-3R Unit (AC Adapter) is required when any of the following adapters is installed in an IPK Terminal:
 - AP(R)-R Unit
 - CT(U)-R Unit
 - HF-R Unit
 - IPR-R Unit (When in-line power is used, power over Ethernet, is not provided.)
 - DCR-60-1 Console
- An AC-2R Unit/AC-3R Unit (AC Adapter) is required for the ITH-16D-2 and ITH-8D-2 telephones when in-line power, power over Ethernet, is not provided or the ILPA-R is not used.
- The distance from the KSU for Electra Elite IPK II DTH Multiline Terminals can be extended when local AC power is provided. When the Electra Elite IPK II terminal is powered by a local AC-2R/AC-3R Unit (AC Adapter), a built in Long Line Adapter allows these telephones to be connected up to 2,000 feet by Twisted 1-pair 24AWG Cable.
- The WM-R Unit (Wall Mount Unit) is required when any adapter is installed in an Electra Elite Terminal and the terminal is to be wall mounted.

Default Settings

None

System Availability

Terminals

- DTH-16LD-1 TEL
- DTH-32D-1 TEL
- DTH-16D-1 TEL
- DTH-8D-1 TEL
- DTH-8-1 TEL
- DCR-60-1 CONSOLE
- IP Terminals
 - ITH-16D-2 TEL
 - ITH-8D-2 TEL

Required Component(s)

- ESI(8)-U() ETU
- ESIB(8)-U() ETU
- ESIE(8)-U() ETU
- IP ETU
 - IPD(4/8)-U() ETU (for IP Terminals)

Optional Component(s)

- AD(A)-R UNIT (Adapter for Call Recording)
- AP(A)-R UNIT (Analog Port Adapter [without Ringer])
- AP(R)-R UNIT (Analog Port Adapter [with Ringer])
- CT(A)-R UNIT (TAPI Adapter connect by RS-232C)
- CT(U)-R UNIT (TAPI Adapter connect by USB)
- HF-R UNIT (Adapter for Full Duplex Handsfree)
- IP-R(IPK) UNIT (IP Adapter)
- BS(F)-R UNIT (French Button Unit)
- BS(S)-R UNIT (Spanish Button Unit)

- WM-R UNIT (Wall Mount Unit)
- AC-2R Unit (AC Adapter)
- AC-3R Unit (AC Adapter, PC Type)
- IP Units
 - AD(A)-2R UNIT (Adapter for Call Recording)
 - ILPA-R (In Line Power Adapter)

Related Features

Ancillary Device Connection

Electra Elite Terminal Migration

IP Station (MEGACO) – MG 16

Off-Hook Signaling

Softkeys

User Programming Ability

Programming

- ➔ **10-03-01 ETU Setup**
Terminal Type (B1) for ESI
- ➔ **15-07 Programmable Function Keys**
- ➔ **15-20-01 LCD Line Key Name Assignment**
- ➔ **20-07 Class of Service Options (Administrator Level)**
Refer to [Table 2-6 Class of Service Options \(Administrator Level\)](#), Program 20-07 on page 2-205.
- ➔ **20-08 Class of Service Options (Outgoing Call Service)**
Refer to [Table 2-7 Class of Service Options \(Outgoing Call Service\)](#), Program 20-08 on page 2-207.
- ➔ **20-09 Class of Service Options (Incoming Call Service)**
Refer to [Table 2-8 Class of Service Options \(Incoming Call Service\)](#), Program 20-09 on page 2-209.

- ➔ **20-10 Class of Service Options (Answer Service)**
Refer to [Table 2-9 Class of Service Options \(Answer Service\)](#), Program 20-10 on page 2-210.
- ➔ **20-11 Class of Service Options (Hold/Transfer Service)**
Refer to [Table 2-10 Class of Service Options \(Hold/Transfer Service\)](#), Program 20-11 on page 2-211.
- ➔ **20-12 Class of Service Options (Charging Cost Service)**
Refer to [Table 2-11 Class of Service Options \(Charging Cost Service\)](#), Program 20-12 on page 2-214.
- ➔ **20-13 Class of Service Options (Supplementary Service)**
Refer to [Table 2-12 Class of Service Options \(Supplementary Service\)](#), Program 20-13 on page 2-215.
- ➔ **20-14 Class of Service Options for DISA/E&M**
Refer to [Table 2-13 Class of Service Options for DISA/E&M Service](#), Program 20-14 on page 2-219.

Operation

Refer to individual features for details.

Table 2-19 Electra Elite IPK Telephone Specifications

	DTH-16LD-1	DTH-32D-1	DTH-16D-1	DTH-8D-1	DTH-8-1	ITH-16D-2	ITH-8D-2
Dedicated Function Keys	11	11	11	11	11	11	11
Line/Feature Access/ Programmable Feature Access Key	16	32*	16	8	8	16	8
LCD (3x24)	Yes	Yes	Yes	Yes	No	Yes	Yes
Call/Message Indicator	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Adjustable Base	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Built-in Wall Mount	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Headset Jack (Built-in)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
DESI Label by LCD	Yes	No	No	No	No	No	No
Receiver Volume Control for:							
Handset	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Speakerphone	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Headset	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Ring Volume Control	Yes	Yes	Yes	Yes	Yes	Yes	Yes

Table 2-19 Electra Elite IPK Telephone Specifications (Continued)

	DTH-16LD-1	DTH-32D-1	DTH-16D-1	DTH-8D-1	DTH-8-1	ITH-16D-2	ITH-8D-2
LCD Contrast Control	Yes	Yes	Yes	Yes	No	Yes	Yes
Housing Color	White or Black	White or Black	White or Black	White or Black	White or Black	White or Black	White or Black
Softkeys	Yes	Yes	Yes	Yes	No	Yes	Yes

* A maximum of 32 keys may be programmed as Function Keys.

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Electra Elite Terminal Migration

Description

Electra Elite Terminal Migration allows an Electra Elite customer to protect their investment in terminals when purchasing Electra Elite IPK II systems. Electra Elite multiline terminals can easily be used with the Electra Elite IPK II systems. With very few exceptions, all telephone features and abilities that are possible on Electra Elite 48/192 are also possible with the Electra Elite IPK II system.

Conditions

Restrictions

- Dedicated Function Keys for Microphone, Directory, and Message are not provided with Electra Elite Multiline Terminals.
- Electra Professional Multiline Terminals are not supported with the Electra Elite IPK II system.

Default Settings

None

System Availability

Terminals

Electra Elite Multiline Terminals

Required Component(s)

- ESI(8)-U() ETU
- ESIB(8)-U() ETU
- ESIE(8)-U() ETU

Related Features

Ancillary Device Connection

Electra Elite IPK Terminals

Handsfree and Monitor

Off-Hook Signaling

Softkeys

Programming

- **10-03-01 ETU Setup**
Terminal Type (B1) for ESI
- **15-07 Programmable Function Keys**
- **20-07 Class of Service Options (Administrator Level)**
Refer to [Table 2-6 Class of Service Options \(Administrator Level\)](#), Program 20-07 on page 2-205.
- **20-08 Class of Service Options (Outgoing Call Service)**
Refer to [Table 2-7 Class of Service Options \(Outgoing Call Service\)](#), Program 20-08 on page 2-207.
- **20-09 Class of Service Options (Incoming Call Service)**
Refer to [Table 2-8 Class of Service Options \(Incoming Call Service\)](#), Program 20-09 on page 2-209.
- **20-10 Class of Service Options (Answer Service)**
Refer to [Table 2-9 Class of Service Options \(Answer Service\)](#), Program 20-10 on page 2-210.
- **20-11 Class of Service Options (Hold/Transfer Service)**
Refer to [Table 2-10 Class of Service Options \(Hold/Transfer Service\)](#), Program 20-11 on page 2-211.
- **20-12 Class of Service Options (Charging Cost Service)**
Refer to [Table 2-11 Class of Service Options \(Charging Cost Service\)](#), Program 20-12 on page 2-214.
- **20-13 Class of Service Options (Supplementary Service)**
Refer to [Table 2-12 Class of Service Options \(Supplementary Service\)](#), Program 20-13 on page 2-215.



20-14

Class of Service Options for DISA/E&M

Refer to [Table 2-13 Class of Service Options for DISA/E&M Service, Program 20-14](#) on page 2-219.

Operation

Refer to individual features for details.

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EliteApps – Interactive Voice Response

Description

EliteApps – Interactive Voice Response (IVR) is a software application that accepts a combination of voice telephone input, database information, and telephone keypad selection to provide audio (usually voice) information to callers and databases, place calls, transfer calls, and send e-mail messages. IVRs also allow callers to provide voice and data information to be stored in databases used by other user applications. Common IVR applications include:

- Bank and stock account balances and transfers
- Surveys and polls
- Call center hold and forwarding
- Order entry tracking
- Simple order entry transactions
- Selective information lookup (movie schedules, etc.)

The IVR application uses pre-recorded voice, optional text-to-speech, call flow logic, access to relevant data, and records voice input for later handling. Using computer telephony integration (CTI), the IVR can hand off a call to someone that can view data related to the caller at a display.

The programmable IVR uses open database connectivity (ODBC) connections to databases to allow complete customizing of call flows and information anytime. The IVR can generate e-mail messages and can be remotely monitored and configured using a LAN or WAN in a totally secure environment.

Basically, the IVR receives calls, collects information, and forwards the call.

A customer can use NEC Professional Services to customize the IVR to meet a unique user application.

The following major functions are provided:

- Call control
 - Answer an inbound call.
 - Place an outbound call.
 - Collect DTMF digits from a caller.
 - Play one or more audio files to a caller.
 - Play text to a caller as digits, number, date, time, or money.
 - Record from a caller into a file.
 - Select a menu item.

- Access caller ID information.
- Set all call control parameters.
- Supervise normal and Blind transfers.
- Announce a call prior to connection to another port.
- Multilingual support.
- Database Access (ODBC)
 - Open a Data Source Name (DSN).
 - Close an opened DSN.
 - Execute a direct SQL statement to query, insert, update, and delete records.
 - Get data fields from a selected record.
 - Get the next selected record.
 - Use database information to control program logic or present data to callers.
- Program Control
 - Script driven for total control.
 - if, else, else if, end if logic control with up to 20 levels of nesting.
 - goto for branching to a label.
 - exec to start a whole new program.
 - run and return to execute a function (with nesting) and continue.
 - User embedding of comments and tracing information.
 - User generation of events.
- Logging
 - Unlimited number of user definable logs.
 - User specified information in logs.
 - Date and time stamping of all log entries.
- Caching
 - Audio files are automatically cached in Zeus cache.
 - Memory is available for caching that can be configured.
 - No user actions are required.
 - Modified audio files are automatically updated in cache.
- Timers
 - Pause for configured number of milliseconds.
 - Mark the current date and time.
 - Get the number of seconds from a marked time now.
 - Assign a new script to be executed after user specified number of seconds.
 - Suspend a timer.

- Cancel a timer.

Variables

- Variables are user definable.
- Variables can be integer or string type.
- Set a variable to the result of an expression.
- Set a variable to a string.
- Set a variable to another variable.
- Set a variable to the results of a function().
- Use variables anywhere in script syntax.

Strings

- String extraction: Left, Mid, Right.
- String replacement.
- System information: current day of month, day of week, year, month, hour, minute, date.
- Temporary unique name generation.
- File Control.
- Rename files.
- Remove files.
- Copy files.
- Test the existence of a file or directory.

Conditions

- The IVR(8)-U() ETU is assigned as VMS 8 in Program 90-34-01 Firmware Information.

Default Setting

None

System Availability

Terminals

All Stations

Required Component(s)

IVR(8)-U() ETU

Related Features

Digital Voice Mail

IPK II In-Mail

Voice Mail Integration (Analog)

Programming

- ➔ 11-02-01 Extension Numbering
- ➔ 11-07-01 Department Group Pilot Numbers – Dial
- ➔ 15-03-03 Single Line Telephone Basic Data Setup – Terminal Type
- ➔ 16-01-01
through
16-01-03 Department Group Basic Data Setup
- ➔ 16-02 Department Group Assignment for Extensions

Operation

Operating procedures depend on the application.

Elite CallAnalyst

Description

Elite CallAnalyst is an easy to use, graphically oriented software package that allows you to monitor and analyze telephone calls, understand telephone usage, and cut costs. Incoming and outgoing calls are tracked accurately along with the date and time of the call. When the incoming telephone call must be tracked with name and/or telephone numbers, Elite CallAnalyst requires Caller ID service from the local telephone company.

Elite CallAnalyst increases productivity, facilitates billing, and helps detect toll fraud and telephone abuse. It also has powerful tabular (text) and graphic report generating ability. Reports include extension/line summaries, date, time, and department summaries, longest/most expensive calls, and most frequently called numbers. These reports can be used to analyze your telephone as a critical business communication tool, improve its business effectiveness, and reduce your telephone costs. A report can be generated showing calling patterns by volume or duration on a color-coded United States map. This can help a Customer Support, Sales Order, or Telemarketing business become more focused, more productive, and more cost effective.

Elite CallAnalyst keeps track of:

- The date and time calls were made or received
- The duration of each call
- Which extension made or received the call
- The CID/ANI, DNIS of the caller
- The trunk or line numbers which handled the call
- Account codes and authorization codes used for the call
- CCIS calls are now logged with extension number and trunks used for CCIS. These trunks can be placed in a different line group to track usage across a CCIS link using the Traffic Analysis add-on feature.

Highlights of Elite CallAnalyst and Elite CallAnalyst Lite:

- Network based.
- Fraud Alert. This module can generate alarms by e-mail, pager screen, screen pop-up, wav file etc. when it detects user defined patterns in the call records.
- Automatic report and data archival scheduling, to include automatic e-mailing of reports to predetermined destinations.
- Real-time inbound/outbound call monitoring.

- Changes can now be made to the call record such as Account Code Entry, DNIS, and comments field.
- Call costing and user configurable rate plans
- Time billing

Included Reports:

- Date and time summaries
- Most frequently called numbers
- Department summaries
- Extension and line summaries
- Longest and most expensive calls
- And many more

Conditions

- The following software items are installed on the PC:
 - Call Data Manager (CDM) to collect call records from the system.
 - Elite CallAnalyst 1.0 (version depends on license purchased, lite or full) to allow reports and other CallAnalyst features.
 - There is no lite package for Elite CallAnalyst versions 2.0 or 2.5.
 - Scheduler (default installed) to allow reports and database archival depending on the Elite CallAnalyst version.
 - Elite CallAnalyst Manual (default installed, PDF format).
- The following optional modules of Elite CallAnalyst Software require license upgrade:
 - Network Client
Network clients must then call NEC for additional licensing. The license is issued on the CallAnalyst Server installation. All license information is maintained on the server PC. Because the license is unique to the PC where the server is installed, you must follow the help files in the license screen to move the license to another location.
 - Traffic Analysis
This tool allows users to view and analyze trunk capacity usage by date, time, and call direction.
- Refer to the Elite CallAnalyst Manual installed with the software for more detailed information.
- SMDR does not print Intercom calls.

- The SMDR call buffer stores 500 calls. The buffer stores calls when the SMDR device is unavailable. When the buffer fills, the oldest record is deleted to allow the new record to be saved.
- The IPK II SMDR does not provide data to support the tracking of tandem calls or conference calls. Tandem calls appear as one call in with extension number shown as the trunk it was answered on, and out with the extension number as the trunk used to make the call. Conference calls show only the last party to join the conference and the party that answered the call.

Default Setting

None

System Availability

Terminals

All Terminals:

- Incoming CO/PBX Call
- Outgoing CO/PBX Call
- Conference CO/PBX Call
- Transferred CO/PBX Call

Required Component(s)

LAN connection for SMDR over Ethernet

Minimum PC Requirements:

- PC with Pentium Processor
- 64 MB RAM
- VGA monitor 800 X 600 resolution (SVGA 1024 X 768 recommended)
- Windows 98 (SE), Windows 2000/XP Professional, Windows 2000/2003 server
- 80 MB free hard drive space
- CD-ROM drive (When using CD for installation)
- Available serial port and RS-232 cable

- Printer for reports

The Elite CallAnalyst software should be installed on a separate PC, but other applications may be installed. Performance depends on the need for memory and processing for the other applications.

Related Features

Account Code – Forced/Verified/Unverified

Account Code Entry

Central Office Calls, Answering

Central Office Calls, Placing

Clock/Calendar Display

E-911 Compatibility

ISDN Compatibility

Multiple Trunk Types

Station Message Detail Recording

T1 Trunking (with ANI/DNIS Compatibility)

Traffic Reports

Trunk Groups

Programming

For additional SMDR programming options, see [Station Message Detail Recording on page 2-913](#).

Programming Notes:



- If the system is programmed to display the date (Program 35-02-14=1), the date is displayed regardless of the setting for display of trunk name (Program 35-02-03) and only the trunk number is printed.

-
-
- For example, if trunk port 049 has a trunk name of PRI Ch1, if Program 35-02-03 = 0 (name) and PRG 35-02-14 = 1 (display date), then SMDR shows 8/19 049. However, if Program 35-02-14 = 0 (date not displayed), the SMDR shows PRI Ch1.
 - For proper handling of DNIS calls, the name field (Program 22-11-03) must be the same as the received DNIS digits (Program 22-11-01). If this is not set, CallAnalyst cannot track transferred calls since the system displays the DNIS number when a call is received and displays the DNIS name for transferred calls. This setting has no impact on outgoing calls, which display the trunk name instead of the DNIS name.
 - Caller ID name can be displayed in SMDR records. Program 35-02-17 must be set to 0 and Program 35-02-18 set to 1.
 - 14-01-06 Basic Trunk Data Setup – SMDR Printout**
Have the system include (1) or exclude (0) the trunk being programmed for the SMDR printout.
 - 15-01-03 Basic Extension Data Setup – SMDR Printout**
For each extension, enter 1 if an extension call should appear on the SMDR report. Enter 0 for the extension if the calls should not appear.
 - 35-01-01 SMDR Options – Output Port Type**
Specify the type of connection used for SMDR

0 = No Setting (default)
1 = CPU II ()-() COM Port 1
3 = LAN,
4 = CTA
The baud rate for the serial COM port should be set in PRG 10-21-02 or PRG 15-02-19.
 - 35-01-02 SMDR Options – Output Destination Number**
Specify (up to 8 digits) the SMDR printer output extension (CTA/CTU extension number).
Default is No Setting
 - 35-01-03 SMDR Options – Header Language**
Specify the language in which the SMDR header should be printed.

0 = English (default)
1 = German
2 = French
3 = Italian
4 = Spanish.
Set this entry to 0 for Elite CallAnalyst.
 - 35-01-04 SMDR Options – Omit Digits**
The number of digits entered for this option do not print on the SMDR Report (0~24). For example, if the entry is 10, the first 10 digits a user dials do not appear on the SMDR report.
 - 35-01-05 SMDR Options – Minimum Digits**
Outgoing calls must be at least this number of digits (0~24) for inclusion in the SMDR report.
Default is 0.
-
-

-
-
- ➔ **35-01-06 SMDR Options – Minimum Call Duration**
A call must last at least this time (0~65535 seconds) to be included in the SMDR report. Default is 0 (All).
 - ➔ **35-01-07 SMDR Options – Minimum Ring Time (For Incoming Calls)**
A call must ring for at least this time (0~65535 seconds) to be included in the SMDR report. Default is 0 (All).
 - ➔ **35-01-08 SMDR Options – Format Selection**
For North America, set this option to 0 (default).
 - ➔ **35-02-01 SMDR Output Options – Toll Restricted Call**
Enter 1 if you want the SMDR report to Display calls blocked by Toll Restriction. Enter 0 to Not Display blocked calls.
 - ➔ **35-02-02 SMDR Output Options – PBX Calls**
If system is behind a PBX, enter 1 to have SMDR Display all calls to the PBX. Enter 0 to have SMDR Display only calls dialed using PBX trunk access code.
 - ➔ **35-02-03 SMDR Output Options – Trunk Number or Name**
Select whether the system should display the trunk name (0) or the number (1) on SMDR reports. Default is 1. *If this option is set to 0, Program 35-02-14 must be set to 0.*
 - ➔ **35-02-04 SMDR Output Options – Summary (Daily)**
 - ➔ **35-02-05 SMDR Output Options – Summary (Weekly)**
 - ➔ **35-02-06 SMDR Output Options – Summary (Monthly)**
Enter 1 (default) to enable a summary report. Enter 0 to disable a summary report. The daily report prints every day at midnight. The weekly report prints every Sunday night at midnight. The monthly report prints at midnight on the last day of the month.
 - ➔ **35-02-07 SMDR Output Options – Toll Charge Cost**
Set this option to 1 to have the SMDR report Display toll charges.
 - ➔ **35-02-08 SMDR Output Options – Incoming Call**
Enter 1 (default) if you want the SMDR report to Display incoming calls. Enter 0 if you want the SMDR report to Not Display incoming calls.
 - ➔ **35-02-09 SMDR Output Options – Extension Number or Name**
Enter 0 if you want the SMDR report to Display the extension name. Enter 1 (default) if you want the SMDR report to Display the extension number.
 - ➔ **35-02-10 SMDR Output Options – All Lines Busy (ALB) Output**
Enter 1 to Display information when all lines in a group are busy and an extension user tries to access the group. Enter 0 (default) if this information should not be Displayed.
 - ➔ **35-02-11 SMDR Output Options – Walking Toll Restriction Table Number**
Enter 1 (default) to Display Table Number
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-

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-
- ➔ **35-02-12 SMDR Output Options – DID Table Name Output**
Determine if the DID table name should be Displayed for incoming DID calls.
- DID Table Name Display:
0 = Not Displayed (default)
1 = Displayed
- ➔ **35-02-13 SMDR Output Options – CLI Output When DID to Trunk**
Determine if the Caller ID should be displayed when the incoming DID number is transferred to an outgoing trunk.
- Caller ID Display:
0 = Not Displayed (default)
1 = Displayed
- ➔ **35-02-14 SMDR Output Options – Date**
Determine whether or not the date should be displayed on SMDR reports. *This option must be set to 0 if the trunk name is set to be displayed in Program 35-02-03.*
- Date Display:
0 = Not Displayed (default)
1 = Displayed
- ➔ **35-02-15 SMDR Output Options – CLI/DID Number Switching**
Determine if the Caller ID number 0 (default) or DID calling number (1) should be displayed in the SMDR output.
- ➔ **35-02-16 SMDR Output Options – Trunk Name or Received Dialed Number**
Determine how the SMDR should print incoming calls on ANI/DNIS or DID trunks. If set to 1, ANI/DNIS trunks can print DNIS digits. For DID trunks, if the received number is not defined in Program 22-11-01, a number is not printed. If set to 0 (default) trunk names are printed instead (as assigned in Program 14-01-01).
- ➔ **35-02-17 SMDR Output Options – Print Account Code or Caller Name of Incoming Call**
Determine whether the Account Code (0) or Caller ID name (1) should print in the SMDR record. By default, the Account Code is displayed.
-  *Program 35-01-08 must be set to 0 for this entry to be followed.*
- ➔ **35-02-18 SMDR Output Options – Print Mode for Caller Name of Incoming Call**
Select whether to display up to 16 characters of the Caller Name on the same line as the call record (0) or if a line feed should be added and up to 24 characters of the Caller Name are displayed on the following line (1). If the line feed option is selected, the Caller Name is displayed on the next line as: NEXT Caller Name. The default is 0.
- This setting works regardless of the setting in Program 35-02-15.
-  *With this option set to 1, if your communications program (such as HyperTerminal) has the line wrap option enabled in the ASCII setup, an additional line break may appear above the Caller name line.*
-
-

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- ➔ **35-03-01 SMDR Port Assignment for Trunk Group**
Assign the SMDR port for each trunk group. This is the SMDR port where the incoming call information should be sent.

Trunk Group: 1~100 (default is 1)
SMDR Port: 1~8

- ➔ **35-04-01 SMDR Port Assignment for Department Groups**
Assign the SMDR port for each department group. This is the SMDR port where the outgoing call information should be sent.

Department Group: 01~64 (default is 1)
SMDR Port: 1~8

CTA Adapter:

- ➔ **15-02-19 Multiline Telephone Basic Data Setup – CTA Data Communication Mode**
Change the system programming to match the CTA adapter dip switch settings.

- ➔ **15-02-20 Multiline Telephone Basic Data Setup – Baud Rate for CTA Port**
Select the baud rate to be used by the CTA Adapter.

0 = 4800 bps
1 = 9600 bps
2 = 19200 bps (default)

LAN Connection:

- ➔ **10-12-01 CPUII Network Setup – IP Address**
When using an IP connection, set up the IP address used to connect from the CallAnalyst PC to the system (default: 172.16.0.10).

- ➔ **10-20-01 LAN Setup for External Equipment – TC Port**
When using an IP connection, define the TCP port used for communicating to the CallAnalyst (External Device 5 = SMDR, Entries: 0~65535). This entry must match the entry made in the CDM setup with the CallAnalyst program. Default is 0.

- ➔ **10-20-03 LAN Setup for External Equipment – Keep Alive Time**
Define Keep Alive Time

Serial Connection:

- ➔ **10-21-02 CPUII Hardware Setup – Baud Rate for COM Port**
If the SMDR connection is made using the COM port on the CPUII()-U10, define the baud rate.

0 = 4800 bps
1 = 9600 bps
2 = 19200 bps (default)
3 = 38400 bps
4 = 56000 bps (Set in PCPRO only)
5 = 115200 bps (Set in PCPRO only)



Operation

Refer to [Central Office Calls, Answering on page 2-187](#) and [Central Office Calls, Placing on page 2-195](#) features for detailed operations for placing or answering calls.

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Facsimile CO Branch Connection

Description

The Electra Elite IPK II system provides branch connection of locally provided facsimile machines to CO/PBX lines. Additional dedicated CO/PBX lines are not required for a facsimile to operate. The facsimile shares the last CO/PBX line on the COI(4)-U(), or COIB(4)-U() ETU through the Main Distribution Frame (MDF) where the CO line is connected from Telco.

Conditions

- The Electra Elite IPK II systems require a COI(4)-U(), or COIB(4)-U() ETU to connect a facsimile in branch to a direct CO/PBX line.
- Connection of the facsimile machine does not require extra system ports.
- The Electra Elite IPK II systems cannot distinguish between an incoming facsimile call and a CO/PBX call. Ringing assignments should be turned off for fax lines.
- A facsimile call terminating at the CO/PBX line where the Fax Branch Unit is connected is automatically answered by the Fax Machine. Delayed Ringing Line assignment to stations for Fax Branch Line are Recommended.
- When the facsimile is in use, the Fax Branch CO/PBX line can be used as an outside line.
- Code restriction does no apply to outgoing calls from the Fax machine.

Default Settings

None

System Availability

Terminals

None

F

Required Component(s)

COI(4)-U() ETU or COIB(4)-U() ETU

Related Features

None

Programming

- ➔ **14-02-14 Analog Trunk Data Setup – Loop Start/Ground Start**
0 = Loop Start
1 = Ground Start
- ➔ **14-05-01 Trunk Group**
Trunk Ports = 1~200
Trunk Groups = 0~100
- ➔ **22-01-04 System Options for Incoming Calls – DIL No Answer Recall Time**
Time – 0~64800 sec. (default is 0)
- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Maximum: eight digits (up to 32 extensions per group)
Default: Extensions 101~108 (first 8 ports) ring for incoming Ring Group 1 calls. No other extensions ring for incoming Ring Group 1 calls.
- ➔ **22-05-01 Incoming Trunk Ring Group Assignment**
0 (No Setting)

1~8 or 001~100 Incoming Group (Default is 1)
102 (In-Skin/External Voice Mail)
- ➔ **22-08-01 DIL/IRG No Answer Destination**
0 = No Setting
001~100 = Incoming Group (Default is 1)
102 = In-Skin/External Voice Mail
103 = (Centralized Voice Mail)

Operation

None

Flash

Description

Flash allows an extension user to access certain CO and PBX features by interrupting the trunk loop current. Flash lets an extension user take full advantage of whatever features the connected Telco or PBX offers. You must set the Flash parameters for compatibility with the connected Telco or PBX.

Conditions

The system does not provide a ground flash.

Default Setting

Enabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Drop Key

PBX Compatibility

Voice Mail Message Indication on Line Keys

Programming

- ➔ **11-12-42 Service Code Setup (for Service Access) – Flash on Trunk Lines**
Assign Service Code, default = #3.
- ➔ **14-02-03 Analog Trunk Data Setup – Flash Type**
Always set this to 0 for Open Loop Flash.
- ➔ **14-02-04 Analog Trunk Data Setup – Hooking Type**
Set to 0 for Timed Flash (hooking) or set to 1 for Disconnect (cut). Default is 0. The user can press the Flash key while on a call to implements this.
- ➔ **15-02-05 Multiline Telephone Basic Data Setup – Transfer Key Operation Mode**
For the Cordless Lite/Cordless Lite II telephone user to use the flash function, this option must be set to 2. This changes the Transfer key to a Flash key.
- ➔ **15-03-04 Single Line Telephone Basic Data Setup – Flashing**
Select Yes (1)/No (0) for the ability to hookflash for system feature access. Default is 1.
- ➔ **81-10-07 COI Initial Data Setup – Hookflash Time Selection 1**
Set the Flash duration to 0 (20 ms)~15(5 seconds) for analog trunk [COI()-U() ETU] circuits in PRG 81-10-07. Default is 9 (600ms)
- ➔ **81-10-08 COI Initial Data Setup – Hookflash Time Selection 2**
Set the open loop disconnect duration to 0 (20 ms)~15(5 seconds) for analog trunk [COI()-U() ETU] circuits PRG 81-10-08. Default is 14 (3 seconds)

Operation

To flash the trunk you are on:

From a Multiline Terminal

1. Press **Recall**.

From a Single Line Telephone

1. Hookflash.
2. Dial **#3**.

Flexible System Numbering

Enhancements

Version 2000 and higher supports station numbers to be assigned by the 10s group for 4-digit station numbers, 100s group for 5-digit, 1,000s group for 6-digit station numbers, and 10,000s group for 7-digit station numbers.

Description

Flexible System Numbering lets you reassign the system port-to-extension assignments. This allows an employee to retain their extension number if they move to a different office. Also, factory technicians can make the following comprehensive changes to your system number plan:

- Set the number of digits in internal (Intercom) functions. For example, extension numbers can have up to eight digits.
- Change your system Service Code numbers.
- Assign single digit access to selected Service Codes.

Talk to your sales representative to find out if this program is available to you.

You can also use Flexible System Numbering to change the system Trunk Group Routing code. Although the default code of 9 is suitable for most applications, you can alter the code if needed.

The system provides a completely flexible system numbering plan. Refer to the chart below and the Electra Elite IPK II Programming Manual for more details.

Flexible System Numbering	
Program	Description
11-01-01 System Numbering	Set the system internal (Intercom) numbering plan. The numbering plan includes the digits an extension user must dial to access features and other extensions.
11-09-01 Trunk Access Code	Assign the single-digit trunk access code (normally 9). This is the code users dial to access Automatic Route Selection or Trunk Group Routing.

Flexible System Numbering (Continued)	
Program	Description
11-20-01 Dial Extension Analyze Table (Version 2000 or higher required)	Use tables 01 ~ 128 to assign the digits to be dialed using the Dial Extension Analyze Tables. These tables are used when Program 11-01-01 is set to option 9 = Dial Extension Analyze. (Up to eight digits can be assigned and the valid entries are: 0, 1 ~ 9, #, *,)
11-20-02 Dial Extension Analyze Table (Version 2000 or higher required)	Assign the Type of Dial for the Dial Extension Analyze Table from Program 11-20-01. (Svc Code, Intercom, Operator, or F-Route)
11-10 Service Code Setup (for System Administrator) 11-11 Service Code Setup (for Setup/Entry Operation) 11-12 Service Code Setup (for Service Access) 11-13 Service Code Setup (for ACD) 11-14 Service Code Setup (for Hotel) 11-15 Service Code Setup, Administrative (for Special Access)	Customize the Service Codes.
11-16 Single Digit Service Code Setup	Assign the Single Digit Service Code. This is the post-dialing code a user can dial after placing an Intercom call to a co-worker.

Conditions

- Programming follows a telephone extension number, not the port number in most cases. If you relocate a telephone, you may need to change additional programming.
- If the extension numbering plan is changed from 3xx to 2xx, and you want to consecutively press two DSS keys without toggling the hookswitch, Program 11-16-05 must be removed. If not, press the second DSS key to actually camp you on to the first extension.
- Since making changes in Program 11-01 does not automatically make any changes in any other program, changing the number plan after the system is in operation may cause problems in the following programs:

PRG 11-01 Type 2 (Extension Number)				PRG 11-01 Type 1 (Service Codes)		
11-02	11-08	15-12	22-11	11-10	11-14	21-11
11-04	11-17	16-01-01	25-06	11-11	11-15	30-03
11-06	15-01-01	15-14	30-03	11-12	15-07	
11-07	15-07	21-11		11-13	15-14	

- Any feature which requires dialing a code or extension number can be affected.
- When the system searches the Dial Extension Analyze Table (PRG 11-20-01), it uses prefix searching, giving the lower table number the higher priority. For example, if the user programs 211 in table 1 and 2113 in table 2, then dials 2113, the system selects table 1.

Example for 310X**Example for 3100X**

10s Group (4-digit)	100s Group (5-digit)
11-01-01 = Dial 3 31 Digit 4 = (9)Dial Extension Analyze Table	11-01-01 = Dial 3 31 Digit 7 = (9)Dial Extension Analyze Table
11-20-01 Table 1 = Dial 310	11-20-01 Table 1 = Dial 3100
11-20-02 Table 1 = Intercom	11-20-02 Table 1 = Intercom

Example for 31000X**Example for 310000X**

1000s Group (6-digit)	10,000s Group (7-digit)
11-01-01 = Dial 3 31 Digit 4 = (9)Dial Extension Analyze Table	11-01-01 = Dial 3 31 Digit 7 = (9)Dial Extension Analyze Table
11-20-01 Table 1 = Dial 31000	11-20-01 Table 1 = Dial 310000
11-20-02 Table 1 = Intercom	11-20-02 Table 1 = Intercom

- Program 11-01-01 9: System Numbering – Dial Extension Analyze is not supported in **Version 1600**.
- Program 11-20-01: Dial Extension Analyze Table – Dial and Program 11-20-02: Dial Extension Analyze Table – Type of Dials are not supported in **Version 1600**.

Default Setting

Extensions and Virtuals are numbered in the following order:

Program 11-02-01 and Program 11-04-01

- Physical Extensions:
 - Extn Port 1 = 101 ~ Extn Port 99 = 199
 - Extn Port 100 = 3101 ~ Extn Port 256 = 3257

- Virtual Extensions/CAR Keys:
 - VE Port 1 = 201 ~ VE Port 99 = 299
 - VE Port 100 = 3301 ~ VE Port 256 = 3457

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

None

Programming

- ➔ **11-01-01 System Numbering**
Customize the system internal (Intercom) numbering plan.
- ➔ **11-02-01 Extension Numbering**
Assign extension numbers to extension ports. The telephone programming identity follows the port number – not the extension number.
- ➔ **11-09-01 Trunk Access Code**
Assign the single-digit trunk access code (normally 9) for ARS/Trunk Group Routing.
- ➔ **11-09-02 Trunk Access Code – 2nd Trunk Route Access Code**
Assign the Service Code set up in PRG 11-01-01 for Alternate Trunk Route Access.
- ➔ **11-10 Service Code Setup (for System Administrator)**
Refer to [Table 2-20 Service Code Setup \(for System Administrator\)](#), Program 11-10 on page 2-451.
- ➔ **11-11 Service Code Setup (for Setup/Entry Operation)**
Refer to [Table 2-21 Service Code Setup \(for Setup/Entry Operation\)](#), Program 11-11 on page 2-452.

- **11-12 Service Code Setup (for Service Access)**
Refer to [Table 2-22 Service Code Setup \(for Service Access\)](#), Program 11-12 on page 2-455.
- **11-13 Service Code Setup (for ACD)**
Refer to [Table 2-23 Service Code Setup \(for ACD\)](#), Program 11-13 on page 2-457.
- **11-14 Service Code Setup (for Hotel)**
Refer to [Table 2-24 Service Code Setup \(for Hotel\)](#), Program 11-14 on page 2-458.
- **11-15 Service Code Setup, Administrative (for Special Access)**
Refer to [Table 2-25 Service Code Setup, Administrative \(for Special Access\)](#), Program 11-15 on page 2-459.

Table 2-20 Service Code Setup (for System Administrator), Program 11-10

Item No.	Item	Terminals	Default	Related Program
01	Night Mode Switching	MLT, SLT	718	12-xx 20-07-01
02	--- Not Used ---			
03	Setting the System Time	MLT	728	
04	Storing Common Speed Dialing Numbers	MLT	753	
05	Storing Group Speed Dialing Numbers	MLT	754	
06	Setting the Automatic Transfer for Each Trunk Line	MLT	733	24-04-01
07	Canceling the Automatic Transfer for Each Trunk Line	MLT	734	24-04-01
08	Setting the Destination for Automatic Trunk Transfer	MLT	735	24-04-01
09	--- Not Used ---			
10	--- Not Used ---			
11	--- Not Used ---			
12	Night Mode Switching for Other Group	MLT	618	12-xx 20-07-01
13	--- Not Used ---			
14	--- Not Used ---			
15	--- Not Used ---			
16	Leaving Message Waiting (Requires CPU to be licensed for Hotel/Motel)	MLT	626	11-11-09
17	Dial Block by Supervisor	MLT	601	90-19
18	Off-Premise Call Forward by Door Box	MLT	722	13-05
19	--- Not Used ---			

Table 2-20 Service Code Setup (for System Administrator), Program 11-10 (Continued)

Item No.	Item	Terminals	Default	Related Program
20	VRS - Record/Erase Message	MLT	616	20-07-13
21	VRS - General Message Playback	MLT	611	20-07-14
22	VRS - Record or Erase General Message	MLT	612	20-07-15
23	SMDR - Extension Accumulated Printout Code	MLT	621	20-07-18
24	SMDR - Group Accumulated Printout Code	MLT	622	20-07-19
25	Account Code Accumulated Printout Code	MLT	623	20-07-20
26	Forced Trunk Disconnect	MLT, SLT	*3	20-07-11
27	Trunk Port Disable for Outgoing Calls	MLT	645	20-07-12
28	--- Not Used ---			
29	--- Not Used ---			
30	Register DECTPP		No Setting	
31	Delete DECTPP		No Setting	
32	Set Private Call Refuse		No Setting	
33	Entry Caller ID Refuse		No Setting	
34	Set Caller ID Refuse		No Setting	
35	Dial-In Mode Switching		No Setting	
36	Change the Guidance Message Number on Voice Mail Auto Attendant		No Setting	

Table 2-21 Service Code Setup (for Setup/Entry Operation), Program 11-11

Item No.	Item	Terminals	Default	Related Program
01	Call Forward – All	MLT, SLT	741	
02	Call Forward – Busy	MLT, SLT	742	
03	Call Forward – No Answer	MLT, SLT	743	
04	Call Forward – Busy/No Answer	MLT, SLT	744	
05	Call Forward – Both Ring	MLT, SLT	745	
06	--- Not Used ---			
07	Call Forwarding – Follow-Me	MLT, SLT	746	
08	Do Not Disturb		747	

Table 2-21 Service Code Setup (for Setup/Entry Operation), Program 11-11 (Continued)

Item No.	Item	Terminals	Default	Related Program
09	Answer Message Waiting	MLT, SLT	*0	11-10-16
10	Cancel All Messages Waiting	MLT, SLT	773	
11	Cancel Message Waiting	MLT, SLT	771	
12	Alarm Clock	MLT, SLT	727	20-01-06
13	Display Language Selection for Multiline Terminal	MLT	678	15-02
14	Text Message Setting	MLT	No Setting	
15	Enable Handsfree Incoming Intercom Calls	MLT	721	20-09-05 20-02-12
16	Force Ringing of Incoming Intercom Calls	MLT	723	20-09-05 20-02-12
17	Programmable Function Key Programming (Dialing 751 Service Code)	MLT	751	15-07 11-11-38
18	BGM On/Off	MLT	725	
19	Key Touch Tone On/Off	MLT	724	
20	Change Incoming CO and ICM Ring Tones	MLT	720	15-02
21	Check Incoming Ring Tones	MLT	711	
22	Extension Name Programming	MLT	700	15-01
23	Second Call for DID/DISA/DIL	MLT	679	
24	Change Station Class of Service Allows an extension user to change the COS of another extension. Must be allowed in Program 20-13-28.	MLT	677	20-13-28
25	Automatic Transfer Setup for Each Extension Group	MLT, SLT	602	20-11-17 24-05
26	Automatic Transfer Cancellation for Each Extension Group	MLT, SLT	603	
27	Destination of Automatic Transfer Each Extension Group	MLT	604	20-11-17 24-05
28	Delayed Transfer for Every Extension Group	MLT, SLT	605	20-11-17 24-05 24-02-08
29	Delayed Transfer Cancellation for Each Extension Group	MLT, SLT	606	20-11-17
30	DND Setup for Each Extension Group	MLT, SLT	607	
31	DND Cancellation for Each Extension Group	MLT, SLT	608	
32	--- Not Used ---			

Table 2-21 Service Code Setup (for Setup/Entry Operation), Program 11-11 (Continued)

Item No.	Item	Terminals	Default	Related Program
33	Dial Block	MLT, SLT	600	
34	Temporary Toll Restriction Override	MLT, SLT	775	21-07
35	Pilot Group Withdrawing	MLT, SLT	650	
36	Toll Restriction Override	MLT, SLT	663	21-14
37	Ring Volume Set	MLT	729	
38	Programmable Function Key Programming (Dialing 852 Service Code)	MLT	752	15-07 11-11-17
39	Station Speed Dial Number Entry	MLT	755	
40	--- Not Used ---			
41	Tandem Ringing	MLT, SLT	No Setting	15-07 30-03
42	Electra Elite IPK II Wireless Transferring When Out of Range Customize the service code to be used when setting a Electra Elite IPK II Wireless telephone to transfer calls when out of range.	Electra Elite IPK II Wireless	689	
43	Headset Mode Switching	MLT, SLT	688	
44	Auto Attendant	MLT	No Setting	
45	Set/Cancel Call Forward All (Split)	MLT, SLT	No Setting	
46	Set/Cancel Call Forward Busy (Split)	MLT, SLT	No Setting	
47	Set/Cancel Call Forward No Answer (Split)	MLT, SLT	No Setting	
48	Set/Cancel Call Forward Busy No Answer (Split)	MLT, SLT	No Setting	
49	Set/Cancel Call Forward Both Ring (Split)	MLT, SLT	No Setting	
50	Set Message Waiting Indication (Version 1500 or higher required)	MLT, SLT	Up to 8 digits	
51	Cancel Message Waiting Indication (Version 1500 or higher required)	MLT, SLT	Up to 8 digits	
52	Set/Cancel Call Forward All Destination (No Split)	MLT, SLT	790	
53	Set/Cancel Call Forward Busy Destination (No Split)	MLT, SLT	791	
54	Set/Cancel Call Forward No Answer Destination (No Split)	MLT, SLT	792	
55	Call Forward Busy No Answer Destination (No Split)	MLT, SLT	793	
56	--- Not Used ---			
57	Set Do Not Call Table		No Setting	

Table 2-21 Service Code Setup (for Setup/Entry Operation), Program 11-11 (Continued)

Item No.	Item	Terminals	Default	Related Program
58	Call Forward with Personal Greeting	MLT, SLT	713	

Table 2-22 Service Code Setup (for Service Access), Program 11-12

Item No.	Item	Terminals	Default	Related Program
01	Bypass Call Activating Call Forwarding/Do Not Disturb Override. This code is available only if you disable the voice mail Single Digit dialing code in Program 11-16-09.	MLT, SLT	707	11-16-09
02	Conference	MLT, SLT	#1	
03	Override (Off-Hook Signaling)	MLT, SLT	709	
04	Set Camp-On	MLT, SLT	750	
05	Cancel Camp-On	MLT, SLT	770	
06	Switching of Voice Call and Signal Call	MLT, SLT	712	
07	Step Call	MLT, SLT	708	
08	Barge-In	MLT, SLT	710	
09	Change to STG (Department Group) All Ring	MLT, SLT	No Setting	16-02
10	Station Speed Dialing	MLT, SLT	#2	
11	Group Speed Dialing	MLT, SLT	#4	
12	Last Number Dial	MLT, SLT	#5	
13	Saved Number Dial	MLT, SLT	715	
14	Trunk Group Access	MLT, SLT	704	
15	Specified Trunk Access	MLT, SLT	#9	
16	Trunk Access Via Networking	MLT	No Setting	
17	Clear Last Number Dialing Data	MLT, SLT	776	
18	Clear Saved Number Dialing Data	MLT, SLT	785	
19	Internal Group Paging	MLT, SLT	701	31-01-01
20	External Paging	MLT, SLT	703	
21	Meet-Me Answer to Specified Internal Paging Group	MLT, SLT	764	
22	Meet-Me Answer to External Paging	MLT, SLT	765	

Table 2-22 Service Code Setup (for Service Access), Program 11-12 (Continued)

Item No.	Item	Terminals	Default	Related Program
23	Meet-Me Answer in Same Paging Group	MLT, SLT	763	
24	Combined Paging	MLT, SLT	*1	31-07
25	Direct Call Pickup - Own Group	MLT, SLT	756	
26	Call Pickup for Specified Group	MLT, SLT	768	
27	Call Pickup	MLT, SLT	* #	
28	Call Pickup for Another Group	MLT, SLT	769	
29	Direct Extension Call Pickup	MLT, SLT	* *	
30	Specified Trunk Answer	MLT, SLT	672	
31	Park Hold	MLT, SLT	#6	24-03
32	Answer for Park Hold	MLT, SLT	*6	24-03
33	Group Hold	MLT, SLT	732	
34	Answer for Group Hold	MLT, SLT	762	
35	Station Park Hold	MLT, SLT	757	
36	Door Box Access	MLT, SLT	702	
37	Common Canceling Service Code	MLT, SLT	620	
38	General Purpose Indication	MLT	783	
39	--- Not Used ---			
40	Station Speed Dialing	MLT, SLT	#7	
41	Voice Over	MLT	690	11-16-08
42	Flash on Trunk lines	SLT	#3	
43	Answer No-Ring Line (Universal Answer)	SLT	#0	14-05 14-06
44	Callback Test for SLT	SLT	799	
45	Enabled On Hook When Holding (SLT)	SLT	749	15-03-07
46	Answer On Hook When Holding (SLT)	SLT	759	15-03-08
47	Call Waiting Answer/Split Answer Splitting (switching) between calls	MLT,SLT	794	11-12-03
48	Account Code	SLT	# #	
49	--- Not Used ---			
50	General Purpose Relay	MLT	780	

Table 2-22 Service Code Setup (for Service Access), Program 11-12 (Continued)

Item No.	Item	Terminals	Default	Related Program
51	VM Access (In-Mail and VMS)	MLT, SLT	*8	
52	Live Monitoring (In-Mail)	MLT	No Setting	
53	Live Recording at SLT	SLT	654	
54	VRS Routing for ANI/DNIS Use when setting up ANI/DNIS Routing to the VRS Automated Attendant. Using the Transfer feature, this also allows a call to be transferred to the VRS.		782	
55	--- Not Used ---			
56	E911 Alarm Shut Off Enter the Service Code that an extension user can dial to shut off the E911 Alarm Ring.	MLT, SLT	786	20-08-16 21-01-13
57	Tandem Trunking	MLT, SLT	#8	
58	Transfer Into Conference Assign the Service Code users dial to Transfer a call to a Conference call.	MLT, SLT	624	20-13-10 20-13-15 20-13-16
59	Trunk Drop Operation for SLT	SLT	No Setting	

Table 2-23 Service Code Setup (for ACD), Program 11-13

Item No.	Item	Terminals	Default	Related Program
01	ACD Log In/Log Out (for KTS)	MLT, SLT	* 5	
02	ACD Log Out (for SLT)	SLT	655	
03	Set ACD Wrap-Up Time (for SLT)	SLT	656	
04	Cancel ACD Wrap-Up Time (for SLT)	SLT	657	
05	Set ACD Off Duty (for SLT)	SLT	658	
06	Cancel ACD Off Duty (for SLT)	SLT	659	
07	--- Not Used ---			
08	Agent ID Code Login Allows an AIC Agent to log into a group.	MLT	No Setting	
09	Agent ID Code Logout Allows an AIC Agent to log out of a group.	MLT	No Setting	
10	ACD Agent Login by Supervisor Allows an ACD Supervisor to log into a group.	MLT	667	

Table 2-23 Service Code Setup (for ACD), Program 11-13 (Continued)

Item No.	Item	Terminals	Default	Related Program
11	ACD Agent Logout by Supervisor Allows an ACD Supervisor to log out of a group.	MLT	668	
12	Change Agent ACD Group by Supervisor When using service code 669 to change an agent ACD group, the supervisor must enter a 2-digit number for the group. For example, to change to ACD group 4, the entry would be 669 04.	MLT	669	
13	ACD Agent Changing Own ACD Group Using this service code, an ACD Agent can reassign themselves to another ACD Group.	MLT	670	

Table 2-24 Service Code Setup (for Hotel), Program 11-14

Item No.	Item	Terminals	Default	Related Program
01	Set DND for Own Extension	MLT, SLT	627	
02	Cancel DND for Own Extension	MLT, SLT	628	
03	Set DND for Other Extension	MLT, SLT	629	
04	Cancel DND for Other Extension	MLT, SLT	630	
05	Set Wake Up Call for Own Extension	MLT, SLT	631	
06	Cancel Wake Up Call for Own Extension	MLT, SLT	632	
07	Set Wake Up Call for Other Extension	MLT, SLT	633	
08	Cancel Wake Up Call for Other Extension	MLT, SLT	634	
09	Set Room to Room Call Restriction	MLT, SLT	635	
10	Cancel Room to Room Call Restriction (Hotel)	MLT, SLT	636	
11	Change Toll Restriction Class for Other Extension	MLT, SLT	637	
12	Check-In	MLT, SLT	638	
13	Check-Out	MLT, SLT	639	
14	Room Status Change for Own Extension	MLT, SLT	640	
15	Room Status Change for Other Extension	MLT, SLT	641	
16	Room Status Output	MLT, SLT	642	
17	Hotel Room Monitor	MLT, SLT	675	
18	Set Hotel PMS Code Restriction	MLT, SLT	666	

Table 2-25 Service Code Setup, Administrative (for Special Access), Program 11-15

Item No.	Item	Terminals	Default	Related Program
01	Remote Maintenance		730	
02	ACD Access in Dial-In Conversion Table		760	22-04 22-11
03	Backup Data Save This option saves the user's soft key settings (extension programmed Call Forwards, DND, etc.). This feature should be used before upgrading the system software.	MLT	# * # 9	
04	--- Not Used ---			
05	System Programming Mode, Log-On	MLT	# * # *	11-01-01
06	Wake on LAN to APSU Unit	MLT	No Setting	10-22
07	--- Not Used ---			
08	--- Not Used ---			
09	Transfer to Incoming Ring Group		No Setting	
10	--- Not Used ---			
11	Ethernet Port Reset	MLT, SLT	No Setting	
12	Extension Data Swap	MLT, SLT	No Setting	
13	--- Not Used ---			
14	Modem Access	MLT, SLT	740	

- ➔ **11-16-01 Single Digit Service Code Setup – Step Call**
Assign the Single Digit (post-dialing) Service Code.
- ➔ **11-16-02 One-Digit Service Code Setup – Barge-In**
Set up Item 02 for single digit Barge-In. For example, you can assign Item 02 to use digit 5 for Barge-In. This allows you to program a function key with an extension number plus the Barge-In code (i.e., 5). This allows one-touch access to the Barge-In feature for extension.
- ➔ **11-16-03 Single Digit Service Code Setup – Switching of Voice/Signal Call**
(Default: 1).
- ➔ **11-16-04 Single Digit Service Code Setup – Intercom Off-Hook Signaling**
Assign a one-digit service code to be used for Off-Hook Signaling.
- ➔ **11-16-05 Single Digit Service Code Setup – Camp-On**
Customize the 1-digit Service Code used for setting Camp-On.
(default: #)

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- ➔ **11-16-06 Single Digit Service Code Setup – DND/Call Forward Override Bypass**
Customize the 1-digit Service Code used for DND/Call Forward Override.
 - ➔ **11-16-07 Single Digit Service Code Setup – Message Waiting**
(Default: 0).
 - ➔ **11-16-08 Single Digit Service Code Setup – Voice Over.**
Service code used for the Voice Over feature (default: 6).
 - ➔ **11-16-09 Single Digit Service Code Setup – Access to Voice Mail**
(Default: 8).
 - ➔ **11-16-10 Single Digit Service Code Setup – (Department) STG All Ring Mode**
Assign the Single Digit (post-dialing) Service Code for All Member Ring.
 - ➔ **11-16-11 Single Digit Service Code Setup – Station Park Hold**
Customize the one-digit service code used when placing a call in Personal Park (default: No setting).
 - ➔ **11-20-01 Dial Extension Analyze Table – Dial Digits**
Use tables 01 ~ 128 to assign the digits to be dialed using the Dial Extension Analyze Tables. These tables are used when Program 11-01-01 is set to option 9 = Dial Extension Analyze. Up to eight digits can be assigned and the valid entries are: (0, 1 ~ 9, #, *)
 - ➔ **11-20-02 Dial Extension Analyze Table – Type of Dials**
Assign the Type of Dial for the Extension Analyze Table from Program 11-20-01.

Operation

None

Flexible Timeouts

Description

The Flexible Timeouts feature provides a variety of timers in the Resident System Program to allow the system to operate without initial programming. The system timers can be changed to meet customer needs according to the system application requirements.

A Timer Class allows terminals and trunks to have different timers for the same feature. There are 16 timer Classes (0~15). The following table shows the Programs that are used depending on the Timer Class used:

Timer Class 0	Timer Class 1~15	Title	Comment
20-01-08	20-31-01	Trunk Queuing Callback Time	Trunk Queuing callback rings an extension for this time. Station Timer Class is referred by the station that sets trunk queuing.
20-01-09	20-31-02	Callback / Trunk Queuing Cancel Time	The system cancels an extension Callback or Trunk Queuing request after this time. Station Timer Class is referred by the station that sets an extension Callback or Trunk Queuing.
20-04-03	20-31-03	CAR/SIE/Virtual Extension Delay Interval	If CAR/VE is set for Delayed Ringing (Program 15-11-01), ring the covering extension after this time. Station Timer Class is referred by the station assigned to CAR/VE.
21-01-02	20-31-04	Intercom Interdigits Time	When placing Intercom calls, users must dial each digit during this time. Station Timer Class is referred by stations. Trunk Timer Class is referred by DID/Automatic Answer Trunk/E&M trunks.
21-01-03	20-31-05	Trunk Interdigits Time	When placing CO calls, users must dial each digit during this time. Station Timer Class is referred by stations. Trunk Timer Class is referred by DID/Automatic Answer Trunk/E&M trunks.
21-01-09	20-31-06	Hotline Time Start Time	A Ringdown extension automatically calls its programmed destination after this time. Station Timer Class is referred by the station which sets Hotline.
22-01-03	20-31-07	Ring No Answer Alarm Time	If a trunk rings a key telephone longer than this time, the system changes the ring cadence. This indicates to the user that the call has been ringing too long. Trunk Timer Class is referred by the trunk.

Timer Class 0	Timer Class 1~15	Title	Comment
22-01-04	20-31-08	DIL No Answer Recall Time	A DIL that rings its programmed destination longer than this time diverts to the DIL No Answer Ring Group (Program 22-08-01). Trunk Timer Class is referred by the trunk.
22-01-06	20-31-09	DID Ring-No-Answer Time	In systems with DID Ring No Answer Intercept, this time sets the Ring No Answer time. This time is how long a DID call rings the destination extension before rerouting to the intercept ring group. Trunk Timer Class is referred by DID trunk.
24-01-01	20-31-10	Hold Recall Time (Non Exclusive Hold)	A call on Hold recalls the extension that placed it on Hold after this time. Station Timer Class is referred by held call.
24-01-02	20-31-11	Hold Recall Callback Time (Non Exclusive Hold)	A Hold recall rings an extension for this time. Station Timer Class is referred by held call.
24-01-03	20-31-12	Exclusive Hold Recall Time	A call on Hold recalls the extension that placed it on Hold after this time. Station Timer Class is referred by held call.
24-01-04	20-31-13	Exclusive Hold Recall Callback Time	An Exclusive Hold Recall rings an extension for this time. If not picked up, the call goes back on Non exclusive Hold. Station Timer Class is referred by held call.
24-01-06	20-31-14	Park Hold Time – Normal	A call left parked longer than this time recalls the extension that initially parked it. Trunk or Station Timer Class is referred by held call.
24-02-03	20-31-15	Delayed Call Forwarding Time	If activated at an extension, No Answer Call Forwarding occurs after this time. Station Timer Class is referred by the station sets No Answer Call Forward.
24-02-04	20-31-16	Transfer Recall Time	A transferred call recalls to the extension that initially transferred it after this time. Station Timer Class is referred by a transferred call.
25-07-02	20-31-17	VRS/DISA No Answer Time	After this time expires, the call follows the programmed Ring No Answer routing (Program 25-03 and 25-04-01). Trunk Timer Class is referred.
25-07-03	20-31-18	Disconnect after VRS/DISA Re-transfer to IRG	Disconnect after re-transfer to Incoming Ring Group. Trunk Timer Class is referred.
25-07-07	20-31-19	Long Conversation Warning Tone Time	Determine the time trunk-to-trunk conversation can talk before the Long Conversation tone is heard. Trunk Timer Class is referred.

Timer Class 0	Timer Class 1~15	Title	Comment
25-07-08	20-31-20	Long Conversation Disconnect	This determines the time the system waits before disconnecting a trunk-to-trunk conversation call after the Long Conversation tone is heard. Trunk Timer Class is referred.
25-07-09	20-31-21	DISA Internal Paging Time	This is the maximum time for an Internal Page placed by a DISA caller. If the Page continues longer than this time, the system terminates the DISA call. Trunk Timer Class is referred.
25-07-10	20-31-22	DISA External Paging Time	This is the maximum time for an External Page placed by a DISA caller. If the Page continues longer than this time, the system terminates the DISA call. Trunk Timer Class is referred.
31-01-02	20-31-23	Page Announcement Duration	This timer sets the maximum time for External Page announcements. Station or Trunk Timer Class is referred by the caller makes announcement.

Conditions

- Timer Classes are used for CAR/VE also.
- When Timer Class is set to 0 it uses the system-wide timers.
- All stations and trunks are assigned to Timer Class 0 at default.
- Both system-wide timers (Timer Class 0) and Timer Class timers (Timer Class 1~15) can be used in the same system.

Default Setting

Timer Class set to 0 for all trunks and extensions.

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

None

Programming

- **20-29-01 Timer Class for Extensions**
Assign the timer class (0~15) to each extension for each Night mode. This entry includes virtual extension number.
- **20-30-01 Timer Class for Trunks**
Assign the timer class (0~15) to each trunk for each Night mode.
- **20-31-01 through 20-31-23 Timer Class Timer Assignment**
Assign times. These timers are referred when a class is set any number from 1~16 in PRG 20-29-01/PRG 20-30-01. When the timer class is set to 0 in PRG 20-29-01/PRG 20-30-01, current timer (system-wide basis) is referred refer to the table above for more detailed information about the programs.

Operation

Please refer to the feature for the operation.

Forced Trunk Disconnect

Description

Forced Trunk Disconnect allows an extension user to disconnect (release) another extension active outside call. The user can then place a call on the released trunk. Forced Trunk Disconnect lets a user access a busy trunk in an emergency, when no other trunks are available. Maintenance technicians can also use Forced Trunk Disconnect to release a trunk on which there is no conversation. This can happen if a trunk does not properly disconnect when the outside party hangs up.

Caution

Forced Trunk Disconnect abruptly terminates the active call on the line. Only use this feature in an emergency and when no other lines are available.

Conditions

This feature only works on an analog trunk. ISDN trunks do not have Forced Trunk Disconnect available.

Default Setting

- COS 15 = Enabled
- COS 1~14 = Disabled

System Availability

Terminals

All Terminals

Required Component(s)

Analog Trunks

Related Features

Central Office Calls, Placing




Programming

- **11-10-26 Service Code Setup (for System Administrator) – Forced Trunk Disconnect**
Assign the Service Code.
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- **20-07-11 Class of Service Options (Administrator Level) – Forced Trunk Disconnect (analog trunk only)**
Enable (1)/Disable (0) an extension user ability to use Forced Trunk Disconnect.
- **21-01-18 System Options for Outgoing Calls – Reset Dial After Failure of Trunk Access (Version 1500 or higher required)**
Enable (1)/Disable (0) the ability to continue to dial codes or extensions after receiving Trunk Busy. This must be set to 1 for this feature to work.




Operation

To disconnect a busy trunk:

Multiline Terminal

1. Press line key for trunk.
 - OR -
 - Dial trunk access code (**#9** + trunk number).
 -  *You hear busy tone. Trunk numbers are 001~200.*
2. Dial the Service Code (not set at default).
 -  *You hear confirmation beeps as the system disconnects the trunk.*
 -  *You can now place a call on the free trunk.*
3. Press the line key for the trunk disconnected in Step 2.
 - OR -
 - Dial the trunk access code (**#9** + trunk number) for the trunk disconnected in Step 2.

Single Line Telephone

1. Dial trunk access code (**#9** + trunk number).
 -  *You hear busy tone. Trunk numbers are 001~200.*
2. Dial Service Code (not set at default).
 -  *You hear confirmation beeps as the system disconnects the line.*
3. Hookflash.
 -  *You can now place a call on the free line.*
4. Dial the trunk access code (**#9** + trunk number) for the trunk disconnected in Step 2.

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Group Call Pickup

Description

Group Call Pickup allows an extension user to answer a call ringing another extension in a Pickup Group. This permits co-workers in the same work area to easily answer each other's calls. The user can intercept the ringing call by dialing a code or pressing a programmed Group Call Pickup key. If several extensions in the group are ringing at the same time, Group Call Pickup intercepts the call based on the extension priority in the Pickup Group.

With Group Call Pickup, a user can intercept the following calls:

- A call ringing the user's own pickup group
- A call ringing another pickup group when the user knows the group number
- A call ringing another pickup group when the user does not know the group number

There are 64 Call Pickup Groups available.

Conditions

- A Call Pickup Group cannot have an associated name.
- Group Call Pickup cannot be used to answer calls recalling from Hold or Park.
- Group Call Pickup cannot be used to answer calls ringing Call Arrival Keys or Virtual Extensions.
- Virtual Extensions can use Group Call Pickup to answer calls ringing a multiline terminal or single line telephone.
- Users can pickup calls regardless of their access map programming.
- Directed Call Pickup provides another way of answering a co-worker's call.
- Function keys simplify Group Call Pickup operation.

Default Setting

Enabled



System Availability

Terminals

Any Station

Required Component(s)

None

Related Features

Central Office Calls, Answering

Directed Call Pickup

Programmable Function Keys

Programming

- ➔ **11-12-25 Service Code Setup (for Service Access) – Direct Call Pickup – Own Group**
If the default service code (756) for Direct Call Pickup for Own Group is not acceptable, change the code as required.
- ➔ **11-12-26 Service Code Setup (for Service Access) – Call Pickup for Specified Group**
If the default service code (768) for Call Pickup for Specified Group is not acceptable, change the code as required.
- ➔ **11-12-27 Service Code Setup (for Service Access) – Call Pickup**
If the default service code (*#) for Call Pickup is not acceptable, change the code as required.
- ➔ **11-12-28 Service Code Setup (for Service Access) – Call Pickup for Another Group**
If the default service code (769) for Call Pickup for Another Group is not acceptable, change the code as required.

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- ➔ **15-07-01 Programmable Function Keys**
Assign Group Call Pickup keys:
 - Code 24 for an extension own Pickup Group and ring group calls (Service Code ******)
 - Code 25 for a telephone ringing in another Pickup Group (Service Code 769) (used when the caller does not know the group number)
 - Code 26 (+ group) for a telephone ringing in another specific Pickup Group (Service Code 768)

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service to an extension (1~15).

 - ➔ **20-10-01 Class of Service Options (Answer Service) – Group Call Pickup (Within Group)**
Enable (1)/Disable (0) an extension user ability to pick up calls ringing their pickup group (Service Code 756).

 - ➔ **20-10-02 Class of Service Options (Answer Service) – Group Call Pickup (Another Group)**
Enable (1)/Disable (0) an extension user ability to pick up calls ringing telephones that are not in their pickup group (Service Code 769).

 - ➔ **20-10-03 Class of Service Options (Answer Service) – Group Call Pickup for Specific Group**
Enable (1)/Disable (0) an extension user ability to pick up calls ringing a specific group (Service Code 768).

 - ➔ **20-10-04 Class of Service Options (Answer Service) – Telephone Call Pickup**
Turn Off or On an extension ability to pick up a call ringing into a pickup group (Service Code ******).

 - ➔ **20-10-05 Class of Service Options (Answer Service) – Directed Call Pickup for Own Group**
Enable (1)/Disable (0) an extension user ability to use Directed Call Pickup for calls ringing their own group (Service Code 756).

 - ➔ **23-02-01 Call Pickup Groups**
Assign extensions to Pickup Groups. Also, use this option to assign an extension priority in a Pickup Group (Priority Number 1~999).

Call Pickup Groups: 1~9 or 01~64


Operation

To answer a call ringing another telephone in your Pickup Group:

1. Pick up the handset or press **Speaker**.
2. At multiline terminal only, press the **Group Call Pickup** key (PRG 15-07 or SC 751: 24).

- OR -

Dial **756** or ******.

 *Service Code ****** can pick up any call in the group, plus any Ring Group calls. Service Code 756 cannot pick up Ring Group calls.*

To answer a call ringing a telephone in another Pickup Group when you do not know the group number:

1. Pick up the handset or press **Speaker**.
2. At multiline terminal only, press the **Group Call Pickup** key (PRG 15-07 or SC 751: 25).
- OR -
Dial **769**.

To answer a call ringing a telephone in another Pickup Group when you know the group number:

1. Pick up the handset or press **Speaker**.
2. At multiline terminal only, press the **Group Call Pickup** key (PRG 15-07 or SC 751: 26 + group).
- OR -
Dial **768** and the group number (1~9 or 01~64).

Group Listen

Description

Group Listen permits a multiline terminal user to talk on the handset and have their caller's voice broadcast over the telephone speaker. This lets the multiline terminal user's co-workers listen to the conversation. Group Listen turns off the multiline terminal handsfree microphone so the caller does not pick the co-worker's voices during a Group Listen.

Conditions

- An extension in headset mode cannot use Group Listen.
- Group Listen is not available to single line telephones.

Default Setting

Disabled

System Availability

Terminals

Any Multiline Terminal

Required Component(s)

None

Related Features

Handset Operation

Programming

- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

- ➔ **20-13-26 Class of Service Options (Supplementary Service) – Group Listen**
Enable (1)/Disable (0) an extension user ability to activate Group Listen.

Operation

To initiate Group Listen:

1. Place or answer call using the handset.
2. Press **Speaker** twice (but do not hang up).
 - 📎 *Speaker flashes slowly.*
 - 📎 *You can talk to the caller through your handset. Your co-workers hear your caller's voice over your telephone's speaker after you press **Speaker** twice. Press **Speaker** a third time to turn off Group Listening.*

To talk Handsfree after initiating Group Listen:

1. Hang up the handset.

To cancel Group Listen (without hanging up your call):

1. Do not hang up.
2. Press the flashing **Speaker**.
 - 📎 *You can talk to the caller over the handset. Your co-workers can no longer hear the caller's voice.*

Handset Mute

Description

Handset Mute is provided to most terminals connected to the Electra Elite IPK system. While talking on the Multiline Terminal handset, a station user can dial a feature code or press MIC to mute the transmit speech path. The station user can still hear the outside (or intercom) voice.

Conditions

- MIC and/or the Handset Transmission Cut Off key flashes when active.
- Two service set tones are heard when Handset Mute is activated or deactivated.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Programmable Function Keys

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Handset Transmission Cut Off (code 40).

- ➔ **80-01-01 Service Tone Setup – Service Tone 25 (Service Set Tone)**
Allow the service set tone to be disabled.
Refer to [Table 2-26 Service Tone Setup, Program 80-01-01 on page 2-476](#) (Service Tone #25).

Table 2-26 Service Tone Setup, Program 80-01-01

Service Tone No.	Service Tone	Repeat Count	Unit Count	Basic Tone No.	Duration	Gain Level (dB)
1	No Tone	0	Basic 1	0	10	32 (0dB)
2	Intercom Dial Tone	0	Basic 1	9	10	32 (0dB)
3	Stutter Dial Tone	0	Basic 6	0 9 0 9 0 9	2 1 1 1 1 77	32 (0dB)
4	Internal Recall Dial Tone	2	Basic 2	9 0	1 1	32 (0dB) 32 (0dB)
5	Trunk Dial Tone	0	Basic 1	9	10	32 (0dB)
6	Internal Busy Tone	0	Basic 2	0 11	5 5	20 (-6dB) 20 (-6dB)
7	DND Busy Tone and Selectable Display Message Tone	0	Basic 2	0 1	2 2	32 (0dB) 32 (0dB)
8	Busy Tone	0	Basic 2	0 11	5 5	20 (-6dB) 20 (-6dB)
9	Internal Reorder Tone	0	Basic 2	11 0	3 2	20 (-6dB) 20 (-6dB)
10	Internal Interrupt Tone	0	Basic 2	0 1	1 1	32 (0dB) 32 (0dB)
11	Internal Confirmation Tone	3	Basic 2	0 6	5 1	32 (0dB) 32 (0dB)
12	Internal Hold Tone	0	Basic 0	0	0	32 (0dB)
13	External Hold Tone	0	Basic 0	0	0	32 (0dB)
14	Intercom Ringback Tone	0	Basic 2	9 0	10 20	32 (0dB) 32 (0dB)
15	Override Tone	1	Basic 1	12	5	32 (0dB)

Table 2-26 Service Tone Setup, Program 80-01-01

Service Tone No.	Service Tone	Repeat Count	Unit Count	Basic Tone No.	Duration	Gain Level (dB)
16	Lock-Out Tone	0	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
17	Clock Alarm Tone	0	Basic 4	6 0 6 0	1 1 1 7	32 (0dB) 32 (0dB) 32 (0dB) 32 (0dB)
18	BGM	0	Basic 0	0	0	32 (0dB)
19	Door Box Chime 1	3	Basic 6	4 4 2 2 2 0	2 2 3 4 6 5	38 (+3dB) 26 (-3dB) 38 (+3dB) 26 (-3dB) 14 (-9dB) 32 (0dB)
20	Door Box Chime 2	3	Basic 6	7 7 5 5 5 0	2 2 3 4 6 5	38 (+3dB) 26 (-3dB) 38 (+3dB) 26 (-3dB) 14 (-9dB) 32 (0dB)
21	Door Box Chime 3	3	Basic 6	8 8 6 6 6 0	2 2 3 4 6 5	38 (+3dB) 26 (-3dB) 38 (+3dB) 26 (-3dB) 14 (-9dB) 32 (0dB)
22	Door Box Chime 4	3	Basic 6	4 4 2 2 2 0	1 1 2 2 3 2	38 (+3dB) 26 (-3dB) 38 (+3dB) 26 (-3dB) 14 (-9dB) 32 (0dB)
23	Door Box Chime 5	3	Basic 6	7 7 5 5 5 0	1 1 2 2 3 2	38 (+3dB) 26 (-3dB) 38 (+3dB) 26 (-3dB) 14 (-9dB) 32 (0dB)
24	Door Box Chime 6	3	Basic 6	8 8 6 6 6 0	1 1 2 2 3 2	38 (+3dB) 26 (-3dB) 38 (+3dB) 26 (-3dB) 14 (-9dB) 32 (0dB)
25	Service Set Tone	3	Basic 2	0 9	1 1	32 (0dB) 32 (0dB)
26	Service Clear Tone	3	Basic 2	0 9	1 1	32 (0dB) 32 (0dB)

Table 2-26 Service Tone Setup, Program 80-01-01

Service Tone No.	Service Tone	Repeat Count	Unit Count	Basic Tone No.	Duration	Gain Level (dB)
27	Talkback Tone	2	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
28	Speaker Monitor Tone This tone is what the originator hears when placing a handsfree speaker ICM call.	1	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
29	Door Relay Tone	1	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
30	Door Box Call Tone	1	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
31	Paging Tone	2	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
32	Splash Tone 1	1	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
33	Splash Tone 2	2	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
34	Splash Tone 3	3	Basic 2	0 6	1 1	32 (0dB) 32 (0dB)
35	1 Second Signal Tone	1	Basic 1	6	10	32 (0dB)
36	Sensor Alarm Tone 1	0	Basic 2	7 0	2 2	32 (0dB) 32 (0dB)
37	Sensor Alarm Tone 2	0	Basic 2	7 0	5 5	32 (0dB) 32 (0dB)
38	Sensor Alarm Tone 3	0	Basic 2	7 0	7 7	32 (0dB) 32 (0dB)
39	Ring Busy Tone	0	Basic 6	0 11 0 11 10 0	5 5 5 5 10 20	32 (0dB) 32 (0dB) 32 (0dB) 32 (0dB) 32 (0dB) 32 (0dB)
40	Internal Call Waiting Tone	1	Basic 1	12	2	32 (0dB)
41	Intrusion Tone	1	Basic 1	12	5	32 (0dB)
42	Conference Tone	0	Basic 0	0	0	32 (0dB)
43	Intrusion Tone 2	0	Basic 1	2	8	32 (0dB)
44	External Dial Tone	0	Basic 1	9	1	26 (-3dB)
45	External Ring Back Tone	0	Basic 2	10 0	10 30	32 (0dB) 32 (0dB)

Table 2-26 Service Tone Setup, Program 80-01-01

Service Tone No.	Service Tone	Repeat Count	Unit Count	Basic Tone No.	Duration	Gain Level (dB)
46	DID Error Tone	0	Basic 2	11 0	5 5	32 (0dB) 32 (0dB)
47	External Busy Tone	0	Basic 1	11	0	32 (0dB)
48	Voice Mail Message Waiting, Special Dial Stutter Dial Tone (Analog Sets)	0	Basic 2	9 0	1 1	32 (0dB) 32 (0dB)
49	--- Not Used ---					
50	External Special Audible Ring Tone	0	3	10 12 0	10 2 30	32 (0dB) 32 (0dB) 32 (0dB)
51	External Intercept Tone	0	2	12 4	3 2	32 (0dB) 32 (0dB)
52	External Call Waiting Tone	1	1	12	3	32 (0dB)
53	External Executive Override Tone	1	1	12	10	32 (0dB)
54	--- Not Used ---					
55	Generate tone for TAPI2.1	0	Basic 1	3	0	32 (0dB)
56	Warning Beep Tone Signaling	1	Basic 1	2	8	32 (0dB)
57	Headset Ear Piece Ringing Tone	0	Basic 5	0 2 0 2 0	2 1 1 1 20	32 (0dB) 32 (0dB) 32 (0dB) 32 (0dB) 32 (0dB)
58	Opening Chime Tone, External Paging	1	Basic 8	2 2 14 14 15 15 16 16	2 2 2 2 2 2 6 4	32 (0dB) 26 (-3dB) 32 (0dB) 26 (-3dB) 32 (0dB) 26 (-3dB) 32 (0dB) 26 (-3dB)
59	Ending Chime Tone, External Paging	1	Basic 8	20 20 19 19 18 18 17 17	2 2 2 2 2 2 6 4	32 (0dB) 26 (-3dB) 32 (0dB) 26 (-3dB) 32 (0dB) 26 (-3dB) 32 (0dB) 26 (-3dB)
60	Splash Tone 1 (Mute)	1	Basic 2	0 6	1 1	8 (-12dB) 8 (-12dB)
61	Splash Tone 2 (Mute)	2	Basic 2	0 6	1 1	8 (-12dB) 8 (-12dB)

Table 2-26 Service Tone Setup, Program 80-01-01

Service Tone No.	Service Tone	Repeat Count	Unit Count	Basic Tone No.	Duration	Gain Level (dB)
62	Splash Tone 3 (Mute)	3	Basic 2	0 6	1 1	8 (-12dB) 8 (-12dB)
63	EXT SPK Ring-back Tone	0	Basic 2	10 0	10 30	32 (0dB) 32 (0dB)
64	Music on Hold (MOH)	0	0	0	0	32 (0dB)

- ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
 (Default: 0).
 Refer to [Table 2-27 Service Tone Setup, Program 80-01-02](#) on page 2-480.

Table 2-27 Service Tone Setup, Program 80-01-02

Item No.	Item	Repeat Count
02	Basic Tone Number	1~33 0 = No Tone 33=Default Time Slot
03	Duration Count	1~255 (100~25500ms)
04	Gain Level (dB)	1~63 (-15.5 ~ +15.5)

Operation

While talking on a terminal handset:

- Press **MIC** (DTH/DTR terminals only).
- OR -
- Press **Feature** + dial 1.
- OR -
- Press the **Handset Transmission Cut-Off** key (PRG15-07-01; Key 40 or SC 751 Key Code 40).

Handsfree and Monitor

Description

Handsfree allows a Multiline Terminal user to process calls using the speaker and microphone in the telephone instead of the handset. Handsfree is a convenience for workers who do not have a free hand to pick up the handset. For example, a terminal operator could continue to enter data with both hands while talking on the telephone.

Three variations of Handsfree are available.

Handsfree

The user can press Speaker to place and answer calls instead of using the handset.

Automatic Handsfree

The user can press a trunk line key or virtual extension key without first lifting the handset or pressing Speaker. An extension can have Automatic Handsfree for only outgoing calls or both outgoing calls and incoming calls.

Monitor

User can place a call without lifting the handset, but must lift the handset to speak.

Conditions

- Handsfree and Monitor are not available for single line telephones.
- Prime Line Selection affects how incoming and outgoing calls are handled and thus determines what happens when the user presses speaker.
- Monitoring volume may be adjusted using the volume control on the Multiline Terminal.
- When a Multiline Terminal user lifts the handset, the monitoring condition is automatically released, and the Speaker LED goes off.
- A Multiline Terminal is considered off-hook by the system when this feature is used.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Programming

- ➔ **15-02-08 Multiline Telephone Basic Data Setup – Automatic Handsfree**
Set whether pressing a key accesses a One-Touch Key (1) or preselects the key (0).
- ➔ **15-02-16 Multiline Telephone Basic Data Setup – Handsfree Operation**
Enable (1)/Disable (0) an extension user ability to use the speakerphone on outside calls. When disabled, users can hear the conversion, but cannot respond handsfree.
- ➔ **20-06-01 Class of Service for Extensions**
Assign Class of Service (1~15) for extensions.
- ➔ **20-02-12 System Options for Multiline Telephones – Forced Intercom Ring (ICM Call Type)**
Enable (1)/Disable (0) Forced Intercom Ringing. If enabled, incoming Intercom calls normally ring. If disabled, Intercom calls voice-announce.
- ➔ **20-09-05 Class of Service Options (Incoming Call Service) – Signal/Voice Call**
Allow (1)/Prevent (0) an extension user from enabling Handsfree Answerback or Forced Intercom Ringing for their incoming intercom calls.

Related Features

Central Office Calls, Answering

Central Office Calls, Placing

Microphone Cutoff

Prime Line Selection

Operation

To talk Handsfree:

1. Press **Speaker**, **Trunk Line** key or **Virtual Extension** key.
2. Place the call.
3. Speak toward the telephone when the called party answers.

To change a handset call into a Handsfree call:


1. Press **Speaker** and hang up the handset.
2. Press **Speaker** again to hang up.

To change a Handsfree call into a handset call:

1. Lift the handset.

To turn on/off Monitor:

1. Press **MIC**, Feature + 1, or the Microphone Function Key (PRG 15-07 or SC 751 : 02) to turn on or off the Microphone.

 *Monitor is off when **MIC LED** is lit, the Microphone Function Key is lit, or the handset is lifted.*

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Handsfree Answerback/Forced Intercom Ringing

Description

Handsfree Answerback permits an extension user to respond to a voice-announced Intercom call by speaking toward the telephone, without lifting the handset. Like Handsfree, this is a convenience for workers who do not have a free hand to pick up the handset.

Conditions

- Handsfree Answerback does not require the Speaker phone to be enabled (PRG 15-02-16).
- A Multiline Terminal user can process calls using the speaker and microphone in the telephone (instead of the handset).
- With Microphone Cutoff enabled, Handsfree Answerback callers to an extension hear a single beep (instead of two).
- Incoming Intercom calls always ring single line telephones.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Handsfree and Monitor


Microphone Cutoff

Programming


- ➔ **11-11-15 Service Code Setup (for Setup/Entry Operation) – Enable Handsfree Incoming Intercom Calls**
If required, change the service code used for setting an extension to voice announce for incoming ICM calls (default: 721).
- ➔ **11-11-16 Service Code Setup (for Setup/Entry Operation) – Force Ringing of Incoming Intercom Calls**
If required, change the service code used for setting an extension to forced ringing for incoming ICM calls (default: 723).
- ➔ **11-12-06 Service Code Setup (for Service Access) – Switching of Voice Call and Signal Call**
If required, change the service code used for toggling an outgoing ICM call between a voice call and signal call (default: 712).
- ➔ **15-02-16 Multiline Telephone Basic Data Setup – Handsfree Operation**
Enable (1)/Disable (0) an extension user ability to use the speakerphone on outside calls. Users can hear the conversation, but cannot respond handsfree.
- ➔ **20-02-12 System Options for Multiline Telephones – Forced Intercom Ring (ICM Call Type)**
Enable (1)/Disable (0) Forced Intercom Ringing. If disabled, Intercom calls voice-announce.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-08-10 Class of Service Options (Outgoing Call Service) – Signal/Voice Call**
Enable (1)/Disable (0) an extension user ability to toggle between Handsfree Answerback and Forced Intercom Ringing for outgoing Intercom calls (dial 1 or Service Code 712).
- ➔ **20-09-05 Class of Service Options (Incoming Call Service) – Signal/Voice Call**
Enable (1)/Disable (0) an extension user ability to set Handsfree Answerback (Service Code 721) and Forced Intercom Ringing (Service Code 723) for incoming Intercom calls.

Operation


To enable Handsfree Answerback for your incoming Intercom calls:

1. Press idle **Speaker**.
2. Dial **721**.
3. Press **Speaker** to hang up.
 *This disables Forced Intercom Ringing.*

To enable Forced Intercom Ringing for your incoming Intercom calls:

1. Press idle **Speaker**.
2. Dial **723**.
3. Press **Speaker** to hang up.
 *This disables Handsfree Answerback.*

To change the way your Intercom call signals the extension you are calling:

1. Dial **1**.
 *If ringing, your call voice-announces. If voice-announced, your call starts to ring the destination. This option is also available on single line telephones.*

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Headset Operation

Description

A multiline terminal user can use a customer-provided headset in place of the handset. Like using Handsfree, using the headset frees up the user's hands for other work. However, Headset Operation provides privacy not available from Handsfree.

As the headset plugs into a separate jack on the bottom of the telephone, the handset can still be connected to the telephone. This provides you with the option to use the handset, headset or the speakerphone for calls.

Conditions

- While using the headset, the Headset function key becomes a release (disconnect) key and no dial tone is heard from the speaker.
- While in the headset mode, the hook switch is not functional.
- The Headset Programmable Function key (05) and Headset service code (688) are not available for the Electra Professional telephones.
- An extension with a headset can still receive voice-announced Intercom calls and respond handsfree when idle.
- A Headset Function key is required to answer or place a call in headset mode.

Default Setting

Disabled

System Availability

Terminals

None

Required Component(s)

Headset

Related Features

Handsfree Answerback/Forced Intercom Ringing

Programmable Function Keys

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Headset Operation (code 05).

- ➔ **20-02-05 System Options for Multiline Telephones – Headset Busy Mode**
Set the conditions under which a headset extension is busy to incoming callers:
 - The Headset extension is busy to incoming callers when only one extension appearance is busy (0).
 - OR -
 - Headset extension is busy to incoming callers only when both extension appearances are busy (1).


- ➔ **20-02-12 : System Options for Multiline Telephones – Forced Intercom Ring (ICM Call Type)**
Determine whether intercom calls should ring or voice-announce extensions.

0 = Voice
1 = Ring



- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension to manually (0)/automatically (1) receive Off-Hook signals.

Operation

To enable the headset:

1. Plug in the headset into the headset jack on the bottom of the telephone.
2. Program a **Headset** key (PRG 15-07 or SC 751: 05).
 -  *You hear a confirmation beep.*

To use the headset:

-  *The Headset key lights when on a call. To disconnect, press the Headset key again.*
-  *You can still use the handset for calls or respond to voice-announced Intercom calls with the headset plugged in. The headset only activates when the Headset key is pressed.*
- Answer a ringing call by pressing the **Headset** key.
 - OR -
- Press the **Headset** key and then a line key or press **Speaker** then **9** to make a outgoing call.
 - OR -
- Press the **Headset** key to get intercom dial tone.
 - OR -
- If on a call, press the **Headset** key to hang up.

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Hold

Description

Hold lets an extension user put a call in a temporary waiting state. The caller on Hold hears silence or Music on Hold, not conversation in the extension user's work area. While the call waits on Hold, the extension user may process calls or use a system feature. Calls left on Hold too long recall the extension that placed them on Hold. Four types of Hold are available:

System Hold

An outside call a user places on Hold flashes the line key (if programmed) at all other multiline terminals. Any multiline terminal user with the flashing line key can pick up the call.

Exclusive Hold

When a user places a call on Exclusive Hold, only that user can pick up the call from Hold. The trunk appears busy to all other multiline terminals that have a key for the trunk. Exclusive hold is important if a user does not want a co-worker picking up their call on Hold.

Group Hold

If a user places a call on Group Hold, another user in the Department Group can dial a code to pick up the call. This lets members of a department easily pick up each other's calls.

Intercom Hold

A user can place an Intercom call on Hold. The Intercom call on Hold does not indicate at any other extension.

Hold Recall to Operator

Hold Recall to Operator enhances how the system handles calls that have been left on hold too long. With Hold Recall to Operator:

- A trunk call recalls the extension that placed it on Hold after the Hold/Exclusive Hold Recall Time.
- The recalling trunk rings the extension that placed it on Hold for the Hold/Exclusive Hold Recall Callback Time.
- After the Hold/Exclusive Hold Recall Callback Time, the trunk call rings the operator.

Hold Recall to Operator applies to trunk calls placed on System Hold, Exclusive Hold and Group Hold. It does not apply to Intercom calls.

Conditions

- The called extension must lift the handset or press Speaker before the call can be placed on hold.
- Callers on Hold hear Music on Hold, if programmed.
- An extension can have function keys for System Hold and Exclusive Hold.
- Analog single line telephones can use Exclusive Hold and Group Hold only.
- If station A calls station B, and station A puts station B on hold and then calls station C, station C cannot transfer the call.

Default Setting

Enabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Music on Hold

Programmable Function Keys

Single Line Telephones, Analog 500/2500 Sets

Programming

- ➔ **11-12-30 Service Code Setup (for Service Access) – Specified Trunk Answer**
If required, redefine the service code used to answer a specific trunk which is either ringing or on hold (default: 672).
- ➔ **11-12-33 Service Code Setup (for Service Access) – Group Hold**
If required, redefine the service code used to answer a specific trunk which is either ringing or on hold (default: 732).
- ➔ **11-12-34 Service Code Setup (for Service Access) – Answer for Group Hold**
If required, redefine the service code used to answer a specific trunk which is either ringing or on hold (default: 762).
- ➔ **14-01-16 Basic Trunk Data Setup – Forced Release of Held Call**
Enable (1)/Disable (0) Forced Release of Held Call.
- ➔ **14-07-01 Trunk Access Map Setup**
Set up the Trunk Access Maps. This sets the access options for trunks on Hold.

Trunk Access Maps: 1~200
- ➔ **15-02-06 Multiline Telephone Basic Data Setup – Hold Key Operating Mode**
Set the function of the extension Hold key:

0 = Normal Common
1 = Exclusive Hold
- ➔ **15-02-07 Multiline Telephone Basic Data Setup – Automatic Hold for CO Line**
Determines whether an extension disconnects the trunk line (1) or automatically holds it (0) when you direct select a CO line without placing the call on hold first. Automatic Hold does not work until the elapse call timer is displayed.
- ➔ **15-02-11 Multiline Telephone Basic Data Setup – Callback Automatic Answer**
Determine whether an extension answers (1) or does not answer (0) a call that is recalled when you go off-hook.
- ➔ **15-06-01 Trunk Access Map for Extensions**
Assign Trunk Access Maps to extensions.

Trunk Access Maps: 1~200
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Exclusive Hold (code 45). If an extension has its fixed Hold key reassigned (in Program 15-02-06), assign a function key for System Hold (code 44).
- ➔ **16-02-01 Department Group Assignment for Extensions**
Assign extensions to Department Groups.

Department Groups: 1~64

-
-
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
 - ➔ **20-11-09 Class of Service Options (Hold/Transfer Service) – Group Hold Initiate**
Enable (1)/Disable (0) an extension user ability to initiate Group Hold (Service Code 732).
 - ➔ **20-11-10 Class of Service Options (Hold/Transfer Service) – Group Hold Answer**
Enable (1)/Disable (0) an extension user ability to pick up a call placed on Group Hold (Service Code 762).
 - ➔ **20-11-13 Class of Service Options (Hold/Transfer Service) – Operator Transfer after Hold Callback**
Turn Off or On an extension ability to have a call which recalls from Hold transfer to the operator.
 - ➔ **20-17-01 Operator Extension – Operator Extension Number**
Assign an operator extension number if Hold Recall to Operator is used.
 - ➔ **24-01-01 System Options for Hold – Hold Recall Time**
Set the Hold Recall Time (0~64800 seconds). A call on Hold recalls the extension that placed it on Hold after this time. Default is 90.
 - ➔ **24-01-02 System Options for Hold – Hold Recall Callback Time**
Set the Hold Recall Callback Time (0~64800 seconds). A trunk recalling from Hold rings an extension for this time. Default is 30.
 - ➔ **24-01-03 System Options for Hold – Exclusive Hold Recall Time**
Set the Exclusive Hold Recall Time (0~64800 seconds). A call on Exclusive Hold recalls the extension that placed it on Hold after this time. Default is 90.
 - ➔ **24-01-04 System Options for Hold – Exclusive Hold Recall Callback Time**
Set the Hold Recall Time (0~64800 seconds). A trunk recalling from Hold rings an extension for this time. If still unanswered, the call changes to System Hold. Default is 30.
 - ➔ **24-01-05 System Options for Hold – Forced Release of Held Call**
Set the Forced Release of Held Calls time (0~64800 seconds). If enabled in Program 14-01-16, the system disconnects a call if on Hold longer than this time. Default is 1800.

Operation

System Hold

To place an outside call on System Hold:

Press **Hold**.

To pick up an outside call on System Hold:

Press the flashing **CAP** key.

- OR -

If you know the specific line number, dial **672** + Line number (**001~200**).

Exclusive Hold

To place an outside call on Exclusive Hold:

Press the **Exclusive Hold** key (PRG 15-07-01 or SC 751: 45).

- OR -

Press **Feature + Hold**.

Single Line Telephone

1. Hookflash.
2. Dial **749**.

To pick up an outside call on Exclusive Hold:

Press flashing **CAP** key.

Single Line Telephone

Dial **759**.

Group Hold

To place a call on Hold so anyone in your Department Group can pick it up:

1. Press **Hold**.
2. Dial **732**.
3. Press **Speaker** to hang up.

Single Line Telephone

1. Hookflash.
2. Dial **732**.
3. Hang up.

To pick up a call on Group Hold:

1. Press **Speaker**.

2. Dial **762**.

Single Line Telephone

1. Lift the handset.
2. Dial **762**.

Intercom Hold

To place an Intercom call on Intercom Hold:

1. Press **Hold**.
2. Press **Speaker** to hang up.

To pick up an Intercom call on Intercom Hold:

1. Press **Speaker**.
2. Press the flashing **Conf**.

Hotel/Motel

Enhancements

Version 2000 or higher provides the following feature enhancements:

Function Code 95: Page Switching

A Direct Station Selection (DSS) Console can have a shift key to expand the capacity of a console. Two pages (or sheets) are available. In Page 1 mode, keys 01~54 are available for DSS/BLF function, feature access, etc. In Page 2 mode, an additional 54 keys (01~54) are available for DSS/BLF function, feature access, etc. Keys 55~60 do not shift. A total of 113 keys are available when using the Page key ($54 + 54 + 6 - 1 = 113$). The Page key (shift key) must be assigned on key 55~60.

DSS Console page switching which allows one DSS Console to monitor up to 114 rooms for wake up call and room status. This function code can be assigned to DSS Console buttons 55~60 only. When enabled the button indicates the following:

If the DSS Key is:	The DSS Console Indicates:
Red	Buttons 1~54 are Rooms 1~54
Green	Buttons 61~114 are Rooms 61~114

PMS Integration: With Version 2000 software and PMS Interface Box (PMS-U10), the Electra Elite IPK II supports third party PMS applications. The PMS-U10 serves as a gateway between the PMS applications, the Electra Elite IPK II and EliteMail LX voice mails. The Electra Elite IPK II and EliteMail LX must be licensed for this feature to work in Hotel/Motel and the Property Management System (PMS). Refer to the Hotel/Motel and EliteMail LX manuals for more information.

Description

Your Elite IPK II telephone system provides Hotel/Motel services in addition to the many features available to business users. These Hotel/Motel services help you run your facility more efficiently, save you time and money, **and** provide your guests with more responsive service.

Hotel/Motel features include:

Wake Up Call

Wake Up Call is like having an alarm clock in each room – with some unique advantages:

- Guests can set or cancel Wake Up Calls for themselves, or you can set and cancel Wake Ups for them.
- Unanswered Wake Up Calls can automatically call the operator and print on the Room Status Printout report.

- Use Wake Up Call as a meeting reminder (e.g., for convention attendees).

Single Digit Dialing

Single Digit Dialing gives your guests one-touch access to your important Hotel/Motel services. They can lift the handset and press a single key for:

- Extensions such as the front desk, reservation services, housekeeping or the maitre d' of your restaurant.
- Feature Access Codes for one-button access to selected features and outside lines.
- Voice Mail, so your guests can leave requests even when your service providers are unavailable.

A Department Calling Group

A Department Calling Group, allowing, for example, your guests to reach the first available agent in your reservation desk group.

Message Waiting

If you call a guest while they are away from their room, leave them a Message Waiting. When the guest returns, they see the lamp on their phone flashing and can automatically call you back. You can use Message Waiting when you have parcels for a guest dropped off at your front desk. Do not keep redialing the guest if they are not in – just send them a Message Waiting. (Your DSS Console can show all the rooms that have messages waiting.)

PMS Integration

With **Version 2000** software and a PMS Interface Box (PMS-U10) the Electra Elite IPK II supports third party Property Management System (PMS) applications. The PMS-U10 serves as a gateway between the PMS applications, the Electra Elite IPK II and EliteMail LX voice mails.

The EliteMail LX VMP/CTP, EliteMail CTI LX and EliteMail CTI LX Lite are supported. When using a CTP()-U10 based voice mail you must have the 5x2 to DB-9 serial adapter. In addition, any voice mail used must be licensed for the Hotel feature and have PMS enabled. Refer to the appropriate voice mail installation manual for information on configuring the voice mail.

- The Electra Elite IPK II and EliteMail LX must be licensed for Hotel/Motel and PMS for this feature to work.
- The supported PMS protocols are NEAX 90 and KTSi.
- The KSU to PMS-U10 connection is via the LAN and an IP port only (default is 5129).
- The PMS-U10 to voice mail connection is via serial port COM 2 only.
- The PMS-U10 to PMS System can be done via LAN or serial port COM 1.

- ❑ The NEAX-90 with and without NAK protocol is compatible with property management systems that support NEAX-90 protocol. Not all messages or functionality supported by NEAX Model 90 protocol are implied or provided by the PMS-U10. The PMS-U10 and the IPK II provide a subset of features supported by NEAX Model 90 protocol.

Refer to the Electra Elite IPK II Hotel/Motel Services Guide for complete programming information.

Room to Room Calling Restriction

Prevent guests in one room from calling guests in another – a handy feature for guests that want to maintain their privacy. If you need to, you can always allow inter-room calling (e.g., for families or groups that have separate rooms).

Toll Restriction (When Checked In)

Control your guest's long distance dialing automatically when they check in. Use this feature to set up two different Toll Restriction modes. The first mode is for you and your staff when the room is checked out. The second mode is for your guests when they check in. You may want to restrict the outside numbers guests can dial, but allow your staff to call vendors and suppliers.

Room Status

You can use your phone and DSS Console to set and monitor the status of all guest rooms: *Checked In*, *Checked Out*, *Maid Required* and *Maid in Room*. Maximize room usage by coordinating your cleaning staff and reservation desk. For example, you can dial simple codes to set a room status.

Room Status Printouts

The Room Status Printout gives a concise overview of the status of all guest rooms at a glance. The printout provides up to the minute reports showing Room Status, Room Call Restriction, Do Not Disturb, Message Waiting and Wake Up Calls. If your cleaning staff needs to know which rooms to clean up, for example, print out the report showing Room Status.

DSS Console Monitoring

The DSS Console provides room monitoring. You can see at a glance which rooms have Wake Up Calls set or messages waiting, and you can still use your console for business mode features.

Do Not Disturb

A guest can activate DND anytime they need privacy (for example, if they need to work uninterrupted). Do Not Disturb (DND) blocks the room telephone incoming calls and Paging announcements. This can be set from the room phone or attendant phone.

Flexible Numbering Plan

To simplify dialing guests and services in your facility, customize your system to have room numbers match phone extension numbers. For example, if the rooms on the first floor are numbered 100~120, the corresponding room extensions can also be 100~120.

Conditions

- A CTU adapter USB connection should be safely stopped in Windows before the following procedures are performed:
 - CPUII Upgrade
 - ESIB/E Reset
 - CPUII Reset
- Function codes 92 and 93 can be assigned only to a DSS Console that is in Hotel Mode. These features do not work when programmed on multiline telephone line keys or on a DSS Console in Business mode.
- When multiple DSS Consoles are used for Hotel/Motel, function keys must be assigned to each DSS console for Wake Up Call Indication and Room Status Indication.
- The Message Waiting status of a room cannot be seen when the console is in Wake Up Call or Room Status mode.
- The BLF indication for each room is always available no matter what mode the console is in.
- The Hotel/Motel feature requires the CPUII be licensed for Hotel. The following dial access codes can be used only if the CPUII is licensed for the Hotel/Motel Feature:

Dial Access Codes that Require CPUII Hotel License		
Program	Dial Access Code	Description
11-10-16	626	Leave Message Waiting
11-14-01	627	Enable DND at Own Extension
11-14-02	628	Disable DND at Own Extension
11-14-03	629	Enable DND for Other Extension
11-14-04	630	Disable DND for Own Extension
11-14-05	631	Enable Wake Up Call for Own Extension
11-14-06	632	Disable Wake Up Call for Own Extension
11-14-07	633	Enable Wake Up Call for Other Extension
11-14-08	634	Disable Wake Up Call for Other Extension

Dial Access Codes that Require CPU11 Hotel License (Continued)		
Program	Dial Access Code	Description
11-14-09	635	Enable Room to Room Call Restriction
11-14-10	636	Disable Room to Room Call Restriction
11-14-11	637	Change Code Restriction Class for Other Extension
11-14-12	638	Check In Room
11-14-13	639	Check Out Room
11-14-14	640	Change Room Status for Own Extension
11-14-15	641	Change Room Status for Other Extension
11-14-16	642	Room Status Output
11-14-17	675	Hotel Room Monitor
11-14-18	666	Use to set/change the default Toll Restriction Class on check-in for a room (PRG 42-02-02)

- Program 30-03-01: DSS Console Key Assignment, Function Number 95: Page Switching is not supported in **Version 1600**.
- Program 42-06-01~Program 42-06-06: PMS Service Setting, are not supported in **Version 1600**.

Default Settings

Not Enabled

System Availability

Terminals:

All Terminals

Required Component(s)

DSS Console

Related Features

Code Restriction

Department Calling

Do Not Disturb

Flexible System Numbering

Programming

Refer to the Electra Elite IPK II Hotel/Motel Services Guide for complete programming information.

Operation

Refer to the Electra Elite IPK II Hotel/Motel Services Guide for complete operation information.

Hot Key-Pad

Enhancements

This feature added with **Version 1500**.

Description

The Hot Key-Pad feature allows the user to place a call without lifting the handset or pressing Speaker. When the user dials another extension number on an idle telephone with Hot Key-Pad enabled, Speaker lights and the internal call is made. When the user dials the trunk access code from a telephone with Hot Key-Pad enabled, Speaker lights, a trunk is seized and the outgoing call is made.

Conditions

- When a user dials any digit on a station with Hot Key-Pad enabled, Speaker lights.
- After a user dials the trunk access code on a station with Hot Key-Pad enabled, a trunk is seized on dialing the first digit of the called party number.
- When both Hot Key-Pad and Dialing Number Preview are turned on, Hot Key-Pad has priority and Dialing Number Preview does not work.
- When both Hot Key-Pad and Hotline are turned on, Hot Key-Pad has priority and Hotline does not work.
- When placing an outgoing call with the Hot Key-Pad, the user must dial the trunk access code before dialing the called party number.
- The ARS feature can be used when placing outside calls with the Hot Key-Pad. When both Hot Key-Pad and VRS Fixed Messaging are turned on, VRS fixed messaging does not work.
- The Hot Key-Pad also works when dialing service codes.

Default Settings

Disabled

System Availability

Terminals

All Multiline Terminals

Required Software:

Version 1500 or higher

Related Features

Central Office Calls, Placing

Class of Service

Dialing Number Preview

Hotline

Intercom

Voice Response System (VRS)

Programming

- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service to extensions (1~15). Any Class of Service assignment you change using Service Code 677 automatically updates this program.

- ➔ **20-08-20 Class of Service Options (Outgoing Call Service) – Hot Key-Pad**
Enable (1)/Disable (0) an extension user ability to use Hot Key-Pad. Default is 0.



Operation

To place an intercom call using Hot Key-Pad:

Dial the extension.

To place a trunk call using Hot Key-Pad:

Dial the trunk access code, **9** by default, and the number you want to call.

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Hotline

Description

Hotline gives a multiline terminal user one-button calling and Transfer to another extension (the Hotline partner). Hotline helps co-workers that work closely together. The Hotline partners can call or Transfer calls to each other by pressing a single key.

The Hotline feature has two applications.

- Hotline (Hotline partner)
- Ringdown Extension, Internal/External (Refer to [Ringdown Extension, Internal/External on page 2-849.](#))

In addition, the Hotline key shows the status of the partner's extension.

When the key is . . .	The extension is . . .
Off	Idle
On	Busy or ringing
Fast Flash	DND – All calls (option 3) or Intercom calls (option 2)
Double Wink On	ACD Agent logged onto the group
Wink Off	ACD Agent logged off

There are 512 internal Hotline extensions available.

Conditions

- An extension user cannot use Hotline to pick up a call ringing their partner's extension.
- If a station is an ACD agent, the Hotline key blinks to indicate the ACD agent's status.
- Hotline keys can be assigned to the DSS consoles.
- Hotline does not override Do Not Disturb.
- Hotline always follows the Handsfree Answerback/Forced Intercom Ringing mode set at the called extension. The Hotline caller can override the setting, if desired.
- External Hotline automatically dials a telephone number or Speed Dial – System/Group/Station number when the handset is lifted.
- If the partner's extension is busy, Hotline does not automatically activate Off-Hook Signaling.

- A Hotline is a uniquely programmed function key.

Default Setting

Disabled

Related Features

Automatic Call Distribution (ACD)

Distinctive Ringing, Tones and Flash Patterns

Direct Station Selection

Do Not Disturb

Handsfree Answerback/Forced Intercom Ringing

Off-Hook Signaling

Programmable Function Keys

Ringdown Extension, Internal/External

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Hotline (code 01 + partner's extension number).
- ➔ **15-02-22 Multiline Telephone Basic Data Setup – Multiple Incoming From Intercom and Trunk**
When this option is set to 0 (Disabled), incoming calls to an extension indicate on any Hotline keys for that extension as solid (busy). When this option is set to 1 (Enabled), lighting is determined by the setting of Program 22-01-01 Incoming Call Priority. If set to trunk priority (1), the Hotline key lights solid when a trunk call rings in. If set to intercom priority (0), the Hotline key does not light for incoming trunk calls, but lights solid for intercom calls.

➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**

Program 20-02-03 and Program 20-13-06 set the conditions under which a Hotline, Reverse Voice Over or DSS Console key indicates that an extension is busy. With condition 1 in the following chart, the BLF LED is on only when both extension line appearances are busy. In conditions 2~4, the BLF LED is on when one line appearance is busy.

	Program 20-13-06	Program 20-02-03	BLF ¹ Status	Busy Status
1	1	0	Off	No
2	1	1	On	Yes
3	0	0	On	Yes
4	0	1	On	Yes

¹ BLF is on for extension receiving a voice announced Intercom call.

➔ **20-06-01 Class of Service for Extensions**

Assign a Class of Service to an extension (1~15).

➔ **20-08-19 Class of Service Options (Outgoing Call Service) – Hot line for SPK**

0 = Off

1 = On

➔ **20-11-06 Class of Service Options (Hold/Transfer Service) – Unscreened Transfer (Ring Inward Transfer)**

0 = Off

1 = On

➔ **20-11-11 Class of Service Options (Hold/Transfer Service) – Automatic On-Hook Transfer**

0 = Off

1 = On

➔ **22-01-01 System Options for Incoming Calls – Incoming Call Priority**


Determine if intercom calls or trunk calls have answer priority when both are ringing simultaneously (0 = Intercom Call Priority, 1 = Trunk Call Priority). Refer to Program 15-02-22 for light conditions.

➔ **30-05 DSS Console Lamp Table**

If required, customize the system DSS Console flash rates. This table also affects DSS/Hotline key lights for multiline terminals.

Operation

To place a call to your Hotline partner:


1. Press the **Hotline** key (PRG 15-07 or SC 751: 01 + partner's extension number).
 *You can optionally lift the handset after this step for privacy.*

To transfer your outside call to your Hotline partner:

1. Press the **Hotline** key.
2. Announce the call and hang up.

- OR -

Hang up to have the call wait at your Hotline partner unannounced.

 *If unanswered, the call recalls like a regular transferred call.*

To answer a call from your Hotline partner:

1. If you hear two beeps, speak toward the telephone.

- OR -

If your telephone rings, lift the handset.

Howler Tone Service

Description

Howler Tone Service provides a Howler Tone when a station remains off-hook after a call is completed or when a station is off-hook and digits are not dialed in a programmed time.

Conditions

Howler tone is generated 30 seconds after a call is disconnected and the telephone is left off-hook or the telephone is left off-hook without dialing.

Default Setting

Enabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

None

Programming

Changing either of these two programs will require a KSU reset for the changes to take affect.

- ➔ **80-01-01 Service Tone Setup – Repeat Count**
Set repeat count for tone 16 Lockout, default = 0 (endless).
Refer to [Table 2-26 Service Tone Setup, Program 80-01-01 on page 2-476](#).
- ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
(Default: 0).
Refer to [Table 2-27 Service Tone Setup, Program 80-01-02 on page 2-480](#).
- ➔ **80-02 DTMF Tone Setup**
Define the Basic Tone, duration (on time) and pause (off time) for Service Tone 16 Lockout. This option affects all internal stations system wide. Make separate entry for duration and pause. It is also possible to adjust both high and low frequency tones.

Operation

Not applicable.

Intercom

Description

Intercom gives extension users access to other extensions, and provides the system with complete internal calling ability.

Handsfree Answerback/Forced Intercom Ringing

Handsfree Answerback permits an extension user to respond to a voice-announced Intercom call by speaking toward the telephone, without lifting the handset. Like Handsfree, this is a convenience for workers who do not have a free hand to pick up the handset. Refer to [Handsfree Answerback/Forced Intercom Ringing on page 2-485](#) feature for more information.

Busy Status Display

When a display multiline terminal user places an Intercom call to a busy extension, the details of the busy status (who it talking to the extension or which line is in use by the extension) can be displayed. The details of the trunk busy status (the extension using the line) can be displayed after trying to access the trunk. This feature provides a user information which can determine whether or not they should use Barge-In for the extension or trunk. This information automatically displays for a multiline terminal once programmed.

Conditions

- Preventing ICM calls does not affect dialing other service codes, including 911.
- Intercom calls can ring or be voice-announced at the called extension.
- Intercom Abandoned Call Display remembers the last five Intercom calls to an extension.
- Ringing Line Preference can automatically answer ringing Intercom or trunk calls when the user lifts the handset.
- An extension can have a name assigned that identifies the extension to callers.
- Dialing 9 or any other trunk access code after dialing an extension results in termination of the Intercom call and a trunk is seized.

Default Setting

Enabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Handsfree Answerback/Forced Intercom Ringing

Intercom

Line Preference

Name Storing

Programming

For Intercom:

- **11-11-13 Service Code Setup (for Setup/Entry Operation) – Display Language Selection for Keypad**
Select the service code to be used at an extension to change the displayed language on a multiline terminal display (default: 678).
- **15-02-01 Multiline Telephone Basic Data Setup – Display Language Selection**
Select the language to be displayed on a multiline terminal display.

0 = Japanese	5 = Spanish	10 = Swedish
1 = English	6 = Dutch	11 = Turkish
2 = German	7 = Portuguese	12 = Latin American Spanish
3 = French	8 = Norwegian	
4 = Italian	9 = Danish	
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

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-
- ➔ **20-08-01 Class of Service Options (Outgoing Call Service) – Intercom Calls**
Enable (1)/Disable (0) an extension user ability to place Intercom calls.
 - ➔ **20-08-11 Class of Service Options (Outgoing Call Service) – Protect for the Call Mode Switching from Caller**
When an extension is set to ring mode for ICM calls, enable this option to prevent callers from changing the call to voice announce mode [Disable (0)/Enable (1)].
 - ➔ **20-11-07 Class of Service Options (Hold/Transfer Service) – Transfer Without Holding**
Enable (1)/Disable (0) an extension user ability to Transfer a call without holding. If the Busy Status Display feature is used, set this option to 0.
 - ➔ **20-13-22 Class of Service Options (Supplementary Service) – Busy Status Display (Called Party Status)**
Enable (1)/Disable (0) an extension user ability to use the Busy Status Display feature.
 - ➔ **20-17-01 Operator Extension – Operator’s Extension Number**
Assign the operator’s extension. Default is 101.
 - ➔ **20-18-01 Service Tone Timers – Extension Dial Tone Time**
Set the Extension Dial Tone Time (0~64800 seconds). After getting Intercom dial tone, a multiline terminal user has this time to dial the first digit of the Intercom call. Default is 30.
 - ➔ **21-01-02 System Options for Outgoing Calls – Intercom Interdigit Time**
Set the Intercom Interdigit Time (0~64800 seconds). When placing Intercom calls, users must dial each digit within this time. Default is 10.
 - ➔ **82-01-01 Incoming Ring Tone – Frequency 1**
Customize the Intercom ring tone.
 - ➔ **82-01-02 Incoming Ring Tone – Frequency 2**
Customize the Intercom ring tone.
 - ➔ **82-01-03 Incoming Ring Tone – Modulation**

For Handsfree Answerback/Forced Intercom Ringing:

- ➔ **20-02-12 System Options for Multiline Telephones – Forced Intercom Ring (ICM Call Type)**
Determine if an extension ICM calls should be set as Forced Intercom Ringing (1). If disabled (0), Intercom calls voice-announce. For all NEC Cordless telephones, this option must be set to 1 as voice announce is impossible. Default is 0.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
Default: Assign 101 as Class 15 and all others as Class 1.
- ➔ **20-08-10 Class of Service Options (Outgoing Call Service) – Signal/Voice Call**
Enable (1)/Disable (0) an extension user ability to toggle between Handsfree Answerback and Forced Intercom Ringing for outgoing Intercom calls (dial 1). Default is 1.

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-
- ➔ **20-09-05 Class of Service Options (Incoming Call Service) – Signal/Voice Call**
Enable (1)/Disable (0) an extension user ability to set Handsfree Answerback (Service Code 721) and Forced Intercom Ringing (Service Code 723) for incoming Intercom calls. Default is 1.
 - ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Enable (1)/Disable (0) an extension user ability to automatically send Off-Hook Signals while busy on a handset call. Setting this option to 0 prevents Intercom calls from providing Off-Hook Signaling – instead the caller hears a busy signal. Default is 1.
 - ➔ **80-01-01 Service Tone Setup – Tone 28 (Speaker Monitor Tone)**
Change the tone the originator of an ICM call hears. (The tone cannot be changed for what is heard by the user when receiving an ICM call.)
Refer to [Table 2-26 Service Tone Setup, Program 80-01-01 on page 2-476](#) (Service Tone #28).
 - ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
Refer to [Table 2-27 Service Tone Setup, Program 80-01-02 on page 2-480](#).

Operation

To place an Intercom call:

1. At multiline terminal, press **Speaker**.
- OR -
At single line telephone, lift the handset.
2. Dial extension number (or **0** for your operator).
 - 📎 *Your call may voice-announce or ring the called extension. Dial 1 to change the way your call alerts the called extension.*
 - 📎 *If the extension you call is busy or does not answer, you can dial another extension without hanging up.*

To answer an Intercom call:


1. If you hear two beeps, speak toward telephone.
 - 📎 *Your telephone picks up your voice.*
- OR -
- If your telephone rings, lift the handset.

To check the extension data (multiline terminal only):

1. Press the **Help** key.

2. Dial the extension number.

 *You display shows your telephone's extension number, port number and extension/Department Group.*

 *You can also check any other extension numbers information y pressing Help + the extension number.*

3. Press **Exit** to return the normal time/date display.

To change how Intercom calls ring the extension:

1. Press **Speaker** or lift the handset.

2. Dial **723** to have calls ring your extension.

- OR -

3. Dial **721** to have calls voice announce to your extension.

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Internal Hub

Description

A network switch is a computer networking device that connects network segments. It uses the logic of a network bridge but allows a physical and logical star topology. The switch determines how and to which port the data is forwarded, based on layer 2 MAC addresses. It is often used to replace network hubs. A switch is also often called an intelligent hub.

Switches, unlike hubs, use *microsegmentation* to create collision domains, one per connected segment. This way, only the Ethernet devices which are directly connected via a point-to-point link, or directly connected hubs are contending for the medium. By eliminating the possibility of collisions, full-duplex point-to-point connections on the switch become possible.

When multiple ETUs requiring Ethernet data connections are installed in an Electra Elite IPK II KSU, the HUB ETU can provide a neat and simple installation.

The HUB(8)-U() ETU is an in-skin, fast Ethernet switching hub unit that provides the following services:

Basic LAN Settings:

Auto Negotiation

This feature can allow/deny Auto Negotiation for port speed and duplex mode for each port. When this function is allowed, the HUB(8)-U() ETU negotiates with linked equipment to determine the speed (10Mbps or 100Mbps) and operating duplex mode (Full or Half).

Link Speed Selection

Data switching speed can be manually set to 10Mbps or 100Mbps for each port. This feature is allowed only when Auto Negotiation is denied.

Duplex Mode Selection

Duplex mode can be manually set to Half Duplex or Full Duplex for each port. This feature is allowed only when Auto Negotiation function is denied. Currently only Auto Negotiate and Half Duplex settings are supported on IPK II.

MDI/MDIX Mode Selection

MDI/MDIX mode can be set to MDI, MDIX, or Auto detection. When this mode is set to Auto detection, the HUB(8)-U() ETU can automatically determine whether or not it needs to crossover between pairs so that an external crossover cable is not required. If Auto Negotiation for port configuration is not allowed, this Auto MDIX mode must not be used. Only fixed MDI or MDIX mode can be used.

Back Pressure

For congestion on a half-duplex port, back pressure flow control is supported. Back pressure generates collision-detection signals in the CSMA/CD MAC-layer (media access control) protocol of Ethernet when a port is overloaded. These collision-detection signals from the congested port trick the sending port into believing a collision is imminent, and cause it to back off. This, in turn, forces a throttling back of the LAN segments experiencing congestion. When the sending ports try to transmit after the standard Ethernet delay time, the congested port may again send collision-detection signals or accept the incoming frames if ready.

 Flow Control

For congestion on a full-duplex port, the ETU supports IEEE 802.3x flow control that allows a device to momentarily stop the flow of traffic from a transmitting device, by sending a pause frame.

VLAN Configuration

 VLAN Tagging

A Virtual LAN (VLAN) subdivides a local area network by logical, rather than physical or geographical, means. For example, voice traffic can be put on one VLAN, and data traffic can be put on another VLAN. Ports on the same switching hub assigned to different VLANs are logically separated, and a router is required to pass traffic from one VLAN to another. A router would be put on a port configured to allow traffic from multiple VLANs.

Quality of Service

-
- Using the Quality of Service (QoS) related memory blocks, the VLAN carrying voice traffic could be assigned to the high priority queue and data traffic to other VLANs destined for the low priority queue.

Traffic with VLAN tags already assigned by the sending device can be directed to either the High Priority queue or the Low Priority queue according to thresholds to be determined for each HUB port.

Traffic without VLAN tags may be tagged with a default priority assigned to that port.

Port Mirroring

-
- For troubleshooting network issues with a protocol analyzer, port mirroring can be enabled to have traffic on a source port reflected to the target port.

Conditions

- When Auto Negotiation is denied and port speed is set to 100Mbps, the yellow LED (located on the RJ45 connector) is **ON**. When port speed is set to 10Mbps, this LED is **OFF**.
- Only one HUB(8)-U() ETU can be assigned and installed per cabinet.

Default Setting

None

System Availability

Terminals

None

Required Components:

HUB(8)-U() ETU

Related Features

None

Programming

- **85-01-01 Hub(8) LAN Setup – Auto Negotiation**
Enable (1)/Disable (0) Auto Negotiation. When enabled; Link Speed, Duplex and MDI/MDIX settings are negotiated with the other connected device. Default is 0.
- **85-01-02 Hub(8) LAN Setup – Link Speed**
Set the Link Speed to 10Mbps (0) or 100Mbps (1) if Auto Negotiation is disabled in PRG 85-01-01. Default is 1.
- **85-01-03 Hub(8) LAN Setup – Half Duplex/Full Duplex**
Set the Duplex to Half (0) or Full (1) if Auto Negotiation is disabled in PRG 85-01-01. Default is 1.
- **85-01-04 Hub(8) LAN Setup – Auto MDI/MDIX**
When PRG 85-01-01 is enabled, Auto MDI/MDIX functions. Auto MDI/MDIX (Media Dependent Interface/Media Dependent Interface Crossover) enables a switch port to sense the appropriate transmit/receive pairs of an Ethernet cable. Default is 2.
- **85-01-05 Hub(8) LAN Setup – Back Pressure, Half Duplex**
When Enabled (1), a switch applies back pressure to a half-duplex ingress port while an output queue is congested. (Only applicable when PRG 85-01-01 is disabled and PRG 85-01-03 is half-duplex.) Default is 0.

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- ➔ **85-01-06 Hub(8) LAN Setup – Flow Control, Full-Duplex**
Full-Duplex – 802-3x. When Enabled (1), a switch applies flow control to a full-duplex ingress port while an output queue is congested. (Only applicable when PRG 85-01-01 is disabled and PRG 85-01-03 is full-duplex.) Default is 0.
 - ➔ **85-02-01 Hub(8) VLAN Setup – VLAN Mode**
Enable (1)/Disable (0) VLAN. Default is 0.
 - ➔ **85-02-02 through 85-02-09 Hub(8) VLAN Setup – Port VLAN ID**
Define the VLAN ID for each port (1~4095). Default is 1 for all programs.
 - ➔ **85-03-01 Priority Setup – Default Priority**
Disable (0) assigns untagged frames to either the Low (2) or High (3) queue and tags them with a priority assigned in PRG 85-03-03 (high) or PRG 85-03-04 (Low). Default is 0.
 - ➔ **85-03-02 Priority Setup – RX High**
Establish the minimum threshold (1~7) for frames designated for the high priority queues. Default is 1.
 - ➔ **85-03-03 Priority Setup – TX High**
When PRG 85-03-01 is set to High, untagged frames are marked with this priority setting (1~7). Previously tagged frames are unchanged. Default is 7.
 - ➔ **85-03-04 Priority Setup – TX Low**
When PRG 85-03-01 is set to Low, untagged frames are marked with this priority setting (1~7). Previously tagged frames are unchanged. Default is 0.
 - ➔ **85-04-01 Port Mirroring Setup – Port Mirroring**
Enable (1)/Disable (0) Port Mirroring. Default is 0.
 - ➔ **85-04-02 Port Mirroring Setup – Source Port**
Define the port to be monitored (1~8). Default is 1.
 - ➔ **85-04-03 Port Mirroring Setup – Target Port**
Define the port where the monitored ports data should be simultaneously transmitted (1~8). Default is 1.
 - ➔ **85-05-01 Hub(8) VLAN Group Settings – VLAN ID**
Define the 802.1q VLAN ID(0~4095) . Default is 0.
 - ➔ **85-05-02 Hub(8) VLAN Group Settings – Port**
Input data represents the eight physical ports of the Hub(8), numbered from left to right as 8 to 1 (00000000~11111111). Setting a port to 1 enables the port to allow traffic from the VLAN ID specified in PRG 85-05-01. Default is 00000000.
 - ➔ **85-05-03 Hub(8) VLAN Group Settings – Tag Egress Retention**
Input data represents the eight physical ports of the Hub(8), numbered from left to right as 8 to 1 (00000000~11111111). At the point of egress, should VLAN tags be retained or removed? (Many Ethernet device do not process VLAN tagged packets.) Setting a port to 1 permits VLAN tags to be retained. Default is 00000000.
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Operation

None

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IP Extenders/Mobile ConneX

Enhancements

Version 2000 or higher is required for the Mobile ConneX feature.
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Description

The IP Extenders allow for a Digital Station to be extended to anywhere in the world, allowing seamless connection and functionality with the Elite IPK II system.

The following are some supported features for Extended Extensions:

- Place and receive calls
- Use Speed Dials
- Receive and send Caller ID
- Make Conference Calls
- Transfer calls
- Integrate with Voice Mail
- Act as an ACD Agent
- Use some adapters on the Digital Extension

Gateway Unit

The IP Extender-Gateway is the central receiving point for routing voice conversations from one or more locations to the IPK II. Extended Extension can place a call through the IPK II by dialing a number as if the user was in the main office.

- Two types of Gateway units are available – dual port (two ports) or 8-port Gateway.
- Gateways can connect to Branch units via an IP connection.
- Gateways can connect to Branch units via a WAN connection (8-port Gateway to 8-port Branch only).
- The Gateway only has two WAN connections (8-port Gateway only).
- The Gateway uses a proprietary protocol.

- Supported compression rates and bandwidth are used. This is defined on a per-port bases.

	G.7.11	ADPCM 32	ADPCM 24	G.729A
ISDN Connection	64 (72) Kbps	32 (40) Kbps	24 (32) Kbps	8 (16) Kbps
IP Connection	90 Kbps	60 Kbps	50 Kbps	30 Kbps

Branch (Remote) Unit

The IP Extender-Branch is the remote unit to which the extended extension is connected to. There is an optional port to allow for a local PSTN to be connected to allow for emergency and local calls without going through the main office.

- Four types of Branch units are available: 8-port, dual port (two ports), single port with PSTN, and Single port without PSTN Branch.
- A local PSTN line can be assigned to ring and flash on a line key, flash only on a line key, or neither.
- Emergency calls (911) are set to automatically dial out the local PSTN line. (Dual and single port Branches only). Emergency calls (911) can be set to go through the main system.
- The branch unit can connect to a Gateway unit via an IP Connection.
- The branch unit can connect to a Gateway unit via WAN connections (8-port branch to 8-port Gateway only).
- The branch unit uses a proprietary protocol.
- Supported compression rates and bandwidth are used. This is defined per port.

	G.7.11	ADPCM 32	ADPCM 24	G.729A
ISDN Connection	64 (72) Kbps	32 (40) Kbps	24 (32) Kbps	8 (16) Kbps
IP Connection	90 Kbps	60 Kbps	50 Kbps	30 Kbps

- APA, APR and HFU adapters are supported on the extended telephones.

Mobile ConneX

Mobile ConneX allows users to have their cell phone ring when their extension is called. When the call is answered, it allows the user to have access to system features. The cell phone user can also call into the system and is then able to dial internal and external numbers (**an 8-port Gateway and Version 2000 or higher are required**).

Mobile ConneX users can access the following system features:

- Transfer Calls (##T)
- Place Calls on Hold (##H)
- Make Conference Calls (##C)
- Make Internal Calls (** + internal extension)
- Make Trunk Calls (** + 9+ number)

The following conditions apply for Mobile ConneX:

- Two IP Gateway ports are required.
- Channels 1&2, 3&4, 5&6, and 7&8 are married together for Mobile ConneX.
- Mobile ConneX can tell you when you have new voice mail. (Only when dialing into IVR).
- An e-mail can be sent when a voice mail is left.
- A user can call in and enable or disable the Mobile ConneX.
- A user can change the number for the Mobile ConneX to call.
- Mobile ConneX ports and Extended Extensions can be on the same Gateway unit.

Default Setting

Not set up at default

System Availability

Terminals

All multiline display terminals and DTH-4R-2

Required Software

Version 2000 or higher (Mobile ConneX)

Required Components:

- Digital Station Card (ESI, ESIB/E)

Related Features

Electra Elite IPK Terminals

Programming

Extended Extensions

- ➔ **10-03-01 ETU Setup**
Terminal Type (B1) for ESI
- ➔ **15-07-01 Programmable Function Keys**
Assign function keys on multiline telephones of telephone that is extended.

Mobile ConneX

- ➔ **10-03-01 ETU Setup**
Terminal Type (B1) for ESI
- ➔ **15-07-01 Programmable Function Keys**
Assign VE/SE keys (*03), CAP keys (*08), and DSS/BLF keys (01 + Extension) on Mobile ConneX ports. Assign function keys on multiline telephones of telephone that is extended.
- ➔ **15-02-21 Multiline Telephone Basic Data Setup – Virtual Extension Access Mode (when idle Virtual Extension key is pressed)**
Determine whether an extension Virtual Extension key should be used as a DSS key to the extension and for receiving calls (0), answering incoming calls and ability to place outgoing ICM or CO calls (1), or just receiving incoming calls (2). If the key is to be used for outgoing calls, the extension number of the key must be a real extension or virtual extension number. When the extension number of the key is a real extension number, and the key is pressed, the real extension cannot be used. Must be set to 1.
- ➔ **15-09-01 Virtual Extension Ring Assignment**
Individually program an extension Virtual Extension key(s) to either ring (1) or not ring (0).
- ➔ **15-18-01 Virtual Extension Key Enhanced Options – Virtual Extension Key Operation Mode**
Assign if a call to a Virtual Extension holds (1) on the Virtual Extension or release (0) to the telephone that answered the Virtual Extension. Must be set to 1.

Operation

Please refer to the IP Gateway Branch Manual(s) for more information.

IP Station (MEGACO) – IAD (Integrated Access Device)

Enhancements

This feature was added with **Version 1100**.

Version 1500 or higher allows IP stations, which have call forwarding set, to continue forwarding calls, even if the station loses power or LAN connection.

Description

The IAD(8)-U() ETU with IP Station (MEGACO) application loaded is an optional interface package for the Electra Elite IPK II system that converts digital station ports into MEGACO IP station ports.

An on-board 10 Base-T/100 Base-TX connector provides a WAN/LAN connection. Voice and signaling data from/to the IP stations are converted into IP packets and transmitted through the Data Communication IP Network Intranet or Internet. The IAD(8)-U() ETU supports station to station direct RTP connections (peer-to-peer) for calls between IP telephones. Each IAD(8)-U() ETU supports up to eight IP telephones.

The IAD ETU contains a regular TCP/RTP/IP stack that can handle real time media, support industry standard MEGACO (H.248) communication on the WAN side, and interface with the Electra Elite IPK as a regular Electronic Station Interface board [ESI(8)-U() ETU].

From the network administration perspective, the IAD ETU is an end point on the IP network.

This interface can provide:

- MEGACO (H.248) signaling Protocol
- DTMF generation
- RTP port number designation
- ToS field QoS support
- Tone generation
- General Tone detection
- G.711 and G.729a voice compression
- 10 Base-T/100 Base-TX LAN interface
- Echo Cancellor
- Remote configuration and maintenance

NAT Beater – Network Address Translation

For this feature, the IAD(8)-U() ETU is installed and assigned as an ESI(8)-U() ETU. Each IAD(8)-U() ETU supports up to eight IP Phones and reduces the maximum capacity of stations in the system by eight.

A maximum of 14 IAD(8)-U() ETUs can be installed supporting the maximum of 112 stations (refer to [Table 2-44 KSU Power-Based Calculator Chart on page 2-1044](#) in the [Universal Slots](#) feature).

The IAD(8)-U() supports only those vocoders that are approved to provide toll-quality speech path. The following voice compression methods are supported for the IP Station (MEGACO) application:

- G.711 μ Law – Highest Bandwidth
- G.729 – Mid-Range Bandwidth

Each voice call requires at least the specifications listed in the following table. This includes all the overheads of VoIP communication, including signaling).

Codec	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G711 μ Law	90 Kbps	90 Kbps	20 ms*	1.5 ms	2 datagrams (40 ms)	4.4
G729	34 Kbps	34 Kbps	20 ms*	15.0 ms	2 datagrams (40 ms)	4.07

* When an IP Soft Phone is connected, set Time Between Packets to 100 ms.

- The IAD(8)-U() ETU contains a regular TCP/RTP/IP stack that can handle real time media. The IAD ETU is an end-point on the IP network from the network administration perspective.
- The IAD(8)-U() ETU uses MEGACO (H.248) Protocol to provide telephony services between remote stations through the IP Network. This is an IETF/ITU standards-based protocol. Currently, this protocol does not natively allow communications across networks where Network Address Translation (NAT) is performed.
- Speech-connection audio quality depends greatly on the available bandwidth between the stations in the data network. Because Internet is an uncontrolled data network compared to an Intranet, using this application in Intranet WAN environment with known (or controlled and assured) Quality of Service (QoS) is highly recommended.
- When any network where the IAD(8)-U() ETU or any of its IP terminals is connected uses NAT for connecting the voice calls (including firewall), consulting your network provider/administrator to specifically request service for VoIP or real-time media support on your networks is recommended.
- A static IP Address is required for each IAD(8)-U() ETU in the network.

- ❑ The IAD(8)-U() ETU supports station to station direct RTP connections (peer-to-peer) for calls between IP Phones that provide the following benefits:
 - Improved voice quality for calls between two IP Phones.
 - Improved bandwidth performance for voice between IP Phones.
 - The operating mode can be configured per IAD(8)-U() ETU on each port.
 - Pad and echo canceller controls are built-in and are programmable in the IAD(8)-U() ETU.
 - The LAN/WAN connection is provided by a 10 Base-T/100 Base-TX Ethernet.
- ❑ In voice communication, particularly Internet telephony, the **Mean Opinion Score (MOS)** provides a numerical measure of the quality of human speech at the circuit destination. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. The maximum MOS is 5.0.

Conditions

- The Synchronous Ringing feature is not supported for IP Terminals. The ITH-16D-2/3 and ITH-8D-2/3 Multiline Terminals have user programmable ringing options. Group listening is not supported by (MEGACO) IP stations.
- Station Relocation is not supported by (MEGACO) IP stations or IPK Soft Phones.
- Not all data networks can support Voice over Internet Protocol (VoIP). A good VoIP network solution requires low-latency, low jitter, and low packet loss. Accordingly, the network must be evaluated for these factors to determine whether or not it can provide toll-quality speech paths.
- When 5-, 6-, or 7-digit station numbers are used, this feature is not available.
- Night Mode is indicated as **N** on an IPK Soft Phone.
- If call forwarding is set for an IP station, the system continues to follow the call forwarding, even if the station loses power or LAN connection.
- If both Call Forward Busy and Call Forward No Answer are set, the system follows the Call Forward No Answer setting, even if the station loses power or LAN connection.
- MEGACO station registration is solely dependant upon extension registration and not MAC registration. Program 15-05-02 can only be viewed after an IP station has been registered (read only).

Default Setting

None

System Availability

Terminals

ITH-16D-2/3 Multiline Terminal

- OR -

ITH-8D-2/3 Multiline Terminal

- OR -

Any IPK Multiline Terminal with IP-R (IPK) Unit installed.

- OR -

IPK Soft Phone

Required Software

Version 1100 or Higher

Required Component(s)

- IAD(8)-U() ETU with IP Station (MEGACO) application loaded.
- AC-R UNIT (AC Adapter) (When in-line power is used, Power over Ethernet (POE), is not provided.)
- IP Terminals – must be updated to **release 1.25 or higher** to support this application.
- IP Terminals – must be updated to **release 1.23 or higher** to support this application.

Related Features

None

Programming

- ➔ **11-02-01 Extension Numbering**
Assign extension numbers to extension ports. The telephone programming identity follows the port number – not the extension number.



Operation

None

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IP Station (MEGACO) – MG 16

Enhancements

Version 1500 or higher:

- Allows IP stations, which have call forwarding set, to continue forwarding calls, even if the station loses power or LAN connection.
- The Registration Override feature gives users access to their IP telephone from any location by using the override login function. Users have the flexibility of logging into their IP Station in the office as well as remotely at the home office. Refer to [Registration Override for MEGACO Phones on page 2-557](#) in the Operation section.
- The Center Download feature provides users with the ability to locally update the firmware on IP Terminals/Adapters. Each IP Terminal/Adapters can download the firmware through a file server, thus reducing time and cost required to update each Terminal individually. The Center Download feature, when configured completely, allows the MG16 to communicate with the IP Terminals via DRS listening port 161, gathering the IP Terminal Firmware information. The MG16 then sends an Initial Setting Request to the Elite IPK II CPUII. The CPUII compares both the IP Phone firmware versions and the information programmed in system data to verify if the need for firmware is required on the IP Terminal. If a firmware upgrade is required, the CPUII sends a download instruction. Refer to [Firmware Download Setup on page 2-546](#) in the System Data programming section and to [To Configure an IP Terminal for Center Download on page 2-558](#) in the Operation section.
- The MG16 web server has a new HTTP page to display the network statistics based on values entered in PRG 84-06.
- A 4-port license registration provides users with the flexibility of adding (MG16) DSP resources in increments of four (4, 8, 12 or 16).
- The Elite IPK II supports Echo Cancellation for both stations and trunks. Type 4 refers to analog trunks, Type 5 refers to both digital trunks and MEGACO stations.

Version 2000 and higher:

- Supports station numbers to be assigned by the 10s group for 4-digit station numbers, 100s group for 5-digit, 1,000s group for 6-digit station numbers, and 10,000s group for 7-digit station numbers.
- Supports the Combined functionality of a MG16 and K-CCIS over IP and SIP Stations/trunks on one card.

DESCRIPTION

The Media Gateway (MG16) is a IP Application loaded on the PVA(X)-U() ETU. This is an optional interface package for the Electra Elite IPK II system that supports MEGACO IP stations.

An on-board 10 Base-T/100 Base-TX connector provides a WAN/LAN connection. Voice and signaling data to/from the IP stations are converted into IP Frames and transmitted through the Data Communication IP Network Intranet or Internet. Media Gateway 16 supports station-to-station direct RTP connections (peer-to-peer) for calls between IP telephones. Each Media Gateway 16 application can support up to 16 TDM Talk paths. Although, one Media Gateway Card can support 256 IP MEGACO Stations, but only provides 16 simultaneous talk paths across the TDM highway.

The MG Application contains a regular TCP/RTP/IP stack that can handle real-time media, supports industry standard MEGACO (H.248) communication on the WAN side, and interfaces with the Electra Elite IPK II.

For this feature, the Media Gateway 16 is installed and assigned as a VoIP MG16 ETU. Each Media Gateway 16 supports IP signaling for up to 16 Phones and reduces the maximum capacity of IP stations in the system by 16.

There can be only one Media Gateway Controller assigned in the Electra Elite IPK II. This determination is made in KSU Programming [PRG 84-05-03] Master/Slave determination. The Media Gateway Controller interprets UDP signaling messages between the Elite IPK II Processor and IP Stations. Any additional MG16 ETUs added to the system are called Media Gateway cards. The media gateway card controls and interprets RTP messaging from the IP Phone to the Elite IPK II Processor.

If a non-IP Phone (e.g., POTS, multiline terminal), or trunk line is required, a DSP resource is needed. If while on a peer-to-peer call, DSP resources are not used and the MG16 port is not accessed, only Media Gateway Controller processing is used. If, while on a peer-to-peer call, a conference call is formed, the peer-to-peer connection is released and a new non peer-to-peer connection is created using the MG16 DSP resources – two ports are used on the MG16 ETU. If the third party drops out of the conversation, the call reverts to a peer-to-peer call.

A maximum of 16 PVA(X)-U() ETUs can be installed supporting the maximum of 256 IP stations (refer to [Table 2-44 KSU Power-Based Calculator Chart on page 2-1044](#) in the [Universal Slots](#) feature).


The MG supports only those vocoders that are approved to provide toll-quality speech path. The following voice compression methods are supported for the IP Station (MEGACO) application:

- G.711 μ Law – Highest Bandwidth
- G.729 – Mid-Range Bandwidth

Each voice call requires at least the specifications listed in the following table. This includes all the overheads of VoIP communication, including signaling).

Codec	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G711 μ Law	90 Kbps	90 Kbps	20 ms*	1.5 ms	2 datagrams (40 ms)	4.4
G729	34 Kbps	34 Kbps	20 ms*	15.0 ms	2 datagrams (40 ms)	4.07

* When an IP Soft Phone is connected, set Time Between Packets to 100 ms.

- The MG16 contains a regular TCP/RTP/IP stack that can handle real time media. The MG16 ETU is an end-point on the IP network from the network administration perspective.
- The MG16 uses MEGACO (H.248) Protocol to provide telephony services between remote stations through the IP Network. This is an IETF/ITU standards-based protocol.
- Speech-connection audio quality depends greatly on the available bandwidth between the stations in the data network. Because Internet is an uncontrolled data network compared to an Intranet, using this application in Intranet WAN environment with known (or controlled and assured) Quality of Service (QoS) is highly recommended.
- When connecting the voice calls across NAT/Firewall, consulting your network provider/administrator to specifically request service for VoIP or real-time media support on your networks is recommended.
- A static IP Address is required for each MG16 in the network.
 -  *The MG16 ETU and Elite IPK II CPU II must be configured to communicate on the same network. This may require a second static IP Address for the IPK II CPU II.*
- The MG16 supports station to station direct RTP connections (peer-to-peer) for calls between IP Phones that provide the following benefits:
 - Improved voice quality for calls between two IP Phones.
 - Improved bandwidth performance for voice between IP Phones.
 - Pad and echo canceller controls are built-in and are programmable via Elite IPK II KSU Programming.
 - The LAN/WAN connection is provided by a 10 Base-T/100 Base-TX Ethernet connector. This connector is a single 10/100 MPS Ethernet Connector. This port has Auto-Medium Dependent Interface Crossover (MDIX) to allow using either a straight-through Ethernet cable for connection to a PC or a crossover Ethernet cable.
 - The Ethernet Interface setting allows manual configuration of the Ethernet port from the Key System programming, and can be set to operate in the following port speed and duplex mode combinations: 10MB/Full Duplex, 10MB/HalfDuplex, 100MB/Half Duplex, or 100MB/Full Duplex, Auto-Negotiate via Elite IPK II KSU Programming.

- Registration Override is a feature for overriding the registration of an IP phone from various locations. You have to know the IP address of the IP phone and the extension number of the phone you want to Override.

In voice communication, particularly Internet telephony, the **Mean Opinion Score (MOS)** provides a numerical measure of the quality of human speech at the circuit destination. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. The maximum MOS is 5.0.

This interface supports via KSU configuration:

- MEGACO (H.248) signaling Protocol
- DTMF generation
- RTP port number designation
- TOS field QoS support
- Tone generation
- General Tone detection
- G.711 and G.729a voice compression
- 10 Base-T/100 Base-T LAN interface
- Echo Cancellor
- Registration Override

Conditions

- Group listening is not supported by (MEGACO) IP stations.
- Station Relocation is not supported by (MEGACO) IP stations or IPK Soft Phones.
- Not all data networks are suitable to support Voice over Internet Protocol (VoIP). ***A good VoIP network solution requires low-latency, low jitter, and low packet loss. Accordingly, the network must be evaluated for these factors to determine whether or not it can provide toll-quality speech paths.***
- When 6- or 7-digit station numbers are used, this feature is not available.
- Night Mode is indicated as **N** on an IPK Soft Phone.
- Soft Phone is not supported when using Registration Override.
- If call forwarding is set for an IP station, the system continues to follow the call forwarding, even if the station loses power or LAN connection.
- If both Call Forward Busy and Call Forward No Answer are set, the system follows the Call Forward No Answer setting, even if the station loses power or LAN connection.

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- Program 11-01-01 9: System Numbering – Dial Extension Analyze is not supported in **Version 1600**.
 - Program 11-20-01: Dial Extension Analyze Table – Dial and Program 11-20-02: Dial Extension Analyze Table – Type of Dials are not supported in **Version 1600**.
 - The IPK II CPU algorithm only compares IP terminals firmware against the files stored in the TFTP server. If the firmware comparison does not match, the firmware is automatically updated.
 - MEGACO station registration is solely dependant on extension registration and not MAC registration. Program 15-05-02 can be viewed only after an IP station is registered (read only).

PVA Combo Card Service Conditions

- **Version 2000 or higher** is required to support the PVA Combo Card Application.
- If changing package types on the PVA (for example MG16 to Combo), the previous card must be deleted from the slot in Program 90-05-01. Even if both are the same physical card, PVA, the system sees them differently.
- Echo Cancellation in Program 84-17-01 defines settings for Type 4 and Type 5. When dealing with MEGACO telephones, refer to Type 5, which also affects digital trunks.
- The PVA Combo Card Application can support 24 DSP resources which may be configured to support the MG16 or the CCISoIP application.
- The PVA Combo Card CCISoIP Trunks and SIP Trunks Logical Port Numbers are determined by the following:
 - ❑ **SIP Trunks**
 - PVA(X)-U() ETU License Registration
 - SIP Client License Registration
 - Elite IPK II Program 10-41-01 SIP Trunk Availability
 - Elite IPK II Program 10-40-02 Number of SIP Ports
 - ❑ **CCISoIP Trunks**
 - PVA(X)-U() ETU License Registration
 - Elite IPK II Program 10-40-03 CCISoIP Trunk Availability
 - Elite IPK II Program 10-40-04 Number of CCISoIP Ports
 - The maximum number of Logical Trunk Ports is 48 ports per PVA Combo Card package
- The PVA(X)-U() ETU with PVA Combo Card Application may configure DSP usage criteria with Elite IPK II Program 10-19-01 DSP Resources. Each VoIP ETU can flag individual DSP resources as:
 - ❑ IP Ext – IP Extensions (includes MEGACO Station or SIP Stations)

-
-
- SIP Trk – SIP Trunks
 - CCIS – CCIS Networking
 - Common – Common usage for CCIS Networking, SIP Station, MEGACO Stations, SIP Trunks
 - PVA Combo Card CCISoIP Application supports trunks configured in increments of four contiguous DSP resources.
 - ETU Configuration – The PVA Combo Card Application package is identified as a MG_CCIS ETU.
 - The PVA Combo Card – VoIP ETU supports up to 200 CCIS Logical Trunks and 200 SIP Logical Trunk Port Numbers (read only).
 - IP Application DSP Resource configuration:
 - CCIS Trunks are defined as CCIS or Common
 - IP MEGACO Stations are defined as IP Ext or Common
 - SIP Trunks are defined as SIP Trk or Common
 - SIP Station are defined as IP Ext or Common
 - When the PVA Combo Card is installed in the Electra Elite IPK II system, the system assigns the first available Logical Port Numbers for CCISoIP trunks and SIP trunks are allotted the next available logical port numbers.
 - If the Elite IPK II CPUII system does not have registered SIP licenses, the Logical SIP Trunk is not assigned.
 - If the PVA Combo Card cannot establish the common highway, the maximum number of trunk ports is 16 ports for one package.
 - Changes in Program 10-40-03 and or Program 10-40-04 requires the PVA(X)-U() ETU to be reset for the new changes to take effect.
 - Changes in Program 10-19-01 requires PVA(X)-U() ETU to be reset for the changes to take effect.
 - PVA(X)-U() ETU/Combo Card LED Indications:
 - Non Peer-to-Peer MEGACO IP Station and SIP IP Station calls are indicated by a Flashing RED LED
 - CCISoIP Trunk and SIP Trunk calls are indicated by a Solid RED LED

Default Setting

None

System Availability

Terminals

ITH-16D-2/3 Multiline Terminal

- OR -

ITH-8D-2/3 Multiline Terminal

- OR -

Any IPK Multiline Terminal with IP-R (IPK) Unit installed.

- OR -

IPK Soft Phone

Required Component(s)

PVA(X)-U() ETU with MG16 Application Package **1.01 or Higher**.

PVA(X)-U() ETU with MG16 Application Package **1.5 or Higher** to support Registration Override.

PVA(X)-U() IP Resource CD (stock number 750874) can be used for full (MG16) 16-DSP resource licensing.

PVA(X)-U() 4-Port IP Resource CD (stock number 750895) can be used for 4-port license increments.

AC-R UNIT (AC Adapter) [When in-line power is used, Power over Ethernet (POE), is not provided.]

IP Terminals – must be updated to **Release 1.27 or higher** to support this application.

IP Adapters – must be updated to **Release 1.27 or higher** to support this application.

PVA(X)-U() ETU with PVA Combo Application Package **1.00 or Higher** and Elite IPK II CPU II **Version 2000 or Higher**.

Related Features

None

Programming

- ➔ **10-16-03 Option Information Setup – TFTP Server (Version 1600 and 2100 or higher required)**
 IP Address
 0.0.0.0 ~ 126.255.255.254
 128.0.0.1 ~ 191.255.255.254
 (default: 0.0.0.0)

VoIP Settings:

When the PVA(X)-U() ETU is first purchased it has a factory default package installed. You have to load the MG16 Application for the Elite IPK II KSU to recognize the VoIP MG16 ETU. [Refer to the PVA(X)-U() ETUs Firmware upgrade installation procedure].

- ➔ **84-05-01 VoIP IP Address – IP Address**
 Define the IP Address of the VoIP Application supported (MG16, CCISoIP). (default: Slot 1: 172.16.0.20)
- ➔ **84-05-02 VoIP IP Address – LAN Interface**
 Define the LAN interface Speed and Mode of the VoIP Application supported (MG16, CCISoIP). (default: 0 = AUTO)
- ➔ **84-05-03 VoIP IP Address – Master/Slave**
 Define the Media Gateway Controller for the MG16 application.
 (default: 1 = MASTER)
☞ The Elite IPK II system supports only one Media Gateway controller (determined by setting the MG16 Application to Master).
- ➔ **84-05-04 VoIP IP Address – Subnet Mask**
 Define the Media Gateway Card Subnet Mask Address.
 (default: 255.255.0.0)
- ➔ **84-05-05 VoIP IP Address – Default Gateway**
 Define the MG16 Application Default Gateway Address.
 (default: 0.0.0.0)
- ➔ **84-06-01 VoIP Info – RTP Port Number**
 Define the Media Gateway Card starting RTP Port Number.
 (default: 10020)

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- ➔ **84-06-02 VoIP Info – RTCP Port Number**
Define the Media Gateway Card Starting RTCP Port Number . The RTCP Port Number has to be the (RTP port number + 1).
(default: 10021)
 - ➔ **84-06-04 VoIP Info – Fract Lost Threshold**
Define the fractional lost threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)
 - ➔ **84-06-05 VoIP Info – Packets Lost Threshold**
Define the packet lost threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)
 - ➔ **84-06-07 VoIP Info – Jitter Threshold**
Define the Jitter Threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)
 - ➔ **84-06-09 VoIP Info – Delay LSR Threshold**
Define the Delay threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)

VoIP Echo Cancellor Control Setup

The Elite IPK II supports Echo Cancellation for both stations and trunks. Type 4 refers to analog trunks, Type 5 refers to both digital trunks and MEGACO stations.

 *Changes made to Type 5 affect both MEGACO stations and digital trunks simultaneously.*

- ➔ **84-17-01 VoIP Echo Cancellor Setup – Echo Cancellor Mode**
Enable/Disable the Echo Cancellor.
Type 4 (default: 1: Enable)
Type 5 (default: 0: Disable)
- ➔ **84-17-02 VoIP Echo Cancellor Setup – Echo Cancellor Tail Size**
Select the time threshold for the Echo Cancellor.
Type 4 (default: 5: 128 ms)
Type 5 (default: 1: 8 ms)
- ➔ **84-17-03 VoIP Echo Cancellor Setup – Echo Cancellor NLP Mode**
Enable/Disable the NLP Mode.
Type 4 (default: 1: Enable)
Type 5 (default: 0: Disable)

VoIP ToS Setup

The Elite IPK II supports Quality of Service Marking for the Type of Service and DiffServe on the following VoIP Applications: (MG16 and CCISoIP packages)

- ➔ **84-10-01 ToS Setup – ToS Mode**
Define the logic used for marking the ToS Field.
(default: 0)
- ➔ **84-10-02 ToS Setup – Priority, IP Precedence**
Define the IP Precedence data bit.
(default: 0)
*Typically, only **one** of bits 3 ~ 6 are set to 1 and the other three bits are set to 0. For example, to maximize route reliability, set PRG 84-10-05 to 1, and leave PRG 84-10-03, PRG 84-10-04 and PRG 84-10-06 at 0.*
- ➔ **84-10-03 ToS Setup – Low Delay**
1 = Optimize for low delay routing.
(default: 0)
- ➔ **84-10-04 ToS Setup – Wideband (Throughout)**
1 = Optimize for high bandwidth routing.
(default: 0)
- ➔ **84-10-05 ToS Setup – High Reliability**
1 = Optimize for reliability routing.
(default: 0)
- ➔ **84-10-06 ToS Setup – Low Cost**
1 = Optimize for low cost routing
(default: 0)
- ➔ **84-10-07 ToS Setup – Priority (D.S.C.P. - Differential Services Code Point)**
Define the Diffserve – decimal value (valid values 0~63)
(default: 0)

Firmware Download Setup

Use this data program to configure the required settings for Center Download file server access.

- ➔ **84-07-01 Firmware Download Setting – Server Mode**
Select the mode type.
0 = TFTP, 1 = FTP
(default: 0)
- ➔ **84-07-02 Firmware Download Setting – File Server IP Address**
Set the IP Address for the file server.
(default: 0.0.0.0)

➔ **84-07-03 Firmware Download Setting – Login Name**
Assign the user login name.
(Up to 24 characters)

➔ **84-07-04 Firmware Download Setting – Password**
Assign the password.
(Up to 24 characters)

IP Terminal Version Information defines the hardware and firmware version of the IP Terminal and Adapter.

➔ **90-22-01 Terminal Version Information – Hardware Version**
Define the hardware of the IP Terminal.
(default: 00)

➔ **90-22-02 Terminal Version Information – Firmware Version**
Define the firmware of the IP Terminal.
(default: 00.00)

IP Phone Configuration

IP Telephone Terminal Basic Data Setup:

➔ **15-01-01 Basic Extension Data Setup – Extension Name**
Review the type protocol support by the IP phone.
(default: MEGACO)

➔ **15-05-02 IP Telephone Terminal Basic Data Setup – IP Phone Fixed Port Assignment**
Review the registered IP Phone Mac Address.
(default: 00-00-00-00-00-00)

➔ **15-05-06 IP Telephone Terminal Basic Data Setup – IP Phone Terminal Type**
Define the type of IP Phone being used – valid options are IP70, IP80 or IP Soft Phone.
(default: 0)

➔ **15-05-07 IP Telephone Terminal Basic Data Setup – Using IP Address**
Review the registered IP Phones IP Address (Informational Only).
(default: 0.0.0.0)

➔ **15-05-15 IP Telephone Terminal Basic Data Setup – CODEC Type**
Set the registered IP Phone Codec type – Reference PRG 84-11 Dterm IP Codec Basic Information.
(default: Type:1).

IP Phone Information Basic Setup:

➔ **84-03-06 IP Phone Information Basic Setup – Area Number**
Set the country of origin.
(default: 1:USA).

IP Extension Numbering:

➔ 11-02-01 Extension Numbering

Define the IP Phone extension Number.

This designated extension is used to register your IP Phone and is programmed IP Phone programming. Refer to the Installation Instructions for Elite ITH-4D/8D/16D-2/3 Dterm IPK Terminals and the IP-R (IPK) Adapters.

IP Phone Codec Information Basic Setup:

➔ 84-03-10 IP Phone Information Basic Setup – Signaling Server Port

(default: 5029).

➔ 84-11-01 *D*^{term} IP CODEC Information Basic Setup – G.711 Audio Frame

Define the G.711 audio Frame Size [1 = 10 ms, 2 = 20 ms, 3 = 30 ms].
(default: 3)

➔ 84-11-02 *D*^{term} IP CODEC Information Basic Setup – G.711 VAD Mode

Enable/Disable Voice Activity Detection for G.711.
(default: 0:Disable)

➔ 84-11-03 *D*^{term} IP CODEC Information Basic Setup – G.711 Type

Define the G.711 Type – μ -law is recommended when in USA.
(default: 1: μ -Law)

➔ 84-11-04 *D*^{term} IP CODEC Information Basic Setup – G.711 Jitter Buffer Min

Define G.711 Jitter Buffer minimum accepted value.
(default: 30 ms)

➔ 84-11-05 *D*^{term} IP CODEC Information Basic Setup – G.711 Jitter Buffer Average

Define G.711 Jitter Buffer average accepted value.
(default: 60 ms)

➔ 84-11-06 *D*^{term} IP CODEC Information Basic Setup – G.711 Jitter Buffer Max

Define G.711 Jitter Buffer maximum accepted value.
(default: 120 ms)

➔ 84-11-07 *D*^{term} IP CODEC Information Basic Setup – G.729 Audio Frame

Define the G.729 audio Frame Size (1 = 10 ms, 2 = 20 ms, 3 = 30 ms).
(default: 3=30 ms)

➔ 84-11-08 *D*^{term} IP CODEC Information Basic Setup – G.729 VAD Mode

Enable/Disable Voice Activity Detection for G.729.
(default: 0: Disable)

➔ 84-11-09 *D*^{term} IP CODEC Information Basic Setup – G.729 Jitter Buffer Min

Define G.711 Jitter Buffer minimum accepted value.
(default: 30 ms)

-
-
- ➔ **84-11-10** *D^{term}* **IP CODEC Information Basic Setup – G729 Jitter Buffer Average**
Define G.729 Jitter Buffer average accepted value.
(default: 60 ms)
 - ➔ **84-11-11** *D^{term}* **IP CODEC Information Basic Setup – G729 Jitter Buffer Max**
Define G.729 Jitter Buffer maximum accepted value.
(default: 120 ms)
 - ➔ **84-11-17** *D^{term}* **IP CODEC Information Basic Setup – Jitter Buffer Mode**
Define the Jitter Buffer mode – support Static or Immediate.
[default: 2 (Immediate)]
 - ➔ **84-11-18** *D^{term}* **IP CODEC Information Basic Setup – VAD Threshold**
Define the VAD Threshold Values. Consult the Electra Elite IPK II Programming Manual for Threshold scale to set acceptable values.
[default: 20: 0dB (-30dBm)]
 - ➔ **84-11-26** *D^{term}* **IP CODEC Information Basic Setup – TX Gain**
Define to TX Gain Values – Adjusting this value increases or decreases volume levels for the receiving party. Consult the Electra Elite IPK II Programming Manual for Transmit Gain scale to set acceptable value.
(default: 14)
 - ➔ **84-11-27** *D^{term}* **IP CODEC Information Basic Setup – RX Gain**
Define to RX Gain Values – Adjusting this value increases or decreases volume levels for the sending party. Consult the Electra Elite IPK II Programming Manual for Transmit Gain scale to set acceptable value.
(default: 14)
 - ➔ **84-11-28** *D^{term}* **IP CODEC Information Basic Setup – Audio Capability Priority**
(default: 0)

Operation

Example Configuration 1 – Static IP Addressing, One LAN:

This example shows System IP Phones connected to a single LAN (no routers) with a static IP address.

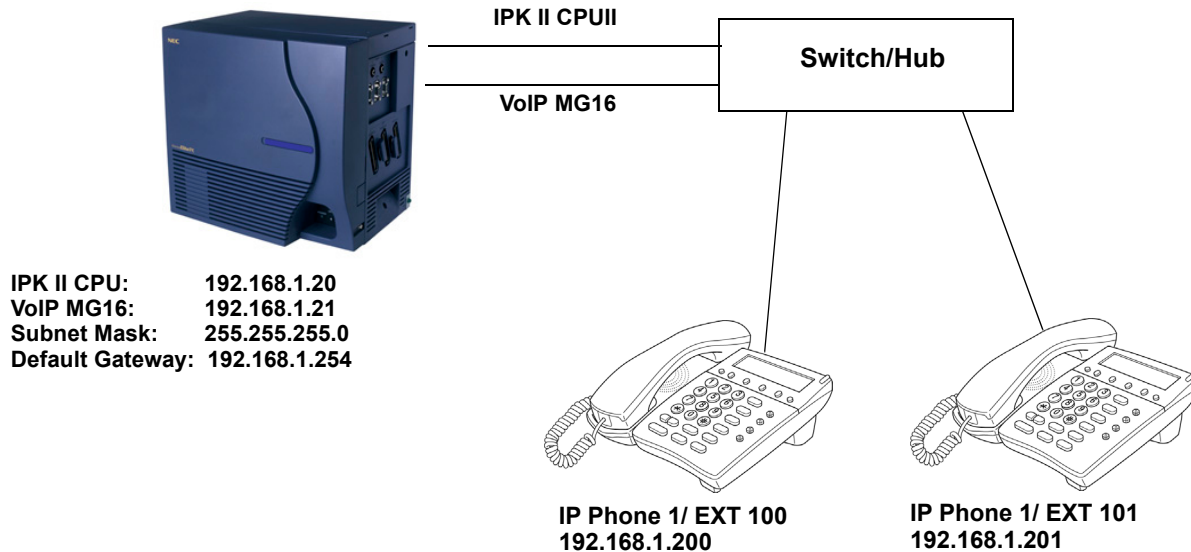


Figure 2-11 Example – Static IP Addressing, One LAN

Programming – IPK II CPUII

- 10-12-01 IPK II CPUII Network Setup – IP Address = 192.168.1.20
- 10-12-02 IPK II CPUII Network Setup – Subnet Mask = 255.255.255.0

Programming – VoIP MG16 ETU

- 84-05-01 VoIP IP Address Setup (Slot No.) = 192.168.1.21
- 84-05-02 LAN Interface = 0: Auto Detect
- 84-05-03 Master/ Slave = 1: Master
- 84-05-04 VoIP Subnet Mask = 255.255.255.0

Programming – System IP Phones

- DHCP Mode (Disabled)

2. IP Address = 192.168.1.200
3. Subnet Mask = 255.255.255.0
4. MGC Address = 192.168.1.21, Ext = 100

Example Configuration 2 – Dynamic IP Addressing, One LAN:

This example shows System IP Phones connected to a single LAN (no routers) with a dynamic IP address. The DHCP server could be:

- Customer supplied (i.e., Windows 2003 Server, etc.)
- Elite IPK II internal DHCP server

In this case, additional programming would be required.

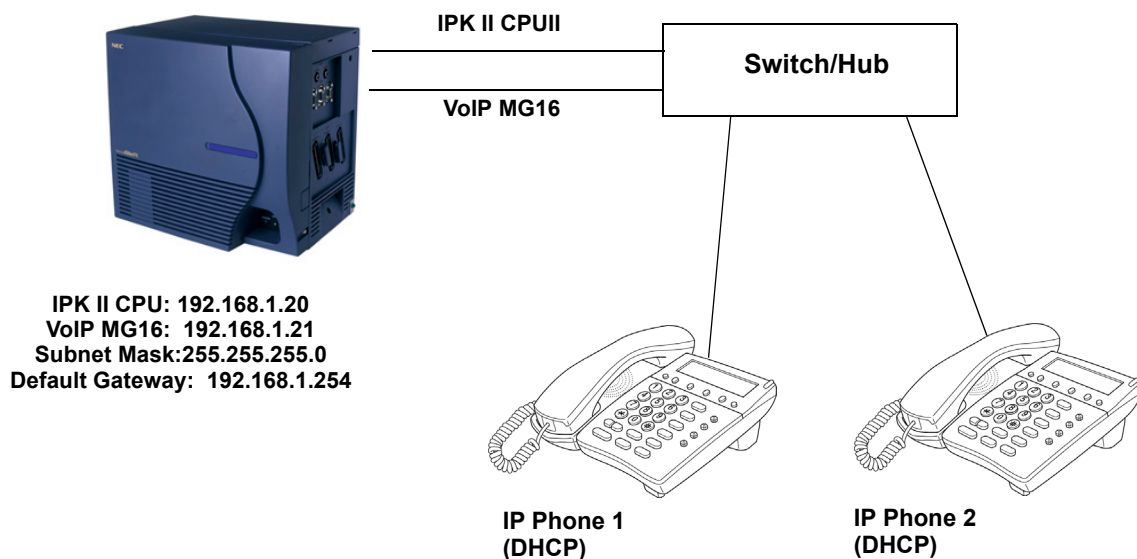


Figure 2-12 Example – Dynamic IP Addressing, One LAN

Programming – IPK II CPU II

1. 10-12-01 CPU II Network Setup – IP Address = 192.168.1.20
2. 10-12-02 CPU II Network Setup – Subnet Mask = 255.255.255.0

Programming – System IP Phones

1. DHCP Mode – Enabled

Example Configuration 3 – Static IP Addressing, Routed LAN:

This example shows System IP Phones connected to an Elite IPK II over a Wide Area Network (WAN) with static addressing. This is a typical scenario – a small branch office connecting to an Elite IPK II at the main office.

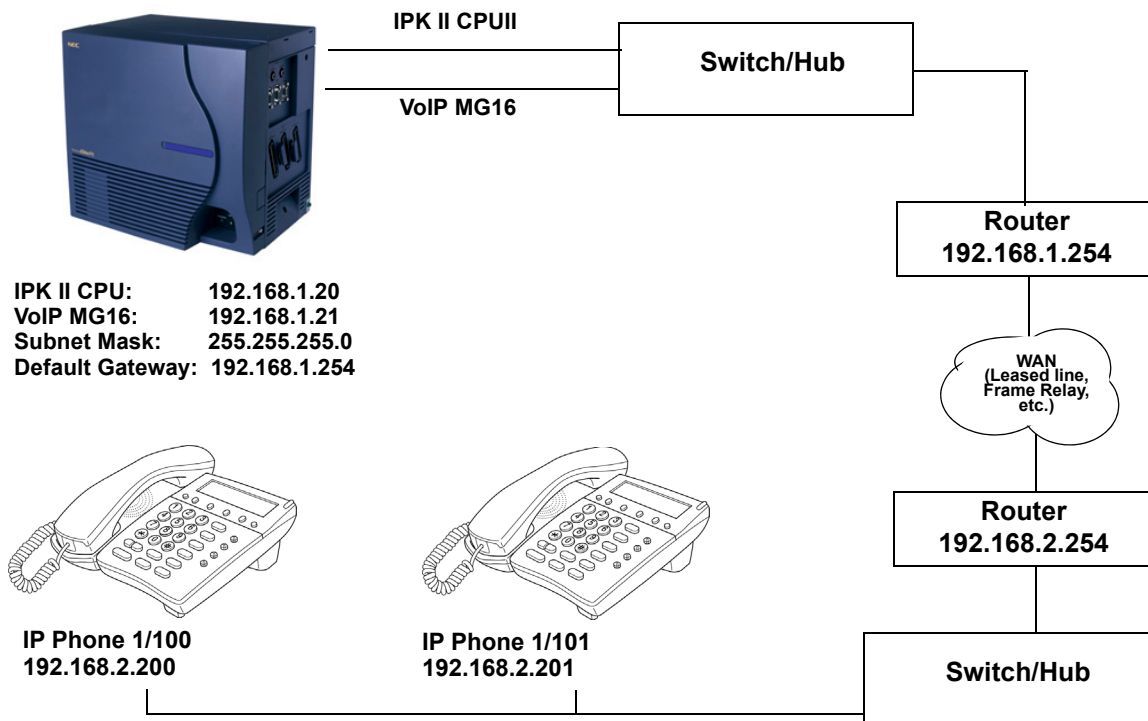


Figure 2-13 Example – Static IP Addressing, Routed WAN

Programming – IPK II CPUII

1. 10-12-01 CPUII Network Setup – IP Address = 192.168.1.20
2. 10-12-02 CPUII Network Setup – Subnet Mask = 255.255.255.0
3. 10-12-03 CPUII Network Setup – Default Gateway = 192.168.1.254

Programming – VoIP MG16

1. 84-05-01 VoIP MG16 IP Address Setup (Slot No.) = 192.168.1.21

Programming – System IP Phones

1. DHCP Mode – Disabled

2. IP Address = 192.168.2.200
3. Subnet Mask = 255.255.255.0
4. Default Gateway = 192.168.2.254

CODEC Selection

IP Phones can use various CODECs (**Coder/Decoder**). A CODEC is a standard for converting an analog signal to a digital signal. This conversion process is handled by the DSP (Digital Signal Processors) on VoIP MG16 cards. Each CODEC has different voice quality and compression properties.

The correct choice of CODEC is based on the available bandwidth, the number of calls required, and the voice quality required.

CODECs:

- G.729. Low bandwidth requirement – Usually used on most Wide Area Network links.
- G.711. High bandwidth requirement – usually used on Local Area Networks.

The Elite IPK II IP Phones have a preferred CODEC, called the Audio Capability Priority. This is the CODEC that is used when calls are initiated. There are five Types (profiles) that can be created and assigned to IP Phones.

This allows for different CODECs to be used for different groups of IP Phones. In most cases, all IP Phones belong to the same Type.

To create the CODEC Type (1 ~ 5), use the following programming:

IP Phone Codec Information Basic Setup:

1. 84-11-01 G.711 Audio Frame (default: 3: 30 ms)
2. 84-11-02 G.711 VAD Mode (default: 0 : Disable)
3. 84-11-03 G.711 Type (default: 1: μ -Law)
4. 84-11-04 G.711 Jitter Buffer Min (default: 30 ms)
5. 84-11-05 G.711 Jitter Buffer Average (default: 60 ms)
6. 84-11-06 G.711 Jitter Buffer Max (default: 120 ms)
7. 84-11-07 G.729 Audio Frame (1 = 10 ms, 2 = 20 ms, 3 = 30 ms) (default: 3)
8. 84-11-08 G.729 VAD Mode (default: 0: Disable)
9. 84-11-09 G.729 Jitter Buffer Min (default: 30 ms)

10. 84-11-10 G.729 Jitter Buffer Average (default: 60 ms)
11. 84-11-11 G7.29 Jitter Buffer Max (default: 120 ms)
12. 84-11-17 Jitter Buffer Mode – support Static or Immediate (default: 2: Immediate)
13. 84-11-18 Voice Activity Detection Threshold [default: 20 = 0dB (-30dBm)]
14. 84-11-20 Echo Canceller Mode (default: 1: Enable)
15. 84-11-22 Echo Canceller NLP Mode (default: 1: Enable)
16. 84-11-26 TX Gain (default: 14)
17. 84-11-27 RX Gain (default: 14)

84-11-28 : *D^{term}* IP CODEC Information Settings – Audio Priority

For each Type, set the codec to use (default = G.711). By default, all IP Phones belong in **Type 1**.

To change the Type for an IP Phone:

15-05-15 : IP Phone Terminal Basic Data Setup – CODEC Type

For each IP Phone (extension number), select the Type to use (default:1).

System Tones and Ring Tones:

System IP Phones do not use Program 80-01 : Service Tone Setup entries. The tones are generated locally by the System IP Phone. When a Door Box chime rings a System IP Phone, the system activates the chimes using a ring command. Because of this, if the volume is adjusted while the door chime is sounding, the ringing volume of the System IP Phone is adjusted.

Music on Hold:

Music on Hold is also provided by a Multicast from the Media gateway Card (MG16). The setting in Program 10-04-01 Music on Hold Setup, is accepted whether or not music should be provided.

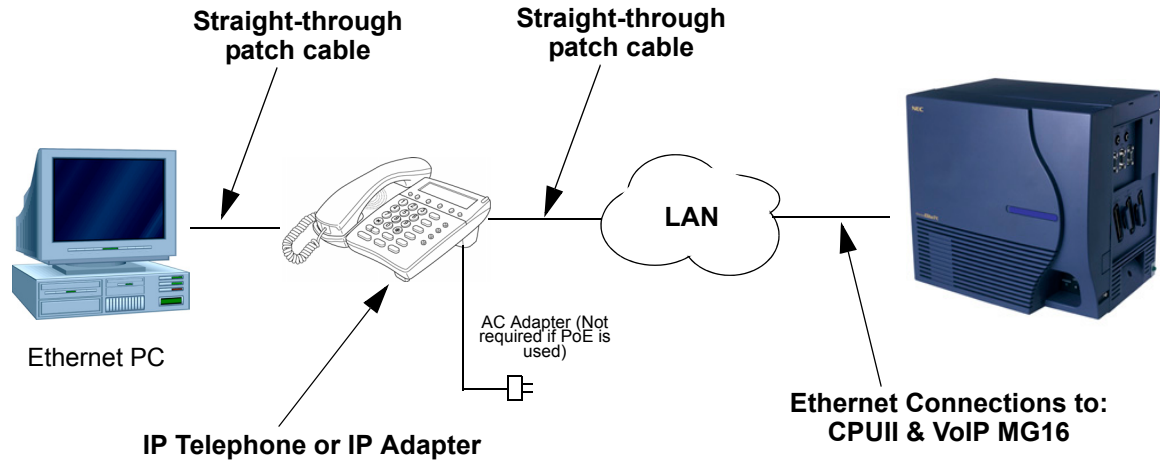



Figure 2-14 IP Phone Programming

Configuring IP Phones:

From a powered IP Phone, enter the telephone program mode by pressing **Hold + Transfer + * + #** to enter the IP User Menu program mode to select the settings for the individual telephone.

The following menu items require programming in your IP Phone:

Program/ Item No.	Description/Selection	Default Assigned Data	Comments
1	Network Setting	DHCP Disable	
2	IP Address	0.0.0.0	Enter a Static IP Address for the IP Phone.
3	Subnet Mask	0.0.0.0	Enter the subnet Mask Address.
4	Default Gateway	0.0.0.0	Enter your Default Gateway address.
5	Speed & Duplex	Auto	
6	MGC Address	0.0.0.0	Enter the Media Gateway Controller IP Address.  <i>When Multiple MG16 ETUs are entered only the MG16 configured as MASTER in Program 84-05-03 is set as MGC IP Address.</i>
7	EXT	0	Assign the IP Phones extension. This information must match Program 11-02-01 Extension Numbering. Extension must be manually defined. Auto setting is not supported.
8	IP Address	0	Overriding an IP telephone must be done through phone programming.


Operation

Registration Override for MEGACO Phones

The registration override feature gives users access to their IP telephone from any location by using the override login function. Users can log into their IP Station in the office or remotely at the home office. Use registration override and enter the override access code.



The above example displays the user using the Override function in the Central Office and remote Home Office. Two IP Phones are required; each IP Phone registers as 3999. When the IP Phone is registered as extension 3999, the other phone is in Registration Override Mode.

 *The IP Phones share the IP extension; one assumes the extension number and the other remains in the override mode.*

Key System Programming:


None required

IP Terminal Programming

1. Access IP Terminal Programming → **Hold + Transfer + * + #**.

2. Select **Option 3 – Extension Numbering**.

3. Enter the extension number you wish to override, press the **Set** and **Save** softkeys.

 *The extension number entered should be an IP Phone that is registered in PRG 15-05 IP Phone Options and PRG 11-02 Extension Numbering.*

4. Observe the IP terminal display prompting the Override Access Code.

5. The Override login code is the **Extension Number**.

6. The IP Phone registers online.

To Configure an IP Terminal for Center Download

1. Access IP Terminal Programming → **Hold + Transfer + * + #**.
2. Select **Option 9 – Advance Setting** and select sub-menu **Option 3 – DRS Setting**.

 *You are required to enable DRS Mode to support this feature.*


3. Select **Option 1 – DRS Mode**.
4. Press **2 – DRS Enable**.

Caution

The Dterm Registration Server (DRS) settings is supported by the Elite IPK II (Center Download) feature only. This setting should be enabled only if IP Phones and PVA(X)-U10 exist in the same IP Subnet. This setting does not support IP Phones residing on a different IP Subnet and must be disabled.

Reference the Elite IPK II Feature and Specifications Manual for Center Download feature.

5. Select the **Set** softkey.
6. Select **Option 2 – Primary DRS Address** (enter the IP Address of the MG16 Card) the extension you want to override.
7. Press the **Set** and **Save** softkeys. DRS Listening Port = 161.

 *The extension number entered should be a registered IP Phone .*

IP Station (SIP) – MG16

Enhancements

This feature requires **Version 1500 or higher**.

Version 1600 and Version 2100 or higher supports:

- Peer-to-Peer.
- Message Waiting Indication via RFC3842.
- Call Forwarding for SIP stations.

Version 2000 or higher:

- Supports the Combined functionality of a MG16 and K-CCIS over IP and SIP Stations/trunks on one card.

Description

SIP (Session Initiation Protocol) is used for Voice over Internet Protocol. It is defined by the IETF (Internet Engineering Task Force) RFC3261. Other RFC designations, such as RFC 3842, refer to a later implementation of SIP and may be supported by the IPK II depending on software revision of the CPUII. Commonly called SIP Station, this feature is used for IP Stations using (SIP) Session Initiation Protocol.

SIP analyzes requests from clients and retrieves responses from servers, then sets call parameters at either end of the communication, handles call transfer, and terminates. Typically, such features, including but not limited to Voice over IP services, are available from a SIP service provider.

Each Media Gateway 16 application can support up to 16 TDM Talk paths. This total may be shared among SIP Station or SIP Trunks. Registered SIP Stations and/or SIP Trunks require a one-to-one relation with the MG16 DSP Resource. This is a required component of SIP implementation in IPK II.

The Electra Elite IPK II CPUII contains a regular TCP/RTP/IP stack that can handle real-time media, supports industry standard SIP (RFC 3261) communication on the WAN side, and interfaces with the MG16 Application.

For this feature, the Media Gateway 16 is installed and assigned as a VoIP MG16 ETU. Each Media Gateway 16 supports IP signaling for up to 16 (SIP Trunks and/or SIP Stations) and reduces the maximum capacity of system stations and/or Trunks in accordance with the number of registered SIP Stations.

SIP IP Stations use the PVA(X)-U10 ETU Media Gateway. The media gateway card controls and interprets RTP messaging from the SIP IP Phone to the Elite IPK II CPUII, therefore KSU Programming (PRG 84-05-03) Master/Slave determination is **NOT** required.

A maximum of 16 PVA(X)-U() ETUs can be installed supporting the maximum of 256 IP stations (refer to [Table 2-43 Maximum System Capacities for Application Interface ETUs](#) in the [Universal Slots](#) feature). This may be a combination of SIP and or MEGACO Stations.

The MG supports only those codecs that are considered to provide toll-quality equivalent speech path. The following voice compression methods are supported for the IP Station SIP feature:

- G.711 μ -Law – Highest Bandwidth
- G.729 – Mid-Range Bandwidth

The minimum bandwidth requirements for each voice call is listed in the following table. This includes all the overhead of VoIP communication, including signaling).

Codec	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G711 μ -Law	90 Kbps	90 Kbps	20 ms*	1.5 ms	2 datagrams (40 ms)	4.4
G729	34 Kbps	34 Kbps	20 ms*	15.0 ms	2 datagrams (40 ms)	4.07

* When an IP Soft Phone is connected, set Time Between Packets to 100 ms.

- The MG16 contains a regular TCP/RTP/IP stack that can handle real time media. The MG16 ETU is an end-point on the IP network from the network administration perspective.
- The Elite IPK II CPU uses SIP Protocol to provide telephony services between remote stations through the IP Network. This is an IETF/ITU standards-based protocol.
- Speech-connection audio quality depends greatly on the available bandwidth between the stations in the data network. Because Internet is an uncontrolled data network compared to an Intranet, using this application in Intranet WAN environment with known (or controlled and assured) Quality of Service (QoS) is highly recommended.
- A static IP Address is required for each MG16 in the network.
- An on-board RJ-45 connector provides a WAN/LAN connection. Voice and signaling data to/from the IP Stations are converted into IP Frames and transmitted through the Data Communication IP Network.
- Media Gateway 16 supports a maximum of 16 Voice over IP connections.
- Duplex mode (Auto, Full, or Half) and speed (10 or 100 mbps) are configured via Elite IPK II KSU Programming.

In voice communication, particularly Internet telephony, the **Mean Opinion Score (MOS)** provides a numerical measure of the quality of human speech at the circuit destination. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. The maximum MOS is 5.0.

- IPK II SIP supports a 100 rel option and Session Timer option.
- IPK II SIP supports Early Media.
- IPK II SIP restricts an outgoing call under the following conditions:
 - SIP configuration failed
 - SIP registration failed
 - CPU11 and MG16 Link down
 - Lack of MG16 DSP resource
 - Lack of a MG16 bandwidth
- Both port numbers can be configured by system data.
- ToS Support.

Conditions

- SIP IP Station to MEGACO IP Station does not support Peer-to-Peer function and results in using a DSP resource. **(Version 1600 and 2100 or higher required)**.
- SIP Stations which support RFC 3842 (Message Waiting) receive Message Waiting Lamp indications. **(Version 1600 and 2100 or higher required)**.
- SIP Station – MG16 does not support NAT traversal.
- SIP Station – MG16 does not support a Blind Transfer feature.
- SIP station does not support Video Stream.
- SIP Station and SIP Trunk required license key for activation. Once activated, the SIP Client license may be shared between SIP Station and SIP Trunks.
- The maximum number of the SIP clients is 256.
- SIP protocol (RFC3261) is used.
- SIP Station uses the MG16 as a media gateway. Any MG16 can be used as a MG. Master/Slave setting is not required.
- Default UDP listen port for a SIP station is 5070. Both port numbers can be configured by system data.
- IPK II SIP Station registration policy supports an authentication feature. Enabling this policy prevents the registered telephone from unexpected override.

- IPK II SIP supports HOLD and TRF feature on the basis of IETF draft.
 - ❑ draft-ietf-sipping-service-examples-09.txt
 - ❑ draft-ietf-sipping-cc-transfer-05.txt
 - ❑ draft-ietf-sip-session-timer-10.txt
- **When all VoIP DSP resources are busy, the SIP phone cannot pre-empt active calls to make a 911 call.**
- The Elite IPK II CPU is the registration server for the SIP stations. The configurable IP Address is located in PRG 10-12-01 (IPK II Network Setup – IP Address).
- **Version 1600** does not support Program 10-03-01: MG/CCIS PKG – Logical Port Number of CCIS over IP.

PVA Combo Card Service Conditions

- **Version 2000 or higher** is required to support the PVA Combo Card Application.
- The PVA Combo Card Application can support 24 DSP resources which may be configured to support the MG16 or the CCISoIP application.
- The PVA Combo Card CCISoIP Trunks and SIP Trunks Logical Port Numbers are determined by the following:
 - ❑ **SIP Trunks**
 - PVA(X)-U() ETU License Registration
 - SIP Client License Registration
 - Elite IPK II Program 10-41-01 SIP Trunk Availability
 - Elite IPK II Program 10-40-02 Number of SIP Ports
 - ❑ **CCISoIP Trunks**
 - PVA(X)-U() ETU License Registration
 - Elite IPK II Program 10-40-03 CCISoIP Trunk Availability
 - Elite IPK II Program 10-40-04 Number of CCISoIP Ports
 - The maximum number of Logical Trunk Ports is 48 ports per PVA Combo Card package
- The PVA(X)-U() ETU with PVA Combo Card Application may configure DSP usage criteria with Elite IPK II Program 10-19-01 DSP Resources. Each VoIP ETU can flag individual DSP resources as:
 - ❑ IP Ext – IP Extensions (includes MEGACO Station or SIP Stations)
 - ❑ SIP Trk – SIP Trunks
 - ❑ CCIS – CCIS Networking
 - ❑ Common – Common usage for CCIS Networking, SIP Station, MEGACO Stations, SIP Trunks

- PVA Combo Card CCISoIP Application supports trunks configured in increments of four contiguous DSP resources.
- ETU Configuration – The PVA Combo Card Application package is identified as an MG_CCIS ETU.
- The PVA Combo Card – VoIP ETU supports up to 200 CCIS Logical Trunks and 200 SIP Logical Trunk Port Numbers (read only).
- IP Application DSP Resource configuration:
 - ❑ CCIS Trunks are defined as CCIS or Common
 - ❑ IP MEGACO Stations are defined as IP Ext or Common
 - ❑ SIP Trunks are defined as SIP Trk or Common
 - ❑ SIP Station are defined as IP Ext or Common
- When the PVA Combo Card is installed in the Electra Elite IPK II system, the system assigns the first available Logical Port Numbers for CCISoIP trunks and SIP trunks are allotted the next available logical port numbers.
- If the Elite IPK II CPU system does not have registered SIP licenses, the Logical SIP Trunk is not assigned.
- If the PVA Combo Card cannot establish the common highway, the maximum number of trunk ports is 16 ports for one package.
- Changes in Program 10-40-03 and or Program 10-40-04 require the PVA(X)-U10 ETU to be reset for the new changes to take effect.
- Changes in Program 10-19-01 require the PVA(X)-U10 ETU to be reset for the changes to take effect.
- PVA(X)-U10 ETU/Combo Card LED Indications:
 - ❑ Non Peer-to-Peer MEGACO IP Station and SIP IP Station calls are indicated by a Flashing RED LED
 - ❑ CCISoIP Trunk and SIP Trunk calls are indicated by a Solid RED LED

Default Setting

None

System Availability

Terminals

SIP Terminals Compliant with RFC 3261, RFC 3262, RFC 3264 (Session Description Protocol), RFC 1889 (Real Time Protocol).

PVA(X)-U20 ETU with MG16 Application Package **1.51 or higher**.

IPK II CPU Release **1.50 or higher**.

PVA(X)-U() ETU with PVA Combo Application Package **1.00 or higher** and Elite IPK II CPU II **Version 2000 or higher**.

Related Features

None

Programming

VoIP Settings:

When the PVA(X)-U() ETU is first purchased it has a factory default package installed. You must load the MG16 Application for the Elite IPK II KSU to recognize the VoIP MG16 ETU. [Reference the PVA(X)-U() ETUs Firmware upgrade installation procedure].

- ➔ **10-19-01 VoIP DSP Resource Selection (Version 1600 or higher required)**
Select type of VoIP ETU DSP Resource. This program setting has no affect on the terminal/trunk port assignment or usage.
Entries: 0 = Common (Used for both IP Extensions and Trunks), 1 = Extension, 2 = Trunks, 3 = Networking
(Default: 0)
- ➔ **10-26-01 IP System Operation Setup – Peer to Per Mode (Version 1600 or higher required)**
0 = Off
1 = On
(default: 1)
- ➔ **10-26-03 IP System Operation Setup – SIP Peer to Per Mode (Version 1600 or higher required)**
0 = Off
1 = On
(default: 1)

-
-
- ➔ **84-05-01 VoIP IP Address – IP Address**
Define the IP Address of the VoIP Application supported (MG16, CCISoIP). (default: Slot 1: 172.16.0.20)
 - ➔ **84-05-02 VoIP IP Address – LAN Interface**
Define the LAN interface Speed and Mode of the VoIP Application supported (MG16, CCISoIP). (default: 0 = AUTO)
 - ➔ **84-05-04 VoIP IP Address – Subnet Mask**
Define the Media Gateway Subnet Mask Address. (default: 255.255.0.0)
 - ➔ **84-05-05 VoIP IP Address – Default Gateway**
Define the MG16 Application Default Gateway Address. (default: 0.0.0.0)
 - ➔ **84-06-01 VoIP Info – RTP Port Number**
Define the Media Gateway starting RTP Port Number. (default: 10020)
 - ➔ **84-06-02 VoIP Info – RTCP Port Number**
Define the Media Gateway Starting RTCP Port Number . The RTCP Port Number is the RTP port number + 1 (default: 10021)
 - ➔ **84-06-04 VoIP Info – Fract Lost Threshold**
Define the fractional lost threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value. (default: 0)
 - ➔ **84-06-05 VoIP Info – Packets Lost Threshold**
Define the packet lost threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value. (default: 0)
 - ➔ **84-06-07 VoIP Info – Jitter Threshold**
Define the Jitter Threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value. (default: 0)
 - ➔ **84-06-09 VoIP Info – Delay LSR Threshold**
Define the Delay threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value. (default: 0)

VoIP ToS Setup

The Elite IPK II supports Quality of Service (QoS) Marking for the Session Initiation Protocol (SIP).

- ➔ **84-10-01 ToS Setup – ToS Mode**
Define your SIP QoS marking for Type of Service(ToS) or Diffserve.
[1 = CPU, 2 = MGC, 3 = MEGACO, 5 = RTP/RTCP, 6 = SIP, 7 = CCISoIP]
(default: 0 = Invalid)

IP Extension Numbering:

- ➔ **11-02-01 Extension Numbering**
Define the IP Phone extension number.
This designated extension is used to register your IP Phone and is programmed IP Phone programming. Refer to the Installation Instructions for Elite ITH-4D/8D/16D-2/3 D^{term} IPK Terminals and the IP-R (IPK) Adapters.

SIP Extension Codec Information:

- ➔ **84-19-01 SIP Extension IP CODEC Information Basic Setup – G.711 Audio Frame Number**
Define the G.711 audio Frame Size (1 = 10 ms, 2 = 20 ms, 3 = 30 ms).
(default: 3)
- ➔ **84-19-02 SIP Extension IP CODEC Information Basic Setup – G.711 Voice Activity Detection Mode**
Enable/Disable Voice Activity Detection for G.711.
(default: 0 = Disable)
- ➔ **84-19-03 SIP Extension IP CODEC Information Basic Setup – G.711 Type**
Define the G.711 Type – μ -law is recommended when in USA.
(default: 1 = μ -Law)
- ➔ **84-19-04 SIP Extension IP CODEC Information Basic Setup – G.711 Jitter Buffer (min)**
Define G.711 Jitter Buffer minimum accepted value.
(default: 30 ms)
- ➔ **84-19-05 SIP Extension IP CODEC Information Basic Setup – G.711 Jitter Buffer (average)**
Define G.711 Jitter Buffer average accepted value.
(default: 60 ms)
- ➔ **84-19-06 SIP Extension IP CODEC Information Basic Setup – G.711 Jitter Buffer (max)**
Define G.711 Jitter Buffer maximum accepted value.
(default: 120 ms)
- ➔ **84-19-07 SIP Extension IP CODEC Information Basic Setup – G.729 Audio Frame Number**
Define the G.729 audio Frame Size [1 = 10 ms, 2 = 20 ms, 3 = 30 ms].
(default: 3)
- ➔ **84-19-08 SIP Extension IP CODEC Information Basic Setup – G.729 Voice Activity Detection Mode**
Enable (1)/Disable (0) Voice Activity Detection for G.729.
(default: 0)

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- ➔ **84-19-09 SIP Extension IP CODEC Information Basic Setup – G.729 Jitter Buffer (min)**
Define G.711 Jitter Buffer minimum accepted value.
(default: 30 ms)
 - ➔ **84-19-10 SIP Extension IP CODEC Information Basic Setup – G729 Jitter Buffer (average)**
Define G.729 Jitter Buffer setting.
(default: 60 ms)
 - ➔ **84-19-11 SIP Extension IP CODEC Information Basic Setup – G729 Jitter Buffer (max)**
Define G.729 Jitter Buffer maximum accepted value.
(default: 120 ms)
 - ➔ **84-19-17 SIP Extension IP CODEC Information Basic Setup – Jitter Buffer Mode**
Define the Jitter Buffer mode – support Static (1), Adaptive during silence (2), or Adaptive Immediate (3).
(default: 3)
 - ➔ **84-19-18 SIP Extension IP CODEC Information Basic Setup – VAD Threshold**
Define the VAD Threshold. Consult the Electra Elite IPK II Programming Manual for Threshold scale to set acceptable values.
[default: 20 = 0dB (-30dBm)]
 - ➔ **84-19-26 SIP Extension IP CODEC Information Basic Setup – TX Gain**
Define to TX Gain Values – Adjusting this value increases or decreases volume levels for the receiving party. Consult the Electra Elite IPK II Programming Manual for Transmit Gain scale to set acceptable value.
(default: 10)
 - ➔ **84-19-27 SIP Extension IP CODEC Information Basic Setup – RX Gain**
Define to RX Gain Values – Adjusting this value increases or decreases volume levels for the sending party. Consult the Electra Elite IPK II Programming Manual for Transmit Gain scale to set acceptable value.
(default: 10)
 - ➔ **84-19-28 SIP Extension IP CODEC Information Basic Setup – Audio Capability Priority**
Define Audio Capability Priority. Consult the Electra Elite IPK II Programming Manual for Transmit Gain scale to set acceptable value.
(default: G.711)
 - ➔ **84-19-31 SIP Extension IP CODEC Information Basic Setup – DTMF Payload Number**
Define the DTMF Payload Number.
(default: 110)

SIP Extension Basic Information Setup:

- ➔ **84-20-01 SIP Extension Information Basic Setup – Registrar/Proxy Port**
Define SIP station Proxy Port (0~65535).
(default: 5070)

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- ➔ **84-20-02 SIP Extension Information Basic Setup – Session Timer Value**
Define the periodic refresh time (0~65535 seconds) that allows both user agents and proxies to determine if the SIP session is still active.
(default: 180)
 - ➔ **84-20-03 SIP Extension Information Basic Setup – Minimum Session Timer Value**
Define the minimum allowed time (0~65535 seconds) for the SIP session timer.
(default: 180)
 - ➔ **84-20-04 SIP Extension Information Basic Setup – Called Party Info**
Define the SIP Extension presented Caller ID information as Request URI (0) or To header (1).
(default: 0)
 - ➔ **84-20-05 SIP Extension Information Basic Setup – Expire Value of Invite**
Define the time out response value (0~256 seconds) for SIP invite.
(default: 60)

IP Phone Configuration

IP Telephone Terminal Basic Data Setup:

- ➔ **15-01-01 Basic Extension Data Setup – Extension Name**
Review the type protocol support by the IP Phone.
(default: MEGACO)
- ➔ **15-05-02 IP Telephone Terminal Basic Data Setup – IP Phone Fixed Port Assignment**
Review the registered IP Phone Mac Address.
(default: 00-00-00-00-00-00)
- ➔ **15-05-06 IP Telephone Terminal Basic Data Setup – IP Phone Terminal Type**
Define the type of IP Phone being used – valid options are IP70 (1), IP80 (2) or IP Smart Phone (3).
(default: 0)
- ➔ **15-05-07 IP Telephone Terminal Basic Data Setup – Using IP Address**
Review the registered IP Phones IP Address [Informational Only].
(default: 0.0.0.0)
- ➔ **15-05-15 IP Telephone Terminal Basic Data Setup – CODEC Type**
Set the registered IP Phone Codec type – Reference PRG 84-11 Dterm IP Codec Basic Information.
(default: Type:1)
- ➔ **15-05-16 IP Telephone Terminal Basic Data Setup – Authentication Password**
Set the registered Basic Information.
(default: Not Assigned)
- ➔ **15-05-17 IP Telephone Terminal Basic Data Setup – Calling Party Display Info**
Set the Calling Party Display as Nickname (0), Display Name (1), User Part (2), or Extension (3).
(default: 0)

- ➔ **15-05-18 IP Telephone Terminal Basic Data Setup – IP Duplication Allowed Group**
 Set the registered IP Phone Codec type – Reference PRG 84-11 Dterm IP Codec Basic Information.
 (default: 0:Not Used)

SIP Phone Example

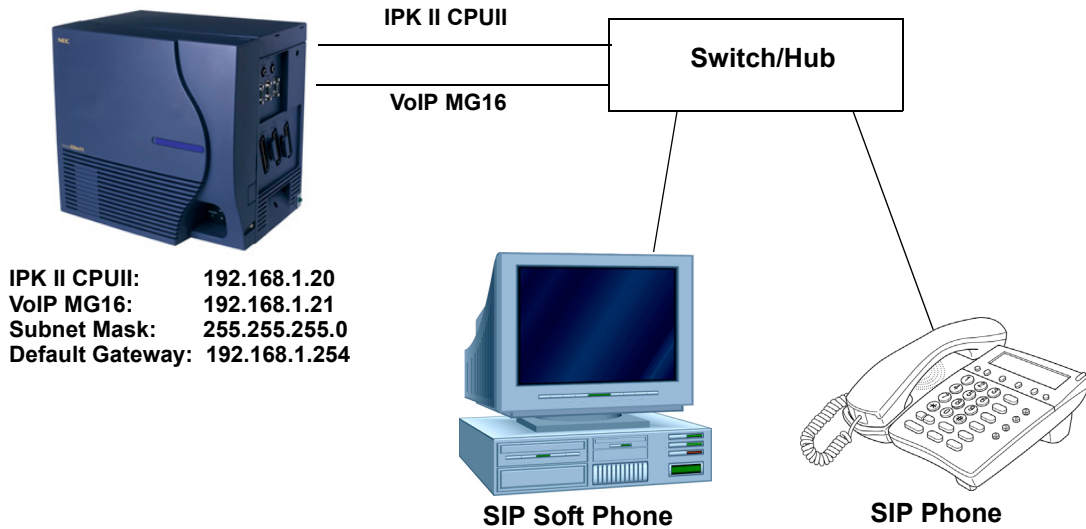


Figure 2-15 Example – SIP Phone

The following menu items require programming in your SIP IP Phone (consult SIP Phone vendor specific documentation):

Program/Item No.	Description/Selection	Default Assigned Data	Comments
1	IP Address	0.0.0.0	Enter a Static IP Address for the SIP Phone.
2	Subnet Mask	0.0.0.0	Enter the Subnet Mask Address.
3	Default Gateway	0.0.0.0	Enter the Default Gateway address.
4	MGC Address	0.0.0.0	Enter the Media Gateway Controller IP Address. ✎ <i>This information can be located in PRG 10-12-01 IPK II Network Setup IP Address.</i>

Program/ Item No.	Description/Selection	Default Assigned Data	Comments
5	Extension Number	0	Assign the SIP Phone extension. This information must match Program 11-02-01 Extension Numbering.

IP Trunk – H.323 Protocol

Description

The Electra Elite IPK II Voice over IP Trunk Card H.323 package sends the real-time voice over the corporate LAN or WAN. The voice from the telephone is digitized and then put into frames to be sent over a network using Internet Protocol.

The Electra Elite IPK II Voice over IP Trunk Card H.323 package allows communication using standard H.323 (Normal and Fast Start) Protocol and allows connectivity to any H.323 standard compliant voice gateway and gatekeeper. This VoIP Trunk Card also provides Registration and Authentication Server (RAS) support with the ability to register with an RAS Server and use Gatekeeper for dynamic call routing.

The IAD(8)-U10 ETU – H.323 is an optional interface that can provide IP trunks and Tie Lines. It can operate in the following modes:

- COI
- COID
- DID
- TLI
- DTI

Depending on the requirements and resource allocation in the LAN/WAN/Internet, the IAD(8)-U10 ETU - H.323 can be configured to use any of the following voice compressions:

- G.711 μ -Law – Highest Bandwidth
- G.723 – Lowest Bandwidth
- G.729 (a) – Most often used

The IAD(8)-U10 ETU – H.323 can be assigned in any of the following configurations:

- A 2-port TLI(2)-U10 ETU
- A 4-port DID(4)/COI(4)/COID(4)-U10 ETU
- An 8-port COI(8)/COID(8)-U10 ETU
- A DTI ETU using eight channels that can be installed in interface slots supporting these ETUs.
- The LAN/WAN or Internet connection is provided by a 10 Base-T/100 Base-TX Ethernet.
- The ETU operating mode can be configured per ETU, but not per port.

System Availability

Terminals

All Multiline Terminals

Required Component(s)

IAD(8)-U10 ETU – H.323 package

Related Features

None

Programming

- ➔ 10-03-01 **ETU Setup**
- ➔ 11-01-01 **System Numbering**
- ➔ 14-02-01 **Analog Trunk Data Setup – Signaling Type (DP/DTMF)**
At default, PRG 14-02-01 is set to 2 (DTMF). For the IAD trunks this must be set for Dial Pulse [either 0 (10PPS) or 1 (20PPS)].
- ➔ 14-05-01 **Trunk Group – Trunk Group Number**
- ➔ 44-02-01
through
44-02-03 **Dial Analysis Table for ARS/F-Route Access**
- ➔ 44-05-01 **ARS/F-Route Table – Trunk Group Number**
IAD(8)-U() assigned as DTI is allocated 24 trunk ports, but can use only the first eight ports. The remaining trunk ports should be assigned to Trunk Group 0.
- ➔ 44-05-09 **ARS/F-Route Table – Maximum Digit**

Operation

None

IP Trunk – (SIP) Session Initiation Protocol

Description

The Electra Elite IPK II Voice over IP Trunk Card SIP package sends real-time voice over the corporate LAN or WAN. The voice from the telephone is digitized and then put into frames to be sent over a network using Internet protocol.

Using VoIP equipment at a gateway (a network point that acts as an entrance to another network), the packetized voice transmissions from users in the company are received and routed to other parts of the company intranet (local area or wide area network) or they can be sent over the Internet using CO lines to another gateway.

The IAD(8)-U10 ETU – SIP is an optional interface that can provide IP trunks and Tie Lines. It can operate in the following modes:

- COI
- COID
- DID
- TLI
- DTI

Depending on the requirements and resource allocation in the LAN/WAN/Internet, the IAD(8)-U10 ETU - SIP can be configured to use any of the following voice compressions:

- G.711 μ -Law – Highest Bandwidth
- G.729 (a) – Most often used

The IAD(8)-U10 ETU – SIP can be assigned in any of the following configurations:

- A 2-port TLI(2)-U10 ETU
- A 4-port DID(4)/COI(4)/COID(4)-U10 ETU
- An 8-port COI(8)/COID(8)-U10 ETU
- A DTI ETU using eight channels that can be installed in interface slots supporting these ETUs.
- The LAN/WAN or Internet connection is provided by a 10 Base-T/100 Base-TX Ethernet.
- The ETU operating mode can be configured per ETU, but not per port.

Conditions

None

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

IAD(8)-U10 ETU – SIP package

Related Features

None

Programming

- ➔ 10-03-01 ETU Setup
- ➔ 11-01-01 System Numbering
- ➔ 14-02-01 **Analog Trunk Data setup – Signaling Type (DP/DTMF)**
At default, PRG 14-02-01 is set to 2 (DTMF). For the IAD trunks this must be set for Dial Pulse [either 0 (10PPS) or 1 (20PPS)].
- ➔ 14-05-01 Trunk Group – Trunk Group Number
- ➔ 44-02-01
through
44-02-03 Dial Analysis Table for ARS/F-Route Access

- ➔ **44-05-01 ARS/F-Route Table – Trunk Group Number**
IAD(8)-U() assigned as DTI is allocated 24 trunk ports, but can use only the first eight ports.
The remaining trunk ports should be assigned to Trunk Group 0.
- ➔ **44-05-09 ARS/F-Route Table – Maximum Digit**

Operation

None

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IP Trunk (SIP) – MG16

Enhancements

This feature added with **Version 1500**.

Version 2000 supports the Combined functionality of a MG16 and K-CCIS over IP and SIP Stations/trunks on one card.

Description

SIP (Session Initiation Protocol) is used for Voice over IP (VoIP). It is described in the IETF (Internet Engineering Task Force) RFC3261. Commonly called SIP Trunking, this feature is used for linking a key telephone system to the publicly available voice services using SIP.

SIP analyzes requests from clients and retrieves responses from servers, then sets call parameters at either end of the communication, handles call transfer, and termination.

Typically, such features, including but not limited to Voice over IP services, are available from an SIP service provider. The SIP trunks on the Electra Elite IPK II can connect to a service provider over a broadband connection for implementing Voice over IP lines to the IPK II user.

The call routing, call features, and speech handling (RTP) are the same as other trunks in the Electra Elite IPK II system – only the signaling protocol is different. With the IPK II, SIP trunks can receive incoming calls with Caller ID, place outgoing calls, and transfer SIP trunks to IP, SIP, analog and digital stations, and across a network.

The Media Gateway (MG16) is an IP Application loaded on the PVA(X)-U() ETU. This is an optional interface package for the Electra Elite IPK II system that supports SIP Trunks, SIP Stations, and MEGACO IP stations. This is a required component of SIP implementation in Elite IPK II.

An on-board RJ-45 connector provides a WAN/LAN connection. Voice and signaling data to/from the IP stations are converted into IP Frames and transmitted through the Data Communication IP Network. Media Gateway 16 supports a maximum of 16 Voice over IP connections.

Each Media Gateway 16 application can support up to 16 TDM Talk paths. This total may be shared among SIP Station or SIP Trunks. Registered SIP Stations and/or SIP Trunks require a one-to-one relation with the MG16 DSP Resource.

The Electra Elite IPK II CPUII contains a regular TCP/RTP/IP stack that handles real-time media, supports industry standard SIP (RFC 3261) communication on the WAN side, and interfaces with the MG16 Application.

For this feature, the Media Gateway 16 is installed and assigned as a VoIP MG16 ETU. Each Media Gateway supports IP signaling for up to 16 (SIP Trunks and/or SIP Stations) and reduces the maximum capacity of SIP Trunk/Stations in accordance with the number of registered SIP Stations or SIP Trunks.

A maximum of 16 PVA(X)-U() ETUs can be installed supporting the maximum of 200 SIP Trunks (refer to [Table 2-44 KSU Power-Based Calculator Chart on page 2-1044](#) in the Universal Slots feature).

SIP Trunks can operate in two modes:

- SIP Trunking – provides connectivity to a SIP service provider.
- SIP Interconnect – provides connectivity to other IPK II systems.

Depending on the requirements and resource allocation in the LAN/WAN, the SIP Trunk can be configured to use any of the following voice compressions:

- G.711 μ -Law – Highest Bandwidth
- G.729 (a) – Most often used

The minimum bandwidth requirements for each voice call is listed below. This includes all the overhead of VoIP communication, including signaling.

Codec	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.711 μ -Law	90 Kbps	90Kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G.729	34 Kbps	34 Kbps	20 ms	15.0 ms	2 datagrams (40 ms)	4.07

Note 1: This includes the overhead of VoIP communication, including signaling.

Note 2: In voice communications, particularly Internet telephony, the mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. The maximum for the Mean Opinion Score is 5.0.

- SIP signaling stack resides on the CPU II.
- The MG16 contains a regular TCP/RTP/IP stack that can handle real time media. The MG16 ETU is an end-point on the IP network from the network administration perspective.
- Speech-connection audio quality depends entirely on the available bandwidth between the Voice over IP end points in the data network. Because Internet is an uncontrolled data network compared to an Intranet, using this application in Intranet WAN environment with known (or controlled and assured) Quality of Service (QoS) is highly recommended.
- A static IP Address is required for each MG16 in the network.

- The LAN/WAN connection is provided by a 10 Base-T/100 Base-TX Ethernet connector. This connector is a single 10/100 mbps Ethernet Connector. This port has Auto-Medium Dependent Interface Crossover (MDIX) to allow using either a straight-through Ethernet cable for connection to a PC or a crossover Ethernet cable.
- Duplex mode (Auto, Full, or Half) and speed (10/100 mbps) are configured via Elite IPK II KSU Programming.
- Default UDP listen port for SIP Trunk is 5060.
- IPK II SIP supports 100 rel option (IETF RFC 3262) and Session Timer option (IETF draft-ietf-sip-session-timer-10.txt).
- IPK II SIP supports Early Media.
- IPK II SIP restricts an outgoing call under the following conditions:
 - SIP configuration failed
 - SIP registration failed
 - CPU11 and MG16 Link down
 - Lack of MG16 DSP resource
 - Lack of MG16 bandwidth
- IPK II SIP supports MD5 Authentication flow for a SIP server. (Refer to IETF RFC 3261 for more information).
- IPK II SIP supports multi-UAs registration for one SIP server.
- IPK II SIP Trunk supports a DNS resolution access and an IP address direct access for SIP server. (Refer to IETF RFC 3261 for more information).
- IPKII SIP Trunk can connect a SIP server over NAPT router by one IP address.
- IPK II SIP can transport DTMF digits in-band according to IETF RFC 2833, or by using uncompressed codec such as G.711 (Configured in system programming).
- IPK II SIP Trunk supports the sub-address feature with SIP trunk interconnection. (IPK II SIP Trunk to IPK II SIP Trunk).

Conditions

- If Program 10-29-01 : SIP Proxy Setup – Outbound Proxy is set to 0 (Off), then all other programs under 10-29 must be manually configured to default values.
- The IPK II does not support the simultaneous use of a SIP Trunk inter-connection and a SIP Trunk carrier connection modes. If connection between IPK II systems is desired while working with the SIP providers, then both IPK II systems must individually register such a service provider.

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- Outgoing calls on SIP are not possible under the following conditions:
 - ❑ SIP configuration failed
 - ❑ SIP registration failed
 - ❑ CPU11 and MG16 link down
 - ❑ Lack of MG16 DSP resource
 - ❑ Lack of MG16 bandwidth
 - IPK II SIP does not support T.38 FAX.
 - SIP IP Trunks use the PVA(X)-U() ETU Media Gateway (MG16 application). The media gateway card controls and interprets RTP messaging from the SIP IP Trunks to the Elite IPK II Processor, therefore KSU Programming (PRG 84-05-03) Master/Slave determination is **NOT** required.
 - The MG16 Application consumes 16 logical trunk ports, only when using an SIP trunk.
 - SIP Trunk does not support multi-SIP Server registration.
 - SIP Station and SIP Trunk required license key for activation. Once activated, the SIP Client license may be shared between SIP Station and SIP Trunks.
 - The maximum number of the SIP Trunks is 200.
 - SIP protocol (RFC3261) is used.
 - When using SIP Trunks as Interconnect, and using it to grab a trunk out of the remote system, the trunk access code must be an F-Route and not a Trunk.
 - **Version 1600** does not support the options for configuring 20 or 24 ports in Program 10-40-02: VoIP Trunk Availability.
 - **Version 1600** does not support Program 10-03-01: MG-CCIS PKS – Logical Port Number of CCIS over IP.

PVA Combo Card Service Conditions

- **Version 2000 or higher** is required to support the PVA Combo Card Application.
- The PVA Combo Card Application can support 24 DSP resources which may be configured to support the MG16 or the CCISoIP application.
- The PVA Combo Card CCISoIP Trunks and SIP Trunks Logical Port Numbers are determined by the following:
 - ❑ **SIP Trunks**
 - PVA(X)-U() ETU License Registration
 - SIP Client License Registration
 - Elite IPK II Program 10-41-01 SIP Trunk Availability

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- Elite IPK II Program 10-40-02 Number of SIP Ports
 - CCISoIP Trunks**
 - PVA(X)-U() ETU License Registration
 - Elite IPK II Program 10-40-03 CCISoIP Trunk Availability
 - Elite IPK II Program 10-40-04 Number of CCISoIP Ports
 - The maximum number of Logical Trunk Ports is 48 ports per PVA Combo Card package
 - The PVA(X)-U() ETU with PVA Combo Card Application may configure DSP usage criteria with Elite IPK II Program 10-19-01 DSP Resources. Each VoIP ETU can flag individual DSP resources as:
 - IP Ext – IP Extensions (includes MEGACO Station or SIP Stations)
 - SIP Trk – SIP Trunks
 - CCIS – CCIS Networking
 - Common – Common usage for CCIS Networking, SIP Station, MEGACO Stations, SIP Trunks
 - PVA Combo Card CCISoIP Application supports trunks configured in increments of four contiguous DSP resources.
 - ETU Configuration – The PVA Combo Card Application package is identified as an MG_CCIS ETU.
 - The PVA Combo Card – VoIP ETU supports up to 200 CCIS Logical Trunks and 200 SIP Logical Trunk Port Numbers (read only).
 - IP Application DSP Resource configuration:
 - CCIS Trunks are defined as CCIS or Common
 - IP MEGACO Stations are defined as IP Ext or Common
 - SIP Trunks are defined as SIP Trk or Common
 - SIP Station are defined as IP Ext or Common
 - When the PVA Combo Card is installed in the Electra Elite IPK II system, the system assigns the first available Logical Port Numbers for CCISoIP trunks and SIP trunks are allotted the next available logical port numbers.
 - If the Elite IPK II CPU system does not have registered SIP licenses, the Logical SIP Trunk is not assigned.
 - If the PVA Combo Card cannot establish the common highway, the maximum number of trunk ports is 16 ports for one package.
 - Changes in Program 10-40-03 and or Program 10-40-04 requires the PVA(X)-U() ETU to be reset for the new changes to take effect.
 - Changes in Program 10-19-01 requires PVA(X)-U() ETU to be reset for the changes to take effect.

- PVA(X)-U() ETU/Combo Card LED Indications:
 - ❑ Non Peer-to-Peer MEGACO IP Station and SIP IP Station calls are indicated by a Flashing RED LED.
 - ❑ CCISoIP Trunk and SIP Trunk calls are indicated by a Solid RED LED.

Default Setting

None

System Availability

Terminals

SIP Trunks Compliant with RFC 3261

Required Component(s)

PVA(X)-U() ETU – MG16 **1.50 package or Higher**

IPK II CPU II **Version 1.50 or Higher**

SIP License

PVA(X)-U() ETU with PVA Combo Application package **1.00 or Higher** and Elite IPK II CPU II **Version 2000 or Higher**

Related Features

Reference the Elite IPK II VoIP (Voice over Internet Protocol) Installation Manual

Programming

VoIP Settings

When the PVA(X)-U() ETU is first purchased it has a factory default package installed. You must load the MG16 Application for the Elite IPK II KSU to recognize the VoIP MG16 ETU. [Reference the PVA(X)-U() ETU Firmware upgrade installation procedure].

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- ➔ **84-05-01 VoIP IP Address – IP Address**
Define the IP Address of the VoIP Application supported (MG16, CCISoIP).
(default: Slot 1: 172.16.0.20)
 - ➔ **84-05-02 VoIP IP Address – LAN Interface**
Define the LAN interface Speed and Mode of the VoIP Application supported (MG16, CCISoIP).
(default: 0 = AUTO)
 - ➔ **84-05-04 VoIP IP Address – Subnet Mask**
Define the Media Gateway Card Subnet Mask Address.
(default: 255.255.0.0)
 - ➔ **84-05-05 VoIP IP Address – Default Gateway**
Define the MG16 Application Default Gateway Address.
(default: 0.0.0.0)
 - ➔ **84-06-01 VoIP Info – RTP Port Number**
Define the Media Gateway Card starting RTP Port Number.
(default: 10020)
 - ➔ **84-06-02 VoIP Info – RTCP Port Number**
Define the Media Gateway Card Starting RTCP Port Number. The RTCP Port Number is the RTP port number + 1.
(default: 10021)
 - ➔ **84-06-04 VoIP Info – Fract Lost Threshold**
Define the fractional lost threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)
 - ➔ **84-06-05 VoIP Info – Packets Lost Threshold**
Define the packet lost threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)
 - ➔ **84-06-07 VoIP Info – Jitter Threshold**
Define the Jitter Threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)
 - ➔ **84-06-09 VoIP Info – Delay LSR Threshold**
Define the Delay threshold – this data is sent to the IPK II CPUII when the value exceeds the defined value.
(default: 0)

VoIP ToS Setup

The Elite IPK II supports QoS Marking for the ToS and DiffServe on the following VoIP Applications: (MG16 and CCISoIP packages).

SIP Trunk Basic Setup

- **10-40-01 SIP Trunk Availability – SIP Trunk Availability**
Enable (1)/Disable (0) SIP trunking.
(Default: 0)
- **10-28-01 SIP System Information Setup – Domain Name**
Define the Domain name. This information is generally provided by the SIP carrier. Entries: 64 characters maximum.
(Default: None)
- **10-28-02 SIP System Information Setup – Host Name**
Define the Host name. This information is generally provided by the SIP carrier. Entries: 48 characters maximum.
(Default: None)
- **10-28-03 SIP System Information Setup – Transport Protocol**
Define the Transport type. This option is always set to UDP.
Entries: 0 = UDP, 1 = TCP
(Default: 0)
- **10-28-04 SIP System Information Setup – User ID**
Define the User ID. Entries: 32 characters maximum.
(Default = No Entry)
- **10-28-05 SIP System Information Setup – Domain Assignment**
Define the Domain Assignment. This entry is determined by what information the SIP carrier provides. If the SIP carrier provides a server name: **SIPconnect-sca@L0.cbeyond.net**, then the domain would be: **L0.cbeyond.net** and the host name would be **SIPconnect-sca**.
Entries: 0 = IP Address, 1 = Domain name
(Default: 0)
- **10-29-14 SIP Server Information Setup – SIP Carrier Choice**
Define the SIP Carrier Choice. This should currently stay at an entry of 0 (Default).
Entries: 0 = Default, 1 = Carrier A, 2 = Carrier B, 3 = Carrier C, 4 = Carrier D, 5 = Carrier E, 6 = Carrier F, 7 = Carrier G.
(Default: 0)
- **14-12-01 SIP Register ID Setup for IP Trunk – Register ID**
Set the IP trunk ID in a networked system. This data is referred to for incoming and outgoing IP trunks. Incoming calls are received on the trunk port of the same ID as the trunk ID of the outgoing call in the networked system (Entries: 0~65535, Default: 0 = ID not notify).
For example, trunk 5 in Site A is programmed with ID 2. Trunk 7 in Site B is programmed with ID 2. An outgoing call from Site A on trunk 5 is received on trunk 7 in Site B since the ID number is a match. Entries: 0 = not notified, 1~65535
(Default: 0)
- **22-02-01 Incoming Call Trunk Setup**
Define the SIP trunks as type 3 (DID). In addition to the SIP trunk programming, refer to the Direct Inward Dialing (DID) feature in the Electra Elite IPK II Features and Specifications Manual and the Electra Elite IPK II Programming Manual for detailed information.

VoIP DSP Resource

- **10-19-01 VoIP DSP Resource Selection (Version 1600 or higher required)**
Select type of VoIP ETU DSP Resource. This program setting has no affect on the terminal/trunk port assignment or usage.
Entries: 0 = Common (Used for both IP Extensions and Trunks), 1 = Extension, 2 = Trunks, 3 = Networking.
(Default: 0)
- **10-40-02 VoIP Trunk Availability – Number of Ports (Version 1600 or higher required)**
0 = 4 ports
1 = 8 ports
2 = 12 ports
3 = 16 ports
(Default: 0)

SIP Authentication Information

- **10-30-02 SIP Authentication Information – User Name**
Define the authentication User name provided by the SIP carrier.
Entries: 48 characters maximum
(Default: None)
- **10-30-03 SIP Authentication Information – Password**
Enter the authentication password provided by the SIP carrier. When the Electra Elite IPK II registers its own ID with the carrier SIP server or makes an outgoing call via the carrier SIP server, the SIP server requests the authentication. This data is used as Register ID 0.
Entries: 24 characters maximum
(Default: None)
- **10-30-04 SIP Authentication Information – Authorization Trial**
Define the Authorization Trial. When a call tries to register with the SIP carrier and they refuse, this entry determines how many times the Electra Elite IPK II sends authorization.
Entries: 1~9
(Default:1 time)

SIP Caller ID

- **14-01-24 Basic Trunk Data Setup – Trunk-to-Trunk Outgoing Caller ID Through Mode**
Enable (1)/Disable (0) the Trunk-to-Trunk Outgoing Caller ID Through Mode. This option allows Caller ID from the original outside caller to be displayed when a trunk is forwarded off premise. This option can be used only with PRI and SIP trunks.
(Default: 0)

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- **21-17-01 IP Trunk (SIP) Calling Party Number Setup for Trunk**
Assign the Caller Party Number for each IP trunk. The assigned number is sent to the central office when the caller places an outgoing call. If the Calling Party Number is assigned by both PRG 21-17 and PRG 21-18/PRG 21-19, then the system uses the entry in PRG 21-18/PRG 21-19.
Entries: 1~0, *, # (to equal up to 16 digits)
(Default: None)
 - **21-19-01 IP Trunk (SIP) Calling Party Number Setup for Extension**
Assign the Calling Party Number for each extension. The assigned number is sent to the central office when the caller places an outgoing call. If the Calling Party Number is assigned by both Program 21-17 and Program 21-18/Program 21-19, the system uses the data in Program 21-18/Program 21-19.
Entries: 1~0, *, # (to equal up to 16 digits)
(Default: None)

SIP CODEC Trunk

- **84-13-01 SIP Trunk CODEC Information Basic Setup – G.711 Audio Frame Number**
Set the G.711 Audio Frame Number.
Entries: 2~3
(Default: 2)
- **84-13-02 SIP Trunk CODEC Information Basic Setup – G.711 Voice Activity Detection Mode**
Enable (1)/Disable (0) the G.711 VAD Detection Mode.
(Default: 0)
- **84-13-03 SIP Trunk CODEC Information Basic Setup – G.711 Type**
Define the G.711 type.
Entries: 0 = A-law, 1 = μ -law
(Default: 1)
- **84-13-04 SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (min)**
Set the minimum (0~200 ms) G.711 Jitter Buffer.
(Default: 20)
- **84-13-05 SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (Average)**
Set the average (0~200 ms) G.711 Jitter Buffer type.
(Default: 40)
- **84-13-06 SIP Trunk CODEC Information Basic Setup – G.711 Jitter Buffer (max)**
Set the maximum (0~200 ms) G.711 Jitter Buffer.
(Default: 60)

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- ➔ **84-13-07 SIP Trunk CODEC Information Basic Setup – G.729 Audio Frame Number**
Set the G.729 Audio Frame Number. Entries: 2~6
(Default: 2)
2 = 20 ms
3 = 30 ms
4 = 40 ms
5 = 50 ms
6 = 60 ms

 - ➔ **84-13-08 SIP Trunk CODEC Information Basic Setup – G.729 Voice Activity Detection Mode**
Enable (1)/Disable (0) the G.729 VAD Detection Mode.
(Default: 0)

 - ➔ **84-13-09 SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (min)**
Set the minimum (0~200 ms) G.729 Jitter Buffer.
(Default: 20)

 - ➔ **84-13-10 SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (Average)**
Set the average (0~200 ms) G.729 Jitter Buffer.
(Default: 40)

 - ➔ **84-13-11 SIP Trunk CODEC Information Basic Setup – G.729 Jitter Buffer (max)**
Set the maximum (0~200 ms) G.729 Jitter Buffer.
(Default: 60)

 - ➔ **84-13-17 SIP Trunk CODEC Information Basic Setup – Jitter Buffer Mode**
Set the Jitter Buffer Mode.
Entries: 1= Static, 2 = Adaptive during silence, 3 = Adaptive Immediately
(Default: 3)

 - ➔ **84-13-18 SIP Trunk CODEC Information Basic Setup – VAD Threshold**
Set the VAD (Voice Activity Detection) threshold.
Entries: 0~30 (-19dB~+10dB)
(Default: 20)
1 = -19dB (-49dBm)
20 = 0dB (-30dBm)
29 = 9dBm (-21dBm)
30 = 10dBm (-20dBm)

 - ➔ **84-13-26 SIP Trunk CODEC Information Basic Setup – TX Gain**
Set the transmit gain.
Entries: 0~28 (-14dBm ~ +14dBm)
(Default: 10)
0 = -14 dBm
1 = -13 dBm
14 = 0 dBm
27 = 13 dBm
28 = 14 dBm

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- **84-13-27 SIP Trunk CODEC Information Basic Setup – RX Gain**
Set the receive gain.
Entries: 0~28 (-14dBm ~ +14dBm)
(Default: 10)
0 = -14dBm
1 = -13 dBm
14 = 0 dBm
27 = 13 dBm
28 = 14 dBm

 - **84-13-28 SIP Trunk CODEC Information Basic Setup – Audio Capability Priority**
Define the CODEC Priority.
Entries: 0=G.711, 2=G.729
(Default: 0)

 - **84-13-31 SIP Trunk CODEC Information Basic Setup – DTMF Payload Number**
Define the DTMF Payload Number.
Entries: 96~127
(Default: 110)

 - **84-13-32 SIP Trunk CODEC Information Basic Setup – DTMF Relay Mode**
Determine the DTMF setup.
Entries: 0 = Disable, 1 = RFC2833
(Default: 0)

SIP DNS Setup

- **10-29-08 SIP Server Information Setup – DNS Server Mode**
Define the DNS Mode. If the SIP carrier provides a domain name, set this option to 1.
Entries: 0 = Off, 1 = On
(Default: 0 = Off)

- **10-29-09 SIP Server Information Setup – DNS Server IP Address**
Define the DNS IP Address (normally provided by the SIP carrier). Enter the carrier-provided information or enter a valid DNS server IP address.
Entries:
0.0.0.1~126.255.255.254
128.0.0.1~191.255.255.254
192.0.0.1~223.225.255.254
(Default: 0.0.0.0)

- **10-29-10 SIP Server Information Setup – DNS Port Number**
Define the DNS Transport port.
Entries: 0~65535
(Default: 53)

SIP NAPT Router Setup

- **10-12-06 CPUII Network Setup – NAPT Router**
Enable (1)/Disable (0) the NAPT Router Setup. With SIP trunking behind a NAPT Router, this must be set to 1.
(Default: 0)
- **10-12-07 CPUII Network Setup – NAPT Router IP Address**
Set the NAPT Router IP address. With SIP trunking, the IP address of the WAN side of the router must be entered.
(Default=0.0.0.0)
- **10-12-08 CPUII Network Setup – ICMP Redirect**
Enable (1)/Disable (0) ICMP (Internet Control Message Protocol) Redirect.
(Default: 0)

SIP System Interconnect Setup

- **10-23-01 SIP System Interconnection Setup – System Interconnection**
Enable (1)/Disable (0) the system interconnection to another system
(Default: 0)
- **10-23-02 SIP System Interconnection – IP Address**
Define the IP address of another system.
Entries:
0.0.0.0 ~ 126.255.255.254
128.0.0.1 ~ 191.255.255.254
192.0.0.1 ~ 223.255.255.254
(Default: 0.0.0.0)
- **10-23-04 SIP System Interconnection Setup – Dial Number**
Determine another system call control port number
Entries: Up to 12 digits (0~9)
(Default = None)

SIP Protocol

- **84-14-06 SIP Trunk Basic Information Setup – SIP UA Trunk Port**
Set the SIP UA (User Authorized) Trunk port number (1~65535) receiving transport for Electra Elite IPK II SIP.
(Default: 5060)
- **84-14-07 SIP Trunk Basic Information Setup – Session Timer Value**
Set the Session Time (0~65535 seconds).
(Default: 0 (session timer Off))
- **84-14-08 SIP Trunk Basic Information Setup – Minimum Session Timer Value**
Set the Minimum Session Time (0~65535 seconds).
(Default: 1800)

➔ **84-14-09 SIP Trunk Basic Information Setup – Called Party Information**

Set the Called Party Information.
Entries: 0 = Request URI, 1 = To Header
(Default: 0)

SIP Server Information Setup

➔ **10-29-01 SIP Server Information Setup – Default Proxy (Outbound)**

Define the SIP Proxy setup, Default Proxy (Outbound). When SIP trunking is used, set this to 1 (On).

If entries are made in Program 10-29-xx for an SIP Server and the SIP Server is then removed or not used, the entries in Program 10-29-xx must be set back to their default settings. Even if PRG 10-29-01 is set to 0 (Off), the Electra Elite IPK II still checks the settings in the remaining 10-29 programs.

➔ **10-29-02 SIP Server Information Setup – Default Proxy (Inbound)**

Turn Off (0) or On (1) the Default Proxy (Inbound).
(Default: 0)

➔ **10-29-03 SIP Server Information Setup – Default Proxy IP Address**

Enter the default Proxy IP Address if the SIP carrier is using an IP address for the proxy. In most cases, this is left at the default entry as the domain name is used.

Entries:
0.0.0.1 ~ 126.255.255.254
128.0.0.1 ~ 191.255.255.254
192.0.0.1 ~ 223.225.255.254
(Default: 0.0.0.0)

➔ **10-29-04 SIP Server Information Setup – Default Proxy Port Number**

Define the Proxy Port Number (0~65535).
(Default: 5060)

➔ **10-29-12 SIP Server Information Setup – Domain Name**

Define the Proxy Domain Name (Electra Elite IPK II domain name).
Entries: 64 characters maximum
(Default: None)

➔ **10-29-13 SIP Server Information Setup – Proxy Host Name**

Define the Proxy Host name (Electra Elite IPK II proxy name).
Entries: 48 characters maximum
(Default: None).

SIP Registrar Setup

➔ **10-29-05 SIP Server Information Setup – Registrar Mode**

Define the Registrar Mode. Always set this to 1 when using SIP trunking.
Entries: 0 = None, 1 = Manual
(Default: 0)

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- ➔ **10-29-06 SIP Server Information Setup – Registrar IP Address**
Define the Registrar IP Address. The carrier may provide an IP address. In most cases, a domain name is used so this entry is left at the default.
0.0.0.0 ~ 126.255.255.254
128.0.0.1 ~ 191.255.255.254
192.0.0.1 ~ 223.225.255.254
(Default: 0.0.0.0)
 - ➔ **10-29-07 SIP Server Information Setup – Registrar Port Number**
Define the Registrar Port Number (0~65535).
(Default: 5060)
 - ➔ **10-29-11 SIP Server Information Setup – Registrar Domain Name**
Define the Registrar Domain Name (normally provided by the SIP carrier).
Example: SIPconnect-sca@L0.cbeyond.net
Entries: 128 characters maximum
(Default: None)
 - ➔ **10-29-15 SIP Server Information Setup – Registration Expiry (Expire) Time**
Define the Registration Expire time – the time (120~65535 seconds) allowed to register with the SIP carrier. This should stay at default.
(Default: 3600)

SIP Server Status

- ➔ **90-10-01 System Alarm Setup – Alarm Type**
Defines alarm type 14 and 60.

Alarm 14 – CPU-LAN Link Error (IP Layer 1)
Assign a Major or Minor alarm status to for the LAN link. This program also assigns whether or not the alarm is displayed to a key telephone and whether or not the alarm information is reported to the predefined destination.

Alarm 60 – SIP Registration Error Notification
Assign a Major or Minor alarm status to for the SIP Registration Error. This program also assigns whether or not the alarm is displayed to a key telephone and whether or not the alarm information is reported to the predefined destination.
- ➔ **90-10-14 System Alarm Setup – CPU-LAN Link Error (IP Layer 1)**
Assign a Major or Minor Alarm status to for the LAN link. This program also assigns whether or not the alarm is displayed to a key telephone and whether or not the alarm information is reported to the predefined destination.
- ➔ **90-10-60 System Alarm Setup – SIP Registration Error Notification**
Assign a Major or Minor Alarm status to for the SIP Registration Error. This program also assigns whether or not the alarm is displayed to a key telephone and whether or not the alarm information is reported to the predefined destination.

- **90-11-01 through 90-11-13 System Alarm Report**
Define the details of the system alarm report (multiline terminal to display alarms and Email notification setup).

SIP Trunk Registration Information Setup

- **10-36-01 SIP Trunk Registration Information Setup – Registration**
Enable (1)/Disable (0) the SIP trunk registration.
(Default: 0)
- **10-36-02 SIP Trunk Registration Information Setup – User ID**
Define the user ID.
Entries: 32 characters maximum
(Default: None)
- **10-36-03 SIP Trunk Registration Information Setup – Authentication User ID**
Define the authentication user ID.
Entries: 48 characters maximum
(Default: None)
- **10-36-04 SIP Trunk Registration Information Setup – Authentication Password**
Define the authentication password.
Entries: 32 characters maximum
(Default: None)

SIP UPnP

- **10-37-01 UPnP Setup – UPnP Mode**
Enable (1)/Disable (0) UPnP task. If UPnP (Universal Plug and Play) task starts, it obtains a NAPT router WAN IP Address by using NAT traversal and saves it in PRG 10-12-07 automatically.
(Default: 0)
- **10-37-02 UPnP Setup – Retry Time**
UPnP task tries to obtain the WAN IP Address of the NAPT router during this time (0~3600 seconds).
(Default: 60)

IPK II In-Mail

Enhancements

With Version 2000 or higher In-Mail Master mailboxes can be used for ACD Delay Announcements. Refer to Automatic Call Distribution (ACD) feature or the In-Mail system guide for more information.

Description

The Elite IPK II In-Mail is a low cost voice mail solution that mounts onto the CPU11. Its programming is fully integrated with KSU programming. This system offers most voice mail system features customers expect.

Automated Attendant automatically answers the system's incoming calls. After listening to a customized message, an outside caller can dial a system extension or use Voice Mail.

There are up to four Elite IPK II In-Mail voice mail ports available. Both 2- and 4 -port In-Mail systems reduce the total station ports available by eight. Integrated Voice Mail enhances the telephone system with the following features:

Call Forwarding to Voice Mail

An extension user can forward their calls to Voice Mail. Once forwarded, calls to the extension connect to that extension mailbox. The caller can leave a message in the mailbox instead of calling back later. Forwarding can occur for all calls immediately, for unanswered calls or only when the extension is busy. When a user transfers a call to an extension forwarded to Voice Mail, the call waits for the Delayed Call Forwarding time before routing to the called extension mailbox. This gives the transferring party the option of retrieving the call instead of having it go directly to the mailbox.

Leaving a Message

Voice Mail lets a multiline terminal extension user easily leave a message at an extension that is unanswered, busy or in Do Not Disturb. The caller can press their Voice Mail key to leave a message in the called extension mailbox. There is no need to call back later.

Transferring to Voice Mail

By using Transfer to Voice Mail, a multiline terminal extension user can Transfer a call to the user's or a co-worker's mailbox. After the Transfer goes through, the caller can leave a message in the mailbox.

 Live Record

While on a call, an extension user can have In-Mail record the conversation. The multiline terminal user can press the In-Mail Record key; and the ESL user dials a code. Once recorded, the Voice Messaging System stores the conversation as a new message in the user's mailbox. After calling their mailbox, a user can save, edit or delete the recorded conversation.

 Personal Answering Machine Emulation

A multiline terminal user can have their idle extension emulate a personal answering machine. This lets In-Mail screen their calls, just like their answering machine at home. If activated, the extension incoming calls route to the user's subscriber mailbox. Once the mailbox answers, the user hears the caller's incoming message. The multiline terminal user can then:

- Let the call go through to their mailbox
- Intercept the call before it goes to their mailbox

 Voice Mail Overflow

If Voice Mail automatically answers trunks, Voice Mail Overflow can reroute those trunks to other extensions when all Voice Mail ports do not answer or, with certain software, are busy. During periods of high traffic, this prevents the outside calls from ringing Voice Mail for an inordinate amount of time. Two types of Voice Mail Overflow are available: Immediate and Delayed. With immediate overflow, calls immediately reroute to other extensions when all Voice Mail ports do not answer or, with certain software, are busy. With delayed overflow, calls reroute after a preset interval. Without overflow, the outside calls ring Voice Mail until a port becomes available or the outside caller hangs up.

 Message Center Mailbox

A Message Center Mailbox is a mailbox shared by more than one extension. Any multiline terminal that has a Message Center Key for the shared mailbox can:

- Listen to the messages stored in the shared mailbox
- Transfer calls to the shared mailbox
- Use many other Voice Mail features previously available only at an extension individual mailbox

A Message Center Mailbox helps co-workers that work together closely – such as members of the same Department Hunt Group or ACD Group. For example, an ACD Group Supervisor can send important messages to the shared Message Center Mailbox, to which any ACD Group member can respond when time allows. Each ACD Agent Message Center Key flashes when messages are waiting. (The Message Center Mailbox can be a mailbox for an installed, uninstalled or virtual extension.)

 Voice Mail Caller ID

In-Mail can use ANI/Caller ID information to identify the outside caller that left a message in a user's mailbox. When the message recipient presses TI after hearing a message, they hear the time the message was sent and the outside telephone number of the message sender. Refer to [Caller ID on page 2-169](#) and [T1 Trunking \(with ANI/DNIS Compatibility\) on page 2-941](#) for more information on setting up this feature.

Voice Mail Queuing

When accessing the voice mail, the system provides a voice mail queue. If all the voice mail ports are busy, any call trying to get to the voice mail is placed in queue. As the voice mail ports become available, the calls are connected to the voice mail in the order in which they were received.

As the Voice Mail Queue follows Department Hunting programming, the queue can hold a maximum of 10 calls. If the queue is full or if the voice mail ports are not assigned to a Department Group, the calls are handled as though there were no voice mail queuing feature enabled. The calls either access voice mail if a port is available or they receive a busy signal.

The Voice Mail Queuing feature does not work with the Conversation Record feature.

Message Key Operates as Voice Mail Key

The system enhances a telephone Message key function when connected to a system which has voice mail installed. When an extension receives a voice mail, the MSG key can be used to check the number of messages in voice mail, and call the voice mail to listen to the messages. If no Voice Mail Programmable Function Key is defined (Program 15-07-01, code 77), the telephone Message Waiting LED flashes to indicate new messages.

This option is not available with a networked voice mail – the voice mail must be local.

In-Mail Available

In-Mail is a plug-in “in-skin” full-featured, DSP-based integrated Voice Mail with Automated Attendant. It is available in two models:

- FM128(2) provides two Voice Mail ports, eight hours of message storage, and up to 160 mailboxes requires a DSPII-U10 Unit.
- FM128(4) provides four Voice Mail ports, eight hours of message storage, and up to 160 mailboxes requires a DSPII-U10 Unit.

The In-Mail Automated Attendant answers incoming calls and routes them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Softkeys guide the display telephone user through the extensive In-Mail feature set.

In-Mail Specifications		
Part Number:	750558	750559
Ports:	2	4
Station Mailboxes:	128	
Routing Mailboxes:	16	
Master Mailboxes:	16	
Total Mailboxes:	160	

In-Mail Specifications (Continued)		
Storage Hours:	8 Hours	8 Hours
Answer Tables:	8	
Dial Action Tables:	16	
Programming Interface:	PCPro or WebPro	
Remote Programming:	Access Via HTML-based WebPro or using customer-provided modems with PCPro.	
Voice Storage Media:	Flash Card (on DSPII-U10 Unit ETU, P/N 750118)	
Languages:	1 (English Mnemonic)	

In-Mail: External Transfer Available

Software allows In-Mail to perform an external transfer. This allows the In-Mail to route an incoming Automated Attendant call out of the Electra Elite IPK II system on a new trunk based on a Speed Dial number stored in a Dial Action Table.

In-Mail: Softkey With Security Code Programming

In-Mail provides softkeys when programming the security code. These softkeys allow a user to select OK, CLEAR or EXIT following an entry of a new code.

In-Mail: Internal Message Notification Timer Lengthened

When Message Notification places a call out, the system waits up to 30 seconds for ringback, reorder, or busy tone from the trunk. If detected, notification call out processing begins normally. If not detected, the system abandons the call and decrements the Ring No Answer (RNA) count. In older software versions, the system waits 15 seconds. This could cause notification callbacks to be inadvertently abandoned.

In-Mail: Directory Dialing

Directory Dialing allows an Automated Attendant caller to reach an extension by dialing the first few letters in the extension user's name. With Directory Dialing, the caller does not have to remember the extension number of the person they wish to reach – just their name. Here is how Directory Dialing works:

1. When the Automated Attendant answers, it sends the call to a Directory Dialing Mailbox. (Optionally, the caller may be asked to dial a digit to access Directory Dialing.)
2. The Directory Dialing Mailbox plays the Directory Dialing Message which asks the caller to dial letters for the name of the person they want to reach.
3. The caller dials the letters for the person's name plus #. They can dial by first name or last name, depending on how the Directory Dialing Message is recorded and the Directory Dialing Mailbox is set up.

4. In-Mail searches the list of programmed extension names for a match of the caller-entered letters.
5. Voice prompts announce the first three matches, and allow the caller to dial a digit (1~3) to reach one of the announced matches. Additionally, the caller can dial 4 to hear additional matches (if any).
6. The caller dials the digit for the extension they wish to reach, and In-Mail sends the call to that extension. The call is sent as a Screened or Unscreened transfer, depending on programming.

For callers to use Directory Dialing, the system must have a name programmed for each extension (up to 15 characters, A~Z, using upper and lower case letters). Each extension should also have a name recorded in their Subscriber Mailbox. In addition, each extension used by Directory Dialing must be installed and must have their Subscriber Mailbox active (Personal or Group).

An outside caller can route to a Master Mailbox or Routing Mailbox programmed as a Directory Dialing Mailbox from:

- The Answer Tables Answer Schedule Override mailbox, Default mailbox, or Routing mailbox.
- A GOTO action in the Dial Action Table of a Call Routing Mailbox.

In-Mail: Multiple Greetings

The mailbox subscriber can record up to three separate greetings and make one of the three active. When a caller leaves a message in the subscriber's mailbox, they hear the active greeting. This allows the subscriber, for example, to record separate greetings for work hours, after work, and during vacation. Instead of rerecording their greeting when they leave the office, they can just activate the after work greeting instead.

If the active greeting is not recorded, a caller leaving a message in the subscriber mailbox hears, *"At the tone, you can leave your message for (extension number or name)."*

Refer to the Electra Elite IPK II In-Mail System Guide for complete details on setting these features.

Conditions

- The Park and Page feature is not available with In-Mail.
- Conversation Record does not work for monitored calls.
- Fixed Call Forwarding can be used to transfer a user's unanswered calls to their voice mail. Call Forwarding does not have to be programmed manually by every user.
- Caller ID information is passed from the Voice Mail to an extension for pre-answer display on an unscreened transfer from Voice Mail.
- Off-premise notification and external extensions require access to outside lines.
- To have the Voice Mail Automated Attendant answer a trunk, program the trunk as a DIL to a Voice Mail port.

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- When the voice mail places a call on hold, it uses Group Hold. Any line appearances for the trunk shows the hold flash rate, however, users cannot pick up these calls (a busy signal is heard).
 - Updating the system time also updates the In-Mail time.
 - Both 2- and 4-port In-Mail systems use eight station ports.
 - VRS Features and In-Mail do not work at the same system.
 - DSPII-U10 Unithas 32 resources which are used for DT/DTMF/Caller ID receiver or Caller ID sender. However, when using In-Mail they do not work.
 - In-Mail and VMS/FMS/VMP/CTI/CTP cannot be used at the same time in the same system.
 - The displayed message count for New and Saved messages does not update until the mailbox user hangs up and calls back into the In-Mail.
 - In-Mail and Analog Voice Mail cannot be used at the same time in the same system.
 - In-Mail does not support Elite IPKII Centralized Voice Mail.
 - When transferring a caller to a mailbox the caller hears the personal greeting from the time the transfer is completed, not from the beginning.
 - KSU must be licensed for ACD and In-Mail for the ACD Delay Announcement feature to work.
 - Master Mailboxes used for ACD Delay Announcements cannot be used for other voice mail features.
 - In-Mail ACD Delay Announcement feature requires KSU software 2.00 or higher.
 - The first port of In-Mail must start with one of the following ports: 1, 5, 9, 12, 16,237, 241, 245, 249, and will use the first port assigned + the next three consecutive ports.
 - Program 41-08-03: ACD Overflow Options – Delay Announcement Source Type is not supported in **Version 1600**.
 - Program 41-19-01~Program 41-19-08: Voice Mail Delay Announcement, is not supported in **Version 1600**.

Default Setting

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

- DSPII-U10 Unit
- IPK II In-Mail – 2 Port, 8 Hours
- IPK II In-Mail – 4 Port, 8 Hours

Related Features

Automatic Call Distribution (ACD)

Barge-In

Caller ID

Call Forwarding

Central Office Calls, Placing

Clock/Calendar Display

Direct Inward Line (DIL)

Hold

Message Waiting

One-Touch Calling

Programmable Function Keys

Transfer

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**

Assign at least one circuit for DTMF reception (type 0 or 1).
Use the following as a guide when allocating DTMF receivers:

 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.

- ➔ **11-07-01 Department Group Pilot Numbers – Dial**

Assign a Department Group pilot number for the Voice Mail (eight digits maximum). The extensions are assigned to the group in Program 16-02-01.

- ➔ **11-12-52 Service Code Setup (for Service Access) – Live Monitoring (In-Mail)**

Define access code used for In-Mail Live Monitoring (VRS). At default this program is not set.

- ➔ **15-02-26 Multiline Telephone Basic Data Setup – MSG Key Operation Mode**

Determine whether an extension MSG key should function as a Message key (0) or Voice Mail key (1) (Default = 0). If set as a Message key, the user can press the key to call the voice mail only when they have new messages. If set as a Voice Mail key, it functions as a normal Voice Mail key (it is not active if Centralized Voice Mail is used).

- ➔ **15-02-28 Multiline Telephone Basic Data setup – Message Waiting Lamp Color**

Select whether the Message Wait LED located at the top of the multiline terminal flashes green (0) or red (1) when a Message Wait indication is flashing. By default, this option is set to 1.

- ➔ **15-03-01 Single Line Telephone Basic Data Setup – SLT Signaling Type**

For each Electra Elite IPK II voice mail extension, this option must be set to 0.

- ➔ **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**

Set all SLIU ports used for Voice Mail as type 1.

- ➔ **15-03-09 Single Line Telephone Basic Data Setup – Caller ID Function – for External Module**

Set to 0 when voice mail is used or the integration code for the disconnect function is incorrect.

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- ➔ **15-07-01 Programmable Function Keys**

Assign a Voice Mail key to an extension. You must enter the Voice Mail key code (code 77) followed by:

 - Your own extension number if you are setting up your own Voice Mail key.
 - A virtual extension number if you are setting up a Message Center key for a virtual extension.
 - A co-worker's extension number if you are setting up a Message Center key for an installed extension.
 - An uninstalled extension number if you are setting up a Message Center key for an uninstalled extension.

(Optional) Assign a Voice Mail Record key to an extension (code 78).
(Optional) Assign a Personal Answering Machine Emulation key (code 16).
(Optional) Use a Call Redirect key (49) to allow a user to transfer a call to another extension or voice mail without answering the call.

 - ➔ **16-02-01 Department Group Assignment for Extensions**

Put all the Voice Mail extensions in a unique Department Group (do not use Department Group 1). No other extensions should be in this group. This allows DILs to Voice Mail to ring other Voice Mail extensions when the DIL assigned port is busy. Only one Voice Mail group is allowed per system. Be sure to set up the Department Group in order. (For example, the first extension should be order 1, the second order 2, etc.)

 - ➔ **20-02-09 System Options for Multiline Telephones – Disconnect Supervision**

Enable (1) disconnect supervision for the system.

 - ➔ **20-06-01 Class of Service for Extensions**

Assign a Class of Service (1~15) to the voice mail extensions. Using COS 14 for all time modes is recommended.

 - ➔ **20-09-02 Class of Service Options (Incoming Call Service) – Caller ID Display**

Set to On (1) for the voice mail COS.

 - ➔ **20-11-01 Class of Service Options (Hold/Transfer Service) – Call Forward All**

Enable (1)/Disable (0) an extension user ability to set Call Forward Immediate.

 - ➔ **20-11-02 Class of Service Options (Hold/Transfer Service) – Call Forward When Busy**

Enable (1)/Disable (0) an extension user ability to set Call Forwarding when Busy.

 - ➔ **20-11-03 Class of Service Options (Hold/Transfer Service) – Call Forwarding When Unanswered**

Enable (1)/Disable (0) an extension user ability to set Call Forwarding when Unanswered.

 - ➔ **20-11-04 Class of Service Options (Hold/Transfer Service) – Call Forwarding (Both Ringing)**

Enable (1)/Disable (0) an extension user ability to set Call Forwarding with Both Ringing.

 - ➔ **20-11-05 Class of Service Options (Hold/Transfer Service) – Call Forwarding with Follow Me**

Enable (1)/Disable (0) an extension user ability to set Call Forward with Follow Me.

 - ➔ **20-11-12 Class of Service Options (Hold/Transfer Service) – Call Forwarding Off-Premise (External Call Forwarding)**

Set to off (0) for the voice mail COS.
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- ➔ **20-13-01 Class of Service Options (Supplementary Service) – Long Conversation Alarm**
- 20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
- 20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
- 20-13-07 Class of Service Options (Supplementary Service) – Message Waiting**
- 20-13-16 Class of Service Options (Supplementary Service) – Barge-In, Receive**
- 20-13-28 Class of Service Options (Supplementary Service) – Allow Class of Service to be Changed**
- 20-13-35 Class of Service Options (Supplementary Service) – Block Camp On**
Set these options to Off (0) for the voice mail COS.

- ➔ **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**
Enable (1) Continued Dialing for all extensions that dial Voice Mail features.

- ➔ **20-13-15 Class of Service Options (Supplementary Service) – Barge-In, Initiate**
Set to On (1) for the voice mail COS.

- ➔ **22-01-04 System Options for Incoming Calls – DIL No Answer Recall Time**
If setting up Delayed Voice Mail Overflow, enter a time greater than 0. Overflow occurs after this interval (provided the other related programming is correct). If setting up Immediate Voice Mail Overflow, set to 0.

- ➔ **24-02-02 System Options for Transfer – MOH or Ringback on Transferred Calls**
Enable (1)/Disable (0) MOH on Transfer. If enabled, a transferred caller hears Music on Hold while their call rings the destination extension. If disabled, a transferred caller hears ringback while their call rings the destination extension. For this option to work with voice mail, the transferred call must be an unscreened transfer.

Assign Trunks As Automated Attendant Trunks – Method 1

- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign Service Type 4 to each trunk you want to ring into Voice Mail as a Direct Inward Line (DIL).

- ➔ **22-07-01 DIL Assignment**
Assign the master/pilot number of the voice mail group from Program 11-07-01 as the DIL destination. If all Voice Mail ports are in the same unique Extension (Department) Group (see Program 16-02 above), the DIL rings another Voice Mail port if its assigned port is busy.

Assign Trunks As Automated Attendant Trunks – Method 2

- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign Service Type 0 to each trunk you want to ring into Voice Mail as a normal line.

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- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign Ring Group 102 for an In-Skin/External Voice Mail, or 103 for a Central Voice Mail as the destination.

For Either Method:

- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
To enable Voice Mail Overflow, assign selected extensions to a Ring Group that rings for unanswered DILs to Voice Mail ports. In Program 22-06, enter 1 to enable overflow ringing.
- ➔ **22-08-01 DIL/IRG No Answer Destination**
For Voice Mail Overflow, enter the Ring Group that unanswered DILs to Voice Mail ring after the DIL Call Waiting time (Program 22-01-04).
- ➔ **24-02-03 System Options for Transfer – Delayed Call Forwarding Time**
Set the time a transferred call waits at a forwarded extension before routing to the called extension mailbox.
- ➔ **45-01-01 Voice Mail Integration Options – Voice Mail Department Group Number**
Assign which Extension (Department) Group number is to be assigned as the voice mail group. An entry of 0 means no voice mail is installed.
Department Groups: 0 (default), 1~64
- ➔ **45-01-02 Voice Mail Integration Options – Voice Mail Master Name**
Enter the Voice Mail master name up to 12 characters.
- ➔ **45-01-04 Voice Mail Integration Options – Park and Page**
Enable (1)/Disable (0) the system ability to process the Voice Mail Park and Page (*) commands. You should normally enable this option. Default is 1.
- ➔ **45-01-05 Voice Mail Integration Options – Message Wait**
Enable (1)/Disable (0) the system ability to process the Voice Mail Message Wait (#) commands. You should normally *enable* this option. If enabled, be sure that the programmed Message Notification strings do not contain #9 for trunk access. When using an external voice mail and centrex transfer, this option should be disabled or the service code #3 in Program 11-12-42 must be changed.
- ➔ **45-01-06 Voice Mail Integration Options – Record Alert Tone Interval Time**
This time (0~64800 seconds) sets the interval between Voice Mail Conversation Record alerts. Default is 30
- ➔ **45-01-14 Voice Mail Integration Options – CCIS Centralized Voice Mail Number**
Assign (for remote stations only) the pilot number (up to eight digits) to Centralized Voice Mail over CCIS Link. Default is No Setting.
- ➔ **45-01-15 Voice Mail Integration Options – Analog Voice Mail Protocol Selection**
Select whether Fixed codes (0) or the codes used in Program 45-04 (1) are used for analog voice mail protocol. Default is 0. **Version 1500 or higher is required.**
- ➔ **45-01-16 Voice Mail Integration Options – Voice Mail Fax Digit Add Assignment**
Assign up to four digits in front of the station number sent to the SLT port when a call is forwarded. Default is None. **Version 1500 or higher is required.**

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- ➔ **45-01-17 Voice Mail Integration Options – Reply Mailbox Number**
Select Yes (1) to include or No (0) to not include the Mailbox number in the analog voice mail protocol. a Reply Mailbox Number. Default is 1. **Version 1500 or higher is required.**
 - ➔ **45-01-18 Voice Mail Integration Options – Trunk Number Mapping**
Assign the digits (2 or 3) of Trunk Number Mapping. Default is 2. **Version 1600 or higher is required.**
 - ➔ **47-01-01 through 47-01-17 IPKI II In-Mail System Options**
 - ➔ **47-02-01 through 47-02-15 IPK II In-Mail Station Mailbox Options**
 - ➔ **47-03-01 through 47-03-03 IPK II In-Mail Master Mailbox Options**
 - ➔ **47-04-01 through 47-04-04 Master Call Routing Mailbox Options**
 - ➔ **47-05-01 through 47-05-03 Master Announcement Mailbox Options**
 - ➔ **47-06-01 through 47-06-13 Master Subscriber Mailbox Options**
 - ➔ **47-07-01 IPK II In-Mail Routing Mailbox Options – Routing Mailbox Active**
47-07-02 IPK II In-Mail Routing Mailbox Options – Routing Mailbox Type
 - ➔ **47-08-01 through 47-08-04 Call Routing Mailbox Options**
 - ➔ **47-09-01 through 47-09-03 Announcement Mailbox Options**
 - ➔ **47-10-01 IPKI II In-Mail Trunk Options – Answer Table Assignment**
 - ➔ **47-11-01 through 47-11-04 IPKI II In-Mail Answer Table Options**
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- ➔ **47-12-01**
through
47-12-08 IPKI II In-Mail Answer Schedules
 - ➔ **47-13-01 IPKI II In-Mail Dial Action Tables**
Refer to the Electra Elite IPK II In-Mail System Guide, for complete programming details.
 - ➔ **80-03-01 DTMF Tone Receiver Setup – Detect Level**
80-04-01 Call Progress Tone Detector Setup – Detect Level
Set the criteria for DTMF dial, ringback and busy tones. Default for Type 1~5 for 80-03-01 is 0.
Default for Type 1~3 for 80-04-01 is 0 (-25 dBm); for Type 4 and 5 default is 0.

ACD Delay Announcement:

Use only if Master Mailboxes are used for ACD Delay Announcements.

- ➔ **41-19-01 ACD Voice Mail Delay Announcement – Delay Message Start Timer**
Determine the time the system waits before playing the Delay Message. **Version 2000 or higher is required.**
Options
0 ~ 64800 seconds
(default: 0)
- ➔ **41-19-02 ACD Voice Mail Delay Announcement – Mailbox Number for 1st Announcement Message**
Assign Voice Mail ACD Announcement Mailbox as the message source for the 1st Announcement Message. **Version 2000 or higher is required.**
Options
Dial up to eight digits
(default: No Setting)
- ➔ **41-19-03 ACD Voice Mail Delay Announcement – 1st Delay Message Sending Count**
Determine the 1st Delay Message Sending Count. This entry must be set to 1 or higher for the message to play. **Version 2000 or higher is required.**
Options
0 = No message played, 1 ~ 255
(default: 0)
- ➔ **41-19-04 ACD Voice Mail Delay Announcement – Mailbox Number for 2nd Announcement Message**
Assign Voice Mail ACD Announcement Mailboxes as the message source for the 2nd Announcement Message.
Options
Dial up to eight digits
(default: No Setting)
- ➔ **41-19-05 ACD Voice Mail Delay Announcement – 2nd Delay Message Sending Count**
Determine the 2nd Delay Message Sending Count. This entry must be set to 1 or higher for the message to play.
Options
0 = No message played, 1 ~ 255
(default: 0)

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- ➔ **41-19-06 ACD Voice Mail Delay Announcement – Wait Tone Type at Message Interval. Version 2000 or higher is required.**
Options
0 = Ring Back Tone
1 = MOH Tone
2 = BGM Source
(default: 0)
- ➔ **41-19-07 ACD Voice Mail Delay Announcement – ACD Forced Disconnect Time after 2nd Announcement.**
Assign the time the system should wait after the end of the ACD delay message before disconnecting. **Version 2000 or higher is required.**
Options
0 ~ 64800 seconds
(default: 0)
- ➔ **41-19-08 ACD Voice Mail Delay Announcement – Delayed Message Interval Time.**
Set the time between the Delayed Messages.
Options
0 ~ 64800 seconds
(default: 20)
- ➔ **47-03-01 IPK II In-Mail Master Mailbox Options – Master Mailbox Active**
Enable (1)/Disable (0) the Master Mailbox. A Master Mailbox is not accessible when it is disabled.
(default: 0)
- ➔ **47-03-02 IPK II In-Mail Master Mailbox Options – Master Mailbox Number**
The Master Mailbox Number is the same as the Department Group master (pilot) number. Select the Department Group master (pilot) number associated with the Master Mailbox you are programming.
Options
Digits (seven maximum, using 0~9)
No Setting (entered by pressing Hold)
(default: No Setting)
- ➔ **47-03-03 IPK II In-Mail Master Mailbox Options – Master Mailbox Type**
Set the Master Mailbox type. There are four types of Electra Elite IPK II In-Mail mailboxes: Call Routing boxes are used for ACD Delay Announcements.
Options
0 - Undefined
1 - Call Routing
2 - Announcement
3 - Subscriber
4 - Directory
(default: 0)

- ➔ **47-05-01 Master Announcement Mailbox Options – Next Call Routing Mailbox**
If you set up an Announcement Mailbox for ACD Delay Announcements, provide additional routing options to the Automated Attendant callers.
- Options
 - Number (01~16)
 - Undefined (0)
 - (default: 0)

Operation

Calling Your Mailbox

To call your mailbox:

With a multiline terminal, your Voice Mail key flashes green and your Message Center keys flash red when they have messages waiting. If you do not have a Voice Mail key, your Message Waiting LED flashes instead.

Multiline Terminal


1. Press your **Voice Mail** key (PRG 15-07 or SC 751: 01 + *8).

- OR -

Press the **Vmsg** softkey.

- OR -


Press the **Message** key on the telephone, if equipped.


 *Your mailbox number is normally the same as your extension number. You may optionally dial a co-worker's mailbox - or use this procedure to call your mailbox from a co-worker's telephone.*

- OR -

Press **Speaker** and dial *8.


2. If requested by Voice Mail, enter your security code.

 *Ask your Voice Mail system administrator for your security code.*

 *Normally, your Message Waiting (MW) LED goes out (if applicable). If it continues to flash, you have unanswered Message Waiting requests or a new General Message. Go to To check your messages below.*

Single Line Telephone






1. Lift the handset and dial *8.

 *If you are at a co-worker's telephone, you can dial the Voice Mail master number and your mailbox number instead. You can also use this procedure from your own telephone to call a co-worker's mailbox.*




2. If requested by Voice Mail, enter your security code.

Checking Messages

If PRG 15-02-26 = 0 (Message Key):


1. Press the **Message** key once.
 -  *The user can use the VOL UP and VOL DOWN keys to view the new messages. If there are both voice mail messages and Message Waiting calls, the display indicates the number of new voice mail messages and then each Message Waiting call is shown.*
 -  *When there are new messages, the Message Waiting LED on the telephone flashes red.*
 -  *To return a displayed Message Waiting, press Speaker or lift the handset.*
2. To return a displayed Message Waiting, press **Speaker** or lift the handset.
To listen to the voice messages, with Voice Message displayed, press **Speaker** or lift the handset.
 -  *The voice mail is called.*
 -  *The voice mail is only called if there are new messages. If the display indicates Check Messages No Messages, then press the Exit key to return the telephone to idle.*

If PRG 15-02-26 = 1 (Voice Mail Key):

1. Press the **Message** key once.
 -  *The voice mail is called.*
 -  *When there are new messages, the Message Waiting LED on the telephone flashes red.*
 -  *With this option set, the MSG key can be used as a Voice Mail key for any function [calling voice mail or transfer call a to voice mail (Hold + MSG + Extension Number), etc.].*

Leaving A Message (Multiline Terminal Only)

To leave a message in the mailbox of an unanswered extension (the extension you call can be busy, in DND or unanswered):

1. Press the **Voice Mail** key (PRG 15-07 or SC 751: code 77 + In-Mail pilot).
 - OR -
 - Press the **Message** key on telephone, if equipped.
 - OR -
 - Dial **8**.
 -  *The Voice Mail System will prompt you to leave a message.*

Forwarding Calls to Your Mailbox

To activate or cancel Call Forwarding:

1. Press **Speaker** (or lift the handset at the single line telephone) and choose from the following dial access codes:
 - 741 = Call Forward – Immediate (PRG 15-07 or SC 751: code 10)
 - 742 = Call Forward – Busy (PRG 15-07 or SC 751: code 11)

743 = Call Forward – No Answer (PRG 15-07 or SC 751: code 12)
744 = Call Forward – Busy/No Answer (PRG 15-07 or SC 751: code 13)

2. Dial the Voice Mail master number.
3. Press **Speaker** to hang up (or hang up handset at the single line telephone).

Transferring Calls to a Mailbox

To transfer your active call to a mailbox:

Multiline Terminal

1. Press **Hold**.
2. Press the **Voice Mail** key (PRG 15-07 or SC 751: code 77 + In-Mail pilot).

- OR -


Press the **Message** key on the telephone, if equipped.

3. Dial the number of mailbox to receive the transfer.

 *This number can be a mailbox number or a co-worker's mailbox number.*

- OR -

Press the **DSS Console** or **One-Touch** key for extension user's mailbox, which receives the transfer.

 *If the Transfer destination is an extension forwarded to Voice Mail, the call waits before routing the called user's mailbox. This gives you the option of retrieving the call instead of having it picked up by Voice Mail.*

4. Hang up.

 *Voice Mail prompts your caller to leave a message in the mailbox you selected.*


- OR -

1. Dial extension number or press a DSS Console key for the extension mailbox which receives the transfer.
2. Press the Voice Mail key (PRG 15-07 or SC 751: code 77 + In-Mail pilot)

- OR -

Press the **Message** key on the telephone, if equipped.


3. Hang up.

 *Voice Mail prompts your caller to leave a message in the mailbox you selected.*

Single Line Telephone

1. Hookflash.

Dial Voice Mail master number followed by destination mailbox.



 *If the Transfer destination is an extension forwarded to Voice Mail, the call waits before routing to the called user's mailbox. This gives you the option of retrieving the call instead of having it picked up by Voice Mail.*

2. Hang up.



Recording Your Call

To record your active call in your mailbox:



Multiline Terminal

1. Press the **Voice Mail Record** key (PRG 15-07 or SC 751: code 78)
 -  You hear a beep and your Record key flashes. The system beeps periodically to remind you that you are recording.
 -  To stop recording, press the Voice Mail Record key again. You can restart and stop recording as required.

- OR -

1. Press **Hold** to Place the call on hold.
2. Dial **654**.
 -  The system automatically reconnects you to your call.
 -  To stop recording, place the call on hold then pick the call back up. You can restart and stop recording as required.

Single Line Telephone

1. Hookflash.
2. Dial **654**.
 -  The system automatically reconnects you to your call.
 -  To stop recording, hookflash twice. You can restart and stop recording as required.

Personal Answering Machine Emulation (Multiline Terminal Only)


To enable or cancel Personal Answering Machine Emulation:

1. Press **Speaker** (or lift the handset at the single line telephone) and choose from the following dial access codes:
 - 741 = Call Forward – Immediate
 - 742 = Call Forward – Busy
 - 743 = Call Forward – No Answer
 - 744 = Call Forward – Busy/No Answer
 - 745 = Call Forward – Both Ring
 - 746 = Call Forwarding – Follow Me
2. Dial the Voice Mail master number.
3. Press **Speaker** to hang up (or hang up handset at the single line telephone).

When Personal Answering Machine Emulation broadcasts your caller's message, you can:


Your telephone must be idle (not on a call).

1. Do nothing.

 The message is automatically being recorded in your mailbox. The broadcast stops when your caller hangs up.


- OR -

1. Lift the handset to intercept the call.

 You connect to the caller. The system records the first part of the message in your mailbox. The line key changes from red to green.

- OR -

Press **Speaker** to cut off the message broadcast and send the call to your mailbox.


 Voice Mail records the entire message in your mailbox.

Checking Your Messages (Multiline Terminal Only)

To check your messages:

1. Press the **Message** key.

2. Dial ***0**.

 You can have any combination of the message types in the table below on your telephone.

If you see. . .	You have. . .
VOICE MESSAGE n MESSAGES	New messages in your Voice Mail mailbox
CHECK MESSAGE VRS GENERAL MESSAGE	Not listened to the current General Message
CHECK MESSAGE (name)	Message Waiting requests left at your telephone by your co-workers

3. Press VOL **▲** or VOL **▼** to scroll through your display.

4. When you find the message you want to answer, press **Speaker**. You can either:

- Go to your Voice Mail mailbox.
- Listen to the new General Message.
- Automatically call the extension that left you a Message Waiting.

Directory Dialing

Recording a Directory Dialing message:

1. Log onto the System Administrator's mailbox: **SA** (72) or press **0** to play a Help message.
2. Select Instruction Menus: **I** (4).

3. Enter the Directory Dialing Mailbox number or press # to go back to the System Administrator Options.
4. Select one of the following options:
 - L (5)** = Listen to the current Directory Dialing Message (if any)
 - #** = Exit listen mode
 - R (7)** = Record a new Directory Dialing Message
 - * = *Pause or restart recording*
 - **E (3)** = *Erase recording*
 - **#** = *Exit recording mode*
 - E (3)** = Erase the Directory Dialing Message
 - #** = Go back to the System Administrator options

Using Directory Dialing:

1. After the Automated Attendant answers, wait for the Directory Dialing Message. The Automated Attendant may ask you to dial a digit for Directory Dialing.
2. Dial the letters that correspond to the name of the person you want to reach + #.
 - The Directory Dialing Message tells how many letters you need to dial, and whether you should enter the person's first name or last name.
 - To exit Directory Dialing without selecting a name, just dial #.
3. The Automated Attendant announces the name matches, and tells you which digit to dial (1~3) to reach each of the announced names.
 - To hear additional name matches (if any), dial 6 instead.
4. After you make your selection, the Automated Attendant routes your call to the name you select.

IPK II – PC Assistant

Enhancements

Version 1600 supports:

- An updated 3rd party TAPI drivers to provide screen pops for incoming calls to Virtual extensions. **Version 1600 system software and version 2.0 TAPI drivers are required.**
- A Microsoft Outlook Add-in to provide screen pops and dial out abilities without full PC Assistant installation.

Description

The Elite PC Assistant enhances the operation of the NEC digital telephone set by providing easy access to common, and not so common, IPK II voice control features. This software application provides a very intuitive user interface that can be conveniently located at the top, side, or bottom of the PC screen. The user interface can even "shrink" into the edge of the screen and become visible when a call arrives, or when the user moves the mouse to the edge of the display.

In addition to quick access to these IPK II features, the Assistant provides a call log for easy viewing of recent received, missed, or made calls – just like your cell phone. It also includes a directory to keep your commonly dialed numbers close at hand, and optional features like voice recording, personal greeting, and screen pops using Microsoft Outlook, ACT! 2005, or Goldmine 6.7 or higher or Elite (this option is available only when using the Professional version and a CTU is required).

PC Assistant has the following main components:

1. Elite PC Assistant Application Software
This application runs on a PC and provides the PC-based GUI (Graphical User Interface) and features.
2. Telephony Admin
This application interfaces between the PC Assistant and the 3rd party TAPI drivers.
3. CTU/CTA Adapter
This adapter is installed on the multiline telephone and interfaces the Electra Elite IPK II KSU with the USB (Universal Serial Bus), or Serial port on the PC. Using the optional CTU adapter is required for voice recording and personal greeting features.
4. Headset (Optional)
The headset can be plugged into the multiline telephone and used when making or receiving calls with the Elite PC Assistant.

Elite PC Assistant runs on a PC and communicates with the Electra Elite IPK II through a normal digital station port using the CTU Adapter attached to the telephone. When calls come into this station, the PC Assistant displays it on the PC, and provides several features that allow the user to handle the call quickly. Elite PC Assistant can be minimized to run in the background and pop to the front when call activity occurs. Calls can then be handled using either the keyboard or the mouse. The user speaks to the caller through the telephone handset, headset, or speakerphone of the multiline telephone the application is running on.

Conditions

- The PC Assistant requires a CTU Adapter installed on the multiline terminal with connection directly to the client PC for Call Recording and Personal Greeting voice functions.
- If using a CT(U)-R for recording, the adaptor is supported only on the ESIB(8)-U20 or ESIE(8)-U20 ETU.
- Incoming calls to a virtual extension that appear on stations that are used with the CTI applications, PC Assistant, or PC Attendant, do not show up as a second call in the CTI application.
- The CPUII()-U10 ETU must be licensed for the CTI feature.
- The Microsoft Outlook Add-in uses the same license as PC Assistant. If PC Assistant is already licensed on the machine the Add-in is installed, it is automatically activated.
- PC Assistant cannot dial digits while a call is in progress.
- When parking calls through PC Assistant, parked Intercom calls do not show.

Default Setting

Not Applicable

System Availability

Terminals

All Multiline Terminals

Required Component(s)

- PC running Windows XP or Windows 2000
- PC Assistant Client License
- 3rd Party TAPI Drivers (IPKII TSP)

- Red Phoenix Telephony Admin (CTI Server)
- CTI License for CPUII()-U10 ETU

Optional Component(s)

The CT(U)-R Unit is only supported on the ESIB(8)-U20 or ESIE(8)-U20 ETU.

Related Features

Refer to the Electra Elite IPK II PC Assistant Installation Manual for detailed feature information.

Programming

- ➔ **10-12-01 CPUII Network Setup – IP Address**
IPK II IP Address = 0~255.0~255.0~255.0~255
- ➔ **10-12-02 CPUII Network Setup – Subnet Mask**
IPK II Subnet Mask = 0~255.0~255.0~255.0~255
- ➔ **10-12-03 CPUII Network Setup – Default Gateway**
IPK II Default Gateway = 0~255.0~255.0~255.0~255
- ➔ **10-20-01 LAN Setup for External Equipment – TCP Port**
Select the TCP port number used by the Telephony Admin on the CTI Server (8181 is recommended).

In addition to the above programming, define the programming options as required for the system features. Refer to the Electra Elite IPK II Programming Manual for programming details. Callback, Callback Request, and Auto Redial are not supported.

Operation

Refer to the Electra Elite IPK II PC Assistant Installation Manual for detailed operations.

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IPK II – PC Attendant

Enhancements

Version 1500 supports updated 3rd party TAPI drivers to provide screen pops for incoming calls to Virtual Extensions.

Version 1600 supports:

- Phone Display Messages (from PC Attendant to phone). **Version 1600 system software and version 2.0 TAPI drivers are required.**
- Speed Dial buttons (Programmable Speed Dial buttons on PC Attendant).

Description

The PC Attendant Console is a software product that provides access to the most common functions required by an operator or receptionist. By using this application in conjunction with an IPK II S, M, L, or XL system, the attendant can easily manage their call handling tasks without having to switch their attention between the telephone and the PC. A company directory, recording ability, and PC-to-PC messaging, provide additional features to further enhance the operation. (The attendant telephone requires a CTU Adapter installed with connection directly to the end-user PC for the Call Recording and Personal Greeting voice functions).

The PC Attendant application uses a CTI (Computer Telephony Integration) service that is provided with the product to integrate with the IPK II telephone systems. Through the CTI service, an ethernet connection over the company LAN, and an IPK II multiline terminal for audio, the PC Attendant application can monitor all extensions on the phone system and control the actions of the attendant phone, including placing calls.

The CTI service on the PC communicates with the IPK II system through the CTI port on the telephone system. An administration utility (Telephony Administrator) is provided which allows the system administrator to configure the global settings for the console application.

The PC Attendant application can be installed also on multiple PCs for installations that need to support more than one attendant position (each attendant PC requires a licensed version of the PC Attendant installed). Up to eight PC Attendant positions can be installed.

The PC Attendant application also includes a supporting application, call Quick Message. By installing the Quick Message client on individual PCs, the attendant can quickly send short messages to other employees, who can respond with a single keystroke.

The host PC used as the CTI Server requires Windows XP, Windows 2000, or Windows Server 2003 and an interface to the IPK II system through the 3rd-party CTI link to monitor and control the telephone activity. When installing the PC Attendant Console on multiple PCs for more than one attendant position, the PC requires Windows XP or Windows 2000.

Conditions

- The attendant telephone requires a CT(U)-R Unit installed with connection directly to the end-user PC for the Call Recording and Personal Greeting voice functions).
- If using a CT(U)-R, the unit is supported only on the ESIB(8)-U20 or ESIE(8)-U20 ETU.
- Incoming calls to a virtual extension that appear on stations that are used with the CTI applications, PC Assistant, or PC Attendant, do not show up as a second call in the CTI application.
- The CPUII()-U10 ETU must be licensed for the CTI feature.
- PC Assistant cannot dial digits while a call is in progress.
- When parking calls through PC Assistant/PC Attendant, parked Intercom calls do not show.
- When initiating a conference in PC Attendant, if the 3rd party hangs up before the conference is completed, the original call no longer shows in Attendant.
- If the Attendant presses the disconnect button in PC Attendant to abort a transfer, the call shows up in Attendant as a held call. The Attendant is not automatically connected back to the caller.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)


The CT(U)-R Unit is supported only on the ESIB(8)-U20 or ESIE(8)-U20 ETU.

CTI license for CPUII()-U10 ETU

Related Features

Refer to the IPK II PC Attendant Manual for detailed feature information.

Programming

- ➔ **10-12-01 CPUII Network Setup – IP Address**
IPK II IP Address = 0~255.0~255.0~255.0~255
 - ➔ **10-12-02 CPUII Network Setup – Subnet Mask**
IPK II Subnet Mask = 0~255.0~255.0~255.0~255
 - ➔ **10-12-03 CPUII Network Setup – Default Gateway**
IPK II Default Gateway = 0~255.0~255.0~255.0~255
 - ➔ **10-20-01 LAN Settings for External Equipment – TCP Port**
Select the TCP port number to be used by the PC Attendant (8181 is recommended).
-  *In addition to the above programming, define the programming options as required for the system features. Refer to the Electra Elite IPK II Programming Manual for programming details.*

Operation

Refer to the Electra Elite IPK II PC Attendant Installation Manual for detailed operations.

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IPK II VoIP Management System

Enhancements

Version 1500 or higher is required to support this feature.
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Description

The Electra Elite IPK II VoIP Management (IVM) System efficiently monitors and manages the performance of enterprise VoIP networks and provides multiple monitoring tools for the NEC IPK Key Systems with IAD(8)-U10 ETUs. The IPK VoIP Management Remote Unit and the IPK VoIP Management software are included in the system that is installed on a centralized server (Qovia Central). Together these products closely monitor voice performance in the converged network, notify system administrators concerning performance anomalies, and allow Information Technology (IT) teams to manage voice infrastructure components remotely.

 *The IPK VoIP Management System currently supports only the IAD(8)-U10 ETU VoIP applications. The PVA(X) applications are not currently supported. The PVA(X) ETU with CCIS application can be added to the database, but the MoS scores are inaccurate. The PVA(X) ETU with MG16 and associated terminals cannot be added to the database.*

HIGHLIGHTS

- Centralized monitoring using one simple user interface
- Intuitive, remote system and network performance monitoring
- Secure, role-based system user administration

ARCHITECTURE

The IVM Remote Unit connects to the network using Ethernet and captures the Voice Network call quality metrics. The “virtual hands” of the Management Remote Unit collect the real-time call quality data and report it to Qovia Central for aggregation and display. The IVM Remote Unit interfaces with the NEC IPK Key system, LAN switches, and a UPS to provide dedicated VoIP monitoring and management abilities. Depending on the size and design of the customer network, a solution may include one or more IVM Remote Units.

The WEB-based IVM Software integrates with the IVM Remote Unit and provides a consolidated Instrument panel view to monitor the VoIP Infrastructure for one or more locations. The software aggregates the real-time data collected by the IVM Remote Unit and displays it in easy-to-read graphs and charts.

The IVM Software, Qovia Central, supports display of the VoIP infrastructure in both a user-defined hierarchy and an application generated topology. The operation/performance of various VoIP devices is displayed in Red, Yellow, or Green to provide an instant visual status check. An intuitive point-and-click interface gives quick and easy access to any device requiring attention. Qovia Central, is easy to install and configure on a Microsoft Windows™ workstation or server.

For detailed information, refer to the IVM System Users Manual provided with the software.

Conditions

General

The IVM Owner's Guide includes a chapter devoted to the configuration and use of each of the following functions:

Application Administration

Provides basic tools to monitor and manage the voice network including:

- Firewall traversal ability to remote management unit
- Ability to create users, assign privileges and assign multiple users
- Ability to create, modify, or delete nodes in the Device Administration Tree

Network Discovery

Auto discovery of handsets and soft phones using SNMP and proprietary software

- Discovered VoIP infrastructure elements are displayed in a configurable topology diagram
- Import of diagram or map for topology background is supported

Call Quality Monitoring

Calculates and displays the call quality metrics that reflect the overall call quality experienced by the caller

- A real-time MOS score for every call is provided, even during the call.
- Near real-time data on packet loss, packet jitter, and packet delay is provided in graphic format with the option to view historical data.
- Threshold-based email notifications and/or SNMP traps are supported. The user can customize threshold values to minimize false alarms.

Active Packet Testing

Allows user to see how voice packet traffic is flowing in real-time. It can be used to:

- Verify and trace passively identified problems
- Proactively test the network during pre-deployment or day-to-day operations
- Trigger active packet testing when needed to prevent testing packets from clogging the voice network traffic

- Schedule active packet testing at regular intervals or when a particular threshold is met
- Run on-demand active packet tests with configurable settings, including test duration
- UPS Monitoring
 - Power status for all local and remote locations is provided.
 - When line power is lost proactive notification is provided
 - UPS load and battery level are monitored using a serial port on the IVM Remote Unit
- Configuration Management
 - A Universal Command Line Interface with two serial ports per IVM Remote Unit is provided for remote serial port access for Routers, Switches, Call Server, and Gateway.

Service Conditions

General:

- An IVM Remote Unit must be installed on the LAN of a MEGACO IAD ETU. A remote unit is recommended for each LAN with monitored VoIP endpoints. A remote unit is not required on the LAN of the Qovia Central site.
- Call quality is measured from VoIP endpoint to VoIP endpoint. The endpoint can be an IAD ETU or an ITH terminal.
- Monitoring call quality requires VoIP traffic to be visible to the monitoring port of the IVM Remote Unit. This requires port mirroring of the applicable port on an Ethernet switch. A HUB can be used only when no other voice or data device is connected to the HUB. Adding other voice or data devices on the HUB causes voice quality degradation.
- Call quality is measured using the RTCP data included in the RTP streams of VoIP calls. The IAD ETU RTCP reporting supports delay with a granularity of two milliseconds.
- Call quality can be accomplished only after the discovery process is successfully completed. This process involves Telnet queries to the IAD ETUs and SNMP queries to the Ethernet switches.

Restrictions:

- For Call Quality Monitoring, Set History feature, a 30-day history, is available, but it can be set only in the current calendar month.
- Client port customization is not yet supported. Client connections are fixed using TCP port 8080 for unencrypted browsing and TCP port 8443 for encrypted browsing.
- The following sites are not covered:
 - MEGACO station sites with low-end switches (without SNMP and Multiport/VLAN mirroring)
 - Multiple Trunk sites with low-end switches (without SNMP and Multiport/VLAN mirroring)
 - IAD sites with old software versions that do not have RTCP calculation corrections
 - PVA(X)-U() ETU based applications

- Monitoring packets without compliant RTCP data (NEAX, most 3rd party vendor systems) is not supported.

System Availability

Server Minimum Hardware Requirements:

- Pentium III 1.5 GHz or faster CPU
- 512MB Memory (1GB recommended)
- 40 GB Hard Disk Space
- Microsoft Windows XP Professional with SP1, Windows 2000 or 2003 Server Software, Microsoft Internet Explorer 5.5 or higher on Windows XP Pro and 2003 server
- Browsers to support Java Plug-in 1.4.1_04
- VGA Color Monitor with 1024x768 resolution
- Mouse or similar pointing device
- Network adapter card configured with a static IP address and connected to the network

Client Minimum Hardware Requirements:

- Pentium III 450 MHz or faster CPU
- Minimum 256 MB physical RAM
- 200 MB hard disk drive (for installation; decreases after installation is complete)

Minimum Integration Access Device (IAD) Software Release:

- MEGACO Station 4.01
- IP-CCIS Trunk 4.00
- SIP Trunk 2.07
- H.323 Trunk 2.01

Network Requirements:

- NTP server to provide time synchronization Source for NTP service
 - Local Server Preferred
 - Qovia Central server that includes option to provide NTP service
 - Public Server (e.g., time.nist.gov) that requires internet access and UDP port 123 open on firewall

- ❑ Switch support for SNMP with bridge MIB to enable discovery of IP terminals, IAD station ETUs, or multiple IAD Trunks in a single KSU (not required at single IP Trunk sites)
- ❑ Switch support for port mirroring of all VoIP devices (Mirror all ports or mirror a VLAN)

Operating Procedures

Refer to the product manuals for core applications: Application Administration, Network Discovery, Call Quality Monitoring, Active Packet Testing, UPS Monitoring, and Configuration Management.

IVM Configuration Scenarios

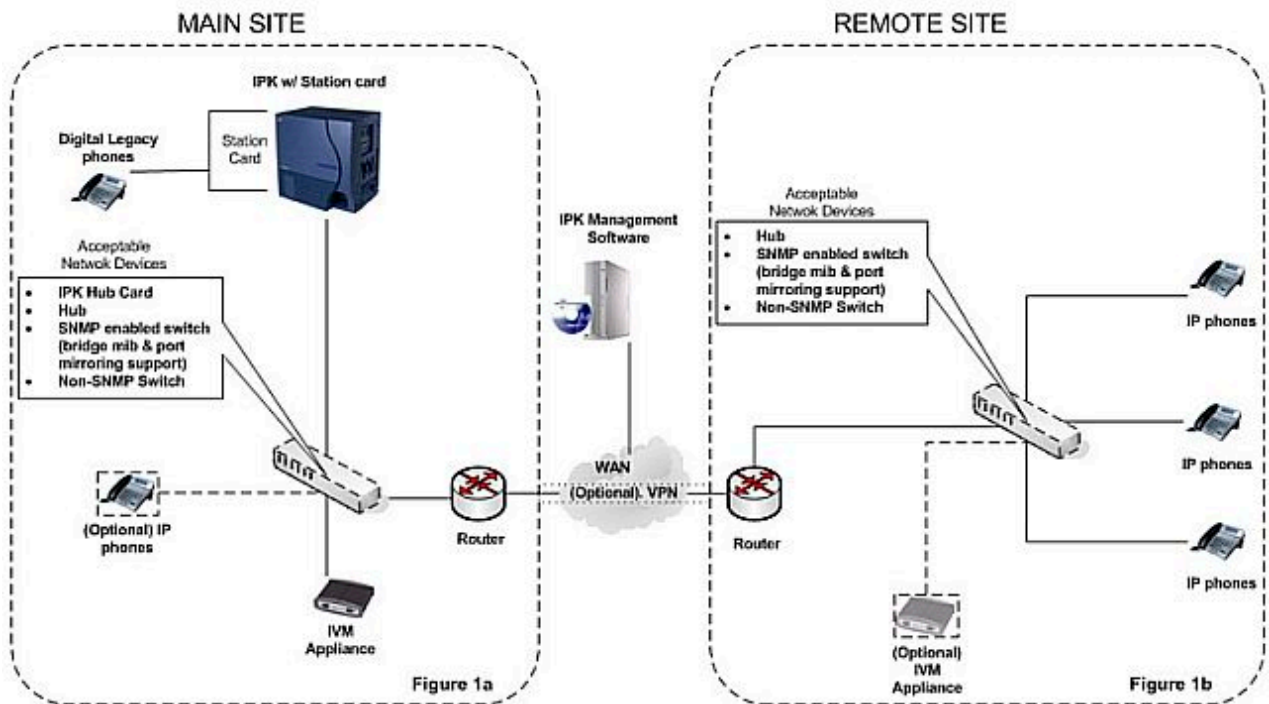


Figure 2-16 MEGACO Station Card

MEGACO Station Card Scenarios Optional Device Configurations	Features Summary (Main Site)					
	IVM Main	Discover Phones	Monitor Phones	Monitor Station Card	Perform Active Tests	Notification
Network Equipment						
IPK HUB ETU/HUB	YS	NO	NO	NO	YS	NO
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	YS	YS	YS
SNMP switch (Bridge MIB – No Mirroring)	YS	YS	NO	NO	YS	NO
Non-SNMP enabled switch (Mirroring)	YS	NO	NO	NO	YS	NO
Non-SNMP enabled switch (No Mirroring)	YS	NO	NO	NO	YS	NO

MEGACO Station Card Scenarios Optional Device Configurations	Features Summary (Remote Site)					
	IVM Main	Discover Phones	Monitor Phones	Monitor Station Card	Perform Active Tests	Notification
Network Equipment						
HUB	YS	NO	NO	N/A	YS	NO
HUB	NO	NO	NO	N/A	NO	NO
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	N/A	YS	YS
SNMP switch (Bridge MIB and Mirroring)	NO	NO	NO	N/A	NO	NO
SNMP switch (Bridge MIB – No Mirroring)	YS	YS	NO	N/A	YS	NO
SNMP switch (Bridge MIB – No Mirroring)	NO	NO	NO	N/A	NO	NO
Non-SNMP enabled switch (Mirroring)	YS	NO	NO	N/A	YS	NO
Non-SNMP enabled switch (Mirroring)	NO	NO	NO	N/A	NO	NO
Non-SNMP enabled switch (No Mirroring)	YS	NO	NO	N/A	YS	NO
Non-SNMP enabled switch (No Mirroring)	NO	NO	NO	N/A	NO	NO

- ✎ *The IVM should always be optimally positioned where it can monitor a majority of the telephone traffic.*
- ✎ *Active tests can be performed only between switches that are configured to be managed by an IVM appliance. When the Discovery process is unsuccessful, only schedule-triggered Active Tests can be performed between these switches.*
- ✎ *When there is an IVM at the MEGACO site, but not at the remote site, calls from the remote are recorded under the MEGACO Station Card, but no summary reporting for the individual telephones at the remote site is provided. The IP address of the remote telephones is displayed on the MEGACO Station call records.*

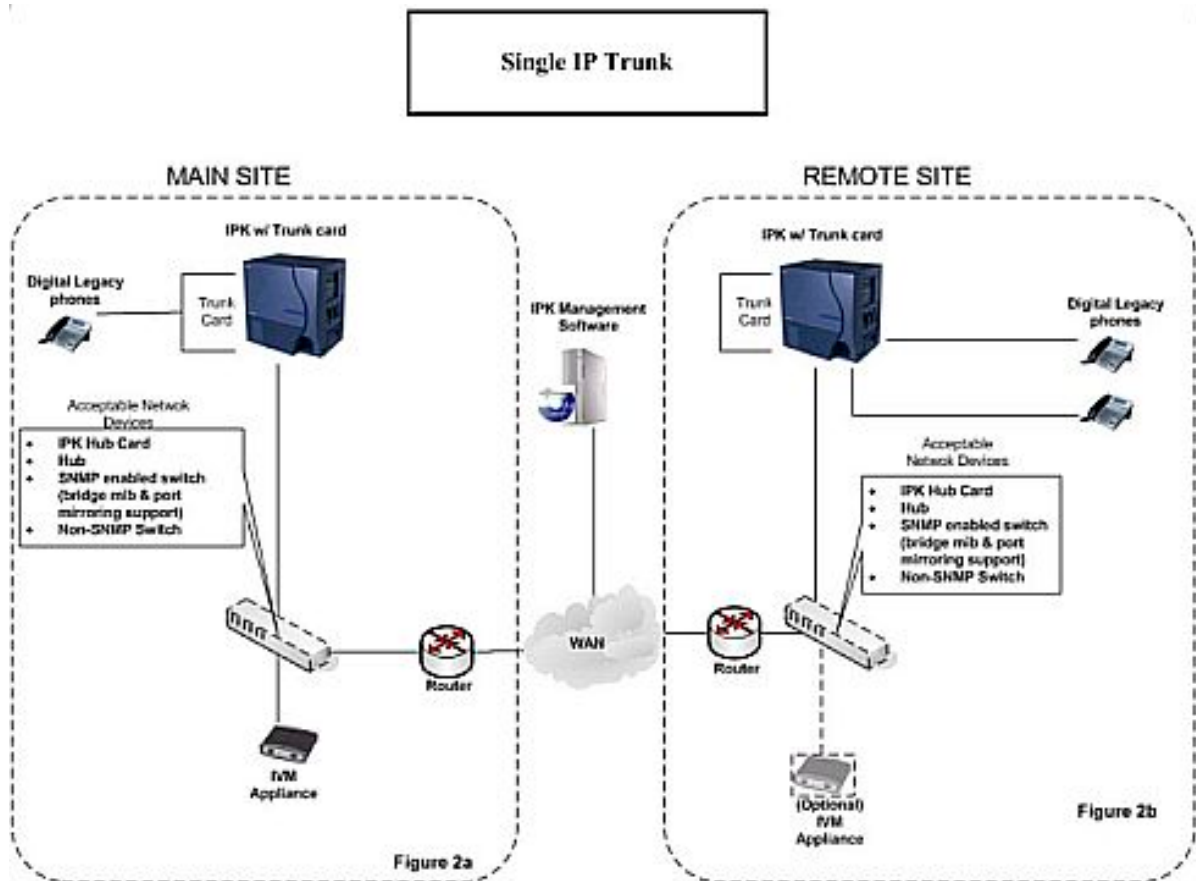





Figure 2-17 Single IP Trunk

IAD Trunk (one IAD at a site) Optional Device Configurations	Features Summary (Any Site)			
Network Equipment	IVM	Monitor Trunk	Perform Active Tests	Notification
IPK HUB ETU/HUB	YS	YS	YS	YS
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	YS
SNMP switch (Bridge MIB – No Mirroring)	YS	NO	YS	NO
Non-SNMP enabled switch (Mirroring)	YS	YS	YS	YS
Non-SNMP enabled switch (No Mirroring)	YS	NO	YS	NO
Router and dumb HUB	YS	YS	YS	YS

-  *The IVM should always be optimally positioned where it can monitor a majority of the telephone traffic.*
-  *Active tests can be performed only between switches that are configured to be managed by an IVM appliance. When the Discovery process is unsuccessful, only schedule-triggered Active Tests can be performed between these switches.*
-  *A work-around solution for a switch that does not support mirroring is to add a 4-port dumb HUB without switching that connects the IAD trunk, the IVM ports, and the upstream port to the router or the switch. HUBs should not be used to connect more than one VoIP device. When other devices are included voice degradation results.*

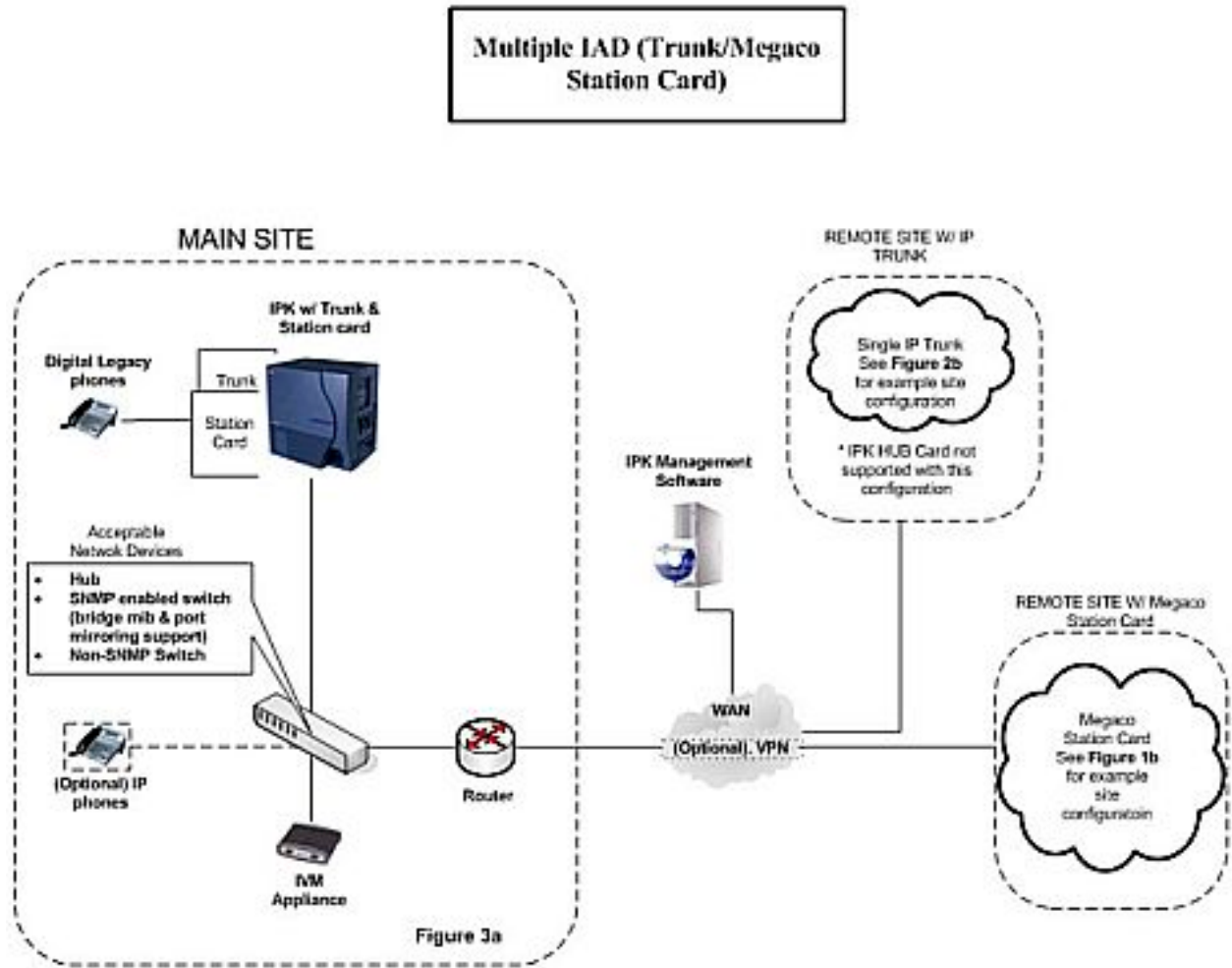


Figure 2-18 Multiple IAD (Trunk/MEGACO Station Card)

Multiple IAD (Trunk/MEGACO Station Card) Scenarios Optional Device Configurations	Features Summary (Main)					
Network Equipment	IVM Main	Discover Phones	Monitor Phones	Monitor Station Card	Perform Active Tests	Notification
IPK HUB ETU/HUB	YS	NO	NO	NO	YS	NO
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	YS	YS	YS
SNMP switch (Bridge MIB – No Mirroring)	YS	YS	NO	NO	YS	NO
Non-SNMP enabled switch (Mirroring)	YS	NO	NO	NO	YS	NO
Non-SNMP enabled switch (No Mirroring)	YS	NO	NO	NO	YS	NO

Remote MEGACO Station Card site Optional Device Configurations	Features Summary (Remote)					
Network Equipment	IVM Main	Discover Phones	Monitor Phones	Monitor Station Card	Perform Active Tests	Notification
IPK HUB ETU/HUB	YS	NO	NO	N/A	YS	NO
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	N/A	YS	YS
SNMP switch (Bridge MIB – No Mirroring)	YS	YS	NO	N/A	YS	NO
Non-SNMP enabled switch (Mirroring)	YS	NO	NO	N/A	YS	NO
Non-SNMP enabled switch (No Mirroring)	YS	NO	NO	N/A	YS	NO

Remote IP Trunk Card site Optional Device Configurations	Features Summary (Remote)			
Network Equipment	IVM	Monitor Trunk	Perform Active Tests	Notification
IPK HUB ETU/HUB	YS	YS	YS	YS
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	YS
SNMP switch (Bridge MIB – No Mirroring)	YS	NO	YS	NO
Non-SNMP enabled switch (Mirroring)	YS	YS	YS	YS
Non-SNMP enabled switch (No Mirroring)	YS	NO	YS	NO
Router and dumb HUB	YS	YS	YS	YS

- 📎 *The IVM should always be optimally positioned where it can monitor a majority of the telephone traffic.*
- 📎 *Active tests can be performed only between switches that are configured to be managed by an IVM appliance. When the Discovery process is unsuccessful, only schedule-triggered Active Tests can be performed between these switches.*
- 📎 *Telephones connected to a HUB cannot be discovered and therefore cannot be monitored.*

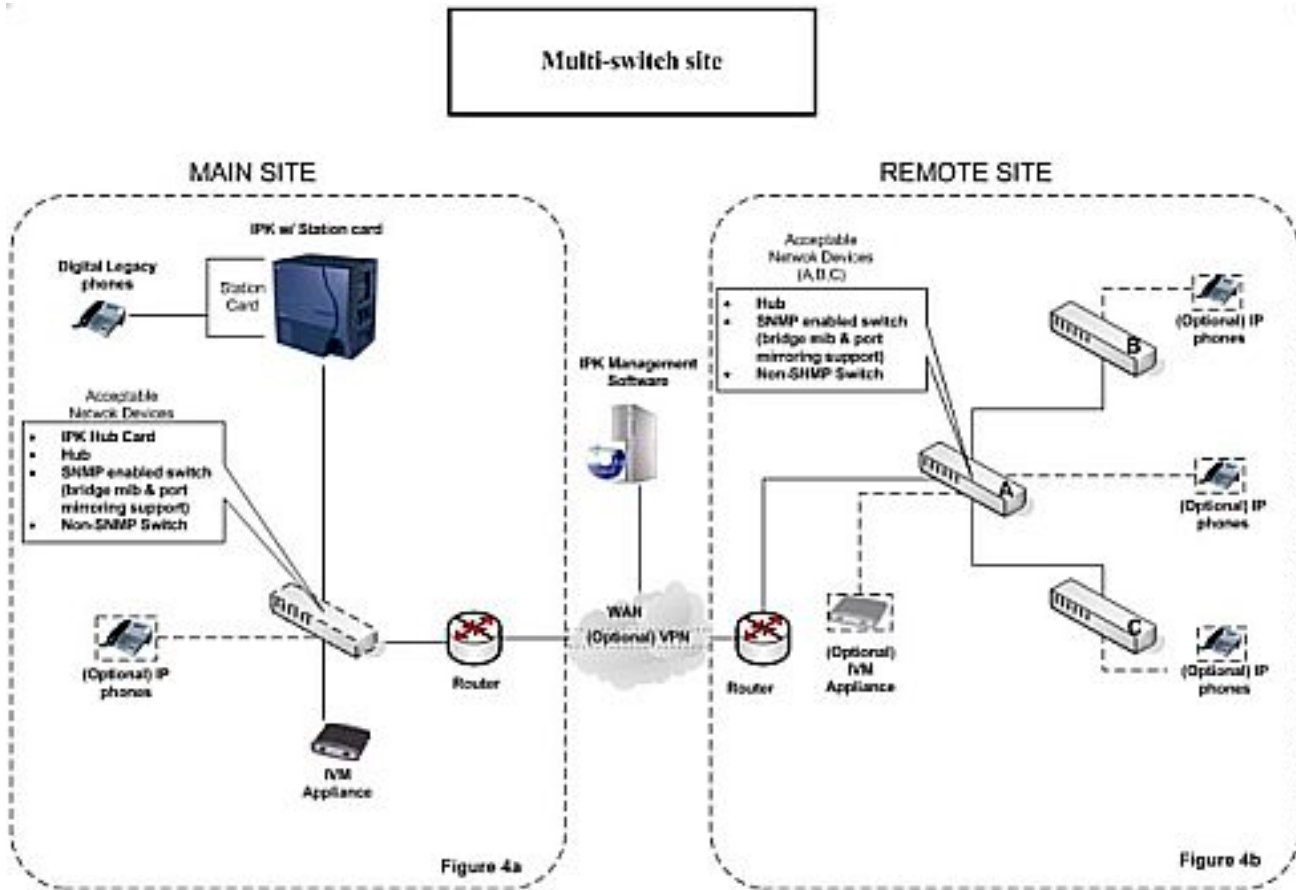





Figure 2-19 Multiswitch Site


Multiswitch Station Card Scenarios Optional Device Configurations	Features Summary (Main Site)					
	IVM Main	Discover Phones	Monitor Phones	Monitor Station Card	Perform Active Tests	Notification
IPK HUB ETU/HUB	YS	NO	NO	NO	YS	NO
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	YS	YS	YS
SNMP switch (Bridge MIB – No Mirroring)	YS	YS	NO	NO	YS	NO
Non-SNMP enabled switch (Mirroring)	YS	NO	NO	NO	YS	NO
Non-SNMP enabled switch (No Mirroring)	YS	NO	NO	NO	YS	NO


Multiswitch Station Card Scenarios Optional Device Configurations	Available Features (Remote Site)					
	IVM Main	Discover Phones	Monitor Phones	Monitor Station Card	Perform Active Tests	Notification
HUB	YS	NO	NO	N/A	YS	NO
SNMP switch (Bridge MIB and Mirroring)	YS	YS	YS	N/A	YS	YS
SNMP switch (Bridge MIB – No Mirroring)	YS	YS	NO	N/A	YS	NO
Non-SNMP enabled switch (Mirroring)	YS	NO	NO	N/A	YS	NO
Non-SNMP enabled switch (No Mirroring)	YS	NO	NO	N/A	YS	NO


 None of the available features are supported unless at least one switch on the Main Site LAN supports SNMP with bridge MIB support.

 Monitoring of telephones requires at least one switch at the remote site that supports SNMP with bridge MIB support.

 **HUBs are never recommended for use with VoIP except to mirror a single port.**

 Telephones connected to a HUB cannot be discovered and therefore cannot be monitored.

 The IVM should always be optimally positioned where it can monitor a majority of the telephone traffic.


 Active tests can be performed only between switches that are configured to be managed by an IVM appliance. When the Discovery process is unsuccessful, only schedule-triggered Active Tests can be performed between these switches.

Switch Configuration Specifics

Switch A

When the IVM is positioned on Switch A, as shown in [Figure 2-19 Multiswitch Site](#), a majority of the Telephone communication at the remote site can be monitored because it is a gateway for switches B and C.

The IVM cannot monitor the inter-switch communication that occurs on Switches B and C, but it can monitor external calls destined for or originating from telephones connected to switches B and C. To monitor the inter-switch telephone communication an additional IVM must be placed on each respective switch.

 *The inter-switch communication limitations apply only when peer-to-peer communication is enabled on the MEGACO station application.*

Switch B

When the IVM is positioned on Switch B, only the inter-switch communication on switch B and any external call destined for or originating from a telephone connected to Switch B can be monitored.

Switch C

When the IVM is positioned on Switch C, only the inter-switch communication on Switch C and any external call destined for or originating from a telephone connected to Switch C can be monitored.

ISDN Compatibility

Enhancements

Version 1500 or higher:

- The T1/PRI Interface ETU can be programmed as a 4/8/12/16/20/24-port Fractional T1/PRI (when PRG 10-39-01 is Enabled).
- Supports Called Party Number Enhancement so the IPK II can edit the Called Party Number Parameter using the Network Specified Parameter Table settings (PRG 26-12-01 and PRG 26-12-02).

Description

ISDN-BRI

Integrated Service Digital Network – Basic Rate Interface (ISDN-BRI) is a Public Switched Telephone Network (PSTN) service that provides two B channels and a D channel (2B + D) for voice call trunking. The B channels provide two voice path connections. Caller ID is usually a standard feature on ISDN-BRI provided trunks. Caller ID indication displays the calling party telephone number on the LCD of the Multiline Terminal for CO incoming calls. This interface provides voice communication path only.

ISDN-PRI

ISDN-PRI (Integrated Service Digital Network – Primary Rate Interface) is a Public Switched Telephone Network (PSTN) service that provides 23 B channels and a single D channel (23B+1D) for trunking. Caller ID indication displays the calling party telephone number on the LCD of the Multiline Terminal for CO incoming calls. This interface provides voice communication path only.

ISDN – BRI/PRI Features

- DID Line Service
When configured for DID Line Service, the trunks emulate Loop Start or Ground Start trunks for outgoing calls and DID trunks for incoming calls.
- Calling Line Identification Presentation (CLIP)
PRG 10-03-05: ETU Configuration – CLIP Information Announcement, allows the Calling Party Number IE in the Setup Message for a call when placed out an ISDN Trunk.

Calling Party Number (CPN) Presentation from Station

Calling Party Number (CPN) Presentation from Station allows each unique station or virtual extension 10-digit number (representing the DID number of the originating station) to be sent out over the ISDN Network, if it is programmed. If no Extension Calling Number is assigned, the system sends the calling number for the ISDN trunk. If both the extension and trunk information is programmed, the extension information is sent as it takes priority. When the option for calling party subaddress is on, the extension number is sent as the subaddress information. Both the calling party number and calling party subaddress are sent in a SETUP message as the calling party information element and a calling party subaddress information element. Allow the system to send the subaddress by setting the following programs: 10-03-05=1, 15-01-04=1, 20-08-13=1, 21-13-01 = enter number to be sent.

 Enhanced 911 Service with ISDN PRI

Enhanced 911 Service is supported by defining an available ISDN PRI trunk as a E911 trunk. When a station user dials 911, 9+ 911, or CO line key + 911, the E911 trunk is selected and the Public Safety Answering Point (PSAP) is called. The PSAP is provided callback information generated from the Electra Elite IPK system.

 SMDR Includes Dialed Number

The SMDR report can optionally print the trunk name (entered in system programming) or the number the incoming caller dialed (i.e., the dialed ISDN digits). This gives you the option of analyzing the SMDR report based on the number your caller dials. (This option also applies to a DID trunk as well.)

 Display Shows Why Caller ID is Not Available

With Caller ID enabled, the system provides information for ISDN calls that do not contain the Caller ID information. If the Caller ID information is restricted, the telephone display shows PRIVATE. If the system cannot provide Caller ID information because the Telco information is not available, the display shows OUT OF AREA.

Conditions

 Primary Rate Interface (PRI):

The system is compatible with ISDN Primary Rate Interface (PRI) services. PRI services currently supported include:

- Basic PRI Call Control (BCC)
- Display of incoming caller's name and number when allowed by Telco
- Routing in the system based on the number the caller dialed
- ISDN maintenance functions (such as In Service/Out of Service Messaging)
- Speech and 3.1 KHz audio

PRI ability requires the installation of DTI-U40 ETU with PRT Firmware version 5.XX or higher. Each ETU (also called a PRI circuit) provides 24 PRI channels (23B + D) 4 with 64K Clear Channel response. The T1/PRI Interface ETU uses a single slot. When installed, the T1/PRI Interface ETU uses the first block of 24 consecutive trunks. For example, if you have an COI (8)-U() ETU installed for trunks 1~8, the T1/PRI Interface ETU automatically uses trunks 9~32. If you have COI (8)-U() ETUs installed for trunks 1~8 and 17~24, the T1/PRI ETU uses trunks 25~48. The T1/PRI Interface cannot use trunks 9~16 (even if available) since they are not part of a consecutive block of 24 trunks. (**Version 1100 or lower and Version 1500 or higher** when PRG 10-39-01 is Disabled.)

- When using fractional PRI, the card comes up as zero ports until PRG 10-03-06 is set to the 4/8/12/16/20/24(auto), and then reset.
- If fractional PRI has the number of ports changed, the Trunk Port number might change if they would become split or could fit into an empty gap of trunk ports. For example, if you have a COI(8) ETU for Trunks ports 1~8 and 17~24 and the PRI (12 ports) was assigned as 25~36 and the PRI is changed to be eight ports instead of 12 ports, the new trunk port numbers would be 9~16 because the eight ports can now fit into the gap without being split ports. Another example, if you have a COI(8) ETU for Trunks ports 1~8 and 17~24 and the PRI (8 ports) was assigned as 9~16 and then you change the PRI to be 12 ports instead of eight, the new trunk ports would be 25~36 because the ports would have to be split to keep the original port numbers, and this is not supported. (**Version 1500 or higher is required** when PRG 10-39-01 is Enabled.)
- If using a CSU/DSU, PRG 10-03-12 must be set to Short, and PRG 10-03-13 set to 0. If not using a CSU/DSU, PRG 10-03-12 must be set to Long, and PRG 10-03-13 must be set to 1~7 or anything other than 0.
- Restrictions for Calling Party Name:

The Electra Elite IPK II supports receiving the name from the Network in supported formats only and cannot send the Calling Name. Refer to [Table 2-28 Restrictions for Calling Party Names](#).

Table 2-28 Restrictions for Calling Party Names

Protocols	Name Delivery Formats
NI-2	Facility Information Elements
4ESS (AT&T Custom)	Not Supported
AT&T5ESS Lucent Custom	Facility Information Element
DMS-100 (Custom) *	Display Information Element *
DMS-100 (National; ISDN) **	Facility Information Element **

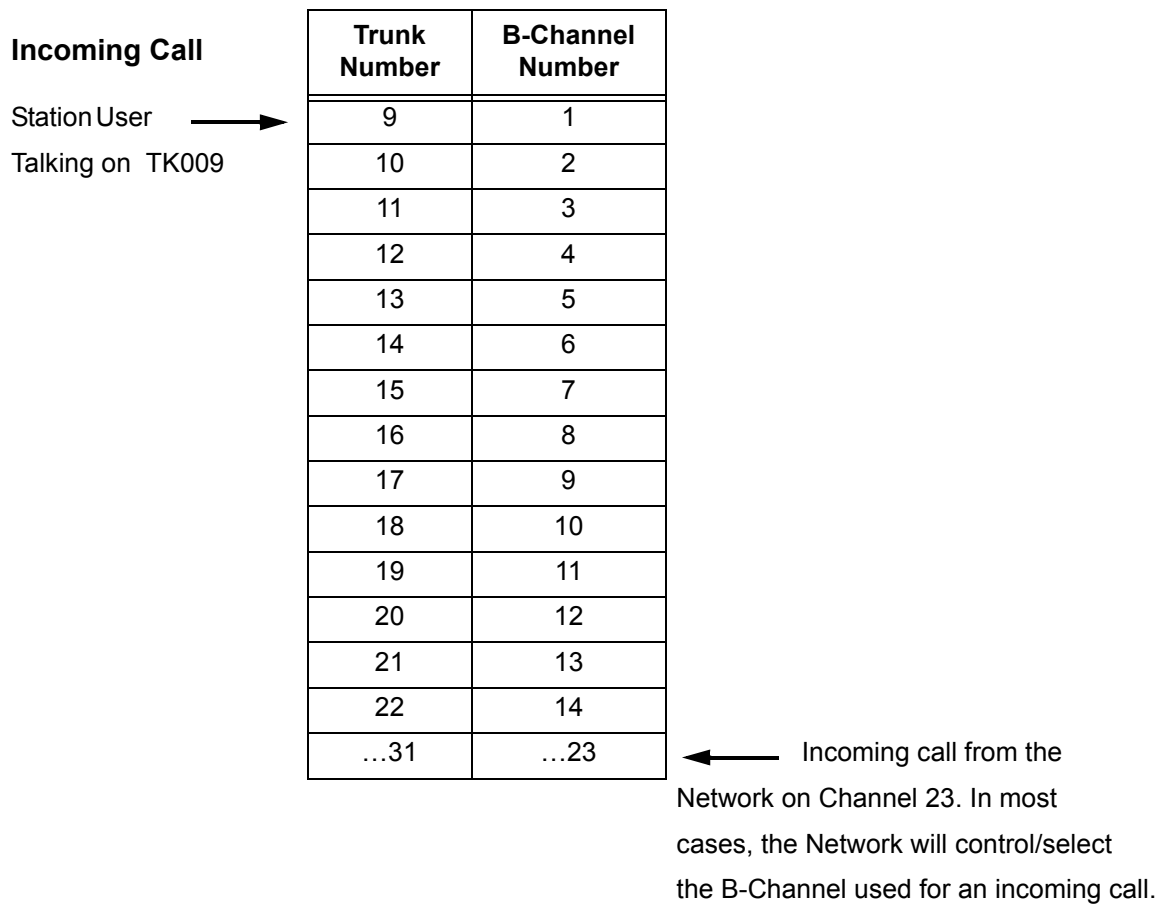
* Nortel Specification NIX-A211-1

** Nortel Specification NIS-A233-1


- CO Line Service is not supported.
ISDN – PRI cannot be configured for CO Emulation
- B-Channel to Trunk Association

When an Incoming ISDN-BRI/PRI call is received, the system assigns the lowest trunk number of the ISDN circuit to the incoming call associated with the B-Channel. When an Outgoing call is placed using the ISDN-PRI/BRI, the system assigns the Trunk and B-Channel association according to the chart below. This is based on the Trunk-to-Trunk Group and Trunk Group Priority assignment in (PRG 14-05-01).

Refer to the charts below for examples:



Outgoing Call	Trunk Number	Trunk Group	Trunk Priority	B-Channel Number
	9	1	9	1
	10	1	8	2
	11	1	7	3
	12	1	6	4
	13	1	5	5
	14	1	4	6
	15	1	3	7
	16	1	2	8
Station user →	17	1	1	9
places outgoing trunk	18	2	3	10
call by dialing Trunk	19	2	2	11
Access code. Outgoing	20	2	1	12
call is placed on the	21	3	11	13
associated B-Channel.	22	3	10	14
	...31	3	...1	...23

 *In addition to T1/PRI interface ETUs, PRI also requires a CSU/DSU Unit and interconnecting cables to interface with the Telco.*

○ Basic Rate Interface (BRI)

Caller ID Name to Single Line Telephone is *NOT* supported for ISDN (BRT) Trunks.

The system is compatible with ISDN Basic Rate Interface (BRI) services. BRI services currently supported include:

- Basic BRI Call Control (BCC)
- Point-to-Point BRI Terminal Connection (no daisy-chaining)
- Multipoint BRI Terminal Connection (daisy-chaining)

BRI services require the installation of BRT(4)-U20 ETUs. Each BRT(4)-U20 ETU has four BRI circuits. The BRT(4)-U20 ETU uses a single universal slot.

For each BRI line, two different Terminal Endpoint Unidentified (TEIs) are assigned to two different Service Profile Identifiers (SPIDs).

The two different SPIDs for each BRI line, are related to different trunk logical port numbers. One BRI provides two trunk logical ports when it is connected to a CO line. Each SPID is assigned to a different TEI. This relationship is made in the initialization of the BRI line when it is connected to the CO.

This relationship between SPID and TEIs are created as follows.

LOGICAL-PORT-NUMBER + 0 = SPID-1

LOGICAL-PORT-NUMBER + 1 = SPID-2

When using the SMDR reports for BRI, all incoming BRI calls are displayed under the CLASS column as IVIN.

○ Automatic Data Link Failure Recovery

If a data link error is detected by the BRI ETU, the system tries to recover the data link and send the SPID to the central office. To provide this enhancement, the BRI ETU must be able to indicate to the system when a data link error has occurred.

In addition to the BRI Interface ETU, BRI Services require the installation of NT1 Network Terminators and interconnecting cabling.

○ CO Line Service is not supported.

ISDN-BRI cannot be configured for CO Emulation.

○ BRI and DID Callers with Non-Matching SPID Numbers

This feature allows you to determine whether the system checks the called party number with the SETUP message and the SPID setup. Depending on the system programming, this can allow DID calls to be received on BRI trunks and direct them according to the DID Translation Table (Program 22-11).

○ Special Conditions Related to Ordering DID Service For ISDN-BRI

Telcos may refer to this in different ways. The reference Verizon uses to order such service is Additional Directory Numbers with no new terminating equipment (only a dialable number). When you want Additional Directory Numbers to hunt when a B-Channel is busy, the service may be called Busy Diversion.

○ Calling Party Number (CPN) presentation from station is available for virtual extensions.

Default Setting

None

System Availability

Terminals

Not Applicable

Required Component(s)

To provide ISDN-PRI trunk connection:

- DTI-U40 ETU

To provide ISDN-BRI trunk connection:

- BRT(4)-U20 ETU
- NT-1 for each BRI (locally provided)

Related Features

Central Office Calls, Answering

Central Office Calls, Placing

Direct Inward Dialing (DID)

Direct Inward Line (DIL)

E-911 Compatibility

Forced Trunk Disconnect

Station Message Detail Recording

Transfer

Programming

ISDN – BRI Installation

- ➔ **10-03-01 ETU Setup – For BRI PKG**
Set up and confirm the Basic Configuration data for each BRT (4)-U20 ETU. Select the ISDN Line Mode

0 = Not Set
1 = T-Bus (Default)
- ➔ **10-03-03 ETU Setup – Connection Type (For BRI PKG)**
Set up and confirm the Basic Configuration data for each BRT(4)-U20 ETU. Confirm the connection type for or each BRT(4)-U20 ETU


0 = Point-to-Multipoint (Default)
1 = Point-to-Point
- ➔ **10-03-04 ETU Setup – Layer 3 Timer Type (For BRI PKG)**
Set up and confirm the Basic Configuration data for each BRT(4)-U20 ETU. This program selects the Layer 3 timer type (1~5). Each timer value of Layer 3 is set up for each type in Program 81-06 (T-Bus).
- ➔ **10-03-08 ETU Setup – Dial Sending Mode (For BRI PKG)**
Set up and confirm the Basic Configuration data for each BRT(4)-U20 ETU. Select either enblock or overlap sending.

0 = Enblock Sending
1 = Overlap Sending (Default)
- ➔ **10-03-09 ETU Setup – Dial Information Element (For BRI PKG)**
Set up and confirm the Basic Configuration data for each BRT(4)-U20 ETU. If Overlap Sending is selected in Program 10-03-08, select either Keypad Facility (0) or Called Party Number (1) for the dial information element.
- ➔ **10-03-12 ETU Setup – Short/Long-Haul (For PRI PKG)**
If there is a CSU/DSU between the Telco and the DTI-U40, set to 0:Short. If there is no CSU/DSU between the Telco and the DTI-U40, set to 1:Long.
- ➔ **10-03-13 ETU Setup – Loss of Signal Detection Limit (For PRI PKG)**
If the transmit/receive voltage is less than the setting in PRG 10-03-13, the system considers this as Loss-Of-Signal and the PRI does not come up. Note that there are different values based on the setting in PRG 10-03-12 for the PRI.
- ➔ **10-06-01 ISDN BRI Setup – TEI Selection**
Select the method the system uses when assigning Terminal Endpoint Identifier (TEI) values to the BRI Circuit.

0 = Select by SPID number (Default)
1 = Select by Channel ID number

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-
- ➔ **10-06-02 ISDN BRI Setup – DID Mode**
Select the method the system uses when assigning DID Mode to the BRI Circuit.

0 = Route by Called Party Number (default).
1 = Route by Redirecting Number
 - ➔ **10-06-03 ISDN BRI Setup – SPID 1**
Assign the SPID Number for B-Channel 1 (maximum of 20 Digits allowed).
 - ➔ **10-06-04 ISDN BRI Setup – SPID 2**
Assign the SPID Number for B-Channel 2 (maximum of 20 Digits allowed).
 - ➔ **21-12-01 ISDN Calling Party Number Setup for Trunks – Calling Party Number Data**
Assign Calling Party Numbers for each trunk (maximum 16 digits per entry).

 *After the above programming is complete a reset of the BRT(4)-U20 ETU is required.*

ISDN – PRI Installation

- ➔ **10-03-01 ETU Setup – For BRI PKG**
Set up and confirm the Basic Configuration data for each DTI ()-U40 ETU. Use this program to select the ISDN Line Mode.

0 = Not Set
1 = T-Bus (Default)
- ➔ **10-03-03 ETU Setup – Connection Type (For BRI PKG)**
Set up and confirm the Basic Configuration data for each DTI()-U40 ETU. This program determines whether or not the CRC Multi-frame (CRC4) is used.

0 = Off (Default)
1 = On
- ➔ **10-03-04 ETU Setup (for PRI PKG Setup) – Layer 3 Timer Type**
Set up and confirm the Basic Configuration data for each DTI-U40 ETU. This program selects the Layer 3 timer type (1~5). Each timer value of Layer 3 is set up for each type in Program 81-06 (T-Bus).
- ➔ **10-03-06 ETU Setup (for PRI PKG Setup) – Length of Cable**
Set up and confirm the Basic Configuration data for each DTI-U40 ETU. Select the length of cable to be used.

0 = 0 ~ 40m (Default)
1 = 40 ~ 81m
2 = 81 ~ 122m
3 = 122 ~ 162m
4 = 162 ~ 200m

- ➔ **10-03-08 ETU Setup (for PRI PKG Setup) – Dial Sending Mode**
Set up and confirm the Basic Configuration data for each DTI-U40 ETU.
0 = Enblock Sending
1 = Overlap Sending (Default)
- ➔ **10-03-09 ETU Setup (for PRI PKG Setup) – Dial Information Element**
Set up and confirm the Basic Configuration data for each DTI-U40 ETU. If Overlap Sending is selected in Program 10-03-08, select either Keypad Facility (0) or Called Party Number (1) for the dial information element.
- ➔ **10-03-18 ETU Setup (for PRI PKG Setup) – Type of Number**
Select the number type for the ISDN circuit.
- ➔ **10-03-19 ETU Setup (for PRI PKG Setup) – Numbering Plan Identification**
Select the Numbering Plan used for the ISDN circuit.
- ➔ **10-03-20 ETU Setup (for PRI PKG Setup) – Network Exchange Selection**
Select the ISDN protocol for the ISDN circuit.
- ➔ **10-03-21 ETU Setup (for PRI PKG Setup) – PRI Number of Ports**
Select the number of ports for the PRI.
- ➔ **10-39-01 Fractional Setup**
Enable (1)/Disable (0) the T1/PRI fractional function.
- ➔ **21-12-01 ISDN Calling Party Number Setup for Trunks – Calling Party Number Data**
Assign Calling Party Numbers for each trunk (maximum 16 digits per entry).
✎ After the above programming is complete a reset of the BRT(4)-U20 ETU is required.
- ➔ **14-05-01 Trunk Group – Trunk Group Number**
Assign trunks to Trunk Groups, also assign the outbound priority for trunks in the group.
- ➔ **21-01-03 System Options for Outgoing Calls – Trunk Interdigit Time (External)**
The system waits for this time to expire before placing the call in a talk state (default: 5 seconds).

DID Services for either ISDN – BRI or PRI

- ➔ **22-02-01 Incoming Call Trunk Setup – Incoming Type**
Assign the incoming trunk type for each trunk.

0 = Normal (Default)	4 = DIL
1 = VRS (Second dial tone if no VRS installed)	5 = E&M Tie Line
2 = DISA	6 = Delayed DID
3 = DID	7 = ANI/DNIS
- ➔ **22-11-01 DID Translation Number Conversion – Received Number**
For each DID Translation Table entry (1~2000), specify the digits received by the system (eight maximum).

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- ➔ **22-11-02 Translation Number Conversion – Target Number**
For each DID Translation Table entry (1~2000), specify the extension the system dials after translation (24 digits maximum).
 - ➔ **22-09-01 DID Basic Data Setup – Expected Number of Digits**
For each DID Translation Table (1~20), enter the number of digits the table expects to receive from the CO (eight maximum). For example, for a table used with 3-digit DID service, enter 3.
✎ For additional DID Services refer to [Direct Inward Dialing \(DID\)](#) on page 2-321.

Calling Party Number Presentation for either ISDN – BRI or PRI

Calling Line Identification Presentation (CLIP)

- ➔ **10-03-05 ETU Setup (for PRI PKG Setup) – CLIP Information Announcement**
Based on this setting, the system includes a Presentation Allowed (1) or Presentation Restricted (0) in the Setup message to allow or deny the Calling Party Number. Program 15-01-04 must also be set to 1 if this option is enabled.
- ➔ **21-12-01 ISDN Calling Party Number Setup for Trunks**
Assign Calling Party Numbers for each trunk (maximum 16 digits per entry). When a call is made by an extension which does not have an Extension Calling Number assigned (Program 21-13), the system sends the calling number for the ISDN trunk defined in 21-12. If the Calling Party Number is assigned in both Program 21-12 and Program 21-13, the system sends the data in Program 21-13.

Calling Party Number Presentation from Station

- ➔ **10-03-05 ETU Setup (for PRI PKG Setup) – CLIP Information Announcement**
Based on this setting, the system includes a Presentation Allowed (1) or Presentation Restricted (0) in the Setup message to allow/deny the Calling Party Number. Program 15-01-04 must also be set to 1 if this option is enabled.
- ➔ **15-01-04 Basic Extension Data Setup – ISDN Caller ID**
If both PRG 15-01-04 and PRG 10-03-05 are enabled (1), the system includes Caller ID in the Setup message as Presentation Allowed. If these options are disabled (0), it is Presentation Restricted.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-08-13 Class of Service Options (Outgoing Call Service) – ISDN CLIP**
Determine if the ISDN calling line identity presentation and screening indicators are allowed:

0 = Disabled (Default)
1 = Enabled

- **21-13-01 ISDN Calling Party Number Setup for Extensions**
Assign each extension a Calling Party Number (maximum 16 digits per entry). The calling number is the subscriber number of the dial-in number. When a call is made by an extension which does not have an Extension Calling Number assigned (Program 21-12), the system sends the calling number for the ISDN trunk defined in Program 21-13. If a Calling Party Number is assigned in both PRG 21-12 and PRG 21-13, the system sends the data in PRG 21-13.

ISDN – PRI Network Specific Assignment

- **26-02-07 Dial Analysis Table for ARS/LCR – Network Specified Parameter Table**
Define the network specified parameter table for each ARS Table

Dial Analysis Table No.: 1 ~ 400
Table No.: 0 ~ 16 (0: no setting)
- **26-12-01 Network Specified Parameter Table for ARS – Type of Number**
Define the type of Number parameter for an ISDN outgoing call.

Table No.: 1 ~ 15
Type: 0 (System Default)
1 (Unknown)
2 (International Number)
3 (National Number)
4 (Network Specific Number)
5 (Subscriber Number)
6 (Abbreviated Number)

(default: 0 (All Tables))
- **26-12-02 Network Specified Parameter Table for ARS – Numbering Plan Identification**
Define the Numbering Plan Identification Parameter for an ISDN outgoing call

Table No.: 1 ~ 15
Type: 0 (System Default)
1 (Unknown)
2 (ISDN Numbering Plan)
3 (Data Numbering Plan)
4 (Telex Numbering Plan)
5 (National Standard Numbering Plan)
6 (Private Numbering Plan)

(default: 0 (All Tables))
- **44-05-11 ARS/F-Route Table – Network Specified Parameter Table**
Define the network specified parameter table for each F-Route table.


F-Route Table No.: 1 ~ 500
Priority No.: 1 ~ 4
Table No.: 0 ~ 15 (0: no setting)

(default: 0 All Tables, All Orders)

SMDR Dialed Digits for either ISDN – BRI or PRI

- ➔ **35-02-15 SMDR Output Options – CLI/DID Number Switching**
Determine if the CLI/DID Number should be displayed.

0 = Caller ID Number (Default)
1 = DID Calling Number
2 = Caller ID Name
- ➔ **35-02-16 SMDR Output Options – Trunk Name or Received Dialed Number**
Determine how the SMDR should print incoming calls on ANI/DNIS or DID trunks. If set to (1), ANI/DNIS trunks can print DNIS digits, if set to (0) trunk names are printed instead.

 For additional SMDR Services refer to [Station Message Detail Recording on page 2-913](#).

General ISDN Programs

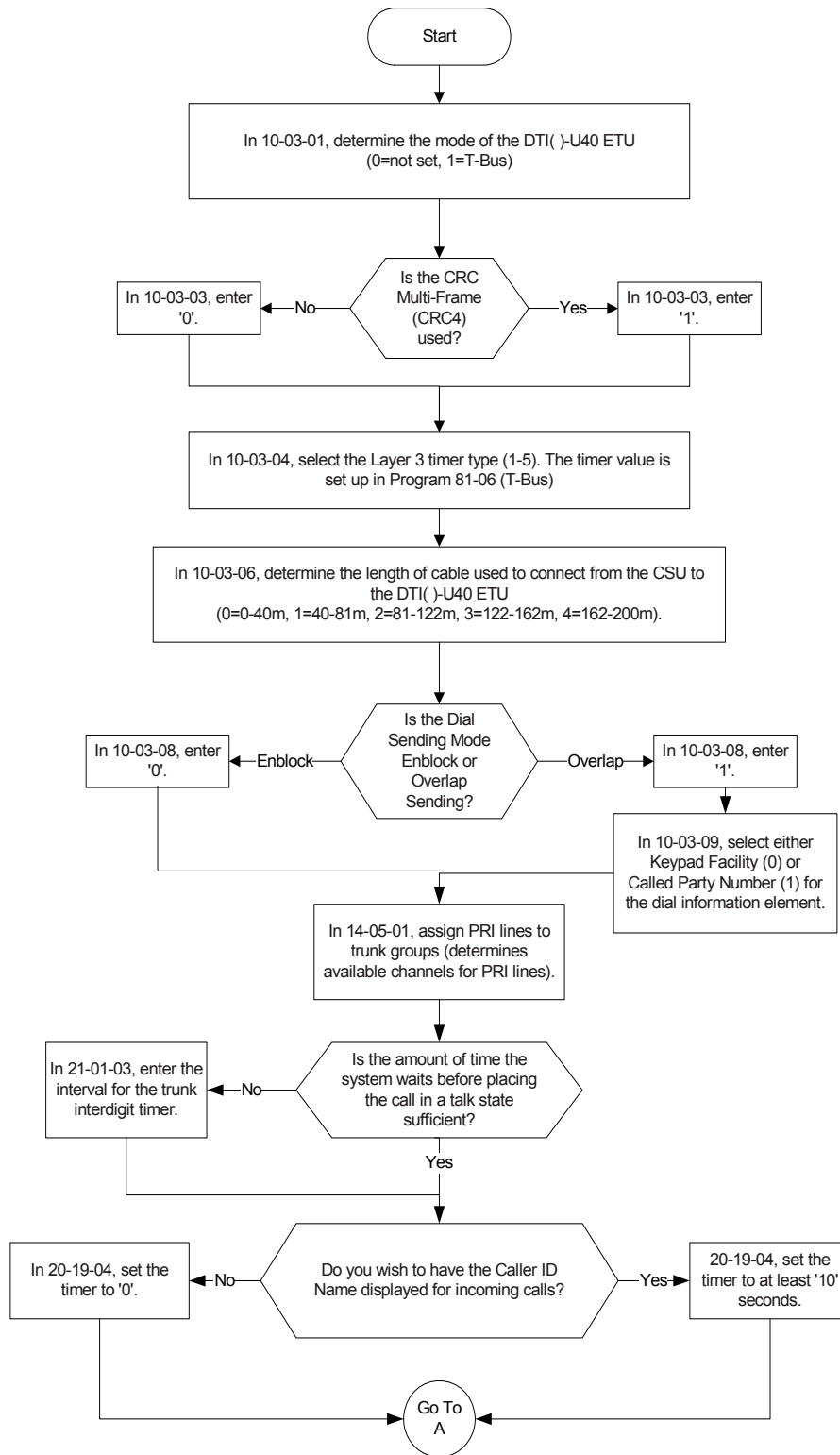
- ➔ **14-01-13 Basic Trunk Data Setup – Trunk-to-Trunk Transfer**
For each trunk that is able to participate in a tandem call, enter 1. To disable a trunk from Tandem Trunking, enter 0 (default setting: 1). Required for 2 B-Channel transfer.
- ➔ **15-02-29 Multiline Telephone Basic Data Setup – PB Back Tone Level**
Adjust the PB Back Tone level when calling an ISDN line. [Entries: 1~63 (-15.5 to +15.5 db in 0.5 dB steps), default: 32].
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Caller ID Block for ISDN (63) if required.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service to extensions.
- ➔ **20-09-02 Class of Service Options (Incoming Call Service) – Caller ID Display**
Enable (1)/Disable (0) the extension ability to display incoming Caller ID name/number information.
- ➔ **20-09-03 Class of Service Options (Incoming Call Service) – Sub Address Identification**
Define whether an extension displays the Caller Sub-Address, 0 = Deny, 1 = Allow.
- ➔ **20-11-11 Class of Service Options (Hold/Transfer Service) – Auto On-Hook Transfer**
Enable (1)/Disable (0) an extension user ability to transfer when the user hangs up.
- ➔ **20-11-14 Class of Service Options (Hold/Transfer Service) – Trunk-to-Trunk Transfer Restriction**
Allow (0)/Deny (1) Trunk-to-Trunk Transfer (default: 1). Required for 2 B-Channel transfer.
- ➔ **20-11-21 Class of Service Options (Hold/Transfer Service) – Restriction for Tandem Trunking on Hang Up**
Allow (0) an extension user to set up a tandem call automatically when they hang up.

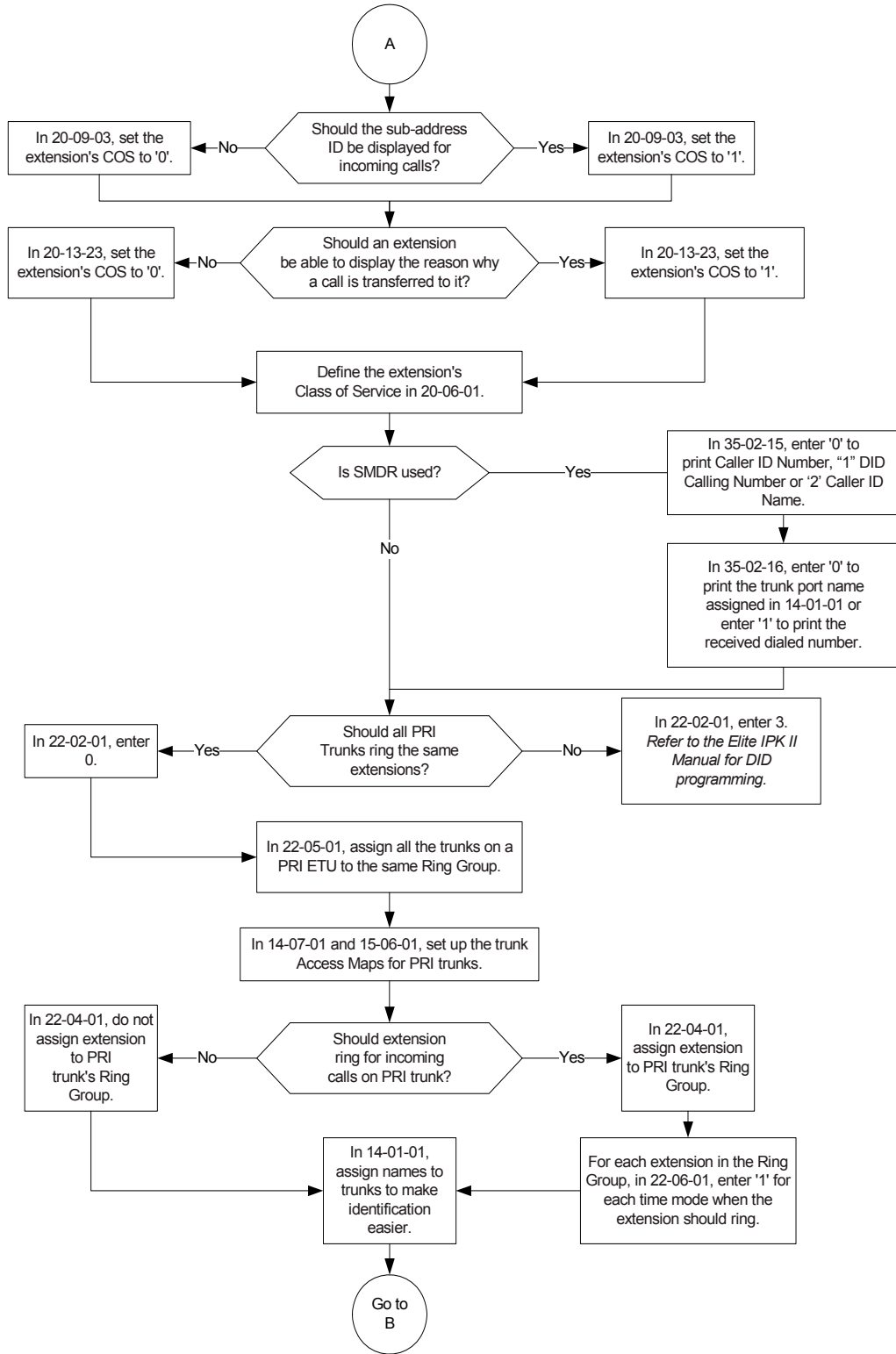
- ➔ **20-13-23 Class of Service Options (Supplementary Service) – Display the Reason for Transfer**
Enable (1)/Disable (0) an extension ability to display the reason (Call Forward, Busy, No Answer or DND) a VRS, DID, DISA, or ISDN call is being transferred to their extension.
- ➔ **20-19-04 System Options for Caller ID – Wait Facility IE Timer**
An ISDN trunk uses this time to determine how long the system waits for the Caller ID name from the Telco, (default: 10 seconds).
- ➔ **20-25-14 ISDN Options – No response Release Send**
Determine whether or not a release message is sent when the second T303 timer expires.
Entries: Normal/Off (0)/Send Release Message (1).
(default: 0).

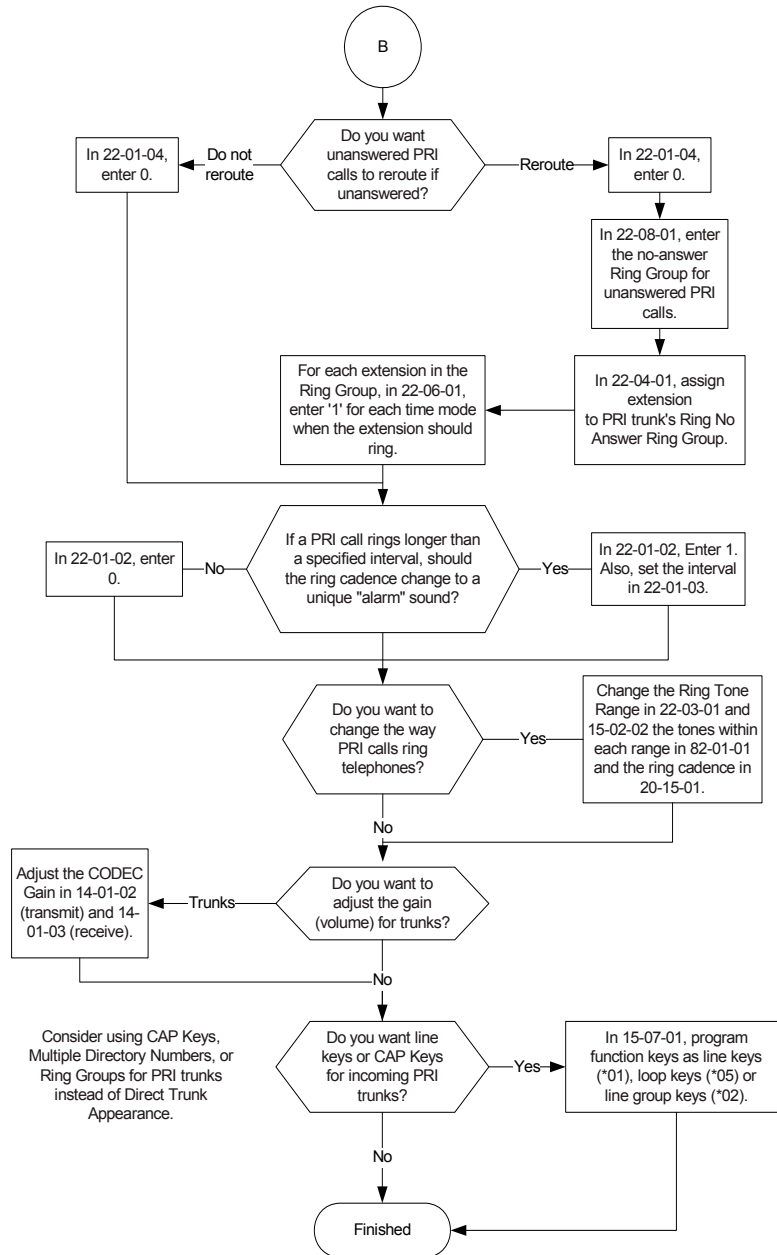
Operation

None

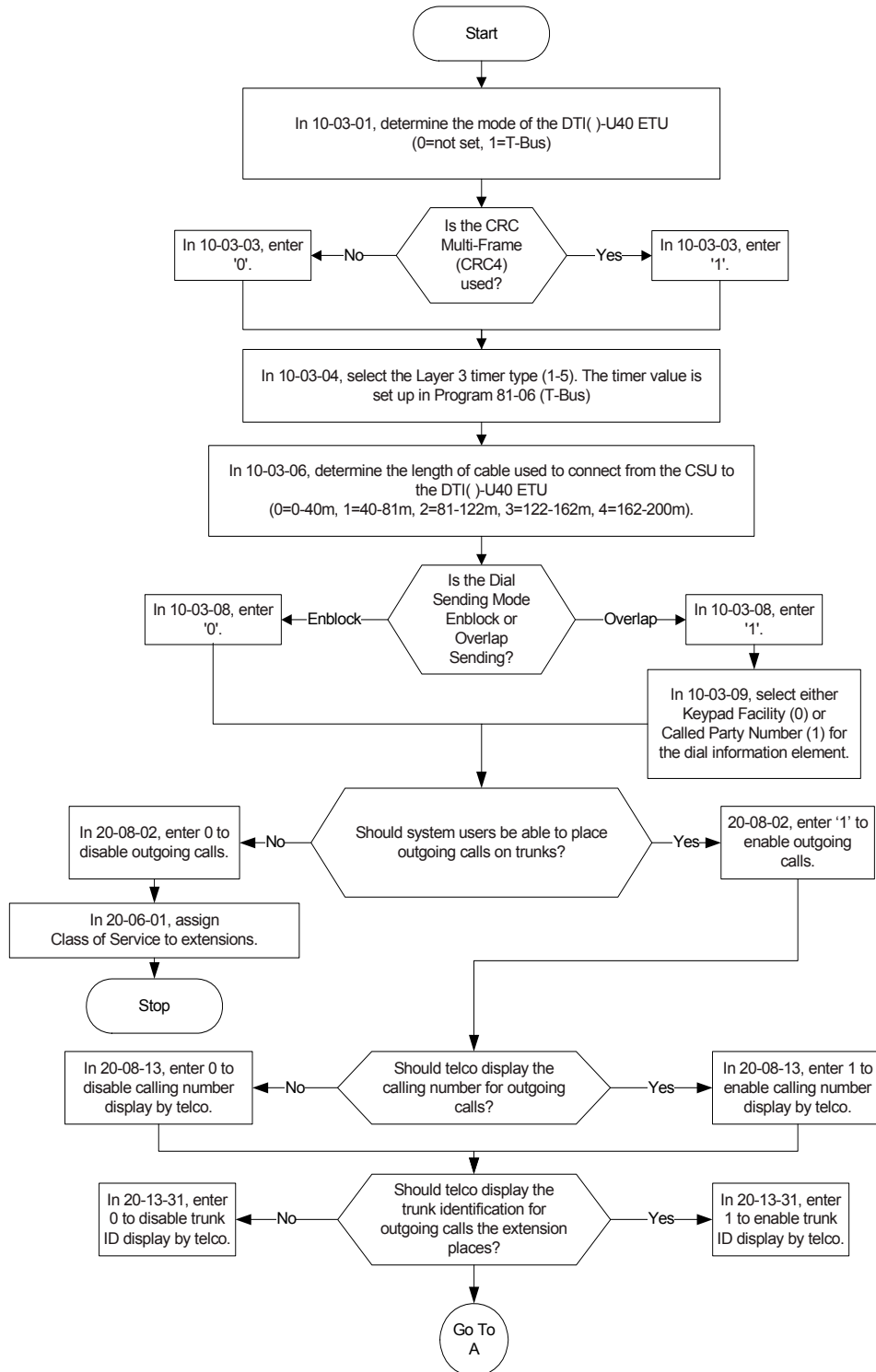
Programming Flowchart for ISDN-PRI – Answering Calls

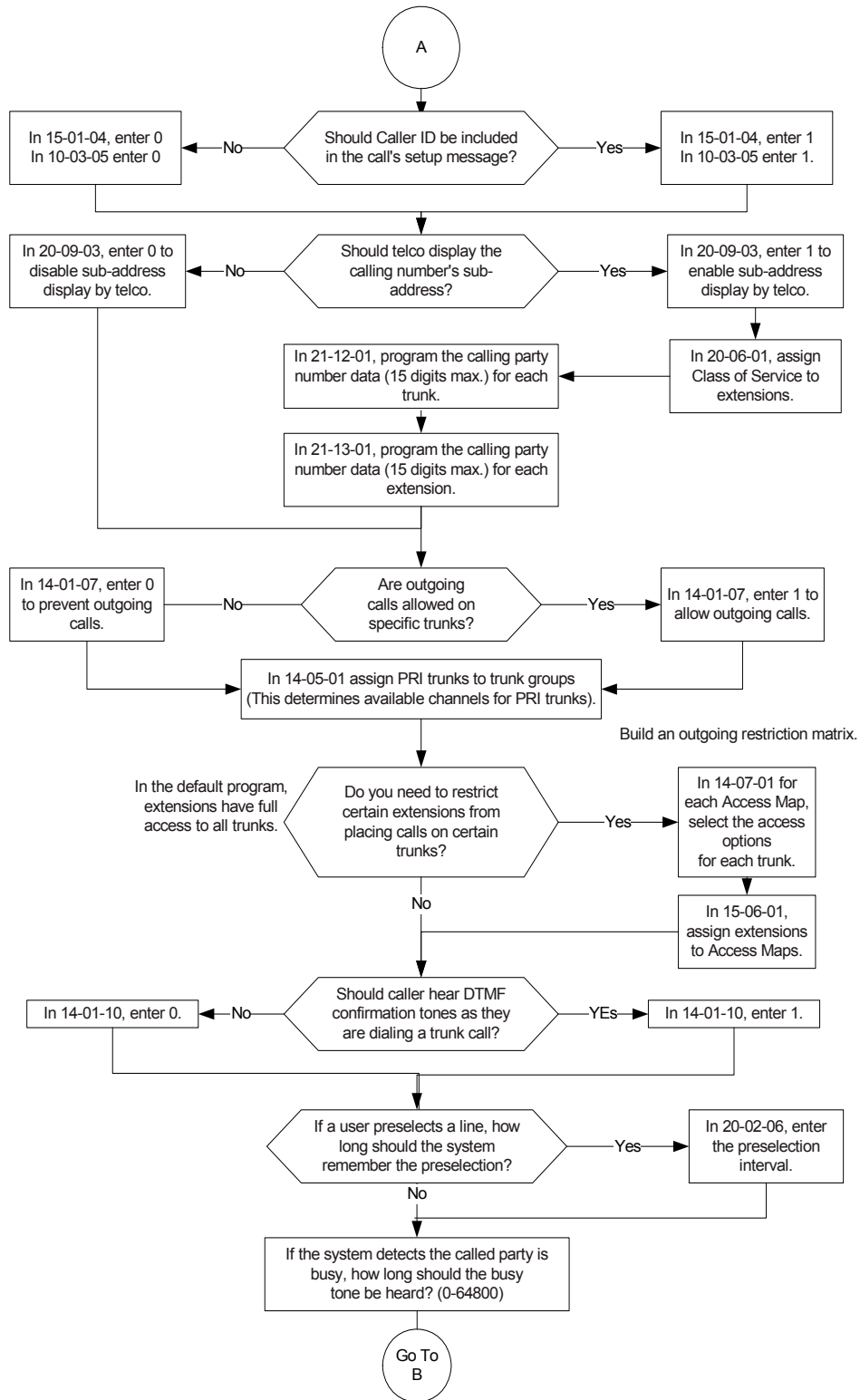


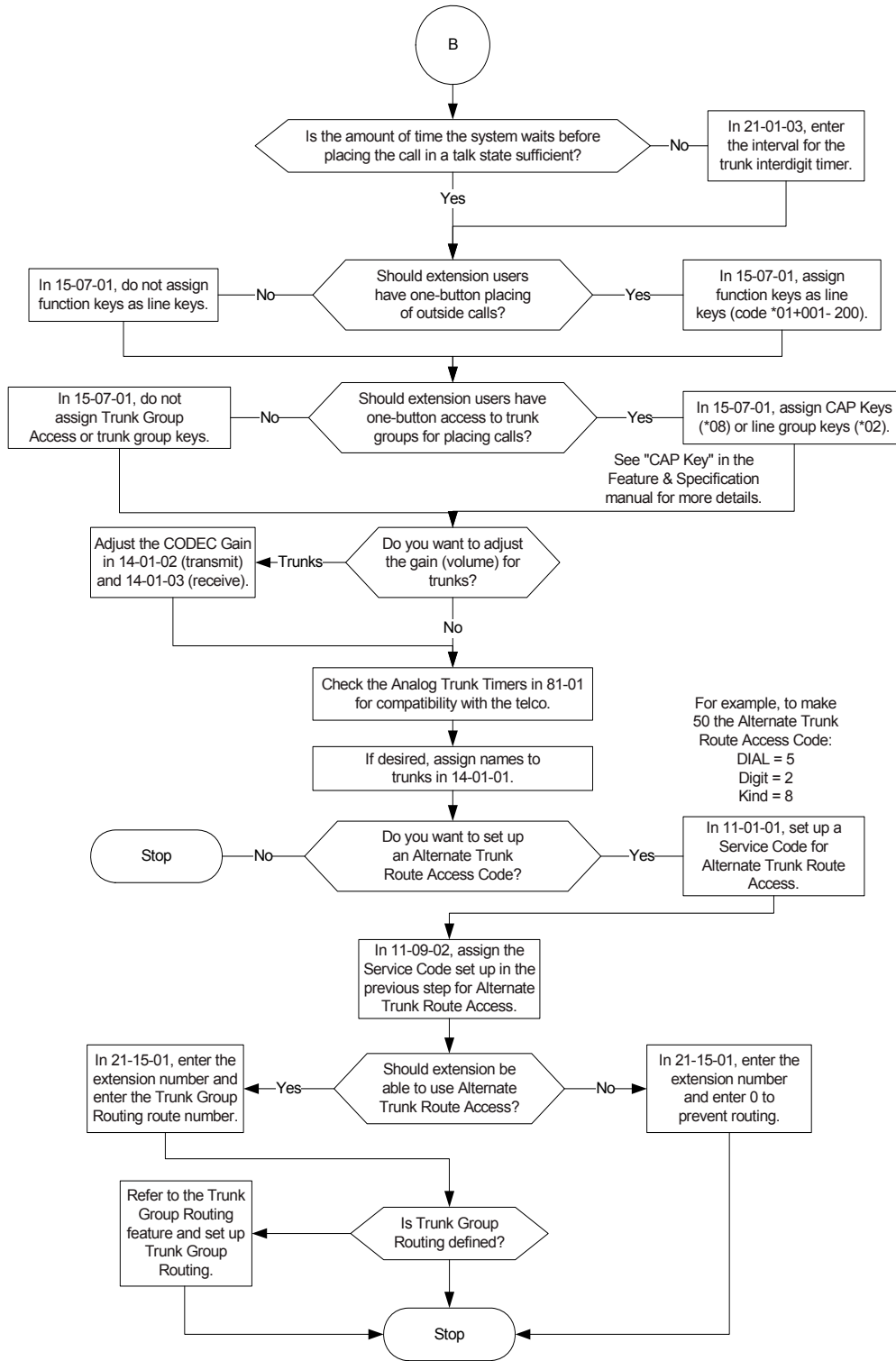




Programming Flowchart of ISDN-PRI – Placing Calls







K-CCIS – IP with IAD

Enhancements

This feature was added with **Version 1100**.

Version 1600 or higher:

- A remote K-CCIS user can transfer a call directly to an ACD Pilot number. Prior to this release, this type of transfer was not allowed.

Version 2000 or higher:

- Supports station numbers to be assigned by the 10s group for 4-digit station numbers, 100s group for 5-digit, 1,000s group for 6-digit station numbers, and 10,000s group for 7-digit station numbers.



Description

This feature provides the benefits and additional feature compatibility of Key-Common Channel Interoffice Signaling (K-CCIS) between multiple systems including NEAX PBX systems connected together over a Data Communication IP Network (Intranet). Voice signals and common signaling data from/to the distant offices are converted into IP Packets and transmitted through the Data IP Network. When using the IP (K-CCIS) to NEAX (Point-to-Multipoint) feature, both voice and data communication lines are integrated into one network and communication costs can be reduced.

This feature is available between Electra Elite IPK II systems and NEAX PBX systems. When connecting to a NEAX system, IPT cards are used and must be installed.

The following K-CCIS features are available with the IP (K-CCIS) to NEAX (Point-to-Multipoint) feature:

- Call Forwarding – All Calls – K-CCIS
- Call Forwarding – Busy/No Answer – K-CCIS
- Call Transfer – All Calls – K-CCIS
- Calling Name Display – K-CCIS
- Calling Number Display – K-CCIS
- Centralized Billing – K-CCIS
- Centralized BLF (K-CCIS)*

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-
- Centralized Day/Night Mode Change (K-CCIS)
 - Dial Access to Attendant (K-CCIS)
 - Direct Inward Dialing – K-CCIS
 - Dual Hold – K-CCIS
 - Elapsed Time Display – K-CCIS
 - Flexible Numbering of Stations – K-CCIS
 - Hands-Free Answerback – K-CCIS
 - Hot Line – K-CCIS
 - Link Reconnect – K-CCIS
 - Multiple Call Forwarding – All Calls – K-CCIS
 - Multiple Call Forwarding – Busy/No Answer – K-CCIS
 - Paging Access – K-CCIS
 - Station-to-Station Calling – K-CCIS
 - Uniform Numbering Plan – K-CCIS
 - Voice Call – K-CCIS
 - Voice Mail Integration – K-CCIS **

* Not supported with NEAX PBXs.

** Not supported with In-Mail.

Conditions

- For this feature, the IAD(8)-U() ETU is installed and assigned as a CCISoIP ETU. Each IAD(8)-U10 ETU reduces the maximum capacity of DTI-U() ETUs and trunks in the system.
- For a Basic Port Package, a maximum of six IAD(8)-U() ETUs can be installed. If the IAD(8)-U() ETU is assigned as a CCISoIP ETU, a maximum of six IAD(8)-U() ETUs can be installed.
- For an Expanded Port Package, a maximum of 22 IAD(8)-U() ETUs can be installed. If the IAD(8)-U() ETUs are assigned as CCISoIP ETUs, a maximum of 22 IAD(8)-U() ETUs can be installed.
- The IAD(8)-U10 supports only those codecs that are approved to provide toll-quality speech paths. The following voice compression methods are supported for the IP CCH for NEAX application:

- G.711 μ -Law – highest bandwidth
- G.729 – mid-range bandwidth
- G.723 – lowest bandwidth
- Each voice call will require at least the bandwidth listed in the following table:

Codec	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.711 μ -Law	90 Kbps	90Kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G.729	34 Kbps	34 Kbps	20 ms	15.0 ms	2 datagrams (40 ms)	4.07
G.723	25 Kbps	25 Kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.87

Note 1: This includes the overheads of VoIP communication, including signaling.

Note 2: In voice communications, particularly Internet telephony, the mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. The maximum for the Mean Opinion Score is 5.0.

- Data calls (Modem Data) across the VoIP connection are only supported when G.711 μ Law Codec is used.
- The IAD(8)-U() ETU contains a regular TCP/RTP/IP stack that can handle real time media. The IAD card, from the network administration perspective, is an end-point on the IP network.
- The IAD(8)-U() ETU with IP CCH for NEAXI application loaded uses Common Channel Interoffice Signaling over IP networks. Currently, this protocol does not allow communications across networks where Network Address Translation (NAT) is performed.
- The audio quality of speech connections depend greatly on the available bandwidth between the IAD(8)-U() ETUs in the data network. As the internet is an uncontrolled data network compared to an Intranet, it is highly recommended to use this application in Intranet WAN environment, with known (or controlled and assured) Quality of Service (QoS).
- If any network where the IAD(8)-U() ETU is connected uses NAT for connecting the voice calls (including firewall), It is recommended that you consult your network provider/ administrator and specifically request service for VoIP or real-time media support on your networks.
- A static IP Address is required for each IAD(8)-U() ETU in the network.

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- This feature allows Point-to-Multipoint, peer-to-peer connections for calls through the IP K-CCIS Network. If a call is transferred or forwarded to a different system in the Network, the trunks in the transferring system are released and a new Point-to-Point connection is established.
 - When a call terminates back to the originating system due to call forwarding or transfer, the intermediate trunks are not released until the call is completed by an answer at the called party or until the call is forwarded to a different system (similar to the link-reconnect function in TDM CCIS).
 - The CCH(4)-U10 ETU is not required to support this feature. It can be installed and used in a system using traditional K-CCIS with point-to-point T1 lines allowing both IP (K-CCIS) to NEAX (Point-to-Multipoint) and traditional K-CCIS to be used with the same system.
 - The LAN connection is provided by a 10 Base-T/100 Base-TX, Auto sensing, full duplex Ethernet.
 - The Centralized E911 (K-CCIS) feature is not supported with IP (K-CCIS) to NEAX (Point-to-Multipoint). Each Electra Elite IPK system in the network must have at least one local trunk for Emergency 911 calls.
 - The Electra Elite IPK can send billing information to a billing center office (NEAX2000/2400), but cannot receive the billing information as the billing center office.
 - Voice Calls (K-CCIS), voice announce is not supported for a forwarded call across IP (K-CCIS) to NEAX (Point-to-Multipoint).
 - This feature is not available when using 6- or 7-digit station numbers.
 - This feature and the IP (K-CCIS) feature are not supported in the same network.
 - Not all data networks can support Voice over Internet Protocol (VoIP). A good VoIP solution requires a low-latency, low jitter, and low packet loss network. Accordingly a network must be evaluated for latency, packet loss, and jitter to qualify and determine if it can provide toll-quality speech paths.
 - With **Version 1600** or higher, a remote K-CCIS user can call, or transfer a call directly to an ACD Pilot number. However, an incoming K-CCIS call to the ACD Pilot number does NOT provide a Link Reconnect.
 - When the system searches the Dial Extension Analyze Table (PRG 11-20-01), the system uses prefix searching, giving the lower table number the higher priority. For example, the user programs 211 in table 1 and 2113 in table 2, then dials 2113, the system selects table 1.
 - Program 11-01-01 9: System Numbering – Dial Extension Analyze is not supported in **Version 1600**.
 - Program 11-20-01: Dial Extension Analyze Table – Dial and Program 11-20-02: Dial Extension Analyze Table – Type of Dials are not supported in **Version 1600**.

Default Setting

None

System Availability

Terminals

All Stations

Required Software

Version 1100 or higher

Required Component(s)

IAD(8)-U10 ETU with IP CCH for NEAXPI application **1.00 or higher** loaded

Related Features

IP Station (MEGACO) – IAD (Integrated Access Device)

Voice Over

Programming

Refer to the Electra Elite IPK II Key-Common Channel Interface Signaling (K-CCIS) Manual.

- ➔ **50-03-04 CCIS Destination System Settings – Point Code Availability (Version 1600 and 2100 or higher required)**
 - 0:Disable
 - 1:Enable
 - (default: 1)

Operation

Refer to the Electra Elite IPK II Key-Common Channel Interface Signaling (K-CCIS) Manual.

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K-CCIS – IP with PVA

Enhancements

This feature was added with **Version 1100**.

Version 1500 or higher:

- A 4-port license registration allows users to add CCISoIP Ports in increments of four (4, 8, 12, 16, 20 or 24).
- The CCISoIP web server has a new HTTP page to display the network statistics based on values entered in PRG 84-06.

Version 1600 or higher:

- A remote K-CCIS user can transfer a call directly to an ACD Pilot number. Prior to this release, this type of transfer was not allowed.

Version 2000 or higher:

- Supports station numbers to be assigned by the 10s group for 4-digit station numbers, 100s group for 5-digit, 1,000s group for 6-digit station numbers, and 10,000s for 7-digit station numbers.
- Supports the Combined functionality of a MG16 and K-CCIS over IP and SIP Stations/trunks on one card.

Description

This feature provides the benefits and additional feature compatibility of Key-Common Channel Interoffice Signaling (K-CCIS) between multiple systems including NEAX PBX systems connected together over a Data Communication IP Network (Intranet). Voice signals and common signaling data from/to the distant offices are converted into IP Packets and transmitted through the Data IP Network. When using the IP (K-CCIS) to NEAX (Point-to-Multipoint) feature, both voice and data communication lines are integrated into one network and communication costs can be reduced.

This feature is available between Electra Elite IPK II systems and NEAX PBX systems. When connecting to a NEAX system, IPT cards are used and must be installed.

The following K-CCIS features are available with the IP (K-CCIS) to NEAX (Point-to-Multipoint) feature:

- Call Forwarding – All Calls – K-CCIS
- Call Forwarding – Busy/No Answer – K-CCIS
- Call Transfer – All Calls – K-CCIS
- Calling Name Display – K-CCIS
- Calling Number Display – K-CCIS

- Centralized Billing – K-CCIS
- Centralized BLF (K-CCIS) *
- Centralized Day/Night Mode Change (K-CCIS)
- Dial Access to Attendant (K-CCIS)
- Direct Inward Dialing – K-CCIS
- Dual Hold – K-CCIS
- Elapsed Time Display – K-CCIS
- Flexible Numbering of Stations – K-CCIS
- Hands-Free Answerback – K-CCIS
- Hot Line – K-CCIS
- Link Reconnect – K-CCIS
- Multiple Call Forwarding – All Calls – K-CCIS
- Multiple Call Forwarding – Busy/No Answer – K-CCIS
- Paging Access – K-CCIS
- Station-to-Station Calling – K-CCIS
- Uniform Numbering Plan – K-CCIS
- Voice Call – K-CCIS
- Voice Mail Integration – K-CCIS **

* Not supported with NEAX PBXs.

** Not supported with In-Mail.

Conditions

- For this feature, the PVA(X)-U() ETU is installed and assigned as a CCISoIP ETU. Each PVA(X)-U() ETU reduces the maximum capacity of DTI-U() ETUs and trunks in the system.
- For a Basic Port Package, a maximum of two PVA(X)-U() ETUs can be installed. If the PVA(X)-U() ETU is assigned as a CCISoIP ETU, a maximum of two PVA(X)-U() ETUs can be installed.
- For an Expanded Port Package, a maximum of 11 PVA(X)-U() ETUs can be installed. If the PVA(X)-U() ETUs are assigned as CCISoIP ETUs, a maximum of 11 PVA(X)-U() ETUs can be installed.

- The PVA(X)-U() ETU supports only those codecs that are approved to provide toll-quality speech paths. The following voice compression methods are supported for the IP CCH for NEAX application:
 - G.711 μ -Law – highest bandwidth
 - G.729 – mid-range bandwidth
 - G.723 – lowest bandwidth
- Each voice call will require at least the bandwidth listed in the following table:

Codec	Transmit Data Rate	Receive Data Rate	Time Between Packets	Packetization Delay	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.711 μ -Law	90 Kbps	90Kbps	20 ms	1.5 ms	2 datagrams (40 ms)	4.4
G.729	34 Kbps	34 Kbps	20 ms	15.0 ms	2 datagrams (40 ms)	4.07
G.723	25 Kbps	25 Kbps	30 ms	37.5 ms	2 datagrams (60 ms)	3.87

Note 1: This includes the overheads of VoIP communication, including signaling.

Note 2: In voice communications, particularly Internet telephony, the mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. The scheme uses subjective tests (opinionated scores) that are mathematically averaged to obtain a quantitative indicator of the system performance. The maximum for the Mean Opinion Score is 5.0.

- Data calls (Modem Data) across the VoIP connection are only supported when G.711 μ Law Codec is used.
- The PVA(X)-U() ETU contains a regular TCP/RTP/IP stack that can handle real time media. The PVA(X)-U() ETU, from the network administration perspective, is an end-point on the IP network.
- The PVA(X)-U() ETU with IP CCH for NEAX application loaded uses Common Channel Interoffice Signaling over IP networks. Currently, this protocol does not allow communications across networks where Network Address Translation (NAT) is performed.
- The audio quality of speech connections depend greatly on the available bandwidth between the PVA(X)-U() ETUs in the data network. As the internet is an uncontrolled data network compared to an Intranet, you should use this application in Intranet WAN environment, with known (or controlled and assured) Quality of Service (QoS).
- If any network where the PVA(X)-U() ETU is connected uses NAT for connecting the voice calls (including firewall), you should consult your network provider/administrator and specifically request service for VoIP or real-time media support on your networks.
- A static IP Address is required for each PVA(X)-U() ETU in the network.

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- This feature allows Point-to-Multipoint, peer-to-peer connections for calls through the IP K-CCIS Network. If a call is transferred or forwarded to a different system in the Network, the trunks in the transferring system are released and a new Point-to-Point connection is established.
 - When a call terminates back to the originating system due to call forwarding or transfer, the intermediate trunks are not released until the call is completed by an answer at the called party or until the call is forwarded to a different system (similar to the link-reconnect function in TDM CCIS).
 - The CCH(4)-U10 ETU is not required to support this feature. It can be installed and used in a system using traditional K-CCIS with point-to-point T1 lines allowing both IP (K-CCIS) to NEAX (Point-to-Multipoint) and traditional K-CCIS to be used with the same system.
 - The LAN connection is provided by a 10 Base-T/100 Base-TX, Auto sensing, full duplex Ethernet.
 - The Centralized E911 (K-CCIS) feature is not supported with IP (K-CCIS) to NEAX (Point-to-Multipoint). Each Electra Elite IPK system in the network must have at least one local trunk for Emergency 911 calls.
 - The Electra Elite IPK can send billing information to a billing center office (NEAX2000/2400), but cannot receive the billing information as the billing center office.
 - Voice Calls (K-CCIS), voice announce is not supported for a forwarded call across IP (K-CCIS) to NEAX (Point-to-Multipoint).
 - This feature is not available when using 6- or 7-digit station numbers.
 - This feature and the IP (K-CCIS) feature are not supported in the same network.
 - Not all data networks can support Voice over Internet Protocol (VoIP). A good VoIP solution requires a low-latency, low jitter, and low packet loss network. Accordingly a network must be evaluated for latency, packet loss, and jitter to qualify and determine if it is able to provide toll-quality speech paths.
 - With **Version 1600 or higher**, a remote K-CCIS user can call, or transfer a call directly to an ACD Pilot number. However, an incoming K-CCIS call to the ACD Pilot number does NOT provide a Link Reconnect.
 - When the system searches the Dial Extension Analyze Table (PRG 11-20-01), the system uses prefix searching, giving the lower table number the higher priority. For example, the user programs 211 in table 1 and 2113 in table 2, then dials 2113, the system selects table 1.
 - FoIP (Fax over Internet Protocol) with T.38 protocol is only supported between an IPK to IPK with V1.02 PVA IP CCHNEAX Package and with IPK II V1.00 PVA Combo Package.
 - Program 11-01-01 9: System Numbering – Dial Extension Analyze is not supported in **Version 1600**.
 - Program 11-20-01: Dial Extension Analyze Table – Dial and Program 11-20-02: Dial Extension Analyze Table – Type of Dials are not supported in **Version 1600**.

PVA Combo Card Service Conditions

- **Version 2000 or higher** is required to support the PVA Combo Card Application.
- If changing packages on the PVA (for example K-CCIS to Combo), the previous card type must be deleted from the slot in Program 90-05. Even if both are the same physical card, PVA, the system sees them differently.
- The PVA Combo Card (MG_CCIS ETU) Application can support 24 DSP resources which may be configured to support the MG16 or the CCISoIP application.
- The PVA Combo Card CCISoIP Trunks and SIP Trunks Logical Port Numbers are determined by the following:
 - ❑ **SIP Trunks**
 - PVA(X)-U() ETU License Registration
 - SIP Client License Registration
 - Elite IPK II Program 10-41-01 SIP Trunk Availability
 - Elite IPK II Program 10-40-02 Number of SIP Ports
 - ❑ **CCISoIP Trunks**
 - PVA(X)-U() ETU License Registration
 - Elite IPK II Program 10-40-03 CCISoIP Trunk Availability
 - Elite IPK II Program 10-40-04 Number of CCISoIP Ports
 - The maximum number of Logical Trunk Ports is 48 ports per PVA Combo Card package
- The PVA(X)-U() ETU with PVA Combo Card Application may configure DSP usage criteria with Elite IPK II Program 10-19-01 DSP Resources. Each VoIP ETU can flag individual DSP resources as:
 - ❑ IP Ext – IP Extensions (includes MEGACO Station or SIP Stations)
 - ❑ SIP Trk – SIP Trunks
 - ❑ CCIS – CCIS Networking
 - ❑ Common – Common usage for CCIS Networking, SIP Station, MEGACO Stations, SIP Trunks
- PVA Combo Card CCISoIP Application supports trunks configured in increments of four contiguous DSP resources.
- ETU Configuration – The PVA Combo Card Application package is identified as a MG_CCIS ETU.
- The PVA Combo Card – VoIP ETU supports up to 200 CCIS Logical Trunks and 200 SIP Logical Trunk Port Numbers (read only).
- IP Application DSP Resource configuration:
 - ❑ CCIS Trunks are defined as CCIS or Common
 - ❑ IP MEGACO Stations are defined as IP Ext or Common

- SIP Trunks are defined as SIP Trk or Common
- SIP Station are defined as IP Ext or Common
- When the PVA Combo Card is installed in the Electra Elite IPK II system, the system assigns the first available Logical Port Numbers for CCISoIP trunks and SIP trunks are allotted the next available logical port numbers.
- If the Elite IPK II CPU system does not have registered SIP licenses, the Logical SIP Trunk is not assigned.
- If the PVA Combo Card cannot establish the common highway, the maximum number of trunk ports is 16 ports for one package.
- Changes in Program 10-40-03 and or Program 10-40-04 require the PVA(X)-U10 ETU to be reset for the new changes to take effect.
- Changes in Program 10-19-01 require the PVA(X)-U10 ETU to be reset for the changes to take effect.
- PVA(X)-U10 ETU/Combo Card LED Indications:
 - Non Peer-to-Peer MEGACO IP Station and SIP IP Station calls are indicated by a Flashing RED LED
 - CCISoIP Trunk and SIP Trunk calls are indicated by a Solid RED LED

Default Setting

None

System Availability

Terminals

All Stations

Required Component(s)

PVA(X)-U() ETU with PVA CCISoIP Package **1.00 or Higher**.

PVA(X)-U() IP Resource CD (stock number 750874) can be used for full 24-CCISoIP port licensing.

PVA(X)-U() 4-Port IP Resource CD (stock number 750895) can be used for 4-port license increments.

PVA(X)-U() ETU with PVA Combo Application Package **1.00 or Higher** and Elite IPK II CPUII **Version 2000 or Higher**.

Related Features

IP Station – (MEGACO) MG 16

Voice Over

Programming

Refer to the Electra Elite IPK II Key-Common Channel Interoffice Signaling (K-CCIS) Manual.

- ➔ **50-03-04 CCIS Destination System Settings – Point Code Availability (Version 1600 and 2100 or higher required)**
 - 0:Disable
 - 1:Enable
 - (default: 1)

Operation

Refer to the Electra Elite IPK II Key-Common Channel Interoffice Signaling (K-CCIS) Manual.

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K-CCIS – T1

Enhancements

Version 1600 or higher:

- A remote K-CCIS user can transfer a call directly to an ACD Pilot number. Prior to this release, this type of transfer was not allowed.

Version 2000 or higher:

- Supports station numbers to be assigned by the 10s group for 4-digit station numbers, 100s group for 5-digit, 1,000s group for 6-digit station numbers, and 10,000s for 7-digit station numbers.

Description

Key-Common Channel Interoffice Signaling (K-CCIS) allows multiple systems to be connected together to provide additional feature compatibility, above what normal Tie Lines provide. The system is configured with the 24-channel Digital Trunk Interface (DTI), a Common Channel Handler (CCH) for receiving or transmitting common signaling data from/to a distant office. The system can provide a variety of interoffice service features such as Calling Name display, Centralized Voice Mail Integration, or Link Reconnect.

The following features are provided:

- Call Forwarding – All Calls – K-CCIS
- Call Forwarding – Busy/No Answer – K-CCIS
- Call Park Retrieve – K-CCIS
- Call Transfer – All Calls – K-CCIS
- Calling Name Display – K-CCIS
- Calling Number Display – K-CCIS
- Calling Party Number (CPN) Presentation from Station – K-CCIS
- Centralized Billing – K-CCIS
- Centralized BLF (K-CCIS)
- Centralized Day/Night Mode Change – K-CCIS
- Centralized E911 (K-CCIS)
- Dial Access to Attendant – K-CCIS

- Direct Inward Dialing – K-CCIS
- Dual Hold – K-CCIS
- Elapsed Time Display – K-CCIS
- Flexible Numbering of Stations – K-CCIS
- Hands-Free Answerback – K-CCIS
- Hot Line – K-CCIS
- IP (K-CCIS)
- IP (K-CCIS) to NEAX (Point-to-Multipoint)
- Link Reconnect – K-CCIS
- Multiple Call Forwarding – All Calls – K-CCIS
- Multiple Call Forwarding – Busy/No Answer – K-CCIS
- Paging Access – K-CCIS
- Quick Transfer to Voice Mail – K-CCIS
- Station-to-Station Calling – K-CCIS
- Uniform Numbering Plan – K-CCIS
- Voice Call – K-CCIS
- Voice Mail Integration – K-CCIS *

* Not supported with In-Mail.

Conditions

- Each Electra Elite IPK II system can have up to four K-CCIS routes.
- One DTI-U() ETU can be used to support/connect a maximum of four K-CCIS Links.
- The Basic Port Package can have up to 63 DTI trunks for K-CCIS voice path.
- The Expanded Port Package can have up to 199 DTI trunks for K-CCIS voice path.
- The K-CCIS feature shares the CO/PBX/Tie/DID trunks available for the system.
- When assigning a Closed Numbering Plan and DID conversion across K-CCIS is required, the Electra Elite IPK II uses the ARS/F-Route Tables.
- The Electra Elite IPK II uses the F-Route Tables to assign an Open Numbering Plan.
- When all K-CCIS voice channels are busy, the Electra Elite IPK II originator of a K-CCIS call hears a busy tone from the system.

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- Outgoing CO calls in a K-CCIS network can be routed over the K-CCIS link and use the distant system CO lines.
 - Distant system extension numbers in the K-CCIS network can be assigned to Feature Access or One Touch keys and Speed Dial buffers.
 - When a K-CCIS trunk is on hold, the Specified Line Seizure access codes can be used to retrieve the call from its held state.
 - The Electra Elite IPK II can support only 2~8-digit station numbers.
 - Station Numbers are assigned by the 10s group for 4-digit station numbers, 100s group for 5-digit station numbers, 1,000s group for 6-digit station numbers, 10,000s group for 7-digit station numbers.
 - When Voice Mail Message Waiting status must be sent across the K-CCIS to a remote system, F-Routes must be used.
 - For a Closed Numbering Plan network using F-Routes, a maximum of 120 F-Route Tables are available allowing a maximum of 121 connected systems per K-CCIS network.
 - When a Closed Numbering Plan Network is used, a user can call another station by dialing the distant extension number, but extensions in the network cannot have the same prefix.
 - For an Open Numbering Plan network, a user can dial another station by dialing the office location number plus an extension number and the extension number can have the same prefix, but the office location cannot be the same.
 - When an Electra Elite IPK II system is a tandem system (in the middle) between systems with higher K-CCIS feature support (including NEAX PBXs), only the K-CCIS features supported by the Electra Elite IPK II tandem system are passed through and supported.
 - An Electra Elite IPK II K-CCIS network should never have more than five hops (tandem connections) because of the message delay through each tandem system.
 - A Star topology network supports only five systems because of the CCH channels in the Main/Hub system.
 - A Tree topology network is supported. The maximum number of systems depends on the Numbering Plan used and the maximum number of hops (tandem connections).
 - A Mesh topology network is not supported.
 - K-CCIS requires assigning a point code for each office. Point codes differentiate between an originating office and a destination office in the K-CCIS Network. Assigning point codes requires the following considerations:
 - The point code must be unique in the network.
 - When a system has two or more CCH channels, the same originating point code must be assigned to all channels in the system.
 - The Electra Elite IPK II can have a maximum of 255 codes assigned to distant systems.
 - Using an Electra Elite IPK II-to-Electra Elite IPK II network, centralized voice mail is not supported when an Open Numbering Plan is used.

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- Centralized E911 – K-CCIS is supported.
 - When Centralized E911 – K-CCIS is not used, each Electra Elite IPK II system in a K-CCIS network must have at least one trunk for Emergency 911 calls.
 - Using a NEAX-to-Electra Elite IPK II network, the PBX must supply centralized voice mail.
 - Multiline terminals must have an available Call Appearance (CAP) key to originate or answer a K-CCIS trunk call.
 - Direct access of K-CCIS voice or data channels using Line keys or Specified Line Seizure access codes is prohibited.
 - The Recall key or Drop key does not function on K-CCIS calls. When either key is pressed, operation is ignored, and the call continues.
 - Trunk queuing is prohibited on a K-CCIS trunk route.
 - Routing an incoming DID call directly across a K-CCIS link (Direct Inward Dialing - K-CCIS) is supported only when a Closed Numbering Plan using F-Routes is used.
 - This feature is not supported by the TLI(2)-U() Analog Line interface.
 - Only one CCH(4)-U() ETU can be assigned per system.
 - Extension numbers cannot start with 0 or 9.
 - Internal Calls, transferred calls, and K-CCIS calls do not provide Caller ID to single line telephones.
 - Caller ID Call Return feature is not supported with K-CCIS calls.
 - Call Park Searching is supported only in the local system.
 - With **Version 1600 or higher**, a remote K-CCIS user can call, or transfer a call directly to an ACD Pilot number. However, an incoming K-CCIS call to the ACD Pilot number does NOT provide a Link Reconnect.
 - When the system searches the Dial Extension Analyze Table (PRG 11-20-01), the system uses prefix searching, giving the lower table number the higher priority. For example, the user programs 211 in table 1 and 2113 in table 2, then dials 2113, the system selects table 1.
 - Program 11-01-01 9: System Numbering – Dial Extension Analyze is not supported in **Version 1600**.
 - Program 11-20-01: Dial Extension Analyze Table – Dial and Program 11-20-02: Dial Extension Analyze Table – Type of Dials are not supported in **Version 1600**.

Default Setting

Not installed

System Availability

Terminals

All Stations

Required Component(s)

- DTI-U30 ETU or higher
- CCH(4)-U() ETU with **firmware 5.0X or higher**
- The following table shows the KSU system software compatibility with CCH(4)-U() firmware and K-CCIS feature compatibility.

KSU Software	CCH V5.XX
IPK II V1000 V1.0X	X

X = Compatible

– = Not Compatible

Related Features

T1 Trunking (with ANI/DNIS Compatibility)

Universal Slots

Programming

Refer to the Electra Elite IPK II Key-Common Channel Interoffice Signaling (K-CCIS) Manual.

Operation

Refer to the Electra Elite IPK II Key-Common Channel Interoffice Signaling (K-CCIS) Manual.

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Last Number Redial

Description

Last Number Redial allows an extension user to quickly redial the last number dialed. For example, a user may quickly recall a busy or unanswered number without manually dialing the digits.

Last Number Redial saves in system memory the last 24 digits a user dials. The number can be any combination of digits 0~9, # and *. The system remembers the digits regardless of whether the call was answered, unanswered or busy. The system normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

When pressing the Redial key, the display indicates REDIAL [#] / SYS. The user can then press # to redial the number displayed, or enter an System Speed Dialing bin number to be dialed. Pressing the Redial key repeatedly scrolls through the last 10 numbers dialed.

Conditions

- Redial List requires a display telephone. Non-display and single line telephones cannot use this feature.
- When using Automatic Route Selection, ARS selects the trunk for the call unless the user preselects.

Default Setting

Enabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Automatic Route Selection

Repeat Redial



Save Number Dialed



Programming

- **11-12-12 Service Code Setup (for Service Access) – Last Number Dial**
Assign a service code (#5) to use Last Number Dial.
- **11-12-17 Service Code Setup (for Service Access) – Clear Last Number Dialing Data**
Assign a service code (776) to clear the Last Number Dial.
- **15-02-13 Multiline Telephone Basic Data Setup – Redial List Mode**
Select the type of numbers that are stored in the Redial List – Internal and External numbers (0) or External only (1).
- **20-08-05 Class of Service Options (Outgoing Call Service) – Dial Number Preview (Preset Dial)**
Turn Off (0) or On (1) an extension user ability to use Dial Number Preview. This program also turns Off or On the Last Number Redial function.


Operation

To redial your last call:

1. Without lifting the handset, press **Redial**.
 *The last dialed number is displayed.*
2. To redial the last number, press #.
- OR -
Search for the desired number from the Redial List by pressing **Redial** or VOLUME ▲ or VOLUME ▼ keys.
3. Lift the handset or press **Speaker** to place the call.
 *The system automatically selects a trunk from the same group as your original call and dials the last number dialed.*
- OR -

1. At the multiline terminal, press **Speaker** or lift the handset (optional).
 *The system automatically selects a trunk from the same group as your original call.*
2. Press **Redial**.
- OR -
At the single line telephone, lift the handset.
3. Dial **#5**.
 *The system automatically selects a trunk from the same group as your original call and dials the last number dialed.*

To check the number saved for Last Number Redial:

1. Press **Redial**.
 *The stored number displays for six seconds. The stored number dials out if you:*
 - Lift the handset,
 - Press an idle line key,
 - or -
 - Press **Speaker**.
2. Press the **Exit** key.

To erase the stored number:

1. At the multiline terminal, press **Speaker** or lift handset.
- OR -
At the single line telephone, lift the handset.
2. Dial **776**.

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Licensing

Enhancements

Version 1500 or higher license allows for the system to have Session Initiation Protocol (SIP) Trunks/Stations.
Version 1600 or higher supports the 30-day promo license.

Description

Licenses are used to activate certain features and applications for the Electra Elite IPK II. The IPK II provides the following licenses:

CPU Licenses

The IPK II CPU can be licensed for the following features:

- InACD – This license allows the system to run the In ACD feature.
- SMDR – This license allows the system to print SMDR reports.
- Remote Software Upgrade – This license allows the system to be upgraded remotely.
- MEGACO Stations – 16 MEGACO Station licenses are provided with every PVA card with MG16 or MGCCoIP applications installed when purchased, but more licenses can be purchased in increments of one, four, or eight with a maximum of 256.
- SIP – This license allows the system to have SIP Trunks/Stations. The licenses can be purchased in increments of one, four, or eight with a maximum of 256. (**Version 1500 or higher required.**)
- Hotel/Motel – This license allows the system to run the Hotel/Motel feature.
- CTI – This license allows the system to have a third party CTI Server connected.
- CPUII Feature Version License is required for the system to be upgraded to system software 2.0 or higher.

Temporary 10-Day License

The Temporary 10-Day license turns on ACD, CTI, Remote Software Upgrade, Hotel/Motel, 256 MEGACO stations, and SMDR license for up to 10 days.

Conditions

- Temporary 10-Day license can be activated only from a Multiline terminal, not PCPro, or WebPro.
- Each time the temporary license is activated, the program is read only until the temporary license expires.
- Each time the temporary license expires, it can be set again for up to 10 additional days.
- After setting a number of days in the program, subsequent days show one less as it counts down to expiration.
- When PRG 90-37-01 shows 1, the license expires at midnight on that day. When the license expires, the system resets.
- If the date is changed in PRG 10-01-XX, while the temporary license is in effect, one day is subtracted from the license period.
- If the date is changed in PRG 10-01-XX, when the temporary license shows 1 day, the system resets when it is applied (transfer key), not when exiting programming mode.
- If any CPU license is activated when the 10 days licensing is being used, the license is reset with only the activated license(s).

30-Day Promo License

The 30-Day Promo license turns on ACD, CTI, Remote Software Upgrade, Hotel/Motel, 256 MEGACO stations, and SMDR license for up to 30 days. (**Version 1600 or higher required.**)

To request for a 30 Day Promo license, go to <http://activate.necii.com/promo> and enter in the CPUII generated key and your Tech ID. This can be done only twice per CPUII.

Conditions

- This can be used only twice for each IPK II CPUII.
- When the Promo License is generated, it gives the date that the license ends (30 days **include** the day generated).
- If the date is changed in PRG 10-01-XX, while the license is in effect, to a date before the generated date it subtracts one day from the Promo license.
- If the date is changed in PRG 10-01-XX, while the license is in effect, to a date after the generated date, it runs until the End Date specified when the License was generated.
- If the date is changed in PRG 10-01-XX, while the license is in effect, to a date after the End Date specified when the License was generated, the system resets when it is applied (transfer key pressed), not when exiting program mode.
- When the System time turns to midnight of the End Date, the system resets and comes back with no licenses.

- The 30 Day Promo License can be activated only from PCPro or WebPro, not a Multiline Terminal.
- If any CPUII license is activated when the Promo licensing is being used, the CPUII license resets with only the activated license(s).

Application Licenses

The IPK II system has extra applications that can be purchased and other application software that works with the IPK II system. Please refer to the specific applications manual for more information about licensing each of the following features:

- Elite Call Analyst
- PC Assistant
- PC Attendant
- PC Pro
- IPK II ACD MIS

Voice Mail Licenses

The Voice Mail Card that is supported in the IPK II has additional features that can be purchased to allow for more advanced applications. Please refer to the Voice Mail manual for further information about the features and how to license them.

PVA Licenses

The PVA card in the IPK II system can be used for multiple applications such as:

- IP-CCIS (firmware version **SP01H** is required)
- MEGACO Station Card (firmware version **SP01F** is required)
- Conference Card (firmware version **SP01Q** is required)
- MG_CCIS Combo Package (**version 1.0** and **SP01J** license are required)

Refer to the Electra Elite IPK II VoIP Reference Manual and Electra Elite IPK and Electra Elite IPK II Elite Multimedia Conference Bridge Installation Manual (PVA ETU) for licensing procedures on these features.

Default Settings

No licensing.

System Availability

Terminals

N/A

Required Component(s)

Refer to the particular Feature for required component(s)

Related Features

Programming from a Multiline Terminal

Programming

- **90-37-01 Temporary License – Number of Days**
Assign the number of days for the Temporary License to run (up to 10 days). Once set, this program is Read Only.
- **90-37-02 Temporary License – Valid Days for Temporary License (Version 1600 or higher required)**
This is a read only program. This tells you how many days you have left on the 30 Day Promo license. Only viewable through handset programming.

Operation

Please refer to the following manuals for how to activate each feature:

- ACD MIS – ACD MIS Activation Procedure
- CPUII Features – CPUII()-U10 ETU Feature Activation Procedure
- PC Assistant – PC Assistant Activation Procedure
- PC Attendant – PC Attendant Feature Activation Procedure
- PC Pro – PCPro Activation Procedure
- PVA – PVA(X)-U() ETU License Instructions

Line Preference

Description

Line Preference determines how a multiline terminal user places and answers calls. There are two types of Line Preference: Incoming Line Preference and Outgoing Line Preference.

Incoming Line Preference

Incoming Line Preference establishes how a multiline terminal user answers calls. When a call rings the multiline terminal, lifting the handset answers either the ringing call (for Ringing Line Preference) or seizes an idle line (for Idle Line Preference). The idle line can provide either Intercom or trunk dial tone (see Outgoing Line Preference below). Ringing Line Preference helps users whose primary function is to answer calls (such as a receptionist). Idle Line Preference is an aid to users whose primary function is to place calls (such as a telemarketer).

Outgoing Line Preference

Outgoing Line Preference sets how a multiline terminal user places calls. If a multiline terminal has Outgoing Intercom Line Preference, the user hears Intercom dial tone when they lift the handset. If a multiline terminal has Outgoing Trunk Line Preference, the user hears trunk dial tone when they lift the handset. Outgoing Line Preference also determines what happens at extensions with Idle Line Preference. The user hears either trunk (dial 9) or Intercom dial tone.

Auto-Answer of Non-Ringing Lines

With Auto-Answer of Non-Ringing Lines, an extension user can automatically answer trunk calls that ring other extensions (not their own). This helps a user that has to answer calls for co-workers that are away from their desks. When the user lifts the handset, they automatically answer the ringing calls based on Trunk Group Routing programming. The extension user's own ringing calls, however, always have priority over calls ringing other co-worker's extensions.

Conditions

- If a multiline terminal extension has more than one call ringing its line keys, Ringing Line Preference answers the calls on a first-in first-answered basis.
- DILs do not affect Incoming Line Preference operation.
- Trunks ring extensions according to Ring Group programming.
- If an extension gets trunk dial tone when the user lifts the handset, the system uses the dial 9 routing to select the trunk.

Default Setting

Enabled

System Availability

Terminals

Multiline Terminals

Required Component(s)

None

Related Features

Direct Inward Line

Ring Groups

Trunk Groups

Programming

- **14-05-01 Trunk Group – Trunk Group Number**
For Auto-Answer of Non-Ringing Lines, assign trunks to trunk groups. This is part of Trunk Group Routing programming.

Trunks: 1~100
- **14-06-01 Trunk Group Routing – Priority Order Number**
For Auto Answer of Non-Ringing Lines, program the Trunk Group Routing table. Auto Answer uses this table to determine the answer sequence for ringing calls.

Trunk Groups: 1~100
- **14-07-01 Trunk Access Map Setup**
For Outgoing Line Preference and Auto-Answer of Non-Ringing Lines, set up the Trunk Access Maps.

Trunk Access Maps: 1~200

-
-
- ➔ **15-01-02 Basic Extension Data Setup – Outgoing Trunk Line Preference**
Enable (1)/Disable (0) Outgoing Trunk Line Preference for extensions.
 - ➔ **15-02-10 Multiline Telephone Basic Data Setup – Ringing Line Preference for Trunk Calls**
Enable Idle (0)/Ringing (1) Line Preference for trunk calls. Program 22-01-01 sets Intercom (0) or trunk (1) call priority.
 - ➔ **15-06-01 Trunk Access Map for Extensions**
For Outgoing Line Preference and Auto-Answer of Non-Ringing Lines, assign trunk Access Maps to extensions.

Trunk Access Maps: 1~200
 - ➔ **20-10-07 Class of Service Options (Answer Service) – Automatic Off-Hook Answer**
Enable (1)/Disable (0) an extension user ability to use Universal Auto Answer for non-ringing lines.
 - ➔ **22-01-01 System Options for Incoming Calls – Incoming Call Priority**
Determine if ringing Intercom calls (0) or ringing trunk calls (1) should have answer priority.
 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign extensions to ring groups. Auto-Answer for Non-Ringing Lines only works for trunks that do not ring an extension.

Ring Groups:
1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
 - ➔ **22-05-01 Incoming Trunk Ring Group Assignment**
Assign trunks to ring groups. Auto-Answer for Non-Ringing Lines only works for trunks that do not ring an extension.

Ring Groups:
1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
 - ➔ **23-03-01 Universal Answer/Auto Answer**
For each extension, select the route that it uses for Auto-Answer for Non-Ringing Lines. The routes correspond to the Trunk Group Routing table set in Program 14-06.

Trunk Group Routes: 1~100

Operation

Ringing Trunk or intercom (ICM) call:

Lift the handset or press **Speaker**.

 *The setting assigned for PRG 15-02-10 and PRG 22-01-01 determines which call is answered first.*

Long Conversation Cutoff

Description

For incoming and outgoing central office calls, each trunk can be programmed to disconnect after a defined time. The time begins when the trunk is seized and the call disconnects after the time expires.

When used with the Warning Tone for Long Conversation feature, the system can provide a warning tone on outgoing trunks calls before the call is disconnected.

Conditions

- Long Conversation Cutoff can disconnect incoming and outgoing CO calls after a set time period.
- Long conversation cutoff is controlled separately for DISA and Tie Lines.
- Using the Warning Tone for Long Conversation feature allows users on outgoing calls to hear a warning tone prior to the call disconnecting.

Default Setting

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Central Office Calls, Answering

Central Office Calls, Placing

Direct Inward System Access (DISA)

Tie Lines

Warning Tone for Long Conversation

Programming

- **14-01-14 Basic Trunk Data Setup – Long Conversation Cutoff**
Enable (1)/Disable (0) a trunk user ability to disconnect incoming and outgoing central office calls automatically.
- **14-01-15 Basic Trunk Data Setup – Long Conversation Alarm Before Cut Off**
Enable (1)/Disable (0) the Long Conversation Alarm for each trunk.
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to extensions.
- **20-13-02 Class of Service Options (Supplementary Service) – Long Conversation Cutoff (Incoming)**
Enable (1)/Disable (0) the ability of an extension Class of Service to use Long Conversation Cutoff for incoming calls.
- **20-13-03 Class of Service Options (Supplementary Service) – Long Conversation Cutoff (Outgoing)**
Enable (1)/Disable (0) the ability of an extension Class of Service to use Long Conversation Cutoff for outgoing calls.
(default: 0)
- **20-21-03 System Options for Long Conversation – Long Conversation Cutoff for Incoming Call**
Enter the time the system waits before disconnecting incoming trunks (0~64800 seconds).
(default: 0)
- **20-21-04 System Options for Long Conversation – Long Conversation Cutoff for Outgoing Call**
Enter the time the system waits before disconnecting outgoing trunks (0~64800 seconds).
(default: 0)



Operation

This feature is automatic once it is programmed.

Meet Me Conference

Description

With Meet Me Conference, an extension user can set up a Conference with their current call and up to 31 other internal or external parties. Each party joins the Conference by dialing a Meet Me Conference code. Meet Me Conference lets extension users have a telephone meeting – without leaving the office.

The CPUII provides two blocks of 32 conference circuits, allowing each block to have any number of internal or external parties conferenced up to the block limit of 32.

Conditions

None

Default Setting

Enabled

System Availability

Terminals

Multiline Terminals

Required Component(s)

None

M


Related Features

Conference

Meet Me Paging

Programmable Function Keys

Programming

 For additional programming for Paging, refer to the *Paging External* and *Paging Internal* features.

- **15-07-01 Programmable Function Keys**
Assign function keys for Conference (code 07).
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- **20-10-06 Class of Service Options (Answer Service) – Meet-Me Conference and Paging**
Enable (1)/Disable (0) an extension user ability to initiate a Meet Me Conference or Meet Me Page.
- **31-01-04 System Options for Internal/External Paging – Privacy Release Time**
Set the Privacy Release Time (0~64800 seconds). After the user initiates Meet Me Conference, the system waits this time for the Paged party to join the conversation.
(default 90)

Operation

Meet Me External Conference

To make a Meet Me External Conference:


Multiline Terminal

1. While on a call, press **Conf**.
2. Dial *1 and the Combined Paging Zone code **1~8** (for Internal/External Zones 1~8) or 0 (for Internal/External All Call).
3. Announce the zone.
4. When a co-worker answers your page, press **Conf** twice.
5. Repeat steps 1~4 for each co-worker you want to add.

Single Line Telephone

1. While on a call, hookflash and dial **#1**.
2. Dial **703** and the External Paging zone code (**1~8** or **0** for All Call).
- OR -
Dial ***1** and the Combined Paging Zone code **1~8** (for Internal/External Zones 1~8) or **0** (for Internal/External All Call).
3. Announce the zone.
4. When a co-worker answers your page, press hookflash twice.
5. Repeat steps 1~4 for each co-worker you want to add.

To join a Meet Me External Conference:

1. At the multiline terminal, press **Speaker**.
- OR -
At a single line telephone, lift the handset.
2. Dial **765**.
3. Dial the announced External Paging Zone code (**0~8**).
 *You connect to the other parties.*

Meet Me Internal Conference**To make a Meet Me Internal Conference:**Multiline Terminal

1. While on a call, press **Conf**.
2. Dial ***1** and the Combined Paging Zone code **1~8** (for Internal/External Zones 1~8) or **0** (for Internal/External All Call).
3. Announce the zone.
4. When a co-worker answers your page, press **Conf** twice.
5. Repeat steps 1~4 for each co-worker you want to add.

Single Line Telephone

1. While on a call, hookflash and dial **#1**.
2. Dial ***1** and the Combined Paging Zone code **1~8** (for Internal/External Zones 1~8) or **0** (for Internal/External All Call).
3. Announce the zone.

4. When a co-worker answers your page, press hookflash twice.
5. Repeat steps 1~4 for each co-worker you want to add.

To join a Meet Me Internal Conference:

1. At the multiline terminal, press **Speaker** (or lift the handset).

- OR -

At the single line telephone, lift the handset.

2. Dial **763** (if your extension is in the zone called).

- OR -

Dial **764** and the zone number (if your extension is not in the zone called).

- OR -

Press the Meet Me Conference/Paging Pickup key (PRG 15-07 or 23) if your extension is in the zone called.

Meet Me Paging

Description

Meet Me Paging allows an extension user to Page a co-worker and privately meet with them on a Page zone. The Paging zone is busy to other users while the meeting takes place. While the co-workers meet on the zone, no one else can hear the conversation, join in or make an announcement using that zone. Meet Me Paging is a good way to talk to a co-worker when their location is unknown. If the co-worker can hear the Page, they can join in the conversation.

Conditions

- With Meet Me Paging Transfer, a user can page a co-worker and have the call automatically transfer when the co-worker answers the page.
- An extension access to internal and external page zones affects the Meet Me Paging feature.
- Internal and External Paging keys simplify Meet Me Paging operation.

Default Setting

Enabled

System Availability

Terminals

Multiline Terminals

Required Component(s)

External paging requires a PGD(2)-U10 ADP be installed in the system.

Related Features

Meet Me Conference


Meet Me Paging Transfer




Paging, External

Paging, Internal

Programmable Function Keys

Programming

 For additional programming information on Paging, refer to the [Paging, External](#) and [Paging, Internal](#) features.

-  **15-07-01 Programmable Function Keys**
Assign function keys for External Zone Paging (code 19 + zone), External All Call Paging (code 20), Internal Zone Paging (code 21 + zone) or Meet Me Conference/Paging Pickup (code 23).
-  **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
-  **20-10-06 Class of Service Options (Answer Service) – Meet-Me Conference and Paging**
Enable (1)/Disable (0) an extension user ability to initiate a Meet Me Conference or Meet Me Page.

Operation

Meet Me External Page

To make a Meet Me External Page:

1. At multiline terminal, press **Speaker** or pick up the handset.

- OR -

At the single line telephone, lift the handset.

2. Dial **703** and the External Paging Zone code (1~8 or 0 for All Call).

- OR -

Dial ***1** and the Combined Paging Zone code 1~8 (for Internal/External Zones 1~8) or 0 (for Internal/External All Call).

3. Announce the zone.

- OR -

1. At the multiline terminal, press the **External Paging Zone** key (PRG 15-07 or SC 751: 19 + zone).
2. Announce the zone.


To join a Meet Me External Page:

1. At the multiline terminal, press **Speaker** or pick up the handset.

- OR -

At the single line telephone, lift the handset.

2. Dial **765**.
3. Dial the announced External Paging Zone (0~8).

 *You connect to the other party.*

Meet Me Internal Page

To make a Meet Me Internal Page:

1. At the multiline terminal, press **Speaker** or pick up the handset.

- OR -

At the single line telephone, lift the handset.

2. Dial **701** and dial the Internal Paging Zone code (0~9, 00~32 or 00~64).

- OR -

Dial ***1** and the Combined Paging Zone code 1~8 (for Internal/External Zones 1~8) or 0 (for Internal/External All Call).

3. Announce the zone.

- OR -

1. At the multiline terminal, press the **External Paging Zone** key (PRG 15-07 or SC 751: 19 + zone).
2. Announce the zone.

To join a Meet Me Internal Page:

1. At the multiline terminal, press **Speaker** or pick up the handset.

- OR -

At the single line telephone, lift the handset.

2. Dial **763** (if your extension is in the zone called).

- OR -

Dial **764** and the zone number (if your extension is not in the zone called).

- OR -

Press the Meet Me Conference/Paging Pickup key (PRG 15-07 or SC 751: 23) if your extension is in the zone called.

Meet Me Paging Transfer

Description

If a user wants to Transfer a call to a co-worker but they do not know where the co-worker is, they can use Meet Me Paging Transfer. With Meet Me Paging Transfer, the user can Page the co-worker and have the call automatically Transfer when the co-worker answers the Page. Since Meet Me Paging Transfer works with both Internal and External Paging, a call can be quickly extended to a co-worker anywhere in the facility.

Conditions

- An extension user can set up a conference with their current call and up to 31 other inside parties.
- An extension user can Page a co-worker and meet with them on a page zone.
- With External Paging, an extension user can broadcast an announcement over paging equipment connected to external paging zones.
- Internal Paging lets extension users broadcast announcements to other multiline terminals.
- Function keys simplify Meet Me Paging Transfer operation.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

External paging requires a PGD(2)-U10 ADP be installed in the system.

Related Features

Meet Me Conference


Meet Me Paging




Paging, External

Paging, Internal

Programmable Function Keys

Programming

 For additional programming information on Paging, refer to the [Paging, External](#) and [Paging, Internal](#) features.


-  **15-07-01 Programmable Function Keys**
Assign function keys for External Zone Paging (code 19 + zone), External All Call Paging (code 20), Internal Zone Paging (code 21 + zone) and Meet Me Conference/Paging Pickup (code 23).
-  **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
-  **20-10-06 Class of Service Options (Answer Service) – Meet-Me Conference and Paging**
Enable (1)/Disable (0) an extension user ability to initiate a Meet Me Paging Transfer.

Operation



Meet Me External Paging Transfer

To make a Meet Me External Paging Transfer:

1. At the multiline terminal, while on a call, press **Hold**.
- OR -
At the single line telephone, while on a call, hookflash.

2. Press the **External Paging Zone** key (PRG 15-07 or SC 751: 19 + zone).
 - OR -
 - Dial **703** and the External Paging Zone code (1~8 or 0 for All Call).
 - OR -
 - Dial ***1** and the Combined Paging Zone code 1~8 (for Internal/External Zones 1~8) or 0 (for Internal/External All Call).
3. Announce the call.
4. From a multiline terminal, when the paged party answers, press **Transfer** or the **Transfer** softkey.
 - OR -
 - From a single line telephone, when the paged party answers, hang up.
 -  *The party is transferred.*


To join a Meet Me External Paging Transfer:

1. At the multiline terminal, press **Speaker** or pick up handset.
 - OR -
 - At single line telephone, lift the handset.
2. Dial **765**.
3. Dial the announced External Paging Zone (0~8).
 -  *The Paging party is connected.*
4. Stay on the line.
 - From a multiline terminal, press **Transfer** or the **Transfer** softkey.
 - OR -
 - From a single line telephone, hang up.
 -  *The party is transferred.*


Meet Me Internal Paging Transfer

To make a Meet Me Internal Paging Transfer:

1. At multiline terminal, while on a call, press **Hold**.
 - OR -
 - At the single line telephone, while on a call, hookflash.

2. Press **Internal Paging Zone** key (PRG 15-07 or SC 751: 20 + zone).
 - OR -
 - Dial **701** and the Internal Paging Zone code (0~9 or 00~64).
 - OR -
 - Dial ***1** and the Combined Paging Zone code 1~8 (for Internal/External Zones 1~8) or 0 (for Internal/External All Call).
3. Announce the call.
4. From a multiline terminal, when the paged party answers, press **Transfer** or the **Transfer** softkey.
 - OR -
 - From a single line telephone, when the paged party answers, hang up.
 -  *The party is transferred.*

To join a Meet Me Internal Paging Transfer:

1. At the multiline terminal, press **Speaker** or pick up handset.
 - OR -
 - At the single line telephone, lift the handset.
2. Dial **763** (if your extension is in the zone called).
 - OR -
 - Dial **764** and the zone number (if your extension is not in the zone called).
 - OR -
 - Press the Meet Me Conference/Paging Pickup key (PRG 15-07 or SC 751: 23) if your extension is in the zone called.
3. Stay on the line.
 - From a multiline terminal, when the paged party answers, press **Transfer** or the **Transfer** softkey.
 - OR -
 - From a single line telephone, when the paged party answers, hang up.
 -  *The party is transferred.*

Memo Dial

Description

While on an outside call, Memo Dial lets a multiline terminal user store an important number for easy redialing later on. The telephone can be like a notepad. For example, a user could dial Directory Assistance and ask for a client's telephone number. When Directory Assistance plays back the requested number, the caller can use Memo Dial to jot the number down in the telephone memory. They can quickly call the Memo Dial number after hanging up.

When a user enters a Memo Dial number, the dialed digits do not output over the trunk. Dialing Memo Dial digits does not interfere with a call in progress.

Conditions

- When Memo Dial calls out, it outdials the entire stored number. Memo Dial does not automatically strip out trunk or PBX access codes if entered as part of the stored number.
- Only one number can be stored at a time.
- If a number is already stored in Memo Dial and you are on a internal or external call and the Dial Memo Key is pressed, the number is erased.
- A user's outgoing dialing options affect how a Memo Dial call is placed.
- Memo Dial is not available at single line telephones.

Default Setting

Disabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Central Office Calls, Placing

Last Number Redial

Save Number Dialed

Programming

- **15-07-01 Programmable Function Keys**
Assign a function key for Memo Dial (code 31).


Operation

To store a number while you are on a call:

1. While on a call, press Memo Dial key (PRG 15-07 or SC 751: 31).
2. Dial number you want to store.
3. Press Memo Dial key again and continue with conversation.


To call a stored Memo Dial number:

1. Do not lift the handset.
2. Press the Memo Dial key (PRG 15-07 or SC 751: 31).
3. Press **Speaker**.

 *The stored number dials out only if you store a trunk access code before the number.*


- OR -

Press the **line** key.

 *The stored number dials out.*

To check to see the stored Memo Dial number:

1. Do not lift the handset.
2. Press **Memo Dial** key (PRG 15-07 or SC 751: 31).

 *The stored number displays.*

To cancel (erase) a stored Memo Dial number:

1. Press **Speaker**.
2. Press the **Memo Dial** key (PRG 15-07 or SC 751: 31).

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Message Waiting

Enhancements

Version 1600 or higher:

- Allows the flash rate and color of the Message Waiting LED to be configured for the following conditions:
 - Message Waiting Lamp Cycle for Calling Extension (PRG 15-02-35)
 - Message Waiting Lamp Cycle for Called Extension (PRG 15-02-36)
 - Voice Mail Message Wait Lamp Color (PRG 15-02-37)
 - Voice Mail Message Wait Lamp Cycle (PRG 15-02-38)

This provides more distinction between incoming calls, Message Waiting (Set/Received) and a VM Message Waiting indication.

- Allows the flash rate and color to be configured for the Message Waiting LED. This provides more distinction between incoming CO calls and Message Waiting Indication.

Description

An extension user can leave a Message Waiting indication at a busy or unanswered extension requesting a return call. The indication is a flashing MW lamp at the called extension and a lighted MW lamp on the calling extension. Answering the Message Waiting automatically calls the extension which left the indication. Message Waiting ensures that a user does not have to recall an unanswered extension. It also ensures that a user does not miss calls when their extension is busy or unattended. Additionally, Message Waiting lets extension users:

- View and selectively answer messages left at their extension (display multiline terminal only)
- Cancel all messages left at their extension
- Cancel messages they left at other extensions

An extension user can leave Messages Waiting at any number of extensions. Also, any number of extensions can leave a Message Waiting at the same extension. A periodic VRS announcement may remind users that they have Messages Waiting.

Message Key Operates as Voice Mail Key


The system enhances a telephone Message key function when connected to a system which has voice mail installed. When an extension receives a voice mail, the Message key can be used to check the number of messages in voice mail, as well as call the voice mail to listen to the messages. If no Voice Mail Programmable Function Key is defined (Program 15-07-01, code 77), the telephone Message Waiting LED flashes to indicate new messages.

This option is not available with a networked voice mail – the voice mail must be local.

Refer to the [Voice Mail Integration \(Analog\) on page 2-1057](#) feature for the feature operation.

LED Color Indication

The software allows you to select whether the Message Wait LED located at the top of the multiline terminal flashes green (0) or red (1) when a Message Wait indication is flashing. By default, this option is set to flash red.

 *If this LED is also used for voice mail indications (no Programmable Function Key programmed for voice mail), if there are both voice mail messages and Message Wait indications, the color set for Message Wait overrides the color used for voice mail indications (red).*

Conditions

- Reminder messages require a DSP daughter board for VRS messages.
- Analog ports from APA or APR adapters do not provide Message Waiting lamps.
- When a user responds to a Message Waiting, the system does not cancel the Message Waiting indication if the called party uses Handsfree Answerback. The system cancels the indication only if the called party lifts the handset or presses Speaker.
- With the Hotel/Motel set up, an employee with a multiline terminal can send a Message Waiting to a room telephone if allowed in system programming.
- A Message Waiting key simplifies this feature operation.
- Reminder messages require a DSP daughter board for Voice Response System (VRS).
- Telephone-to-telephone Message Waiting works when the voice mail is installed.
- The MW (Message Waiting) LED may be used to indicate voice mail messages if no extension number is assigned to the voice mail key in system programming.
- If the following programs are changed while the phone is online, a reset of the feature is required before the setting takes effect.
 - ❑ PRG 15-02-35 Message Waiting Lamp Cycle for Calling Extension
 - ❑ PRG 15-02-36 Message Waiting Lamp Cycle for Called Extension
 - ❑ PRG 15-02-37 Voice Mail Message Wait Lamp Color
 - ❑ PRG 15-02-38 Voice Mail Message Wait Lamp Cycle
- If both Voice Mail Message and Message Wait indication is set, the color set for Message Wait overrides the color used for Voice Mail Message indication.

Default Setting

Enabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Handsfree Answerback/Forced Intercom Ringing

Hotel/Motel

Programmable Function Keys

Voice Mail Message Indication on Line Keys

Voice Response System (VRS)

Programming

- **15-02-28 Multiline Telephone Basic Data Setup – Message Waiting Lamp Color**
Select whether the Message Wait LED located at the top of the multiline terminal flashes green (0) or red (1) when a Message Wait indication is flashing. If this LED is also used for voice mail indications (no Programmable Function Key programmed for voice mail), if there are both voice mail messages and Message Wait indications, the color set for Message Wait overrides the color used for voice mail indications.
(default: 1)


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-
- ➔ **15-02-35 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Calling Extension (Version 1600 or higher required)**
This option allows you to select the Message Waiting flash pattern for the station that set the Message Waiting reminder.
- 1 = Cycle 1
 - 2 = Cycle 2
 - 3 = Cycle 3 (default)
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7
- ➔ **15-02-36 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Called Extension (Version 1600 or higher required)**
This option allows you to select the Message Waiting flash pattern for the station that receives the Message Waiting reminder
- 1 = Cycle 1
 - 2 = Cycle 2 (default)
 - 3 = Cycle 3
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7
- ➔ **15-02-37 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Color (Version 1600 or higher required)**
This option allows you to select the Message Waiting flash pattern for the station that set the Message Waiting reminder.
- 0 = Green
 - 1 = Red
- (default: 1)
- ➔ **15-02-38 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Cycle (Version 1600 or higher required)**
This option allows you to select the Message Waiting flash pattern for the station that set the Message Waiting reminder.
- 1 = Cycle 1
 - 2 = Cycle 2 (default)
 - 3 = Cycle 3
 - 4 = Cycle 4
 - 5 = Cycle 5
 - 6 = Cycle 6
 - 7 = Cycle 7
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Message Waiting (code 38).
-
-

- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-07 Class of Service Options (Supplementary Service) – Message Waiting**
Enable (1)/Disable (0) an extension user ability to leave Messages Waiting.
- ➔ **24-02-05 System Options for Transfer – Message Wait Ring Interval Time**
For single line telephones without message waiting lamps, this determines the time between intermittent ringing. If this time is set to 0, the system rings once.
- ➔ **80-01-01 Service Tone Setup – Repeat Count**
Set repeat count for tone 16 Lockout, default = 0 (endless).
Refer to [Table 2-26 Service Tone Setup, Program 80-01-01 on page 2-476](#).
- ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
(Default: 0).
Refer to [Table 2-27 Service Tone Setup, Program 80-01-02 on page 2-480](#).
- ➔ **80-01-48 Service Tone Setup – Voice Mail Message Waiting, Special Dial**
If the default tone is not acceptable for the Message Wait Dial Tone, a new tone can be assigned.

Operation

To leave a Message Waiting:

1. Call busy or unanswered extension.
2. Dial **0** or press the **Message Waiting** key (PRG 15-07 or SC 751: 38)
3. Hang up.

 *With multiline terminal telephones, the Message Waiting LED lights.*

To answer a Message Waiting:

 *When you have a message, your Message Waiting LED flashes fast for multiline terminals.*


1. At the multiline terminal, press **Speaker** and dial ***0**.


- OR -

Press the **Message Waiting** key (PRG 15-07 or SC 751: 38).

- OR -

At the single line telephone, lift the handset and dial ***0**.

 *If the called extension does not answer, dial 0 or press your **Message Waiting** key to automatically leave them a message.*

 *Normally, your Message Waiting LED goes out. If it continues to flash, you have new messages in your Voice Mail mailbox or a new General Message. Go to [To check your messages below](#).*

To cancel all your Messages Waiting:

 This includes messages you have left for other extensions and messages other extension have left for you.

1. At the multiline terminal, press **Speaker**.

- OR -

At the single line telephone, lift the handset.

2. Dial **773**.
3. Hang up.

To cancel the Messages Waiting you have left at a specific extension:

1. At the multiline terminal, press **Speaker**.


- OR -

At the single line telephone, lift the handset.

2. Dial **771**.
3. Dial the number of the extension you do not want to have your messages.
4. Hang up.

To check your messages:

1. Press **Message** key or the **MW** (Message Waiting) softkey.
2. Dial ***0**.

 You can have any combination of the message types in the table below on your telephone.

If you see. . .	You have. . .
VOICE MESSAGE n MESSAGE	New messages in your Voice Mail mailbox.
CHECK MESSAGE VRS GENERAL MESSAGE	A General message in Voice Mail that has not been heard.
CHECK MESSAGE (name)	Message Waiting requests left at your telephone by your co-workers.

3. Press VOL ▲ or VOL ▼ to scroll through your display.
4. When you find the message you want to answer, press **Speaker**. You either:
 - Go to your Voice Mail mailbox.
 - Listen to the new General Message.
 - Automatically call the extension that left you a Message Waiting.

Microphone Cutoff

Description

Microphone Cutoff lets a multiline terminal user turn off their telephone handsfree or handset microphone anytime. When activated, Microphone Mute prevents the caller from hearing conversations in the user's work area. The user may turn off the microphone while their telephone is idle, busy on a call or ringing. The microphone stays off until the user turns it back on.

Conditions

- Microphone Cutoff does not operate if the user calls another extension and the called extension responds without lifting the handset or pressing Speaker.
- When using the Handset Transmission Cutoff key during an intercom call with the handset on-hook, you hear three beep tones and the LED is on solid. This also occurs when on an outside call.
- When using the Handset Transmission Cutoff key during an intercom call with the handset off-hook, you hear three beep tones through the handset and the Handset Transmission Cutoff and MIC keys flash. This also occurs when on an outside call.
- When Handset Transmission Cutoff is activated and the handset is off-hook, press Speaker and then return the handset to the cradle to turn off the Handset Transmission Cutoff key. Three beep tones are heard over the telephone speaker.

Default Setting

Enabled (using MIC key)

System Availability

Terminals

Any Multiline Terminal

Required Component(s)

None

Related Features

Handsfree Answerback/Forced Intercom Ringing

Handset Mute

Programmable Function Keys

Programming

- **15-07-01 Programmable Function Keys**
If an extension needs handset microphone cutoff, program a Handset Transmission Cutoff key (code 40). To program a MIC Cutoff key, use code 02 as the entry. The MIC Cutoff key mutes conversation on a handsfree call, while the Handset Transmission Cutoff key mutes the handset transmission on a non-handsfree call.

- **20-02-11 System Options for Multiline Telephones – Handsfree Microphone Control**
Control the setting for multiline terminal handsfree microphone after being disconnected and reconnected. If set to 0, the microphone is always off when the terminal is reconnected. If set to 1, the microphone remains in the same state it was in when the terminal is reconnected.

Operation

To mute your telephone handset or Handsfree microphone while on a call:

1. Press **MIC**.
 - ✎ *This only turns off the Handsfree microphone.*
 - OR -
 - Press the **Microphone Cutoff** key (PRG 15-07 or SC 751: 40).
 - ✎ *This turns off both the handset and Handsfree microphone.*

To turn your telephone microphone back on:

1. Press **MIC**.
 - ✎ *Use **MIC** only if you pressed it initially to turn off your Handsfree microphone.*
 - OR -
 - Press the **Microphone Cutoff** key (PRG 15-07 or SC 751: 40).
 - ✎ *Use the **Microphone Cutoff** key only if you pressed it initially to turn off your handset or Handsfree microphone.*

Multiline Conference Bridge

Description

Multiline Conference Bridge allows any intercom or outside caller to call the CNF(8)-U() ETU to place a multiparty conference call. Each CNF(8)-U() ETU supports one 8-party conference or two 4-party conferences regulated by a switch setting. Two CNF(8)-U() ETUs may be installed. DSP-based amplification provides a higher quality conference call.

Conditions

- The CNF(8)-U() ETU can be used with any version of software on the Electra Elite IPK II system.
- When CNF is set for two, four-party conferences, ports 1~4 can be set to the same Master Hunt Group, and ports 5~8 can be set to a different Master Hunt Group using Program 11-07-01 (Department Group Pilot Numbers) and Program 16-02-01 (Department Group Assignment for Extensions). This allows two different Pilot Numbers for each conference.
- OR -
- If only one Pilot number is needed, put all eight ports in one hunt group using Program 16-02-01 (Department Group Assignment for Extensions). When the password is entered, the conference selected is the conference you enter.
- The supervisor must perform the Setting Procedures before the Conference Bridge can be used.
- Up to 16 Voice Mail ports are available. The CNF(8)-U() ETU reduces this number by eight.
- Each CNF(8)-U() ETU reduces the number of stations by eight.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

CNF(8)-U() ETU

Related Features

None

Programming

- **10-09-01 DTMF and Dial Tone Circuit Setup**

Assign at least one circuit for DTMF reception (type 0 or 1). Use the following as a guide when allocating DTMF receivers:

 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.
- **11-07-01 Department Group Pilot Numbers – Dial**

Assign a Department Group pilot number for the CNF(8)-U ETU (eight digits max). The extensions are assigned to the group in Program 16-02-01.
- **15-03-01 Single Line Telephone Basic Data Setup – SLT Signaling Type**

For each SLT extension, this option must be set to 0.
- **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**

Set all SLIU ports used for Conference as type 1 to allow the SLT port to receive DTMF tones after the initial call setup.
- **16-02-01 Department Group Assignment for Extensions**

Put all the CNF(8)-U ETU extensions in a unique Department Group (do not use Department Group 1). No other extensions should be in this group. This allows DILs to Voice Mail to ring other Voice Mail extensions when the DILs assigned port is busy. Only one Voice Mail group is allowed per system. Be sure to set up the Department Group in order. (For example, the first extension should be order 1, the second order 2, etc.)
- **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**

Enable Continued Dialing (1) for all extensions that dial the Conference ports.
- **22-02-01 Incoming Call Trunk Setup**

Assign Service Type 4 to each trunk you want to ring into the CNF(8)-U ETU as a Direct Inward Line (DIL).
- **22-07-01 DIL Assignment**

Assign the master/pilot number of the Conference group from Program 11-07-01 as the DIL destination. If all Conference ports are in the same unique Extension (Department) Group (see Program 16-02 above), the DIL rings another Conference port if its assigned port is busy.

Operation

Setup Procedures

To set the Supervisor Password:

1. Call the Conference Bridge extension number, and wait for the voice prompt.
2. Dial the default Supervisor Password (**0 0 0 0**), then dial #.
3. Follow the voice prompt, and enter the setting verification mode (#). Then enter system set mode.
4. Follow the voice prompt to change Supervisor Password (4~8 digits).

To set the Conference 1 and/or the Conference 2 Password:

1. Call the Conference Bridge extension number, and wait for the voice prompt.
2. Dial the default Supervisor Password (**0 0 0 0**), then dial #.
3. Skip the steps until Conference Setup mode is available for conference 1 or conference 2.
4. Follow the voice prompt, and set the conference 1 or the conference 2 password (4~8 digits).

To record new Customized Greeting:

1. Call the Conference Bridge extension number, and wait for the voice prompt.
2. Dial the default Supervisor Password (**0 0 0 0**), then dial #.
3. Skip options until the Customized Greeting option is selected.
4. Follow the voice prompt and record new a Customized Greeting.

Operating Procedures

To start a Conference Call at an internal extension:

1. Call the Conference Bridge extension number.
2. When you hear the voice prompt, enter the Conference Bridge 1 or 2 password, and dial #.
3. Start the conference.

To start a conference call using outside DID:

1. Call the DID number for the Conference Bridge.
2. When the voice prompt is heard, enter the Conference Bridge 1 or 2 password, and dial #.
3. Start the conference.

To start a conference call using outside DIT:

1. Call a trunk that is set as DIT to Conference Bridge.
2. When the voice prompt is heard, enter the Conference Bridge 1 or 2 password, and dial #.
3. Start the conference.

To start a conference call on an incoming CO call using an Automated Attendant:

1. Call a trunk that is set as an Automated Attendant.
2. Select the option for an extension connected to the Conference Bridge.
3. When the voice prompt is heard, enter the Conference Bridge 1 or 2 password, and dial #.
4. Start the conference.

To start a conference call from an incoming CO call using an Attendant:

1. Call the Attendant, and ask to be transferred to an extension connected to the Conference Bridge.
2. When the voice prompt is heard, enter the Conference Bridge 1 or 2 password, and dial #.
3. Start the conference.

Multimedia Conference Bridge

Enhancements

The following enhancements are available with CNF(16)-U20 ETU **V1.10 firmware or higher**.

- A conference organizer can end a conference prior to its scheduled end.
- DTMF signals are consumed by the conference bridge and therefore are not played to all participants.
- Conference participants can adjust the transmit volume level by pressing 1 (increase volume) or 3 (decrease volume) on the telephone dial pad.
- Conference participants can adjust the receive volume level by pressing 7 (increase volume) or 9 (decrease volume) on the telephone dial pad.
- Conference participants can press 0 on the telephone dial pad to toggle mute (on/off).
- Help voice prompts were added for the above options.
- A conference organizer can extend the conference call time by pressing ## on the telephone dial pad.
- The conference host can dial a programmed DTMF digit to lock the conference bridge so that no additional participants can join the conference. This provides a secured conference bridge.
- The conference host can selectively admit or reject participants from entering a conference by dialing a programmed DTMF digit.
- A conference organizer can add ports to the conference (if free ports are available) by accessing the web interface and editing the conference.
- The administrator can set the password length for all conference participants that use simple mode.
- Automatic Gain Control Settings for web interface.
- A conference organizer can force a participant to exit using the web interface.
- The conference network configuration is stored when firmware is updated.
- Refer to the Electra Elite IPK II Multimedia Conference Bridge manual for detailed information regarding this feature.
- This feature is now available when the IPK II Conference Bridge software is installed on the PVA(X)-U() ETU.

Description

The CNF(16)-U20 ETU is a Multimedia Conference Bridge that is used in the Electra Elite IPK II. This ETU can be configured as an 8-port or 16-port conference bridge. The Multimedia Conference Bridge is configured using an Internet Browser. The Login page allows user name and password access to the web browser. Conferences can be setup to send E-mail notification to each participant.

Simple Mode

When configured as an 8-port ETU the following conferences are available:

- One 8-party conference
- Two 4-party conferences (default)

When configured as a 16-port ETU the following conferences and conditions are available:

- One 16-party conference
- Two 8-party conferences (default)
- Three conferences (one 6-party and two 5-party)
- Four 4-party conferences
- Password protection for each conference
- Applicable voice messages and announcements (e.g., Entry, password request, exit)

Advanced Mode

- Password protection is provided for each conference.
- Applicable voice messages and announcements (e.g., Entry, password request, exit) are available.
- E-mail notification, when enabled, requires the organizer to enter the E-mail address of each participant to be sent notification of a pending conference. This option is selectable when setting up new conferences.
- Host Required, when enabled, requires the host/organizer to be logged into the conference before any other participant can enter. This option is selectable when setting up new conferences.
- Admission Control, when enabled, requires the organizer to dial a digit allowing each participant to enter the conference. This option is selectable when setting up new conferences (FUTURE).
- One customer greeting can be recorded for each CNF(16)-U20 ETU. Predefined password is necessary to record personal greetings.

Conditions

- Each CNF(16)-U20 ETU reduces the total number of station ports available in the Electra Elite IPK II system by eight or 16 depending on the assignment in Program 10-03-01 Unit Configuration.
- The CNF(16)-U20 ETU uses Digital Voice Mail (DVM) ports assigned in Program 10-03-01 Unit Configuration.
- The CNF(16)-U20 ETU shares the total number of available DVM ports with Voice Mail, Elite ACD Plus, CTI, and Elite Q-Master.

- The CNF(16)-U20 ETU is a standalone Conference Bridge that cannot be linked with another CNF ETU to increase the number of conferences.
- Conference ports should be placed in a Class of Service (COS) with no enabled alert tones.
- In Simple Mode, the call duration time starts when the last person joins the conference.
- A scheduled conference with nine or more participants automatically disables the participant announce feature.
- In Simple Mode, all participants have the same conference password. Default passwords are:
 - ❑ Conference 1 111111
 - ❑ Conference 2 222222
 - ❑ Conference 3 333333
 - ❑ Conference 4 444444
- If password is entered before the prompt finishes speaking, the first digit is not recognized. Enter password after the prompt is completed to prevent failed password entry.
- Custom Greeting is automatically enabled after the user accepts the newly recorded greeting.
- Entry or Exit announcement can be disabled during conference setup per conference. Participants are not requested to record name when Entry/Exit announcement is disabled.
- All passwords must have six digits.
- In Simple Mode, only the number of conferences, duration, password, and participant announce can be edited from the web browser interface.
- Edit options do not take effect until the conference is idle.
- When adding a CNF(16)-U20 ETU to a KSU, the slot must be assigned to match SW1 setting on the ETU or it does not initialize.
- A comparison between simple and advanced mode is shown below:
- DID calls to a Multimedia Conference Bridge Department Group Pilot are not answered if Pre-Ringing is enabled.

Feature	Simple Mode	Advanced Mode
Passwords	Per Conference	Per Participant
Number of Participants	Fixed, depending on CNF(16)-U20 ETU configuration	Selected by conference originator. From one to 15 participants can be accommodated.
E-Mail Notification	N/A	Selected by originator per conference.

Feature	Simple Mode	Advanced Mode
Admission Control	N/A	Selected by originator per conference (FUTURE).
Host Required	N/A	Selected by originator per conference.
Conferences Scheduled by Date and Time	N/A	Selected by originator per conference.
Custom Greeting	N/A	One custom greeting per CNF(16)-U20 ETU.
Entry/Exit Announcement Disable	N/A	Announcement can be disabled during conference setup.

Default Setting

None

System Availability

Terminals

All Terminals

Required Component(s)

CNF(16)-U20 ETU

-OR-

PVA(X)-U10 ETU with Required Service Pack and Application Firmware

Setup Procedures

After the Multimedia Conference Bridge ETU is installed in the KSU, the PC that is communicating with it must be configured to recognize the CNF ETU.

The default IP address of the CNF(16)-U20 ETU is:

IP Address: 192.168.1.100

Subnet Mask: 255.255.255.0

Gateway: 0.0.0.0

The PC and CNF ETU must be connected using either a straight Ethernet cable, crossover cable, or a small HUB. The CNF(16)-U20 ETU uses Medium Dependent Interface Crossover (MDIX) sensing protocol that allows the ETU to automatically sense the type (straight or crossover) of Ethernet cable used. To ensure communication between the PC and the ETU, they must be attached to the same network.

Related Features

None

Programming

- **11-02-01 Extension Numbering**
Set the extension number for each multimedia conference port.
- **11-07 Department Group Pilot Number**
Assign a pilot number to the Multimedia Conference bridge department group.
- **16-01 Department Group Basic Data Setup**
Assign basic function mode for each department group.
- **16-02 Department Group Assignment for Extensions**
Assign Extension to Department Groups.
- **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**
Allow a single line port to receive DTMF tones after the initial call setup, for voice mail always enter 1.

Operation

To join a conference using intercom:

1. Dial the Conference pilot number.
2. Dial conference password, and press #.
3. Record name at prompt, and press #.
4. Enter the conference.

To join a conference from an outside location:

1. Dial the number associated with the conference pilot number.
2. Dial conference password, and press #.
3. Record name at prompt, and press #.
4. Enter the conference.

Multiple Trunk Types

Description

The IPK II supports many different Trunks in the system (DID, E&M Tie Lines, Loop Start, Ground Start, ISDN BRI, ISDN PRI, and T-1 trunks). The system supports up to 200 trunks in the system, with the expanded port package, and a maximum of 56 trunks in the basic port package.

DID

Refer to the [Direct Inward Dialing \(DID\)](#) feature for related information.

E&M Tie Lines

E&M Tie Lines (4-Wire) can be connected to the system to provide communication between remote systems and facilities. The system can receive and/or transmit DTMF or DP signals on E&M Tie Lines.

Ground Start Trunks

Ground Start Trunks can be connected to the system. Ground and Loop Start Trunks can be mixed in the system per trunk. Ground Start Trunks are provided with line supervision to reduce call collisions. Refer to the following table to see which CO ETUs support Loop Start Trunks.

Loop Start Trunks

Loop Start Trunks can be connected to the IPK II system. Loop Start is assigned per trunk at the associated ETU. Ground Start and Loop Start Trunks can be mixed in the system per trunk. Refer to the following table to see which CO ETUs support Loop Start Trunks.

ETU	Caller ID*	CO Message Waiting Indication	Pad Control	Loop Start	Ground Start	Fax Branch
COI(4)-U10				X		X
COI(8)-U10				X	X	
COID(4)-U10	X			X		X
COID(8)-U10	X			X		
COIB(4)-U10	X			X	X	X
COIB(4)-U20	X		X	X		X
COIB(4)-U30	X	X	X	X		X
COIB(8)-U20	X		X	X		
COIB(8)-U30	X	X	X	X		

* Caller ID is not supported for Ground Start trunks.

ISDN BRI

Refer to the [ISDN Compatibility on page 2-635](#) feature for related information.

ISDN PRI

Refer to the [ISDN Compatibility on page 2-635](#) feature for related information.

T-1 Trunks

Refer to the [T1 Trunking \(with ANI/DNIS Compatibility\) on page 2-941](#) feature for related information.

Conditions

- Each TLI(2)-U() ETU supports two 4-wire E&M Tie Lines.
- Each COI(8)-U() ETU has eight switches (one for each port) that can be used to select either Loop or Ground Start per trunk.
- Each COIB(4)-U10 ETU has four switches (one for each port) that can be used to select either Loop or Ground Start per trunk.
- Ground Start Trunks do not support Caller ID.

Default Setting

None

System Availability

Terminals

All Terminals

Required Component(s)

Any Trunk ETU

Related Features

Automatic Route Selection

Call Appearance (CAP) Keys

Caller ID

Direct Inward Dialing (DID)

ISDN Compatibility

T1 Trunking (with ANI/DNIS Compatibility)

Programming

Loop Start/Ground Start Trunks

- **14-04-01 Behind PBX Setup – Type of Connection**
Indicate if the trunk is installed behind a PBX (1) or not (0). There is one item for each Night Service Mode.
- **14-05-01 Trunk Group – Trunk Group Number**
Assign trunks to Trunk Groups. You can also assign the outbound priority for trunks in the group. When users dial up the trunk group, they seize the trunks in the order you specify in the outbound priority entry.

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- **22-02-01 Incoming Call Trunk Setup**
For each Night Service mode, enter service type for each trunk.
 - **81-10-01 COI Initial Data Setup – DP Interdigit Time Selection**
Select the DP Interdigit Time (minimum pause time between Dial Pulses).
[default: 1, (Pattern B)]
 - **81-10-02 COI Initial Data Setup – Prepause Time Selection**
Specify the loop open time for a hookflash signal sent to the CO or PBX when the Recall key on a multiline terminal is pressed. A single line telephone generates a hookflash to the CO or PBX line when a single line telephone hookflash is assigned.
[default: 9, (600 ms)]
 - **81-10-03 COI Initial Data Setup – Incoming Signal Detect Time Selection**
Specify the time after the incoming signal from another system is detected before the acknowledge signal is sent out.
[default: 3 (200 ms)]
 - **81-10-04 COI Initial Data Setup – Disconnect Recognition Time Selection**
Specify the minimum time before a disconnected circuit can be accessed again.
[default: 3 (200 ms)]
 - **81-10-05 COI Initial Data Setup – Auto Release Signal Detection Time**
Specify the signal detection time for release of a CO/PBX line after a disconnect signal is received from the distant Central Office or PBX.
[default: 7 (350 ms)]

Tie Lines

- **10-09-01 DTMF and Dial Tone Circuit Setup**
If the system has DTMF Tie Lines, be sure to reserve at least one circuit for analog trunk DTMF reception (type 0 or 2).
 - Use the following as a guide when allocating DTMF receivers:
 - ❑ In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - ❑ In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.
- **14-01-02 Basic Trunk Data Setup – Transmit Level**
- **14-01-03 Basic Trunk Data Setup – Receive Level**
Customize the transmit and receive levels of the CODEC Gain Types for each trunk as required.
[default: 32 (0dB)]
- **14-01-13 Basic Trunk Data Setup – Trunk-to-Trunk Transfer**
Enable (1) loop supervision for each Tie Line that should be able to place outgoing calls.
- **14-02-01 Analog Trunk Data Setup – Signaling Type (DP/DTMF)**
Set the outgoing signaling type for the tie trunk. The options are 0 (DP 10 pps), 1 (DP 20 pps) and 2 (DTMF). To set incoming signaling, refer to Program 34-01-02.
(default: 2)

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- ➔ **14-05-01 Trunk Group – Trunk Group Number**
 Program Tie Lines of similar type into the same trunk group. The system uses trunk groups for outgoing access to Tie Lines (i.e., Service Code 704 + group). Also see Program 34-05-01.
 Options:
 Trunk group numbers 1~200
 Priority: 1~200
 (default: All ports in Group 1; Trunk Port 1~200 = Priority 1~200)

 - ➔ **14-06-01 Trunk Group Routing – Priority Order Number**
 When a Tie Line user dials 9, the system uses the routes defined in this program. Refer to Program 34-03-01.
 Options:
 Priority Order Number 1~4
 0 = Not Specify
 1~100 Trunk Groups
 1001~1100 (1000 + Route Table No).
 [default: Route 1, Order Number 1 =Trunk Group 1;Order Numbers 2~4 = 0; All other routes (2~100) and Order numbers 1~4 = 0]

 - ➔ **20-01-05 System Options – DTMF Receive Active Time**
 After answering the Tie Line call, the system attaches a DTMF receiver to the Tie Line for this time (0~64800 seconds).
 (default: 10)

 - ➔ **20-06-01 Class of Service for Extensions**
 Assign a Class of Service (1~15) to extensions.
 (default: Extension 101 in Class 15 – All others in Class 1)

 - ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
 Enable (1)/Disable (0) an extension ability to receive a second call from a DID, DISA, DIL, or Tie Line caller.
 (default: 1)

 - ➔ **20-14-01 through 20-14-12 Class of Service Options for DISA/E&M**
 Enable (1)/Disable (0) the following options for each Tie Line Class of Service (1~15):
 - 01 First Digit Absorption (Delete First Digit Dialed)
 - 02 Trunk Group Routing/ARS Access
 - 03 Trunk Group Access
 - 04 Outgoing System Speed Dial
 - 05 Operator Calling
 - 06 Internal Paging
 - 07 External Paging
 - 08 Direct Trunk Access
 - 09 Forced Trunk Disconnect <Not for ISDN T-point>
 - 10 Call Forward Setting by Remote via DISA
 - 11 DISA/Tie Trunk Barge-In
 - 12 Retrieve Park Hold
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- ➔ **21-05-01 Toll Restriction Class – International Call Restriction Table**
 - ➔ **21-05-02 Toll Restriction Class – International Call Permit Code Table**
 - ➔ **21-05-04 Toll Restriction Class – Maximum Number of Digits Table Assignment**
 - ➔ **21-05-05 Toll Restriction Class – Common Permit Code Table**
 - ➔ **21-05-06 Toll Restriction Class – Common Restriction Table**
 - ➔ **21-05-07 Toll Restriction Class – Permit Code Table**
 - ➔ **21-05-08 Toll Restriction Class – Restriction Table**
 - ➔ **21-05-09 Toll Restriction Class – Restriction for Common Speed Dials**
 - ➔ **21-05-10 Toll Restriction Class – Restriction for Group Speed Dials**
 - ➔ **21-05-11 Toll Restriction Class – Intercom Call Restriction**
 - ➔ **21-05-12 Toll Restriction Class – PBX Call Restriction**
If Program 34-01-05 is set to (1), define the toll restriction settings for Tie Lines.
 - ➔ **21-05-13 Toll Restriction Class – Restriction of Tie Line Calls**
Enable (1)/Disable (0) toll restriction for Tie Line calls (defined in Program 34-08-01).
 - ➔ **22-02-01 Incoming Call Trunk Setup**
For each Night Service mode, enter service type 5 when the trunk should be a tie trunk.
 - ➔ **34-01-01 E&M Tie Line Basic Setup – DID/E&M Start Signaling**
Enter 2nd dial tone (0), Wink (1), Immediate (2), or Delay (3) start signaling. To set outgoing signaling, refer to Program 14-02-01
(default: 1)
 - ➔ **34-01-03 E&M Tie Line Basic Setup – E&M Dial-In Mode**
Determine if the incoming Tie Line call should be directed as an intercom call (0) or if it should follow the DID Translation Table in Program 22-11-01 (1).
(default: 0)
 - ➔ **34-01-04 E&M Tie Line Basic Setup – E&M Line Dial Tone**
Enable (1)/Disable (0) the Tie Line dial tone.
(default: 1)
 - ➔ **34-01-05 E&M Tie Line Basic Setup – System Toll Restriction**
Turn Off (0) or On (1) the System Toll Restriction for Tie Line calls to follow the system toll restriction entries in Program 21-05-01 through Program 21-05-13. If Off, Tie Line toll restriction is determined by PRG 21-05-13.
(default: 0)
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- ➔ **34-02-01 E&M Tie Line Class of Service**
Assign the Tie Line Class of Service (1~15). Use Program 20-14-01 to set the Tie Line Class of Service options. You cannot use Program 20-06 ~ Program 20-17 to assign Class of Service to Tie Lines.
(default: 1)

 - ➔ **34-03-01 Trunk Group Routing for E&M Tie Lines**
Assign the trunk group route chosen when a user seizes a Tie Line and dials 9. Set Trunk Group Routing in Program 14-06-01. If the system has ARS, dialing 9 accesses ARS.
Options:
Route Table Number (0~100
0 = No Setting
(default: 1)

 - ➔ **34-04-01 E&M Tie Line Toll Restriction Class**
If the system uses Toll Restriction, enter a Toll Restriction Class (1~15) for each Tie Line. The system uses the class you enter in Program 21-05-01. You cannot use Program 21-04 to assign Toll Restriction to Tie Lines.
(default: 2)

 - ➔ **34-05-01 Tie Line Outgoing Call Restriction**
Build a restriction matrix for trunk calls placed over a Tie Line. For each Tie Line trunk group, Enable (0)/Disable (1) outgoing access to each CO trunk group.
(default: 0)

 - ➔ **34-06-01 Add/Delete Digit for E&M Tie Line – Delete Digit**
Some Tie Line networks pass the location number and extension number to the remote side. If the system should ignore these digits, use this program to define the number of digits which should be deleted (0~255, 255 deletes all digits) for a call.
(default: 0)

 - ➔ **34-06-02 Add/Delete Digit for E&M Tie Line – Additional Dial Digits**
If a Tie Line network requires additional digits to reroute the call to a location, enter the digits for the location which should be added to the received digits (up to four digits).
(default: No Setting)

 - ➔ **34-07-01 through 34-07-05 E&M Tie Line Timer**
Define the system timers (0~64800 seconds) as needed for use with the Tie Line trunks
(default: 3 for 01, 0 for 02, 3 for 03, 0 for 04, and 30 for 05)

 - ➔ **34-08-01 Toll Restriction Data for E&M Tie Lines**
Define the toll restriction data (up to 10 digits) for E&M Tie Lines if required. This should be defined if toll restriction is enabled in Program 21-05-13.
(default: No Setting)

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- ➔ **80-03-01 through 80-03-08 DTMF Tone Receiver Setup**

 - ➔ **80-04-01 through 80-04-09 Call Progress Tone Detector Setup**
If required, modify the criteria for dial tone detection and call progress tone detection for Tie Line calls.

 - ➔ **81-11-01 Tie Line Initial Setup – DP Interdigit Time Selection**
Select the DP Interdigit Time (minimum pause time between Dial Pulses).
[default: 1 (Pattern B)]

 - ➔ **81-11-02 Tie Line Initial Setup – Prepause Time Selection**
Specify the loop open time for a hookflash signal sent to the Tie Line when the Recall key on a multiline terminal is pressed. A single line telephone generates a hookflash to the Tie Line when a single line telephone hookflash is assigned.
(default: 0)

 - ➔ **81-11-03 Tie Line Initial Setup – Tie Line Answer Detect Time Selection**
Specify the time before a Post IPK System answer (off-hook) is recognized as an answer.
[default: 4 (520 ms)]

 - ➔ **81-11-04 Tie Line Initial Setup – Tie Line Release Detect Time Selection**
Specify the before the circuit disconnect detected on the Tie Line on the distant system side is recognized as Tie Line release.
[default: 4 (520 ms)]

 - ➔ **81-11-05 Tie Line Initial Setup – Incoming Signal Detect Time Selection**
Specify the time after the incoming signal from another system is detected before the acknowledge signal is sent out.
[default: Wink 3(390 ms) ;Delay 3(90 ms)]

 - ➔ **81-11-06 Tie Line Initial Setup – Loop Off-Guard Time Selection**
Assign the loop off-guard time to prevent noise that could cause the system to be unable to answer an incoming Tie Line.
[default: 4 (2.0 sec.)]

 - ➔ **81-11-07 Tie Line Initial Setup – Length of Wink Signal Selection**
Specify the time a wink pulse is sent to another system.
[default: 5 (180 ms)]

 - ➔ **81-11-08 Tie Line Initial Setup – Length of Delay Signal Selection**
Specify the time a delay pulse is sent to another system.
[default: 1 (300 ms)]

 - ➔ **81-11-09 Tie Line Initial Setup – Incoming Interdigit Timeout Selection**
Specify the time that an address signal is missing during the incoming call detection process before an error tone is returned to the other system.
(default: 6 seconds)
-
-

- ➔ **81-11-10 Tie Line Initial Setup – Wink/Delay Signal Detect Timeout Selection**
Specify a maximum time for receiving an acknowledgment signal from a distant system before sending a busy tone.
(default: 7 seconds)
- ➔ **81-11-11 Tie Line Initial Setup – Disconnect Recognition Time Selection**
Specify the minimum time before a disconnected circuit can be accessed again.
[default: 3 (0.3 seconds)]
- ➔ **81-11-12 Tie Line Initial Setup – Automatic Release Signal Detection Selection**
Specify the signal detection time for release of a Tie Line after a disconnect signal is received from the distant Central Office or PBX.
[default: 7 (350 ms)]

Operation

None

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Music on Hold

Description

Music on Hold (MOH) sends music to calls on Hold and parked calls. The music lets the caller know that his call is waiting, not forgotten. Without Music on Hold, the system provides silence to these types of calls. The Music on Hold source can be internal (tone) or from an external customer-provided music source (i.e., tape deck, or receiver). The customer-provided source can connect to a PGD(2)-U10 ADP analog port or to a connector on the side of the cabinet.

Option Available for Using System Tone

The Music on Hold feature is enhanced to allow callers to hear a system tone instead of playing the internal or external music.

 *In accordance with U.S. copyright law, a license may be required from the American Society of Composers, Authors and Publishers (ASCAP) or other similar organizations, if radio, television broadcasts or music other than material not in the public domain are transmitted through the Music on Hold feature of telecommunications systems. NEC America, NEC Unified Solutions, Inc., and NEC Infrontia hereby disclaim any liability arising out of the failure to obtain such a license.*

Conditions

- A maximum of 97 Music on Hold sources are possible; 96 from PGD(2)-U10 ports and one from the side of the KSU.
- External music on hold source for internal calls is provided only via audio input on the side of the KSU. Program 10-04-01 must be set to 1 (External Source).
- No music is provided to internal calls on hold via the ACI input.
- Use the combination of PRG 10-04, PRG 10-38 and PRG 14-08.
- The PGD(2)-U10 ADP can be connected only to an ESIB(8)-U20 ETU. If the PGD(2)-U10 ADP is connected to an ESIE(8)-U20 ETU, the PGD(2)-U10 ADP module is NOT recognized.
- When a ESIB(8)-U20 ETU has an ESIE(8)-U20 ETU installed [ESIB/E(16)], the PGD(2)-U10 ADP has both channels available for Paging, MOH, External Ringers and Door Box according to strapping on ports 1~8. The corresponding port on the ESIE(8)-U20 ETU does not support a station or other hardware (it is used by the PGD(2)-U10 ADP for channel 2).
- When a ESIB(8)-U20 ETU is installed without an ESIE(8)-U20 ETU installed, a PGD(2)-U10 ADP installed on ports 1~8 is recognized, but only channel 1 on the PGD(2)-U10 ADP is supported. Channel 2 is not available.

- The PGD(2)-U10 ADP can be connected only to an ESIB(8)-U20 ETU. If the PGD(2)-U10 ADP is connected to an ESIE(8)-U20 ETU, the PGD(2)-U10 ADP is NOT recognized.
- When an ESIB(8)-U20 ETU has an ESIE(8)-U20 ETU installed [ESIB/E(16)], the PGD(2)-U10 ADP has both channels available for Paging, Music on Hold, External Ringers and Door Box according to strapping on ports 1~8. The corresponding port on the ESIE(8)-U20 ETU does not support a station or other hardware (it is used by the PGD(2)-U10 ADP for channel 2).
- When an ESIB(8)-U20 ETU is installed without an ESIE(8)-U20 ETU installed, a PGD(2)-U10 ADP installed on ports 1~8 is recognized, but only channel 1 on the PGD(2)-U10 ADP is supported. Channel 2 is not available.
- ESIB(8)-U10 ETUs and ESIE(8)-U10 ETUs do not support the PGD(2)-U10 ADP.
 - 📎 A maximum of 56 PGD(2)-U10 ADP units can be installed in an IPK II system. Refer to the Electra Elite IPK II System Hardware Manual for more information.

Default Setting

Disabled

System Availability

Terminals

None

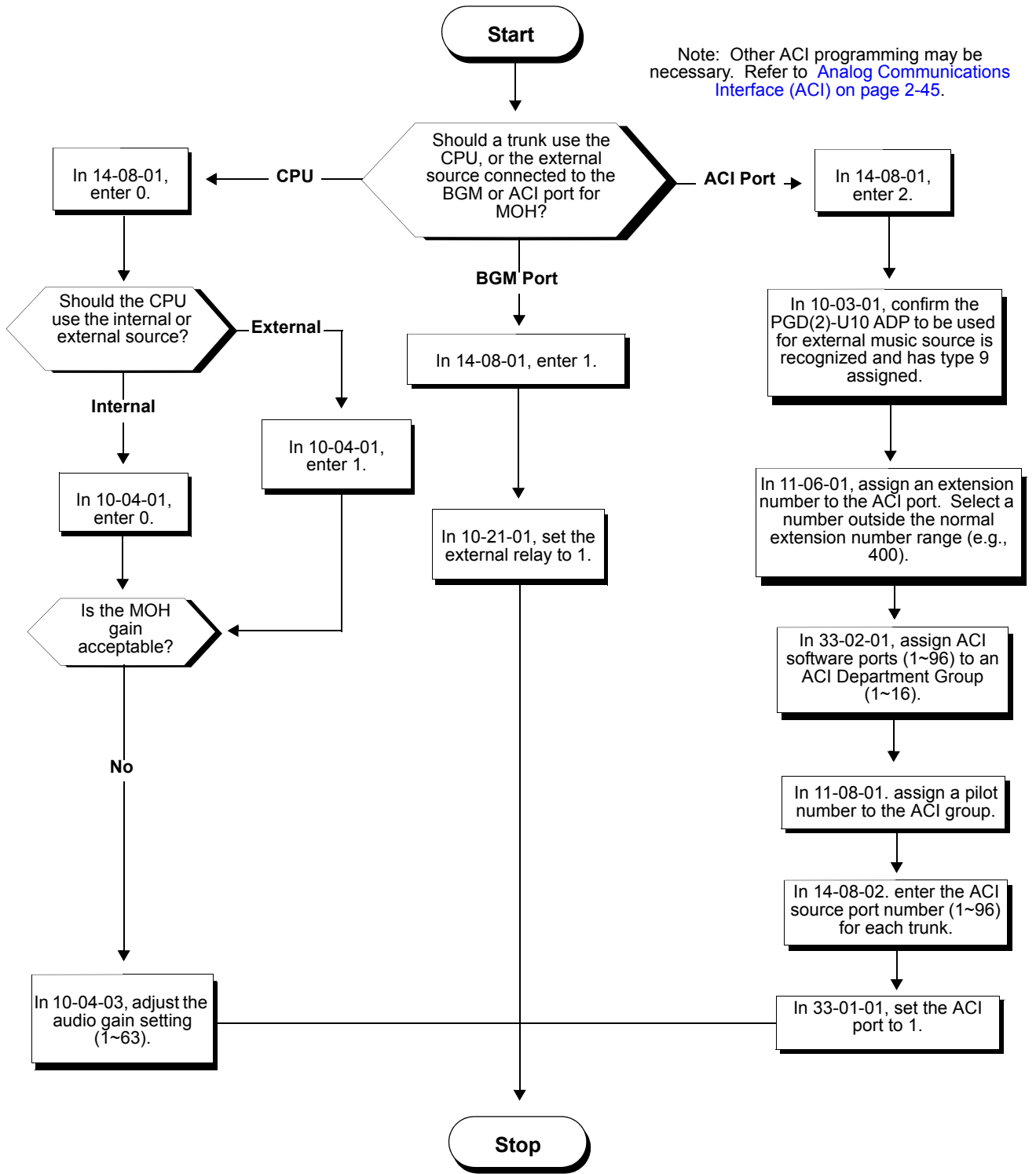
Required Component(s)

PGD(2)-U10 ADP

Related Features

None

Programming




-
-
- ➔ **10-04-01 Music on Hold Setup – Music on Hold Source Selection**
Determine whether the system should use No Tone (0), External (1) Music Source, or Internal Service Tone (2). If set to 1, Program 14-08-01 must be set to 0 or 1.
(default: 2)
 - ➔ **10-04-03 Music on Hold Setup – Audio Gain Setup**
Set the Music on Hold audio gain (1~63).
[default: 32 (0dB)]
 - ➔ **14-08-01 Music on Hold Source for Trunks – MOH Type**
For each trunk, set the Music on Hold source.
0 = Internal/External MOH
1 = Customer Provided Source Connected to BGM Port
2 = Customer Provided Source Connected to ACI Port
(default: 0)
 - ➔ **14-08-02 Music on Hold Source Port Number – Source Port Number**
If the MOH type is 2 in PRG 14-08-01, for each trunk enter the ACI source port number (0~96).
(default: 0)
 - ➔ **33-01-01 ACI Port Type Setup**
If ACI software port is designated for MOH in PRG 14-08-01, set port function to 1 (input).
(default: 2)
 - ➔ **80-01-01 Service Tone Setup, Music On Hold Tone (Service Tone 64)**
If required, change the tone heard by the user when placed on hold when Program 10-04-01 set to 2.
[default: 32 (0dB)]

When Using a PGD(2)-U10 ADP:

- ➔ **10-03-01 ETU Setup – Terminal Type (B1) for ESI PKG**
If a PGD(2)-U10 ADP is used for the external music source, the module is automatically assigned type 9 if the jumper straps in the module are set prior to connecting it to the system. If another type is assigned, disconnect the PGD(2)-U10 ADP from the system, delete the type setting, and, with the jumper straps positioned correctly in the PGD(2)-U10 ADP, reconnect the module to the system. Refer to the Electra Elite IPK II System Hardware Manual for the jumper strap settings.
- ➔ **11-06-01 ACI Extension Numbering**
Each ACI port must be assigned an extension number. Use this program to assign the extension numbers to ACI software ports. Select a number outside of the normal extension number range.

ACI Ports: 1~96
(default: No extensions assigned)
- ➔ **11-08-01 ACI Group Pilot Number**
Assign pilot numbers to ACI groups. When a user dials the pilot number, they reach an available ACI software port in the group.

ACI Groups: 1~16
(default: No Setting)

- ➔ **33-01-01 ACI Port Type Setup**
Set each ACI software ports for input (1) or input/output (2). Use input ports for Music on Hold sources. Use output ports for External Paging/Ringer Control.
- ACI Ports: 1~96
(default: 2)
- ➔ **33-02-01 ACI Department Calling Group**
Assign ACI software ports to an ACI Department Group. This lets ACI callers connect to ACI software ports by dialing the group pilot number (set in Program 11-08).
- ACI Ports: 1~96
ACI Groups: 1~16
Priority: 1~96
-  Refer to [Analog Communications Interface \(ACI\) on page 2-45](#) for additional information.
- (default: All ports assigned to Group 1; ports 01~96 have priority 01~96)
- ➔ **80-01-64 Service Tone Setup – Music on Hold**
Refer to [Music on Hold on page 2-735](#).
- ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
(Default: 0).
Refer to [Table 2-27 Service Tone Setup, Program 80-01-02 on page 2-480](#).

Operation

None

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Name Storing

Enhancements

With **Version 1500 or higher**, telephones that are not connected or uninstalled, do not show up in the Extension Directory.

With **Version 1600 or higher**:

- Station Number and Name can be removed from the display when the phone is idle based on new Class of Service options (PRG 20-13-47 and PRG 20-13-48). Prior to this release, the station Number and Name were always displayed when the phone was idle.
- Extensions (including Virtual Extensions) can be removed from the Extension Directory list based on a new Class of Service option (PRG 20-13-51). Prior to this release, installed extensions and Virtual Extensions with a Name Assigned were always displayed in the Extension Directory list.
- Allows a Class of Service option to remove extensions from the Extension Directory of the telephones. This applies to both physical and Virtual Extensions.
- Allows a Class of Service option to remove the Station Number and Name from the display when the telephone is idle.

A red vertical bar on the right side of the page, containing a white capital letter 'N'.

Description

Extensions and trunks can have names instead of just circuit numbers. These names show on a multiline terminal display when the user places or answers calls. Extension and trunk names make it easier to identify callers. The user does not have to refer to a directory when processing calls. A name can have up to 12 digits consisting of alphanumeric characters, punctuation marks and spaces.

Additional Characters Available

When using the Name Storing feature, the system now provides additional characters which can be used. These characters are available with any option which allows Name Storing - Speed Dial – System/Group/Station, One-Touch Keys, Extension Name, Trunk Naming.

Conditions

- Display telephones use extension names for Directory Dialing.
- Single line extension users cannot program names.

- If a name is not assigned to the Extension/Virtual Extensions, it does not show in the Extension Directory.
- Extension Directory only shows telephones/virtual extensions that have a name assigned in PRG 15-01-01.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals with Display

Required Component(s)

None

Related Features

Directory Dialing

Single Line Telephones, Analog 500/2500 Sets

Speed Dial – System/Group/Station

Programming

- ➔ **14-01-01 Basic Trunk Data Setup – Trunk Name**
Program names for system trunks.
- ➔ **15-01-01 Basic Extension Data Setup – Extension Name**
Define the extension/Virtual extension name (up to 12 characters) for system extensions (default: STA 101 = Ext 101; STA 102 = Ext 102, etc.)
- ➔ **20-09-02 Class of Service Options (Incoming Call Service) – Caller ID Display**
Enable (1)/Disable (0) the displaying of Caller ID.
(default: 0)

- ➔ **20-13-47 Class of Service Options (Supplementary Service) – Station Number Display (Version 1600 or higher required)**
Enable (1)/Disable (0) the Station Number Display when the telephone is idle.
(default: 1)
- ➔ **20-13-48 Class of Service Options (Supplementary Service) – Station Name Display (Version 1600 or higher required)**
Enable (1)/Disable (0) the Station Name Display when the telephone is idle.
(default: 1)
- ➔ **20-13-51 Class of Service Options (Supplementary Service) – Number and Name appear in the Directory (Version 1600 or higher required)**
Enable (1)/Disable (0) the displaying of Station Name and Number in the Directory.
(default: 1)

Operation

Refer to [Table 2-29 Keys for Entering Names](#) for and explanation for using the keypad to enter names.

Table 2-29 Keys for Entering Names

Use this keypad digit . . .	When you want to . . .
1	Enter characters: 1 @ [¥] ^ _ ` { } Æ " Á À Â Ã Ç É Ê Ì Ó
2	Enter characters: A-C, a-c, 2.
3	Enter characters: D-F, d-f, 3.
4	Enter characters: G-I, g-i, 4.
5	Enter characters: J-L, j-l, 5.
6	Enter characters: M-O, m-o, 6.
7	Enter characters: P-S, p-s, 7.
8	Enter characters: T-V, t-v, 8.
9	Enter characters: W-Z, w-z, 9.
0	Enter characters: 0 ! “ # \$ % & ’ () ô Õ ú ä ö ü α ε θ
*	Enter characters: * + , - . / : ; < = > ? π Σ σ Ω ∞ € £
#	# = Accepts an entry (only required if two letters on the same key are needed – ex: TOM). Pressing # again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.)
CONF	Clear the character entry one character at a time.
HOLD	Clear all the entries from the point of the flashing cursor and to the right.

Night Service

Description

Night Service lets system users activate one of the Night Service modes. Night Service redirects calls to their night mode destination, as determined by Assigned and Universal Night Answer programming. A user typically activates Night Service after normal working hours, when most employees are unavailable to answer calls. The system also provides external contacts to enable Night Service.

- There are eight Service Modes. At default, the mode names are assigned as follows:
 - Mode 1 = No setting
 - Mode 2 = Night
 - Mode 3 = Midnight
 - Mode 4 = Rest
 - Mode 5 = Day2
 - Mode 6 = Night2
 - Mode 7 = Midnight2
 - Mode 8 = Rest2

There are 32 Service Patterns/Groups available.

Assigned Night Answer (ANA)

With Assigned Night Answer (ANA), Night Service has calls ring extensions directly. Assigned Night Answer provides an answering point for Night Service calls. For certain applications, this may be more appropriate than Universal Night Answer. For example, you could program trunks to ring the security station telephone during off hours.

Universal Night Answer (UNA)

Universal Night Answer makes incoming calls ring over the External Paging speakers. With UNA, an employee can go to a telephone and press the flashing line key or use Universal Answer to pick up the call. Only ring groups calls can be used with Universal Night Answer. For more on setting up Universal Answer, refer to [Central Office Calls, Answering on page 2-187](#).

You may also use Transfer to UNA. An extension user can transfer their call to UNA (i.e., External Paging at night). Once transferred, the call rings the External Paging speakers like any other UNA call and can be picked up at any extension. You can also set up Transfer to UNA through the Voice Response System (VRS). This lets outside callers, answered by the VRS, dial a code to have their call ring External Paging.

Automatic Night Service

The system allows or denies Automatic Night Service. If allowed, the calls route according to the service patterns programmed. The Night Service programming is stored in the RAM memory. This means that if the system is not using the Automatic Night Service, for a power failure while in night mode, when the power is restored, the system continues to be in night mode.

Programmable Function Key Can Toggle Night Modes

The software allows a Night Service Programmable Function Key (PRG 15-07-01 or SC 751: 09 + 0) to toggle night modes. You can determine in programming (PRG 12-08-01) how many modes through which the user will toggle. Note that the additional data for the Programmable Function Key must be set to 0 for the toggle function to work.

Conditions

- Almost all features are affected by Night Mode except for the following:
 - Dial Tone Detection
 - External Alarm Sensors
 - Flexible System Numbering
 - Pulse to Tone conversion
 - SMDR
 - Volume Control
- Call Arrival (CAR) Keys and Virtual Extension keys do not support Day/Night Mode (09) Programmable Function keys.
- Universal Night Answer works only when a Call is sent to a ring group.
- Separate Access Map and Ring Group programming entries are available for each Night Service mode (modes 1~8). Also, Universal Answer allows an extension user to pick up a Universal Night Answer (UNA) call.
- Mode Keys can be assigned as required for DSS Consoles.
- With Universal Night Answer, outside calls can ring External Paging Zones.
- Programmable Function Keys simplify activating Night Service.
- The relay circuits (5~8) are on the PGD(2)-U10 ADP are programmed and used for General Purpose Relays.
- When programming Night Service function keys, multiple keys must be used for switching between each Night Service Mode.

Default Setting

System is always in Mode 1

System Availability

Terminals

Not Applicable

Required Component(s)

None

Related Features

Central Office Calls, Answering

Central Office Calls, Placing

Direct Station Selection

Paging, External

Programmable Function Keys

Ring Groups

Programming

- ➔ 10-05-01 **General Purpose Relay Setup**
- ➔ 11-10-01 **Service Code Setup (for System Administrator) – Night Mode Switching**
Customize the service code (718) used for day/night mode switching.
- ➔ 11-10-12 **Service Code Setup (for System Administrator) – Night Mode Switching for Other Group**
Customize the service code (618) used for Day/Night mode switching for another Night service group.

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-
- ➔ **11-12-43 Service Code Setup (for Service Access) – Answer No-Ring Line (Universal Answer)**
Customize the service code (#0) used to manually answer a Universal Night Answer.
 - ➔ **11-12-50 Service Code Setup (for Service Access) – General Purpose Relay**
Define the service code used for turning the general purpose relay on and off (default = 780).
 - ➔ **12-01-01 Night Mode Function Setup – Manual Night Mode Switching**
Allow (1)/Prevent (0) any extensions from activating Manual Night Service.
 - ➔ **12-01-02 Night Mode Function Setup – Automatic Night Mode Switching**
Enable (1)/Disable (0) Automatic Night Service for the system. Make sure to set the Service Patterns in PRG 12-02-01, PRG 12-02-02 and PRG 12-02-03.
 - ➔ **12-02-01 Automatic Night Service – Start Time**
For each Night Service Group, enter up to 20 start times for each Time Pattern (1~10).
The pattern first start time (Pattern 1) should begin at 00:00 (midnight).
 - ➔ **12-02-02 Automatic Night Service – End Time**
For each Night Service Group (01~32), enter up to 20 end times (0000~2359) for each Time Pattern (1~10).
 - ➔ **12-02-03 Automatic Night Service – Operation Mode**
For each Night Service Group (01~32), define the Night Service Mode (1~8) for up to 20 start/end times for each Time Pattern (1~10).
 - ➔ **12-03-01 Weekly Night Service Switching**
Assign one of the 10 Time Patterns programmed in PRG 12-02-01 to each day of the week.
 - ➔ **12-04-01 Holiday Night Service Switching**
Assign one of the 10 Time Patterns to holidays.
(default: No Setting)
 - ➔ **12-05-01 Night Mode Group Assignment for Extensions**
Assign a Day/Night Mode Group (01~32), for each extension.
(default: 1)
 - ➔ **12-06-01 Night Mode Group Assignment for Trunks**
Assign a Day/Night Mode Group (01~32), for each trunk port (1~200).
(default: 1)
 - ➔ **12-07-01 Text Data for Night Mode**
Create an original text message (up to 12 characters) which is displayed on an LCD of multiline terminal in each Night Mode.
 - ➔ **12-08-01 Night Mode Service Range**
For each Night Mode Group (01~32), determine how many night modes (0~9) a user toggles through when the Night Mode key is pressed.
(default: 2)
 - ➔ **14-07-01 Trunk Access Map Setup**
To allow for Universal Night Answer (UNA) answering, set up the Trunk Access Maps (1~200).
For UNA, extension must have incoming access to trunk ringing the External Paging speakers.

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-
- ➔ **15-06-01 Trunk Access Map for Extensions**
For Universal Night Answer (UNA) answering, assign Trunk Access Maps (1~200) to extensions. Make one entry for each Night Service mode.

 - ➔ **15-07-01 Programmable Function Keys**
Assign Night Service function keys (09) to extensions and set the key for the proper mode (Day, Night, Rest, etc.). If the additional data is set to 0, the toggle mode is assigned.

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

 - ➔ **20-07-01 Class of Service Options (Administrator Level) – Manual Night Service Enabled**
Enable (1)/Disable (0) an extension user ability to manually switch the Night Mode (Service Code 718). This option must be enabled for an extension to display the Night indication.
[default: 0 (COS 1 ~14); 1(COS 15)]

 - ➔ **20-10-07 Class of Service Options (Answer Service) – Automatic Off-Hook Answer**
Allow (1)/Deny (0) Universal Auto Answer.
(default: 0)

 - ➔ **22-02-01 Incoming Call Trunk Setup**
Assign the incoming trunk type for each trunk. There is one item for each Night Service Mode (1~8).
[default: 0 (Normal)]

 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
To have trunks ring extensions during the different Night Service modes (for ANA), assign extensions to Ring Groups. For each extension in the Ring Groups (1~100), indicate in Program 22-06-01 if trunk should ring (1) or not ring (0).
[default: Ext. 101~108 (first 8 ports) ring for incoming Ring Group 1 calls. No other extensions ring for Ring Group1 calls.

 - ➔ **22-05-01 Incoming Trunk Ring Group Assignment**
To have trunks ring extensions for ANA, assign trunks to Ring Groups (1~100), You make a different entry for each Night Service mode.
Ring Groups:
1~100
101 = Not used
102 = In-Skin/External Voice Mail
103 = Not Used
(default: 1)

 - ➔ **22-08-01 DIL/IRG No Answer Destination**
If a Universal Answer call rings longer than the DIL No Answer Time (Program 22-01-04), it routes to the Ring Group specified in this option.
Ring Groups:
1~100
101 = Not used
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
(default: 1)
-
-

-
-
- ➔ **31-05-01 Universal Night Answer/Ring Over Page**
For each Night Service Mode, assign which trunks should ring which External Paging Zones.

Operation



To activate Night Service by dialing codes:

1. At a Multiline Terminal, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **718**. To change a different group's mode, dial **718** + the group number (01~32).
3. Dial the Night Service Code:
1 = Day 1 Mode
2 = Night 1 Mode
3 = Midnight 1 Mode
4 = Rest 1 Mode
5 = Day 2 Mode
6 = Night 2 Mode
7 = Midnight 2 Mode
8 = Rest 2 Mode
4. Press **Speaker** or hang up.

To activate Night Service by using programmable keys:

1. Press **Night Service** key (PRG 15-07-01 or SC 751:09 + Mode code number below).
1 = Day 1 Mode
2 = Night 1 Mode
3 = Midnight 1 Mode
4 = Rest 1 Mode
5 = Day 2 Mode
6 = Night 2 Mode
7 = Midnight 2 Mode
8 = Rest 2 Mode

To transfer a call to the Universal Answer External Page zones:

1. Place the CO call on hold and dial the Transfer to Trunk Ring Group code (assigned in PRG 11-15-09).
 *You hear a confirmation tone.*
2. Hang up.
 *The call rings over the External Paging, enabling anyone to answer the call.*

Off-Hook Signaling

Description

Off-Hook ringing alerts a Multiline Terminal user that an incoming outside call is ringing to that station during another call. Off-Hook Signaling helps important callers get through, without waiting in line for the called extension to become free. The system provides the following Off-Hook Signaling options:

Called Extension Block

The called extension Class of Service may block incoming Off-Hook Signaling attempts. This is beneficial to users that do not want interruptions while on a call.

Automatic Signaling

Calling a busy extension automatically initiates Off-Hook Signaling. This option is useful to receptionists, operators and others that must quickly process calls. This is set in the called extension Class of Service.

Manual Signaling

After reaching a busy extension, manual signaling gives the caller the choice of using Off-Hook Signaling or activating other features. Extensions without automatic signaling have manual signaling. The users can dial a service code or press a Programmable Function Key to send Off-Hook Signaling to the called telephone.

Selectable Off-Hook Signaling Mode

The Off-Hook Signal can be muted ringing, no off-hook ringing or a beep in the handset - based on the caller's programming.

Off-Hook Ringing

Use this option to enable/disable an extension Off-Hook Signaling for incoming calls. If enabled, Off-Hook Signaling occurs normally. If disabled, calls queue behind the extension busy line appearance and the user gets no Off-Hook Signaling indication. The second line appearance stays idle. The caller hears ringback tone while their call waits. This is set in the called extension Class of Service.

DID Call Waiting

An extension can optionally have DID calls camp-on with Off-Hook/Call Wait signaling, without Off-Hook/Call Wait signaling or no signaling. This is set in the called extension Class of Service.

❑ Block Manual Off-Hook Signals

This Class of Service option enables/disables a busy extension user ability to block off-hook signals manually sent from a co-worker. If disabled (not blocked), callers can dial * at busy or busy/ring to signal the extension. If enabled (blocked), nothing happens when the caller dials * to off-hook signal.

❑ Block Camp-On

If an extension has Block Camp-On enabled, callers to the extension cannot dial 2 to Camp-On after hearing busy or busy/ring. If the extension has Block Camp-On disabled, callers are not prevented from dialing 2 to Camp-On after hearing busy or busy/ring. This is set in the called extension Class of Service.

Conditions

- An extension user cannot Camp-On to a busy extension or leave a callback if Off-Hook Signaling has already gone through. Off-Hook Signaling allows an extension to block a caller's ability to dial # to camp-on.
- You cannot send off-hook signals to an extension busy on a Handsfree (Speakerphone) call. The called extension large LED flashes fast, with no ringing.
- The setting of Program 20-13-06 affects the BLF display for Hotline and Reverse Voice Over. Refer to [Hotline](#) and [Reverse Voice Over](#) features for additional information.
- You cannot send off-hook signals to an extension that is already receiving a voice announcement.
- An extension user can store the Off-Hook Signaling Service Code (709) under a One-Touch Key to provide quick Off-Hook Signaling access.
- An extension set as Operator in PRG 20-17-01 does not follow settings in PRG 20-13-05, PRG 20-13-06 or PRG 20-09-07 and always receives Off-Hook Signaling.
- Program 20-09-07 and 20-13-06 must be set to 1 in Class of Service for a normal extension to receive automatic Off-Hook Signaling.
- Off-Hook signaling is not supported for Wireless – DECT telephones.

Default Setting

Enabled

System Availability

Terminals

All Multiline and Single Line Terminals

Required Component(s)

None

Related Features

Callback

Call Waiting/Camp-On

Direct Inward Dialing (DID)

Handsfree and Monitor

Hotline

Intercom

One-Touch Calling

Programmable Function Keys

Reverse Voice Over

Single Line Telephones, Analog 500/2500 Sets

Programming

- ➔ 11-12-03 **Service Code Setup (for Service Access) – Override (Off-Hook Signaling)**
Assign a service code (709 by default) used for Off-Hook Signaling Override.
- ➔ 11-16-04 **Single Digit Service Code Setup – Intercom Off-Hook Signaling**
Assign a one-digit service code used for Off-Hook Signaling.

-
-
- ➔ **15-02-12 Multiline Telephone Basic Data Setup – Off-Hook Ringing**
For each extension, set Off-Hook Ringing type: 0 (muted), 1 (none), 3 (beep in speaker), 4 (beep in handset), 5 (Speaker & Handset Beep). DID, DNIS and DIL trunks can use any of the four options - normal/ring group trunks can use only options 0 or 1.

 - ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Off-Hook Signaling (code 33).

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

 - ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Enable (1)/Disable (0) station call queuing.
(default: 1)
☞ Must be set to 1 to enable automatic Off-Hook Signaling.

 - ➔ **20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
Enable (1)/Disable (0) the extension ability to receive Off-Hook Signaling.
(default: 1)

 - ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Enable (1)/Disable (0) the extension ability to automatically send Off-Hook Signals while busy on a handset call.
(default: 1)

 - ➔ **20-13-34 Class of Service Options (Supplementary Service) – Block Manual Off-Hook Signaling**
Turn off (0) or on (1) an extension ability to block off-hook signals manually sent from a co-worker.
(default: 0)


 - ➔ **20-18-06 Service Tone Timers – Interval of Call Waiting Tone**
Set the time (0~64800 seconds) between Off-Hook Signaling alerts.
(default: 10)

 - ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
(Default: 0).
Refer to [Table 2-27 Service Tone Setup, Program 80-01-02 on page 2-480](#).

 - ➔ **80-01-39 Service Tone Setup – Ring Busy Tone (Service Tone 39)**
Define Ring Busy Tone.
Refer to Service Tone 39 [Ring Busy Tone on page 2-478](#)
☞ Changing this program requires a KSU reset to take effect.

Operation


To send Off-Hook signals to an extension busy on a call:


 Your extension may send off-hook signals automatically.

1. Dial 7.

- OR -


Press **Off-Hook Signaling** key (PRG 15-07 or SC 751: 33).

 You hear ringback.

 To have your call voice-announce, dial 1.

Receiving Off-Hook Signaling on a single line telephone while engaged on an internal or external call:

1. When Off-Hook Signaling is heard in the receiver, press the **Flash Key** to answer the call. The first call is placed on hold.
2. Press the **Flash Key** again to toggle between the two calls.

 If the single line phone hangs up with the active call, the other call on hold rings back to the single line.

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One-Digit Dial Option

Enhancements

Version 1600 and higher is required.

Description

When a caller to an ACD Group is being played a Voice Response System (VRS) Delay Announcement they can dial a one-digit option to dial out of the Group. The One-Digit Dial Out option can be pressed during the Announcement only, for X seconds after the Announcement only, or both. (**Version 1600 or higher required.**)

There is one 1-digit option (0~9, * & #) that can be set to transfer the call to an Extension in the system, VM with integration (Group #), Ring Group, Speed Dial Bin, or another ACD Group.

Conditions

- The One-Digit Dial Out Option cannot transfer out to a Department Group other than a VM Department Group.
- Delay Announcement with In-Mail and ACI ports does not allow for the One-Digit Dial Out option.
- PRG 41-14-14 is used for **both** the 1st and 2nd Delay Announcements. You cannot allow one and not the other.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Automatic Call Distribution (ACD)

Voice Response System (VRS)

Programming

- ➔ **41-08-03 ACD Overflow Options – Delay Announcement Source Type**
For each ACD Group (1~64), assign the announcement message types. Delay Announcement functions are not available for ACD pilot number call. Each ACD Group can have unique overflow options.
- ➔ **41-08-04 ACD Overflow Options – ACD Overflow Transfer Time**
For each ACD Group (1~64), assign the overflow transfer time (0~64800 seconds).
(default: 30)
- ➔ **41-14-13-1 ACD Options – ACD Queue 1-Digit Assignment (Version 1600 or higher required)**
For each ACD Queue (1~64) assign the One-Digit number (0~9, *, #) used for the One-Digit Dial Out Option.
(default: Blank)
- ➔ **41-14-13-2 ACD Options – Destination Number Type (Version 1600 or higher required)**
For each ACD Queue (1~64), assign the Destination Number Type.
0 = None
1 = Ext/VM
2 = Ring Group
3 = Speed Dial Bin
4 = ACD Group
(default: 0)
- ➔ **41-14-13-3 ACD Options – Destination Number (Version 1600 or higher required)**
For each ACD Queue (1~64), assign the destination number for the assigned Destination Type.
(Up to eight digits).
(default: Blank)
- ➔ **41-14-14 ACD Options – DTMF Detection Assignment during Delay Announcement (Version 1600 or higher required)**
For each ACD Queue (1~64), assign if the One-Digit Dial Out option **can** (1 = Yes) or **cannot** (0 = No) be pressed during the Delay Announcements.
(default: 1)
- ➔ **41-14-15 ACD Options – DTMF Detect Time after Delay Announcement Message (Version 1600 or higher required)**
For each ACD Queue (1~64), assign the time (0~64800 seconds) after the Delay Announcement that the One-Digit Dial Out option works.
(default: 0)

-
-
- ➔ **41-11-01 VRS Delay Announcement – Delay Message Start Timer**
For each ACD Group (01~64), determine the time (0~64800 seconds) the system waits before playing the delay message. This program is activated when the delay announcement source and options are assigned as VRS in Program 41-08-03.
(default: 0)
 - ➔ **41-11-02 VRS Delay Announcement – 1st Delay Message Number**
For each ACD Group (01~64), assign the VRS message number used as the message source for the 1st Delay Announcement Message (0~49). This program is activated when the delay announcement source and options are assigned as VRS in Program 41-08-03.
(default: 0)
 - ➔ **41-11-03 VRS Delay Announcement – 1st Delay Message Sending Count**
For each ACD Group (01~64), determine the 1st Delay Message Sending Count (0~255).
(default: 0)
 - ➔ **41-11-04 VRS Delay Announcement – 2nd Delay Message Number**
For each ACD Group (01~64), assign the VRS message number used as the message source for the 2nd Delay Announcement Message (0~49). This program is activated when the delay announcement source and options are assigned as VRS in Program 41-08-03
(default: 0).
 - ➔ **41-11-05 VRS Delay Announcement – 2nd Waiting Message Sending Count**
For each ACD Group (01~64), determine the 2nd Waiting Message Sending Count (0~255).
(default: 0)
 - ➔ **41-11-06 VRS Delay Announcement – Tone Kind at Message Interval**
For each ACD Group (01~64), determine what the caller should hear between messages
0 = Ring Back
1 = MOH
2 = BGM.
(default: 0)
 - ➔ **41-11-07 VRS Delay Announcement – ACD Forced Disconnect Time After 2nd Delay Message**
For each ACD Group (01~64), when using ACD Overflow modes 3, 5, 8 and 9, enter the time (0~64800 seconds) after the caller hears the 2nd Announcement the system disconnects (drops) the call. This prevents callers from waiting in queue an excessive time. To disable this option (and allow callers to wait forever), enter 0.
(default: 60)
 - ➔ **41-11-08 VRS Delay Announcement – Queue Depth Announcement**
For each ACD Group (01~64), assign if the Queue Depth Announcements don't play (0), after the 1st Delay Announcement only (1), 2nd Delay Announcement only (2), or both the 1st and 2nd (3).
(default: 0)



Operation

None

One-Touch Calling

Description

One-Touch Calling gives a multiline terminal user one-button access to extensions, trunks, speed dial bins and selected system features. This saves users time when accessing co-workers, clients and features they use most often. Instead of dialing a series of codes, the user need only press the One-Touch key. An extension user can have One-Touch keys programmed for:

- Direct Station Selection** – one-button access to extensions
- Station Speed Dial** – one-button access to stored numbers (up to 24 digits)
- Speed Dial – System/Group/Station** – one-button access to stored speed dialing numbers
- Trunk Calling** – one-button access to trunks or trunk groups
- Service Codes** – one-button access to specific Service Codes

An extension user can chain dial with One-Touch Keys. For example, a user can store the number for a company Automated Attendant in key 1 and employee extension numbers in keys 2~5. The user can press key 1 to call the company, then one of keys 2~5 to ring the employee they want to reach.

An extension user or system administrator can optionally store a Flash command under a One-Touch key. This is helpful for One-Touch Keys used as Station Speed Dial bins. The stored Flash may be helpful to access features of the connected Telco, PBX or Centrex.

Conditions

- One-Touch keys provide a Busy Lamp Field (BLF).
- When a multiline terminal user is on a call, they can press a DSS key for that station to transfer to another station. It is not necessary to press Transfer to transfer to another station using a DSS key.
 - ✎ A multiline terminal user on a call, must press transfer to transfer a call off site with a DSS key.
 - ✎ A multiline terminal user on a call, must press transfer to transfer a call to a destination that is not a station (ACD/Voice Mail/Department group pilot, etc.).
- Pauses can be entered in the dial string of a DSS/One Touch button. The pause is entered as P in the dial string and causes the system to wait three seconds before sending the rest of the digits that follow the P (pause). Multiple pauses can be entered.

- The @ symbol can be entered in the dial string of a DSS/One Touch button. The @ applies only to ISDN and Intercom calls. When using the @, the system waits for the destination to answer (answer supervision), and then sends the rest of the digits.
- Entering a P (pause) in a DSS/One Touch dial string can be used for CO calls, Intercom calls, or after the @ for ISDN calls.

Default Setting

None

System Availability

Terminals

All Multiline Terminals and DSS Consoles

Required Component(s)

None

Related Features

Programmable Function Keys

Transfer

Programming

- ➔ **15-07-01 Programmable Function Keys**
Define a Programmable Function Key for One-Touch Calling by defining the key as a DSS/One-Touch key (01).
- ➔ **30-03-01 DSS Console Key Assignment**
Customize DSS Console keys to function as DSS keys, Service Code keys, Programmable Function Keys, and One-Touch Calling keys.
 - ✎ *When programming a feature with a One-Touch Key, refer to the feature description for additional programming options.*

Operation

Programmable Function Keys

To define a Programmable Function Key as a One-Touch Key:

1. Dial the service code for Function Key Programming (PRG 11-11-17, 751 by default).
2. Press the key to be defined.
3. Dial **01** (DSS/One-Touch Key Operation).
4. For Direct Station Selection (Extension):
 - a. Dial extension number you want assigned to that key.
 - b. Press **Hold**.
 - c. Press **Speaker**.

For Personal Speed Dial:

- a. Dial the general trunk access code (**9**).
 - **OR** -
 - Dial the Specific Trunk Service Code (**#9**) plus the trunk number (e.g., 005).
 - **OR** -
 - Dial the Trunk Group Service Code (**704**) plus the trunk group number (e.g., 1).
- b. Dial the number you want to store.
 - ☞ *The total of the digits stored in steps 3 and 4 cannot exceed 24.*
 - ☞ *Valid entries are 0~9, # and *. To enter a pause, press **MIC**. To store a Flash, press the **Redial** key.*
- c. Press **Hold**.
- d. Press **Speaker**.

For Speed Dial – System/Group:

- a. Dial **#2** to store a Speed Dial – System dialing number.
 - **OR** -
 - Dial **#4** to store a Speed Dial – Group dialing number.
- b. Dial Speed Dial number storage code (e.g., 001).
- c. Press **Hold**.
- d. Press **Speaker**.

For Central Office Calls, Placing (Trunk Calling):

- a. a. Dial the general trunk access code (**9**).

- OR -

Dial the specific Trunk Service Code (**#9**) plus the trunk number (e.g., 005).


- OR -

Dial the Trunk Group Service Code (**704**) plus the trunk group number (e.g., 1).

- b. Dial the telephone number to be stored.
- c. Press **Hold**.
- d. Press **Speaker**.

For Service Codes:



- a. Dial the Service Code you want stored.

 *For example, if you want a One-Touch Key to automatically clear your Last Number Redial, enter 776.*

- b. Press **Hold**.
- c. Press **Speaker**.

Checking the One-Touch Keys

To check the function of a One-Touch key:

1. Press the **Help** key.
2. Press the **One-Touch** key.
 -  *The stored function displays.*
 -  *Repeat this step to check additional keys.*
3. Press the **Exit** key.

Operator

Enhancements

Version 1500 and higher allows multiple Operator Groups.

Description

When an extension user dials 0, calls are routed to a main system operator. The operator can answer and route outside calls or locate employees using the Page feature.

A maximum of eight operators are available.

Conditions

- Tandem Ringing is not supported on an Operator or Attendant Extension.
- Attendant extensions can have up to 32 incoming calls queued before additional callers hear busy tone.
- The operator extension cannot be a CAR Key or virtual extension.
- When dialing 0 from In-Mail across CCIS and CCISoIP, it follows Program 47-13-01 Key 10.
- When dialing 0 from the in-skin Voice Mail across CCIS and CCISoIP, it follows what is in the operator's set up.
- With **Version 1500 and higher**, extensions and trunks can be assigned to an operator group. A call to an operator that is busy rolls to the next operator in the operator group.

Default Setting

Extension 101 is an operator.

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Attendant Call Queuing

Programming

- **20-01-01 System Options – Operator Access Mode**
Set up priority of a call when calling an operator telephone (0 = Step, 1 = Circular).
(default: 0)
- **20-17-01 Operator Extension – Operator's Extension Number**
Designate an extension an operator. When an extension user dials 0 or 9 (defined by Program 11-01, Type 5), calls go to the operator selected in this program.
(default: 101)
If you do not assign an extension in Program 90-11-01, system alarms appear on the extension assigned in this option.
- **20-35-01 extension Operator Setting**
Assign an extension to an operator group.
(Input: 0~15)
(default: 0)
- **20-36-01 Trunk's Operator Mode**
Assign a trunk to an operator group.
(Input: 0~15)
(default: 0)
- **20-37-01 Operator Extension Group Setup**
Define the initial operator extension in the operator group.
(default: None)
- **20-38-01 Operator Group Setting**
Set up priority of a call when calling an operator telephone.
(0 = Step, 1 = Circular)
(default: 0)

Operation

Refer to the individual features for operation.

(OPX) Off-Premise Extension

Description

Off-Premise Extension allows a single line telephone, located remotely from the main installation site, to access the system features with the same abilities as an on-premise single line telephone.

Conditions

- Each OPX(2)-U() ETU provides two circuits.
- The maximum loop resistance between an OPX(2)-U() ETU and an Off-Premise Extension Single Line Telephone is 1600 ohms (including single line telephone set resistance).
- The OPX(2)-U() ETU has a built-in ringer (RSG). This ETU supports Synchronous Ringing and detects Dial Pulse/DTMF tones.
- The OPX(2)-U() ETU does not support an interface to a Voice Mail unit.

Default Setting

None

System Availability

Terminals

Single Line Telephones

Required Component(s)

OPX(2)-U() ETU

Related Features

Single Line Telephones, Analog 500/2500 Sets

Programming

Refer to [Single Line Telephones, Analog 500/2500 Sets on page 2-883](#) for related programming.

Operation

Normal call handling procedures for single line telephones apply.

Paging, External

Description

With External Paging, a user can broadcast announcements over paging equipment connected to external Paging zones. When a user pages on of these external zones, the system broadcasts the announcement over the speakers. Like Internal Paging, External Paging allows a user to locate another employee or make an announcement without calling each extension individually.

The Electra Elite IPK II system allows up to eight External Paging zones, or a common zone output provided by the KSU (Speaker 9). All other speakers (1~8) require a port on a PGD(2)-U10 ADP, with a maximum of two external paging circuits per module. You must have four PGD(2)-U10 ADPs to get the eight external zones. In addition, each external zone has an associated relay contact. When a user pages to a zone, the corresponding contact activates (closes). This provides for Paging amplifier control.

Combined Paging

Use Combined Paging when you want to simultaneously Page into an internal and corresponding external zone. For example, you can Page your company warehouse and outside loading dock at the same time. Combined Paging is available for zones 1~8 and All Call. Refer to [Paging, Internal on page 2-775](#) for more on setting up Combined Paging. In addition, you can program a Function Key as a Combined Paging key. Using the External Page Function Key, when an All Call External Page Function Key is programmed, it includes both the external zones and the assigned internal zone(s). If the internal page zone is busy or there are no extensions in a page group, the announcement is made on the external zones only.

Conditions

- The Electra Elite IPK II provides a common zone output provided by the KSU. For more than one external page zone, External Paging requires PGD(2)-U10 ADPs and customer-provided paging equipment.
- Talkback paging requires using a PGD(2)-U10 ADP. The Electra Elite IPK II common zone output provided by the KSU does not allow talkback.
- A common zone output is provided by the KSU and is considered Zone #9 when programming.
- A Class of Service option is available in system programming to prevent display telephones from showing incoming paging information. This allows the system to save processor time and speed up system operation.

- DID and DIL trunks do not ring external page speakers. Only trunks defined as normal in Program 22-02-01 ring external page speakers.
- Paging keys can be assigned on Programmable Function Keys and Direct Station Selection (DSS) Consoles to simplify External Paging operation.
- If a PGD(2)-U10 ADP circuit has a Door Box connected, you cannot use that circuit for External Paging.
- To have outside calls ring External Paging Zones at night, refer to the Night Service feature and Program 31-05.
- The PGD(2)-U10 ADP can be connected only to an ESIB(8)-U20 ETU. If the PGD(2)-U10 ADP is connected to an ESIE(8)-U20 ETU, the PGD(2)-U10 ADP is NOT recognized.
- When a ESIB(8)-U20 ETU has an ESIE(8)-U20 ETU installed [ESIB/E(16)], the PGD(2)-U10 ADP has both channels available for Paging, MOH, External Ringers and Door Box according to strapping on ports 1~8. The corresponding port on the ESIE(8)-U20 ETU do not support a station or other hardware (it is used by the PGD(2)-U10 ADP for channel 2).
- When a ESIB(8)-U20 ETU is installed without an ESIE(8)-U20 ETU installed, a PGD(2)-U10 ADP installed on ports 1~8 is recognized, but only channel 1 on the PGD(2)-U10 ADP is supported. Channel 2 is not available.
- ESIB(8)-U10 ETUs and ESIE(8)-U10 ETUs do not support the PGD(2)-U10 ADP.
- The maximum number of PGD(2)-U10 ADPs is 56. Refer to the Hardware Manual for more information which describes how many of the 56 can be for paging, door box or Music on Hold (MOH).
- Phones that have an APR/APA installed do not pass voice to a trunk used for paging until the interdigit time expires (Program 21-01-03).

Default Setting

No External Paging defined.

System Availability

Terminals

All Stations

Required Component(s)

- ESIB(8)-U20 ETU for PGD(2)-U10 ADP
- ESIE(8)-U20 ETU for 2nd channel support
- PGD(2)-U10 ADP for Zone Paging
- 1- or 2-way amplifier and speakers (locally provided)

Related Features

Central Office Calls, Placing

Direct Inward Dialing (DID)

Direct Inward Line (DIL)

Direct Station Selection

Door Box

Night Service

Paging, Internal

Programmable Function Keys

Transfer

Programming

- ➔ **10-05-01 General Purpose Relay Setup**
Define which relay circuits (5~8) on the PGD(2)-U10 ADP are used for General Purpose Relays.
- ➔ **11-12-50 Service Code Setup (For Service Access) – General Purpose Relay**
Specify the service code to be used for toggling the relay open and closed (default: 780).
- ➔ **15-07-01 Programmable Function Keys**
Assign function keys for External Paging zones (19 + zone) and External All Call Page (20). If required, define a function key for a multiline terminal to use the general purpose relay (51).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

- ➔ **20-13-29 Class of Service Options (Supplementary Service) – Paging Display**
Turn Off (0) or On (1) an extension user ability to display incoming internal paging information. (default: 1)
- ➔ **31-01-02 System Options for Internal/External Paging – Page Announcement Duration**
Set the maximum allowable time (0~64800 seconds) for a Paging announcement. (default: 1200)
- ➔ **31-04-01 External Paging Zone Group**
Assign each External Paging Speaker to an External Paging Group (0~8) used for accessing the zone. If zones 1~8 are not connected to PGD(2)-U10 ADP, set these unused zones to External Paging Group 0.
External Paging Speaker/Zones: 1~9
- ➔ **31-05-01 Universal Night Answer/Ring Over Page**
Assign Universal Night Answer ringing (1)/ No ringing (0) to each External Paging zone. For each trunk port, make a separate entry for each External Paging Zone (1~9). (default: 1200)
- ➔ **31-06-01 External Speaker Control – Broadcast Splash Tone before Paging (Paging Start Tone)**
Enable Splash Tone (1) or Chime Tone (2)/Disable both (0) for each External Speaker Number (1~9). (default: 2)
- ➔ **31-06-02 External Speaker Control – Broadcast Splash Tone after Paging (Paging End Time)**
Enable Splash Tone (1) or Chime Tone (2)/Disable both (0) for each External Speaker Number (1~9). (default: 2)
- ➔ **31-06-04 External Speaker Control – CODEC Transmit Gain Setup**
- ➔ **31-06-05 External Speaker Control – CODEC Receive Gain Setup**
Select the CODEC gain types (1~32) for each External Speaker (1~9). (default: 32)
- ➔ **31-07-01 Combined Paging Assignments**
Assign an External Paging Group (0~8) to an Internal Paging Zone (0 = All Call, Zones 1~64) for Combined Paging. When an extension user makes a Combined Page, they simultaneously broadcast into both the External and Internal Zone. (default: 1).
- ➔ **31-08-01 BGM on External Paging – BGM**
Enable (1)/Disable (0) the Background Music option for each External Paging Zone (1~9). If set to 1, the system plays Background Music over the zone when it is idle. (default: 0)

Operation

To Page into an external zone:

1. Press External Paging key (PRG 15-07 or SC 751: 19 for External Paging zones or 20 for External All Call Paging).
2. Make announcement.

- OR -

1. At the multiline terminal, press **Speaker** or pick up the handset.


- OR -


At single line telephone, lift the handset.

2. Dial **703** and the External Paging Zone code (1~8 or 0 for All Call).

- OR -

Dial ***1** and the Combined Paging Group code (1~8 or 0 for Internal/External All Call).

 *Display indicates the Combined Paging as an External Page.*


 *If the Internal Page Zone is busy or if there are no extensions in a page group, the page may be announced as an External Page only.*


3. Make an announcement.

4. Dial **703** and the External Paging Zone code (1~8 or 0 for All Call).

- OR -

Dial ***1** and the Combined Paging Group code (1~8 or 0 for Internal/External All Call).

 *Display indicates the Combined Paging as an External Page.*

 *If the Internal Page Zone is busy or if there are no extensions in a page group, the page may be announced as an External Page only.*

5. Make an announcement.

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Paging, Internal

Description

Internal Paging lets extension users broadcast announcements to other multiline terminal users. When a user makes a Zone Paging announcement, the announcement broadcasts to all idle extensions in the zone dialed. With All Call Paging, the announcement broadcasts to all idle extensions programmed to receive All Call Paging. An extension can be a member of only one Internal Paging Zone. Like External Paging, Internal Paging allows a user to locate another employee or make an announcement without calling each extension individually.

Combined Paging

Use Combined Paging when you want to simultaneously Page into an internal and corresponding external zone. For example, you can Page your company warehouse and outside loading dock at the same time. Combined Paging is available for Paging zones 1~8 and All Call. Optionally, you can change the Combined Paging assignments. For example, you can associate External Paging Zone 1 with Internal Paging Zone 4. You can program a Function Key as a Combined Paging key. When an All Call External Page Function Key is programmed, it includes both the external zones and the assigned internal zone(s). If the internal page zone is busy or there are no extensions in a page group, the announcement is made on the external zones only.

Conditions

- Internal Paging does not require a PGD(2)-U10 ADP.
- You can assign up to 50 extensions to an Internal or All Call Paging Group.
- A system must have at least one extension port idle to make an Internal Page. If no extension port is idle, the extension user performing the Page hears a busy signal.
- There are 64 available Internal Paging Groups (Zones).
- A Class of Service option is available in system programming to prevent display telephones from showing incoming internal paging information. This allows the system to save processor time and speed up system operation.
- An extension user can broadcast an announcement over an External Paging Zone.
- Function keys simplify Internal Paging operation.
- You must assign an extension to be in a two-digit zone in PRG 31-02-01 before you can assign a function key using the 751 service code as a two-digit Internal Group Paging Zone key.
- If Auto Hold in PRG 15-02-07 is set to Cut(1), when a user presses the page key while on a trunk call, the trunk call is put on hold.

- A single line telephone can initiate an Internal Zone page, but cannot receive an Internal Zone Page.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Single Line Telephones

Required Component(s)

None

Related Features

Meet Me Paging

Meet Me Paging Transfer

Paging, External

Programmable Function Keys

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign function keys for Internal Paging Zones (code 21 + page zone) and Internal All Call Paging (code 22).
Internal Page Zones: 0, 1~9, 00, 01~64
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

- ➔ **20-13-29 Class of Service Options (Supplementary Service) – Paging Display**
Turn Off (0) or On (1) an extension user ability to display incoming internal paging information.(Default: 1)
- ➔ **31-01-01 System Options for Internal/External Paging – All Call Paging Zone Name**
Assign a name to the All Call Internal Paging Zone. The name shows on the display of the telephone making the announcement.
(default: Group All)
- ➔ **31-01-02 System Options for Internal/External Paging – Page Announcement Duration**
Set the maximum allowable duration (0~64800 seconds) for a Paging announcement.
(default: 1200)
- ➔ **31-02-01 Internal Paging Group Assignment – Internal Paging Group Number**
Assign extensions to Internal Paging Zones. An extension must be assigned to a 2-digit zone to access any 2-digit zone.
Internal Page Zones: 0, 1~9, 00, 01~64
[default: 0 (IP Station); 1 (TDM)]
- ➔ **31-02-02 Internal Paging Group Assignment – Internal All Call Paging Receiving**
Allow (1)/Prevent (0) All Call Internal Paging for each extension. If allowed, extensions can make and receive All Call Internal Paging announcements. If prevented, extension can make only All Call Internal Paging announcements.
(default: 0)
- ➔ **31-03-01 Internal Paging Group Settings – Internal Paging Group Name**
Program names (up to 12 characters) for the Internal Paging Zones.
(default: External Paging Group 0~64 = Group 1~ Group 64)
- ➔ **31-03-02 Internal Paging Group Settings – Internal Paging Splash Tone**
For each Internal Paging Zone, have normal (0), muted (1) or no (2) Internal Paging alert tones. If enabled (0 or 1), extensions hear two beeps before Paging announcements. If disabled, the extensions hear the announcement without the beeps.
Internal Page Zones: 0, 1~9, 00, 01~64
(default: 0)
- ➔ **31-07-01 Combined Paging Assignments**
For each External Paging Group (1~8 and 0 for All Call), assign a corresponding Internal Zone for Combined Paging.
Internal Page Zones: 0, 1~9, 00, 01~64
(default:1)

Operation

To make an Internal Page announcement:


Multiline Terminal

1. Press the zone **Internal Paging** key (PRG 15-07 or SC 751: 21 + 0 or 1~9 or 01~64 for zones (0 or

00 for All Call).


- OR -


1. Press **Speaker** or lift the handset.
2. Dial **701** and the Paging Zone number (0~9 or 00~64).

 *Dialing 0 or 00 calls All Call Internal Paging.*

- OR -

Dial ***1** and the Combined Paging Group code 1~8 or 0 (for Internal/External All Call).


 *Display indicates the Combined Paging as an External Page.*


 *If the Internal Page Zone is busy or if there are no extensions in a page group, the page is announced as an External Page only.*

3. Make an announcement.
4. Press **Speaker** to hang up.

Single Line Telephone

1. Lift the handset.
2. Dial **701** and the Paging Zone number (0~9 or 00~64).

 *Dialing 0 or 00 calls All Call Internal Paging.*

 *Dial ***1** and the Combined Paging Group code 1~8 or 0 (for Internal/External All Call).*

3. Make an announcement.
4. Hang up.

Park

Enhancements

Version 2000 or higher provides the ability to perform Call Park – Step Call by pressing one key to find and park a call in an idle park location. The park location is displayed.

Description

Park places a call in a waiting state (called a Park Orbit) so that an extension user may pick it up. There are two types of Park: System and Personal. Use System Park when you want to have the call wait in a system orbit. Personal Park allows a user to Park a call at their extension so a co-worker can pick it up. After parking a call in orbit, a user can Page the person receiving the call and hang up. The paged party dials a code or presses a programmed Park key to pick up the call. With Park, it is not necessary to locate a person to handle their calls. A call parked for too long recalls the extension that initially parked it, however the call remains in the park orbit until it is answered. There are 64 Park Orbits (1~64) available for use.

Extended Park

An extension Class of Service determines whether it uses the normal Park Orbit Recall time or the Extended Park Orbit Recall time. The timers are set up in system programming. When an extension with Extended Park Recall Class of Service option parks a call, it recalls after the Extended Park Orbit Recall time. When an extension with the Normal Park Orbit Recall Class of Service option parks a call, it recalls after the normal Park Orbit Recall time, however the call remains in the park orbit until it is answered.

Programmable Function Key and Service Code Available for Personal Park

The Personal Park feature is enhanced by using a Programmable Function Key or service code (3-digit or 1-digit) to place a call in Personal Park. This option is available for multiline terminals, single line sets, and Electra Elite IPK II Wireless telephones and can be used for analog or ISDN trunks.

Conditions

- An extension can park a call in any Park Orbit. However, an extension user can pick up only a call Parked by a member of its own Park group (see Program 24-03).
- When a 2-button telephone user parks a call, they must wait the Interdigit Time (normally 10 seconds) before trying to retrieve it.
- An extension can have only one Personal Park key.

- When the terminal that has a call in Personal Park is unplugged, the Personal Park is released and the held caller is placed on Non-Exclusive Hold.
- The following table indicates what condition the service codes and Programmable Function key can be used.

Status	Using 3-Digit Service Code	Using 1-Digit Service Code	Using Personal Park Key
Speaking	Not Available	Not Available	Available
ICM Dial Tone or Busy Tone	Available	Not Available	Available
Calling Another Extension	Not Available	Available (with outside call on hold – when called extension does not answer)	Available
Receiving a Personal Park Recall	Not Available	Not Available	Available

- A user can display the Caller ID of a call in Park if Caller ID is enabled (1) in Program 20-09-02.
- Park keys can be assigned on DSS consoles.
- Calls on virtual extension keys cannot be put in Personal Park if program 15-18-01 is set to Land on the key (1).
- Function keys simplify Park operation.
- Calls on Virtual Extension keys cannot be Call Parked.
- One Touch keys programmed for Park Hold Service Code cannot be used to park calls without first using Hold or Transfer.
- Call Park – Step Call is supported in the local system only.
- Program 20-11-27: Class of Service Options (Hold/Transfer Service) – Call Park Automatically Searched is not supported in **Version 1600**.

Default Setting

Enabled

System Availability

Terminals

AI Terminals

Required Component(s)

None

Related Features

Caller ID

Call Arrival (CAR) Keys

Direct Station Selection

Hold

Programmable Function Keys

Programming




- **11-12-31 Service Code Setup (for Service Access) – Park Hold**
Set the service code which should be used for placing a call in Park (default: #6).
- **11-12-32 Service Code Setup (for Service Access) – Answer for Park Hold**
Set the service code which should be used for answering a call in Park (default: *6).
- **11-12-35 Service Code Setup (for Service Access) – Station Park Hold**
Set the service code to be used for placing a call in a Personal Park (default: 757).
- **11-16-11 Single Digit Service Code Setup – Station Park Hold**
Customize the one-digit service code to be used when placing a call in Personal Park (default: No setting).
- **15-02-08 Multiline Telephone Basic Data setup – Automatic Handsfree**
Set whether pressing a key accesses a One-Touch key (1) or Preselects the key (0). (default: 1)
- **15-07-01 Programmable Function Keys**
Assign a key as a Park Orbit key (code *04 plus Park orbit number [01~64]) or as a Personal

Park key (code *07).


- ➔ **15-18-01 Virtual Extension Key Enhanced Options – Virtual Extension Key Operation Mode**
Set whether an incoming call to a Virtual Extension/CAR resides on the Virtual Extension/CAR key once answered (1) or appears on a CAP Key/CO Appearance Line key (0). This setting applies to Multiline Terminals, single line telephones and virtual extension numbers (default: 0).
- ➔ **15-18-02 Virtual Extension Key Enhanced Options – Display Mode when pacing a call on Virtual Extension Key**
(default: 0)
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to extensions.
- ➔ **20-11-19 Class of Service Options (Hold/Transfer Service) – Hold/Extended Park**
Determine whether an extension Class of Service should allow normal (0) or extended Park (1). (default: 0)
- ➔ **20-11-27 Class of Service Options (Hold/Transfer Service) – Call Park Automatically Search**
Enable (1)Disable (0) using the Call Park Automatically Search option. (default: 1)
- ➔ **24-01-02 System Options for Hold – Hold Recall Callback Time**
A call left parked too long recalls the extension that initially parked it for this time (0~64800 seconds). (default: 30)
- ➔ **24-01-06 System Options for Hold – Park Hold Time - Normal**
Set the Park Hold Time (0~64800 seconds). A call left parked longer than this time recalls the extension that initially parked it. (default: 90)
- ➔ **24-01-07 System Options for Hold – Park Hold Time - Extended (Recall)**
Set the Extended Park Hold Time (0~64800 seconds). A call left parked longer than this time recalls the extension that initially parked it. (default: 300)
- ➔ **24-03-01 Park Group – Park Group Number**
Assign an extension to a Park Group (01~64). An extension can only pick up a call parked by a member of its own Park Group. (default: 1)

Operation

To Park a call in a system orbit:

-  *You can Park Intercom or trunk calls.*
- 1. Press the **Park** key (PRG 15-07 or SC 752: *04 + orbit).
 -  *The Park key LED lights.*
 -  *If you hear busy tone, the orbit is busy. Try another orbit.*
- 2. Use Paging to announce call.

3. Press **Speaker** to hang up.

 *If not picked up, the call recalls to you.*


- OR -


1. At the multiline terminal or 2-button telephone, press **Hold**.

- OR -

At a 500/2500 single line telephone, hookflash.


2. Dial **#6** and the Park orbit (01~64).


 *If you hear busy tone, the orbit is busy. Try another orbit.*

 *With **Version 2000 or higher**, if you hear a busy tone, the orbit is busy. Dial #6 if enabled in PRG 20-11-27 (Call Park AutoSearching) to search for an idle park location in ascending order.*

3. Use Paging to announce the call.

4. Press **Speaker** to hang up.

 *If not picked up, the call recalls to you.*

 *The parked call recalls after the Park Hold Time (Program 24-01-06). The call rings the extension to which it recalled for the Hold Recall Callback Time (Program 24-01-02). The call then goes on Hold for the Park Hold Time, then recalls again for the Hold Recall Callback Time. The call continues to cycle between Hold and recall until the extension user answers the call or the outside party hangs up.*

To pick up a parked call:

1. Lift the handset.
2. Press the **Park** key (PRG 15-07 or SC 752: *04 + orbit).

- OR -

1. At the multiline terminal or 2-button telephone, press **Speaker**.

- OR -

At single line telephone, lift the handset.

2. Dial *6 and the Park orbit (01~64).

To park a call at your extension:

1. Do not hang up.




2. Press the **Personal Park** key (PRG 15-07 or SC 752: *07).


- OR -

Press **Hold** and dial 757.




- OR -

Press **Hold** and the **Personal Park** key (PRG 15-07 or SC 752: *07).




-  *At a 500/2500 single line telephone, hookflash instead of pressing HOLD.*
-  *A confirmation tone is heard and the call is parked at your extension. If the extension has a Personal Park key, the key flashes.*
-  *The Personal Park single-digit service code (PRG 11-16-11) cannot be used in this operation.*

3. Page your co-worker to pick up the call.
 4. Press **Speaker** to hang up (or hang up at the single line telephone).
-  *If not picked up, the call recalls to you.*


To Park an outside call at your extension after trying to call a co-worker:

1. While on a call, press **Hold**.
2. Dial a co-worker's extension number.
 -  *The co-worker does not answer.*
3. Press the **Personal Park** key (PRG 15-07 or SC 752: *07).
 - OR -
 - Dial the Personal Park single digit code (PRG 11-16-11).
 -  *The Intercom call to the co-worker is dropped. A confirmation tone is heard and the outside call is parked at your extension.*
 -  *If the co-worker answers the call, the outside call rings back after the intercom call is completed. The call can then be placed in Personal park if desired.*


To pick up a call parked at your extension:

1. Press the **Personal Park** key (PRG 15-07 or SC 752: *07).
 - OR -
 - Press **Speaker** and dial 757.
 -  *At a single line telephone, do not press **Speaker**.*
 -  *The Personal Park single-digit service code (PRG 11-16-11) cannot be used in this operation.*
 -  *If it recalls the extension, press the **Personal Park** key or flashing **Speaker** to answer the call.*

To answer a call parked at a co-worker's extension:

1. Press **Speaker**, dial ** plus the co-worker's extension number.
 -  *At a single line telephone, do not press **Speaker**.*

To display Caller ID for a call in Park:

 With Program 15-02-08 set to 0 (preselect) for this feature.

1. With Program 15-02-08 set to 0 (preselect) and a call in Park, press the **Park** key. (PRG 15-07 or SC 752: *04).

- OR -


With Program 15-02-08 set to 1 (One-Touch), and a call in Park, press **Feature**, then the **Park** key (PRG 15-07 or SC 752: *04).

Call Park – Step Call: (Version 2000 or higher required)**To Park a call in the first available system orbit:**


 You can Park Intercom or trunk calls.

1. Press **Hold** or **Transfer**.


2. Dial **#6**.

 If you hear a busy tone, the orbit is busy. Proceed to step 3.

3. Dial *****.

 Program 20-11-27 must be enabled in the Multiline Terminals Class of Service.

4. Press **Speaker** to hang up.

 If not picked up, the call recalls to you.

- OR -

1. Press **Hold** or **Transfer**.

2. Press the DSS/BLF key programmed as **#6***
(The Park location is displayed in the LCD).

3. Press **Speaker** to hang up.

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PBX Compatibility

Description

You can connect your telephone system trunks to Centrex/PBX lines, rather than Telco trunk circuits. This makes the trunk inputs to the system 500/2500 type compatible Centrex/PBX extensions, rather than Telco circuits. PBX Compatibility lets the system be a node (i.e., satellite) in a larger private telephone network. To place outside calls when the system is behind a PBX, telephone system users must first dial the PBX trunk access code (usually 9).

The system provides the following PBX Compatibility options:

PBX Trunk Access Code Screening

The system can monitor the numbers users dial and screen for PBX trunk access codes. The system can screen up to four groups of trunk access codes. The codes can have one or two digits, consisting of the digits 0~9, # and *. (You use Line Key 1 as a wild card entry.)

PBX Trunk Toll Restriction

The system can provide the Toll Restriction for the PBX trunk, or restriction can be handled solely by the connected PBX. If the telephone system provides the restriction, it restricts the digits dialed after the PBX access code.

PBX Call Restriction

When the telephone system does the Toll Restriction, it can further restrict users from dialing PBX extensions. In this case, the only valid numbers are those dialed after the PBX trunk access code. The only PBX facility telephone system users can access are the PBX outside trunks.

Automatic Pause

The system automatically pauses when it sees a PBX trunk access code during manual dialing, Speed Dialing, Last Number Redial, Repeat Redial and Save Number Dialed. This gives the connected PBX time to set up its trunk circuits.

Conditions

- When using Account Codes, do not use * in a PBX access code. Otherwise, after the *, the trunk stops sending digits to the central office.
- The system automatically pauses after it finds a PBX access code in a Speed Dialing bin.
- If Speed Dialing routes a call to a PBX trunk, it does not automatically insert a PBX access code. It outdials the digits just as they are stored.
- Users answer incoming calls on PBX trunks just like other trunks. All relevant access and Ring Group programming applies.

- Except for dialing the PBX access code, users place calls on PBX trunks just like other trunks. All relevant access programming applies. Refer to the [Central Office Calls, Placing on page 2-195](#) feature for more details.
- You can have DILs route from the connected PBX. Users can access these trunks for outgoing PBX calls. All PBX Compatibility restrictions and programming apply.
- Flash may allow access to certain PBX features – like Transfer. Make sure you program Flash for compatibility with the connected PBX.
- The system does not provide automatic Pulse to Tone Conversion after outdialing the PBX trunk access code.
- You can program incoming DISA trunks to be outgoing PBX trunks. All PBX Compatibility restrictions and programming apply.
- PBX trunks can follow normal system Toll Restriction.
- Users can get outbound access to PBX trunks through Trunk Groups and/or Trunk Group Routing. All PBX Compatibility restrictions and programming apply.
- If the system routes a call to a PBX trunk, it does not automatically insert the PBX access code. It outdials the call just as the user dialed it.

Default Setting

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Account Code Entry

Call Forwarding – Centrex

Central Office Calls, Answering

Central Office Calls, Placing

Code Restriction

Direct Inward Line (DIL)

Direct Inward System Access (DISA)

Flash

Pulse to Tone Conversion

Ring Groups

Speed Dial – System/Group/Station

Trunk Group Routing

Trunk Groups

Programming

- **14-01-01 Basic Trunk Data Setup – Trunk Name**
Set the names for trunks. The trunk name displays at display multiline terminals for incoming and outgoing calls. Up to 12 characters are allowed.
- **14-01-02 Basic Trunk Data Setup – Transmit Level**
Set these options for compatibility with the connected PBX.
- **14-01-08 Basic Trunk Data Setup – Toll Restriction**
For each PBX trunk port, Enable (0)/Disable (1) Toll Restriction.
- **14-02-01 Analog Trunk Data Setup – Signaling Type (DP/DTMF)**
- **14-02-02 Analog Trunk Data Setup – Ring Detect Type**

- ➔ **14-04-01 Behind PBX Setup – Type of Connection**
For each PBX trunk port, enter 1. Make a separate entry for each Night Service mode.
- ➔ **21-04-01 Toll Restriction Class for Extensions**
Assign a Toll Restriction Class (1~15) to each extension. (default: 2)
- ➔ **21-05-12 Toll Restriction Class – PBX Call Restriction**
For each Toll Restriction Class, enter 1 to restrict calls on the PBX trunk to outside calls only. Enter 0 to allow users to dial PBX extensions. (default: 0)
- ➔ **21-06-08 Toll Restriction Table Data Setup – PBX Access Code**
Enter the system PBX access codes. The system can have up to four groups of codes. A code can have one or two digits. Valid entries are 0~9, # and *. Use Line Key 1 as a “don’t care” digit. If using Account Codes, do not use * in the PBX Access Code.

Operation

To place a call over a PBX trunk:

1. At multiline terminal, press **Speaker** and dial **704**.
- OR -
At single line telephone, lift the handset and dial **704**.
2. Dial PBX trunk group number (**1~9** or **001~100**).
3. Dial PBX access code and number.
- OR -
 1. At the multiline terminal only, press **PBX trunk group** key (PRG 15-07 or SC 752: *02 + group).
 2. Dial PBX access code and number.
- OR -
 1. At the multiline terminal, press **Speaker** and dial **9**.
- OR -
At the single line telephone, lift the handset and dial **9**.
 2. Dial the PBX access code and number.
- OR -
 1. At the multiline terminal, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
 2. Dial **#9**.
 3. Dial the PBX trunk number (e.g., 005 for line 5).

4. Dial the PBX access code and number.

- OR -

1. Press the **PBX trunk key** (PRG 15-07 or SC 752: *01 + 1 to 200).

2. Dial the PBX access code and number.

 *In all cases above, Toll Restriction may prevent your call.*

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PC Programming

Enhancements

Version 1100 or higher provides updated PCPro software that includes a database conversion tool and supports DESI.

Version 1600 or higher: PC Programming software allows the Terminal Type to be assigned and saved for DESI export.

Description

IPK II introduces three different methods for programming. The first is via the handset, the second is by PCPro (SAT) and third by WebPro.

PCPro is a Microsoft Windows™ based application. It stems from the SAT application in IPK. It allows the technician/system administrator to download a database from the KTS, make changes, and then upload.

New to IPK II is WebPro. This application is a web server running on the CPU card of the KTS. No special installation program is required. A user programs the KTS using their standard web browser.

An overview of the three programming applications is given below.

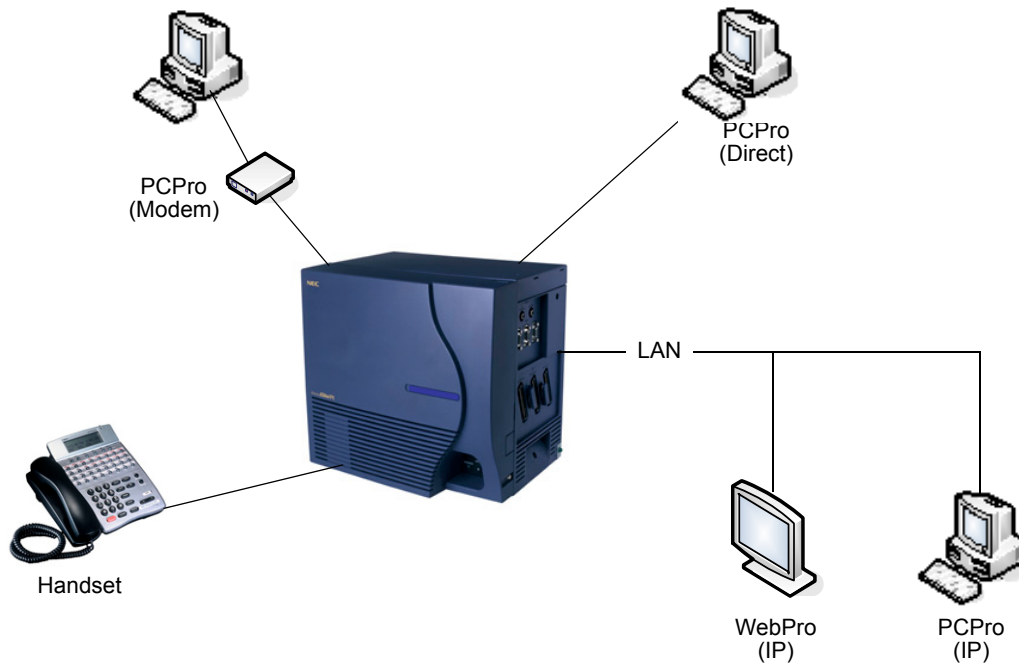


Figure 2-20 PC Programming Overview

Enhancements

IPK to IPK II Conversion – With **Version 1100 or higher**, PCPro, in conjunction with a new version of IPK SAT, can convert an IPK database to IPK II. Not every memory block is converted, but some of the more tedious programming is converted. Refer to the IPK to IPK II Conversion document for more information.

Connecting

As can be seen in [Figure 2-20 PC Programming Overview](#), three connection types are available to PCPro/WebPro.

If using PCPro, a user can connect directly, remotely using a modem or via LAN. A connection with the KTS is made via the Connection Dialog in the application. (Refer to [Figure 2-21 PCPro Connection Dialog Box](#).)

- Direct* connections are established using the RS232 serial port (COM Port 1) on the side of the KTS.
- Modem* (remote) connections are established via the internal CPUII modem. You need to dial in to an extension on the KTS and redirect that extension to the internal modem.

- ❑ IP (LAN) connections are established via the Ethernet connector on the CPUII card.

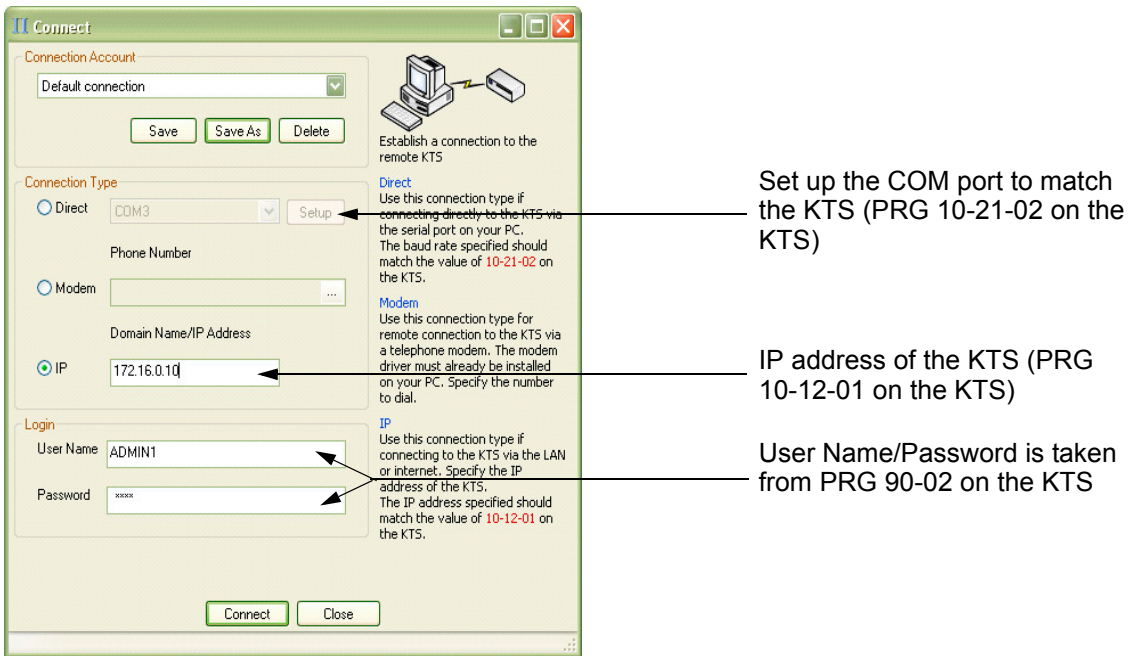


Figure 2-21 PCPro Connection Dialog Box

If using WebPro, a user can connect only via IP. To connect, launch a web browser (i.e., IE or Mozilla) and enter the IP address of the switch (refer to [Figure 2-22 WebPro Login Screen.](#))

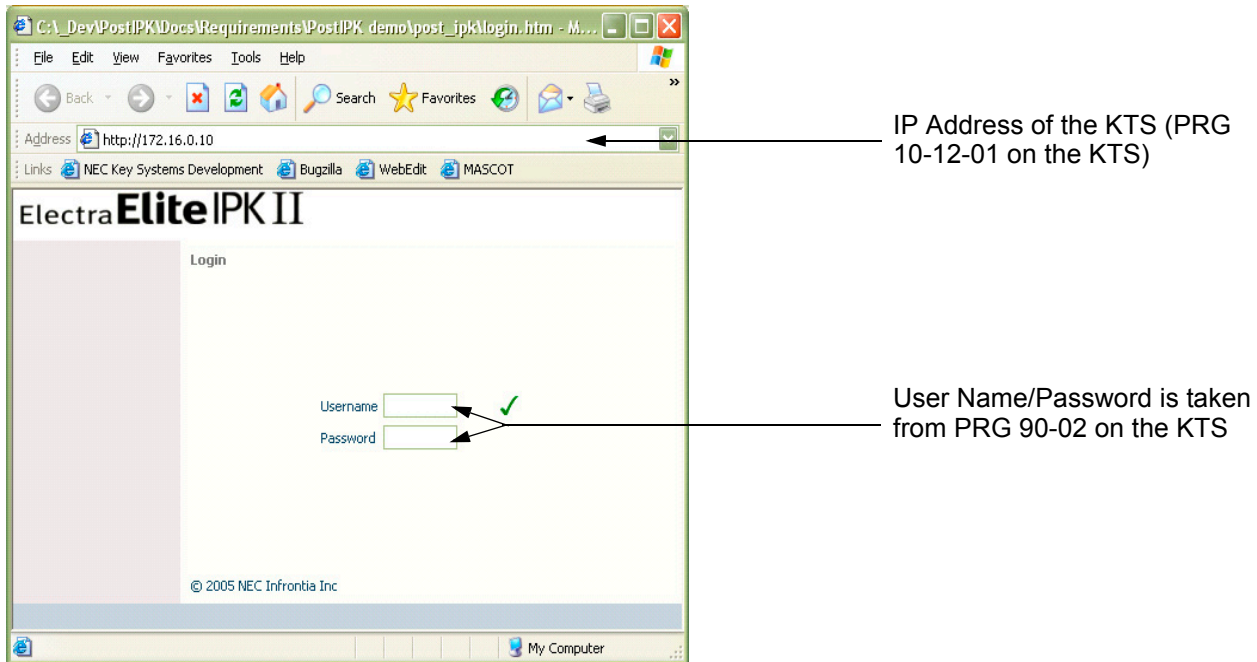


Figure 2-22 WebPro Login Screen

Conditions

- The hardware/software requirements for the host PC running the PCPro application are:

Item	Requirement
CPU	Pentium III 598MHz (minimum) Pentium 4 2.5GHz (recommended)
Memory	128 MB of RAM, 256 MB (recommended)
Disk Space	25MB (minimum)
Operating System (OS)	MS Windows 2000 or higher Windows XP (recommended)
Browser	MS Internet Explorer 6.0
Network	IP and/or RS232 connection to the KTS

Item	Requirement
TCP Port	If using IP connections, PCPro needs TCP port 8000 to be open between the KTS and the host PC, because communication between PCPro and the KTS occurs on this port.
Screen Resolution	800 x 600 (minimum) 1024 x 760 (recommended)

- The hardware/software requirements for the host PC running WebPro are:

Item	Requirement
Browser	MS Internet Explorer 6.0 (or higher) Mozilla Firefox 1.0.3 (or higher)
Network	IP connection to the KTS
Screen Resolution	800 x 600 (minimum) 1024 x 760 (recommended)

- You can have a maximum of four users logged into WebPro anytime.
- You can have up to two phones in programming mode anytime.
- You can have four WebPro users and two phone programming users logged in at the same time for a **total of six users** in programming mode simultaneously. However, the two phone programming users do not show up in session management in WebPro.
- PCPro can be logged in with only one user. This is allowed only if no other users are logged into programming mode (PCPro, WebPro, or Phone). Also, if a user is connected to the switch via PCPro, no other user can log in through PCPro, WebPro, or Phone Programming.
- Only one PCPro/WebPro/Handset can be programming the switch anytime.
- When programming via WebPro/PCPro, some data requires you to logout before the switch fully applies the changes. These programs are: 10-21-2, 11-02 (for directory dial), 11-04 (for directory dial), 13-04 (for directory dial), 14-04, 15-05, 15-15, 16-02, 23-02, 31-02, 41-02, 41-17, 47-02, 47-03, 82-11 and 83-11.

- To access the modem over K-CCIS, route the modem access service code to the target switch. Do not call a station that is call forwarded to the service code. When accessing the modem over K-CCIS, enter the service code to be dialed in PC Pro.
 - ✎ PC Pro follows the PC dialing properties. If dialing a service code, you must turn off the dial 9 for outside line and area code inclusion or PC Pro dials these digits as well.
- Some program items require second initialization of the KTS before they take effect. These programs are: 10-12-01, 10-12-02, 10-12-03, 10-12-04, 10-13-01, 10-13-02, 10-13-03, 10-14, 10-15, 10-16-01, 10-16-02, 10-16-03, 10-16-04, 20-01-03, 47-01-01, 80-01, 80-02-01, 80-02-02, 80-02-03, 80-02-04, 80-03, 80-04, 84-03-01, 84-03-02, 84-03-06, 84-03-07, 84-03-08, 84-05-01, 84-05-02, 84-06-01, 84-06-02, 84-06-03, 84-06-04, 84-06-05, 84-06-06, 84-06-07, 84-06-08, 84-06-09, 84-06-10, 84-06-11, 84-09 and 84-10.

Table 2-30 PC Pro to CPUII Version Comparison Chart

		PC Pro Version				
		1.00	1.10	1.50	2.00	2.10
CPUII Version	1000	X	X	X	X	X
	1100	-	X	X	X	X
	1500	-	-	X	X	X
	1600	-	-	-	-	X
	2000	-	-	-	X	X
	2100	-	-	-	-	X

X = PC Pro and CPUII Version is compatible
 - = PC Pro and CPUII Version is not compatible

Default Setting

None

System Availability

Terminals

Not Applicable

Required Component(s)


Each copy of installed PC Programming requires a license.

Related Features

None

Programming

- **10-12-01 CPUII Network Setup – IP Address**
Assign the CPU IP address if PC Programming is required via LAN. This program item must be set if WebPro access is required.
- **10-12-02 CPUII Network Setup – Subnet Mask**
Assign the CPU Sub Net Mask address if PC Programming is required via LAN. This program item must be set if WebPro access is required.
- **10-12-03 CPUII Network Setup – Default Gateway**
Assign the CPU Gateway address if PC Programming is required via LAN. This program item must be set if WebPro access is required.
- **10-21-02 CPUII Hardware Setup – Baud rate for COM Port**
Assign the Baud rate for COM Port 1 (Side of KSU) COM 1, (Default 2, 19200). If a direct or remote (modem) connection is required via PC Programming this program must be set.
- **11-15-14 Service Code Setup, Administrative (For Special Access) – Modem Access**
Assign the service code to be used to access the internal modem on the CPUII.
- **22-02-01 Incoming Call Trunk Setup**
Define the service type for the trunk intended to access the internal modem as 4:DIL.
- **22-07-01 DIL Assignment**
Assign the Modem Access service code set in PRG 11-15-14 as the destination extension for the DIL trunk for modem access.
- **90-02 Programming Password Setup**
Configure the administrator accounts that are used when connecting to the KTS via PCPro/ WebPro
If using PCPro, these are the accounts used to *connect*.
If using WebPro, these are the accounts that are used to login.

 *If calls are answered by an Auto attendant first, instead of the DIL setup on PRG 22-01 and PRG 22-07, set the transfer destination in the Auto Attendant to the Modem Access Service Code.*

- ➔ **90-26 Program Access Level Setup**
 Define access levels to each program. This program defines which administrator accounts in PRG 90-02 can access the program. If a program is not accessible, it does not appear in PCPro/ WebPro.

Operation

Refer to the Electra Elite IPK II PC Programming Manual for details of operation.

PCPro and WebPro Comparison

The table below gives a quick feature comparison of PCPro and WebPro. For further details refer to the Electra Elite IPK II PC Programming Manual.

Feature		Feature Application		Comments
		PCPro	WebPro	
Installation Program		Y	–	
File Handling	File New/Open/Save/Save As	Y	–	
	File Properties	Y	–	PCPro supports save/view/modify KTS Site Information, password protect files, add notes, connection settings.
	Version Conversion	Y	–	PCPro can convert databases between different KTS versions.
Programming Modes	Offline	Y	–	Ability to program offline and upload to the KTS at a later date.
	Live Update	–	Y	Changes made in WebPro apply immediately. Upload is not required.
Remote Connection	Upload	Y	–	PCPro downloads the database from the KTS to allow backups.
	Download	Y	–	
	Connection Accounts	Y	–	PCPro supports Direct, Modem and IP connections. WebPro supports only IP.
Accounts		Y	Y	WebPro: Refer to PRG 90-02 in the Programming Manual.

Feature	Feature Application		Comments	
	PCPro	WebPro		
Programming	Screen Help Text: System Data Help Text	Y	Y	Help in WebPro is more simplified than in PCPro.
	Control Hint Text	Y	Y	
	Smart Links	Y	–	WebPro has more simplified links than PCPro.
	Smart Labels	Y	Y	WebPro has more simplified labels than PCPro.
	Smart Controls	Y	–	WebPro has more simplified controls than PCPro.
	Validation	Y	Y	
	Multi-Assignments: Extension Numbers	Y	–	PCPro provides special screens that allow multiple values to be set easily. This applies mainly to table data. These screens shorten the programming time.
	Line Keys (CAP)	Y	–	
	Line Keys (General)	Y	–	
	Account Codes	Y	–	
	Defaults: View	Y	–	
	Copy: System Data Level	Y	Y	Copy items in an individual program.
	Group Level	Y	Y	Copy data for ports (telephone/trunk).
Modification Tracking (See also Modification History.)	Y	–	PCPro keeps track of changes made to a database. This includes: <ul style="list-style-type: none"> 1. Changes made to a database that are not yet saved. 2. Changes made to database that are not yet uploaded. 	
Wizards	Y	Y		
Configuration Screens	Card Configuration	Y	–	PCPro provides special screens that shorten the programming time to setup core KTS features.
	Class of Service	Y	–	
	Night Mode Switching	Y	–	
	Trunk Access Maps	Y	–	
	Trunk Groups	Y	–	
	Department Groups	Y	–	
	Direct Inward Dialing	Y	–	
	Ring Groups	Y	–	
	Timers (Trunk/Telephone)	Y	–	

Feature		Feature Application		Comments
		PCPro	WebPro	
QuickSearch		Y	Y	WebPro has a simplified search facility. It applies only to programs. PCPro provides extensive searching on programs, Wizards and IPK cross-referencing.
Reports	System Data	Y	–	PCPro can generate various reports based on values in the database.
	Verify	Y	–	
	Maintenance	Y	–	
	CAP Keys	Y	–	
	Numbering Plan	Y	–	
	Class of Service	Y	–	
	Modification History	Y	–	
Simulators	LCR/ACR	Y	–	
Import/Export	Speed Dials	Y	–	PCPro allows import/export of speed dials (csv file). It can also import converted IPK databases.
	IPK Converted File	Y	–	
Program Help	Help Pages	Y	Y	WebPro has more simplified help than PCPro.
	Context Sensitive Help	Y	Y	
Security	Application Login	Y	Y	User name/password protection to login to PCPro/WebPro.
	KTS Connection Login	Y	–	PCPro connections to a KTS are user name/password protected.
	File Open	Y	–	You can password protected a PCPro saved database.
Debug/Capture	CPU Debug Capture	Y	–	PCPro provides a tool for capturing debug information from the KTS CPU.
	SMDR Capture	Y	–	PCPro provides a tool for capturing SMDR reports from the KTS.
Modification History		Y	–	PCPro keeps a running list of all modifications made to a system databases. It also tracks uploads/downloads.
System Initialize		Y	Y	This is the ability to initialize the KTS.
System Time Setting		Y	Y	This is the ability to set the time on the KTS.

Feature		Feature Application		Comments
		PCPro	WebPro	
Software Updates	Firmware Upload	Y	–	The KTS CPU firmware can be upgraded via PCPro.
Licensing / Feature Activation	PCPro Registration	Y	–	PCPro requires product registration.
	KTS Feature Activation	Y	Y	Licensed KTS features can activated via PCPro/WebPro. You can also see what is licensed.

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Power Failure Transfer

Description

Power Failure Transfer ensures that a customer has access to the Central Office network during a power outage. The CO/PBX tip and ring are automatically transferred to the time and ring of a preselected single line telephone. The single line telephone can function in the system during normal operation or be used during a power failure.

Conditions

- The installed single line telephone must provide a dialing signal accepted by the outside exchange (Dial Pulse or Dual Tone Multifrequency).
- Only single line telephones cross-connected at the MDF can be used for this feature.
- Single line telephones and outside lines connected during power failure are fixed one-to-one.
- Single line telephones must have a ground start button for use with Ground Start Trunks.
- System features cannot be activated from single line telephone when Power Failure Transfer is in operation.
- When power is restored to the system, Power Failure Transfer is cancelled. Calls in progress on Power Failure Transfer lines are disconnected.
- Refer to the Electra Elite IPK II System Hardware Manual for the MDF Pin Numbers and PFT Connections (Power Failure Transfer Relay 1).

Default Setting

None

System Availability

Terminals

Single Line Telephones

Required Component(s)

SLI(4)/(8)-U() ETU

Related Features

None

Programming

- ➔ **14-02-09 Analog Trunk Data Setup – Busy Tone Detection**
 - 0 = Disable (default)
 - 1 = Enable
 - (default: 0)

- ➔ **14-02-14 Analog Trunk Data Setup – Loop Start/Ground Start**
 - 0 = Loop Start (default)
 - 1 = Ground Start

Operation

None

Prime Line Selection

Description

Prime Line Selection allows a multiline terminal user to place or answer a call over a specific trunk by lifting the handset. The user does not have to press a key or dial a code. This simplifies handling calls on a frequently used trunk.

Prime Line Selection has the following two modes of operation:

Outgoing Prime Line Preference

Lift the handset to seize the Prime Line. Outgoing Prime Line Preference helps a telemarketer who always needs a free line to call prospective clients. The telemarketer just lifts the handset and the Prime Line is always available. (Outgoing Prime Line Preference may be affected by Incoming Prime Line Preference – refer to the Programming section of this feature.)

Incoming Prime Line Preference

When the Prime Line rings the extension, lift the handset to answer the call. Incoming Prime Line Preference could benefit the Service Department dispatcher who must quickly answer customer's service calls and then dispatch repair technicians. Dispatchers have the assurance that when a customer calls in, they can lift the handset to get their call. (Incoming Prime Line Preference can optionally seize an idle line appearance – Refer to the Programming section of this feature.)

Conditions

- Prime Line Selection can be assigned for Electra Elite IPK II Wireless – DECT and single line telephones (Analog 500/2500), however, the telephones cannot access ICM dial tone.
- Direct Inward Lines (DILs) and Direct Inward System Access (DISA) calls also ring extensions directly, even if they are not assigned in ring group programming.
- Prime Line Selection directly interacts with line preference.

Default Setting

Disabled

System Availability

Terminals

Any Station

Required Component(s)

None

Related Features

Central Office Calls, Placing

Direct Inward Dialing (DID)

Direct Inward Line (DIL)

Direct Inward System Access (DISA)

Line Preference

Programming

- **14-05-01 Trunk Group – Trunk Group Number**
Assign Prime Line to a separate trunk group for outgoing Prime Line selection. (Also refer to PRG 14-06 and PRG 21-02.)
- **14-06-01 Trunk Group Routing – Priority Order Number**
Set up outbound route for trunk group that contains the Prime Line. (Also refer to PRG 14-05 and PRG 21-02.)

Trunk Group Routing: 1~100
- **14-07-01 Trunk Access Map Setup – Trunk Port Number**
For outgoing Prime Line selection, assign each Prime Line trunk to a different Access Map and deny outbound access to all trunks except the Prime Line trunk.

Trunks: 1~200
- **15-01-02 Basic Extension Data Setup – Outgoing Trunk Line Preference**
Enter 1 for this option so extension user seizes Prime Line when they lift the handset.

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-
- ➔ **15-02-10 Multiline Telephone Basic Data Setup – Ringing Line Preference for Trunk Calls**
Enter 1 if lifting the handset should answer ringing Prime Line; enter 0 to seize idle line appearance. (default: 1)

 - ➔ **15-06-01 Trunk Access Map for Extensions**
Set assignment so extension(s) can have access to Prime Line. Deny outbound access to extensions that should not have Prime Line.

Trunks: 1~200

 - ➔ **21-02-01 Trunk Group Routing for Extensions**
Assign extension(s) to a Prime Line route for outgoing Prime Line access.

Trunk Groups: 1~100

 - ➔ **22-01-01 System Options for Incoming Calls – Incoming Call Priority**
Set incoming Prime Line preference. Enter 1 to answer ringing Prime Line; enter 0 to answer ringing Intercom call. (default: 1)

 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign extension(s) to a ring group that consists of a Prime Line.


Ring Groups: 1~100

 - ➔ **22-05-01 Incoming Trunk Ring Group Assignment**
Assign a Prime Line to a ring group.


Ring Groups: 1~100
(default: 1)

Operation

To place a call on your Prime Line:

1. Lift the handset.
 -  *You hear dial tone on your Prime Line.*

To answer a call on your Prime Line:

1. Lift the handset.
 -  *Depending on your Line Preference programming, you either answer the Prime Line or get dial tone on the idle line appearance.*

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Private Line

Description

A Private Line is a trunk reserved for a multiline terminal for placing and answering calls. A user with a Private Line always knows when important calls are for them. Additionally, the user has their own trunk for placing calls that is not available to others in the system.

Conditions

- Incoming Only – The multiline terminal has a Private Line only for incoming calls. The user cannot place calls on the Private Line.
- Outgoing Only – The multiline terminal has a Private Line only for outgoing calls. The Private Line does not ring for incoming calls.
- Both Ways – The multiline terminal has a Private Line for both incoming and outgoing calls.
- Private Lines do not follow Call Forwarding if not Direct Inward Line (DIL).
- Other programmed options for outgoing calls also affect a Prime Line.
- Calls to extensions with DND active do not follow Call Forwarding programming. Direct Inward Line (DIL) calls ring an idle Department Group member, then follow 22-08 programming then 22-05 programming.
- An extension user can have Line Preference options applied to their Private Line.
- A Private Line can also be a Prime Line.
- You should always program a line key for each Private Line.
- Private Lines are available on single line telephones.
- Private Lines follow normal Toll Restriction.
- An extension user can transfer their Private Line. Since other users have hold access, the destination can answer the transferred Private Line and place it on Hold.
- NEC does not recommend assigning ringdown to a private line.

Default Setting

Disabled

System Availability

Terminals

AI Stations

Required Component(s)

None

Related Features

Call Forwarding

Central Office Calls, Placing

Do Not Disturb

Line Preference

Prime Line Selection

Programmable Function Keys

Single Line Telephones, Analog 500/2500 Sets

Toll Restriction

Transfer

Voice Mail Message Indication on Line Keys

Programming

- ➔ **14-07-01 Trunk Access Map Setup – Trunk Port Number**
Assign Private Line to the Private Line Access Map (refer to Program 15-06 in this section). Use option 5 for Incoming, option 7 for Both Ways and option 4 for Outgoing. In all other Access Maps, assign option 3 to the Private Line.

Trunk Access Maps: 1~200

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-
- **14-01-09 Basic Trunk Data Setup – Private Line**
Determine if a trunk should be used as a normal (0) or private (1) line.
(default: 0)
 - **15-06-01 Trunk Access Map for Extensions**
Assign extension to have Private Line to an unused Private Line Access Map.

Trunk Access Maps: 1~200
 - **15-07-01 Programmable Function Keys**
Make sure extension has a line key (e.g., 012) for the Private Line.

Trunks: 1~200
 - **21-02-01 Trunk Group Routing for Extensions**
Change the routing as needed.
 - **22-02-01 Incoming Call Trunk Setup**
Set the Trunk Service Type to 4 if routing unanswered Private Lines to voice mail or 0 if not routing to voice mail.

Ring Groups: 1~100
 - **22-04-01 Incoming Extension Ring Group Assignment**
Assign extension to Private Line ring group. Set the ringing in Program 22-06 – use option 1 for Incoming or Both Ways Private Lines. Use option 0 for Outgoing Private Lines. Do not assign any other extensions to the Private Line ring group.
 - **22-05-01 Incoming Trunk Ring Group Assignment**
Assign Private Line to an unused Private Line ring group (i.e., a ring group just for the Private Line)
(default: 1)

Ring Groups: 1~100
 - **22-07-01 DIL Assignment**
If routing unanswered Private Lines to voice mail, assign DILs to the extensions.

Operation

To place a call on your Private Line:

1. Press **Private Line** key and then press **Speaker** or lift the handset.
2. Dial the number.

To answer a call on your Private Line:

1. Press **Private Line** key and then press **Speaker** or lift handset.


To place a call from your Multiline Terminal on you Private Line:

1. Press the **Private Line** key, then press **Speaker** or lift the handset.
2. Dial the number.

To answer a call from your Multiline Terminal on your Private Line:

1. Press **Private Line** key or press **Speaker** or lift handset.

To place a call on your Private line from a single line telephone:

1. Pick up handset.
 *Private Line dial tone is heard.*
2. Dial the number.

To answer a call on you Private Line from a single line telephone:

1. Lift the handset.

Programmable Function Keys

Enhancements

Version 1500 or higher:

- Allows DSS/One-Touch keys to be used for one-touch transfer without using Transfer.
- DSS keys distinguish whether the telephone is set for DND/Call Forward All Calls or if the telephone is off-hook.

In **Version 1600 or higher**, when a Ring Group call rings a station, a BLF Indication for this station shows idle or busy based on a new Class of Service option (20-13-49). Prior to this release, the BLF Indication showed busy.

Description


Each multiline terminal has Programmable Function Keys. Programmable Function Keys simplify placing calls, answering calls and using certain features. You can customize the function of a multiline terminal's programmable keys from each multiline terminal. Depending on your telephone style, you can have either 4, 8, 16 or 32 Programmable Function keys.


Conditions

- When a key is programmed using service code 752, it cannot be programmed with a function using the 751 code until the key is undefined (000). For example with a Park Key programmed by dialing 752 + *04 must be undefined by dialing 000 before it can be programmed as a Voice Over key by dialing 751 + 48.
- Using Program 92-01 to copy a multiline terminal Programmable Function Keys copies all the keys whether or not they exist on the telephone to which the programming is being copied. This may cause confusion when trying to define a key which is already defined but which does not exist on the telephone (displays as DUPLICATE DATA). It is recommend to either clear these non-existent keys or to only copy from an extension which has the same or fewer number of keys than the extension to which the programming is being copied.
- When using Program 15-07-01 to program a D16(LD)-R ADM, regardless of the type of multiline terminal connected, *start programming the D16(LD)-R ADM keys at key number 17*. Service codes 751 and 752 can also be used to program these keys.
- Speed Dialing and One-Touch Calling also offer quick access to calls and features.
- Programming a 60-button console requires separate programming.

- If the feature key is not listed below, the LCD shows ALL-BLANK. (PRG 15-07-01 Line Key Assign).

Function Number	Function	Display
00	None	[All Blank]
01	DSS/One-Touch	DSS (xxxxxxxx xxxxxxxx = Extension Number
02	Microphone Key (ON/OFF)	MIC
03	DND Key	DND
04	BGM (ON/OFF)	BGM
05	Headset	HSET
07	Conference Key	CONF
10	Call Forward – Immediate	CFA
11	Call Forward – Busy	CFB
12	Call Forward – No Answer	CFNA
13	Call Forward – Busy/No Answer	CFBNA
14	Call Forward – Both Ring	CFBOTH
15	Follow Me	FLWME

 If a key is programmed as a DSS/One-Touch key for a station that is set for Call Forward All Calls or Do Not Disturb, the DSS/One-Touch key flashes. If a station is busy, the DSS/One-Touch for that station does not light solid, even if Call Forward All Calls or DND is set.

 Refer to the Electra Elite IPK II Programming Manual for a complete list of Function Numbers.

- One-Touch keys programmed for Park Hold Service Code cannot be used to park calls without first using Hold or Transfer.
- Pauses can be entered in the dial string of a DSS/One Touch button. The pause is entered as P in the dial string and causes the system to wait three seconds before sending the rest of the digits that follow the P (pause). Multiple pauses can be entered.
- The @ symbol can be entered in the dial string of a DSS/One Touch button. The @ only applies to ISDN and Intercom calls. When using @, the system waits for the destination to answer (answer supervision), and then sends the rest of the digits.
- Entering a P (pause) in a DSS/One Touch dial string can be used for CO calls, Intercom calls, or after @ for ISDN calls.

Default Setting

The first eight keys on a telephone are line keys (e.g., key 1 = line 001). The remaining keys are unassigned.

System Availability

Terminals

All Multiline Terminals

Required Component(s)

D16(LD)-R ADM

Related Features

Direct Station Selection

One-Touch Calling

Speed Dial – System/Group/Station

Programming


- ➔ **15-07-01 Programmable Function Keys**
Assign the functions of a multiline terminal's Programmable Function Keys. When using Program 15-07-01 to program *the function keys 17~32 on the D16(LD)-R ADM start with key 17*. Service codes 751 and 752 can also be used to program these keys.
- ➔ **15-20-01 LCD Line Key Name Assignment**
The name can be entered using the dial pad keys 1~32.
- ➔ **20-06-01 Class of Service for Extensions**
Assign Class of Service (1~15) to extensions.
- ➔ **20-07-10 Class of Service Options (Administrator Level) – Programmable Function Key Programming (Appearance Level)**
Enable (1)/Disable (0) an extension user ability to program their own function keys using service code 752.

📎 *When programming a feature as a Programmable Function Key, refer to PRG 15-07-01 in the Electra Elite IPK II Programming Manual.*
- ➔ **20-13-49 Class of Service for Options (Supplementary Service) – BLF Indication on CO Incoming State (Version 1600 or higher required)**
Turn Off (0) or On (1) the BLF Indication on CO Incoming State.

Operation

To change a 2-digit programmable key:


1. Press **Speaker**.
2. Dial **751** for 2-digit codes.
3. Press the key you want to program.
4. Enter the 2-digit key function, any additional information needed for the key and press **Hold**.


 For available functions codes refer to PRG 15-07 in the Electra Elite IPK II Programming Manual.


 To undefine a key, enter 00.

To change a 3-digit programmable key:

1. Press **Speaker**.
2. Dial **752** for 3-digit codes.
3. Press the key you want to program.
4. Enter the 3-digit key function and any additional information needed for the key.


 For available functions codes, refer to PRG 15-07 in the Electra Elite IPK II Programming Manual.

 To undefine a key, enter 000.

 When a key is programmed using service code 752, that key cannot be programmed with a function using the 751 code until the key is undefined (000). For example with a Park Key programmed by dialing 752 + *04 must be undefined by dialing 000 before it can be programmed as a Voice Over key by dialing 751 + 48.

To check the function of a programmable key:

1. Press the **Help** key.
2. Press the programmable key.

 The programmed function displays.

Programming from a Multiline Terminal

Enhancements

<p>Version 1100 or higher provides a temporary license (10 days) for the following: ACD, CTI, firmware upgrade, Hotel/Motel, 256 MEGACO stations and SMDR.</p>

Description

System Programming can be performed from any display multiline terminal. Most programming changes become effective immediately. Other programming changes become effective after the data is backed up from temporary memory to permanent memory.

Enhancements

Version 1100 or higher provides Temporary Licensing. Features that require a license before they can function can be licensed temporarily for up to 10 days. The temporary license turns on ACD, CTI, Firmware Upgrade, Hotel, 256 MEGACO stations, and SMDR.

Conditions

- You can have up to two telephones in programming mode anytime.
- You can have a maximum of four users logged into WebPro anytime.
- You can have four WebPro users and two phone programming users logged in at the same time for a **total of six users** in programming mode simultaneously. However, the two phone programming users do not show up in session management in WebPro.
- PCPro can be logged in only with one user. This is only allowed if no other users are logged into programming mode (PCPro, WebPro, or Phone). Also, if a user is connected to the switch via PCPro, no other user can log in through PCPro, WebPro, or Phone Programming.
- Programming from a Multiline Terminal can require a password to enter programming.
- Temporary License can be activated only from a Multiline Terminal, not PCPro or WebPro.
 - ❑ When activated, the system is temporarily licensed for ACD, CTI, Firmware Upgrade, Hotel, 256 MEGACO stations, and SMDR.
 - ❑ Each time the temporary license is activated, the program is read only until the temporary license expires.
 - ❑ Each time the temporary license expires, it can be set again for up to 10 additional days.

- After setting a number of days in the program, subsequent days show one less as it counts down to expiration.
- When PRG 90-37-01 shows 1, the license expires at midnight on that day. When the license expires, the system resets.
- If the date is changed in PRG 10-01-01 while the temporary license is in effect, one day is subtracted from the license period.
- If the date is changed in PRG 10-01-01 when the temporary license shows 1 day, the system resets when it is applied (transfer), not when exiting programming mode.

Default Setting

Enabled

System Availability

Terminals

Multiline Terminals with Display

Required Component(s)

- ESI(8)-U() ETU
- ESIB(8)-U() ETU
- ESIE(8)-U() ETU

Required Software

Version 1100 or higher is required for temporary licensing.

Related Features

PC Programming

Programming

- ➔ 90-02 **Programming Password Setup**
- ➔ 90-37 **Temporary License**

Operation

Refer to the Electra Elite IPK II Programming Manual for additional information.

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Pulse to Tone Conversion

Description

An extension can use Pulse to Tone Conversion on trunk calls. Pulse to Tone Conversion lets a user change their extension dialing mode while placing a call. For systems in a Dial Pulse area, this permits users to access dial-up OCCs (Other Common Carriers – such as MCI) from their DP area. The user can, for example:

- Place a call to an OCC over a DP trunk.
- Depending on programming:
 - Manually implement Pulse to Tone Conversion
 - OR -**
 - Wait 10 seconds.
- Dial the OCC security code and desired number. The system dials the digits after the conversion as DTMF.

Conditions

Pulse to Tone Conversion is valid only for Dial Pulse trunks (Program 14-02-01, options 0 or 1).

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

COI(4/8)-U10 ETU, COID(4/8)-U10 ETU, COIB(4/8)-U() ETU

- OR -

TLI(2)-U() ETU

- OR -

DTI-U() ETU assigned for CO or Tie Line mode

Related Features

Central Office Calls, Placing

Multiple Trunk Types

Programming

- ➔ **14-02-07 Analog Trunk Data Setup – DP to DTMF Conversion Options**
For each trunk, set the type of DP to DTMF Conversion required: automatic (0), automatic and manual (1), or manual (2).

Operation

To convert your telephone's dialing to tone after placing your call on a pulse line:

1. Place a call over pulse line.
2. Dial # to switch the DP trunk to DTMF dialing.

Quick Transfer to Voice Mail

Enhancements

With **Version 1600 or higher**, a call which is initiated or answered on a Virtual Extension (talking on the key) can be quick Transferred to Voice Mail. Prior to this release, a Quick Transfer to Voice Mail was not allowed.

Description

A station user transferring a call can transfer the call to the called party voice mail box after an internal station number is dialed while performing a screened transfer, or during intercom calls.

Conditions

The Quick Transfer to Voice Mail feature is allowed when:

- Listening to the Ring Back Tone (RBT)
- Listening to the Call Waiting Tone (CWT)
- In Handsfree Answerback Mode
- In Voice Over Mode
- When Quick Transfer to Voice Mail is accessed, the Voice Over feature is canceled.
- While on an intercom (ICM) call, dial the Quick Transfer Access Code (default: 8) to automatically transfer to that station Voice Mail box.

The Quick Transfer to Voice Mail is not allowed when caller is:

- Listening to the busy tone (BT)
- Talking on an internal line
- Talking on an outside line
- Making a conference call



Default Setting

Not Enabled

System Availability

Terminals

All Terminals

Required Component(s)

Any supported voice mail

Related Features

Digital Voice Mail


IPK II In-Mail

Voice Mail Integration (Analog)

Programming

- ➔ **45-01-01 Voice Mail Integration Options – Voice Mail Department Group Number**
Assign which Extension (Department) Group number is to be assigned as the voice mail group. An entry of 0 means no voice mail is installed.

Department Groups: 0, 1~64

 *For complete voice mail system programming refer to programming section of the applicable voice mail.*

- ➔ **45-01-14 Voice Mail Integration Options – CCIS Centralized Voice Mail Number**
Assign the pilot number to Centralized Voice Mail over K-CCIS Link. This is assigned only in remote switches. If using this program 45-01-01 should be 0, 1~64.

Operation

To Quick Transfer a call while talking with an outside or internal party:

1. Press **Transfer**, and wait for an internal dial tone.
2. Enter a station number, and wait for a ringback tone.
3. Dial the Quick Transfer Access Code (default: **8**).
4. Hang up.

 *The Voice Mail answers and the outside party is transferred to the station user's Voice Mail box.*

To leave a message using Quick Transfer to voice mail during an intercom call:

1. Make the intercom call.
2. Dial the Quick Transfer Access Code (default: **8**).
3. Leave a voice mail message.
4. Hang up.

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Redial Key

Description

Users can press the Redial Key to cycle through the last 10 outside numbers dialed. Press the # key to redial the number displayed. Users can also press the Redial Key and dial a System Speed Dial bin number to access System Speed Dial.

Conditions

- Redial List requires using a display telephone.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Last Number Redial

Speed Dial – System/Group/Station




Programming

None

Operation



To redial the last number dialed:

1. Press **Redial**.
 *REDIAL [#] SYS is displayed along with the last dialed number.*
2. Dial the # key.

To scroll through the last 10 outside numbers dialed:

1. Press **Redial**. Each time the Redial key is pressed, it displays the next most recently dialed number.
2. When the desired number is displayed, press the # key.

To access a System Speed Dial bin:

1. Press **Redial**.
 *REDIAL [#] SYS is displayed.*
2. Dial the System Speed Dial bin number.
 *The number stored in that bin is displayed for your preview.*
3. Press **Speaker** to dial the number.
- OR -
Press an idle trunk line key.

Remote (System) Upgrade

Description

With PC Programming, the Electra Elite IPK II can be remotely upgraded to a newer version of main system software. When a new version of main system software is released, a firmware package file is provided. Using either the WebPro or PCPro application, a technician can remotely upgrade the firmware on the CPUII. The upgrade can be applied immediately, or at a scheduled date and time. Remote Upgrade is supported only via a LAN connection. A modem or serial connection is NOT supported for Remote Upgrade.

Conditions

- When doing a Firmware Upgrade, the telephone system can become sluggish during the file transfer portion of the update. It is recommended to perform updates after hours, even if the update is scheduled. The file transfer happens when the update is set. For example, at 2:00PM a technician schedules an update to happen at 12:00AM. When he/she clicks start (2:00PM), it begins transferring the file to the Compact Flash on the CPUII. At this time the telephone system experiences sluggishness until the file transfer is complete. When the time turns to 12:00AM, the telephone system resets and switches to the new firmware.
- The Package file needed is provided by NEC at the time the new version of main system software is released.
- A Compact Flash (CF) must be connected to the CN5 connector for Remote (System) Upgrade. During the Firmware Upgrade, the Package file is copied to the CF and extracted. The system then resets and boots off the compact flash.
- Booting from the CF does not replace the firmware in Flash Memory on the CPUII.
- Up to two versions of firmware are kept on the CF card. One version is the current version that the CPU used to boot up from. The other version is the new version that is used on the next boot up. If the boot up fails when switching to the new version, the CPUII can revert back to the old version.
- The CPUII boots from its Flash Memory if no CF is installed.
- The time entered on the Firmware Upgrade screens is relative to the time on the CPUII, not the PC that PCPro or WebPro was launched from. The user should take into account time zone differences when using this feature.
- The time to upload a firmware package file is directly related to the file size. Generally it takes a few minutes.
- Remote Upgrade is supported only via an LAN connection. A modem or serial connection is not supported for Remote Upgrade.

Default Setting

At default, both PCPro and WebPro are set to *Update Immediately* after the upload.

System Availability

Terminals

Not applicable

Required Component(s)

PCPro or WebPro

Related Features

PC Programming

Programming

Refer to [PC Programming on page 2-793](#).

Operation

PCPro

1. Obtain the firmware package file from NEC.
2. Open and login to PCPro.
3. Connect to the switch.
4. Under the Communications menu, choose the **Firmware Update** option.
5. In the firmware update window, browse to the location of the Firmware Package file. For example, the file name might be MainSWv0.36.pkg.

6. Select the schedule type:
 - Immediately after upload
 - At the time...
 - *If you choose At the time..., select the date and time you want the CPU to reset and switch over to the new software version.*
7. Click **Start**. PCPro uploads the firmware package file, and updates the system at the time you specified in step 6.

WebPro

1. Obtain the firmware package file from NEC.
2. Open Internet Explorer and connect to the following URL where xxx.xxx.xxx.xxx is the IP address of the CPU.
<http://xxx.xxx.xxx.xxx>
3. Login to WebPro with a username and password defined in PRG 90-02.
4. In the Administration section of the Home page, select the Firmware Upgrade Icon.
5. On the Firmware Update page, browse to the location of the firmware upgrade package file. For example, the file name might be MainSWv0.36.pkg.
6. Select the schedule type:
 - Immediately after upload.
 - At the time...
 - *If you choose At the time..., select the date and time you want the CPU to reset and switch over to the new software version.*
7. Click **Start**. WebPro uploads the firmware package file, and updates the system at the time you specified in step 6.

For additional information, refer to the Electra Elite IPK II PC Programming Manual.

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Repeat Redial

Description

If a multiline terminal user places a trunk call that is busy or unanswered, they can have Repeat Redial try it again later on. The user does not continually have to try the number again – hoping it goes through. Repeat Redial automatically retries it until the called party answers (the number of retries is based on system programming).

Conditions

- Lifting the handset during a callout cycle cancels Repeat Redial.
- Other programmed options for outgoing calls can affect how a Repeat Redial call is placed. Refer to Central Office Calls, Placing options as needed.
- For systems with Automatic Route Selection (ARS), ARS selects the trunk for the Repeat Redial call.
- Single line telephones cannot use Repeat Redial.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Automatic Route Selection

Central Office Calls, Placing

Last Number Redial

Save Number Dialed





Single Line Telephones, Analog 500/2500 Sets

Programming

- **15-07-01 Programmable Function Keys**
Assign a function key for Repeat Redial (code 29).
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- **20-08-07 Class of Service Options (Outgoing Call Service) – Repeat Redial**
Enable (1)/Disable (0) an extension ability to use Repeat Redial.
- **21-08-01 Repeat Dial Setup – Repeat Redial Count**
Set how many times (0~255) Repeat Redial automatically repeats if the call does not go through. (default: 3)
- **21-08-02 Repeat Dial Setup – Repeat Redial Interval Time**
Set the time between Repeat Redial attempts (0~64800 seconds).
(default: 60)
- **21-08-03 Repeat Dial Setup – Repeat Dial Calling Timer**
Set the time (0~64800 seconds) the system waits for the called party to answer after a Repeat Redial. If the called party does not answer during this time, the system hangs up and tries again (after the Repeat Redial Interval Time). For unanswered calls, the total time between retries is the sum of PRG 21-08-02 and PRG 21-08-03.
(default: 30)
- **21-08-04 Repeat Dial Setup – Time for Send Busy Tone for ISDN Trunk**
Set the time (0~64800 seconds) to send out Busy Tone with an ISDN line, when called party is busy.
(default: 0)

Operation

To use Repeat Redial (if the outside party you call is unavailable or busy):

1. Place a trunk call.
 -  *Listen for busy tone or ring no answer.*
2. Press **Feature + Redial**.
 - OR -
 - Press the **Repeat Redial** key (PRG 15-07 or SC 751: 29).
 -  *Repeat Redial key flashes while you wait for the system to redial.*
3. Press **Speaker** to hang up.
 -  *The system periodically redials the call.*
4. Lift the handset when called party answers.
 -  *When using trunks with answer supervision the Repeat Redial feature automatically cancels.*

To cancel Repeat Redial:

1. Press **Feature**.
2. Press **Redial**.
 - OR -
 - 1. Press **Repeat Redial** key (PRG 15-07 or SC 751: 29).
(Also refer to [Last Number Redial on page 2-675](#).)

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Resident System Program

Description

When power is supplied to the system, the hardware configuration is scanned and Resident System Program default values are assigned including terminal types (e.g. PGD(2)-U10 ADP, DSS Console). This enables immediate operation, even before the system is programmed to accommodate the individual site requirements.

Conditions

- Default assignments for Multiline Terminals are: LK 01~LK 08 corresponds to CO 01~CO 08.
- DSS Console to Extension assignments for Attendant Add-On Consoles are not assigned.
- Default Attendant Add-On Console key assignments are:

DSS Keys = 001~060
Stations = 101~160
- First Initialization of the system returns all programming values to default. Without a PC-ATA Compact Flash Card installed, press and hold the SW1 (Load Switch) and toggle the Reset (SW2) switch. Continue to hold the SW1 switch for approximately 5~10 seconds before releasing. The system boots loading Resident System Programming.

Default Setting

None

System Availability

Terminals

Not Applicable

Required Component(s)

None

Related Features

None

Programming

- ➔ **14-02-09 Analog Trunk Data Setup – Busy Tone Detection**
0 = Disable (Default)
1 = Enable

- ➔ **14-02-14 Analog Trunk Data Setup – Loop Start/Ground Start**
0 = Loop Start (Default)
1 = Ground Start

Operation

None

Reverse Voice Over


Description

While on a call, Reverse Voice Over lets a busy multiline terminal user make a private Intercom call to an idle co-worker. The idle co-worker can be at a multiline terminal or single line telephone. The busy user can press a programmed Reverse Voice Over key to make a private call to a specified co-worker. The initial caller cannot hear the Reverse Voice Over conversation. The private Intercom call continues until the Reverse Voice Over caller can press the key again. The initial call can be an outside call or an Intercom call.

Reverse Voice Over could help a salesman, for example, when placing a call to an important client. The salesman can talk with the client and give special instructions to a secretary – without interrupting the initial call.

When the multiline terminal is idle, the Reverse Voice Over key functions the same as a Hotline or One-Touch key. A multiline terminal Reverse Voice Over key also shows at a glance the status of the associated extension:

When the key is . . .	The associated extension is . . .
Off	Idle
On	Busy or call ringing
Fast Flash	In Do Not Disturb

 *When the destination extension is idle, the Reverse Voice Over provides one button calling to the associated extension (like a Hotline key). An extension user cannot, however, use the Reverse Voice Over key to Transfer calls by one-touch operation.*

Conditions

- An extension can have Reverse Voice Over keys for more than one extension (limited only by the number of available function keys).
- When the destination extension is in Do Not Disturb, a Reverse Voice Over placed to an extension always rings, regardless of how Handsfree Answerback/Forced Intercom Ringing is set at the destination.
- When the destination extension is not in Do Not Disturb, Reverse Voice Over follows Handsfree Answerback/Forced Intercom Ringing programming.
- Reverse Voice Over requires a uniquely programmed function key.

- Reverse Voice Over is not available from single line telephones, but a single line telephone can be a Reverse Voice Over destination.
- If an extension user places a Reverse Voice Over to a busy destination extension, the system sets up a Voice Over. The Voice Over continues until the Reverse Voice Over key is pressed again.
- When a Reverse Voice Over call is placed to a destination station, while the originator is on a CO call, the Reverse Voice Over is dropped if the destination station is involved in another call and this call is terminated.

Default Setting

Disabled

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Do Not Disturb

Handsfree Answerback/Forced Intercom Ringing

Hotline

One-Touch Calling

Programmable Function Keys

Single Line Telephones, Analog 500/2500 Sets

Voice Over

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Reverse Voice Over (code 47 + destination extension).

Assign a function key for Voice Over to the destination extension (code 48). This allows the user at the destination to switch between calls if they were busy when the Reverse Voice Over was initiated.
- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Define whether a busy extension Manually (0) or Automatically (1) receives Off-Hook signals. (default: 1)
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (COS) to each extension (1~15).

Operation

When on a call:

To place a Reverse Voice Over call:

1. Press your **Reverse Voice Over** key (PRG 15-07 or SC 751: 47 + destination extension).
 - ✍ *Your Reverse Voice Over key lights steadily (red) and you can talk with the programmed Reverse Voice Over destination.*

To receive a Reverse Voice Over Call while busy:

1. Press the **Voice Over** key (PRG 15-07 or SC 751: 48).
 - ✍ *The voice over key allows you to switch back and forth between the initial call and the Reverse Voice Over Call.*


To return to your initial caller:

1. Press the **Reverse Voice Over** key again.
 - ✍ *If the co-worker you call hangs up, you return to the initial call automatically.*

When the telephone is idle:

To place a call to your Reverse Voice Over destination:

1. Press your **Reverse Voice Over** key (PRG 15-07 or SC 751: 47 + destination extension).

 *You can optionally lift the handset after this step for privacy.*

Ring Groups

Enhancements

<p>In Version 1600 or higher, when a Ring Group call rings a station, a BLF Indication for this station shows idle or busy based on a new Class of Service option (PRG 20-13-49). Prior to this release, the BLF Indication showed busy.</p>

Description

Ring Groups determine how trunks ring extensions. Generally, trunks ring extensions only if Ring Group programming allows. For example, to make a trunk ring an extension:

- Assign the trunk and the extension to the same Ring Group.
- In the extension Ring Group programming, assign ringing for the trunk.

Any number of extensions and trunks can be in a specific group. The system allows:

- Ring Groups = 1~100
- In-Skin Voice Mail = 102
- Centralized Voice Mail = 103

If an extension has a line key for the trunk, Ring Group calls ring the line key. If the extension does not have a line key, the trunk rings the line appearance key. If an extension has a key for a trunk that is not in its ring group, the trunk follows Access Map programming.

Conditions

- DIL trunks disregard ring group programming until DIL overflow.

Default Setting

All trunks are in Ring Group 1, the first 16 extensions ring for trunk calls and all other extensions only flash.

System Availability

Terminals

All Multiline Terminals and Single Line Telephones

Required Component(s)

None

Related Features

Automatic Call Distribution (ACD)

Automatic Route Selection

Direct Inward Line (DIL)

Direct Inward Dialing (DID)

Direct Inward System Access (DISA)

ISDN Compatibility

Night Service

Programmable Function Keys

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign function keys as line (code *01 + trunk number) or Call Appearance (CAP) Keys [code *08 + CAP Key orbit 0001~9999 (or 0000 for auto assign)].

Trunks: 1~200
- ➔ **20-13-49 Class of Service Options (Supplementary Service) – BLF Indication on CO Incoming State (Version 1600 or higher required)**
Turn On (1) or Off (0) BLF Indication on CO Incoming State.
(default: 0)

-
-
- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign the incoming trunk type (0) for each trunk. There is one item for each Night Service Mode.
✎ This option must be set to 0 for Ring Groups to work.
Night Service Time Mode: 1~8

 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign extensions to ring groups.

Ring Groups: 1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail

 - ➔ **22-05-01 Incoming Trunk Ring Group Assignment**
Assign trunks to ring groups.

Ring Groups: 1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
(default: 1)

 - ➔ **22-08-01 DIL/IRG No Answer Destination**
For DIL Delayed Ringing, assign the DIL No Answer Ring Group. An unanswered DIL rings this group after the DIL No Answer Time (Program 22-01-04) expires. DIL Delayed Ringing can also reroute outside calls ringing a Ring Group In-Skin/External Voice Mail, or Centralized Voice Mail.

Ring Groups: 1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
(default: 1)

 - ➔ **22-12-01 DID Intercept Ring Group (Version 1600 or higher required)**
For each DID Translation Table, assign the destination for DID Intercept. The destination can be a Ring Group, In-Skin/External Voice Mail, or Centralized Voice Mail. For each table, make a separate entry for each Night Service mode.

Ring Groups: 1~100
102 = In-Skin/External Voice Mail
(default: 1)

 - ➔ **25-03-01 VRS/DISA Transfer Ring Group With Incorrect Dialing**
Set the transfer destination for each DISA and Automated Attendant (OPA) trunk. The destination can be a Ring Group or Voice Mail. Make a separate entry for each Night Service mode.

Ring Groups: 1~100
102 = In-Skin/External Voice Mail
103 = Centralized Voice Mail
(default: 0)

✎ For incoming calls, Ring Group programming (PRG 22-04/PRG 22-05) overrides Access Map programming (PRG 14-07/PRG 15-06).
-
-

Use the charts below to program the following example:

For this extension . . . ¹			
301	Trunk 1 Rings	Trunk 2 Flashes	Trunk 3 Flashes
302	Trunk 1 Flashes	Trunk 2 Rings	Trunk 3 Flashes
303	Trunk 1 Flashes	Trunk 2 Flashes	Trunk 3 Rings

¹ Trunks ring the same in the day as at night.

Program 22-04 : Incoming Extension Ring Group Assignment			
Ring Group ^{1>}	1	2	3
Trunk 1	X	-	-
Trunk 2	-	X	-
Trunk 3	-	-	X

X = Trunk assigned to indicated Ring Group

¹ Make the same PRG 22-04 entry for all Night Service modes.

Program 22-05 : Incoming Trunk Ring Group Assignment			
Ring Group >	1	2	3
Ext. 301	1	0 ¹	0 ¹
Ext. 302	0 ¹	1	0 ¹
Ext. 303	0 ¹	0 ¹	1

1 = Extension rings

0 = Extension does not ring

¹ To allow extension user to answer flashing line, be sure to give extension incoming access to the trunk in PRG 14-07 and PRG 15-06.

Operation

Refer to [Central Office Calls, Answering on page 2-187](#).

Ringdown Extension, Internal/External

Description

With a Ringdown Extension, a user can call another extension, outside number, or Speed Dialing number by just lifting the handset. The call automatically goes through – there is no need for the user to dial digits or press additional keys. Ringdown Extensions are frequently used for lobby telephones, where the caller just lifts the handset to get the information desk or off-site Reservation Desk.

After the Ringdown Extension user lifts the handset, ringdown occurs after a programmable interval. Depending on the setting of this interval, the extension user may be able to place other calls before the ringdown goes through.

Conditions

- Ringdown extension has no effect on an extension current (active) call.
- The Ringdown Extension user can lift the handset or press the Speaker key to initiate ringdown.
- If the Ringdown/Hotline destination is a speed dial bin, the appropriate service code must proceed the bin number.
- Ringdown Extension can use Speed Dial – System/Group/Station numbers (and follow their trunk routing) as the destination number.
- Ringdown Extension follows Call Forwarding. For example, the ringdown destination can forward their calls. When the Ringdown Extension user lifts the handset, ringdown automatically calls the extension to which calls are forwarded.
- If the Ringdown Extension user hears busy tone when they lift the handset, they can Camp-On to the destination, leave a Callback or activate Off-Hook Signaling.
- The ringdown destination user can activate Do Not Disturb. When the Ringdown Extension user lifts the handset, they hear DND. If enabled, the Ringdown Extension user can override the destination's DND.
- If the destination extension has Handsfree Answerback enabled, the call voice announces. If the destination extension has Forced Intercom Ringing enabled, the call rings.
- A Call Arrival (CAR) Key or Virtual Extension can be a ringdown destination. This would allow a front door key to be programmed on every extension.
- Delayed Ringdown can occur by setting the Hotline Start Timer. However, Ringdown does not occur if the Hotline Start Timer is set longer than the Extension Dial Tone Timer.

- The @ code is used to make an outbound call automatically forward to a DISA Trunk or to VM Auto Attendant. This code can be used only on ISDN outbound calls. Internal calls and analog outbound calls are not supported.

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Call Arrival (CAR) Keys

Callback

Call Forwarding

Call Waiting/Camp-On

Do Not Disturb

Handsfree Answerback/Forced Intercom Ringing

Off-Hook Signaling

Speed Dial – System/Group/Station




Virtual Extensions

Programming

- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-08-09 Class of Service Options (Outgoing Call Service) – Hotline/Extension Ringdown**
Enable (1)/Disable (0) Hotline (Ringdown). If disabled in Class of Service, the settings in Program 21-11 below have no effect.
(default: 0)
- ➔ **21-01-09 System Options for Outgoing Calls – Ringdown Extension Timer (Hotline Start)**
After the user lifts the handset, the extension automatically calls the ringdown destination after this time (0~64800 seconds). A setting of 0 immediately rings the programmed extension. Any other setting delays the ringdown the time programmed.
(default: 5)
- ➔ **21-11-01 Extension Ringdown (Hotline) Assignment**
Program the ringdown (Hotline) source and destination (target) number, up to 24 digits (512 Hotline assignments). Remember to include the trunk access code (usually 9) in front of the number when dialing outside numbers. When programming Speed Dial – System numbers as the destination, the entry should be 753 + bin number (the service code for Speed Dialing and the Speed Dial bin number).
(default: No Setting)

Operation

To place a call if your extension has ringdown programmed:

1. Lift the handset.
 -  *If you want to place a trunk call, press a line key before lifting the handset.*
 -  *Depending on the setting of your ringdown timer, you may be able to dial an Intercom call before your ringdown goes through.*
 -  *If the destination has Handsfree Answerback enabled, your call voice announces. If the destination has Forced Intercom Ringing enabled, your call rings.*

To answer a call if you are another extension ringdown destination:

1. Speak toward telephone to answer incoming voice announcement.
- OR -
Lift the handset or press **Speaker** to answer ringing Intercom call.

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Room Monitor

Description

Room Monitor lets an extension user listen to the sounds in a co-workers area. For example, the receptionist could listen for sounds in the warehouse when it is left unattended. To use Room Monitor, the initiating extension **and** the receiving extension must activate it.

When using multiline terminals for monitoring, an extension user can Monitor only one extension at a time. However, many extensions can Monitor the same extension at the same time. However, only one single line telephone can monitor another single line telephone at a time.

Room Monitor for Single Lines

This option enables you to monitor the room status through your single line telephones. Between multiline terminals, the monitored room status is picked up by the telephone microphone and the activity is heard through the speaker of the monitoring multiline terminal. Between single line telephones, at the station to be monitored, a user goes off-hook and dials a service code and the extension number of the monitoring telephone. At the monitoring station, a user goes off-hook and dials a service code and the extension number of the monitored telephone. The activity of the area where the monitored telephone is placed can then be heard at the monitoring telephone. This service is available until the handset of the monitored telephone is placed on-hook.

Caution

The use of monitoring, recording, or listening devices to eavesdrop, monitor, retrieve, or record telephone conversation or other sound activities, whether or not contemporaneous with transmission, may be illegal in certain circumstances under federal or state laws. Legal advice should be sought prior to implementing any practice that monitors or records any telephone conversation. Some federal and state laws require some form of notification to all parties to a telephone conversation, such as using a beep tone or other notification methods or requiring the consent of all parties to the telephone conversation, prior to monitoring or recording the telephone conversation. Some of these laws incorporate strict penalties.

Conditions

- Room Monitor is for listening only. It does not allow for conversation between the monitoring and monitored extensions.
- An extension user cannot monitor an Attendant.
- A multiline terminal user cannot monitor a single line telephone and a single line telephone cannot monitor a multiline terminals.

- Call Arrival (CAR) Key (virtual extension) keys do not support Room Monitor Programmable Function keys (code 39).
- Room Monitor for single line telephones can be used with the Hotel/Motel feature.
- For a multiline terminal, Room Monitor requires uniquely programmed function keys.

Default Setting

Disabled

System Availability

Terminals

Multiline Terminals and Single Line Telephones

Required Component(s)

None

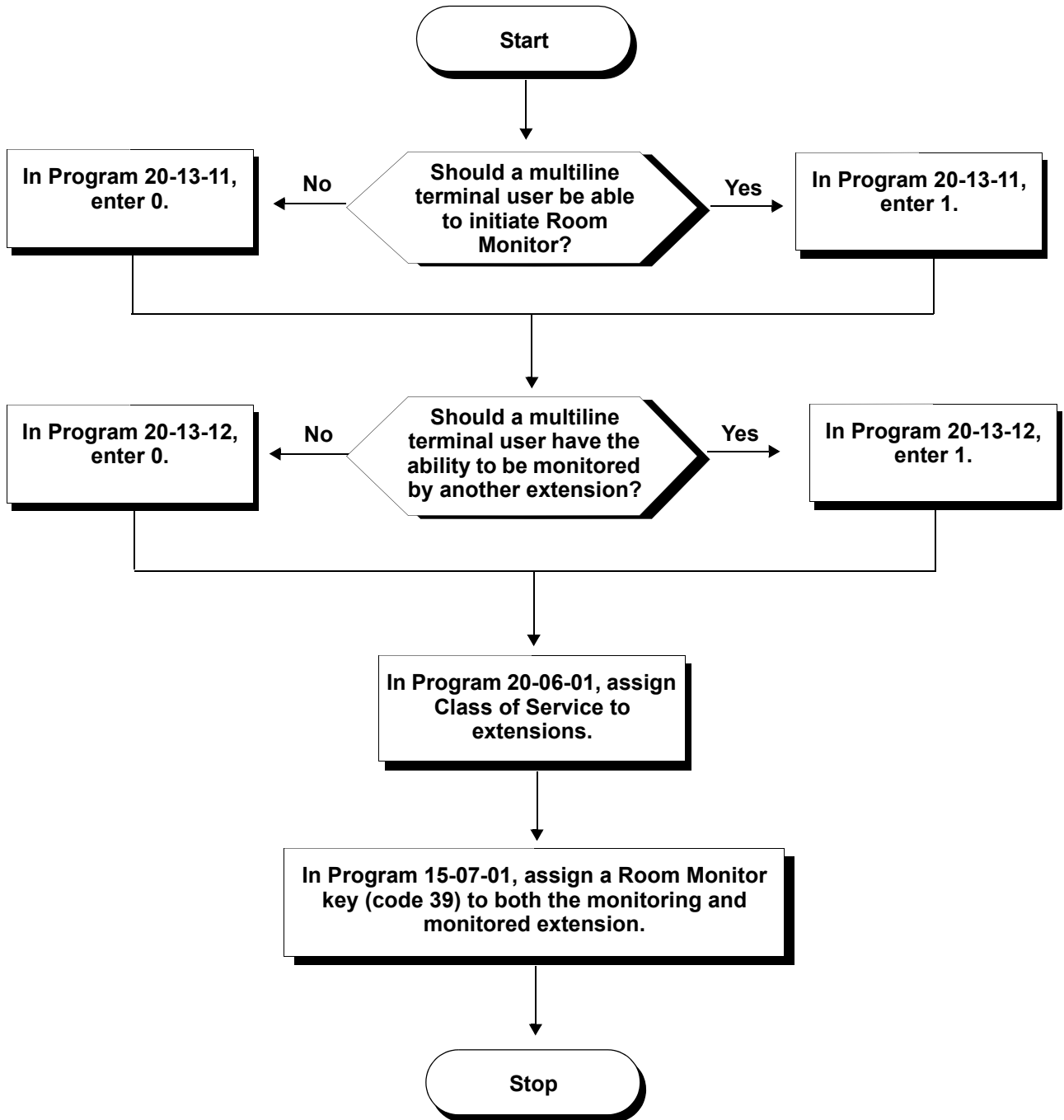
Related Features

Hotel/Motel

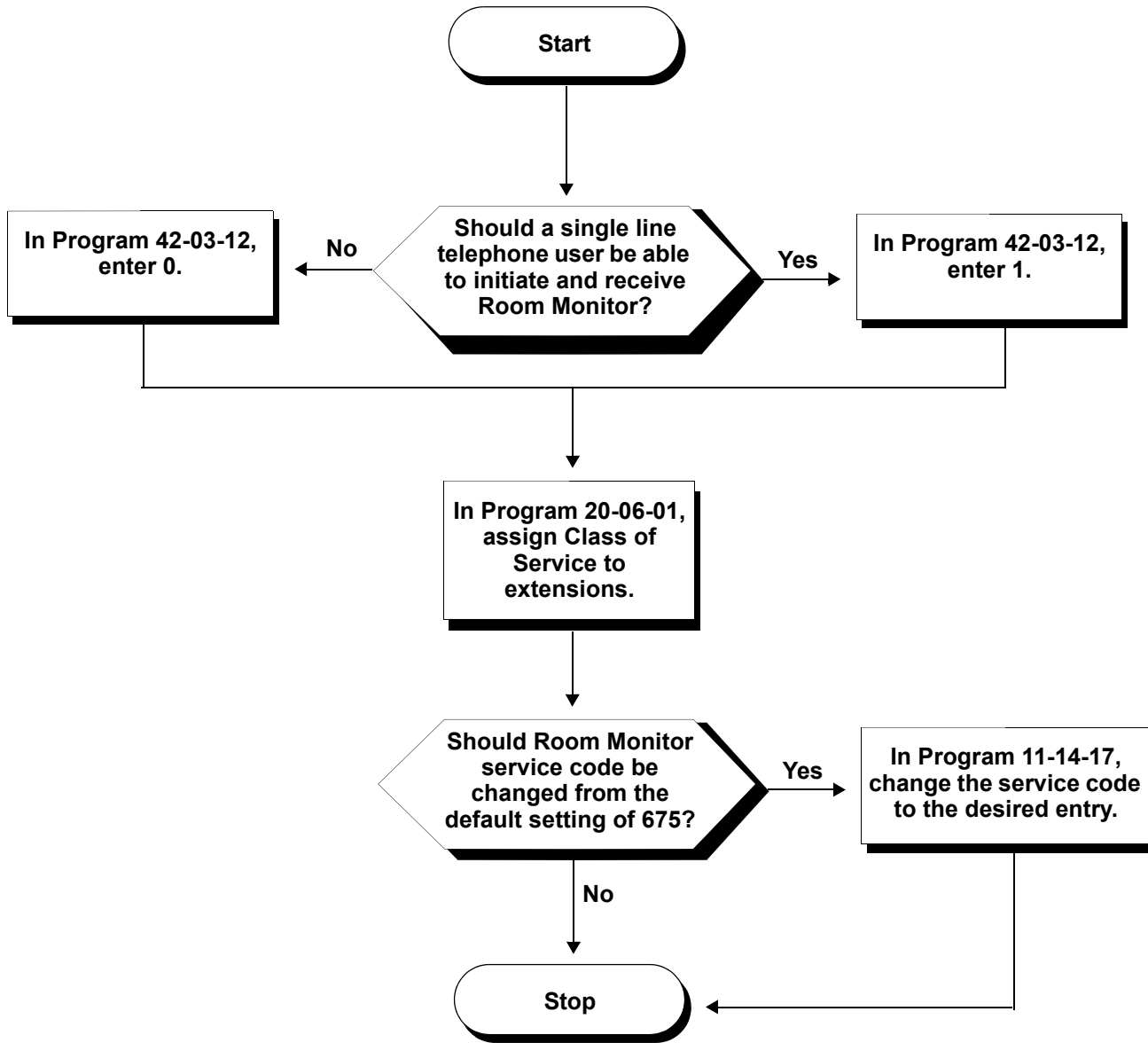
Programmable Function Keys

Programming

Multiline Room Monitoring




Single Line Telephone Room Monitoring



Programming

- ➔ **11-14-17 Service Code Setup (for Hotel) – Hotel Room Monitor**
Customize the service code (675 by default) to be used for Room Monitor.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key as a Room Monitor key (code 39) for both the extension being monitored and the extension initiating Room Monitor.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-11 Class of Service Options (Supplementary Service) – Room Monitor, Initiating Extension**
Enable (1)/Disable (0) an extension user ability to initiate Room Monitor.
(default: 0)
- ➔ **20-13-12 Class of Service Options (Supplementary Service) – Room Monitor, Extension Being Monitored**
Enable (1)/Disable (0) an extension user ability to be monitored.
(default: 0)
- ➔ **42-03-12 Class of Service Options (Hotel/Motel) – SLT Room Monitor**
Enable (1)/Disable (0) a single line telephone user ability to use Room Monitor.
(default: 0)

Operation

 *You must activate Room Monitor at the extension initiating the monitor and at the extension you want to monitor. You can only listen to one extension at a time.*

Multiline Terminals:


To activate Room Monitor from an idle Multiline Terminal (initiating extension):


1. Press the **Room Monitor** key (PRG 15-07 or SC 751: 39).
2. Dial the number of extension you want to monitor.

 *You can place and answer other calls while Room Monitor is active.*

To activate Room Monitor from an idle Multiline Terminal (extension to be monitored):

1. Press **Room Monitor** key (PRG 15-07 or SC 751: 39).
2. Dial the number of the extension where you are located.

 *For example, if you are at extension 106, dial 106.*


 *You can place and answer other calls while Room Monitor is active.*

To cancel Room Monitor (at either extension):

1. Press the **Room Monitor** key at both the initiating extension and the monitored extension.


Single Line Telephones:**To activate Room Monitor (at the extension to be monitored):**

1. Lift the handset at the telephone to be monitored.
2. Dial **675**.
3. Dial **1**.
4. Dial number of the extension number, which is monitoring the telephone.
5. Place the handset on the desk, placing the handset's transmitter towards the room.

 *You cannot place or answer other calls while Room Monitor is active.*

To activate Room Monitor (at the initiating extension):

1. Lift the handset at the telephone which is monitoring another telephone.
2. Dial **675**.
3. Dial **2**.
4. Dial number of extension number, which will be monitored.

 *You cannot place or answer other calls while Room Monitor is active.*

To cancel Room Monitor (at either extension):

1. Hang up the handsets for both the monitored and the monitoring telephones.

Save Number Dialed

Description

Save Number Dialed allows an extension user to save their last outside number dialed and easily redial it later on. For example, an extension user can recall a busy or unanswered number without manually dialing the digits. The system retains the saved number until the user stores a new one in its place or clears the stored one.

Save Number Dialed saves in system memory a dialed number up to 24 digits. The number can be any combination of digits 0~9, # and *. The system remembers the digits regardless of whether the call was answered, unanswered or busy. The system normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

Conditions

- For systems with Automatic Route Selection, ARS selects the trunk for the call unless the user preselects.
- Function keys simplify Save Number Dialed operation.

Default Setting

Enabled

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Automatic Route Selection

Central Office Calls, Placing

Dial Tone Detection

Last Number Redial

Programmable Function Keys


Repeat Redial

Programming

- ➔ **11-12-13 Service Code Setup (for Service Access) – Saved Number Dial**
Customize the service code (715 by default) to be used for dialing a saved number.
- ➔ **11-12-18 Service Code Setup (for Service Access) – Clear Saved Number Dialing Data**
Define the service code for Clear Save Number Dialing List (785 by default) if it is not acceptable.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key as a Save key (code 30).

Operation

To save the outside number you just dialed (up to 24 digits):

 *Use this feature before hanging up.*





Multiline Terminal

1. Press the **Save Number Dialed** key (PRG 15-07 or SC 751: 30).

Single Line Telephone

1. Hookflash.
2. Dial **715**.



To redial a saved number:Multiline Terminal

1. Press an idle trunk line key.
 -  *This selects a specific trunk for the call.*
2. Press the **Save Number Dialed** key (PRG 15-07 or SC 751: 30).
 -  *The stored number dials out.*
 - OR -**
1. Press **Speaker**.
2. Dial **715**.
 - OR -**
 - Press **Save Number Dialed** key (PRG 15-07 or SC 751: 30).
 -  *Save Number Dialed automatically selects a trunk from the same group as your original call.*
 -  *The stored number dials out.*

Single Line Telephone

1. Go off-hook.
2. Dial **715**.

To view the number you have saved from a multiline terminal with a display:

1. Press the **Save Number Dialed** key (PRG 15-07 or SC 751: 30).
 -  *The stored number displays for 10 seconds.*
 -  *The stored number dials out if you:*
 - Lift the handset,*
 - Press an idle line key,*
 - Press the Speaker key.*
2. Press the **Exit** key.

To clear your saved number:Multiline Terminal

1. Press **Speaker**.
2. Dial **785**.
3. Press **Speaker** to hang up.

Single Line Telephone

1. Lift the handset and dial **785**.
2. Hang up.

Secondary Incoming Extension

Description

Secondary Incoming Extensions (SIE) are incoming appearance keys of actual stations assigned in the system. SIE keys are assigned to programmable function keys and can appear on an individual station, or multiple stations. Incoming internal calls, ringing DIL/Tie/DID/CO Transfer calls, or call forwarded calls can be picked up from an SIE.

Conditions

- Calls cannot be originated from a Secondary Incoming Extension.
- Off-Hook ringing is provided with calls ringing into Secondary Incoming Extensions.
- Secondary Incoming Extensions are forwarded when the actual station is set for call forwarding.
- SIE keys can appear on an individual station, or multiple stations.
- A station can have more than one SIE key assigned.
- Up to 32 calls can be queued waiting on an SIE key.
- When a Secondary Incoming Extension call is received and answered while the user is on an outside line call, the first call can be automatically put on hold.
- If a trunk call rings a Secondary Incoming Extension, to answer the call, the station must be programmed with the direct trunk appearance key or an available CAP key and the SIE must be programmed to allow the call to come off the SIE key and appear on the line or CAP key.
- The same SIE key cannot be programmed on multiple programmable function keys on the same Multiline Terminal.
- An SIE key does not ring during an Intercom Voice call to the actual station.
- If multiple CAR/SIE/VE keys are ringing on a station at the same time, the CAR/SIE/VE key on the lowest Line Key is answered first.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Call Queuing

Call Appearance (CAP) Keys

Call Arrival (CAR) Keys

Virtual Extensions

Programming

- ➔ **11-04-01 Virtual Extension Numbering**
V-Port = 1~256
Virtual Ext No. = Extension Number
- ➔ **15-01-01 Basic Extension Data Setup – Extension Name**
Ext Name = Extension Name
- ➔ **15-02-07 Multiline Telephone Basic Data Setup – Automatic Hold for CO lines**
0 = Hold
1 = Disconnect
- ➔ **15-02-21 Multiline Telephone Basic Data Setup – Virtual Extension Access Mode (When idle Virtual Extension key pressed)**
V-Ext Key Mode:
0 = DSS (default)
1 = OTG (Outgoing – Not supported for SIE)
2 = Ignore

-
-
- **15-07-01 Programmable Function Keys**
KYxx = Key Number 01~32
KY01 = *03 (Virtual Extension, *01 by default)
KY01 = *08 (CAP Key)

 - **15-08-01 Incoming Virtual Extension Ring Tone Setup**
ICM Tone Pattern:
0 = Pattern 1 (default)
1 = Pattern 2
2 = Pattern 3
3 = Pattern 4

 - **15-09-01 Virtual Extension Ring Assignment**
Mode1.
0 = No (default)
1 = Yes

 - **15-10-01 Incoming Virtual Extension Ring Tone Order Setup**
Order 1 Patrn0 = Pattern 1 (default)
Order 2 Patrn1 = Pattern 2 (default)
Order 3 Patrn2 = Pattern 3 (default)
Order 4 Patrn3 = Pattern 4 (default)

 - **15-11-01 Virtual Extension Delayed Ring Assignment**
KY01 Mode1:
1 = Yes
0 = No
(default: 0)

 - **15-18-01 Virtual Extension Key Enhanced Options – Virtual Extension Key Operation Mode**
0 = Release
1 = Land On The Key
(default: 0)

 - **20-04-03 System Options for Virtual Extensions – CAR/SIE/Virtual Extension Delay Interval**
Delay = 10 sec (default)

 - **20-06-01 Class of Service for Extensions**
Mode1 Class No. = 1 (default)

 - **20-07-10 Class of Service Options (Administrator Level) – Programmable Function Key Programming (Appearance Level)**
1 = On (default)
0 = Off

 - **20-10-08 Class of Service Options (Answer Service) – Virtual Extension Off-Hook Answer**
0 = Off
1 = On (default)

 - **20-13-27 Class of Service Options (Supplementary Service) – Busy on Seizing Virtual Extension**
1 = On (default)
0 = Off

-
-
- ➔ **21-01-15 System Options for Outgoing Calls – Outgoing Disable on Incoming Line (Toll Restriction)**
0 = Off (default)
1 = On

 - ➔ **23-04-01 Ringing Line Preference for Virtual Extensions**
Order 1 Stg Group = 0~64 (0 by default)
Order 2 Stg Group = 0~64 (0 by default)
Order 3 Stg Group = 0~64 (0 by default)
Order 4 Stg Group = 0~64 (0 by default)

Operation

To answer a call ringing a SIE key:

1. Press the flashing **SIE** key

To program a SIE key on a phone:

1. Press **Speaker**.
2. Dial **752**.
3. Press the key you want to program.
4. Dial ***03**.
5. Dial the number of the extension you want to appear on the key.
6. Press **Hold** once for Immediate Ring, (skip to step 8 for Delayed Ring).
7. Dial the mode number in which the key rings.
8. Press hold a second time for Delayed Ring, or Skip to step 10.
9. Dial the mode number in which the key delays ringing.
10. Press **Speaker**.

Secretary Call (Buzzer)

Description

Secretary Call lets two co-workers alert each other without disturbing their work. To have Secretary Call, both co-workers must have multiline terminals with Secretary Call buzzer keys. When a user presses their buzzer key, the system alerts the called extension by sending a splash tone and flashing the called extension buzzer key. The called user can respond by placing an intercom call to the calling party.

The called extension buzzer key continues to flash and the splash tone is heard until either user cancels the Secretary Call. A secretary could use this feature, for example, to get a message through to the boss in an important meeting. After being alerted, the boss could call the secretary when it is most convenient.

An extension can have Secretary Call keys for any number of extensions, limited only by the available number of programmable keys.

Conditions

- Secretary Call is not available to single line telephone users.
- Secretary Call does not set up an Intercom call.
- When assigning Secretary Call, a user enters the associated extension numbers, not port numbers.
- Secretary Call requires a uniquely programmed function key.
- Secretary Call cannot be used with single line telephone.

Default Setting

Disabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features





Programmable Function Keys

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign function keys for Secretary Call buzzer (code 41 + the destination extension number). Both co-workers must have buzzer keys for each other.

Operation

To buzz your secretary or boss:

1. Do not lift the handset.
2. Press the buzzer key (PRG 15-07 or SC 751: 41 + secretary extension).
 -  *Your boss or secretary hears ringing.*
 -  *Your buzzer key lights steadily.*
 -  *Your boss's or secretary's buzzer key flashes fast.*
 -  *The telephone continues to ring until the Secretary Call key is pressed.*

To check to see who left you a Secretary Call:

1. Do not lift the handset.
2. Press the **Help** key.
3. Press the **Secretary Call** key that flashed.
4. Press the **Exit** key.

To answer your Secretary Call indication:

1. Place an Intercom call to the extension that called you.

To cancel a Secretary Call you left at another extension:

1. Press the lit **Secretary Call** key.

To cancel a Secretary Call left at your extension:

1. Do not lift the handset.
2. Press the flashing **Secretary Call** key.

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Secretary Call Pickup

Description

Secretary Call Pickup lets a multiline terminal user easily reroute calls intended for a co-worker to themselves. By pressing a Secretary Call Pickup key, the user can have all calls to a co-worker's telephone ring or voice-announce theirs instead. Secretary Call Pickup is a simplified type of Call Forward with Follow Me for employees that work closely together. This feature could be helpful to customer service representatives that must frequently cover each other's clients. When a representative leaves their desk, an associate could press the Secretary Call Pickup key to intercept all their calls.

An extension can have Secretary Call Pickup keys for any number of extensions, limited only by the available number of programmable keys.

Conditions

- Secretary Call Pickup is not available to single line telephone users.
- A Call Arrival (CAR) Key (virtual extension) cannot be programmed as the boss's extension.
- An extension user can also have Call Forwarding with Follow Me reroute a co-worker's calls to themselves.
- A multiline terminal can have a Secretary Call Pickup key for a single line telephone.

Default Setting

Disabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Call Forwarding with Follow Me

Programmable Function Keys

Secretary Call (Buzzer)



Single Line Telephones, Analog 500/2500 Sets

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign function keys for Secretary Call Pickup (42 + boss ext). Unlike Secretary Call, you do not have to program a corresponding key at the source and destination extensions.

Operation

To activate Secretary Call Pickup:

1. Press your **Secretary Call Pickup** key (PRG 15-07 or SC 751: 42 + boss extension).
 -  *Your Secretary Call Pickup key lights and the boss's telephone display shows "BOSS FWD>>".*
 -  *Calls intended for covered extension ring your telephone instead.*

To cancel Secretary Call Pickup:

1. Press your lit **Secretary Call Pickup** key (PRG 15-07 or SC 751: 42 + boss extension).

To check a key's Secretary Call Pickup assignment:

1. Press the **Help** key.
2. Press your **Secretary Call Arrival (CAR)** key (PRG 15-07 or SC 751: 42 + boss's extension).
3. Press the **Exit** key.

Selectable Display Messaging

Description

An extension user can select a programmed Selectable Display Message for their extension. Display multiline terminal callers see the selected message when they call the user's extension. Selectable Display Messaging provides personalized messaging. For example, an extension user could select the message GONE FOR THE DAY. Any display multiline terminal user calling the extension may hear a DND signal and then see the message. See table below for a list of the standard messages.

An extension user can add digits for date, time or telephone number after messages 1~8 and 10 (up to 24 characters). For example, an extension user could select the message ON VACATION UNTIL and then enter the date. Callers see the original message followed by the appended date. They would then know when the user is coming back from vacation. The system allows all telephones to use the Selectable Display Messaging feature at the same time.

All telephones can use Selectable Display Messaging at the same time.

The default messages are:

Table 2-31 Selectable Display Messaging Defaults

No.	Message	Change “#” to...
1	IN MEETING UNTIL ##:##	Time (when meeting done)
2	MEETING ROOM - #####	Room Name or extension
3	COME BACK ##:##	Time (when returning)
4	PLEASE CALL #####	11 digits (telephone number)
5	BUSY CALL AFTER ##:##	Time (when returning)
6	OUT FOR LUNCH BACK ##:##	Time (when returning)
7	BUSINESS TRIP BACK ##/##	Date (when returning)
8	BUSINESS TRIP #####	10 digits (where reached)
9	GONE FOR THE DAY	
10	ON VACATION UNTIL ##/##	Date (when returning)
11~20	MESSAGE 11~20	

Conditions

- The # cannot be used in a Message.
- When Selectable Display Messaging is set as DND All, all other DND modes are canceled when Selectable Display Messaging is canceled.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals with Display

Required Component(s)

None

Related Features

Do Not Disturb

Programmable Function Keys

Programming

- ➔ **11-11-14 Service Code Setup (for Setup/Entry Operation) – Text Message Setting**
Define the service code to be used when setting a text message.
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Text Message (code 18). The Text Message key automatically selects the message used when programming the key.

- ➔ **20-01-02 System Options – Text Message Mode**
Select whether and intercom caller should hear busy (1) or ring through (0) for extensions which have Selectable Display Messaging set.
Any extension previously set with Selectable Display Messaging must cancel the feature and reactivate for a change in this option to take affect.
- ➔ **20-02-07 System Options for Multiline Telephones – Time and Date Display Mode**
Set the System Time and Date display mode. The time that displays in Selectable Display Messages follows this setting.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-19 Class of Service Options (Supplementary Service) – Selectable Display Messaging (Text Messaging)**
Enable (1)/Disable (0) an extension user ability to use Selectable Display Messaging.
(default: 1)
- ➔ **20-16-01 Selectable Display Messages**
Program the Selectable Display Messages (1~20). Refer to the chart below for character entry.

Table 2-32 Selectable Display Message – Character Entry Chart




Use this keypad digit . . .	When you want to . . .
1	Enter characters: 1 @ [¥] ^ _ ‘ { } > <
2	Enter characters: A-C, a-c, 2.
3	Enter characters: D-F, a-f, 3.
4	Enter characters: G-I, g-i, 4.
5	Enter characters: J-L, j-l, 5.
6	Enter characters: M-O, m-o, 6.
7	Enter characters: P-S, p-s, 7.
8	Enter characters: T-V, t-v, 8.
9	Enter characters: W-Z, w-z, 9.
0	Enter characters: 0 ! “ # \$ % & <space> ()
*	Enter characters: * + , - . / : ; < = > ?
#	Accepts a numeric entry from the user when setting a display message. e.g., time or date. Back at ##:##
Conf	Clear the character entry one character at a time.

Table 2-32 Selectable Display Message – Character Entry Chart (Continued)

Use this keypad digit . . .	When you want to . . .
Hold	Clear all the entries from the point of the flashing cursor and to the right.

Operation


To select a message:

1. Press **Speaker** + press the **Text Message** key (PRG 15-07 or SC 751: 18) + enter digits to append (if needed) + **Speaker** to hang up. Skip the remaining steps.
2. (Optional for messages 1~8 and 10.)
Dial the digits you want to append to the message.
 -  *You can append messages 1~8 and 10 with digits (e.g., the time when you will be back). Enter the time in 24-hour format.*
3. Press **Speaker** to hang up.
 -  *Intercom calls to extensions with Selectable Display Messaging set receive a DND signal and receive the display message on their telephone's display instead of ringing the extension based on the setting in Program 20-01-02.*
 -  *To allow calls to ring through and have the message displayed on the calling extension display, cancel DND by pressing DND + 0.*

To cancel a message:

1. Press **Speaker** and the **Text Message** key (PRG 15-07 or SC 751: 18).
2. Press **Speaker** to hang up.

Using the Text Message Service Code to select a message:

1. Press **Speaker** and dial the Text Message service code (Program 11-11-14).
2. Dial the Selectable Display Message number to be used (**01~20**).
(Optional messages 1~8, and 10, dial the digits you want to append to the message.)
3. Press **Speaker** to hang up.
 -  *To cancel, repeat Step 1 and hang up.*

Selectable Ring Tones

Description

An extension user can change the way trunks or internal calls ring their telephone. Selectable Ring Tones allow an extension user to set up unique ringing for their calls. This is important in a crowded work area where several telephones are close together. Because their telephone has a characteristic ring, the user always can tell when their telephone is ringing.

Conditions

- Only ring tones 1-High, 2-Medium, 3-Low are available Electra Elite and *D^{term}* Series E terminals (DTU/DTP). If the other tones are selected, the terminal rings with a 2-Medium tone.

Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Distinctive Ringing, Tones and Flash Patterns

Programming

- ➔ **11-11-20 Service Code Setup (for Setup/Entry Operation) – Change Incoming CO and ICM Ring Tones**
If required, change the service code used for changing the incoming ring tones heard for CO and ICM calls (Default: 720).
- ➔ **11-11-21 Service Code Setup (for Setup/Entry Operation) – Check Incoming Ring Tones**
If required, change the service code used for checking how the incoming ring tones sound. (Default: 711).
- ➔ **15-02-02 MultiLine Telephone Basic Data Setup – Trunk Ring Tone**
Set the tone (pitch) of the incoming trunk ring for the extension port you are programming. (default: 2)

1 = High	5 = Ring Tone 2
2 = Mid Range	6 = Ring Tone 3
3 = Low	7 = Ring Tone 4
4 = Ring Tone 1	8 = Ring Tone 5
- ➔ **15-02-03 Multiline Telephone Basic Data Setup – Extension Ring Tone**
Set the tone (pitch) of the incoming extension call ring for the extension port you are programming. Also refer to program 15-08. (default: 8)

1 = High	5 = Ring Tone 2
2 = Mid Range	6 = Ring Tone 3
3 = Low	7 = Ring Tone 4
4 = Ring Tone 1	8 = Ring Tone 5
- ➔ **15-08-01 Incoming Virtual Extension Ring Tone Setup**
Assign a ring tone pattern (0~4) to incoming virtual extensions assigned to a Virtual Extension key (Program 15-07). If you enable ringing for the key in Program 15-09, the key rings with the tone you set in this program. Also see Program 22-03. (default: 0)
- ➔ **15-10-01 Incoming Virtual Extension Ring Tone Order Setup**
Set the priority (1~4) for the Virtual Extension Ring Tones set in Program 15-08. When Virtual Extension calls ring an extension simultaneously, the tone with the highest order number (e.g., 1) rings. The other keys just flash. (default: 1~4 = 0~3)

**22-03-01 Trunk Ring Tone Range**

Select the ring tone range for the trunk. The trunk uses a ring tone in the range selected when it rings an extension. Four ring tones are available.
(default: 0)

Operation

To change your extension incoming ring tones:

1. Press **Speaker**.
2. Dial **720**.
3. Dial **1** to set Intercom ring; **2** to set trunk ring.
4. Dial code for the desired ring pattern (**1~8**).
5. Press **Speaker** to hang up.

To listen to the incoming ring choices:

1. Press idle **Speaker**.
2. Dial **711**.
3. Dial **1** to listen to Intercom ring; **2** to listen to trunk ring.
4. *For Intercom Ring:*
Dial the code for the ring pattern you want to hear (**1~8**).

- OR -

For Trunk Ring:

Dial code for the ring pattern you want to hear (Ring 1~3, Melody 4~8). If you select Ring 1~3, a second screen prompts for the tone pattern (1~4).

5. Press **Speaker** to hang up.

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Serial Call

Description

Serial Call is a method of transferring a call so it automatically returns to the transferring extension. Serial Calling saves transferring steps between users. For example, a Customer Service Representative (CSR) has a client on the telephone who needs technical advice. The CSR wants to send the call to Technical Service, but needs to advise the client of certain costs when Technical Service is done. Rather than transferring the call back and forth, the CSR can use Serial Call to Technical Service and announce, "I have Ted on the telephone. I need to talk to him again. Just hang up when you're done and I'll get him back."

Conditions

- The transferring extension can remain off-hook to auto-receive the callback or hang up and it rings back to them.
- Serial Call requires a uniquely programmed function key.
- Serial Call is not available to single line telephones.

Default Setting

Disabled

System Availability

Terminals

Multiline Terminals

Required Component(s)

None

Related Features

Programmable Function Keys

Transfer

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign a programmable key as a Serial Call key (code 43).

Operation

To place a Serial Call to a co-worker:

1. Place or answer a call.
2. Press **Hold**.
3. Dial co-worker's extension number.
 - 📎 *Co-worker must lift the handset to respond to your announcement.*
4. Press the Serial Call key (PRG 15-07 or SC 751: 43) but do not hang up.
 - 📎 *When your co-worker hangs up the call, the system makes an automatic live transfer back to your extension.*

Single Line Telephones, Analog 500/2500 Sets

Description

The system is compatible with 500 type (Dial Pulse) and 2500 type (DTMF) analog single line telephones (SLTs). You can install single line telephones as On-Premise or Off-Premise extensions. Single line telephone users can dial codes to access many of the features available to multiline terminal users. With single line telephones, you can have your system simulate PBX operation.

There are 176 single line telephones available (note that this number may be restricted due to system power requirements).

When installing single line telephones you must have:

- A port on an SLIU ETU for each single line telephone installed.
- If you have 2500 sets, at least one block reserved on the CPUII for analog extension DTMF reception.

DTMF Dial Out Timer Added

A program is added for DTMF dialing, Program 20-03-07 : System Options for Single Line Telephones - Trunk Call Dial Forced Sending Start Time (Forced Dial). When Program 20-03-03 : System Options for Single Line Telephones – SLT DTMF Dial to Trunk Lines is set to 0 (receive all digits before sending), the system follows the timer in Program 20-03-04 and Program 20-03-07.

The timer in Program 20-03-04 System Options for Single Line Telephones – Dial Sending Start Time for SLT or ARS resets when the user dials another digit.

The timer in Program 20-03-07 System Options for Single Line Telephones – Trunk Call Dial Forced Sending Start Time (Forced Dial) does not reset when a digit is dialed. The user must finish dialing all the digits before this time expires (entries: 0~64800 seconds, default: 0).

Conditions

- When an analog device is a member of a Department Group assigned as the VM Department Group in PRG 45-01-01, it cannot perform an unsupervised transfer. When the party where the call is being transferred answers, the transferred party is connected to that party, and the analog device is released.
- Dial Pulse (500 type) single line telephones cannot access any features that require the user to dial # or *.
- A single line telephone can initiate an Internal Zone page, but cannot receive an Internal Zone Page.

- With **Version 2000 or higher**, when a Ring Group call rings a single line station, the BLF indication shows busy.
- Stutter Dial Tone is supported to Single Line Telephones for Voice Mail Message Waiting.

Default Setting

Single line telephones function as soon as they are installed and properly programmed.

System Availability

Terminals

DTR-1-1

DTH-1HM-1

DTH-1-1

DTP-1-2

DTP-1HM-2

DTR-1R-2

Required Component(s)

SLI()-U() ETU

Related Features

Single line telephone users have access to the following features:

- Speed Dialing
- Automatic Route Selection
- Call Forwarding with Follow Me
- Callback
- Conference
- Directed Call Pickup
- Flash
- Hold
- Last Number Redial
- Meet Me Paging
- Night Service
- PBX Compatibility
- Save Number Dialed
- Transfer
- Voice Over
- Account Codes
- Barge-In
- Call Forwarding/DND Override
- Central Office Calls, Answering
- Department Calling
- Do Not Disturb
- Forced Trunk Disconnect
- Intercom
- Line Preference
- Meet Me Paging Transfer
- Off-Hook Signaling
- Pulse to Tone Conversion
- Selectable Display Messages
- Trunk Queuing and Camp-On
- Warning Tone for Long Conversation
- Alarm
- Call Forwarding
- Call Waiting/Camp-On
- Central Office Calls, Placing
- Department Step Calling
- Door Box
- Group Call Pickup
- Handsfree Answerback/Forced Intercom Ringing
- Meet Me Conference
- Message Waiting
- Paging
- Ringdown Extension
- Toll Restriction
- Voice Mail

Data Communications

APA and APR modules can be used with multiline terminals to provide an analog port.

Refer to the individual features for additional descriptive, programming and operational information.

Programming

- ➔ **10-03-01 ETU Setup – Terminal Type (B1) for ESI PKG**
Program all on-premise 500/2500 type single line telephones with circuit type 2. Set the DIOPU trunk to type 1 when trunks should be defined for off-premise extension (OPX) use.
- ➔ **10-03-03 ETU Setup – For SLI PKG – Transmit Gain Level (S-Level)**
10-03-04 ETU Setup – For SLI PKG – Receive Gain Level (R-Level)
Assign transmit and receive levels for 500/2500 type single line telephones.
- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**
If the system has 2500 type (DTMF) single line extensions, allocate at least one circuit for analog extension DTMF reception (entry 0 or 1).
Use the following as a guide when allocating DTMF receivers:
 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.

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- ➔ **15-03-01 Single Line Telephone Basic Data Setup – SLT Signaling Type**
Enter 0 if single line telephone is a 500 type (dial pulse). Enter 1 if single line telephone is a 2500 type (DTMF).
(default: 1)
 - ➔ **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**
Enter 0 for normal 500/2500 type telephones. Enter 1 if a Voice Mail port (refer to Voice Mail for more information).
(default: 0)
 - ➔ **20-03-02 System Options for Single Line Telephones – Ignore Received DP Dial on DTMF SLT Port**
Define whether the system should receive dial pulse and DTMF signals (0) or ignore dial pulse and only accept DTMF signals (1).
(default: 0)
 - ➔ **20-03-03 System Options for Single Line Telephones – SLT DTMF Dial to Trunk Lines**
Set the SLT phones to (0). Collect all digits before sending or (1), send out immediately after receiving. When using a third-party external paging device, set this option to 1. In addition, set Program 20-03-04 to 1.
(default: 0)
 - ➔ **20-03-04 System Options for Single Line Telephones – Dial Sending Start Time for SLT or ARS**
Set the time (0~64800 seconds) before the first digit is sent out. When using a third-party external paging device, set this option to 1. In addition, set Program 20-03-03 to 1.
(default: 3)
 - ➔ **20-06-01 Class of Service for Extensions**
Assign a unique Class of Service for Dual OPX telephones only when using Continued Dialing.
 - ➔ **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**
Set option to 0 to prevent dual DTMF tones when using Continued Dialing.
 - ➔ **20-15-01 Ring Cycle Setup – Normal Incoming Call on Trunk**
Define the ringing cycle (1~13) for normal incoming trunk calls (DIL, ring group, etc.).
(default: 8)
 - ➔ **20-15-03 Ring Cycle Setup – Incoming Internal Call**
Define the ringing cycle (1~13) for ICM calls.
(default: 12)
 - ➔ **20-15-05 Ring Cycle Setup – DID/DDI**
Define the ringing cycle (1~13) for DID calls.
(default: 8)
 - ➔ **80-03-01 through 80-03-08 DTMF Tone Receiver Setup**
 - ➔ **80-04-01 Call Progress Tone Detector Setup – Detect Level**
If required, modify the criteria for dial tone detection and call progress tone detection for the DTMF tones received at a single line telephone.
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- **82-11-01 SLI Initial Setup – Bounce Protect Time**
- **82-11-02 SLI Initial Setup – Hookflash Start Time**
- **82-11-03 SLI Initial Setup – Hookflash End Time**
Define various timers for the SLI ETUs. The entries you make in this program affect all SLI ETUs installed.

Operation

Refer to the individual features listed in the Related Features section above in this feature.

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SLT Adapter

Description

The SLT (Single Line Telephone) Adapter allows a port of an ESIB(8)-U() ETU or ESIE(8)-U() ETU to support a single line telephone. A single line telephone can be connected to the ESIB(8)-U() ETU using the SLT Adapter and 2-wire cable. Eight SLTII(1)-U() ADP Single Line Telephone Adapters can be installed in the Electra Elite IPK II system.

Conditions

- A maximum of eight SLTII(1)-U() ADPs can be used in the Electra Elite IPK II system.
- Dial Pulse and Dual-Tone Multifrequency Single Line Telephones are supported.
- The SLTII(1)-U() ADP does not support voice mail.
- Message Waiting LED is not supported.
- A single line telephone connected to an SLTII(1)-U() ADP cannot perform a Trunk-to-Trunk Transfer or support a 1-terminal to 2-outside party conference call.

Default Setting

None

System Availability

Terminals

Single Line Telephones

Required Component(s)

SLTII(1)-U() ADP

Related Features

Ancillary Device Connection

Cordless Telephone Connection

Single Line Telephones, Analog 500/2500 Sets

Programming

- ➔ 10-03-01 **ETU Setup – Terminal Type (B1) for ESI PKG**
- ➔ 11-02 **Extension Numbering**
- ➔ 20-07 **Class of Service Options (Administrator Level)**
Refer to [Table 2-6 Class of Service Options \(Administrator Level\)](#), Program 20-07.
- ➔ 20-08 **Class of Service Options (Outgoing Call Service)**
Refer to [Table 2-7 Class of Service Options \(Outgoing Call Service\)](#), Program 20-08.
- ➔ 20-09 **Class of Service Options (Incoming Call Service)**
Refer to [Table 2-8 Class of Service Options \(Incoming Call Service\)](#), Program 20-09.
- ➔ 20-10 **Class of Service Options (Answer Service)**
Refer to [Table 2-9 Class of Service Options \(Answer Service\)](#), Program 20-10.
- ➔ 20-11 **Class of Service Options (Hold/Transfer Service)**
Refer to [Table 2-10 Class of Service Options \(Hold/Transfer Service\)](#), Program 20-11.
- ➔ 20-12 **Class of Service Options (Charging Cost Service)**
Refer to [Table 2-11 Class of Service Options \(Charging Cost Service\)](#), Program 20-12.
- ➔ 20-13 **Class of Service Options (Supplementary Service)**
Refer to [Table 2-12 Class of Service Options \(Supplementary Service\)](#), Program 20-13.
- ➔ 20-14 **Class of Service Options for DISA/E&M**
Refer to [Table 2-13 Class of Service Options for DISA/E&M Service](#), Program 20-14.

Operation

To originate internal calls:

1. Lift the handset, and wait for internal dial tone.
2. Dial the applicable internal number.
3. Talk when called party answers.

To originate outside calls:

1. Lift the handset, and wait for dial tone.
2. Dial Trunk Access Code (default: 9).
3. Dial the number of outside party.
4. Talk when the called party answers.

To answer outside or internal calls:

Lift the handset, and talk.

To transfer an outside call or internal call with a call in progress:

1. Press the hookswitch momentarily, and wait for second dial tone.
2. Dial the station number where call is to be transferred.
3. Hang up.

To perform a Trunk-to-Trunk Transfer with an outside call in progress:

1. Provide hookflash. The call is placed on Exclusive Hold. Receive internal dial tone.
2. Dial the Trunk Access Code for the applicable trunk.
3. Dial the applicable number.
4. Hang up.

To initiate a conference with a call in progress:

1. Provide hookflash and dial #1.
2. Dial the applicable number and wait for the party to answer.
3. Provide hookflash and repeat the second step to add parties to the conference.

- OR -

1. Provide hookflash twice to set up the conference.

To access the feature:

1. Lift the handset, and wait for internal dial tone.
2. Dial the applicable Feature Access Code.

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SNMP Simple Network Management Protocol

Enhancements

Version 1600 and higher is required.

Description

Simple Network Management Protocol (SNMP) is an application-layer protocol designed to facilitate the exchange of management information between devices which are on the network. By using SNMP-transported data (such as packets per second and network error rates), network administrators can easily manage network performance, find and solve network problems, and plan for network growth.

SNMP is part of the Internet network management architecture. This architecture is based on the interaction of many entities, as described in the following Elite IPK II SNMP Installation Manual.

The SNMP Agent is located on the Elite IPK II CPU II. Since the Elite IPK II CPU II controls most of the ETUs, this provides a great advantage of capturing most of the ETU status, failure and alarm details.

The major components of SNMP Implementations on the Elite IPK II system support MIB-II as well as private MIB to accommodate all necessary KTS related data types.

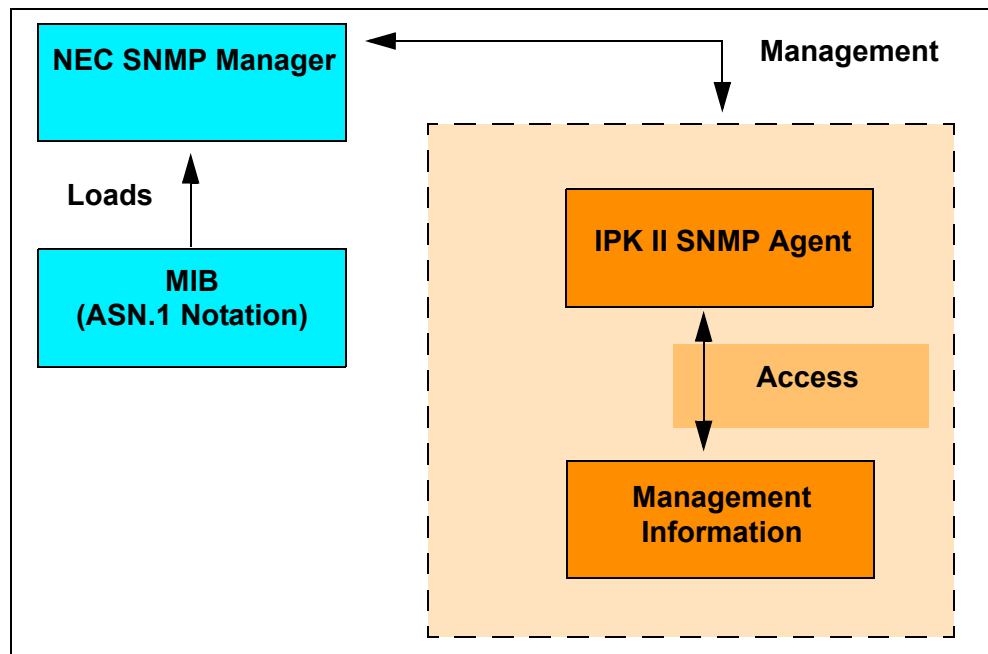


Figure 2-23 SNMP Managed System

Conditions

- The SNMP software module (Agent) located on the Elite IPK II CPUII supports remote monitoring and debugging of Elite IPK II system by using the SNMP command set.
- SNMP Agent uses port 161 for communicating with NMS manager for Get/Set Getnext and walk operations.
- SNMP Agent uses port 162 for sending the traps and alarms. These ports can be configured through MB IDs.
- If multiple ETUs of the same type exist in a system, by default the SNMP Agent provides interface for the first available ETU. To get the details of other ETUs, the user must set other ETU locations in the system by setting cabinet number and slot number definitions.
- A User only needs to set the NMS Manager's IP address to receive the traps and alarms TrapDestIP for one ETU. This setting is applied for every card in the Elite IPK II System.
- A system reset is required to enable the SNMP feature.
- SNMP V1, V2 and V3 with MD5 authentication are supported.
- The MIB tree interfaces with Elite IPK cards like Main CPUII card, PRI card, BRI card, DTI card, COI card, PVA, SLI card and Intra Mail cards.
- Support for MIB

It provides a complete MIB-II (RFC 1213) support. The package also supports the MIBs for monitoring the system and measuring the system performance and extended private MIB.

- Monolithic Agent – The default behavior of the SNMP Agent is as a monolithic agent.

MIBs

The Elite IPK II SNMP Agent implemented a set of Managed Objects described in MIB documents written in ASN.1 data description language.

The Elite IPK II MIBs to support the Elite IPK II SNMP Agent may be downloaded for free from the cng web site: www.cng.nec.com.

Default Setting

None

System Availability

Terminals

None

Required Component(s)

IPK II CPUII()-U10 ETU Release **2.00 or higher**.

SNMP Manager which supports **SNMP V1, V2 and V3 with MD5 authentication**.

Related Features

None

Programming

- ➔ **10-43-01 SNMP Setting – SNMP Agent (Version 1600 or higher required)**
Enable (1)/Disable (0) the SNMP Agent located on the Elite IPK II CPUII.
(default: 0)
- ➔ **10-43-02 SNMP Setting – SNMP Port Number (Version 1600 or higher required)**
Define the SNMP Agent Listening port.
(default: 161)
- ➔ **10-43-03 SNMP Setting – Trap Destination IP Address (Version 1600 or higher required)**
Define the IP Address of the Network Management Station (NMS). Sometimes called *consoles*, these devices execute management applications that monitor and control network.
(default: None)
- ➔ **10-43-04 SNMP Setting – Trap Destination Port Number (Version 1600 or higher required)**
Define Network Management Station (NMS) destination port number for sending traps and alarms.
(default: 162)
- ➔ **10-43-05 SNMP Setting – Read Only Community Name (Version 1600 or higher required)**
Define the Read Only Community Name.
(default: necpublic)

- ➔ **10-43-06 SNMP Setting – Read Write Community Name (Version 1600 or higher required)**
Define the Read and Write Community Name.
(default: necprivate)
- ➔ **10-43-07 SNMP Setting – SNMPv3 Authentication Method (Version 1600 or higher required)**
Define the method of authentication using MD5 or SHA(Not Supported Release).
(default: MD5)
- ➔ **10-43-08 SNMP Setting – SNMPv3 User ID (Version 1600 or higher required)**
Define the Authentication User ID.
(default: IPK II)
- ➔ **10-43-09 SNMP Setting – SNMPv3 Password (Version 1600 or higher required)**
Define the Authentication Password.
(default: password)

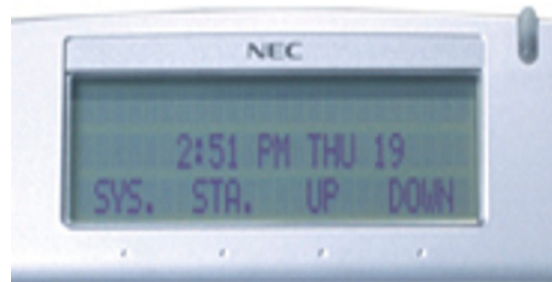
Operation

None

Softkeys

Description

Each display telephone provides interactive softkeys for intuitive feature access. It is no longer necessary to remember feature codes to access the telephone advanced features because the function of the softkeys change as the user processes calls. For example, just press a softkey to Page, Park a call, leave a message or Camp-On to a busy co-worker.



Additional options allow you to “fine tune” the multiline terminal volume levels for handset receive and transmit, speaker volume, ringer and handset volume, and headset volume levels. You can also customize the point at which the built-in speakerphone switches from transmit to receive; a boon for noisy environments. The display telephones also have a contrast control for the LCD display.

Conditions

- If a feature is restricted by an extension Class of Service, though the Softkey menu still displays the option, the user cannot set the feature.
- Using the Directory Dialing Softkeys, the Recall key can toggle the language display from English to Japanese.
- The feature must be active to change the volume (e.g., telephone must be ringing, or page must be heard.). Press either volume key when the telephone is idle to adjust the display contrast.

Default Setting

Display shows time/date/extension/Softkey menu information.

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Directory Dialing

Volume Controls

Programming

None

Speed Dial – System/Group/Station

Description

Speed Dialing gives an extension user quick access to frequently called numbers. This saves time, for example, when calling a client with whom they deal often. Instead of dialing a long telephone number, the extension user just dials the Speed Dialing code.

There are three types of Speed Dialing: System, Group and Station. All co-workers can share the System Speed Dialing numbers. All co-workers in the same Speed Dialing Group can share the Group Speed Dialing numbers. Station Speed Dialing numbers are available only at a user's own extension. The system has 2000 Speed Dialing bins that you can allocate between System and Group Speed Dialing and a maximum of 65 Speed Dialing Groups are available. Each extension has 10 Station Speed Dial bins.

Each Speed Dialing bin can store a number with up to 24 digits.

When placing an Speed Dialing call, the system normally routes the call through Trunk Group Routing or ARS (whichever is enabled). Or, the user can preselect a specific trunk for the call. In addition, the system can optionally force System Speed Dialing numbers to route over a specific Trunk Group. User preselection always overrides the system routing.

System Bins Limited to 1000 with Speaker Key or #2 Service Code

Though 2000 Speed Dialing bins are available in the system, once programmed, these bins can currently be dialed only using the Directory Dial feature (Press Directory key + SYS softkey + use arrow keys to locate number or enter the Speed Dial bin name + Speaker to place call.)

Speaker and service code #2 operations are not available for any 4-digit Speed Dial System bin number.

DSS Console Chaining

DSS Console chaining allows an extension user with a DSS Console to chain to an Speed Dialing number stored under a DSS Console key. The stored number dials out (chains) to the initial call. This can, for example, simplify dialing when calling a company with an Automated Attendant. You can program the bin for the company number under one DSS Console key (e.g., #200) and the client's extension number under the other (e.g., #201). The DSS Console user can press the first key to call the company, wait for the Automated Attendant to answer, then press the second key to call the client (extension 400). See the Programming section below for additional details.

The DSS Console user can also chain to an Speed Dialing number dialed manually, from a Programmable Function Key or a One-Touch Key.

Storing a Flash

To enhance compatibility with connected Centrex and PBX lines, Speed Dialing bin can have a stored Flash command. For example, storing 9 Flash 926 5400 causes the system to dial 9, flash the line and then dial 926 5400. The Flash can be stored by the user from their telephone or by the system administrator during system programming.

Using a Programmable Function Key

To streamline frequently-called numbers, a Speed Dialing Programmable Function Key can also store a Speed Dialing bin number. When the extension user presses the key, the telephone automatically dials the stored number. This provides true one-touch calling via a telephone function key.

Conditions

- Speed Dial bins can contain stored Account Codes. To prevent them from being displayed use PRG 20-07-04.
- ARS selects the trunk for the call unless the user preselects.
- A user can implement Speed Dial only if their extension has outgoing access to trunks.
- An extension can have a One-Touch Key for Speed Dial operation.
- If you enter a PBX trunk access code in a Speed Dial bin, the system automatically inserts a pause after the bin.
- Single line telephones can only dial System and Group Speed Dial numbers.
- Toll Restriction may prevent a user from using a stored Speed Dial number.
- Unless a user preselects a trunk, Trunk Group Routing selects the trunk Speed Dial uses for trunk calls.
- If the Speed Dial bin does not have a name assigned it does not show when scrolling through the directory of speed dials.
- If Program 13-01-01 is set to 1 (Intercom Access mode), system speed dial bins require insertion of a trunk access code.

Default Settings

Available (No Speed Dialing bins are assigned).

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Account Code Entry

Automatic Route Selection

Central Office Calls, Placing

Code Restriction

Dial Tone Detection

One-Touch Calling

PBX Compatibility

Programmable Function Keys

Single Line Telephones, Analog 500/2500 Sets

Trunk Group Routing

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**
If dial tone detection is enabled, be sure to allocate at least one circuit for dial tone detection (Type 0 or 2).
- ➔ **11-12-10 Service Code Setup (for Service Access) – Station Speed Dialing**
Assign Service code for accessing System Speed Dial bins (default #2).

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- **13-01-01 Speed Dialing Option Setup – Speed Dialing Auto Outgoing Call Mode**
Designate trunk or intercom outgoing mode.
(default: 0, trunk)
 - **13-01-03 Speed Dialing Option Setup – Number of Common Speed Dialing Bins**
Designate the bins the system uses for System Speed Dialing.
(default: 1000)
 - **13-02-01 Group Speed Dialing Bins**
Designate the bins the system uses for Group Speed Dialing.
(default: No Setting)
 - **13-03-01 Speed Dialing Group Assignment for Extensions**
For Group Speed Dialing, assign extensions to Speed Dialing groups (1~64).
(default: 1)
 - **13-04-01 Speed Dialing Number and Name – Speed Dialing Data**
Enter the System and Group Speed Dialing numbers and names.
(default: No Setting)
 - **13-04-02 Speed Dialing Number and Name – Name**
Assign a name to each System Speed Dial bin.
(default: No Setting)
 - **13-04-03 Speed Dialing Number and Name – Transfer Mode**
Assign the transfer mode for each System Speed Dial bin.
(default: 0)
 - **13-04-04 Speed Dialing Number and Name – Transfer Destination Number**
(default: No Setting)
 - **13-04-05 Speed Dialing Number and Name – Incoming Ring Pattern**
(default: 0)
 - **13-05-01 Speed Dialing Trunk Group – Trunk Group Number**
For each System Speed Dialing number, enter the routing option Trunk Group Number (1~100)
to dial out on.
(default: No Setting)
 - **14-02-06 Analog Trunk Data Setup – Pause at 1st digit after Line Seize in Manual Dial Mode**
Enable (1)/Disable (0) the system ability to pause after dialing the first digit.
(default: 1)
 - **15-02-04 Multiline Telephone Basic Data setup – Redial (Speed Dial) Control**
Assign the extension Redial key for either Common (0) or Group (1) Speed Dialing.
(default: 0)

-
-
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for System Speed Dialing (27) or Group Speed Dialing (28). You can program the key as either a general Speed Dialing key or you can choose to store a bin number with the function key. This key would then always dial the associated bin number. If storing a bin number along with the code, do not store 0, 00 or 000. To bypass entering a bin number, press **Hold** (**Hold** is also required if programming the function key using the service code 751).

 - ➔ **15-14-01 Programmable One-Touch Keys – Station Speed Dial Data**
Assign the extension (up to 24 digits) Speed Dial number (1~10).
(default: No Setting)

 - ➔ **15-14-02 Programmable One-Touch Keys – Station Speed Dial Name**
Assign the name (up to 12 digits) associated with the extension Speed Dial Bin (1~10).
(default: No Setting)

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

 - ➔ **20-07-04 Class of Service Options (Administrator Level) – Storing Speed Dialing Entries**
Allow (1)/Prevent (0) the storing of Speed Dialing entries (codes 753 and 754). Preventing this option also prevents the user's display from showing the telephone number. This can be used when you want to prevent account codes from being displayed.
(default: 1)

 - ➔ **20-08-03 COS Options (Outgoing Call Service) – System Speed Dialing**
Enable (1)/Disable (0) System Speed Dialing.
(default: 1)

 - ➔ **20-08-04 COS Options (Outgoing Call Service) – Group Speed Dialing**
Enable (1)/Disable (0) Group Speed Dialing.
(default: 1)

 - ➔ **30-03-01 DSS Console Key Assignment**
For DSS Console Chaining, assign an Speed Dialing Service Code (or) plus a 2-digit bin number to a DSS Console key.

 - ➔ **80-03-01 through 80-03-08 DTMF Tone Receiver Setup**

 - ➔ **80-04-01 through 80-04-09 Call Progress Tone Detector Setup**
If required, modify the criteria for dial tone detection and call progress tone detection for outgoing Speed Dialing calls.

Operation

To store an Speed Dialing number (display telephones only):



1. Press **Speaker**.
2. Dial **753** (for system) or **754** (for group).
3. Dial system or group storage code.
 -  *Initially, there are 1000 System Speed Dialing codes. There are Group Speed Dialing codes only if you define them in programming.*
4. Dial telephone number you want to store (up to 24 digits).
 -  *Valid entries are 0~9, # and *. To enter a pause, press **Transfer**. To store a Flash, press **Recall**.*
5. Press **Hold**.
6. Enter the name associated with the Speed Dialing number.

Table 2-33 Keys for Entering Names



Use this keypad digit . . .	When you want to . . .
1	Enter characters: 1 @ [¥] ^ _ ` { } Æ " Á À Â Ã Ç É Ê Ì Ó
2	Enter characters: A-C, a-c, 2.
3	Enter characters: D-F, d-f, 3.
4	Enter characters: G-I, g-i, 4.
5	Enter characters: J-L, j-l, 5.
6	Enter characters: M-O, m-o, 6.
7	Enter characters: P-S, p-s, 7.
8	Enter characters: T-V, t-v, 8.
9	Enter characters: W-Z, w-z, 9.
0	Enter characters: 0 ! “ # \$ % & ’ () ô Õ ú ä ö ü α ε θ
*	Enter characters: * + , - . / : ; < = > ? π Σ σ Ω ∞ € £

Table 2-33 Keys for Entering Names

Use this keypad digit . . .	When you want to . . .
#	# = Accepts an entry (only required if two letters on the same key are needed – ex: TOM). Pressing # again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.)
CONF	Clear the character entry one character at a time.
HOLD	Clear all the entries from the point of the flashing cursor and to the right.

7. Press **Hold**.
8. Press **Speaker** to hang up or repeat steps 3~7 to program another System or Group Speed Dial bin.


To dial a System Speed Dialing number:

1. Go off-hook.
2. Press **Redial**.
3. Dial the System Speed Dialing storage code.
 -  Unless you preselect, Trunk Group Routing selects the trunk for the call. The system may optionally select a specific Trunk Group for the call.
 -  If you have a DSS Console, you may be able to press a DSS Console key to chain to a stored number.





To store a System Speed Dialing number under a Programmable Function Key:

1. At multiline terminal, press **Speaker**.
2. Dial **751**.
3. Press the key where the number is to be stored.
4. Dial **27**.
5. Dial System Speed Dial Bin number to put under the key.
6. Press **Speaker** to hang up.



To dial a System Speed Dialing number under a Programmable Function Key:

1. At the multiline terminal, press **Speaker**.
2. Press the key, which has the stored number to be dialed.
 -  The number seizes an outside line and dials out.


To dial a Group Speed Dialing number:

1. Go off-hook.
2. Press **Redial**.
- OR -
3. Press the **Group Speed Dialing** key (PRG 15-07-01 or SC 751: 28).
 *To preselect, press a line key in step 1 (instead of **Speaker**) before pressing **Redial** or **Speed Dialing** key.*
4. Dial the Group Speed Dialing code.
 *The stored number dials out.*
 *Unless you preselect, Trunk Group Routing selects the trunk for the call.*
 *If you have a DSS Console, you may be able to press a DSS Console key to chain to a stored number.*

To check your stored Speed Dialing numbers (display telephone only):

1. Press the **Help** key.
2. For System Speed Dialing, press **Redial**.
Dial the Speed Dialing Code (e.g., common code **001**).
 *If the entire stored number is too long for your telephone's display, press* to see the rest of the number.*
- OR -
For Group Speed Dialing, press the **Group Speed Dialing** key.
- OR -
For System Speed Dialing key, press the **System Speed Dialing** key.
3. Press the **Exit** key.
 *To display additional numbers, repeat from step 1.*

To store a Station Speed Dialing number (display telephones only):

1. Press **Speaker**.
2. Dial **755**.
3. Dial the Station Speed Dial buffer number to be programmed (**0~9**).
1 = Station Speed Dial buffer 1
2 = Station Speed Dial buffer 2
" " " " " "
0 = Station Speed Dial buffer 10
4. Dial the telephone number you want to store (up to 24 digits).
 *Valid entries are 0~9, # and *. To enter a pause, press MIC. To store a Flash, press Recall.*
5. Press **Hold**.

6. Enter the name associated with the Speed Dialing number (display telephones only).

Key for Entering Names	
Use this keypad digit . . .	When you want to . . .
1	Enter characters: 1 @ [¥] ^ _ ` { } → ← Á À Â Ã Ç É Ê Ì Ó
2	Enter characters: A-C, a-c, 2.
3	Enter characters: D-F, d-f, 3.
4	Enter characters: G-I, g-i, 4.
5	Enter characters: J-L, j-l, 5.
6	Enter characters: M-O, m-o, 6.
7	Enter characters: P-S, p-s, 7.
8	Enter characters: T-V, t-v, 8.
9	Enter characters: W-Z, w-z, 9.
0	Enter characters: 0 ! “ # \$ % & ’ () ô õ ú ä ö ü α ε θ
*	Enter characters: * + , - . / : ; < = > ? π Σ σ Ω ∞ € £
#	# = Accepts an entry (only required if two letters on the same key are needed – ex: TOM). Pressing # again = Space.
Conf	Clears the character entry one character at a time (used when entering the name).
Exit	Clears all the entries from the point of the flashing cursor and to the right (used when entering the number, this clears both the number and name).


7. Press **Hold**.
8. Press **Speaker** to hang up.

To dial a Station Speed Dialing number:

1. Press **Speaker**.
2. Dial **#7** (default Service Code).




- OR -

Press the **System Speed Dialing** key (Service Code 751: 27).

 *To preselect, press a line key in step 1 (instead of Speaker).*

3. Dial the Station Speed Dial buffer number (**0 ~9**).

- 1 = Station Speed Dial buffer 1
- 2 = Station Speed Dial buffer 2
- : : : : : :
- 0 = Station Speed Dial buffer 10

-  *The stored number dials out.*
-  *Unless you preselect, Trunk Group Routing selects the trunk for the call. The system may optionally select a specific Trunk Group for the call.*
-  *If you have a DSS Console, you may be able to press a DSS Console key to chain to a stored number.*

Station Add-On Console

Enhancements

Version 1500 or higher:

- Allows DSS/One-Touch keys to be used for one-touch transfer without using the Transfer key.
- DSS keys distinguishes whether the telephone is set for DND/Call Forward All Calls or if the telephone is off-hook.

Description

The Station Add-On Console functions with a Multiline Terminal to provide an additional 16 DSS/BLF keys. The Busy Lamp Field status is shown by icons for each station or feature. This console also has an additional 100 programmable speed dials that are separate from the System or Station Speed dials.

Conditions

- The Add on Module (ADM) Directory (PC application) must be used to edit the directory database. Refer to the documentation provided with the D16(LD)-R() ADM unit.
- A maximum of 100 Directories can be stored in the D16(LD)-R() ADM.

Default Setting

None

System Availability

Terminals

DTR/DTH-8-2

DTR/DTH-8D-2

DTH-16-2

DTR/DTH-16D-2

DTR/DTH-16-2

DTR/DTH-16LD-2

Required Component(s)

D16(LD)-R() ADM

Related Features

One-Touch Calling

Programmable Function Keys


Programming

- ➔ **11-11-17 Service Code Setup (for Setup/Entry Operation) – Programmable Function Key Programming (Dialing 751 Service Code)**
Set the service code (default 751) to assign 2-digit function codes to the Function keys.
- ➔ **11-11-38 Service Code Setup (for Setup/Entry Operation) – Programmable Function Key Programming (Dialing 852 Service Code)**
Set the service code (default 752) to assign 3-digit function codes to the Function keys.
- ➔ **15-07-01 Programmable Function Keys**
Assign function keys to the D16(LD)-R() ADM.
- ➔ **15-20-01 LCD Line Key Name Assignment**
Assign a name to each LCD Line Key of the DTH/DTR-16LD-1 Telephone or D16(LD)-R() ADM.
Up to eight characters can be assigned.
(default: LK01~LK08 = CO001~CO008; LK09~LK32 are blank)

Operation

To program a key on the D16(LD)-R() ADM

1. Go off-hook.
2. Dial the **Service Code for Programming Feature** key 2-digit (751), or 3-digit (752).
3. Press the **Line** key to program.
4. Dial the function number.

 Refer to PRG 15-07-01 for complete operation.

Station Hunt

Description

After calling a busy extension, a call immediately hunts to the next available member of the Hunt Group (Department Group). The caller does not have to hang up and place another Intercom call if the first extension called is unavailable.

Conditions

- If required, use this option to change the Department Step Calling Single Digit Service Code (default: 2).
- A function key for Department Step Calling can be assigned (code 36).
- In PRG 20-08-12, enable (1) or disable (0) an extension user ability to use Department Step Calling.

Default Setting

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

None

Programming

- ➔ **16-01-03 Department Group Basic Data Setup – Department Routing When Busy (Auto Step Call)**
Assign if a call to busy hunts (1) or does not hunt (0) to the next available member of the Hunt Group (Department Group).
(default: 0)
✎ Refer [Department Calling on page 2-277](#) to set up the Department Group.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Enable (1)/Disable (0) Call Queuing to the extension. This must be set to 0 for Station Hunting to work.
(default: 1)
- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow a busy extension to manually (0) or automatically (1) receive off-hook signals. This must be set to 0 for Station Hunting to work.
(default: 1)

Related Features

Department Calling

Department Step Calling

Operation

To make a Step Call:

1. Place a call to a busy extension.

Station Message Detail Recording

Description

Station Message Detail Recording (SMDR) provides a record of the system trunk calls. Typically, the record outputs to a customer-provided printer, terminal or SMDR data collection device. SMDR allows you to monitor the usage at each extension and trunk. This makes charge-back and traffic management easier.

SMDR provides the following options:

Abandoned Call Reporting

The SMDR report includes calls that rang into the system but were unanswered (i.e., abandoned). SMDR can include all abandoned calls or only those abandoned calls that rang longer than the specified duration. The Abandoned Call Report helps you keep track of lost business.

Blocked Call Reporting

When Toll Restriction blocks a call, you can have SMDR print the blocked call information. Or, you can have SMDR exclude these types of calls. With Blocked Call Reporting, you can better customize Toll Restriction for the site application.

Customized Date Format

The SMDR header can show the report date in one of three formats: American, European or Japanese. Set the format for your preference.

Transferred Call Tracking

SMDR shows each extension share of a transferred call. If an outside call is transferred among four extensions, SMDR shows how long each of the callers stayed on the call.

Data Call Tracking

Data Call Tracking can log the system's internal data calls. Since SMDR normally logs external (trunk) data calls, Data Call Tracking lets you get a complete picture of data terminal activity.

Digit Counting

With Digit Counting, SMDR can selectively keep track of toll calls. For example, if the digit count is nine, SMDR does not include toll calls in the home area code. Digit Counting permits SMDR to include only the types of calls you want to monitor.

Digit Masking

Digit Masking lets you X out portions of the number dialed on the SMDR report. A digit mask of seven, for example, masks out all exchange codes (NNXs) and local addresses. Digit Masking makes it easier to keep track of calling patterns, without having to interpret each individual number. You can also use Digit Masking to block out access and security codes.

Duration Monitoring

SMDR can include calls of any duration, or only those that last longer than the interval you specify. If you want to keep track of all trunk activity, use a short duration. To keep track of only significant usage, use a longer duration.

 Extension Exclusion

You can selectively exclude extensions from the SMDR report. This ensures privacy for high-profile callers. For example, the company attorney negotiating a merger may not want his calls to show up on an in-house report.

 PBX Call Reporting

If your system is behind a PBX, you can have SMDR monitor all traffic into the PBX or just calls placed over PBX trunks. The SMDR record can include all PBX calls (including calls to PBX extensions) or just calls that include the PBX trunk access code.

 Serial and USB SMDR Communication

The system is compatible with both serial and USB SMDR devices. This gives you many SMDR output options. For example, you can output the SMDR report to a high speed printer or send it to disk through a PC's serial or USB port.

 Trunk Exclusion

Use Trunk Exclusion to exclude certain trunks not subject to per-call charges (like WATS lines) from the SMDR report. This makes call accounting easier, since you review only those calls with variable costs.

 Usage Summaries

SMDR can automatically print daily, weekly and monthly call activity summaries. Each summary includes the total number of regular trunk calls and ISDN trunk calls, and the costs for each type. The daily report prints every day at midnight. The weekly report prints every Sunday night at midnight. The monthly report prints at midnight on the last day of the month.

 Extension Name or Number

The SMDR report can include an extension name or extension number. Choose the method that makes it easier for you to track call usage.

This feature requires a connection to the system using a CTA or CTU adapter, or through the serial port on the Electra Elite IPK II CPU II (requires the USB driver).

(The LAN port only provides information through LAN-capable programs, such as HyperTerminal. Printing of the SMDR information must be done from that program.)

SMDR Enhanced for Caller ID

The SMDR output is enhanced to include up to 16 or 24 characters of the Caller ID name information (depending on the view option selected in Program 35-02-18). You can select to display the Caller ID number or name or the DID number. If you wish to display the Caller Name in the DIALLED NO./CLI and ACCOUNT area, select 2 in the updated Program 35-02-15 and 1 in Program 35-02-17.

If the Caller ID name is not received, the area for Caller ID Name is left blank.

Sample SMDR Report

For example, with Program 35-01-09 = 0 (Format for NA) and Program 35-02-17 = 1 (Caller ID Name), if a call is received with the Caller ID Name of NEC Infrontia Corporation (24 characters), the following SMDR record is displayed:


CLASS	TIME	DATE	LINE	DURATION	STATION	DIALLED No./CLI	ACCOUNT
POT	10:52	12/09	002	00:00:10	2001	2142623801	08754
PIN	10:52	12/09	001	00:00:20	2017	2142623802	NECinfrontia Corp.
PIN	10:53	12/09	002			2142623801	NO ANSWER

If Program 35-02-18 = 1 (Caller ID Name Output Method) is set to line feed, the SMDR displays as follows:

CLASS	TIME	DATE	LINE	DURATION	STATION	DIALLED No./CLI	ACCOUNT
POT	10:52	12/09	002	00:00:10	2001	2142623801	08754
PIN	10:52	12/09	001	00:00:20	2017	2142623802	NECinfrontia Corp.
							NEXT NECinfrontia Corp.
PIN	10:53	12/09	002			2142623801	NO ANSWER

Table 2-34 SMDR Report Definitions

Report Headings	Definitions
Call Record Number	SMDR record number (consecutive)
CLASS	Type of call (see Class Definitions below)
TIME	Time call placed or answered. (For Transferred calls, shows time user picked up Transfer.)
DATE	Date the call was made
LINE	Trunk number used for call
DURATION	How long call lasted. (For Transferred calls, shows how long user was on call after answering the Transfer.)
STATION	Extension number of call owner (i.e., extension that first placed or answered call) (For Transferred calls, there can be more than one owner – depending on how many extensions shared the call.)
DIALLED No./CLI	For outgoing calls, the number dialed or, for incoming calls, the Caller ID information
ACCOUNT	Account Code number entered by extension user
Class Definitions	
POT	Outgoing trunk call
POTA	Outgoing trunk call placed using Toll Restriction Override
PIN	Incoming trunk calls
ALB	All lines in group are busy (group number follows TIME field)
BRD	Call blocked due to Toll Restriction
PTRS	Transferred call
IVIN	BRI/PRI inbound trunk call

 The following tables are displayed in the SMDR output with these priorities:

PRG 35-02-14 Data Enable (1st Priority) PIN 10:53 12/09 022 2142623801 NO ANSWER

PRG 35-02-16 Received Dial Number Enable (2nd Priority) PIN 10:53 2038887777 2142623801 NO ANSWER

PRG 35-02-03 Name or Number (3rd Priority) PIN 10:53 Trunk 002 2142623801 NO ANSWER

Table 2-35 SMDR Report Format with Program 35-02-14 Set to 0

Character Position	Field Definition
Header Line 1	
1~60	Spaces
61~70	MM/DD/YYYY
71	Space
72~75	PAGE
76	Space
77~79	Report page number (e.g., 001)
CR & LF	Carriage return and line feed
Header Line 2	
1~5	CLASS
6	Space
7~10	TIME
11~14	Spaces
15~18	LINE
19~22	Spaces
23~30	DURATION
31~32	Spaces
33~39	STATION
40~44	Spaces
45~51	DIALLED
52	Space
53~59	No./CLI
60~63	Spaces
64~70	ACCOUNT
CR & LF	Carriage return and line feed
LF	Line feed
SMDR Record	
1~4	Call type (e.g., POT for outgoing)

Table 2-35 SMDR Report Format with Program 35-02-14 Set to 0 (Continued)

Character Position	Field Definition
5	Space
6~10	Time in 24 hour clock (HH:MM)
11	Space
12~21	LINE
22	Space
23~30	Call Duration (HH:MM:SS)
31	Space
32~41	Station number or name
42	Space
43~62	Number dialed (20 digits maximum)
63	Space
64~79	Account number or NO ANSWER

Table 2-36 SMDR Report Format with Program 35-02-14 Set to 1

Character Position	Field Definition
Header Line 1	
1~60	Spaces
61~70	MM/DD/YYYY
71	Space
72~75	PAGE
76	Space
77~79	Report page number (e.g., 001)
CR & LF	Carriage return and line feed
Header Line 2	
1~5	CLASS
6	Space
7~10	TIME
11	Spaces

Table 2-36 SMDR Report Format with Program 35-02-14 Set to 1 (Continued)

Character Position	Field Definition
12~15	DATE
16~17	Spaces
18~21	LINE
22	Space
23~30	DURATION
31~32	Spaces
33~39	STATION
40~44	Spaces
45~51	DIALLED
52	Space
53~59	No./CLI
60~63	Spaces
64~70	ACCOUNT
CR & LF	Carriage return and line feed
LF	Line feed
SMDR Record	
1~4	Call type (e.g., POT for outgoing)
5	Space
6~10	Time in 24 hour clock (HH:MM)
11	Space
12~16	DATE
17	Space
18~21	LINE
22	Space
23~30	Call Duration (HH:MM:SS)
31	Space
32~41	Station number or name
42	Space
43~62	Number dialed (20 digits maximum)

Table 2-36 SMDR Report Format with Program 35-02-14 Set to 1 (Continued)

Character Position	Field Definition
63	Space
64~79	Account number or NO ANSWER

Table 2-37 SMDR Summary Report

<p>OUTGOING CALL/COST SUMMARY FOR DAY OF nn/nn/nn</p> <p>TOTAL NO. OF OUTGOING PSTN CALLS: 0 TOTAL NO. OF OUTGOING ISDN CALLS: 0 NO. OF OUTGOING PSTN CALLS COSTED: 0 COST: 0 NO. OF OUTGOING ISDN CALLS COSTED: 0 COST: 0</p> <p>OUTGOING CALL/COST SUMMARY FOR WEEK ENDING nn/nn/nn</p> <p>TOTAL NO. OF OUTGOING PSTN CALLS: 49 TOTAL NO. OF OUTGOING ISDN CALLS: 0 NO. OF OUTGOING PSTN CALLS COSTED: 0 COST: 0 NO. OF OUTGOING ISDN CALLS COSTED: 0 COST: 0</p> <p>OUTGOING CALL/COST SUMMARY FOR MONTH ENDING nn/nn/nn</p> <p>TOTAL NO. OF OUTGOING PSTN CALLS: 49 TOTAL NO. OF OUTGOING ISDN CALLS: 0 NO. OF OUTGOING PSTN CALLS COSTED: 0 COST: 0 NO. OF OUTGOING ISDN CALLS COSTED: 0 COST: 0</p>

Conditions

- SMDR data does not include Intercom calls.
- The SMDR call buffer stores 500 calls. The buffer stores calls when the SMDR device is unavailable. When the buffer fills, the oldest record is deleted to allow the new record to be saved.
- When SMDR reports are enabled using the same port as the Traffic Reporting feature (example: 147), the SMDR blocks the Traffic reports. Unplugging the cable and plugging it back in again allows Traffic reports to print.

- SMDR requires a connection to the CPUII via a COM port, CTA/CTU or LAN connection. The system can also use a connection to the system via a CTA/CTU adapter (the CTU adapter also requires a USB driver). Once you designate a CTA or CTU for SMDR, you cannot use that extension for placing and answering other data calls. SMDR and the Traffic Reports should not use the same CTA/CTU.
- If no answer is received, NO ANSWER is displayed regardless of the system programming for the Caller ID display option.
- The setting in Program 35-02-18 works regardless of the entry in Program 35-02-15 or 35-02-17.
- When Program 35-02-18 is set to 1, the first and second lines are sometimes separated. When the buffer is full, the overflowed data may not be shown.
- The special characters used in the Electra Elite IPK II system cannot be output to the SMDR – they are converted to _.
- To use the PBX Call Reporting option, program system for behind PBX operation.
- Traffic Management Reports and SMDR should not use the same CTA/CTU port.
- SMDR can be output to eight simultaneous devices. One SMDR Port can be set to the KSU COM port and one SMDR Port can be set for to the LAN port. The remaining six outputs can be set to different CTA/CTU adapters.

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

Software License

Related Features

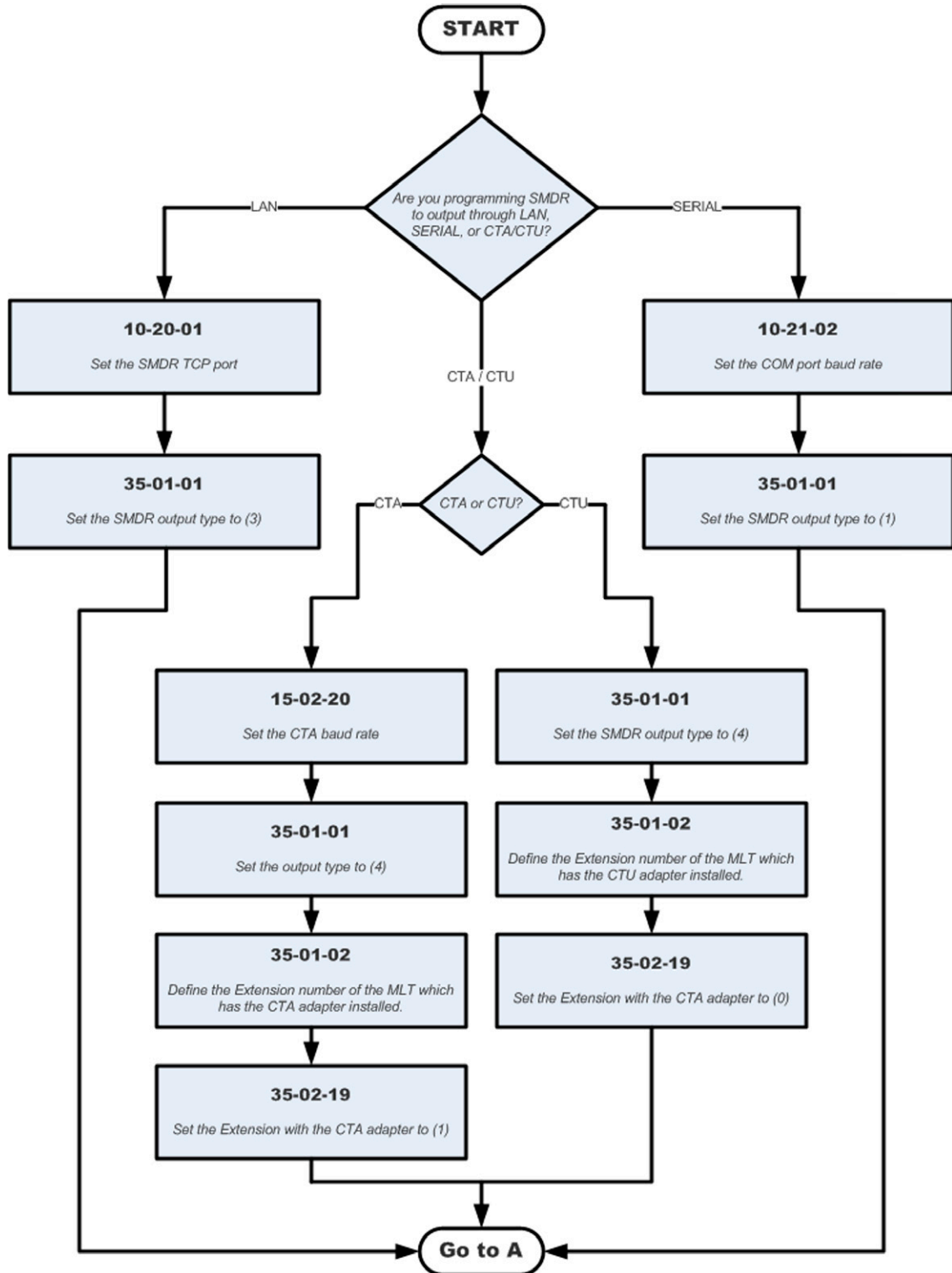
PBX Compatibility

Traffic Reports

Programming

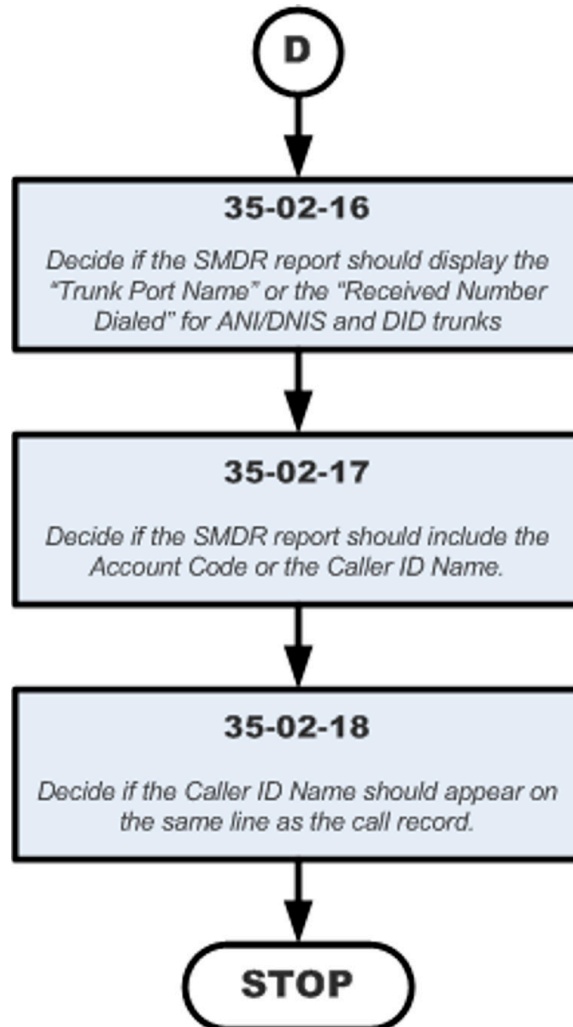
SMDR flowcharts are located on the following pages.

SMDR with a CPU Connection (Serial/CTA/CTU/Ethernet)



SMDR with a CPU Connection (Serial/CTA/CTU/Ethernet) (continued)



SMDR with a CPU Connection (Serial/CTA/CTU/Ethernet) (continued)

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- ➔ **10-20-01 LAN Setup for External Equipment – TCP Port**
Define the TCP port (0~65535) when communicating to the SMDR (type 5).
(default: 0)

 - ➔ **10-20-03 LAN Setup for External Equipment – Keep Alive Time**
Set keep alive time (0~255 seconds).
(default: 30)

 - ➔ **10-21-02 CPUII Hardware Setup – Baud Rate for COM Port**
If the SMDR connection is made using the COM port on the CPUII, define the baud rate.

Baud Rates:
0 = 4800
1 = 9600
2 = 19200 (default)
3 = 38400

 - ➔ **14-01-06 Basic Trunk Data Setup – SMDR Printout**
For each trunk, enter 1 if trunk calls should appear on SMDR report. Enter 0 if trunk calls should not appear on SMDR report.
(default: 0)

 - ➔ **15-01-03 Basic Extension Data Setup – SMDR Printout**
For each extension, enter 1 if extension calls should appear on SMDR report. Enter 0 if extension calls should not appear on SMDR report.

 - ➔ **15-02-19 Multiline Telephone Basic Data Setup – CTA Data Communication Mode**
Select 0 if the dip switch settings are set to PC connection or select 1 if printer/SMDR connection is selected on the CTA/CTU adapter.
(default: 1)

 - ➔ **15-02-20 Multiline Telephone Basic Data Setup – Baud Rate for CTA Port**
Set the baud rate to be used by the CTA.

Baud Rates:
0 = 4800
1 = 9600
2 = 19200 (default)

 - ➔ **35-01-01 SMDR Options – Output Port Type**
Specify the type of connection used for SMDR. The baud rate for the COM port should be set in PRG 10-21-02 or PRG 15-02-20.

Connection Types:
0 = No Setting (default)
1 = COM (CPUII)
3 = LAN (CPUII LAN output)
4 = CTA/CTU


 - ➔ **35-01-02 SMDR Options – Output Destination Number**
Specify the SMDR printer output port (CTA/CTU extension number – up to 8 digits).
(default: No Setting)
-
-

- ➔ **35-01-03 SMDR Options – Header Language**
Specify the language in which the SMDR header should be printed.
- Languages:
0 = English (default)
1 = German
2 = French
3 = Italian
4 = Spanish
- ➔ **35-01-04 SMDR Options – Omit Digits**
Enter the number of digits (1~24) you want SMDR to block (i.e., X out). Enter 0 not to block any digits.
(default: 0)
- ➔ **35-01-05 SMDR Options – Minimum Digits**
Enter the minimum number of digits a user must dial (1~24) before the system includes a call on the SMDR report. Enter 0 to include all outgoing calls, regardless of the number of digits dialed.
(default: 0)
- ➔ **35-01-06 SMDR Options – Minimum Call Duration**
Enter the minimum duration of a call (0~65535 seconds) that prints on the SMDR report. Enter 0 to have calls of any duration print.
(default: 0)
- ➔ **35-01-07 SMDR Options – Minimum Ring Time (For Incoming Calls)**
Enter how long an unanswered call must ring (0~65535 seconds) before SMDR logs it as No Answer). Enter 0 to allow all No Answer calls to print.
(default: 0)
- ➔ **35-01-08 SMDR Options – Format Selection**
Do not change:
This option is added to allow an increased account code field from 8 to 16 when used in the U.K. This allows 16 characters of the Caller ID name to be displayed. For the U.S., this option is set to 0 and should remain at this setting as 16 characters are already provided for the account code field.
(default: 0)
- ➔ **35-02-01 SMDR Output Options – Toll Restricted Call**
Enter 1 if you want the SMDR report to include calls blocked by Toll Restriction. Enter 0 to exclude blocked calls.
(default: 1)
- ➔ **35-02-02 SMDR Output Options – PBX Calls**
If system is behind a PBX, enter 1 to have SMDR include all calls to the PBX. Enter 0 to have SMDR include only calls dialed using PBX trunk access code.
(default: 1)

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- **35-02-03 SMDR Output Options – Trunk Number or Name**
Select whether the system should display the trunk name (0) or the number (1) on SMDR reports
(default: 1)
✎ If this option is set to 0, Program 35-02-14 must be set to 0.
 - **35-02-04 SMDR Output Options – Summary (Daily)**
 - **35-02-05 SMDR Output Options – Summary (Weekly)**
 - **35-02-06 SMDR Output Options – Summary (Monthly)**
Enter 1 to enable a summary report. Enter 0 to disable a summary report. The daily report prints every day at midnight. The weekly report prints every Sunday night at midnight. The monthly report prints at midnight on the last day of the month.
(default: 1)
 - **35-02-07 SMDR Output Options – Toll Charge Cost**
Set this option to (1) have the SMDR report include toll charges.
(default: 1)
 - **35-02-08 SMDR Output Options – Incoming Call**
Enter 1 if you want the SMDR report to include incoming calls. Enter 0 if you want the SMDR report to exclude incoming calls.
(default: 1)
 - **35-02-09 SMDR Output Options – Extension Number or Name**
Enter 0 if you want the SMDR report to include the extension name. Enter 1 if you want the SMDR report to include the extension number.
(default: 1)
 - **35-02-10 SMDR Output Options – All Lines Busy (ALB) Output**
Enter 1 to report information when all lines in a group are busy and an extension user tries to access the group. Enter 0 if this information should not be included.
(default: 0)
 - **35-02-11 SMDR Output Options – Walking Toll Restriction Table Number**
(default: 1)
 - **35-02-12 SMDR Output Options – DID Table Name Output**
Determine if the DID table name should be displayed for incoming DID calls.
0 = Not Displayed
1 = Displayed
(default: 0)
 - **35-02-13 SMDR Output Options – CLI Output When DID to Trunk**
Determine if the Caller ID should be displayed when the incoming DID number is transferred to an outgoing trunk.
0 = Not Displayed
1 = Displayed
(default: 0)

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-
- ➔ **35-02-14 SMDR Output Options – Date**

Determine whether the date should be displayed on SMDR reports.
0 = Not Displayed
1 = Displayed
(default: 0)

 *This option must be set to 0 if the trunk name is set to be displayed in Program 35-02-03.*

 - ➔ **35-02-15 SMDR Output Options – CLI/DID Number Switching**


Enter 0 to display the Caller ID number. Determine if the Caller ID number (0), DID number (1) or Caller ID name (2) should be displayed in the SMDR output.
(default: 0)

 - ➔ **35-02-16 SMDR Output Options – Trunk Name or Received Dialed Number**

Determine how the SMDR should print incoming calls on ANI/DNIS or DID trunks. If set to (1), ANI/DNIS trunks can print DNIS digits. For DID trunks, if the received number is not defined in Program 22-11-01, a number is not printed. If set to (0) trunk names are printed instead (as assigned in Program 14-01-01).
(default: 0)


 - ➔ **35-02-17 SMDR Output Options – Print Account Code or Caller Name of Incoming Call**

Determine whether the Account Code (0) or Caller ID name (1) should appear in the SMDR record.
(default: 0)

 *Program 35-01-08 must be set to 0 for this entry to be followed.*

 - ➔ **35-02-18 SMDR Output Options – Print Mode for Caller Name of Incoming Call**

Select whether to display up to 16 characters of the Caller Name on the same line as the call record (0) or if a line feed should be added and up to 24 characters of the Caller Name will be displayed on the following line (1). If the line feed option is selected, the Caller Name is displayed on the next line as : NEXT Caller Name. The default entry for this option is 0. This setting works regardless of the setting in Program 35-02-15.
(default: 0)

 *With this option set to 1, if your communications program (such as HyperTerminal) has the line wrap option enabled in the ASCII setup, an additional line break may appear above the Caller name line.*

 - ➔ **35-03-01 SMDR Port Assignment for Trunk Group**

Assign the SMDR port for each trunk group. For each Trunk Group, select the SMDR port to which the incoming SMDR information should be sent.
Trunk Groups: 1~100
SMDR Ports: 1~8
(default: 1)

 - ➔ **35-04-01 SMDR Port Assignment for Department Groups**

Assign the SMDR port for each Department Group. For each Department Group, select the SMDR port to which the outgoing SMDR information should be sent.
Department Groups: 1~64
SMDR Ports: 1~8
(default: 1)
-
-

- ➡ **80-05-01 Date Format for SMDR and System – Date Format**
Set the date format for SMDR.
0 = American (default)
1 = Japanese
2 = European

Operation

Once installed and programmed, SMDR operation is automatic.

Station Name Assignment – User Programmable

Description

This feature allows a user to program the Station Name for their telephone extension or any extension within the system. The name is displayed on the multiline terminal LCD when an intercom or K-CCIS call is placed.

Conditions

- Display telephones use extension names for Directory Dialing.
- Single line telephone extensions cannot program names.

Default Setting

Enabled

System Availability

Terminals

All display Multiline Terminals

Required Component

None

Related Features

Directory Dialing


Name Storing

Programming


- ➔ **15-01-01 Basic Extension Data Setup – Extension Name**
Program name for system extensions.
- ➔ **15-07-01 Programmable Function Keys**
Assign an Extension Name Change key (55) to extensions.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-13-21 Class of Service Options (Supplementary Service) – Extension Name**
Enable (1)/Disable (0) an extension user ability to program their name.
(default: 1)

Operation

To program your extension name:

1. Press **Speaker**.
2. Dial **700**.
- OR -
Press the **Extension Name Change** key (PRG 15-07 or SC 751: 55).
3. Press **Hold**.
4. Enter the name. (Refer to [Table 2-31 Selectable Display Messaging Defaults on page 2-873.](#))
 *Your name can be up to 12 digits maximum.*
5. Press **Hold**.
6. Press **Speaker** to hang up.

To program any extension name:

1. Press **Speaker**.
2. Dial **700**
- OR -
Press the **Extension Name Change** key (PRG 15-07 or SC 751: 55).
3. Enter the extension number to be named.
4. Enter a name. (Refer to [Table 2-38 Keys for Entering Names.](#))
 *The name can be have to 12 digits maximum.*

5. Press the **Hold** key.
6. Press **Speaker** to hang up.

Table 2-38 Keys for Entering Names

Use this keypad digit . . .	When you want to . . .
1	Enter characters: 1 @ [¥] ^ _ ` { } Æ " Á À Â Ã Ç É Ê ì ó
2	Enter characters: A-C, a-c, 2.
3	Enter characters: D-F, d-f, 3.
4	Enter characters: G-I, g-i, 4.
5	Enter characters: J-L, j-l, 5.
6	Enter characters: M-O, m-o, 6.
7	Enter characters: P-S, p-s, 7.
8	Enter characters: T-V, t-v, 8.
9	Enter characters: W-Z, w-z, 9.
0	Enter characters: 0 ! “ # \$ % & ’ () ô Õ ú ä ö ü α ε θ
*	Enter characters: * + , - . / : ; < = > ? π Σ σ Ω ∞ ϕ £
#	# = Accepts an entry (only required if two letters on the same key are needed – ex: TOM). Pressing # again = Space. (In system programming mode, use the right arrow Softkey instead to accept and/or add a space.)
Conf	Clear the character entry one character at a time.
Hold	Clear all the entries from the point of the flashing cursor and to the right.

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Station Relocation

Description

Station Relocation allows a station to be moved from one location to another, without having to reprogram the station data. The stations features and extension number are the same after it is moved to the new location.

Conditions

- This feature can be used to swap or relocate multiline and single line terminals.
- Single line includes SLT adaptors.
- The destination extension must be idle. If the station is not idle, busy tone is heard.
- If the Extension Swap service code is dialed from an extension that does not have an extension swap password programmed, busy tone is heard.
- If the Extension Swap service code is dialed from an extension whose Class of Service does not allow Extension Data Swap, busy tone is heard.
- If the destination extension entered is not a valid extension, busy tone is heard.
- The following user setting data is relocated with the extension. All other user setting data is not relocated or cleared.
 - DND
 - Call Forwarding
 - Memo Dial
 - Last Number Dial History
 - Saved Number Dial
 - Incoming History
 - MIC Key LED Status
 - VM MW LED Status

Refer to the Programming section in this feature for system programs that are swapped.

Default Setting

None

System Availability

Terminals

All Multiline Terminals and Single Line Telephones

Required Component(s)

None

Related Features

None

Programming

- **11-15-12 Service Code Setup, Administrative (for Special Access – Extension Data Swap**
Ext. Data Swap = xxx (service code in accordance with PRG 11-01)
- **20-13-42 Class of Service Options (Supplementary Service) – Extension Data Swap Enabling**
0 = Off
1 = On (default)
- **92-05-01 Extension Data Swap Password – Password**
Tel = Extension Number
(default: Blank)
- **The following programs are swapped when Station Relocation is used:**
 - PRG 11-02 Extension Numbering
 - PRG 12-05 Night Mode Group Assignment for Extensions
 - PRG 13-03 Speed Dial Group Assignment for Extensions
 - PRG 15-01 Basic Extension Data Setup
 - PRG 15-02 Multiline Telephone Basic Data Setup
 - PRG 15-03 Single Line Telephone Basic Data Setup
 - PRG 15-06 Trunk Access Map for Extensions
 - PRG 15-07 Programmable Function Keys

- PRG 15-08 Incoming Virtual Extension Ring Tone Setup
- PRG 15-09 Virtual Extension Ring Assignment
- PRG 15-10 Incoming Virtual Extension Ring Tone Order Setup
- PRG 15-11 Virtual Extension Delayed Ring Assignment
- PRG 15-12 Conversation Recording Destination for Extensions
- PRG 15-14 Programmable One-Touch Keys
- PRG 16-02 Department Group Assignment for Extensions
- PRG 20-06 Class of Service for Extensions
- PRG 21-02 Trunk Group Routing for Extensions
- PRG 21-04 Toll Restriction Class for Extensions
- PRG 21-07 Toll Restriction Override Password Setup
- PRG 21-10 Dial Block Restriction Class Per Extension
- PRG 21-11 Extension Ringdown (Hotline) Assignment
- PRG 21-13 ISDN Calling Party Number Setup for Extensions
- PRG 21-15 Individual Trunk Group Routing for Extensions
- PRG 21-19 IP Trunk (SIP) Calling Party Number Setup for Extension
- PRG 21-20 SIP Trunk Call Discernment Setup for Extension
- PRG 23-02 Call Pickup Groups
- PRG 23-03 Universal Answer/Auto Answer
- PRG 23-04 Ringing Line Preference for Virtual Extensions
- PRG 24-03 Park Group
- PRG 26-04 ARS Class of Service
- PRG 26-07 LCR Cost Center Code Table
- PRG 31-02 Internal Paging Group Assignment
- PRG 41-02 ACD Group and Agent Assignments
- PRG 41-17 ACD Login Mode Setup
- PRG 42-02 Hotel/Motel Telephone Setup

Operation

To exchange two terminals:

1. Pick up the handset or press **Speaker**.
2. Dial the Extension Data Swap Service Code – not assigned at default (PRG 11-15-12).
3. Dial the Extension Data Swap Password – not assigned at default (PRG 92-05-01).
4. Dial the extension to be swapped with or relocated to.
5. When successfully completed, confirmation tone is heard and the display shows completed.
6. Press **Speaker** twice to exit.

Synchronous Ringing

Description

Synchronous Ringing synchronizes CO/PBX incoming ringing with the incoming ringing pattern from a Central Office.

Conditions

- When the multiline terminal is ringing at Secondary Extension (SE)/Virtual Extension (VE) key, Synchronous Ring works.
- Synchronous Ringing is not supported for Tie/DID incoming calls, Off-Hook Ringing, or CO/PBX Ring Transfers.
- If Synchronous Ringing is enabled, the VRS Preamble Message cannot be used.

Default Setting

Enabled

System Availability

Terminals

All Stations except single line telephones connected to AP(R)-R or APR-U Unit

Required Component(s)

None

Related Features

None

Programming

- ➔ **14-02-17 Analog Trunk Data Setup – Sync. Ringing**
Enable/Disable per trunk.
0 = Disable
1 = Enable (default)

- ➔ **20-15-01 through 20-15-11 Ring Cycle Setup**
Define the ringing cycles for each ring tone, when Synchronous Ringing is disabled.

- ➔ **22-03-01 Trunk Ring Tone Range – Ring Tone Pattern**
Define the ring tone for each trunk 0~8.
(default: 0).

Operation

None

T1 Trunking (with ANI/DNIS Compatibility)

Enhancements

This feature was added with **Version 1100**.

Version 1500 or higher is required when PRG 10-39-01 is set to Enable. This allows the T1/PRI Interface ETU to be programmed as a 4/8/12/16/20/24 port Fractional T1/PRI.

Description

The T1/PRI Interface ETU gives the system T1 trunking ability. This ETU uses a single universal slot and provides up to 24 trunk circuits. In addition to providing digital-quality trunking, the T1/PRI Interface ETU allows you to have maximum trunking ability with fewer ETUs. This in turn makes more universal slots available for other functions.

You can program each T1/PRI ETU for any combination of the following trunks:

- CO loop start
- CO ground start
- Direct Inward Dialing
- Tie Lines²

With **Version 1100 or lower**, the T1/PRI Interface ETU uses the first block of 24 consecutive trunks. For example, if you have an COI(8) ETU installed for trunks 1~8, the T1/PRI Interface ETU automatically uses trunks 9~32. If you have COI(8) ETUs installed for trunks 1~8 and 17~24, the T1/PRI ETU uses trunks 25~48. The T1/PRI Interface cannot use trunks 9~16 (even if available) since they are not part of a consecutive block of 24 trunks. Each T1/PRI ETU requires that 24 consecutive ports be available in the system even if not all the ports are used otherwise the ETU does not function. This also applies for **Version 1500 or higher** when PRG 10-39-01 is set to Disabled.

The T1/PRI Interface ETU can be programmed as a 4/8/12/16/20/24 port Fractional T1/PRI. **Version 1500 or higher** is required when PRG 10-39-01 is set to Enabled.

2. Two-wire (four-lead) type 1 Tie Lines (FIC TL11M) only.

ANI/DNIS Compatibility

The system is compatible with Telco T1 Automatic Number Identification (ANI) and Dialed Number Information Service (DNIS) services. A compliment to Caller ID service, ANI/DNIS Compatibility provides:

Receive Format

The Receive Format must be set as *ANI*DNIS* in Program 34-09-01 option 4, which is treated as a Feature Group D format. (Example of ANI Information KP009727517645STKP7100ST.)

Flexible Routing

Based on the data received, the system can route the incoming ANI/DNIS call to:

- An extension
- An ACD or Voice Mail master extension number
- A VRS and play a VRS message to the caller
- A Department Group pilot number
- A trunk Ring Group

Route According to DID Translation Table or Speed Dial Bins

Calls can be routed based on either the number of digits defined in Program 22-09-01 (digits 1~8) or by digits entered in Speed Dial bins in Program 13-04-01.

ANI/DNIS Data Displayed as Caller ID Data

Data Error and Unanswered Call Handling

If a call can not be completed, send it to a predetermined Ring Group or play supervisory tones to the caller.

Conditions

- T1 Trunking requires a T1/PRI Interface ETU and a customer-provided CSU/DSU to interface with the Telco. Consult your sales representative and the System Hardware Manual for additional details.
- ANI/DNIS Compatibility requires the use of system DTMF receivers. When all receivers are busy, the incoming ANI/DNIS call waits for a receiver to become available.
- The ANI/DNIS/Address data received from the Telco can have up to 10 digits.
- An extension Class of Service (Program 20-09-02) determines whether ANI information is displayed.
- Refer to [Dial Tone Detection on page 2-293](#) for the specifics on how the system detects dial tone.
- The T1 Tie Line can be used for networking.

- The T1/PRI Interface ETU provides Tie Line service. All programming parameters are the same as those used for analog Tie Lines (except for the additional T1/PRI Interface ETU settings).
- DTI-U40 ETUs can be installed in any Interface slot (1~8) of any cabinet (1~3).
- The DTI-U40 ETU supports Dial Pulse and DTMF signaling.
- With an Electra Elite IPK II – Expanded Port Package, up to 200 T1 trunks can be assigned.
- You can use T1 trunks in place of standard analog trunks. The procedures for placing and answering calls are the same for both types of trunks.
- The T1/PRI Interface ETU provides DID service. All programming parameters are the same as those used for analog DID trunks (except for the additional T1/PRI Interface ETU settings).
- SMDR can print trunk port names or received dialed number for ANI/DNIS or DID trunks. If enabled in programming, DNIS digits can be printed on the SMDR reports instead of the trunk name.
- T1 trunks follow Tie Line toll restriction programming (PRG 34-01-05 and PRG 34-08).
- When using Fractional T-1s, the card comes up with zero ports until PRG 10-03-06 is set to the 4/8/12/16/20/24(auto) and then reset.
- If the number of ports are changed for a fractional T-1 in PRG 10-03-06, the trunk ports could be reassigned if the numerical sequence would split or it could fit into an empty gap of trunk ports. **Version 1500 or higher** with PRG 10-39-01 set to Enabled is required.

Examples:

The COI(8) ETU is assigned for trunk ports 1~8 and 17~24 and the T-1 (12 ports) was assigned as ports 25~36, the number of T-1s change to eight ports instead of 12. The new trunk port numbers are assigned as 9~16 because the eight ports can fit into the gap between ports 8 and 17 without splitting the numerical port number sequence.

The COI (8) ETU is assigned for trunk ports 1~8 and 17~24 and the T-1 (eight ports) was assigned as 9~16 and then the T-1 was changed from eight to 12 ports. The new trunk port numbers are assigned as 25~36 because the port number sequence would have to be split in order to keep them within the original number sequence (9~16) and splitting the numerical port number sequence is not supported.

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

DTI-U40 ETU with **firmware V5.00 or higher**

Locally provided CSU/DSU

Required Software

Version 1100 or higher

Related Features

Caller ID

Central Office Calls, Answering

Central Office Calls, Placing

Code Restriction

Dial Tone Detection

Direct Inward Dialing (DID)

ISDN Compatibility

Station Message Detail Recording

Tie Lines

Programming

- ➔ **10-03-01 ETU Setup (Version 1500 or higher required)**

Set various T1 trunk options for compatibility with the local Telco. For ANI/DNIS, the following settings in Program 10-03 are recommended:

 - Item 02: Frame Type = 0 (D4)
 - Item 03: Zero Suppression = 1 (AMI/ZCS)
 - Item 04: Distance Between ETU and CSU = 0 (0~133')
 - Item 05: Clock Select = 1 (Internal)
 - Item 06: DTI No. of Ports= 0 (Auto/24), 1 (4 Ports), 2 (8 Ports), 3 (12 ports), 4 (16 ports), and 5 (20 Ports)

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**

For ANI/DNIS, reserve at least one DTMF receiver for DTMF reception (entry 0 or 2). Use the following as a guide when allocating DTMF receivers:

 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.

- ➔ **10-39-01 Fractional Setup (Version 1500 or higher required)**

Enable (1)/Disable (0) T1/PRI fractional function.
(default: 0)

- ➔ **14-01-02 Basic Trunk Data Setup – Transmit Level**
- 14-01-03 Basic Trunk Data Setup – Receive Level**

Assign a CODEC Gain Type to each trunk. This sets the amount of gain (amplification) for the selected trunk.
[default:32 (0dB)]

- ➔ **14-01-07 Basic Trunk Data Setup – Outgoing Calls**

For ANI/DNIS, the following additional setting is recommended: Outgoing Calls = 1 (Allowed)

- ➔ **14-02-01 Analog Trunk Data Setup – Signaling Type (DP/DTMF)**

For ANI/DNIS, the following additional settings in Program 14-02 are recommended:

 - Item 1: Signaling Type (DP/DTMF) = 2 (DTMF)
 - Item 2: Ring Detect Type = 1 (Immediate)
 - Item 3: Flash Type = 0 (Open Loop Flash)
 - Item 4: Hooking Type for Timed Flash (0) or Disconnect (1) = 0
 - Item 5: Dial Tone Detection for Manually Accessed Trunks = 1 (Dial Tone Detection Used)

- ➔ **14-02-02 Analog Trunk Data Setup – Ring Detect Type**

Set the trunks for Normal/delayed (0) or Immediate Ringing (1). For T1 loop/ground start trunks, this option must be set to 1 for the trunks to ring and light correctly.

- ➔ **14-04-01 Behind PBX Setup – Type of Connection**

For ANI/DNIS, the following additional setting is recommended: 0 (Stand Alone).

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- ➔ **14-05-01 Trunk Group – Trunk Group Number**
For ANI/DNIS, place all your ANI/DNIS trunks in Trunk Groups as required.
(default: Trunk ports 1~200 to Group 1 and Priority 1~200)
 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
 - ➔ **20-09-02 Class of Service Options (Incoming Call Service) – Caller ID Display**
Turn off (0) or on (1) the Caller ID display at an extension.
(default: 0).
 - ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
Enable (1)/Disable (0) an extension ability to receive a second call from a DID, DISA, DIL, or Tie Line caller. With this option set to 1, the destination extension must be busy for a second DNIS caller to ring through. If the destination extension does not have a line or Call Appearance (CAP) Key available for the second call and a previous call is ringing the extension but has not yet been answered, the second caller hears busy regardless of this program setting.
(default: 1)
 - ➔ **20-14-01 through 20-14-12 Class of Service Options for DISA/E&M**
Enable/disable Tie Line Class of Service options. Assign Tie Line Classes of Service in PRG 34-02. Up to 15 DISA/E&M Classes of Service can be defined.
(default: 0, 1, 1, 0, 1, 1, 1, 0, 0, 0, 0, 1)
 - ➔ **22-02-01 Incoming Call Trunk Setup – Incoming Type**
For each T1 trunk, set the Trunk Service Type to match the Telco's connected T1 service. For each T1 trunk that should support ANI/DNIS service, enter 7. (ANI/DNIS trunks must be immediate start or wink start T1 trunks with E&M signaling.) For T1 loop/ground start trunks defined as 0, Program 14-02-02 must be set to 1 for the trunks to ring and light correctly.
(default: 0)
 - ➔ **22-13-01 DID Trunk Group to Translation Table Assignment**
Assign DID translation tables to trunk groups.
DID Translation Tables: 1~20
Trunk Groups: 1~100
(default: 1)
 - ➔ **34-01-01 E&M Tie Line Basic Setup – DID/E&M Start Signaling**
For each ANI/DNIS trunk, set the start signaling mode to 1 (wink start).
 - ➔ **34-01-03 E&M Tie Line Basic Setup – E&M Dial-In Mode**
Determine if the incoming Tie Line call should be directed as an intercom call (0) or if it should follow the DID Translation Table in Program 22-11 (1).
(default: 0)
 - ➔ **34-01-04 E&M Tie Line Basic Setup – E&M Line Dial Tone**
Determine whether the Tie Line should send dial tone to the calling system once the call is set up (1) or not (0).
(default: 1)
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- ➔ **34-02-01 E&M Tie Line Class of Service**
Assign a Class of Service (1~15) to a Tie Line. The Class of Service options are defined in Program 20-14. For each Tie Line, you make a separate entry for each Night Service mode. (default: 1)

 - ➔ **34-09-01 ANI/DNIS Service Options – Receive Format**
Specify the format of the ANI/DNIS data received from the Telco. Make sure your entry is compatible with the service the Telco provides (4 = *ANI*DNIS* [* = Delimiter Code]). (default: 0)

 - ➔ **34-09-02 ANI/DNIS Service Options – Delimiter Dial Code**
Define the character Telco uses as a delimiter (see entries 1~5 in PRG 34-09-01). Valid entries are: 0~9, #, and *. (default: * 5)

 - ➔ **34-09-03 ANI/DNIS Service Options – Route Setup of Receive Dial**
Specify the source of the data the system uses to route incoming ANI/DNIS calls. If option 2 is selected, refer to Program 34-09-04:
0 = Fixed Route (Item 08)
1 = Routes on Received DNIS or Address Data
2 = Routes on Received ANI Data
(default: 0)

 - ➔ **34-09-04 ANI/DNIS Service Options – Route Table Setup of Target Dial**
Set how the system uses the route data (gathered in Item 3) to route incoming ANI/DNIS calls (0 = ABB Table [Program 13-04], 1 = DID Table [Program 22-11]).
If option 2 is selected and the call is to be routed using the DID table (1), up to eight digits can be matched. The number of expected digits set in Program 22-09-01 must match the ANI digits defined in Program 22-11-01. For example, if an ANI/DNIS number received was *2035551234*3001* and Program 22-09-01=4, then the entry in Program 22-11-01 must be 1234 with the defined target extension.
If the call is to be routed using the ABB table (0), up to 24 digits can be matched. Define the range of the ABB table to be used in Program 34-09-06. The data is then compared to the entries in Program 13-04-01 and then routed according to Program 13-04-03. (default: 0)

 - ➔ **34-09-05 ANI/DNIS Service Options – ANI/DNIS Display as Target Dial Name**
Set if ANI data should appear on telephone displays as part of Caller ID display.
Caller ID Display:
0 = Display Off
1 = Display On
(default: 1)

 - ➔ **34-09-06 ANI/DNIS Service Options – Routing SPD Table Setup**
Define which part of the SPD Dial Table set up in Program 13-04 the system uses for ANI/DNIS Caller ID look-ups and ANI/DNIS routing (Start = 0, 100~1900, End = 0, 99~1999). This is required if Items 4 and 5 above are 1 (Caller ID on). When you specify a starting and end address, the system uses the part of the table for look-ups. When you specify a starting address and length, the system uses that part of the table for routing. If the incoming ANI/DNIS number data matches the Number entry in the table, the system routes according to the associated Name data. That data can be an extension, Department Group pilot number, the voice mail master number or a trunk ring group.
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- ➔ **34-09-07 ANI/DNIS Service Options – Routing on ANI/DNIS Error**
Determine how the system handles an ANI/DNIS call if a data error is detected in the incoming data string (0 = Play busy tone to caller, 1 = Route the caller to the ring group specified in Program 25-03).
(default: 1)
 - ➔ **34-09-08 ANI/DNIS Service Options – Routing When Destination Busy or No Answer**
Determine how the system handles an ANI/DNIS call if destination is busy or does not answer (0 = Play busy or ringback tone to caller, 1 = Route the caller to the ring group specified in Program 25-04).
(default: 0)
 - ➔ **34-09-09 ANI/DNIS Service Options – Calling Number Address Length**
When Item 1=0 (ANI/DNIS receive format is the address), use this option to specify the address length. The choices are from one to eight digits.
(default: 7)
 - ➔ **34-10-01 Digits Delete for T1 ANI Assignment**
Define the number of digits (0~9) to delete from the information element received from Telco.
Requires Version 1.10 or higher CPUII Software.
(default: 2)

Operation

Refer to the operation for the following features:

- Central Office Calls, Answering
- Central Office Calls, Placing
- Direct Inward Dialing (DID)
- Tie Lines

Tandem Ringing

Description

Tandem Ringing allows an extension user to have two telephones with one telephone number. For example, extension 105 (the master telephone) sets Tandem Ringing with extension 106. When extension 105 receives an incoming call, both extensions 105 and 106 ring. Callers would dial the master extension number (extension 105 in this example). When either the master telephone or slave telephone is in use, the other telephone cannot be used for outgoing calls or incoming calls.

The multiline terminal must be paired with either a single line telephone or a Wireless – DECT handset. It cannot be paired with another multiline terminal.

A single line telephone must be paired with another single line telephone or a Wireless – DECT handset. It cannot be paired with a multiline telephone.

Conditions

- Tandem Ringing is not supported on an Operator or Attendant Extension.
- The slave telephone cannot call the master telephone.
- Extension numbers up to eight digits can be registered on the Tandem Ringing key. Extension numbers over nine digits cannot be registered.
- If Tandem Ringing is enabled, and one of the extensions is busy, no additional calls can be received or placed from either telephone.
- Tandem Ringing can support up to 128 pairs of Tandem Ringing extensions.
- The extension user which enables Tandem Ringing is the master, while the slave telephone is the extension entered by the user while setting up the feature.
- A slave telephone ignores the settings for DND and follows the master telephone settings instead.
- Voice Call is not supported on a multiline terminal with Tandem Ringing.
- Calls placed on Hold while Tandem Ringing is active, immediately recall if the handset is placed On-hook.
- When an extension leaves a Message Waiting for a Tandem Ringing extension, both extensions indicate the message until one of the extensions returns the call.
- A slave telephone ignores the settings for Ring Groups and follows the master telephone settings instead.
- To transfer calls between the two Tandem Ringing stations, a System Park Orbit should be used.

Default Setting

Disabled

System Availability

Terminals

Master Telephone:
Multiline Terminals or Single Line telephones

Slave Telephone:
Single Line telephones or Wireless – DECT handsets

Required Component(s)

ID using Wireless – DECT handsets (Refer to [Wireless - DECT on page 2-1115.](#))

Related Features

Call Forwarding

Call Forwarding/Do Not Disturb Override

Direct Station Selection

Do Not Disturb

Hold

Intercom

Message Waiting

Ring Groups

Tie Lines

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Tandem Ringing (code 80).
- ➔ **30-03-01 DSS Console Key Assignment**
Assign a DSS function key for Tandem Ringing (code 80).
- ➔ **11-11-41 Service Code Setup (for Setup/Entry Operation) – Tandem Ringing**
Define a service code to be used to set up Tandem Ringing
(Default: No Setting).

Operation

To set up Tandem Ringing:

1. Press **Speaker** at the extension considered to be the master telephone (optional).
2. Press the Tandem Ringing key (PRG 15-07 or SC 751: 80).
3. Dial **1** to set the feature.
4. Enter the extension number to be considered the slave telephone (the telephone that rings when the master extension rings).
A confirmation tone is heard (if **Speaker** was used).
5. Press **Speaker** to hang up (if the key is lit).
While the feature is active, if either the master or slave telephone is on a call, no calls can be placed or answered at the other extension until the busy telephone has hung up. Multiline Terminals indicate "TANDEM IN USE" in the display and single line telephones hear a busy signal when the handset is lifted.

To cancel Tandem Ringing:

1. Press the **Tandem Ringing** key (PRG 15-07 or SC 751: 80).
2. Dial **0** to cancel the feature.

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Tandem Trunking (Unsupervised Conference)

Description

Tandem Trunking allows an extension user to join two outside callers in a Trunk-to-Trunk Conference. The extension user can then drop out of the call, leaving the trunks in an Unsupervised Conference. The extension user that established the conference is not part of the conversation. The conference continues until either outside party hangs up. In addition, the extension user that set up the conference can end the tandem call anytime.

The number of simultaneous conference calls is limited by the number of conference circuits in the system. Due to this fact, the maximum number of conference calls cannot exceed the limits defined below:

The CPUII provides two blocks of 32 conference circuits, allowing each block to have any number of conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block limit of 32.

Tandem Trunking could help an office manager, for example, put two outside sales people in touch. The office manager could:

- Answer a call from one salesperson
- Place a call to the second salesperson
- Set up the Trunk-to-Trunk Conference
- Drop out of the call

The office manager could terminate the conference anytime.

There are four methods for Tandem Trunking:

- Method A - Tandem Trunking from Conference

An extension user can set up Tandem Trunking (Unsupervised Conference) by dialing a 2-digit service code (#8) or a uniquely programmed Transfer key.

- Method B - Tandem Trunking with Transfer Key

This method allows an extension user to easily set up an Unsupervised Conference with a call they have placed on Hold. It uses a uniquely programmed Transfer key to set up a tandem call.

- Method C - Automatic Tandem Trunking on Hang Up

This method allows an extension user to easily set up an Unsupervised Conference without having to place the conference call on Hold. A Class of Service option is available, which allows or denies an extension user from automatically setting up a Conference/Tandem Trunking call by hanging up the telephone.

Method D - Automatic Tandem Trunking Setup to Speed Dial Number

This method allows an extension user to easily set up an Unsupervised Conference with a call they have placed on Hold. A Class of Service option is available, which allows or denies an extension user from automatically setting up a Conference/Tandem Trunking call upon hanging up the telephone.

Trunk Continue/Disconnect Codes Added

The software enhances the forced trunk release option with the Tandem Trunking and DISA features. Users can be provided with the option to use a Continue or Disconnect service code. The Continue service code extends the conversation a programmed length of time. If the user enters the Disconnect service code, the call is disconnected immediately.

EXAMPLE:

The following example indicates how a call is handled with the system programmed as follows:

- Program 14-01-25: 1 (Continued/Discontinued Trunk-to-Trunk Conversation)
 - Program 20-28-01: # (Conversation Continue Code)
 - Program 20-28-02: No Setting (No Conversation Disconnect Code is entered)
 - Program 20-28-03: 180 (Conversation Continue Time)
 - Program 24-02-07: 600 (Only used with Trunk-to-Trunk Transfer Release Warning Tone)
 - Program 24-02-10: 30 (Only used with Disconnect Trunk-to-Trunk)
 - Program 25-07-07: 600 (Long Conversation Warning Tone Timer)
 - Program 25-07-08: 30 (Long Conversation Disconnect)
1. An external call connects to an external number (either by transferring with Tandem Trunking or by DISA caller).
 2. After 10 minutes (Tandem Trunking = Program 24-02-07 or DISA = Program 25-07-07), a warning tone is heard and the user dials "#" (Program 20-28-01) to extend the conversation.
 3. After three minutes (Program 20-28-03), the warning tone is heard again. After 30 seconds (Tandem Trunking = Program 24-02-10 or DISA = Program 25-07-08), the call is disconnected.

Conditions

- Tandem Trunking requires either loop start trunks with disconnect supervision or ground start trunks.
- The maximum number of simultaneous trunk-to-trunk conferences allowed is determined by the Conference feature setup. Refer to the Programming section for this feature.
- The Continue/Disconnect code must be DTMF.

- With an analog trunk, the Continue/Disconnect code may work using DTMF sounds from the opposite side trunk. With an ISDN trunk, Program 14-01-25 must be enabled to detect the Continue/Disconnect code.
- The Continue/Disconnect code is not accepted while dialing a trunk.
- Continue/Disconnect codes do not work if all receivers are busy.
- When used with the Networking feature, both systems must be programmed the same for the Continue/Disconnect codes.
- A trunk can be set up to automatically tandem trunk/forward to an outside telephone number or Speed Dial – System/Group Dialing bin.
- Other programmed options for incoming and outgoing calls can affect how calls are handled. Refer to Central Office Calls, Answering/Central Office Calls, Placing and check or program these options as needed.
- DISA calls also use the same Continue/Disconnect codes.
- After initiating an unsupervised conference, selecting one of the CAP keys or line keys allows you to barge into the conference.
- If the station that barges into an unsupervised conference hangs up, the conference is terminated.
- A Trunk-to-Trunk transfer can be established by the following operation:
 1. While talking to an outside party, press **Hold**.
 2. Access a second outside line and dial the desired number.
 3. Press **Transfer** key to complete the Trunk-to-Trunk transfer.
 - ✎ When the second call is to be transferred to another station (Not Trunk-to-Trunk), the user should press the Hold key at step 3, then dial the desired station and press Transfer to complete the transfer.

Default Setting

Disabled

System Availability

Terminals

Multiline Terminals and Single Line Telephones

Required Component(s)

None

Related Features

Call Forwarding, Off-Premise

Central Office Calls, Answering

Central Office Calls, Placing

Direct Inward System Access (DISA)

Programming

Tandem Trunking Method A – Tandem Trunking from Conference

- **11-12-57 Service Code Setup (for Service Access) – Tandem Trunking**
If the default service code (#8) for Tandem Trunking is not acceptable, change the code as required.
- **14-01-04 Basic Trunk Data Setup – Transmit Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.
- **14-01-05 Basic Trunk Data Setup – Receive Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.
- **14-01-13 Basic Trunk Data Setup – Trunk-to-Trunk Transfer**
For each trunk that should be able to participate in a tandem call, enter 1.
- **15-07-01 Programmable Function Keys**
(Optional) Assign a function key for Transfer (code 06).
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- **20-07-11 Class of Service Options (Administrator Level) – Forced Trunk Disconnect (analog trunk only)**
Enter 1 to enable Forced Trunk Disconnect. This allows the extension to disconnect an Unsupervised Conference in progress.
- **20-11-14 Class of Service Options (Hold/Transfer Service) – Trunk-to-Trunk Transfer Restriction**
Disable (0) the Trunk-to-Trunk Transfer Restriction option.

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- **20-11-21 Class of Service Options (Hold/Transfer Service) – Restriction for Tandem Trunking on Hang Up**
Deny (1) an extension user ability to set up a tandem call automatically when they hang up.
(default: 0)
 - **20-11-22 Class of Service Options (Hold/Transfer Service) – Restricted Unsupervised Conference**
Enable (1)/Disable (0) an extension user ability to initiate an unsupervised conference.
(default: 0)
 - **20-13-08 Class of Service Options (Supplementary Service) – Conference**
Enable (1)/Disable (0) the extension ability to initiate a conference.
(default: 1)
 - **20-13-10 Class of Service Options (Supplementary Service) – Barge-In Monitor**
Set whether an extension should be able to Barge-In to a call and talk (1) or just monitor the call (0).
(default: 0)
 - **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
This timer (0~64800 seconds) starts when a trunk begins talking with another trunk (for example: Trunk-to-Trunk Transfer, outgoing from trunk, Tandem Trunking).
(default: 1800)
 - **24-02-10 System Options for Transfer – Disconnect Trunk-to-Trunk**
This timer (0~64800 seconds) starts after the Trunk-to-Trunk warning tone is heard.
(default: 0)
 - **25-07-07 System Timers for VRS/DISA – Ling Conversation Warning Tone Time**
Determine the time (0~64800 seconds) a DISA caller or a Tandem Trunking conversation can talk before the Long Conversation tone is heard.
(default: 3600)
 - **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
This timer (0~64800 seconds) determines the time the system waits before disconnecting a call after the Long Conversation tone is heard.
(default: 10)

Tandem Trunking Method B – Tandem Trunking with Transfer Key

- **14-01-04 Basic Trunk Data Setup – Transmit Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.
- **14-01-05 Basic Trunk Data Setup – Receive Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.
- **14-01-13 Basic Trunk Data Setup – Trunk-to-Trunk Transfer**
For each trunk, enter 1 to enable loop supervision.
- **15-07-01 Programmable Function Keys**
Assign a function key for Transfer (code 06).

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- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
 - ➔ **20-07-11 Class of Service Options (Administrator Level) – Forced Trunk Disconnect (analog trunk only)**
Enter 1 to enable Forced Trunk Disconnect. This allows the extension to disconnect an Unsupervised Conference in progress (initially set up using Method B).
 - ➔ **20-11-14 Class of Service Options (Hold/Transfer Service) – Trunk-to-Trunk Transfer Restriction**
Disable (0) the Trunk-to-Trunk Transfer Restriction option.
(default: 0)
 - ➔ **20-11-21 Class of Service Options (Hold/Transfer Service) – Restriction for Tandem Trunking on Hang Up**
Deny (1) an extension user ability to set up a tandem call automatically when they hang up.
(default: 0)
 - ➔ **20-13-10 Class of Service Options (Supplementary Service) – Barge-In Monitor**
Set whether an extension should be able to Barge-In to a call and talk (1) or just monitor the call (0).
(default: 0)
 - ➔ **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
This timer (0~64800 seconds) starts when a trunk begins talking with another trunk (for example: Trunk-to-Trunk Transfer, outgoing from trunk, Tandem Trunking).
(default:1800)
 - ➔ **24-02-10 System Options for Transfer – Disconnect Trunk-to-Trunk**
This timer (0~64800 seconds) starts after the Trunk-to-Trunk warning tone is heard .
(default: 0)
 - ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**
Determine the time (0~64800 seconds) a DISA caller or any trunk-to-trunk (such as Tandem Trunking) conversation can talk before the Long Conversation tone is heard.
(default:3600)
 - ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
This timer (0~64800 seconds) determines the time the system waits before disconnecting a DISA or any trunk-to-trunk (such as Tandem Trunking) call after the Long Conversation tone is heard.
(default: 10)

Tandem Trunking Method C – Tandem Trunking on Hang up

- ➔ **14-01-04 Basic Trunk Data Setup – Transmit Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.
- ➔ **14-01-05 Basic Trunk Data Setup – Receive Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.
- ➔ **14-01-13 Basic Trunk Data Setup – Trunk-to-Trunk Transfer**
For each trunk, enter 1 to enable loop supervision.

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-
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
 - ➔ **20-07-11 Class of Service Options (Administrator Level) – Forced Trunk Disconnect (analog trunk only)**
Enter 1 to enable Forced Trunk Disconnect. This allows the extension to disconnect an Unsupervised Conference in progress (initially set up using Method B).
[default: 0 (COS 1~14); 1 (COS 15)]
 - ➔ **20-11-11 Class of Service Options (Hold/Transfer Service) – Auto On-Hook Transfer**
Disable (0)/Enable (1) an extension user ability to transfer when the user hangs up.
(default: 0)
 - ➔ **20-11-14 Class of Service Options (Hold/Transfer Service) – Trunk-to-Trunk Transfer Restriction**
Disable (0) the Trunk-to-Trunk Transfer Restriction option.
(default: 0)
 - ➔ **20-11-21 Class of Service Options (Hold/Transfer Service) – Restriction for Tandem Trunking on Hang Up**
Allow (0) an extension user to set up a tandem call automatically when they hang up.
(default: 0)
 - ➔ **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
This timer (0~64800 seconds). starts when a trunk begins talking with another trunk (for example: Trunk-to-Trunk Transfer, outgoing from trunk, Tandem Trunking)
(default: 1800)
 - ➔ **24-02-10 System Options for Transfer – Disconnect Trunk-to-Trunk**
This timer (0~64800 seconds). starts after the Trunk-to-Trunk warning tone is heard.
 - ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**
Determine the time a DISA caller or any trunk-to-trunk (such as Tandem Trunking) conversation can talk before the Long Conversation tone is heard (0~64800 seconds).
(default: 3600)
 - ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
This timer (0~64800 seconds). determines the time the system waits before disconnecting a a DISA or any trunk-to-trunk (such as Tandem Trunking) call after the Long Conversation tone is heard.
(default: 10)

Tandem Trunking Method D – Tandem Trunking to Speed Dial Number

- ➔ **11-10-06 Service Code Setup (for System Administrator) – Setting the Automatic Transfer for Each Trunk Line**
If the default service code (733) for enabling Automatic Tandem Trunking feature is not acceptable, change the code as required.
- ➔ **11-10-07 Service Code Setup (for System Administrator) – Canceling the Automatic Transfer for Each Trunk Line**
If the default service code (734) for canceling Automatic Tandem Trunking feature is not acceptable, change the code as required.

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- ➔ **11-10-08 Service Code Setup (for System Administrator) – Setting the Destination for Automatic Trunk Transfer**
If the default service code (735) for setting the destination of the Automatic Tandem Trunking feature is not acceptable, change the code as required.

 - ➔ **13-04-01 Speed Dialing Number and Name – Speed Dialing Data**
Enter the number and names for the bins used to hold the Automatic Tandem Trunking destination.
(default: No Setting)

 - ➔ **13-04-02 Speed Dialing Number and Name – Name**
Assign a name to each System Speed Dial bin.
(default: No Setting)

 - ➔ **13-04-03 Speed Dialing Number and Name – Transfer Mode**
Assign the transfer mode for each System Speed Dial bin.

 - ➔ **13-04-04 Speed Dialing Number and Name – Transfer Destination Number**
(default: No Setting)

 - 13-04-05 Speed Dialing Number and Name – Incoming Ring Pattern**
(default: 0)

 - ➔ **14-01-04 Basic Trunk Data Setup – Transmit Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.

 - ➔ **14-01-05 Basic Trunk Data Setup – Receive Gain Level for Conference and Transfer Calls**
Select the CODEC gain type used by the trunk when it is part of an Unsupervised Conference.

 - ➔ **14-01-13 Basic Trunk Data Setup – Trunk-to-Trunk Transfer**
For each trunk, enter 1 to enable loop supervision.

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

 - ➔ **20-07-05 Class of Service Options (Administrator Level) – Set/Cancel Automatic Trunk-to-Trunk Transfer**
Enable (1)/Disable (0) an extension ability to use the Trunk-to-Trunk Forwarding service codes.
(default: 0)

 - ➔ **20-07-11 Class of Service Options (Administrator Level) – Forced Trunk Disconnect (analog trunk only)**
Enter 1 to enable Forced Trunk Disconnect. This allows the extension to disconnect an Unsupervised Conference in progress.

 - ➔ **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
This timer (0~64800 seconds) starts when a trunk begins talking with another trunk (for example: Trunk-to-Trunk Transfer, outgoing from trunk, Tandem Trunking).
(default:1800)

 - ➔ **24-02-10 System Options for Transfer – Disconnect Trunk-to-Trunk**
This timer (0~64800 seconds) starts after the Trunk-to-Trunk warning tone is heard.
(default: 0)
-
-

-
-
- ➔ **24-04-01 Automatic Trunk-to-Trunk Transfer Target Setup**
Assign the Speed Dialing number (0~1999) to be used as the destination for the Trunk-to-Trunk Transfer.
Trunks: 1~200
(default: 1999)
 - ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**
Determine the time (0~64800 seconds) a DISA caller or any trunk-to-trunk (such as Tandem Trunking) conversation can talk before the Long Conversation tone is heard.
(default: 3600)
 - ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
This timer (0~64800 seconds) determines the time the system waits before disconnecting a DISA or any trunk-to-trunk (such as Tandem Trunking) call after the Long Conversation tone is heard.
(default: 10)

Trunk Disconnect Continue/Disconnect Codes

- ➔ **14-01-25 Basic Trunk Data Setup – Continued/Discontinued Trunk-to-Trunk Conversation**
When Program 24-02-10 is set to disconnect a trunk after the defined time, determine whether or not a user should have the ability to use the continue/disconnect code.
0 = Disabled
1 = Enabled
(default: 0)
- ➔ **20-28-01 Trunk to Trunk Conversation – Conversation Continue Code**
When Program 14-01-25 is enabled, determine the 1-digit code the user should dial (0~9, *, #) to extend the conversation length for the time defined in 20-28-03. If the Continue and Disconnect codes are programmed the same (e.g., #), the system follows the Continue operation. Using the Continue code before the warning tone is heard has no effect.
(default: No Setting)
- ➔ **20-28-02 Trunk to Trunk Conversation – Conversation Disconnect Code**
When Program 14-01-25 is enabled, determine the 1-digit code the user should dial (0~9, *, #) to immediately disconnect their call. Using the Disconnect code before the warning tone is heard disconnects the call.
(default: No Setting)
- ➔ **20-28-03 Trunk to Trunk Conversation – Conversation Continue Time**
When Program 14-01-25 is enabled, determine the time (0~64800 seconds) a call is extended when the user dials the Continue code (defined in PRG 20-28-01).
(default: 0)
- ➔ **24-02-07 System Options for Transfer – Trunk-to-Trunk Transfer Release Warning Tone**
This timer (0~64800 seconds) starts when a trunk begins talking with another trunk (for example: Trunk-to-Trunk Transfer/Tandem Trunking). When this timer expires, a warning tone is heard. If Program 24-02-10 is set, the conversation disconnects after that timer expires. This timer is set again when the external digit timer expires. One of the trunks used must be an analog trunk (or leased line).
(default: 1800)

- ➔ **24-02-10 System Options for Transfer – Disconnect Trunk-to-Trunk**
Determine the time a conversation continues (0~64800 seconds) after the timer in Program 24-02-07 expires. If this option is set to 0, the conversation is disconnected immediately. This program has no affect if Program 24-02-07 is set to 0. One of the trunks used must be an analog trunk (or leased line).
(default: 0)

- ➔ **25-07-07 System Timers for VRS/DISA – Long Conversation Warning Tone Time**
Determine the time (0~64800 seconds) a DISA caller can talk before the Long Conversation tone is heard . If Program 25-07-08 is set to 0, the call is disconnected after the time expires. This time is set again when the external digit timer expires.
(default: 3600)

✎ If this option is set to 0, the settings in PRG 24-02-07 and PRG 24-07-10 are followed – not PRG 25-07-07 and PRG 25-07-08.

- ➔ **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
Determine the time the system waits (0~64800 seconds) before disconnecting a DISA call after the Long Conversation tone is heard. This program has no affect if Program 25-07-07 is set to 0.
(default: 10)

- ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
(Default: 0).
Refer to [Table 2-27 Service Tone Setup, Program 80-01-02 on page 2-480](#).

- ➔ **80-01-35 Service Tone Setup – 1 Second Signal Tone (Service Tone 35)**
If required, adjust the warning tone heard by the user when Program 24-02-07 expires.

Operation

Method A – Tandem Trunking from Conference

To set up a Tandem Call:




1. Place or answer first trunk call.
2. Press **Conf**.
3. Place or answer second trunk call.
4. To set up the tandem call, press **Conf** twice.
✎ This sets up a Conference between you and both outside parties.
5. Press **Transfer**.

- OR -




Press **Hold** and dial **#8** or the service code set for Unsupervised Conference/Tandem Trunking in PRG 11-12-57.

✎ The line keys for the trunks blink green as long as the Unsupervised Conference continues.


To end the Tandem Call:

1. Press either flashing **line** key.
 -  *The line keys light steadily (green). You can listen (i.e., monitor) to the call or rejoin the conversation, based on the setting in Program 20-13-10.*
2. Press **Speaker** or hang up.
 -  *If Program 20-13-10 is set to 0, the Conference ends and the line keys go out.*
 -  *If Program 20-13-10 is set to 1, to manually disconnect the Conference, Forced Trunk Disconnect (i.e., Press the line key + *3 or the service code set of Forced Trunk Disconnect in PRG 11-10-26) must be used by an extension other than the originating extension.*

Method B – Tandem Trunking with Transfer Key**To set up a Tandem Call:**




1. Place or answer first trunk call.
2. Press **Hold** to place the first trunk call on hold.
3. Place a second trunk call.
4. Press **Transfer**.
 -  *This sets up an Unsupervised Conference with both outside parties.*
 -  *The line keys for the trunks light solid red.*
 -  *To disconnect the Conference, use Forced Trunk Disconnect (i.e., Press the line key + *3 or the service code set of Forced Trunk Disconnect in PRG 11-10-26) must be used by an extension other than the originating extension.*

Single Line Telephone**To set up a Tandem Call:**

1. Place or answer first trunk call.
2. Press hookflash and dial **#1**.
3. Place or answer second trunk call.
4. To set up the tandem call, press hookflash and dial **#8**.
5. Hang up.
 -  *This sets up a Conference between both outside parties.*



Method C – Tandem Trunking on Hang up**To set up a Tandem Call:**

1. Place or answer first trunk call.

2. Press **Hold** to place the first trunk call on hold.
3. Place a second trunk call.
4. Hang up.
 -  *This sets up an Unsupervised Conference with both outside parties.*
 -  *The line keys for the trunks light solid red.*
 -  *To disconnect the Conference, use Forced Trunk Disconnect (i.e., Press line key + *3 or the service code set of Forced Trunk Disconnect in PRG 11-10-26).*



Single Line Telephone

To set up a Tandem Call:


1. Place or answer first trunk call.
2. Press hookflash.
3. Place or answer second trunk call.
4. To set up the tandem call, hang up.
 -  *This sets up a Conference between both outside parties.*
 -  *To disconnect the Conference, use Forced Trunk Disconnect [i.e., Dial the trunk access code (#9 + trunk number) + *3 or the service code set of Forced Trunk Disconnect in PRG 11-10-26].*

Method D – Automatic Tandem Trunking Using Speed Dialing

To set Automatic Tandem Trunking:

1. Dial service code **733** (or the service code set for Set Automatic Transfer per Trunk).
2. Dial the desired trunk number (Trunk Number: 001~200).
3. Hang up.
 -  *The line key for the trunk is solid red as long as the Unsupervised Conference continues.*
 -  *To disconnect the Conference, use Forced Trunk Disconnect (i.e., Press the line key or #9 plus the trunk number + *3).*

To cancel Automatic Tandem Trunking:

1. Dial service code **734** (or the service code set for Disable Automatic Transfer per Trunk).
2. Dial the desired trunk number (Trunk Number: 001~200).
3. Hang up.
 -  *To disconnect the Conference, Forced Trunk Disconnect (i.e., Press the line key or #9 plus the trunk number + *3 or the service code set of Forced Trunk Disconnect in PRG 11-10-26).*

To set and change the destination of the Automatic Tandem Trunk call:

1. Dial service code **735** (or the service code set for Set Destination for Automatic Trunk-to-Trunk Transfer).
2. Dial the desired trunk number (Trunk Number: 001~200).
3. Dial the destination Number (trunk access code is not needed).
4. Dial the desired time mode (Time Mode: 1~8).
5. Press **Hold**.
6. Hang up.
 - ☞ To disconnect the Conference, use *Forced Trunk Disconnect* (i.e., Press the line key or **#9** plus the trunk number + ***3**).

Continue/Disconnect Code**To use the Continue code to extend a Tandem Trunk call:**

1. An external call connects to an external number either by transferring with Tandem Trunking or by DISA caller.
2. After the programmed time (Program 24-02-07), a warning tone is heard and the user dials the Continue code (Program 20-28-01) to extend the conversation.
3. After the programmed time (Program 20-28-03), the warning tone is heard again. After the programmed time (Program 24-02-10), the call is disconnected.

TAPI Compatibility

Enhancements

Version 1600 provides new 1st party TAPI drivers for CTA or CTU units.

In versions prior to Version 1600, the 1st party TAPI drivers only supported single line mode. With Version 1600, the CTA or CTU drivers can be configured for single line mode or multiline mode. Multiline mode allows 3rd party applications that support multiline mode to also screen pop on calls to virtual extensions on the station.

Description

The system has Telephony Applications Programming Interface (TAPI) capability. TAPI capability provides:

- Reduced TAPI Feature Set (see the Supported TAPI Commands chart below).
- Caller ID data to the PC for data base lookups and screen pops (see the Caller ID Data chart below).
- Telephone control (Off-Hook, On-Hook and dialing).

The CTA or CTU unit provides an interface that allows the user personalized control of the telephone system from a desktop or laptop PC when used in conjunction with a TAPI-compliant application. The telephone system and PC are connected by installing an adapter on the telephone multiline terminal, allowing the PC user to access sophisticated communications services via the telephone lines.

Conditions

- Caller ID and Call status are available from the TAPI interface functions.
- Refer to [Table 2-39 TAPI Commands](#) for a list of supported TAPI commands.
- The CT(U)-R Unit is supported only on the ESIB(8)-U20 or ESIE(8)-U20 ETU.
- If the CPUII software is at version 1.XX, version 1.0.5 of the 1st party TAPI drivers must be used. If the CPUII software is at version 2.XX, version 2.0.5 of the 1st party TAPI drivers must be used. **Version 2.0.5 drivers are not compatible with V1.XX CPUII software.**

Table 2-39 TAPI Commands

lineAddProvider	lineAnswer
lineConfigDialog	lineBlindTransfer
lineConfigProvider	lineCompleteCall
lineGetAddressCaps	lineCompleteTransfer

Table 2-39 TAPI Commands (Continued)

lineGetDevCaps	lineDevSpecific
lineGetDevConfig	lineDial
lineGetIcon	lineDrop
lineGetID	lineForward
lineInitializeEx	lineHold
lineNegotiateAPIVersion	lineMakeCall
lineNegotiateExtVersion	linePark
lineRemoveProvider	linePickup
lineSetDevConfig	linePrepareAddToConference
lineShutdown	lineRedirect
lineClose	lineReleaseUserUserInfo
lineDeallocateCall	lineRemoveFromConference
lineGetAddressID	lineSecureCall
lineGetAddressStatus	lineSendUserUserInfo
lineGetCallInfo	lineSetupConference
lineGetCallStatus	lineSetupTransfer
lineGetLineDevStatus	lineSwapHold
lineOpen	lineUncompleteCall
lineSetAppSpecific	lineUnhold
lineSetCallData	lineUnpark
lineSetCallTreatment	lineGatherDigits
lineSetLineDevStatus	lineGenerateDigits
lineSetMediaMode	lineGenerateTone
lineSetStatusMessages	lineMonitorDigits
lineAddToConference	

Default Setting

Disabled

System Availability

Terminals

- Electra Elite IPK II multiline terminal containing an RS-232-C CTA Unit/CTU Unit with TAPI compliant firmware

Required Component(s)

- CPUII()-U10 ETU should have LAN connection capability
- Compatible system software version
- PC Driver for the CTA: PC running Windows 2000 Professional or higher
- PC Driver for the CTU: PC running Windows 2000 Professional or higher
- A TAPI compatible Windows application
- ESIB(8)-U20/ESIE(8)-U20 ETU

Related Features

Data Communications

Headset Operation



Programming

- ➔ **15-02-19 Multiline Telephone Basic Data Setup – CTA Data Communications Mode**
Set the extension to use TAPI to 0 for PC connection.

Operation



TAPI operation is automatic once programmed in the telephone system and enabled in the PC TAPI application, unless a headset is used.

Using the Headset with Automatic Answer:

1. With the multiline terminal in an idle state, press the **Check** key.
2. Press the **Headset** key (PRG 15-07 or SC 751: 05) twice.
3. Press the **Clear** key to return the display to idle.
 -  *The Headset key blinks when Automatic Headset is activated.*
 -  *To cancel Automatic Headset, repeat these steps.*

To redirect calls to the headset and disable the hookswitch (required for some TAPI features):

1. With the multiline terminal idle, press the **Check** key.

2. Press the **Headset** key (PRG 15-07 or SC 751: 05) twice.
3. Press the **Clear** key to return the display to idle.
 -  *The Headset key blinks when Automatic Headset is activated.*
 -  *To cancel Automatic Headset, repeat these steps.*
4. Press the **Headset** key (PRG 15-07 or SC 751: 05) to go Off-Hook.

Tone Override

Description

The multiline terminal user that calls a busy station and receives a call waiting tone can generate a Tone Override that is heard by the originator and busy station. The busy station user can place the existing call on hold to answer the Override.

Conditions

- One Tone Override at a time can be received at a multiline terminal.
- Tone Override can be accomplished only after receiving a BUSY tone.
- Tone Override originate is allowed from a single line telephone until the PBR times out.
- Virtual Extensions do not support Tone Override.

Default Setting

None

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Call Waiting/Camp-On

Data Line Security


Programming

- ➔ **11-12-03 Service Code Setup (for Service Access) – Override (Off-Hook Signaling)**
Assign the service code used for Off-Hook Signaling Override.
(default: 709)
- ➔ **11-16-04 Single Digit Service Code Setup – Intercom Off-Hook Signaling**
Assign a one-digit service code used for Off-Hook Signaling.
(default: *)
- ➔ **15-02-12 Multiline Telephone Basic Data Setup – Off-Hook Ringing**
For each extension, set Off-Hook Ringing type: 0 (Muted), 1 (None), 3 (Beep in Speaker), 4 (Beep in Handset), 5 (Beep in Handset and Speaker).
(default: 5)
- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Off-Hook Signaling (code 33).
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- ➔ **20-09-01 Class of Service Options (Incoming Call Service) – Second Call for DID/DISA/DIL/E&M Override**
Enable (1)/Disable (0) an extension ability to receive a second call from a DID, DISA, DIL, or Tie Line caller.
(default: 1)
- ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
For each extension Disable (0)/Enable (1) the extension user ability to have multiple calls queued at the extension. If Allowed, Tone Override is automatic. Refer to PRG 20-13-06.
(default: 1)
- ➔ **20-13-05 Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling**
Enable (1)/Disable (0) the ability to receive Off-Hook Signaling.
- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Enable (1)/Disable (0) the ability to automatically send Off-Hook Signals while busy on a handset call. Used when PRG 20-09-07 is set to disable (0).
(default: 0)
- ➔ **20-13-34 Class of Service Options (Supplementary Service) – Block Manual Off-Hook Signaling**
Turn off (0) or on (1) an extension ability to block Off-Hook signals manually sent from a co-worker.
(default: 0)
- ➔ **20-13-35 Class of Service Options (Supplementary Service) – Block Camp On**
Enable (1)/Disable (0) the extension user ability to block callers from dialing (default #) to Camp On.
(default: 0)

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- ➔ **20-18-06 Service Tone Timers – Interval of Call Waiting Tone**
Use this timer (0~64800 seconds) to set the time between Off-Hook Signaling alerts.
(default: 10)
 - ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
(Default: 0).
Refer to [Table 2-27 Service Tone Setup, Program 80-01-02 on page 2-480](#) (Service Tone #1).
 - ➔ **80-01-01 Service Tone Setup – Ring Busy Tone**
Define Ring Busy Tone.
Refer to [Table 2-26 Service Tone Setup, Program 80-01-01 on page 2-476](#) (Service Tone #39).

Operation

To send Off-Hook signals to an extension busy on a call:

 *Your extension may send Off-Hook signals automatically.*

1. Dial * (PRG11-16-04).


- OR -

Dial **709** (PRG 11-12-02).

- OR -

Press the **Off-Hook Signaling** key (PRG 15-07-01 code 33).

 *You hear Ring Busy Tone.*

 *The called extension hears Call Alert Notification.*

To answer Tone Override:

1. Receive Tone Override.
2. Press **Hold** and talk with the party.

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Traffic Reports

Description

The system provides the ability to send data to a PC connected to the Electra Elite IPK II. The telephone call traffic data for each extension is captured for use with the Station Message Detail Recording (SMDR) feature.

Call Traffic

The total of outgoing call frequency, outgoing call duration, incoming call frequency, answer frequency, incoming call duration, ringing duration for each line and extension, and abandon call frequency for each line is logged. The total of incoming calls, answer frequency, call duration for each line and extension, and abandon call frequency of each line is logged and the data is outputted to the PC. The system totals the hour, day, week, and month for each terminal and trunk number. This information is used by the SMDR feature. The extension which is totaled is determined by system programming. The system outputs this data to the PC for the total period.

Conditions

- The SMDR call buffer stores 500 calls. The buffer stores calls when the SMDR device is unavailable. When the buffer fills, the oldest record is deleted to allow the new record to be saved.
- If connected to the output device, the reports print hourly. If not connected and the data is not outputted at the end of the hour, the traffic data is overwritten by new incoming data.
- The traffic data is lost if a power failure occurs.
- Traffic Reports require connection to the serial connector on the CPU11()-U10 ETU. Additional programming and a customer-provided printer are also required. Refer to the Electra Elite IPK II System Hardware Manual for more on setting up and connecting to the Electra Elite IPK II system.
- SMDR provides additional information about the system trunk calling patterns. Refer to [Station Message Detail Recording on page 2-913](#) for more information.

Default Setting

Disabled

System Availability

Terminals

All Terminals

Required Component(s)

Software Licenses for SMDR

Traffic Total Report – Sample Report

Terminal	OTG	Duration	Cost	ICM	Answer	Duration	Ringling	Abandon
301	54	01:45:14	720	326	115	02:11:52	00:09:36	
301	92	02:37:22	1855	84	84	01:58:31	00:04:19	
LINE001	--	--	--	79	71	01:05:26	--	8

Term	Definition
Terminal	Terminal Number/Called Party Number (maximum 24 digits)
OTG	Outgoing Call Frequency/number of outgoing calls (maximum 65535 calls)
Duration	Call Duration for an Outgoing Call
Cost	Call Charge (Not Used)
ICM	Incoming Call Frequency/number of incoming calls (maximum 65535 calls)
Answer	Answer Frequency (maximum 65535 calls)
Duration	Call Duration for an Incoming Call
Ringling	Ringling Duration
Abandon	Number of Abandoned Calls (maximum 65535 calls)

Related Features

Station Message Detail Recording

Programming

- **90-20-01 Traffic Report Data Setup – Call Traffic Output**
Determine whether or not the Call Traffic Output should be measured.
0 = No
1 = Yes
(default: 0)
- 90-20-03 Traffic Report Data Setup – All Line Busy Output**
- 90-20-04 Traffic Report Data Setup – DTMF Receiver Busy Output**
- 90-20-05 Traffic Report Data Setup – Dial Tone Detector Busy Output**
- 90-20-06 Traffic Report Data Setup – Caller ID Receiver Busy Output**
- 90-20-07 Traffic Report Data Setup – Voice Mail Channel All Busy Output**
- 90-20-08 Traffic Report Data Setup – ACD Operator All Busy Output**
- 90-20-09 Traffic Report Data Setup – Attendant Channel All Busy Output**
- 90-20-10 Traffic Report Data Setup – Base Station All Busy Output**

- **90-21-01 Traffic Report Output – Output Port Type**
Define the output port to be used for the traffic reports. The reports print hourly when connected to the output device.
0 = No Setting
1 = CPU II COM Port
2 = CPU II USB Port
3 = CPU II LAN Port
(default: 0)

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Transfer

Enhancements

Version 1500 or higher allows DSS/One-Touch keys to be used for one-touch transfer instead of using the Transfer key.

Description

Transfer permits an extension user to send an active Intercom or outside call to any other extension in the system. With Transfer, any extension user can quickly send a call to the desired co-worker. A call a user transfers automatically recalls if not picked up at the destination extension. This assures that users do not lose or inadvertently abandon their transfers. While a transferred call is ringing an extension the system can optionally play ringback tone or Music on Hold to the caller.

The system allows the following types of transfers:

- Screened Transfer**
The transferring user announces the call to the destination before hanging up.
- Unscreened Transfer**
The transferring party extends the call without an announcement.
- Extension (Department) Groups Transfer**
The Transferring party sends the call to a Department instead of an extension.
- Transfer Without Holding**
A user presses a busy line key or the same (busy) CAP key and waits for the call to complete. The system automatically sends them the call when the internal caller hangs up.

Automatic On-Hook Transfer Operation

With Automatic On-Hook Transfer, a transfer goes through as soon as the transferring user hangs up. For example, extension 104 can answer a trunk, press Transfer, dial 105 and hang up. The system extends the call to extension 105. Without Automatic On-Hook Transfer, the call would stay on Hold at extension 104 when the user hangs up. To extend the call, the user at extension 104 must press the Transfer key again before hanging up.

Each method has advantages. Automatic On-Hook Transfer makes transferring calls easier. However, users have to be more aware of how they handle their calls on Hold. Without Automatic On-Hook Transfer, extending a call becomes a two-step operation – but separate from placing calls on Hold.

Prevent Recall of Transferred Call

The Class of Service program allows you to prevent a Transferred call from recalling the originating extension if the call is not answered.

Transfer Call into Conference/Existing Call

This feature allows either a multiline terminal or single line telephone user with Barge-In ability to transfer a call into an existing call. This call can be a 2-party call, a Conference call, or a Barge-In Conference. The system allows Intercom and trunk calls to be transferred into a Conference call. This allows, for example, an attendant to locate co-workers and then transfer them to an existing telephone meeting. There is no need for the attendant to locate all the parties at the same time and sequentially add them to the Conference.

Transfer to Trunk Ring Group Available

It is possible to transfer a trunk call to the trunk's defined ring group (defined in Program 22-05-01: Incoming Trunk Ring Group Assignment). The trunk then rings the defined extensions for the ring group.

This also allows the transferred call to ring over the External Paging (Program 31-05: Universal Night Answer/Ring Over Page) so that an employee can answer the call from any available telephone.

To enable this feature, the system has a program option, Program 11-15-09: Service Code Setup Administrative (for Special Access) – Transfer to Trunk Ring Group Code (not assigned at default). When a call is transferred using this service code, it is transferred to the ring group destination for that incoming trunk. For example, trunk 2 is in Ring Group 4. When the call is transferred using this service code, the trunk rings all extensions programmed for Ring Group 4 or rings the External Paging Group for Ring Group 4, depending on how the system is programmed.

Program 22-04-01: Extension Ring Group Assignment and Program 22-05-01: Incoming Trunk Ring Group Assignment must be programmed to allow an extension access to the ring groups. If the call is not answered, it can overflow to the destination defined in Program 22-08-01: DIL/IRG No Answer Destination.

This service code also can be used with the VRS. This provides the caller listening to the VRS message with the ability to transfer their call and have it ring the external page. The code the caller dials is defined in Program 25-06-02: VRS/DISA One-Digit Code Attendant Setup.

Transfer Key Can Place Call on Hold

While on a call, press Transfer to place the call on hold.

Conditions

- When an analog device is a member of a Department Group assigned as the VM Department Group in PRG 45-01-01, it cannot perform an unsupervised transfer. When the party where the call is being transferred answers, the transferred party is connected to that party, and the analog device is released.
- An existing call can be transferred to a call with Barge-In enabled.
- Unscreened Transfers from voice mail show pre-answer Caller ID information.
- With Transfer to Busy Extensions enabled (Program 24-02-01 = 1), Call Forwarding with Both Ringing offers a unique option. A transferred call waits for either the forwarding or destination extension to become free. The call goes through to the extension that first becomes available. If neither extension becomes free during the Transfer Recall Time, the call recalls the transferring extension.
- An existing call can be transferred to a conference call.
- Meet Me Paging Transfer allows the user to page a co-worker and have the call automatically transferred when the co-worker answers the page.
- When transferring, an extension user can press a One-Touch key instead of dialing the extension number.
- Serial calls allow transferring a call so it automatically returns to the transferring extension when completed.
- When a multiline terminal user is on a call, they can press a DSS key for that station to transfer to another station. It is not necessary to press Transfer to transfer to another station with a DSS key.
 - ✎ When a multiline terminal user is on a call, they must press Transfer to transfer a call off site with a DSS key.
- The following features require certain tones be changed in PRG 80-01-02. Refer to the table in the [Digital Voice Mail](#) feature programming section for settings:
 - Call Holding
 - Busy Greeting
 - Call Screening
 - Await Answer Transfer
- A Trunk-to-Trunk transfer can be established by the following operation:
 1. While talking to an outside party, press **Hold**.
 2. Access a second outside line and dial the desired number.
 3. Press **Transfer** to complete the Trunk-to-Trunk transfer.
 - ✎ When the second call is to be transferred (Not Trunk-to-Trunk), the user should press Hold at step 3, then dial the desired station and press Transfer to complete the transfer.

- If station A calls Station B, and station A puts station B on hold and then calls station C, station C cannot transfer the call.
- Transferred calls to ACD Queue hear ringback and do not follow PRG 24-02-02.

Default Setting

Enabled

System Availability

Terminals

All Multiline Stations

Required Component(s)

None

Related Features

Barge-In

Call Waiting/Camp-On

Caller ID

Call Forwarding

Conference

Meet Me Paging Transfer

One-Touch Calling

Quick Transfer to Voice Mail

Serial Call

Programming

- **11-12-58 Service Code Setup (for Service Access) – Transfer into Conference**
Assign the code users dial to Transfer a call into a Conference call. This code is normally 624.
- **11-15-09 Service Code Setup Administrative (for Special Access) – Transfer to Incoming Ring Group**
When a call is transferred using this service code, it is transferred to the ring group destination for that incoming trunk. For example, trunk 2 is in Ring Group 4. When the call is transferred using this service code, the trunk rings all extensions programmed for Ring Group 4 or rings the External Paging Group for Ring Group 4, depending on how the system is programmed.
(default: No Setting)
- **15-07-01 Programmable Function Keys**
Extension users may want a function keys programmed for Transfer (code 06).
- **15-02-05 Multiline Telephone Basic Data Setup – Transfer Key Operation Mode**
Set the operating mode of the extension Conf key. The keys can be for Call Transfer (0), Serial Calling (1) or Flash (2). When selecting the Flash option, refer also to Program 81-10-07 and Program 81-10-08.
(default: 0)

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- ➔ **15-02-24 Multiline Telephone Basic Data Setup – Conference Key Mode**
Program an extension Conf key for Conference or for Transfer. When set for Transfer (1), the user places a call on hold, dials the extension to which it should be transferred, then presses the Conf key. The call is then transferred. When set for Conference (0), with an active call, the user presses the Conf key, places a second call, then presses the Conf key twice. All the calls are then connected.
(default: 0)

 - ➔ **20-02-04 System Options for Multiline Telephones – Retrieve the Line After Transfer**
Enable (1)/Disable (0) an extension user ability to answer a call after it has been transferred, but before it is answered.
(default:1)

 - ➔ **20-03-01 System Options for Single Line Telephones – SLT Call Waiting Answer Mode**
For a busy single line (500/2500 type) telephones, set the mode used to answer a camped-on trunk call.
Modes:
0 = Press and release hookswitch to pick up waiting call
1 = Press and release hookswitch and dial Service Code 794 (Call Waiting Answer/Split Answer for SLT) to answer waiting call.
(default: 0)

 - ➔ **24-02-02 System Options for Transfer – MOH or Ringback on Transferred Calls**
Enable (1)/Disable (0) MOH on Transfer. If enabled (0), a transferred caller hears Music on Hold while their call rings the destination extension. If disabled (1), a transferred caller hears ringback while their call rings the destination extension. For this option to work with voice mail, the transferred call must be an unscreened transfer.
(default: 0)

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to extensions.

 - ➔ **20-11-06 Class of Service Options (Hold/Transfer Service) – Unscreened Transfer (Ring Inward Transfer)**
Enable (1)/Disable (0) the ability to use Unscreened Transfer.
(default: 1)

 - ➔ **20-11-07 Class of Service Options (Hold/Transfer Service) – Transfer Without Holding**
Enable (1)/Disable (0) the ability to use Transfer Without Holding.
(default: 0)

 - ➔ **20-11-08 Class of Service Options (Hold/Transfer Service) – Transfer Information Display**
Enable (1)/Disable (0) the incoming Transfer pre-answer display.
(default: 1)

 - ➔ **20-11-11 Class of Service Options (Hold/Transfer Service) – Automatic On-Hook Transfer**
Enable (1)/Disable (0) the ability to use Automatic On-Hook Transfer. If enabled, user must press Hold and dial the extension number to Transfer the call. If disabled, a user must have a Transfer Programmable Function Key. To transfer the call, the user would press Hold, dial the extension number, then press the Transfer Programmable Function key to complete the transfer.
(default: 0)
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- ➔ **20-11-18 Class of Service Options (Hold/Transfer Service) – No Recall**
Allow (0)/Deny (1) unanswered Transferred calls to the COS from recalling the originating extension.
(default: 0)
✎ For example, with this option set to 1 for COS 1, calls transferred by any COS to any extension with a COS of 1, do not recall.

 - ➔ **20-11-20 Class of Service Options (Hold/Transfer Service) – No Callback**
Allow (0)/Deny (1) unanswered Transferred calls from the COS from recalling the originating extension.
(default: 0)
✎ For example, with this option set to 1 for COS 1, any call transferred by an extension with COS 1, does not recall.

 - ➔ **20-13-10 Class of Service Options (Supplementary Service) – Barge-In Monitor**
Enable the Barge-In Speech Mode (0) at the initiating extension (i.e., Barge-In initiator). This is required to transfer a call to a conference.
(default: 0)

 - ➔ **20-13-15 Class of Service Options (Supplementary Service) – Barge-In, Initiate**
Enable (1)/Disable (0) the ability to Barge-In to a call.
(default: 0)

 - ➔ **20-13-16 Class of Service Options (Supplementary Service) – Barge-In, Receive**
Enable (1)/Disable (0) the ability to receive a Barge-In on a call.
(default: 0)

 - ➔ **20-13-17 Class of Service Options (Supplementary Service) – Barge-In Tone/Display (Intrusion Tone)**
Enable (1)/Disable (0) the Barge-In Tone. If disabled, this also turns off the Barge-In display at the called extension. As this option is for extensions and trunks do not have an option for a Barge-In Tone, the tone is sent to the trunks based on the Class of Service of the transferring extension.
(default: 1)

 - ➔ **20-13-32 Class of Service Options (Supplementary Service) – Deny Multiple Barge-Ins**
Enable (1)/Disable (0) an extension user ability to have multiple users Barge-In to their conversation.
(default: 0)

 - ➔ **20-14-11 Class of Service Options For DISA/E&M – DISA/Tie Trunk Barge-In**
Determine if DISA and tie trunk users should dial a service code to Barge-In to a conference call.
0 = Disabled
1 = Enabled
(default: 0)

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- ➔ **20-18-07 Service Tone Timers – Intrusion Tone Repeat Time**
After a user barges in, the system repeats the Barge-In tone after this time (0~64800 seconds). Normally, you should disable this time by entering 0. (This timer also affects any other type of call interruption features, such as Voice Mail Conversation Recording or Voice Over.)
(default: 0)

 - ➔ **21-01-03 System Options for Outgoing Calls – Trunk Interdigit Time (External)**
Program the time (0~64800 seconds) an extension must wait before using the Barge-In feature can be used on a call (this time expires before putting a call in a talk state). This time also affects Voice Over.
(default: 5)

 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign extensions to ring groups.

 - ➔ **22-05-01 Incoming Trunk Ring Group Assignment**
Define a trunk ring group. When transferring a DID or trunk call to the trunk defined ring group, the trunk then rings the defined extensions for the ring group.
(default: 1)

 - ➔ **22-08-01 DIL/IRG No Answer Destination**
Assign the DIL No Answer Ring Group to which an unanswered call should overflow.
0 = No Setting
1~100 = Incoming Groups
102 = In-Skin/External Voice Mail
(default: 1)

 - ➔ **24-02-01 System Options for Transfer – Busy Transfer**
Prevent (0)/Allow (1) extensions to transfer calls to busy extensions. If disabled, calls transferred to busy extensions recall immediately.
(default: 1)

 - ➔ **24-02-03 System Options for Transfer – Delayed Call Forwarding Time**
If activated at an extension, Delayed Call Forwarding occurs after this time (0~64800 seconds). This also sets how long a Transferred call waits at an extension forwarded to Voice Mail before routing to the called extension mailbox.
(default: 10)

 - ➔ **24-02-04 System Options for Transfer – Transfer Recall Time**
Set the Transfer Recall Time (0~64800 seconds). An unanswered transferred call recalls to the extension that initially transferred it after this time. This time also sets how long a transferred call camps-on to a busy extension.
(default: 30)

 - ➔ **24-02-05 System Options for Transfer – Message Wait Ring Interval Time**
After a trunk is unscreen transferred to a busy Department Group, this time (0~64800 seconds) starts. If the call is not answered, it recalls the originator.
(default: 30)

 - ➔ **25-06-01 VRS/DISA One-Digit Code Attendant Setup – Next Attendant Message Number**
Assign the message number (0~48)
[default: 0 (No Setting)]
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- ➔ **25-06-02 VRS/DISA One-Digit Code Attendant Setup – Destination Number**
Define the Destination Number (up to 8 digits).
(default: No Setting)
- ➔ **31-05-01 Universal Night Answer/Ring Over Page**
For each trunk which should ring the external page, set the External Page zone (1~9) to allow ringing (1).
(default: 0)
- ➔ **81-10-07 COI Initial Data Setup – Hookflash Time Selection 1**
Set the flash duration for analog trunk calls (1~255 in 16 ms steps).

Operation


Transferring Trunk Calls

To Transfer a trunk call to a co-worker's extension:


1. At the multiline terminal, press **Transfer**.


- OR -

At 500/2500 single line telephone, hookflash.


 *You hear Transfer dial tone.*


2. Dial the co-worker's extension number.


 *If the extension is busy or does not answer, you can dial another extension number or press the line key to return to the call. In addition, you may be able to hang up and have the call Camp-On.*

 *SLT users can retrieve the call by pressing hookflash. If a call was transferred and the 500/2500 user has hung up the handset, the call can be retrieved by dialing ****** and the extension number to which it was transferred.*

3. Announce the call and press **Transfer** (PRG 15-07 or SC 751: 06) or hang up.

 *If you do not have Automatic On-Hook Transfer, you must press Conf (Program 15-02-24=1) or your Transfer Programmable Function Key to Transfer the call.*

 *If your co-worker does not want the call, press the flashing line key to return to the call.*

 *Single line telephone users can retrieve the call by pressing hookflash. If a call has been transferred and the 500/2500 user has hung up the handset, the call can be retrieved by dialing ****** and the extension number to which it had been transferred.*

 *If you do not want to screen the call, hang up without making an announcement.*

To answer a call transferred to your extension:

1. Lift the handset or press **Speaker** when a co-worker announces the call.

Transferring without Holding

To Transfer without holding (multiline terminal only):

1. Lift the handset.
2. Press busy line or press **Speaker**.
3. When original caller hangs up, you are connected.

Transferring Intercom Calls



To Transfer your Intercom call:

1. At the multiline terminal, press **Hold**.


- OR -


At single line telephone, hookflash.

2. Dial extension to receive your call.

-  *If the extension is busy, does not answer or does not want the call, you can dial another extension number or press the lit line key to return to the call. In addition, you may be able to hang up and have the call Camp-On.*
-  *Single line telephone users can retrieve the call by pressing hookflash. If a call has been transferred and the 500/2500 user has hung up the handset, the call can be retrieved by dialing ** and the extension number to which it had been transferred.*

3. Announce your call and press **Transfer** (PRG 15-07 or SC 751: 06) or hang up.

-  **With Automatic On-Hook Transfer**
When you hang up, the call is automatically transferred.

-  **Without Automatic On-Hook Transfer**
You must press your Transfer Programmable Function Key to Transfer the call.


To Transfer the call unattended, press your Transfer Programmable Function Key and hang up without making an announcement.


Transferring a Call Into a Conference/Existing Call

1. While on a call, press **Transfer** and dial service code **624**.


-  *The display shows Transfer to Conf. ICM Dial.*


2. Enter the extension number of the co-worker currently on a Conference call to which the call should be transferred.

-  *To cancel the transfer, press the flashing line key to retrieve the call.*

-  *If an error tone is heard, Barge-In is not enabled for the extension and the call does not go through. Retrieve the call by pressing the flashing line key or hang up and the call recalls the extension.*

3. The transferred call is incorporated into the conference call.

-  *The callers hear the Barge-In tone if enabled in Program 20-13-17.*


-  *If a call is transferred into a Barge-In Conference (an existing 2-party call into which an extension user has used the Barge-In feature to join), the Conference becomes a regular 4-party Conference call.*

4. Hang up.


Transferring a Call to a Trunk Ring Group

1. While on a call, press **Transfer**.

2. Dial the Transfer to Ring Group service code defined in PRG 11-15-09.

-  *You hear a confirmation tone.*

3. Hang up.

-  *The call is transferred to the trunk's ring group defined in Program 22-05-01 and all assigned extensions in the group (Program 22-04-01) ring or it rings the External Paging, enabling anyone to answer the call.*

Transferring an Intercom or Trunk Call using a DSS/One-Touch Key

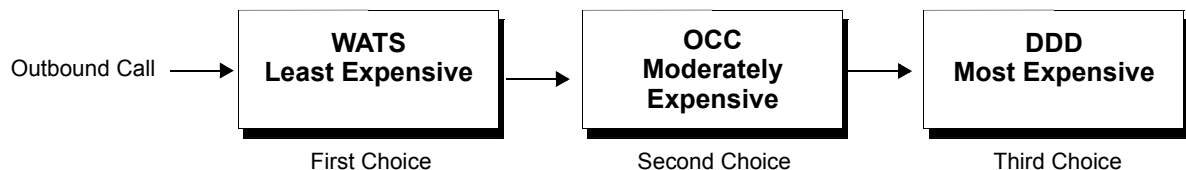
1. While on a call, press the **DSS/One-Touch** key.
2. Announce the call or hang up.

Trunk Group Routing

Description

Trunk Group Routing sets outbound call routing options for users that dial the Trunk Group Routing code (9) for trunk calls. Trunk Group Routing routes calls in the order specified by system programming. If a user dials 9 and all trunks in the first group are busy, the system may route the call to another group. When you are setting up your system, Trunk Group Routing helps you minimize the expense of toll calls. For example, if your system has outbound WATS lines, OCC lines and DDD lines, use Trunk Group Routing to route calls to the WATS lines first.

There are 100 available Trunk Groups and 100 Routes.



Conditions

- DISA (Program 25-10) and Tie Lines (Program 34-03) have separate Trunk Group Routing programs.
- The system uses Trunk Group Routing programming (Program 14-06) when setting up Ringing Line Preference.
- Use trunk group programming to set the order in which users access trunks in a specific trunk group.
- Dialing 9 activates ARS, overriding trunk group routing if ARS service is turned on.

Default Setting

Enabled (All trunks are in Group 1)

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Automatic Route Selection

Central Office Calls, Placing

Direct Inward Dialing (DID)

Dial Tone Detection

Multiple Trunk Types

Programmable Function Keys

Ringing Line Preference

Trunk Groups

Programming

- ➔ **11-01-01 System Numbering**
Set up a Service Code for Alternate Trunk Route Access.
- ➔ **11-09-01 Trunk Access Code**
If required, change the single-digit Trunk Access Code (normally 9). If you change this code, you must also review the settings in PRG 11-01 for the new code selected.
- ➔ **11-09-02 Trunk Access Code – 2nd Trunk Route Access Code**
Assign the Service Code set up in PRG 11-01 for Alternate Trunk Route Access.

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- ➔ **14-05-01 Trunk Group – Trunk Group Number**
Assign trunks to trunk groups.
Trunk Groups: 1~100
(default :All trunks assigned to Trunk Group 1)

 - ➔ **14-06-01 Trunk Group Routing – Priority Order Number**
Set up an outbound routing table for trunk groups assigned in Program 14-05.
Trunk Group Routes: 1~100
Default:
Route 1, Order Number 1 = 1 (Trunk Group 1)
Order Numbers 2~4 = 0 (Not Specified)
All other Routes (2~100) and Order Numbers (1~4) = 0 (Not Specified)

 - ➔ **14-07-01 Trunk Access Map Setup – Trunk Port Number**
Access Map programming may limit Trunk Group Routing options.
Trunk Access Maps: 1~200
Default:
Access Map 1 = Trunk Ports 1~200 assigned with Option 7 (Incoming, Outgoing, and Trunk on Hold access).
Access Maps 2~200, Trunk Ports 1~200 assigned Option 0 (No Access)

 - ➔ **15-06-01 Trunk Access Map for Extensions**
Access Map programming may limit Trunk Group Routing options.
Trunk Access Maps: 1~200

 - ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Trunk Group Routing access (code *02 + trunk group #).
Trunk Groups: 1~100

 - ➔ **21-02-01 Trunk Group Routing for Extensions**
Assign the trunk routes to extensions.
Trunk Group Routes: 1~100
(default: 1)

 - ➔ **21-15-01 Individual Trunk Group Routing for Extensions**
Designate the trunk route accessed when a user dials the Alternate Trunk Route Access Code assigned in Program 11-09-02. Trunk Group Routing is set up in Program 14-06.
Trunk Group Routes: 1~100
[default: 0 (No Setting)]

 - ➔ **23-03-01 Universal Answer/Auto Answer**
Assign trunk routes set in Program 14-06 to extensions for Universal Answer. If the call ringing the paging system is in an extension assigned route, the user can dial the Universal Answer code (#0) to pick up the call. This program also allows an extension user to automatically answer trunk calls ringing other extensions, based on Trunk Group Routing programming (defined in Program 14-06).
Trunk Group Routes: 1~100
(default: 0)


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- ➔ **25-10-01 Trunk Group Routing for DISA**
Assign the Trunk Group Route chosen when a user places a DISA call into the system and dials 9. The Trunk Group Routing is defined in Program 14-06. If the system has ARS, dialing 9 accesses ARS. The route chosen is based on the DISA Class of Service, which is determined by the password the caller dials.
Trunk Group Routes: 1~100
(default: 1)

 - ➔ **25-12-01 Alternate Trunk Group Routing for DISA**
Define the trunk route selected when a DISA caller dials the Alternate Trunk Access Code assigned in Program 11-09-02. The route selected is based on the DISA caller's Class of Service, which is in turn determined by the password the caller dials. Program 14-06 is used to set up the Trunk Group Routing.
Trunk Group Routes: 1~100
(default: 1)

 - ➔ **34-03-01 Trunk Group Routing for E&M Tie Lines**
Assign the Trunk Group Route chosen when a user seizes a Tie Line and dials 9. Set Trunk Group Routing in Program 14-06. If the system has ARS, dialing 9 accesses ARS.
Trunk Group Routes: 1~100
(default: 1)

Operation

To place a call using Trunk Group Routing:

1. At the multiline terminal, press **Speaker**.
- OR -
At single line telephone, lift the handset.
2. Dial **9**.
3. Dial number.
- OR -
 1. At the multiline terminal, press **Trunk Group Routing** key (PRG 15-07 or SC 752: *05).
 Also refer to the [Call Appearance \(CAP\) Keys on page 2-105](#).
 2. Dial the number.

Trunk Groups

Description

Trunk Groups let you optimize trunk usage for incoming and outgoing calls. Each group can be accessed by an Access Code plus the group number. There are 100 available Trunk Groups and you set the access order in trunk group programming. Using Call Appearance (CAP) Keys give an extension user more available function keys, since the user does not need a separate line key for each trunk.

Like Trunk Group Routing, Trunk Groups help you minimize the expense of toll calls. For example, if your system has outbound WATS lines, OCC lines and DDD lines, program the trunk group to route to the WATS lines first.

Priority	Type of Trunk
1	WATS
2	OCC
3	DDD

Conditions

- Unless a user preselects a trunk, Trunk Group programming selects the trunk Speed Dialing used for trunk calls.
- If a user dials a number that is not programmed in ARS, the system can route the call to a trunk group.
- All DID trunks of the same type should be placed in the same trunk group. These trunk groups must then be assigned to a DID Translation Table.
- Trunks ring extensions according to Ring Group programming.

Default Setting

All trunks are in group 1.

System Availability

Terminals

All Stations

Required Component(s)

None

Related Features

Automatic Route Selection

Call Appearance (CAP) Keys

Central Office Calls, Placing

Dial Tone Detection

Direct Inward Dialing (DID)

Programmable Function Keys

Ring Groups

Speed Dial – System/Group/Station

Trunk Group Routing

Programming

- **10-09-01 DTMF and Dial Tone Circuit Setup**
Allocate the circuits on the CPU11 for either DTMF receiving or dial tone detection.
- **11-12-14 Service Code Setup (for Service Access) – Trunk Group Access**
If the service code for Trunk Group Access (704 by default) is not acceptable, change it as necessary.

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- ➔ **14-02-11 Analog Trunk Data Setup – Next Trunk in Rotary if No Dial Tone**
Enable (1)/Disable (0) the system ability to skip over a trunk if dial tone is not detected. This pertains to calls using Call Appearance (CAP) Keys, Speed Dial, ARS, Last Number Redial, or Save Number Dialed. It does not pertain to line keys or Direct Trunk Access calls.
(default: 0)

 - ➔ **14-05-01 Trunk Group – Trunk Group Number**
Assign trunks to trunk groups.
Trunk Groups: 1~200
(default: All trunks Assigned to Trunk Group 1)

 - ➔ **14-06-01 Trunk Group Routing – Priority Order Number**
Set up an outbound routing table for trunk groups assigned in Program 14-05.
Trunk Group Routes: 1~100
Default:
Route 1, Order Number 1 = 1 (Trunk Group 1)
Order Numbers 2~4 = 0 (Not Specified)
All other Routes (2~100) and Order Numbers (1~4) = 0 (Not Specified)

 - ➔ **14-07-01 Trunk Access Map Setup – Trunk Port Number**
Access Map programming may limit Trunk Group Routing options.
Trunk Access Maps: 1~200
Default:
Access Map 1 = Trunk Ports 1~200 assigned with Option 7 (Incoming, Outgoing, and Trunk on Hold access).
Access Maps 2~200, Trunk Ports 1~200 assigned Option 0 (No Access)

 - ➔ **15-06-01 Trunk Access Map for Extensions**
Assign Access Maps to extensions.
Trunk Access Maps: 1~200
(default: 1)

 - ➔ **15-07-01 Programmable Function Keys**
Assign function keys for trunk group access (code *02 + group) or Call Appearance (CAP) Keys (code *08 + CAP Key orbit 0001~9999 (or 0000 for auto assign)).
Trunk Groups: 1~100

 - ➔ **20-02-02 System Options for Multiline Telephones – Trunk Group Access Key Operating Mode**
Set the operating mode of the extension trunk group keys.
Operating Modes:
Incoming and Outgoing Access = 0
Outgoing Access = 1
Incoming Access = 2
(default: 0)

 - ➔ **21-01-02 System Options for Outgoing Calls – Intercom Interdigit Time**
When placing intercom calls, an extension user must dial each digit in this time (0~64800 seconds).
(default: 10)

- ➔ **21-01-05 System Options for Outgoing Calls – Dial Tone Detection Time**
If dial tone detection is enabled, the system waits this time (0~64800 seconds) for the Telco to return dial tone. When the interval expires, the system assumes dial tone is not present. To disable this time (and have the system wait continuously), enter 0.
(default: 3)

Operation

To place a call over a trunk group:

1. At the multiline telephone, press **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **704**.
3. Dial trunk group number (1~9 or 001~100).
4. Dial number.
- OR -
 1. Press the **Trunk Group** key (PRG 15-07 or SC 752: *02 + group).
 2. Dial the number.

To answer an incoming trunk group call:

1. Lift the handset.
2. Press the flashing **Trunk Group** key.

Trunk Queuing/Camp-On

Description

Trunk Queuing permits an extension user to queue (wait in line) on-hook for a busy trunk or trunk group to become free. The system recalls the queued extension as soon as the trunk is available. The user does not have to manually retry the trunk later. Trunk Queuing lets the caller know when the call can go through. If the extension user does not answer the Trunk Queuing ring, the system cancels the queue request.

With Trunk Camp-On, an extension user can queue (wait in line) *Off-Hook* for a busy trunk or trunk group to become free. The caller connects to the trunk when the trunk becomes free. As with Trunk Queuing, the user does not have to manually retry the trunk later.

Any number of extensions may simultaneously queue or Camp-On for the same trunk or trunk group. When a trunk becomes free, the system connects the extensions in the order that the requests were left.

Conditions

- With Automatic Route Selection (ARS), Trunk Queuing automatically queues for the least costly route.
- A user can camp-on or leave a callback request for an extension.
- Other programmed options for outgoing calls can affect how a call is placed. Check or program these options as needed (e.g., access line/Call Appearance (CAP) Keys, etc.).
- Using a Programmable Function Key can simplify the trunk queuing operation.

Default Setting

Enabled

Related Features

Automatic Route Selection

Call Waiting/Camp-On

Callback

Central Office Calls, Placing

Programmable Function Keys

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign a function key for Trunk Queuing and Trunk Camp-On (code 35).
- ➔ **20-01-08 System Options – Trunk Queuing Callback Time**
Set the Trunk Queuing Callback Time (0~64800 seconds). Trunk Queuing Callback rings an extension for this time.
(default: 15)
- ➔ **20-01-09 System Options – Callback/Trunk Queuing Cancel Time**
Set the Callback/Trunk Queuing Cancel Time (0~64800 seconds). The system cancels an extension Callback or Trunk Queuing request after this time.
(default: 64800)
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
(default: Extension 101 Class 15. All other extensions Class 1)
- ➔ **20-11-07 Class of Service Options (Hold/Transfer Service) – Transfer Without Holding**
Disable (0) an extension ability to use Transfer Without Holding.
(default: 0)

Operation

To queue for a busy trunk:

1. Try to access the busy trunk.
2. Dial # or press **Trunk Queuing/Camp-On** key (PRG 15-07 or SC 751: 35).
3. Hang up to leave a Trunk Queuing request.

- OR -

Wait Off-Hook to Camp-On to the trunk.

To answer when Trunk Queuing calls you back:

1. Lift the handset.

To cancel a Trunk Queueing/Camp-On request:

1. At the multiline terminal, press idle **Speaker**.
- OR -
At the single line telephone, lift the handset.
2. Dial **770**.
3. At the multiline terminal, press **Speaker** to hang up.
- OR -
At the single line telephone, hang up.

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Unified Messaging

Description

The CTP()-U10 and VMP-U40 based EliteMail LX Voice Processing systems, using the Electra Elite IPK II system and a Local Area Network, provide Unified Messaging services for voice, fax and e-mail messages with access at either the desktop PC or the telephone. Unified Messaging lets the PC control telephone calls and information about each inbound and outbound call. Both systems include the basic EliteMail LX TeLANophy® Module.

Basic EliteMail LX TeLANophy Module Features

ViewMail® with Live Record Module

All voice and fax messages are visible at a glance on the PC screen and can be sorted in any order. An intuitive Microsoft® Windows interface shows the sender name, subject, and the date and time messages were sent so the user can quickly prioritize them and respond immediately.

ViewCall® Plus

All inbound and outbound calls can be controlled from your PC. Outbound call control requires a TAPI adaptor on the user telephone. By managing calls on the PC instead of the telephone, ViewCall Plus lets you communicate more easily with people inside and outside the office. Three integrated windows are provided to control telephone calls, log all telephone activity, and manage data about each call. With a click of the mouse you can take a call, ask a caller to hold, route the call to another extension, or send the call to voice mail.

Optional EliteMail LX TeLANophy Module Features

ViewFax®

This works in ViewMail to display faxes on screen and lets you send them to any printer. When a fax is received, a fax icon is displayed next to the message in ViewMail. Double click to open the message, and press the play button to listen to any voice annotation sent with the fax. Fax ports are built-in on the CTP()-U10 ETU-based EliteMail LX and are activated as a system option. Up to four Fax ports can be enabled on the EliteMail LX when using the CTP()-U10 ETU.

Hospitality Package

The Hospitality package is used specifically by hotels and resorts to provide guests with personal, accurate, and timely messages. Features include personal greetings, security codes, guest directory, and wake up calls. This feature also supports Property Management System (PMS) integration.

Additional Hospitality Languages

See Multilingual support below for list of supported languages. The Hospitality Package supports one language at default.

Upon purchasing hospitality:

- On VMP-U40, three languages can be unlocked on the system and this is the limit.
- On CTP()-U10 ETU, five are unlocked, additional languages can be purchased (up to the limit of 18).

Text-to-Speech

This converts emails on Exchange-based servers to voice mails.

Networking

This allows the networking of multiple Active Net (AMIS Only) and PlusNet compatible voice mails systems.

Multilingual Support

Add Languages, only United States English is on the drive at default. New languages can be added in the field from the support CD. Additional languages can be added in the field with an upgrade code.

Both systems support one active language at default.

Both systems support up to a maximum of three active system languages.

Supported Languages:

ar = Argentinean	it = Italian
au = Australian English	ja = Japanese (hospitality only)
ca = Catalan Spanish	la = Latin America Spanish
ct = Cantonese Chinese	md = Mandarin Chinese
de = German	nl = Dutch
dk = Danish	nz = New Zealand English
ed = Madrid Spanish	pi = Iberian Portuguese
es = Mexican Spanish	pt = Portuguese
fc = Canadian French	se = Swedish
fr = Parisian French	uk = UK English
he = Hebrew	us = US English

E-Mail Integration

With e-mail integration, subscribers can forward all voice messages to their e-mail inbox automatically and forward all incoming faxes to their e-mail inbox as well. E-mail integration provides users with 24-hour access to e-mail from any touchtone phone. E-mail integration uses standard protocols to access, read and send e-mail messages on the voice messaging system.

Warning: Voice messages forwarded to an e-mail address using e-mail integration are deleted from the user mailbox and the following features are not supported: Pager Notification, Message Waiting Indication.

The following e-mail protocols are supported:

○ IMAP

Internet Message Access Protocol (IMAP) allows the voice messaging system to access an e-mail inbox. Using IMAP, the voice messaging system can obtain e-mail message headers and body information from a variety of e-mail users. This information is then delivered to the text-to-speech engine to convert the text into audio for playback.

○ MIME

Multipurpose Internet Mail Extensions (MIME), ensures that the voice messaging system can read the message header and body information. Multipart MIME messages enable the e-mail system to send enhanced versions of the message for messaging clients such as Lotus Notes or Microsoft Outlook. In addition, multipart MIME messages contain plain text messages that can be read to subscribers over the phone.

Caution

If HTML tags are heard when listening to an e-mail message by phone, the system skips the message. Messages encoded only in HTML are not supported by text-to-speech at this time. Messages containing HTML must be encoded using multipart MIME for text-to-speech to work properly.

○ SMTP

Simple Mail Transport Protocol (SMTP), sends outgoing e-mail messages to e-mail boxes using the voice messaging system. The Forward voice mail to the e-mail system, Forward faxes to the e-mail system, Receive e-mail notification of new fax/voice mail and Reply to e-mail messages via voice mail features use SMTP to send outgoing messages. SMTP can also be configured to restrict the type of messages sent, such as only allowing SMTP mail to be sent to other users on the same domain. Refer to your Exchange, Domino, or GroupWise documentation, or consult your administrator on which settings will work best for your organization.

Conditions

- The following Voice Mail features require changing system tones in PRG 80-01-02 to work. Refer to the programming section for the [Digital Voice Mail](#) feature for details.
- Call Holding
 - Call Screening
 - Busy Greeting
 - Await Answer Transfer

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- The following databases can be migrated to CTP and VMP based EliteMail LX version 2.0.0.5 or higher:
 - OS/2 based EliteMail CTI
 - DOS based EliteMail Q51731 or higher
 - Linux based EliteMail CTI LX
 - Linux based EliteMail CTI LX Lite
 - The following databases can be migrated to CTP based EliteMail CTI LX:
 - OS/2 based EliteMail CTI
 - Linux based EliteMail CTI LX Lite
 - The following databases can be migrated to VMP based EliteMail CTI LX Lite:
 - OS/2 based EliteMail CTI
 - Voice messages forwarded to an e-mail address using e-mail integration are automatically deleted from the user mailbox E-mail integration does not include client applications ViewMail®, VMM, VMG and VML.
 - When voice messages are forwarded to an e-mail address using e-mail integration the following features are not supported: Pager Notification, Message Waiting Indication.
 - Centrex Transfer from voice mail ports requires KSU software version 1.10 or higher.
 - Trunk mapping only works for trunk numbers 1~99. Trunk numbers higher than 99 cannot be trunk mapped.
 - A modem is included for remote maintenance.
 - The EliteMail LX using the CTP()-U10 ETU can be configured for 4, 8, 12, or 16 ports.
 - The EliteMail CTI LX Lite and EliteMail LX using the VMP-U40 can be configured for four or eight ports.
 - The Caller ID display is supported (by name and number, when available) using the telephone, ViewMail, ViewCall Plus, and ViewMail for Microsoft Messaging (VMM).
 - The operating system is Linux.
 - The EliteMail LX using the CTP()-U10 ETU supports up to two Fax ports on 4- and 8-port systems.
 - The EliteMail LX using the CTP()-U10 ETU supports up to four Fax ports on 12- and 16-port systems.
 - The EliteMail CTI LX Lite and EliteMail LX on VMP-U40 do not support the Fax Mail feature.
 - When 5-, 6-, 7 or 8-digit station numbers are used, this feature is not available.

- The OS/2 based EliteMail CTI on the CTI()-U10 board does not support the following features when used in the IPK II:
 - Caller ID Display via telephone.
 - Caller ID Call Return via telephone, ViewMail, ViewCall Plus or ViewMail for Microsoft Messaging.
 - Trunk mapping only works for trunk numbers 1~99. Trunk numbers higher than 99 cannot be trunk mapped.

Optional Feature Package Availability			
Optional Feature	CTI-LX on VMP-U40	CTI-LX on CTP-U10	CTI-LX Lite on VMP-U40
ViewFax™ (Fax Mail)	No	Yes	No
Additional Fax Ports	No	Yes 16-port up to 4 8-port up to 2	No
Additional Voice Ports	4-to 8-port upgrade only	4-port Increments up to 16	4-to 8-port upgrade only
Additional TeLANophy Seats	Yes	Yes	Yes
Hospitality Package	Yes	Yes	No
Text-to-Speech	No	Yes (Up to 4)	No
Additional Systems Languages	Yes (Up to 3)	Yes (Up to 3)	Yes (Up to 3)
Additional Hospitality Languages	No, feature includes the maximum of 3 languages	Yes (Up to 18)	No, feature includes the maximum of 3 languages
Active Net (AMIS Only) and PlusNet	Yes	Yes	No

General Voice Mail Features			
Feature	CTI-LX on VMP-U40	CTI-LX on CTP-U10	CTI-LX Lite on VMP-U40
ViewMail	Yes	Yes	Yes
ViewMail for Outlook (VMM)	Yes	Yes	Yes
ViewMail for Lotus Notes (VML)	Yes	Yes	Yes
ViewMail for Novell GroupWise (VMG)	Yes	Yes	Yes

General Voice Mail Features (Continued)			
Maximum Subscribers	256M CF = 200 512M CF = 250 2G CF = 350 4G CF = 400	4GB = 2000 6GB = 2000	4G CF = 400
Approximate Available Recording Time in Hours	256M CF = 10 512M CF = 20 2G CF = 130 4G CF = 300	4G SSD= 300 6G SSD= 400	4G CF = 300
Maximum Ports	8	16	8
Maximum Number of Active Languages Supported	3	3	3
Live Monitor	Yes	Yes	Yes (via View Call Plus)
Fax-on-Demand	No	No	No
Fax Server	No	No	No
Tape backup hardware or procedures	No	No	No
HTML-based Administration Console	Yes	Yes	Yes
HTML-based Mail Box Manager	Yes	Yes	No
Client-Based Mail Box Manager	No	No	Yes
Softkeys	Yes	Yes	Yes
Live Record using Softkeys	Yes	Yes	Yes
Constant Message Count	Yes	Yes	Yes
Caller ID Display	Yes	Yes	Yes
Caller ID Call Return	Yes	Yes	Yes (via View Call Plus)
Pager Support	Yes	Yes	Yes
FIFO/LIFO (system-wide setting)	Yes	Yes	Yes
Overhead Paging	No	No	No
5-, 6-, or 7-digit station numbers	No	No	No

- Stutter Dial Tone is supported to Single Line Telephones for Voice Mail Message Waiting.

Default Settings

Not Enabled

System Availability

Terminals:

All Terminals

Required Component(s)

EliteMail LX on CTP()-U10 ETU:

- EliteMail CTP()-U10 ETU
- EliteMail LX SSD Drive Kit
- DSP8-U10 Module (for 12-or 16-port installations)
- Desktop PC connected to the Local Area Network (LAN)

EliteMail LX on VMP-U40 ETU:

- EliteMail VMP-U40 ETU
- EliteMail LX CF Drive Kit
- DSP U30 LX Module (for 8-port installations)
- Desktop PC connected to the Local Area Network (LAN)

EliteMail CTI LX Lite:

- EliteMail VMP-U40 ETU
- EliteMail CTI LX Lite Drive Kit
- DSP U30 LX Module (for 8-port installations)
- Desktop PC connected to the Local Area Network (LAN)

Required Software:

When using ViewMail for Microsoft Messaging (VMM) with Office XP/2002 or Office 2000 you must have at least Service Pack 3 for Office installed prior to installing VMM. Failure to do so requires removing and installing the entire Office software suite again. Microsoft Outlook needs Corporate or Workgroup version. When a customer is running the Windows XP operating system, the following versions of Microsoft Outlook work with VMM:

- Outlook 97
- Outlook 98
- Outlook 2000 with Service Pack 3
- Outlook 2002 (XP) with Service Pack 3
- Outlook 2003 with Service Pack 2

The supported TeLANophy applications include:

- ViewMail
- ViewMail for Microsoft Messaging (VMM)
- ViewCall Plus
- ViewFax (EliteMail-LX on CTP()-U10 only)

These TeLANophy applications work on the following operating systems:

- Windows XP
- Window 98
- Windows NT 4.0 with Service Pack 6a
- Windows 2000

Related Features

Central Office Calls, Answering

Central Office Calls, Placing

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**

Assign at least one circuit for DTMF reception (type 0 or 1). Use the following as a guide when allocating DTMF receivers:

 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.

(default: 01~08 = 1, 9~32 = 2, 33~64 = 0)
- ➔ **11-07-01 Department Group Pilot Numbers – Dial**

Assign a Department Group pilot number for the Voice Mail (eight digits maximum). The extensions are assigned to the group in Program 16-02-01.
- ➔ **15-02-26 Multiline Telephone Basic Data Setup – MSG Key Operation Mode**

Determine whether an extension MSG key should function as a Message key (0) or Voice Mail key (1). If set as a Message key, the user can press the key to call the voice mail only when they have new messages. If set as a Voice Mail key, it functions as a normal Voice Mail key (it is not active if Centralized Voice Mail is used).

(default: 0)
- ➔ **15-02-28 Multiline Telephone Basic Data Setup – Message Waiting Lamp Color**

Select whether the Message Wait LED located at the top of the multiline terminal flashes green (0) or red (1) when a Message Wait indication is flashing.

(default: 1)
- ➔ **15-03-01 Single Line Telephone Basic Data Setup – SLT Signaling Type**

For each Electra Elite IPK II voice mail extension, this option must be set to 0.

(default: 0)
- ➔ **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**

Set all SLIU ports used for Voice Mail as type 1.

(default: 0)
- ➔ **15-03-09 Single Line Telephone Basic Data Setup – Caller ID Function - For External Module**

This option *must* be set to 0 when voice mail is used or the integration code for the disconnect function is incorrect.

(default: 0)
- ➔ **15-07-01 Programmable Function Keys**

Assign a Voice Mail key to an extension. You must enter the Voice Mail key code (code 77) followed by:

 - Your own extension number if you are setting up your own Voice Mail key.
 - A virtual extension number if you are setting up a Message Center key for a virtual extension.
 - A co-worker's extension number if you are setting up a Message Center key for an installed extension.
 - An uninstalled extension number if you are setting up a Message Center key for an uninstalled

extension.

- (Optional) Assign a Voice Mail Record key to an extension (code 78).
- (Optional) Assign a Personal Answering Machine Emulation key (code 16).
- (Optional) Use a Call Redirect key (49) to allow a user to transfer a call to another extension or voice mail without answering the call.

- ➔ **16-02-01 Department Group Assignment for Extensions**
Put all the Voice Mail extensions in a unique Department Group (do not use Department Group 1). No other extensions should be in this group. This allows DILs to Voice Mail to ring other Voice Mail extensions when the DILs assigned port is busy. Only one Voice Mail group is allowed per system. Be sure to set up the Department Group in order. (For example, the first extension should be order 1, the second order 2, etc.).
- ➔ **20-02-09 System Options for MultiLine Telephones – Disconnect Supervision**
Enable (1) disconnect supervision for the system.
(default: 1)
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to the voice mail extensions. You should use COS 14 for all time modes.
It is recommended that these options be set to off (0) for the voice mail COS:
 - 20-11-01** Class of Service Options (Hold/Transfer Service) – Call Forward All
 - 20-11-02** Class of Service Options (Hold/Transfer Service) – Call Forward When Busy
 - 20-11-03** Class of Service Options (Hold/Transfer Service) – Call Forwarding When Unanswered
 - 20-11-04** Class of Service Options (Hold/Transfer Service) – Call Forwarding (Both Ringing)
 - 20-11-05** Class of Service Options (Hold/Transfer Service) – Call Forwarding with Follow Me
 - 20-11-12** Class of Service Options (Hold/Transfer Service) – Call Forwarding Off-Premise (External Call forwarding)**It is recommended that these options be set to on (1) for the voice mail COS:**
 - 20-13-01** Class of Service Options (Supplementary Service) – Long Conversation Alarm
 - 20-13-05** Class of Service Options (Supplementary Service) – Intercom Off-Hook Signaling
 - 20-13-06** Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)
 - 20-13-07** Class of Service Options (Supplementary Service) – Message Waiting
 - 20-13-16** Class of Service Options (Supplementary Service) – Barge-In, Receive
 - 20-13-28** Class of Service Options (Supplementary Service) – Allow Class of Service to be Changed
 - 20-13-15** Class of Service Options (Supplementary Service) – Barge-In, Initiate
 - 20-09-02** Class of Service Options (Incoming Call Service) – Caller ID Display
- ➔ **20-13-35 Class of Service Options (Supplementary Service) – Block Camp On**
These options should be set to 0 for the voice mail COS.
(default: 0)
- ➔ **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal On ICM Call**
Enable Continued Dialing (1) for all extensions that dial Voice Mail features.
(default: 1)

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- ➔ **22-01-04 System Options for Incoming Calls – DIL No Answer Recall Time**
A DIL that rings its programmed destination longer than this time (0~64800 seconds) diverts to the DIL No Answer Ring Group set in Program 22-08.
(default: 0)
If setting up Delayed Voice Mail Overflow, enter a time greater than 0. Overflow occurs after this time (provided the other related programming is correct). If setting up Immediate Voice Mail Overflow, enter 0.
 - ➔ **24-02-02 System Options for Transfer – MOH or Ring back on Transferred Calls**
Enable (0)/Disable (1) MOH on Transfer. If enabled (0), a transferred caller hears Music on Hold while their call rings the destination extension. If disabled (1), a transferred caller hears ring back while their call rings the destination extension. For this option to work with voice mail, the transferred call must be an unscreened transfer.
(default: 0)

Assign Trunks As Automated Attendant Trunks – Method 1

- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign Service Type 4 to each trunk you want to ring into Voice Mail as a Direct Inward Line (DIL).
(default: 0)
- ➔ **22-07-01 DIL Assignment**
Assign the master/pilot number of the voice mail group from Program 11-07-01 as the DIL destination. If all Voice Mail ports are in the same unique Extension (Department) Group (see Program 16-02 above), the DIL rings another Voice Mail port if its assigned port is busy.
(default: No Setting)

Assign Trunks As Automated Attendant Trunks – Method 2

- ➔ **22-02-01 Incoming Call Trunk Setup**
Assign Service Type 0 to each trunk you want to ring into Voice Mail as a normal line.
(default: 0)
- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign Ring Group 102 for an In-Skin/External Voice Mail, or 103 for a Central Voice Mail as the destination.
(default: Only extensions 101~108 ring for incoming Ring Group 1 calls)

For Either Method:

- ➔ **11-12-42 Service Code Setup (for Service Access) – Flash on Trunk Lines**
Program the dial access code to use for sending a hook flash to Telco. This code is used for Centrex Transfer using Digital Voice Mail ports. If this code starts with #, PRG 45-01-05 must be set to 0 (Off).
(default: #3)
- ➔ **22-04-01 Incoming Extension Ring Group Assignment**
To enable Voice Mail Overflow, assign selected extensions to a Ring Group that ring for unanswered DILs to Voice Mail ports. In Program 22-06, enter 1 to enable overflow ringing.
(default: Only extensions 101~108 ring for incoming Ring Group 1 calls)

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- ➔ **22-08-01 DIL/IRG No Answer Destination**
For Voice Mail Overflow, enter the Ring Group that unanswered DILs to Voice Mail ring after the DIL Call Waiting time (Program 22-01-04).
(default: 1)

 - ➔ **24-02-03 System Options for Transfer – Delayed Call Forwarding Time**
Set the time (0~64800 seconds) a transferred call waits at a forwarded extension before routing to the called extension mailbox.
(default: 10)

 - ➔ **45-01-01 Voice Mail Integration Options – Voice Mail Department Group Number**
Assign which Extension (Department) Group number is to be assigned as the voice mail group. This program defines the Quick Transfer to Voice Mail destination. An entry of 0 means no voice mail is installed. Department Groups: 0, 1~64
(default: 0)

 - ➔ **45-01-02 Voice Mail Integration Options – Voice Mail Master Name**
Enter the Voice Mail master name up to 12 characters.
(default: Voice Mail)

 - ➔ **45-01-04 Voice Mail Integration Options – Park and Page**
Enable (1)/Disable (0) the system ability to process Voice Mail Park and Page (*) commands.
(default: 1)

 - ➔ **45-01-05 Voice Mail Integration Options – Message Wait**
Enable (1)/Disable (0) the system ability to process Voice Mail Message Wait (#) commands.
(default: 1)
When using Centrex transfer from a voice mail port then the following items must be considered:
 - If the Feature Access Code starts with # in PRG 11-12-42 then set **PRG 45-01-05** to Off.
 - When assigning the dial string in voice mail, one or more Pauses may be needed too, depending on what Telco needs.

 - ➔ **45-01-06 Voice Mail Integration Options – Record Alert Tone Interval Time**
Set the time (0~64800 seconds) between Voice Mail Conversation Record alerts.
(default: 30)

 - ➔ **45-01-14 Voice Mail Integration Options – CCIS Centralized Voice Mail Number**
Assign the pilot number (up to 8 digits) to Centralized Voice Mail over CCIS Link at Remote Sites only.
(default: No Setting)

 - ➔ **45-01-15 Voice Mail Integration Options – Analog Voice Mail Protocol Selection**
(default: 0)

 - ➔ **45-01-16 Voice Mail Integration Options – Voice Mail Fax Digit Add Assignment**
Assign up to 4 digits.
(default: None)

 - ➔ **45-01-17 Voice Mail Integration Options – Reply Mailbox Number**
(default: 1)
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- ➔ **45-01-18 Voice Mail Integration Options – Trunk Number Mapping (Version 2000 or higher required)**
Set if the system sends two digit trunk numbers only or two or three digit trunk numbers to voice mail.
(default: 2)

- ➔ **80-01-02 Service Tone Setup – Basic Tone Number**
The following features require that the system tones listed below be changed to match the table. After changing these settings the KSU must be reset for the changes to take effect.
- Call Screening
 - Call Holding
 - Busy Greeting
 - Await Answer Transfer

Tone 06			
Unit	Basic Tone	Duration	Gain Level
1	11 ~ 480/620Hz -13/-13dB	300ms	32
2	0 - No Tone	300ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

Tone 14			
Unit	Basic Tone	Duration	Gain Level
1	10 ~ 440/480Hz -13/-13dB	1000ms	32
2	0 - No Tone	2100ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

Tone 39			
Unit	Basic Tone	Duration	Gain Level
1	12 ~ 440/620Hz -16dB	500ms	32
2	0 - No Tone	500ms	32
3	0 - No Tone	0ms	
4	0 - No Tone	0ms	
5	0 - No Tone	0ms	
6	0 - No Tone	0ms	
7	0 - No Tone	0ms	
8	0 - No Tone	0ms	

➔ 80-03-01
through
80-03-08 **DTMF Tone Receiver Setup**

➔ 80-04-01
through
80-04-09 **Call Progress Tone Detector Setup**
Set the criteria for DTMF dial, ringback and busy tones.

Operation

Refer to [Digital Voice Mail on page 2-305](#) for complete telephone operation procedures.

Uniform Call Distribution (UCD)

Description

With Uniform Call Distribution (UCD), an extension user can call an idle extension in a preprogrammed UCD Group (Department Group – 64 Department Groups available) by dialing the group pilot number. For example, this would let a caller dial the Sales department just by knowing the Sales department pilot number. The caller would not have to know any of the Sales department extension numbers.

User Log Out/Log In

An extension user can log out and log in to a UCD (Department) group. By logging out, the user removes their extension from the group. Once logged out, UCD (Department Calling) bypasses their extension. When they log back in, UCD (Department Calling) routes to their extension normally. All users can dial a code to log in or log out of their UCD (Department Calling) Group. A multiline terminal can optionally have a function key programmed for one-button log in and log out.

Enhanced Hunting

UCD (Department Calling) is enhanced with expanded hunting abilities. Hunting sets the conditions under which calls to a UCD (Department Group) pilot number cycles through the members of the group. The hunting choices are:

Busy

A call to the pilot number only hunts past a busy group member to the first available extension. A call rings on an unanswered extension until it is answered, or the caller hangs up.

Not Answered

A call to the pilot number cycles through the idle members of a UCD (Department Calling) group. The call continues to cycle until it is answered or the calling party hangs up. However, if the next station in the cycle is busy when a new call comes in, the call queues to the busy agent. New calls do not hunt past a busy agent.

Busy or Not Answered

A call to the pilot number cycles through the idle members of a UCD (Department Calling) group. The call continues to cycle until it is answered or the calling party hangs up.

If all members of the UCD (Department) group are busy, an incoming or transferred call to the group pilot number queues for an available member. Each group has a queue that can hold any number of waiting calls. If a display telephone is waiting in queue, the user sees: *WAITING (group name)*. If a transferred call in queue is an outside call, and the system has a DSP daughter board installed with the VRS compact flash, the queued caller hears, *“Please hold on. All lines are busy. Your call will be answered when a line becomes free.”*

The VRS can also transfer calls to UCD (Department) groups. Refer to the [Voice Response System \(VRS\) on page 2-1089](#) feature for more information on setting up the VRS.

The system prevents hunting to a UCD (Department) group extension if it is:

- Busy on a call
- In Do Not Disturb
- Call Forwarded
- Logged Out

Conditions

- When a DIL rings to a UCD (Department) groups, the DIL may follow overflow programming (Program 22-01-04 and Program 22-08-01).
- If an extension has Call Forwarding set, the system does not hunt to the forwarded extension.

Default Setting

Disabled

Priority Routing

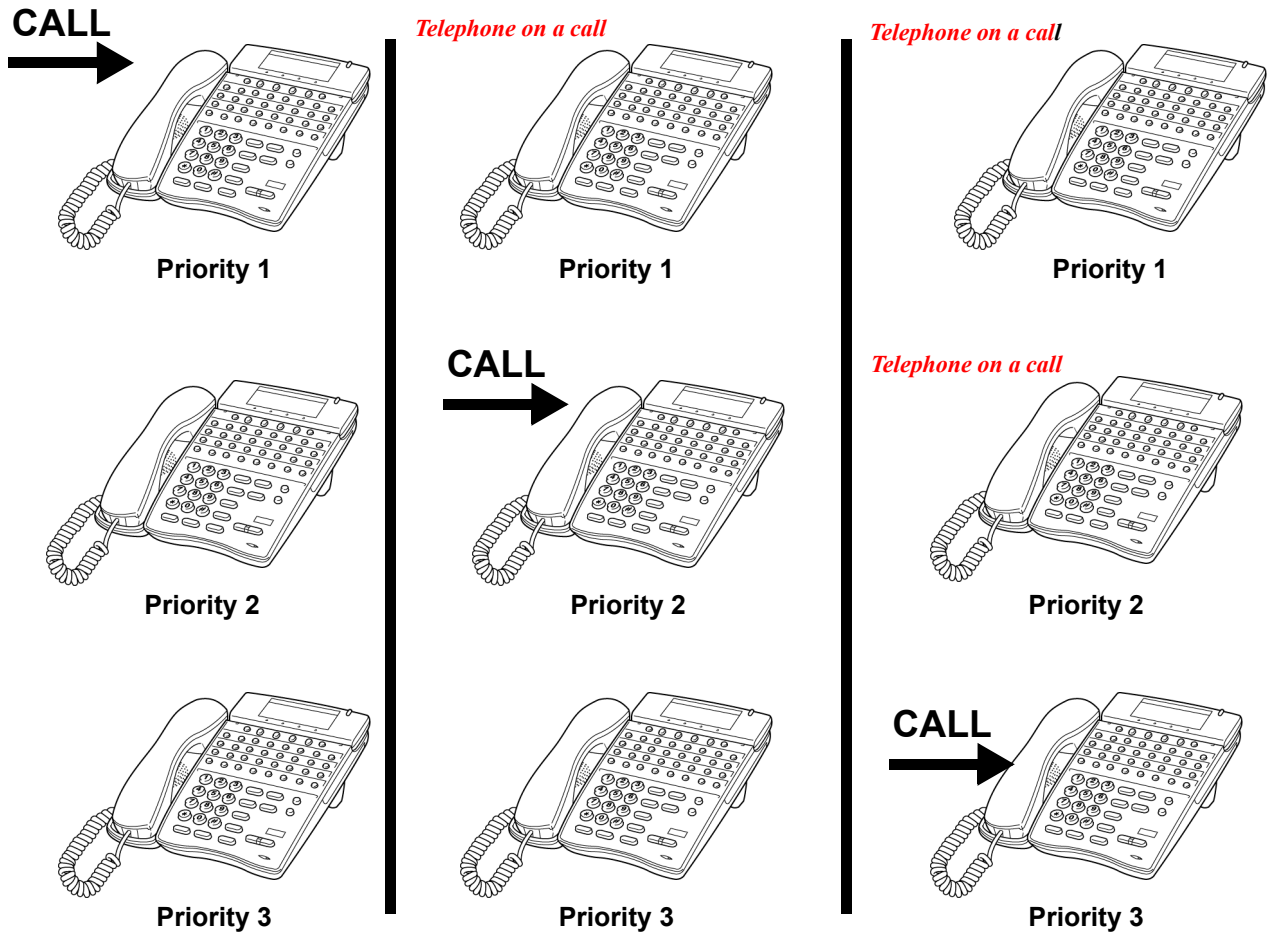


Figure 2-24 Uniform Call Distribution (UCD) Priority Call Routing

Circular Routing

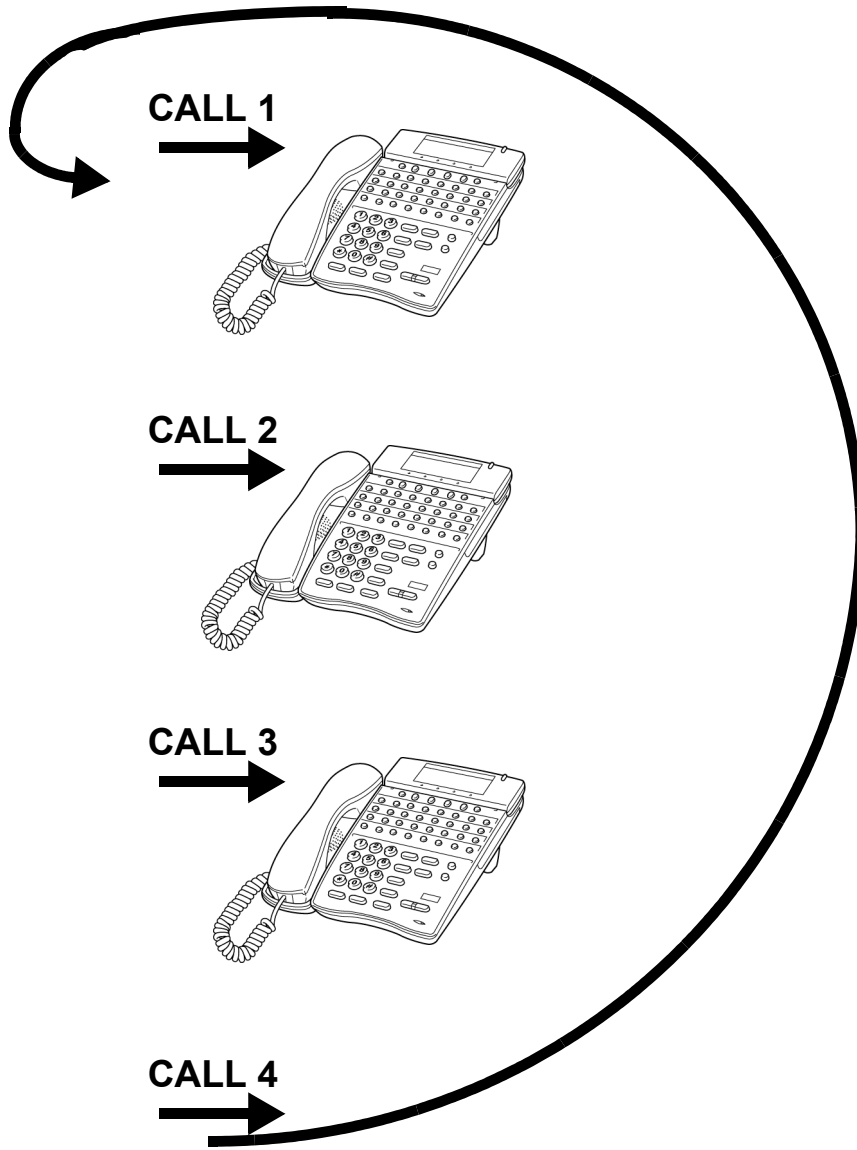


Figure 2-25 Uniform Call Distribution (UCD) Circular Routing

System Availability

Terminals

All Stations

Required Component(s)

DSPII-U10 Unit and VRS Compact Flash (for Delay Announcements)

Related Features

Automatic Call Distribution (ACD)

Call Forwarding

Call Arrival (CAR) Keys

Transfer

Voice Mail Message Indication on Line Keys

Voice Response System (VRS)

Programming

- ➔ **11-07-01 Department Group Pilot Numbers – Dial**
Assign pilot numbers (up to 8 digits) to the Extension (Department) Groups you set up in Program 16-02-01~Program 16-02-10.
(default: No Setting)
- ➔ **15-07-01 Programmable Function Keys**
Assign a Uniform Call Distribution key (46) so extension users can install or remove themselves from the Uniform Call Distribution Group. Additional keys can also be assigned for Department Group features Automatic Transfer (56), immediate calling destination (58), delayed calling destination (59), and DND destination (60).
- ➔ **16-01-01 Department Group Basic Data Setup – Department Name**
Assign a name (up to 12 characters) to the Extension (Department) Groups.
(default: No Setting)

-
-
- ➔ **16-01-02 Department Group Basic Data Setup – Department Calling Cycle**
Set the routing cycle for calls into a department (i.e., when a user dials the department pilot number). The system can ring the highest priority extension available (Priority Routing, 0) or cycle in circular order to a new idle extension for each new call (Circular Routing, 1– set to 1 for UCD).
(default: 0)

 - ➔ **16-01-03 Department Group Basic Data Setup – Department Routing When Busy (Auto Step Call)**
Set how the system routes an Intercom call to a busy Department Group member. The caller can hear busy tone (0) or overflow to the first available Department Group member (1). This option is for Intercom calls to an extension, not a pilot number.
(default: 0)

 - ➔ **16-01-04 Department Group Basic Data Setup – Hunting Mode**
Set whether an unanswered call should hunt once stopping at the last member tried (0) or continually hunt through the idle members (1).
(default: 0)

 - ➔ **16-01-07 Department Group Basic Data Setup – Call Recall Restriction for STG**
Set whether a transferred call to the pilot number should recall (0) or camp-on (1) to the busy or idle Department Group.
(default: 0)

 - ➔ **16-01-09 Department Group Basic Data Setup – Department Hunting No Answer Time**
Set the time (0~64800 seconds) a call rings a Department Group extension before hunting occurs.
(default: 15)

 - ➔ **16-01-10 Department Group Basic Data Setup – Enhanced Hunt Type**
Set the type of hunting for each Department Group:
0 = No Queuing
1 = Hunting When Busy
2 = Hunting When Not Answered
3 = Hunting When Busy or No Answer
(default: 0)

 - ➔ **16-02-01 Department Group Assignment for Extensions**
Assign extensions to Department Groups and set the priority assignment.
Department Groups: 1~64

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.

 - ➔ **20-09-07 Class of Service Options (Incoming Call Service) – Call Queuing**
Allow (1)/Deny (0) an extensions ability to have multiple calls queue to it.
(default: 1)

 - ➔ **20-11-17 Class of Service Options (Hold/Transfer Service) – Department Group Trunk-to-Trunk Transfer (Each Telephone Group Transfer)**
Allow (1)/Prevent (0) an extension user from using the immediate automatic transfer of ICM and transferred calls to the Department Groups feature.
(default: 1)
-
-

-
-
- ➔ **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Allow (1)/Deny (0) automatic Off-Hook Signaling. Set to 0 when PRG 16-01-03 is set to Circular (hunt)
(default: 1)

 - ➔ **20-13-14 Class of Service Options (Supplementary Service) – Department Calling (PLT No Called Extension)**
Allow (1)/Prevent (0) extensions from receiving Department Calls when the pilot number is dialed.
(default: 1)

 - ➔ **22-02-01 Incoming Call Trunk Setup**
If you want a trunk to be a DIL to a Department Group, assign Service Type 4 for each Night Service Mode. Also see PRG 22-07-01.
(default: 0)

 - ➔ **22-07-01 DIL Assignment**
For each trunk assigned Service Type 4 in PRG 22-02 above, assign the DIL destination as the Department Group pilot number (as assigned in Program 11-07-01). (Department: Groups 1~64).
(default: No Setting)

 - ➔ **24-02-05 System Options for Transfer – Message Wait Ring Interval Time**
After a trunk is unscreen transferred to a busy Department Group, this time starts. If the call is not answered, it recalls the originator. Determine the time (0~64800 seconds) the call should ring before recalling.
(default: 30)

 - ➔ **24-02-08 System Options for Transfer – Delayed Transfer Timer for All Department Groups**
Determine the time (0~64800 seconds) a call should ring a Department Group before transferring the call.
(default: 10)

 - ➔ **24-05-01 Department Group Transfer Target Setup**
Assign the Speed Dialing bin to each Department Group to hold the destination for the immediate automatic transfer of ICM and transferred calls to the Department Group feature
(default: 1999).

Operation

To call a UCD Group:

1. At the multiline terminal, press **Speaker**.

- OR -

At single line telephone, lift the handset.

2. Dial the UCD group (department) extension or pilot number.

 *The system routes the call to the first free telephone in the (UCD group) department.*

To log out of your UCD (Department Calling) Group:


 *While you are logged out, UCD (Department Calling) cannot route calls to your extension.*

1. Press **Speaker**.

2. Dial **650** and **1**.

- OR -

Press **Uniform Call Distribution Log In** key (PRG 15-07 or SC 751: 46).

 *The key lights while you are logged out.*

To log back in to your UCD (Department Calling) Group:


 *While you log back in, Uniform Call Distribution routes calls to your extension.*

1. Press **Speaker**.

2. Dial **650** and **0**.

- OR -

Press **UCD (Department Calling) Log In** key (PRG 15-07 or SC 751: 46).

 *The key goes out when you log back in.*

Uniform Numbering Network

Description

Uniform Numbering Network allows multiple or compatible systems to be connected in a network using Tie Lines. A station user can dial a system number and a station number (open numbering) or dial the station number only (closed numbering) to access any station. When the calling and called systems are not directly connected, several Tie Lines may be accessed to route the call. Each system extends the call to the next system until the final destination is reached. Networking provides a seamless connection of multiple systems into a single virtual communications system using Tie Lines with a unified numbering plan. Networking allows many companies to connect their telephone systems so they appear as one. An extension user in the network can easily dial another extension or transfer a call in the Networking System. Calls are passed from network node to network node using a protocol that contains information about the source of the call, the type of call and the destination of the call.

Flexible Network Routing

Use network routes to set up single-channel networking between many separate systems – Or use multiple networking channels per system for greater network performance. Data tables in the system program define the routing for each extension in each network node. These tables are easily customized to meet the requirements of each networking configuration. Users may place an intercom call or transfer a call to any extension at any location by dialing an extension number. The system analyzes each extension number received and determines how to route the call to its final destination. The feature which handles this route selection is called Flexible Routing (F-Routing). F-Routing also can select alternate routes to the destination extension if the primary destination is busy. Up to 120 routes are available for networking. Once an extension number is dialed, the system checks the routing, accesses the assigned trunk group and places the call. Each extension is assigned a route or routes that decide which trunk group to access and any modified dialed data if required.

Conditions

- Monitor the Uniform Numbering Network Access Code plan to avoid loss of Access Codes and to prevent duplicating codes.
- The distant system number can be programmed as 2~8 digits.
- The Electra Elite IPK II system has 500 ARS/F-Route Tables that can be shared by outgoing Tie lines, ISDN CO/PBX, and FT1 lines.
- When a call from/to the remote-end is made to a busy station in the Electra Elite IPK II system, the caller cannot set features such as Callback Message, Step Call, or Camp-On.
- A maximum of 120 Dial Analysis Tables which allows a maximum of 121 connected systems per Uniform Numbering Network.
- DID Full Digit Conversion can access the Uniform Numbering Network.

Default Setting

None

System Availability

Terminals

All Stations

Required Component(s)

TLI(2)-U() ETU or DTI-U() ETU

Related Features

Automatic Route Selection

Flexible System Numbering

K-CCIS – IP with IAD

K-CCIS – IP with PVA

K-CCIS – T1

Tie Lines

Programming

- ➔ **11-01-01 System Numbering**
Set the systems internal (Intercom) numbering plan. The numbering plan assigns the first and second digits dialed and affects the digits an extension user must dial to access other extensions and features, such as service codes and trunk codes. If the default numbering plan does not meet the site requirements, use this program to tailor the system numbering to the site.

Caution

Improperly programming this option can adversely affect system operation. Make sure you thoroughly understand the default numbering plan before proceeding.

Before changing your numbering plan, use PCPro or WebPro to make a backup copy of your system data.

- ➔ **11-02-01 Extension Numbering**
Set the extension number. The extension number can have up to eight digits. The first/second digit(s) of the number should be assigned in Program 11-01-01. This allows an employee to move to a new location (port) and retain the same extension number.
(default: 1~99 = 101~199, 100~256 = 3101~3257)
- ➔ **44-01-01 System Options for ARS/F-Route – ARS/F-Route Time Schedule**
Set this option to 0 so that the F-Route table selected is determined only by the digits dialed without any relation to the day or time of the call.
(default: 0)
- ➔ **44-02-01 Dial Analysis Table for ARS/F-Route Access – Dial**
Set the number of digits (up to eight) to be analyzed by the system for ARS routing.
(default: No Setting)
- ➔ **44-02-02 Dial Analysis Table for ARS/F-Route Access – Service Type**
Select the Service Type (0 = No Setting, 1 = Extension Call, 2 = ARS/F-Route Table, 3 = Dial Extension Analyze Table).
(default: 0)
- ➔ **44-02-03 Dial Analysis Table for ARS/F-Route Access – Additional Data**
Enter the additional data required for the service type selected in Program 44-02-02, either the number of digits to be deleted or the table number to be used.
(default: 0)
- ➔ **44-02-04 Dial Analysis Table for ARS/F-Route Access – Dial Tone Simulation**
If enabled (1), this option sends dial tone to the calling party once the routing is determined. This may be required if the central office at the destination does not send dial tone.
(default: 0)
- ➔ **44-05-01 ARS/F-Route Table – Trunk Group Number**
Select the trunk group number to be used for the outgoing ARS call (0~100).
[default: 0 (No Setting)]

-
-
- ➔ **44-05-02 ARS/F-Route Table – Delete Digits**
Enter the number of digits to be deleted from the dialed number [0~255 (255 = Delete all)].
(default: 0)
 - ➔ **44-05-03 ARS/F-Route Table – Additional Dial Number Table**
Enter the table number (defined in Program 44-06) for additional digits to be dialed (0~1000).
(default: 0)
 - ➔ **44-05-04 ARS/F-Route Table – Beep Tone**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select whether or not a beep is heard if a lower priority trunk group is used (0 = No Beep, 1 = Beep).
(default: 0)
 - ➔ **44-05-05 ARS/F-Route Table – Gain Table Number for Internal Calls**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the gain table number to be used for internal calls (0~500).
(default: 0)
 - ➔ **44-05-06 ARS/F-Route Table – Gain Table Number for Tandem Connections**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the gain table number to be used for tandem connection (0~500).
(default: 0)
 - ➔ **44-05-07 ARS/F-Route Table – ARS Class of Service**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the Class of Service to be used for ARS (0~16). An extensions ARS COS is determined in Program 26-04-01.
(default: 0)
 - ➔ **44-05-08 ARS/F-Route Table – Dial Treatment**
For each ARS/F-Route table (1~500) assign a priority number (1~4). Select the Dial Treatment to be used (0~15). The Dial Treatments are determined in Program 26-03-01.
(default: 0)
 - ➔ **44-05-09 ARS/F-Route – Maximum Digit**
Assign Maximum Digit (0~24).
(default: 0)
 - ➔ **44-05-10 ARS/F-Route Table – CCIS over IP Destination Point Code**
Assign the Point Code (0~16367).
(default: 0)
 - ➔ **44-05-11 ARS/F-Route Table – Network Specified Parameter Table**
Assign Parameter Table (0~16).
(default: 0)
 - ➔ **14-02-09 Analog Trunk Data Setup – Busy Tone Detection**
0 = Disable (default)
1 = Enable
 - ➔ **14-02-14 Analog Trunk Data Setup – Loop Start/Ground Start**
0 = Loop Start (default)
1 = Ground Start
-
-



Operation

None

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Universal Slots

Enhancements

With **Version 1500 or higher**, In-Mail does not take away from the maximum port capacity of a basic system. In-Mail can be assigned to ports greater than 64.

In **Version 1.50**, the DTI/PRI ETUs can be programmed as a 4/8/12/16/20/24-port Fractional T1/PRI (when PRG 10-39-01 is set to be Enabled).

Description

The IPK II has eight universal slots, and up to three cabinets can be installed. The system uses the same KSU for the basic and expansion cabinets to support up to 24 Universal Slots.

Conditions

- Two software packages (Basic Port Package and Expanded Port Package) are available for the IPK II system using the CPU11()-U10 ETU. Refer to [Table 2-40 Maximum Capacities for Basic and Expanded Port Packages](#) for maximum system capacities for each software package.

Table 2-40 Maximum Capacities for Basic and Expanded Port Packages

Item	Basic Port Package	Expanded Port Package
Basic Terminals (Phones)	56*	256
Dedicated CAR/VE	256	256
Basic Terminals + CAR/VE	320*	512
Basic Trunks	56*	200
Basic Terminals + Trunks	64*	416**
Universal Slot	24	24

* 64 maximum Trunk and Station ports combined in the Basic Port Package. The system must have at least one station card and one trunk card installed.

** 16 port slots x 24 Universal Slots = 384 +32 common ports shared by all three cabinets = 416.

- There are 32 common ports in the IPK II that are shared by all three cabinets.
- Each Universal Slot is a 16-port slot.
- The Basic Port Package does not limit the number of slots. It only limits the number of ports.
- The DTI/PRI ETUs can be programmed as a 4/8/12/16/20/24 port Fractional T1/PRI. (**Version 1500 or higher is required** and PRG 10-39-01 must be Enabled).
- AP/ISA Slot is not used in the IPK II.
- If trying to assign an ETU which would exceed the maximum number of ports for the Basic Port Package you do not get an error, but it does not let you program the related programs.
- The available interface cards and maximum capacities for Universal Slots with the Electra Elite IPK II system are shown in [Table 2-41 Maximum System Capacities for Station Interface ETUs](#), [Table 2-42 Maximum System Capacities for Trunk Interface ETUs](#) and [Table 2-43 Maximum System Capacities for Application Interface ETUs](#).

Table 2-41 Maximum System Capacities for Station Interface ETUs

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
CMS(2)/(4)-U() ETU	2- or 4-Port Digital Voice Mail System	1	1	Notes 1~4, 6
CNF(8)-U() ETU	8-port Conference Unit. This Multiline Conference Bridge allows any intercom user or outside party calling to a port of the CNF(16)-U20 ETU to join or make a multiparty Conference Call. Each ETU supports one 8-party conference or two 4-party conferences regulated by a switch setting. This ETU is installed in slots S1~S8 in the B64-U20 KSU. The system recognizes this ETU as an SLI(8)-U() ETU. This ETU shares the total number of station ports in the system.	2	2	Notes 1, 4, 6

Table 2-41 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
CNF(16)-U20 ETU	16-port Multimedia Conference Unit. This Multiline Conference Bridge can support 8 or 16 ports. Each 8-port ETU supports one 8-party conference or two 4-party conferences. Each 16-port ETU supports one 16-party conference, two 8-party conferences, one 6-party and two 5-party conferences, or four 4-party conferences. This ETU is installed in slots S1~S8 in the B64-U20 KSU. The system recognizes this ETU as a CNF()-U20 ETU. This ETU shares the total number of station ports in the system.	1	1	Notes 1, 4, 6
CTI(4)/(8)-U() (System) ETU	This ETU is a 4- or 8-port Digital Voice Mail system with ports that support TeLANophy, inbound or outbound faxing, and Hospitality/HVM applications. It is installed in an interface slot. This ETU shares the total number of station ports in the system.	1	1	Notes 1, 3, 4, 6
CTI(12)/(16)-U() (Daughter) ETU	This ETU and the 4- and 8-port ETU provide a 12- or 16-Port Digital CTI System Digital Voice Mail system with ports that support TeLANophy, inbound or outbound faxing, and Hospitality/HVM applications. It is installed in any interface slot. This ETU shares the total number of station ports in the system.	1	1	Notes 1, 3~5, 6

Table 2-41 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
CTP()-U() ETU	<p>This ETU is a multiplatform system that supports a maximum of 16 ports. It is a PC platform that contains data storage for voice recording and application software. A digital signal processor/voice processing section handles the following functions:</p> <ul style="list-style-type: none"> • DTMF detection and generation • General tone detection • FAX CNG tone detection • PCM compression for audio recording/playback • Automatic Gain Control (AGC) • Two USB 1.0 ports for USB keyboard and mouse support • One 15-pin VGA Connector for VGA monitor support <p>One DSP8-U10 ETU is required for eight ports. Two DSP8-U10 ETUs are required for 12 or 16 ports.</p>	1	1	Notes 1, 3, 4, 6
DSP11-U10 Unit with In-Mail 2-port or In-Mail 4-port Compact Flash card installed.	This unit is a daughter board that is installed on the CPU11()-U10 ETU and is used for the VRS or In-Mail Compact Flash.	1	1	7
ESI(8)-U() ETU	8-port Electronic Station Interface No more than 80 ESI ports are supported in a single KSU.	7	23	
ESIB(8)-U() ETU	<p>This 8-port Electronic Station Interface ETU contains eight circuits. Each circuit can support any Attendant Console, Multiline Terminal, or Single Line Telephone adapter.</p> <p>This ETU is installed in slots S1~S8 in the basic or expansion B64-U20 KSU. The maximum number depends on other station ETUs installed. This ETU shares the total number of extension ports in the system.</p> <p>No more than 80 ESI ports are supported in a single KSU.</p>	7	23	Notes 1, 6

Table 2-41 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
ESIB(8)-U() ETU with ESIE(8)-U() ETU	<p>These ETUs are a 16-port Electronic Station Interface. The ESIE ETU is installed on ESIB ETU. This Electronic Station Interface ETU contains eight circuits. Each circuit can support any Attendant Console, Multiline Terminal, or Single Line Telephone adapter.</p> <p>This ETU is installed in slots S1~S8 in the basic or expansion B64-U20 KSU. The maximum number depends on other station ETUs installed. This ETU shares the total number of extension ports in the system.</p> <p>No more than 80 ESI ports are supported in a single KSU.</p>	3	15	Notes 1, 6
FMS(2)/(4)-U() ETU	<p>This 2- or 4-port Digital Voice Mail System is installed in any interface slot. It has eight channels of built-in Voice Mail. The system recognizes this ETU as a VMS(4)-U() ETU. This ETU shares the total number of station ports in the system.</p>	1	1	Notes 1~4, 6
FMS(8)-U() ETU	<p>This 8-port Digital Voice Mail System is installed in any interface slot. It has two or four channels of built-in Voice Mail. The system recognizes this ETU as a VMS(8)-U() ETU. This ETU shares the total number of station ports in the system.</p>	1	1	Notes 1, 3, 4, 6
IVR Application [VMP(4)/(8)-U() ETU with IVR HDD Kit]	<p>This Interactive Voice Response ETU has four ports to support IVR applications. When the DSP-U() module is attached, eight ports are available. It is installed in any interface slot. This ETU shares the number of station ports in the system.</p>	1	1	Notes 1, 3, 4, 6

Table 2-41 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
OPX(2)-U() ETU	This 2-port Off-Premise Extension Interface ETU provides termination and operation of two off-premise extensions. Each ETU has a built-in ringer signal generator (RSG). Up to 1600 ohms of resistance (including the Single Line instrument) is acceptable between the OPX ETU and the single line telephone. This ETU is installed in slots S1~S8 in any B64-U20 KSU and shares the number of station ports in the system.	6	22	Note 1, 2, 6
SLI(4)-U() ETU	This 4-port Single Line Interface ETU supports four Single Line Telephones and/or analog voice mail ports. Each ETU provides a built-in ringer signal generator (RSG) and Message Waiting (MW) LED voltage to single line telephones. This ETU is installed in slots S1~S8 in any B64-U20 KSU. The maximum number depends on other station ETUs installed. This ETU shares the total number of station ports in the system.	12	22	Notes 1, 6
SLI(8)-U() ETU	This 8-port Single Line Interface ETU supports eight Single Line Telephones and/or analog voice mail ports. Each ETU provides a built-in ringer signal generator (RSG) and Message Waiting (MW) LED voltage to single line telephones. This ETU is installed in slots S1~S8 in any B64-U20 KSU. The maximum number depends on other station ETUs installed. This ETU shares the total number of station ports in the system.	6	22	Notes 1, 6

Table 2-41 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
SLIB(4)-U() ETU	This 4-port Single Line Interface ETU supports four single line telephones. Each ETU provides a built-in ringer signal generator (RSG) and Message Waiting (MW) LED voltage to Single Line Telephones. This ETU is installed in slots S1~S8 in any B64-U20 KSU. The maximum number depends on other station ETUs installed. This ETU shares the total number of station ports in the system.	12	22	Notes 1, 6
SLIB(4)-U() ETU with SLIE(4)-U() ETU installed	These ETUs are an 8-port Single Line Interface. The SLIE ETU is installed on the SLIB ETU, and they support eight single line telephones with built-in ringer signal generator (RSG) and Message Waiting (MW) LED voltage to single line telephones. This combination ETU is installed in slots S1~S8 in any B64-U20 KSU. The maximum number depends on other station ETUs installed. This ETU shares the total number of station ports in the system.	6	22	Notes 1, 6
VMP()-U40 VMP(2)/(4)/(8)-U40	This ETU has two, four or eight channels of built-in Voice Mail. This ETU is a PC platform that contains disk space for voice recording storage and application software. A DSP-U30 Unit must be installed for the 8-port interface.	1	1	Notes 1~4, 6
VMS(2)/(4)-U() ETU	This 2- or 4-port Digital Voice Mail System is installed in any Interface slot. It has two or four channels of built-in voice mail. The system recognizes this ETU as a VMS(4)-U() ETU. This ETU shares the total number of station ports in the system.	1	1	Notes 1~4, 61018

Table 2-41 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
VMS(8)-U() ETU/	This 8-port Digital Voice Mail System is installed in any Interface slot. It has eight channels of built-in voice mail. The system recognizes this ETU as a VMS(8)-U() ETU. This ETU shares the total number of station ports in the system.	1	1	Notes 1, 3, 4, 6
VP(4)/(8)-U() (System) ETU	This 4- or 8-port Digital Voice Mail System has ports that support TeLANophy, inbound/outbound faxing, and Hospitality/HVM applications. It is installed in any interface slot and shares the total number of station ports in the system.	1	1	Notes 1, 3, 4, 6
VP(12)/(16)-U() (Daughter) ETU	This 12- or 16-Port Digital Voice Mail System and the 4- or 8-port provide a 12- or 16-port Digital Voice Mail System with ports that support TeLANophy, inbound/outbound faxing, and Hospitality/HVM applications. It is installed in any interface slot and shares the total number of station ports in the system.	1	1	Notes 1,3~6,

Note 1: Calculating maximum capacities is based on the system having a minimum of eight Electronic Station Interface (ESI) ports, four trunk ports and a PKUII-U Unit installed.

Note 2: When 2-port Station Interface ETUs are installed, the system uses four ports from its maximum port capacity.

Note 3: Only one CMS, FMS, VMS, VP, CTI, CTP, IVR system can be installed in one Electra Elite IPK II system.

Note 4: A maximum of 32 Digital Voice Mail ports are available.

Note 5: Two physical Interface Slots are used for the EliteMail VP 12/16-port system and the EliteMail CTI 12/16-port system.

Note 6: Refer to the KSU Power-Based Calculator Chart.

Note 7: With **Version 1500 or higher**, when the DSPII-U10 Unit with an In-Mail 2- or 4-port Compact Flash is installed on the CPUII()-U() ETU, it does not use any ports from the maximum station port capacity of a basic port package, but it does use eight ports from the maximum station port capacity of an expanded port package.

Table 2-42 Maximum System Capacities for Trunk Interface ETUs

Trunk Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
BRT(4)-U() ETU	This 4-port Basic Rate Interface for eight trunks provides four channels (eight voice channels) for an ISDN-Basic Rate Interface. Caller ID is supported. This ETU is installed in slots S1~S4 in the basic or first expansion B64-U20 KSU. The maximum number depends on other trunk ETUs installed. This ETU shares the total number of CO/PBX lines in the system.	7	23	Notes 1, 6
COI(4)-U() ETU	This 4-port CO/PBX Line Interface has built-in fuses (posistors), supports four outside (CO/PBX) lines, and provides circuitry for ring detection, holding and dialing. The outside lines must be Loop Start DTMF trunks. This ETU is installed in slots S1~S8 in the basic or expansion B64-U20 KSU. The maximum number depends on other trunk ETUs installed. This ETU shares the total number of CO/PBX lines in the system.	14	23	Note 1
COI(8)-U() ETU	This 8-port CO/PBX Line Interface has built-in fuses (posistors), supports eight outside (CO/PBX) lines, and provides circuitry for ring detection, holding and dialing. The outside lines must be Ground Start DTMF trunks. This ETU is installed in slots S1~S8 in the basic or expansion B64-U20 KSU. The maximum number depends on other trunk ETUs installed. This ETU shares the total number of CO/PBX lines in the system.	7	23	Note 1

Table 2-42 Maximum System Capacities for Trunk Interface ETUs (Continued)

Trunk Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
COIB(4)-U() ETU	This 4-port CO/PBX Line Interface can function the same as the COI(4) or COID(4) ETU to provided a Central Office Interface. When set for COID mode, Loop Start DTMF trunks and/or Caller ID trunks are supported, and the ETU supports loop start only. When the ETU is set for COI mode, loop start or ground start DTMF trunks are supported, but Caller ID is not supported. Connections for ground start trunks are polarity sensitive. For COID mode, Caller ID trunks must be installed in slots S1~S4. This ETU shares the total number of CO/PBX lines in the system. Tip and Ring electrical fuses are provided to comply with UL 1459 requirements.	14	23	Notes 1, 3
COIB(8)-U() ETU	8-port CO/PBX Line Interface can function the same as the COI(4) or COID(4) ETU to provided a Central Office Interface. When set for COID mode, Loop Start DTMF trunks and/or Caller ID trunks are supported, and the ETU supports loop start only. When the ETU is set for COI mode, loop start or ground start DTMF trunks are supported, but Caller ID is not supported. Fax CO Branch is not supported. Connections for ground start trunks are polarity sensitive. Only DTMF signaling is supported. For COID mode, Caller ID trunks must be installed in slots S1~S4. This ETU shares the total number of CO/PBX lines in the system. Tip and Ring electrical fuses are provided to comply with UL 1459 requirements.	7	23	Note 1

Table 2-42 Maximum System Capacities for Trunk Interface ETUs (Continued)

Trunk Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
DID(4)-U() ETU	<p>This 4-port Direct Inward Dialing Interface supports four DID or four two-way DID lines. Each ETU requires one interface slot in the KSU. Immediate, wink start, second dial tone, and delay dial signaling can be combined on this ETU.</p> <p>This ETU is installed in slots S1~S8 in any B64-U20 KSU. The maximum number depends on other trunk ETUs installed. This ETU shares the total number of CO/PBX lines in the system.</p>	14	23	Notes 1,4
DTI-U40 ETU for DTI Function DTI-U40 ETU for PRI Function	<p>This T1/FT1 Trunk Interface or ISDN-Primary Rate digital trunk terminates Fractional T1 trunks (Up to 24 DS-0 channels). This ETU supports K-CCIS, ANI/DNIS trunks, and CSU less function on T1. A combination of ground start and loop start signaling can be used on the DTI-U40 ETU. Dial pulse dialing, DTMF, Tie Line (E&M), and DID are supported. This ETU has 24 built-in DTMF detectors. Trunks are assigned in groups of four.</p> <p>When channels are assigned to ANI, Feature Group D is supported. Feature Group D incoming MF/ outgoing DTMF signaling and K-CCIS signaling with point-to-point E&M Tie Lines are also supported.</p> <p>This ETU is installed in slots S1~S8 in any B64-U20 KSU. The maximum number depends on other trunk ETUs installed. This ETU shares the total number of CO/PBX lines in the system.</p>	2	11	Notes 1, 5, 7.

Table 2-42 Maximum System Capacities for Trunk Interface ETUs (Continued)

Trunk Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
TLI (2)-U() ETU	This 2-port Tie Line Interface ETU supports the termination and operation of two E&M Tie Lines (4-wire, type I and type V, and 10/20 pps Dial Pulse or DTMF). Immediate, wink start, second dial tone, and delay dial signaling can be combined on this ETU. This ETU is installed in slots S1~S8 in any B64-U20 KSU. The maximum number depends on other trunk ETUs installed. This ETU shares the total number of CO/PBX lines in the system.	14	23	Note 2

Note 1: Calculating maximum capacities are based on the system having a minimum of eight Electronic Station Interface (ESI) ports, four trunk ports and a PKUII-U Unit installed.

Note 2: When 2-port Trunk Interface ETUs are installed, the system uses four ports from its maximum port capacity.

Note 3: With the Electra Elite IPK II Expanded Port Package, a maximum of 14 COIB(4)-U() ETUs can be installed as COID(4)-U() ETUs.

Note 4: Refer to the KSU Power-Based Calculator Chart.

Note 5: Firmware 5.0 or higher is required.

Note 6: Firmware 3.0 or higher is required.

Note 7: The first four DTI/PRI ETUs are assigned 24 channels, the next six are assigned 16 channels and the eleventh is assigned eight channels.

Table 2-43 Maximum System Capacities for Application Interface ETUs

Application Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
BSU(4M)-U20 ETU	The Master 4-Base Station Unit for Wireless DECT provides connections for up to 16 Base Stations when using two BSU(6S) ETUs. This ETU is installed in slots S1~S8 (S2~S8 when using one slave ETU or S3~S8 when using two slave ETUs) in any B64-U20 KSU.	1	1	Notes 1, 2

Table 2-43 Maximum System Capacities for Application Interface ETUs (Continued)

Application Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
BSU(2S)-U20 ETU	This Slave 2-Base Station Unit for Wireless DECT has connections for two Base Stations and must be used with the BSU(4M)-U20 ETU. This ETU is installed in the first or second slot to the left of the BSU(4)-U20 ETU.	2	2	Notes 1, 2
BSU(6S)-U20 ETU	This Slave 6-Base Station Unit for Wireless DECT has connections for six Base Stations and must be used with the BSU(4M)-U20 ETU. This ETU is installed in the first or second slot to the left of the BSU(4)-U20 ETU.	2	2	Notes 1, 2
CCH(4)-U() ETU	This 4-Channel – Common Channel Handler (CCH) for K-CCIS provides four K-CCIS routes to coordinate receiving common channel data from a distant system or to coordinate sending it to a distant system.	1	1	
HUB(8)-U() ETU	This 8-port Switching HUB is an optional Ethernet interface that supports eight internet ports. Each port has two LEDs that indicate status and activity. One port can be a source port, and another port can be used to mirror source and monitor data traffic. One ETU can be installed in slots S1~S8 in each cabinet. This ETU cannot be installed in a KSU that contains EliteMail VP and IVR or CTI and IVR systems.	1	1	Note 1
IAD(8)-U() ETU configured for ESI	For MEGACO Station	3	14	
IAD(8)-U() ETU configured for CCISoIP	For CCISoIP	6	22	Note 3
PVA(X)-U() ETU as an IP Station (MEGACO)-MG16	For MEGACO Station	2	13	

Table 2-43 Maximum System Capacities for Application Interface ETUs (Continued)

Application Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
PVA(X)-U() ETU as CCISoIP	For CCISoIP Version 1500 or higher – 4-port license registration provides users with the flexibility of adding CCISoIP Ports in increments of four (4, 8, 12, 16, 20 or 24) or (MG 16) DSP Resources in increments of four (4, 8, 12, and 16).	2	11	Note 4

Note 1: Refer to the KSU Power-Based Calculator Chart.

Note 2: A maximum of three Wireless – DECT BSU(4M)/(2S)/(6S) ETUs can be installed.

Note 3: The IAD(8)-U() ETU is assigned as a CCISoIP ETU and counts as eight trunk ports when installed with the IP CCH ETU application loaded.

Note 4: The PVA(X)-U() ETU is assigned as a CCISoIP ETU and counts as 24-trunk ports when installed with the CCISoIP application package.

- Refer to the IPK II General Description Manual or System Hardware Manual for more information.
- COI/COID card will show as COI in 10-03-01, but with show correct in 90-34-01.
- The following Card Calculator allows you to determine the maximum power consumption for each cabinet.
 - ✓ The total point value cannot exceed 1000 points for the +5 voltage.
 - ✓ The total point value cannot exceed 1000 points for the -24 voltage.

Table 2-44 KSU Power-Based Calculator Chart

Package Name	Power Consumption		Number of ETUs	Equivalent Total Power Point	
	+5V	-24V		+5V	-24V
Common					
CPU()-U() ETU	200	0			
CF ETU	7	0			
EXP-U10 ETU	6	0			
MOD-U10 Unit	25	0			
Trunk					
BRT(4)-U() ETU	67	0			
COI(4)-U10 ETU	48	5			

Table 2-44 KSU Power-Based Calculator Chart (Continued)

Package Name	Power Consumption		Number of ETUs	Equivalent Total Power Point	
	+5V	-24V		+5V	-24V
COI(8)-U10 ETU	82	9			
COID(8)-U() ETU	55	9			
COIB(4)-U20 ETU	29	5			
COIB(4)-U30 ETU	29	5			
COIB(8)-U30 ETU	54	9			
DID(4)-U() ETU	24	30			
DTI-U10/20/30 ETU	84	0			
DTI-U40 ETU	73	0			
IAD(8) ETU	31	110			
TLI(2)-U() ETU	15	10			
Station					
CNF(16)-U20 ETU	17	52			
ESI(8)-U() ETU	21	83			
ESIB(8)-U() ETU	21	83			
ESIB(8) plus ESIE(8)-U10 ETU (16 Ports)	32	166			
OPX(2)-U() ETU	22	30			
SLI(4)-U() ETU	29	25			
SLI(8)-U() ETU	52	29			
SLIB(4)-U() ETU	30	17			
SLIE(4)-U() ETU	17	12			
Voice Mail					
CMS(2)-U10 ETU	62	54			
CMS(4)-U10 ETU	62	54			
FMS(2)-U10 ETU	62	54			
FMS(4)-U10 ETU	62	54			
VMS(2)/(4)-U10 ETU	60	57			
VMS(8)-U10 ETU	64	84			
FMS(2)/(4)-U20 ETU	75	97			
FMS(8)-U20 ETU	100	97			
VMS(2)-U20 ETU	74	100			
VMS(4)-U20 ETU	74	100			

Table 2-44 KSU Power-Based Calculator Chart (Continued)

Package Name	Power Consumption		Number of ETUs	Equivalent Total Power Point	
	+5V	-24V		+5V	-24V
VMS(8)-U20 ETU	99	101			
CMS(2)-U30 ETU	55	68			
CMS(4)-U30 ETU	55	68			
FMS(2)-U30 ETU	55	68			
FMS(4)-U30 ETU	55	68			
FMS(8)-U30 ETU	81	68			
VMS(4)-U30 ETU	55	96			
VMS(8)-U30 ETU	80	100			
VMS(4)-U40 ETU	55	96			
VMS(8)-U40 ETU	80	100			
FMS(2)-U40 ETU	55	68			
FMS(4)-U40 ETU	55	68			
FMS(8)-U40 ETU	81	68			
VP/CTI/IVR(8)-U10 ETU	155	193			
VP/CTI/IVR(16)-U10 ETU	274	193			
CTP()-U10 ETU	120	288			
Optional					
BSU(4M)-U20 ETU	77	47			
BSU(2S)-U20 ETU	35	26			
BSU(6S)-U20 ETU	47	69			
CCH(4)-U() ETU	50	0			
HUB-U10 ETU	250	0			
PVA(X)-U() ETU	17	52			
VMP()-U40 ETU	55	96			
Total Points			XX	XXX	XXX

Default Setting

None

System Availability

Terminals

N/A

Required Component(s)

Any ETU

Related Features

None

Programming

- **10-03-01 ETU Setup**
Setup and confirm the Basic Configuration data for each ETU.
- **90-34-01 Firmware Information – Pkg Name**
- **90-34-02 Firmware Information – Firmware Version Number**
Used to view the package name and firmware for each ETU.

Operation

None

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User Programming Ability

Description

A station user can perform programming functions. Speed Group Dialing and Function Keys are just two features programmable from a station.

Conditions

- Multiline Terminals must be idle, Off-Hook and user must enter the service code when programming any function.

Default Setting

None

System Availability

Terminals

All Terminals

Required Component(s)

None

Related Features

Clock/Calendar Display

Code Restriction

One-Touch Calling

Programmable Function Keys

Speed Dial – System/Group/Station

Programming

None

Operation

None

Virtual Extensions

Enhancements

Version 1600 or higher:

- The following functionality has been added for Virtual Extensions:
 - Barge-In to a busy Virtual Extension
 - Conference Call
 - Handset Mute
 - Reverse Voice Over
 - Voice Call/Privacy Release
 - Voice Override to a busy Virtual Extension
- A call which is initiated or answered on a Virtual Extension (talking on the key) can be Quick Transferred to Voice Mail. Prior to this release, a Quick Transfer to Voice Mail was not allowed.
- Caller ID for transferred calls to a Virtual Extension that is programmed to ring is displayed. Prior to this release, Caller ID would not be displayed.



Description

Virtual Extensions are available software extensions on the Basic and Expanded Port Packages. A Virtual Extension assigned to a line key, can appear and ring on an individual station or multiple stations and be used for outbound access.

Virtual Extensions (VE) are shared with Call Arrival (CAR) Keys. In virtual extension mode, the key acts as a secondary extension. Up to 256 CAR/VE keys are provided.

Conditions

- There are 256 available ports/Extensions shared between CAR keys and Virtual Extensions.
- The 256 available ports/Extensions are assigned per extension for CAR key mode or Virtual EXTension key mode.
- More than one extension can share a Virtual Extension key.
- An extension can have more than one Virtual Extension key assigned.
- Up to 32 incoming calls can be queued to busy Virtual Extension key.
- You cannot have a CAR key and Virtual Extension on the same telephone.

- Virtual Extensions do not support the following features (**Version 1500 or lower**):
 - Barge-In
 - Conference
 - Conference, Voice Call/Privacy Release
 - Reverse Voice Over
 - Tone Override
 - Voice Over
- When a valid system station calls a Virtual Extension appearing on another station, Voice and MW softkeys appear in the display of the calling station, but they do not operate.
- When talking on a Virtual Extension you cannot mute the handset.
- Incoming calls to a virtual extension that appear on stations that are used with the CTI applications, PC Assistant, or PC Attendant, do not show up as a second call in the CTI application.
- Calls on Virtual Extension keys cannot be call parked.
- Calls on Virtual Extension keys cannot be put in Personal Park if Program 15-18-01 is set to Land on the key (1).
- If multiple CAR/SIE/VE keys are ringing on a station at the same time, the CAR/SIE/VE key on the lowest Line Key is answered first.
- Virtual Extension Keys assigned as code *03 do not support Voice Mail Message Indication on Line Keys.
- Busy Virtual Extensions cannot be Tone overridden.

Default Settings

Extensions 201~299 and 3301~3457 are the default for CAR/VE.

System Availability

Terminals:

All Multiline Terminals

Required Component(s)

None

Related Features

Call Queuing

Call Arrival (CAR) Keys

Secondary Incoming Extension

Programming

- **11-04-01 Virtual Extension Numbering**
Assign Extension Number for the Virtual Extensions (1~256).
(default: 1~99 = 201~299, 100~256 = 3301~3457)
- **15-01-01 Basic Extension Data Setup – Extension Name**
Assign Names for the Virtual Extensions (1~256).
(default: STA101 = Extension 101 = STA102 = Extension 102, etc.)
- **15-02-21 Multiline Telephone Basic Data Setup – Virtual Extension Access Mode (when idle Virtual Extension key pressed)**
Determine whether an extension Virtual Extension Key should be used as a DSS key to the extension and for receiving calls (0), answering incoming calls and ability to place outgoing ICM or CO calls (1), or just receiving incoming calls (2). If the key is to be used for outgoing calls, the extension number of the key must be a real extension or virtual extension number. When the extension number of the key is a real extension number, when the key is pressed, the real extension cannot be used.
(default: 2)
- **15-02-30 Multiline Telephone Basic Data Setup – Toll Restriction Class**
Assign if the phone uses the Toll Restriction class of the VE (0) or the Real Extension when making outbound calls from the VE (1).
(default: 1)
- **15-07-01 Programmable Function Keys**
Assign Virtual Extension function keys on Multiline telephones (code *03 + extension number).
- **15-08-01 Incoming Virtual Extension Ring Tone Setup**
Assign a ring tone (0~4) to each extension port and virtual extension port assigned to a Virtual Extension key. If ringing is enabled for the key in Program 15-09-01, the key rings with the tone set in this program.
[default: 0 (Tone Pattern 1)]
- **15-09-01 Virtual Extension Ring Assignment**
Individually program an extension Virtual Extension key(s) to either ring (1) or not ring (0).
(default: 0)

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-
- ➔ **15-10-01 Virtual Incoming Extension Ring Tone Order Setup**
Set the priority (1~4) for the Virtual Extensions' Ring Tones set in Program 15-08-01. When simultaneous Virtual Extension calls ring an extension, the tone with the highest priority (e.g., 1) rings and the other keys just flash.
(default: Order 1~4 = Tone 0~3)

 - ➔ **15-11-01 Virtual Extension Delayed Ring Assignment**
Individually program an extension Virtual Extension key(s) for Delayed Ringing (1) or Immediate Ringing (0).
(default: 0)

 - ➔ **15-18-01 Virtual Extension Key Enhanced Options – Virtual Extension Key Operation Mode**
Assign if a call to a VE Holds (1) on the VE or Release (0) to the phone that answered the VE.
(default: 0)

 - ➔ **15-18-02 Virtual Extension Key Enhanced Options – Display Mode when pacing a call on Virtual Extension Key**
Assign if the phone displays the extension name of the Virtual Extension (0) or the Real Extension (1) when making call from the Virtual Extension. PRG 15-02-21 must be set to OTG (1).
(default: 0)

 - ➔ **20-04-03 System Options for Virtual Extensions – CAR/SIE/Virtual Extension Delay Interval**
Assign the Delay Ring Time (0~64800 seconds) for Virtual Extensions.
(default: 10)

 - ➔ **20-06-01 Class of Service for Extensions**
Assign the Extension to a Class of Service.

 - ➔ **20-07-10 Class of Service Options (Administrator Level) – Programmable Function Key Programming (Appearance Level)**
Turn Off (0) or On (1) programming of Appearance Function keys using default service code 752
(default: 1).

 - ➔ **20-10-08 Class of Service Options (Answer Service) – Virtual Extension Off-Hook Answer**
Determine if the extension Class of Service answers (1) or not answers (0) when the extension goes off-hook.
(default: 1)


 - ➔ **20-13-27 Class of Service Options (Supplementary Service) – Busy on Seizing Virtual Extension**
Enable (1)/Disable (0) the ability to call a busy extension which is talking on a virtual extension key. Program 20-13-06 (Call Waiting) must be set to off for this option to work.
(default: 1)

 - ➔ **21-01-15 System Options for Outgoing Calls – Outgoing Disable on Incoming Line (Toll Restriction)**
Enable (1)/Disable (0) the Outgoing Disable on Incoming Line feature.
(default: 0)


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- ➡ 23-04-01 Ringing Line Preference for Virtual Extensions**
When an extension has a virtual extension on Function Key, this program determines the priority (1~4) for a Ring Group for automatically answering ringing calls when the handset is lifted. If (00) is selected for the Ring Group, when the user lifts the handset, the user can answer a ringing call from any group.
(default: 00)

Operation


To answer a call ringing a Virtual Extension:

1. Press the flashing **Virtual Extension** key.
- OR -
Go off-hook.
 *PRG 20-10-08 must be set to 1 for extension Class of Service.*

To place a call to a Virtual Extension:

1. Go off-hook.
2. Dial the Virtual Extension, or press the **Virtual Extension** key.
 *The operation depends on the setting in PRG 15-02-21.*

To place a call from a Virtual Extension:

1. Press the **Virtual Extension** key.
 *The operation depends on the setting in PRG 15-02-21.*
2. Place an intercom call or dial a trunk access code to seize an outside line and place your call.

To program a Virtual Extension key on a telephone:

1. Press **Speaker**.
2. Dial **752**.
3. Press the key you want to program.
4. Dial ***03**.
5. Dial the number of the extension you want to appear on the key.
6. Press **Hold** once for Immediate Ring (skip to step 8 for Delayed Ring).

7. Dial the mode number in which the key rings.
 - 1 = Day 1
 - 2 = Night 1
 - 3 = Midnight 1
 - 4 = Rest 1
 - 5 = Day 2
 - 6 = Night 2
 - 7 = Midnight 2
 - 8 = Rest 2
8. Press **Hold** for a second time for Delayed Ring, or Skip to step 10.
9. Dial the mode number in which the key delay rings.
 - 1 = Day 1
 - 2 = Night 1
 - 3 = Midnight 1
 - 4 = Rest 1
 - 5 = Day 2
 - 6 = Night 2
 - 7 = Midnight 2
 - 8 = Rest 2
10. Press **Speaker**.

Voice Mail Integration (Analog)

Enhancements

Version 1500 or higher supports flexible Voice Mail integration to the Analog Voice Mail ports.

Version 1600 or higher allows the flash rate and color of the Message Waiting LED to be configured for the following conditions:

- Message Waiting Lamp Cycle for Calling Extension (PRG 15-02-35)
- Message Waiting Lamp Cycle for Called Extension (PRG 15-02-36)
- Voice Mail Message Wait Lamp Color (PRG 15-02-37)
- Voice Mail Message Wait Lamp Cycle (PRG 15-02-38)

This provides more distinction between incoming calls, Message Waiting (Set/Received) and a VM Message Waiting indication.

Description

The system provides telephone users with comprehensive Voice Mail features. Voice Mail ends the frustration and cost of missed calls, inaccurate written messages and telephone tag. This frees busy receptionists and secretaries for more productive work.

External voice mail requires available analog station ports based on the number of voice mail ports connected.

Integrated voice mail enhances the telephone system with the following features:

Call Forwarding to Voice Mail

An extension user can forward their calls to Voice Mail. Once forwarded, calls to the extension connect to that extension mailbox. The caller can leave a message in the mailbox instead of calling back later. Forwarding can occur for all calls immediately, for unanswered calls or only when the extension is busy. When a user transfers a call to an extension forwarded to Voice Mail, the call waits for the Delayed Call Forwarding time before routing to the called extension mailbox. This gives the transferring party the option of retrieving the call instead of having it go directly to the mailbox.

Leaving a Message

Voice Mail lets a multiline terminal extension user easily leave a message at an extension that is unanswered, busy or in Do Not Disturb. Callers can press their Voice Mail key to leave a message in the called extension mailbox. There is no need to call back later.

Transferring to Voice Mail

By using Transfer to Voice Mail, a multiline terminal extension user can Transfer a call to the user's or a co-worker's mailbox. After the Transfer goes through, the caller can leave a message in the mailbox.

Voice Mail Queuing

When accessing the voice mail, the system provides a voice mail queue. If all the voice mail ports are busy, any call trying to get to the voice mail is placed in queue. As the voice mail ports become available, the calls are connected to the voice mail in the order in which they were received.

As the Voice Mail Queue follows Department Hunting programming, the queue can hold a maximum of 10 calls. If the queue is full or if the voice mail ports are not assigned to a Department Group, the calls are handled as though there were no voice mail queuing feature enabled. The calls either access voice mail if a port is available or they receive a busy signal.

The Voice Mail Queuing feature does not work with the Conversation Record feature.


MSG Key Operates as Voice Mail Key

The system enhances a telephone MSG key function when connected to a system which has voice mail installed. When an extension receives a voice mail, the MSG key can be used to check the number of messages in voice mail, as well as call the voice mail to listen to the messages.

Analog Voice Mail Protocol Leading and Trailing Digits Assignment

With **Version 1500 or higher**, the Analog Voice Mail Protocol Leading Digits (KSU to VM) and the Trailing Digits format can be changed.

The following chart illustrates the input data for PRG 45-04-01~PRG 45-04-09 (Voice Mail Digit Add Assignment) based on the setting in Program 45-01-15 (Analog Voice Mail Protocol Selection) and Program 45-01-17 (Reply Mailbox Number). If PRG 45-01-15 is set to Fixed (0) it uses the Fixed Memory Location for the Leading Digits or, if set to Program (1) it uses PRG 45-04-01~PRG 45-04-09 for the Leading Digits. If Program 45-01-17 is set to 0 (No), it does not have the calling party in the Trailing Digits.

 *The default values for PRG 45-04-01~PRG 45-04-09 are not assigned.*

Use the chart below to determine what leading and trailing digits will be sent to the Analog Voice Mail System.

Program	Program 45-01-15 (0 = Fixed) Program 45-01-17 (1=Yes or 0=No)	Program 45-01-15 (1 = Program) Program 45-01-17 (1=Yes)	Program 45-01-15 (1=Program) Program 45-01-17 (0=No)	Description
45-04-01 - Remote Logon (Internal) Up to 4 digits * Default not assigned	***1XXX	Up to 4 digits + XXX	Up to 4 digits + XXX	Remote Log-On (Internal) <ul style="list-style-type: none"> ○ Internal call to VM from extension XXX. ○ User has not indicated intent to enter mail box.
45-04-02 - Direct Logon Up to 4 digits * Default not assigned	#XXX	Up to 4 digits + XXX	Up to 4 digits + XXX	Direct Log-On <ul style="list-style-type: none"> ○ Connect user to mail box for extension XXX.
45-04-03 - Transfer Message Up to 4 digits * Default not assigned	***2YYY ***2XXXYYY	Up to 4 digits + YYY Or Up to 4 digits + XXXYYY	Up to 4 digits + YYY	Transfer Message <ul style="list-style-type: none"> ○ User is transferring a call to VM ○ Record a message to be placed in mail box of extension YYY. Record Message for Called Extension (QVM) <ul style="list-style-type: none"> ○ Record a message to be placed in mail box of extension YYY. ○ Store source extension number XXX for automatic reply feature
45-04-04 - Forward-All Up to 4 digits * Default not assigned	***3UUUZZZ	Up to 4 digits + UUUZZZ	Up to 4 digits + ZZZ	Forward-All <ul style="list-style-type: none"> ○ Extension or Trunk UUU that called extension ZZZ and was forwarded to the Voice Mail Box of extension ZZZ.
45-04-05 - Forward-Busy Up to 4 digits * Default not assigned	***4UUUZZZ	Up to 4 digits + UUUZZZ	Up to 4 digits + ZZZ	Forward-Busy <ul style="list-style-type: none"> ○ Extension or Trunk UUU that called extension ZZZ and was forwarded to the Voice Mail Box of extension ZZZ.
45-04-06 - Forward RNA Up to 4 digits * Default not assigned	***5UUUZZZ	Up to 4 digits + UUUZZZ	Up to 4 digits + ZZZ	Forward RNA <ul style="list-style-type: none"> ○ Extension or Trunk UUU that called extension ZZZ and was forwarded to the Voice Mail Box of extension ZZZ.
45-04-07 - Remote Logon Up to 4 digits * Default not assigned	***6TTT	Up to 4 digits + TTT	Up to 4 digits + TTT	Remote Log-on <ul style="list-style-type: none"> ○ External call to Voice Mail from Trunk TTT. ○ Play welcome greeting and connect user to prompt.
45-04-08 - Conversation Recording Up to 4 digits * Default not assigned	***8NNN	Up to 4 digits + NNN	Up to 4 digits + NNN	Conversation Recording <ul style="list-style-type: none"> ○ Record a message to be placed in voice mail box of extension NNN.

Program	Program 45-01-15 (0 = Fixed) Program 45-01-17 (1=Yes or 0=No)	Program 45-01-15 (1 = Program) Program 45-01-17 (1=Yes)	Program 45-01-15 (1=Program) Program 45-01-17 (0=No)	Description
45-04-09 - Clear Down String Up to 4 digits * Default not assigned	9999	Up to 4 digits	Up to 4 digits	Clear down string. ○ Terminate

*=If leading digits are blanks, nothing will be sent to the Analog VM as integration.

Conditions

- When an analog device is a member of a Department Group assigned as the VM Department Group in PRG 45-01-01, it cannot perform an unsupervised transfer. When the party where the call is being transferred answers, the transferred party is connected to that party, and the analog device is released.
- The periodic reminder message requires a DSP daughter board for Voice Response System (VRS).
- Ring Group calls do not follow extension call forwarding to voice mail.
- Only one Voice Mail system can be installed in an IPK II system (Analog or Digital, but not both in same system). This restriction is because only one Department Group can be assigned for Voice Mail.
- If installing an Analog Voice Mail System, any Analog station port (SLT port) can be assigned to support the Analog Voice Mail system. With an Expanded Port Package, the IPK II supports up to 176 Analog station ports (22 x 8 ports = 176).
- If installing a Digital Voice Mail system (In-Skin product), an Analog station port (SLT port) can be assigned to support the sending of DTMF tones and Disconnect Signal to support a Fax server or other like products.
- When using Programmed (45-01-15 = 1) integration and 45-04-XX is blank, no trailing digits are sent. You can only allow the trailing digits to be sent by setting 45-05-XX to 1 (**Version 1600 and Version 2100 or higher is required**).
- Stutter Dial Tone is supported to Single Line Telephones for Voice Mail Message Waiting.

Default Setting

Disabled

System Availability

Terminals

All Stations

Required Component(s)

SLIB(4)-U10 ETU

SLIE(4)-U10 ETU

Related Features

Barge-In

Caller ID

Direct Inward Line (DIL)

Hold

Message Waiting

One-Touch Calling

Programmable Function Keys

Transfer

Programming

- ➔ **10-09-01 DTMF and Dial Tone Circuit Setup**

Assign at least one circuit for DTMF reception (type 0 or 1).
Use the following as a guide when allocating DTMF receivers:

 - In light traffic sites, allocate one DTMF receiver for every 10 devices that use them.
 - In heavy traffic sites, allocate one DTMF receiver for every five devices that use them.

-
-
- ➔ **11-07-01 Department Group Pilot Numbers – Dial**
Assign a Department Group pilot number for the Voice Mail (eight digits maximum). The extensions are assigned to the group in Program 16-02-01.
(default: No Setting)

 - ➔ **11-11-50 Service Code Setup (for Setup/Entry Operation) – Set Message Waiting Indication (Version 1500 or higher required)**
Assign a Service Code (eight digits maximum) to set a Message Waiting light from an Analog Voice Mail port.
(default: No Setting)

 - ➔ **11-11-51 Service Code Setup (for Setup/Entry Operation) – Cancel Message Waiting Indication (Version 1500 or higher required)**
Assign a Service Code (eight digits maximum) to cancel a Message Waiting light from an Analog Voice Mail port.
(default: No Setting)

 - ➔ **15-02-26 Multiline Telephone Basic Data Setup – MSG Key Operation Mode**
Determine whether an extension MSG key should function as a Message key (0) or Voice Mail key (1). If set as a Message key, the user can press the key to call the voice mail only when they have new messages. If set as a Voice Mail key, it functions as a normal Voice Mail key.
(Default = 0)

 - ➔ **15-02-35 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Calling Extension (Version 1600 or higher required)**
Select the Message Waiting flash pattern for the station that set the Message Waiting reminder.

1 = Cycle 1
2 = Cycle 2
3 = Cycle 3 (default)
4 = Cycle 4
5 = Cycle 5
6 = Cycle 6
7 = Cycle 7

 - ➔ **15-02-36 Multiline Telephone Basic Data Setup – Message Waiting Lamp Cycle for Called Extension (Version 1600 or higher required)**
Select the Message Waiting flash pattern for the station that receives the Message Waiting reminder.

1 = Cycle 1
2 = Cycle 2 (default)
3 = Cycle 3
4 = Cycle 4
5 = Cycle 5
6 = Cycle 6
7 = Cycle 7
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-

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- ➔ **15-02-37 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Color (Version 1600 or higher required)**
Select the Message Waiting flash pattern for the station that set the Message Waiting reminder.
0 = Green
1 = Red
(default: 1)

 - ➔ **15-02-38 Multiline Telephone Basic Data Setup – Voice Mail Message Wait Lamp Cycle (Version 1600 or higher required)**
Select the Message Waiting flash pattern for the station that set the Message Waiting reminder.

1 = Cycle 1
2 = Cycle 2(default)
3 = Cycle 3
4 = Cycle 4
5 = Cycle 5
6 = Cycle 6
7 = Cycle 7

 - ➔ **15-03-03 Single Line Telephone Basic Data Setup – Terminal Type**
Set all SLI ports used for Voice Mail as type 1.

 - ➔ **15-03-16 Single Line Telephone Basic Data Setup – Special DTMF Protocol Send (Version 1500 or higher required)**
For each Analog port, Enable (1)/Disable (0) the ability to send the extension number of the phone that forwarded to it when not assigned a VM port.

 - ➔ **15-07-01 Programmable Function Keys**
Assign a Voice Mail key to an extension. You must enter the Voice Mail key code (code 77) followed by:
 - Your own extension number if you are setting up your own Voice Mail key.
 - A virtual extension number if you are setting up a Message Center key for a virtual extension.
 - A co-worker's extension number if you are setting up a Message Center key for an installed extension.
 - An uninstalled extension number if you are setting up a Message Center key for an uninstalled extension.

(Optional) Assign a Voice Mail Record key to an extension (code 78).
(Optional) Assign a Personal Answering Machine Emulation key (code 16).
(Optional) Use a Call Redirect key (49) to allow a user to transfer a call to another extension or voice mail without answering the call.

 - ➔ **16-02-01 Department Group Assignment for Extensions**
Put all the Voice Mail extensions in a unique Department Group (do not use Department Group 1). No other extensions should be in this group. This allows DILs to Voice Mail to ring other Voice Mail extensions when the DIL's assigned port is busy. Only one Voice Mail group is allowed per system.

 - ➔ **20-02-09 System Options for Multiline Telephones – Disconnect Supervision**
Enable (1) disconnect supervision for the system.
(default: 1)
-
-

-
-
- ➔ **20-03-01 System Options for Single Line Telephones – SLT Call Waiting Answer Mode**
For a busy single line (500/2500 type) telephone, set the mode used to answer a camped-on trunk call. The default setting should be used [0 = Hookflash (Hooking)].
(default: 0)

 - ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to the voice mail extensions. You should use COS 14 for all time modes.

 - ➔ **20-11-01 through 20-11-05**
20-11-12 Class of Service Options (Hold/Transfer Service)
Set these options to 0 (Off) for the voice mail COS (recommended).

 - ➔ **20-13-01 through 20-13-07**
20-13-12
20-13-16
20-13-28
20-13-35 Class of Service Options (Supplementary Service)
Set these options to 0 (Off) for the voice mail COS (recommended).

 - ➔ **20-13-13 Class of Service Options (Supplementary Service) – Continued Dialing (DTMF) Signal on ICM Call**
Enable Continued Dialing (1) for all extensions that dial Voice Mail features.
(default: 1)

 - ➔ **22-02-01 Incoming Call Trunk Setup**

 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
To enable Voice Mail Overflow, assign selected extensions to a Ring Group that ring for unanswered DILs to Voice Mail ports. In PRG 22-06, enter 1 to enable overflow ringing.
(default: Extensions 101~108 ring for Incoming Ring Group 1 calls. No other extensions ring for Incoming Ring Group 1 Calls.)

 - ➔ **22-07-01 DIL Assignment**
Assign destination extension (up to eight digits).
(default: No Setting)

 - ➔ **22-08-01 DIL/IRG No Answer Destination**
For Voice Mail Overflow, enter the Ring Group that unanswered DILs to Voice Mail ring after the DIL Call Waiting time (PRG 22-01-04).
(default: 1)

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- ➔ **24-02-02 System Options for Transfer – MOH or Ringback on Transferred Calls**
Enable (0)/Disable (1) MOH on Transfer. If enabled, a transferred caller hears Music on Hold while their call rings the destination extension. If disabled, a transferred caller hears ringback while their call rings the destination extension. For this option to work with voice mail, the transferred call must be an unscreened transfer.
(default: 0)
 - ➔ **24-02-03 System Options for Transfer – Delayed Call Forwarding Time**
Set the time (0~64800 seconds) a transferred call waits at a forwarded extension before routing to the called extension mailbox.
(default: 10)
 - ➔ **40-03-01 through 40-03-03 Message Recording Setup**
 - ➔ **40-07-01 Voice Prompt Language Assignment for VRS**
 - ➔ **45-01-01 Voice Mail Integration Options – Voice Mail Department Group Number**
Assign which Extension (Department) Group number is to be assigned as the voice mail group. An entry of 0 means no voice mail is installed.
Department Groups: 0, 1~64
(default: 0)
 - ➔ **45-01-02 Voice Mail Integration Options – Voice Mail Master Name**
Enter the Voice Mail master name (up to 12 characters).
(default: Voice Mail)
 - ➔ **45-01-04 Voice Mail Integration Options – Park and Page**
Enable (1)/Disable (0) the system ability to process the Voice Mail Park and Page (*) commands.
(default: 1)
 - ➔ **45-01-05 Voice Mail Integration Options – Message Wait**
Enable (1)/Disable (0) the system ability to process the Voice Mail Message Wait (#) commands. You should normally *enable* this option. If enabled, be sure that the programmed Message Notification strings do not contain the code #9 for trunk access. When using an external voice mail and centrex transfer, this option should be disabled or the service code #3 in Program 11-12-42 must be changed.
(default: 1)
 - ➔ **45-01-06 Voice Mail Integration Options – Record Alert Tone Interval Time**
Set the time (0~64800 seconds) between Voice Mail Conversation Record alerts.
(default: 30)
 - ➔ **45-01-14 Voice Mail Integration Options – CCIS Centralized Voice Mail Number**
Assign the pilot number (up to eight digits) to Centralized Voice Mail over CCIS Link.
(default: No Setting)

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- ➔ **45-01-15 Voice Mail Integration Options – Analog Voice Mail Protocol Selection (Version 1500 or higher required)**
Assign whether DTMF sent to the Analog VM follows the Fixed (0) or the Programmed (1) leading digits. PRG 45-04-01~09 is where the programmed digits are assigned.
(default: 0)
 - ➔ **45-01-16 Voice Mail Integration Options – Voice Mail FAX Digit Add Assignment (Version 1500 or higher required)**
Assign the leading digits (up to four) to show in front of extension numbers sent to Analog ports assigned to 1 (Yes) in PRG 15-03-16.
(default: None)
 - ➔ **45-01-17 Voice Mail Integration Options – Reply Mail Box Number (Version 1500 or higher required)**
Enable (1)/Disable (0) the calling party Extension or Trunk to be sent to the Analog VM when PRG 45-01-15 is set to Programmed (1).
 - ➔ **45-01-18 Voice Mail Integration Options – Trunk Number Mapping (Version 1600 or higher required)**
Assign the digits (2 or 3) of trunk mapping.
(default: 2)
 - ➔ **45-04-01 through 45-04-09 Voice Mail Digit Add Assignment (Version 1500 or higher required)**
Assign up to four digits to show in front of call to an Analog VM to indicate what type of call is coming to the VM. This is used only when PRG 45-01-15 is set to Programmed (1).
(default: None)
 - ➔ **45-05-01 through 45-05-09 Voice Mail Send Protocol Signal Without Additional Digits (Version 1600 and 2100 or higher required)**
Allow (1)/Deny (0) the trailing digits to be sent when 45-04-01~09 is left blank.
(default: 0)
 - ➔ **80-03-01 through 80-03-08 DTMF Tone Receiver Setup**
 - ➔ **80-04-01 through 80-04-09 Call Progress Tone Detector Setup**
Set the criteria for DTMF dial, ringback and busy tones.

Operation

Calling your Mailbox


To call your mailbox:

Multiline Terminal

1. Press your **Voice Mail** key (PRG 15-07 or SC 751: 77) or the **Message** key.

- OR -


Press **Speaker** and dial the Voice Mail Master Number. After Voice Mail Answers, dial your mailbox number.


 *Your mailbox number is normally the same as your extension number. You may optionally dial a co-worker's mailbox - or use this procedure to call your mailbox from a co-worker's telephone.*

- OR -

Press **Speaker** and dial ***8**.


2. If requested by Voice Mail, enter your security code.

 *Ask your Voice Mail system administrator for your security code.*

 *Normally, your Message Waiting LED goes out (if applicable). If it continues to flash, you have unanswered Message Waiting requests or a new General Message. Go to To check your messages below.*

Single Line Telephone




1. Lift the handset and dial ***8**.

 *If you are at a co-worker's telephone, you can dial the Voice Mail master number and your mailbox number instead. You can also use this procedure from your own telephone to call a co-worker's mailbox.*

2. If requested by Voice Mail, enter your security code.

Checking Messages

1. Press the **Message** key once.



-  *The voice mail is called.*
-  *When there are new messages, the Large LED on the telephone flashes as red.*
-  *With this option set, the MSG key can be used as a Voice Mail key for any function [calling voice mail or transfer call a to voice mail (Hold + MSG + Extension Number), etc.].*

Recording your Call

To record your active call in your mailbox:

Multiline Terminal



1. Press **Voice Mail Record** key (PRG 15-07 or SC 751: code 78).

-  *You hear two beeps and your Record key flashes. The beeps periodically repeat to remind you that you are recording.*
-  *To stop recording, press the Voice Mail Record key again. You can restart and stop recording as required.*

- OR -

1. Place the call on hold by pressing **Hold**.



2. Dial **654**.

-  *The system automatically reconnects you to your call.*
-  *To stop recording, place the call on hold then pick the call back up. You can restart and stop recording as required.*

Single Line Telephone

1. Hookflash.

2. Dial **654**.

-  *The system automatically reconnects you to your call.*
-  *To stop recording, hookflash twice. You can restart and stop recording as required.*

Voice Mail Message Indication on Line Keys

Description

Voice Mail Message Indication on Line Keys indicates a new voice mail message on Line Keys or DSS/BLF keys.

Conditions

- When a DSS key of an installed extension is pressed when flashing, that extension is called.
- You must use a VM Message key (code 77) to get the indication when there is a new message. It can also be used for installed extensions.
- VM Message key calls the VM and logs into the mail box.
- If an extension VM Message key is placed on the extension, the Large LED does not light.
- VM message LED is a higher priority than any other status for the DSS/BLF key.
- The enabling/disabling of Voice Mail Indication on BLF enables the station with the message to show up on other telephones. It does not enable/disable stations from seeing the BLF indication.
- Virtual Extension Keys assigned as code *03 do not support Voice Mail Message Indication on Line Keys.

Default Setting

Not allowed

System Availability

Terminals

All Multiline Terminals

Required Component(s)

VM (Digital or Analog)

Related Features

Class of Service

Digital Voice Mail

Direct Station Selection (DSS) Console

IPK II In-Mail

Programmable Function Keys

Voice Mail Integration (Analog)

Programming

- ➔ **15-07-01 Programmable Function Keys**
Assign DSS/BLF function keys on Multiline telephones (code 01 + extension number) or Message Key (Code 77 + mailbox number).
- ➔ **20-13-41 Class of Service Options (Supplementary Service) – Voice Mail Message Indication on DSS**
Allow (1)/Deny (0) if the DSS/BLF key indicates when that extension has a new message waiting in VM.
(default: 0)
- ➔ **30-01-01 DSS Console Operating Mode**
Set the mode of the system DSS Consoles. The available options are Regular (Business) Mode (0), Hotel Mode (1), and ACD Monitor Mode (2).
(default: 0)
- ➔ **30-02-01 DSS Console Extension Assignment – Extension Number**
Enter extension numbers (up to eight digits) that have DSS Consoles connected to them.
(default: No Setting)
- ➔ **30-03-01 DSS Console Key Assignment**
Customize DSS Console keys to function as DSS keys, Service Code keys, Programmable Function Keys, and One-Touch Calling keys. The key (when defined as a DSS/One-Touch key [code 01] can have any function with up to four digits (e.g., extension number or Service Code). The function information (such as extension number or Service Code) would then be entered as additional data.
- ➔ **30-05-02 through 30-05-21 DSS Console Lamp Table**
If required, customize the system DSS Console flash rates (0~7). This table also affects DSS/Hotline key lamping for multiline terminals.

Operation

To program a DSS/BLF key on a telephone:

1. Press **Speaker**.
2. Dial **751**.
3. Press the key you want to program.
4. Dial **01**.
5. Dial the number of the extension you want to appear on the key.
6. Press **Hold**.
7. Press **Speaker**.

To program a VM Message key on a telephone:

1. Press **Speaker**.
2. Dial **751**.
3. Press the key you want to program.
4. Dial **77**.
5. Dial the number of the extension you want to appear on the key.
6. Press **Speaker**.

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Voice Over

Description

Voice Over lets a user interrupt a busy station user that is on another call. With Voice Over, the busy extension user hears an alert tone followed by the voice of the interrupting party. The extension user receiving the Voice Over can respond to the interrupting party without being heard by the original caller. If desired, the user can easily switch between their original caller and the interrupting co-worker. The original caller and the interrupting party can never hear each other.

Voice Over could help a lawyer, for example, waiting for an urgent call. While on a call with another client, the lawyer's paralegal could announce the urgent call as soon as it comes in. The lawyer could then give the paralegal instructions how to handle the situation – all without the original client hearing the conversation.

Both multiline terminal users and 500/2500 set users can initiate and receive a Voice Over.

To enable Voice Over, a multiline terminal can have a function key programmed for Voice Over. In addition to one-touch Voice Over operation, the key shows the Voice Over status as follows:

When the key is . . .	You are . . .
Off	Not using Voice Over
Flashing (Red)	Listening to the interrupting party
On (Green)	Responding to the interrupting party

Conditions

- While active, Voice Over uses a Conference circuit on a CPUII()-U10 ETU. Refer to the Conference feature for Conference circuit programming.
- Voice Over can interrupt a trunk call only if the trunk has been set up for at least six seconds.
- Do not use Voice Over to a user on speakerphone as the conversation may be heard by the outside party.
- When a multiline terminal user performs Voice Over, the speech path is 1-way from the originator to the destination.
- The Voice Over Access Code can be assigned on a Programmable Function Key.
- An override tone is sent to both calling and called parties. A single line telephone user can receive Voice Over. After a Tone Override is heard, Voice Over can be set.

- When a Programmable Function Key (programmed with the Voice Over Access Code) is pressed, the LED lights while responding to the page.
- When a multiline terminal has a Handsfree Unit programmed, the Voice Over call can be received and answered handsfree.
- When Data Line Security is assigned to a station, the Voice Over to the station is disabled.
- When a multiline terminal user performs Voice Over, the speech path is 1-way from the originator to the destination.
- An extension user cannot Voice Over to another extension user in a Conference.
- If you place a call on hold and then Voice Over to a busy extension, the call on hold does not transfer to the busy party when you end the Voice Over.
- A station can receive only one Voice Over at a time.
- A multiline terminal user cannot answer a Voice Over with an internal call on hold.
- An attempt to Voice Over a station can be denied if the station is in DND (Do Not Disturb) Mode, Automatic Redial is activated, during Station Programming, during Incoming Ringing, during Internal/External Paging, during a Conference Call, during a conference call on hold, the terminal is on internal hold, or the terminal has a call on internal hold.
- When a single line telephone is on a call and Voice Over is presented, the single line telephone cannot talk back to the party that originated the Voice Over.
- Voice Over to a single line telephone is not recommended because cross talk is inherent in the side tone of analog telephones.
- Voice Over to a user on speakerphone is not recommended because the conversation may be heard by the outside party.
- Answering a Voice Over requires a uniquely programmed Voice Over key.

Default Setting

Disabled

System Availability

Terminals

Multiline and Single Line Terminals

Required Component(s)

None

Related Features

Conference

Off-Hook Signaling

Programmable Function Keys

Programming

- **11-12-41 Service Code Setup (for System Access) – Voice Over**
The service code used for the Voice Over feature (default: 690).
- **11-16-08 Single Digit Service Code Setup – Voice Over.**
Service code used for the Voice Over feature (default: 6).
- **15-07-01 Programmable Function Keys**
Assign a function key for Voice Over (code 48).
- **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
- **20-13-06 Class of Service Options (Supplementary Service) – Automatic Off-Hook Signaling (Automatic Override)**
Enter 1 if you want callers to a busy extension to hear busy/ring tone. Enter 0 if you want callers to hear busy tone. The caller must then dial 7 to activate Intercom Off-Hook Signaling.
(default: 1)
- **21-01-03 System Options for Outgoing Calls – Trunk Interdigit Time (External)**
Set the time (0~64800 seconds) the system waits to expire before placing the call in a talk state (Call timer starts after time expires. Voice Over and Barge-In are not allowed until this time expires).
(default: 5)

Operation

To initiate a Voice Over to a busy extension:


1. Press **Voice Over** key (PRG 15-07 or SC 751: 48)


- OR -

Dial **6**.

- OR -

Dial **790**.


 *You hear an alert tone and the Voice Over key flashes. You can talk to the called party after the alert tone ends.*

 *To use Service Code 790 for Voice Over, Program 11-16-09 (Voice Mail Service Code) must be undefined.*

To respond to a Voice Over alert tone to your extension:

You can respond only if you have a Voice Over key.

1. Press the **Voice Over** key.


 *The Voice Over key lights steadily (green) and you can talk to the interrupting party.*

 *You cannot respond by dialing the Voice Over Service Code (6).*

To return to your original call:

1. Press the **Voice Over** key.

2. Press the **Voice Over** key again.

 *Your Voice Over key flashes red when you are talking to your original call.*

 *To switch between your original call and the interrupting party, just keep pressing the Voice Over key.*

Voice Over Internet Protocol (VoIP)

Description

VoIP (Voice over Internet Protocol) allows the delivery of voice information using the Internet protocol (sending data over the Internet using an IP address). This means that digital voice information can be sent in packets over the Internet rather than through the traditional public switch telephone network (CO lines). A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service.

Using VoIP equipment at a gateway (a network point that acts as an entrance to another network), the packetized voice transmissions from users in the company are received and routed to other parts of the company intranet (local area or wide area network) or they can be sent over the Internet using CO lines to another gateway.³

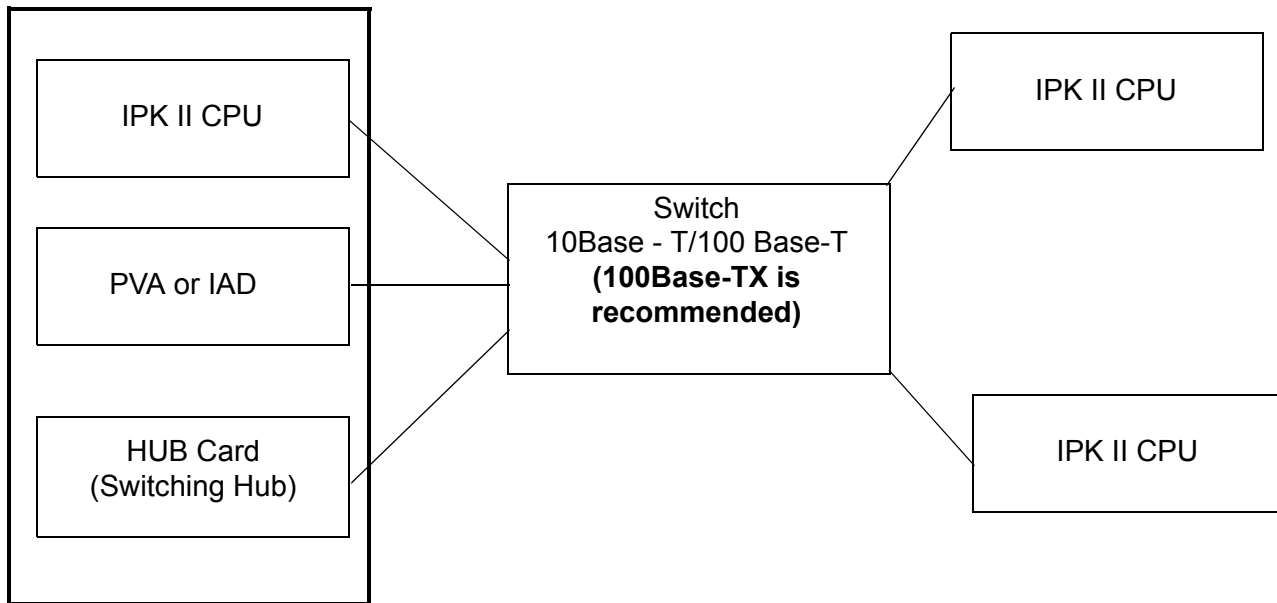
The VoIP supports the following:

- Trunks: IP CCIS, H.323 and SIP Trunks Compressions of G.711, G.723 and G.729
- Stations: MEGACO Compressions of G.711 and G.729

Using LANs

Using a LAN setup (local area network) with the Electra Elite IPK II system complies with the Ethernet standard (10Base-T/100Base-TX).

3. The voice quality of VoIP depends on variables such as available bandwidth, network latency and Quality of Service (QoS) initiatives, all of which are controlled by the network and internet service providers. Because these variables are not in NEC control, it cannot guarantee the performance of the user's IP-based remote voice solution. Therefore, NEC recommends connecting VoIP equipment through a local area network using a Private IP address.



To connect a telephone to a LAN connection, the system allows using an Electra Elite IPK II digital 16-button multiline terminal (called an Electra Elite IPK II IP Phone), an Electra Elite IPK II digital multiline terminal with an IP Adapter installed.

If connecting a LAN to a WAN (wide area network), follow the instructions included with the ADSL modem or gateway device.

IP Address

Equipment/devices used in the Electra Elite IPK II LAN setup must have an IP address assignment. An IP address assigns a unique address for each device. There are two types of IP addresses: Private and Global. A Private IP address is not accessible through the internet – a Global IP address can be accessed through the internet.

With a Private IP address, with equipment that does not access the internet directly, addresses can be assigned to the equipment in Class A, B or C by assigning a number in the class range of numbers.

Class	Allowed IP Address	Recommended Environment
A	10.0.0.0 --- 10.22.255.255	Large Scale Network
B	172.16.0.0 --- 172.31.255.255	Mid Scale Network
C	192.168.0.0 --- 192.168.255.255	Small Scale Network

With a Global IP Address, connected equipment can be accessed through the internet, so each address must be unique. To avoid a conflict, the addresses are controlled by ARIN (American Registry for Internet Numbers). To obtain a Global IP Address, contact ARIN or apply with your local internet service provider (ISP).

The first of three groups of numbers (depending on the subnet mask) identifies the network on which your computer is located. The remaining group(s) of numbers identify your computer on that network.

Subnet Mask

As the IP Address includes information to identify both the network and the final destination, the Subnet Mask is used to set apart the network and destination information.

The default subnet masks are:


Class	Default Subnet Mask
A	255.0.0.0
B	255.255.0.0
C	255.255.255.0

In the above table, the Subnet Mask contains four groups of numbers. When a group contains the number 255, this tells the router to ignore or mask that group of numbers in the IP address as it is defining the network location of the final destination. So, for example, if the IP Address were: 172.16.0.10 and the Subnet Mask used was Class B (255.255.0.0), the first two groups of numbers (172.16) are ignored after they reach the proper network location. The next two groups (0.10) are the final destination in the LAN to which the connection is to be made.

DHCP

DHCP (Dynamic Host Configuration Protocol) assigns a dynamic IP Address. Network control may be easier with DHCP as there is no need to assign and program individual IP Addresses for the LAN equipment. To use a dynamic IP Address, a DHCP server must be provided. The Electra Elite IPK II system allows using DHCP.

When equipment connected to the LAN (the DHCP client) is requesting an IP Address, it searches the DHCP server. When the request for an address is recognized, the DHCP server assigns an IP Address, Subnet definition, and the IP Address of the router, etc., based on the system programming.

 *The IPK II CPU II must always have a static IP address. This address is set in Program 10-12-01 : CPU II Network Setup – IP Address (default: 172.16.0.10).*

Routers

When purchasing a router to use with the Electra Elite IPK II IP feature, the minimum requirements would be that it provide 10/100 Switch and QoS. Currently, VoIP Media Gateway (MG16) application for the Electra Elite IPK II MEGACO telephones can communicate over NAT firewall router. For the best performance, NEC recommends connecting the Elite IPK II CPUII and MG16 10/100 Switch to make it less susceptible to network collision.

The priority control feature prevents RTP packet loss. If a WAN is used for VoIP only and the bandwidth is wide enough for the VoIP channel, the QoS feature might not be required.

VoIP Bandwidth Calculation

A digital telephone system converts an analog voice signal into a stream of bits expressed in K bits per second (K means 1000). For standard PCM digital encoding, this stream of bits is 64 K bits per second. This is 64 K bps in each direction (transmit and receive).

To improve transmission efficiency, this bit stream is compressed using a standard compression algorithm such as G.729. The result is still a bit stream, but with fewer bits per second. For example, G.729 reduces the 64 K bits per second to a bit stream of 8 K bits per second.

This bit stream is then divided into chunks (called Voice Samples or Voice Frames) that can be placed in packets for transmission over a data network.

This reduced bit stream is examined repeatedly in fixed time intervals. This examination time is called the Voice Frame Interval. This is the time used to collect the bits for one Voice Frame. The Voice Frame Interval is expressed in milliseconds (ms). A millisecond is one thousandth of a second.

IP Hardware

ETUs:

- IPK II CPUII** – Signals the gateway with VoIP communication.
- MG16 Application** – VoIP ETU provides a 16-channel voice packet gateway unit and works as a media gateway. VoIP multiline terminal-to-VoIP multiline terminal can talk without using any DSP resources using the peer-to-peer feature.
- HUB Card** – Provides an 8-port switching hub for use with the Electra Elite IPK II system.
- IAD(8)-U10 ETU** – VoIP ETU provides eight channels for supporting MEGACO Stations on an IP CCISoIP network.
- PVA(X)-U()CCISoIP Package** – VoIP ETU provides 24 ports for supporting CCISoIP Network.

Terminals:

- ❑ **IP Phone** – An Electra Elite IPK II ITH 4D/8D/16D-2/3 button multiline IP telephone.
- ❑ **IP Adapter** – An IP adapter connected to a multiline terminal.

The telephone with the adapter installed is also called the Dterm IP telephone.

Power must be supplied to the IP Phone or the Electra Elite IPK II multiline terminal with IP adapter using either a local or central power supply. If a power outage occurs, the IP telephones do not work unless the telephones are plugged into a UPS (uninterruptible power supply).

When the first IP telephone is plugged in, the system automatically assigns the next three consecutive station ports available as IP ports. The next three IP telephones installed use this group of ports. When the fifth IP telephone is connected, the next three consecutive station ports available are assigned as IP ports.

Switches/Hubs for PoE

If PoE (power over ethernet) is used to eliminate the separate power adapters, due to the power requirements, a separate power source is suggested.

Each IP telephone consumes one port. When automatically selected, the port number ranges from 1~256.

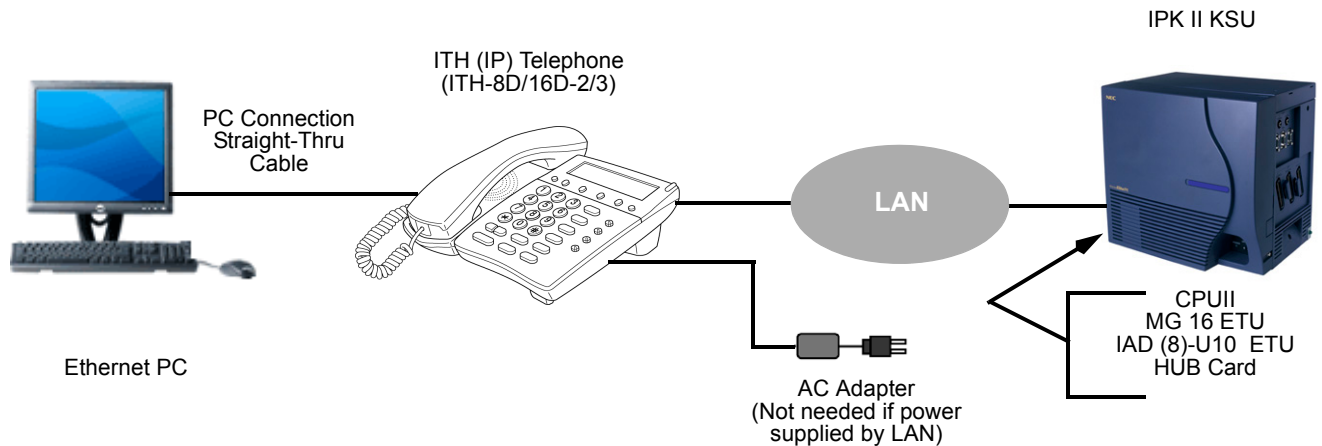


Figure 2-26 IP Telephone Configuration

Table 2-45 Electra Elite IPK II VoIP Specifications

Category	Feature	Note
IP Address	DHCP Server	IPK II CPUII
	DHCP Client	MEGACO IP Phone
QoS	802.1p/1q	
	L3 QoS (ToS)	Diffserv/ToS
Maintenance	HTTP Server	CPUII
VLAN	VLAN Tag and port-based VLAN	
VoCoder	G.711 μ -law/A-law	
	G.729a	
Jitter Buffer Size	Set by System Programming	
RTP Length	Set by System Programming	
Echo Canceller Tail Size	Set by System Programming	
Level Adjustment	Set by System Programming	
Protocol	H.248	
IP Phone	MEGACO Station	MEGACO Station

Conditions

- Electra Elite IPK II systems can be connected together through a LAN using VoIP.
- As the IP telephones require a VoIP resource when receiving a page, the number of IP telephones allowed to receive a page should be minimized to prevent a lack of resources.

Default Setting

Disabled

Related Features

None

System Availability

Terminals

ITH-4D/8D/16D-2/3 TEL

IP Adapter

Required Component(s)

PVA(X)-U() ETU

Programming

Programming – VoIP MG16 ETU

These programs are always required with the VoIP MG16 feature.

Programming – Static IP Address

In addition to the system programming for the MG 16 ETU, the IP telephone also has options which must be programmed from the telephone itself. Be sure to set both the system and telephone options.

- ➔ **10-12-01 through 10-12-03 CPUII Network Setup**
- ➔ **84-05-01 VoIP IP Address – IP Address**
For each ETU, enter the IP address for the **MG 16** ETU (default: slot 1 = 172.16.0.20, slot 2 = 172.16.0.21, etc.). The IP address should be increased according to the number of VoIP MG16 ETUs.
- ➔ **84-05-02 VoIP IP Address – LAN Interface**
For each MG16 ETU, enter the Mode/speed setting (default: Auto).
- ➔ **84-05-03 VoIP IP Address – Master/Slave**
For each MG16 ETU, enter the MASTER/Slave setting.
✎ There can be one MG16 only assigned as MASTER and additional MG16 ETUs must be set as SLAVE.
- ➔ **84-05-04 VoIP IP Address – Subnet Mask**
For each ETU, enter the Subnet Mask used by the MG16 (default: 255.255.0.0).
- ➔ **84-05-05 VoIP IP Address – Default Gateway**
For each ETU, enter the default Gateway used by the MG16 (default: 0.0.0.0).

84-06-01 VoIP Info – RTP Port Number

For each **MG 16** ETU, enter the RTP port number (default: 10020).

Programming – Electra Elite IPK II IP Multiline Terminal or Electra Elite IPK II Multiline Terminal with IP Adapter, Static IP Address

Setting the IP Address on an Electra Elite IPK II IP Phone or Electra Elite IPK II Multiline Terminal with IP Adapter Using a Static IP Address

- On the Electra Elite IPK II IP multiline terminal, enter the telephone program mode by pressing **Hold Conf * #**.
 - This enters the IP User Menu program mode to select the settings for the individual telephone.*
 - To enter the telephone program mode, the display must show the time, error, or Connecting message prior to pressing **Hold Conf * #**.*
- Press **1** (Network Settings).

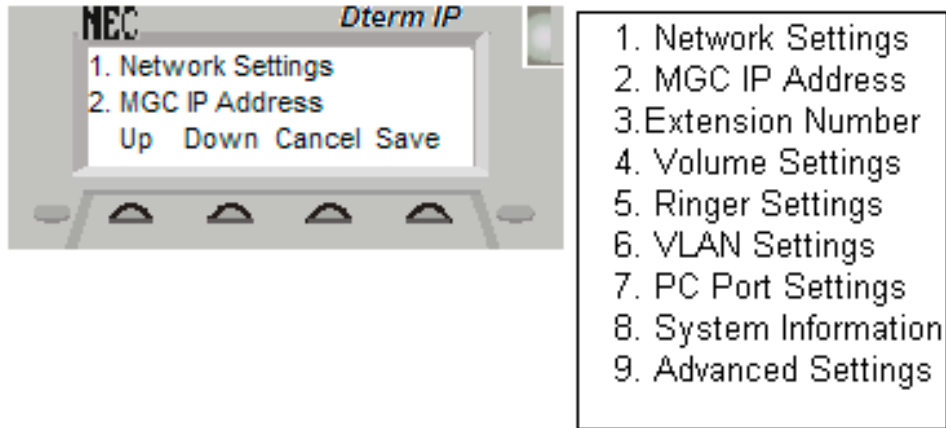


Figure 2-27 Selecting Network Settings Menu

- Select **1** for DHCP mode. If not flashing, press **1** to disable the DHCP mode. Press the **OK** softkey.
 - The flashing item is the current selection.*

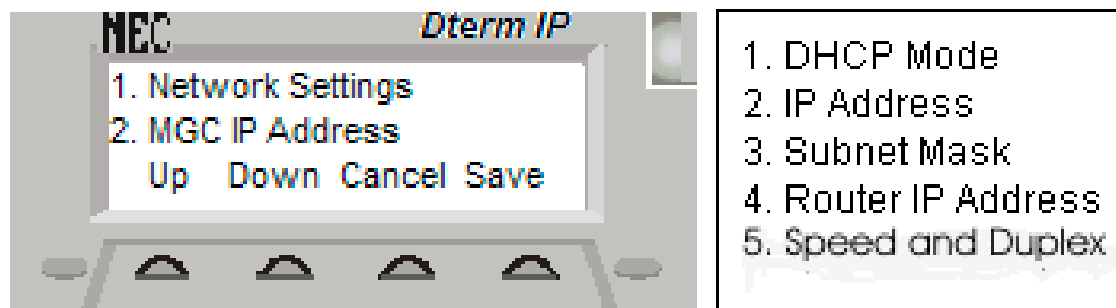








Figure 2-28 Selecting DHCP Mode

4. Access Network Settings and press **2** to enter the IP address for the telephone. The IP address must be unique to each telephone, but the range is dependent on system programming (example: IP Multiline Terminal 1 = 172.16.0.11, IP Multiline Terminal 2 = 172.16.0.12, etc.). The * moves to the next set of digits (example: 172*16*0*10). Press the OK softkey.
 -  For on-site LANs, the IP address must be unique to each telephone but the range depends on system programming (example: IP Phone 1 = 172.16.0.11, IP Phone 2 = 172.16.0.12, etc.). When telephones are off-site (remote LAN), these settings depend on the remote LAN settings – not the telephone system settings.
 -  The BK softkey backs up a character to correct an entry.
5. Access Network Settings and press **4** to enter the Default Gateway address.
 -  For on-site LANs this is the entry in Program 10-12-03. When telephones are off-site (remote LAN), these settings depend on the remote LAN settings - not the telephone system settings.
6. Access Network Settings and press **4** to enter the Subnet Mask.
 -  For on-site LANs this is the entry in Program 10-12-02. When telephones are off-site (remote LAN), these settings depend on the remote LAN settings – not the telephone system settings.
7. Press **3** to enter the IP telephone extension number.
8. Press **2** to enter the MGC IP address (Media Gateway Controller). Enter the IP address used in Program 84-05-01 (default: Slot 1 – 172.16.0.20). The * moves to the next set of digits (example: 172*16*0*10). Press the softkey **OK**.
 -  The BK softkey backs up a character to correct an entry.
9. Press the **SAVE** softkey to save the entries and return the telephone to idle.
 -  Pressing CANCEL instead ignores the changes made and returns the multiline terminal to idle.


Programming – Using the Electra Elite IPK II DHCP




- **10-13-01 In-DHCP Server Setup – DHCP Server Mode**
Enable (1)/Disable (0) the system ability to use the built-in DHCP server. The system must be reset for the change to take effect.
(default: 0)
- **10-13-02 In-DHCP Server Setup – Lease Time**
Enter the lease time of the IP address to a client . The system must be reset for the change to take effect.
(default: 0 days, 0 hours, 30 minutes)
- **10-13-04 In-DHCP Server Setup – Number of Networks**
Select the number of networks in use on the system (0=Single Network, 1=Divide Same Network). The system must be reset for the change to take effect.
(default: 0)
- **10-14-01 Managed Network Setup – The range of the IP address to lease**
Set the range of IP addresses available to be used by the DHCP server for Scope 1.
(default: Minimum = 172.16.0.100, Maximum = 172.16.5.254. Scopes 2~10 = 0.0.0.0)
- **10-15-01 Client Information Setup – The IP address should be assigned out of the scope range of Program 10-14**
Set the IP and MAC addresses for each client (up to 10) when the DHCP server needs to assign a fixed IP address to clients.
(default: MAC:00-00-00-00-00-00, IP:0.0.0.0)
- **10-16-01 Option Information Setup – Router**
Set the router address for the DHCP server to the client.
[default: Code 3 (fixed) IP address = 0.0.0.0]
- **10-16-02 Option Information Setup – DNS Server**
If required, set the DNS server address for the DHCP server to the client.
[default: Code 6 (fixed) IP address = 0.0.0.0]
- **10-16-03 Option Information Setup – TFTP Server (Version 1600 and 2100 or higher required)**
[default: Code 66 (fixed) IP address = 0.0.0.0]
- **10-16-05 Option Information Setup – MGC**
Set the MGC IP address for the DHCP server to the client.
[default: Code 129 (fixed) IP address = 172.16.0.10]

Programming, Electra Elite IPK II IP Phone or Electra Elite IPK II Multiline Terminal with IP Adapter – DHCP


Setting the IP Address on an Electra Elite IPK II IP Phone or Electra Elite IPK II multiline terminal with IP Adapter Using a DHCP Server

1. On the Electra Elite IPK II IP multiline terminal, enter the telephone program mode by pressing **Hold Conf * #**.

 *This enters the IP User Menu program mode to select the settings for the individual telephone.*

2. Access Network settings and press **1** to enter the DHCP mode. If not flashing, press **2** to enable the DHCP mode. Press the **OK** softkey.
 -  *The flashing item is the current selection.*
3. The entries for items 2, 3 and 4 are automatically set by the DHCP server.
4. The MGC IP Address is automatically set by the DHCP Server.
5. Press **3** to enter the IP telephone extension number.
6. Press the **SAVE** softkey to save the entries and return the telephone to idle.
 -  *If a user's LAN is already operating with a DHCP server, disable the Electra Elite IPK II DHCP server. Either DHCP server can be used, but not both.*
 -  *When a user's DHCP server is operating, make sure to define Program 10-16.*

Programming – Using an External DHCP

-  *If a user's LAN is already operating with a DHCP server, disable the Electra Elite IPK II DHCP server. Either DHCP server can be used, but not both.*

Operation

None

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Voice Response System (VRS)

Enhancements

Version 2000 or higher only supports the ACD Delay Announcement function of VRS.

For more details regarding ACD Delay Announcements using In-Mail, refer to the ACD feature.

Description

The DSP daughter board provides the option for the Voice Response System (VRS) which gives the system voice recording and playback ability. The VRS CompactFlash card provides up to 48 system messages (General Message, Automated Attendant greetings, ACD messages, and the 900 Preamble).

- General Message** – provides a recorded message to which any user can listen.
- Automated Attendant (Operator Assistance)** – answers incoming calls, plays a greeting to the caller and then lets the caller directly dial a system extension.
- ACD Messages** – provides announcement and overflow messages for ACD groups.
- Transfer to the VRS** – allows any extension user to Transfer their outside call to the VRS.
- Voice Prompting Messages** – plays call and feature status messages to users.
- 900 Preamble** – alerts callers using 900 lines of the cost and features of the pay-per-call service.
- Time, Date and Station Number Check** – lets a multiline terminal extension user quickly hear a recording for the time, date, or the extension number.

VRS Messages

The VRS allows you to record up to 48 VRS messages. You allocate these messages for Automated Attendant greetings, the General Message, ACD messages and the 900 Preamble message. The total storage time for all messages is approximately 45 minutes. The maximum duration for any message is two minutes – this is not programmable. VRS messages are stored on a Compact Flash drive, and do not require battery back up.

Any on-premise extension caller can listen, record and erase VRS Messages (unless restricted in programming). DISA and DID callers can listen and record VRS messages (unless restricted in programming).

General Message

A General Message is a recorded message available to all callers. A General Message typically contains important company information that all employees should hear. To hear the General Message, an employee can go to any multiline terminal and press 4 (for General Message). You can restrict the ability to record the General Message in an extension Class of Service. This allows you to give recording ability to the System Administrator or Communications Manager, for example, but not any employee. The Message Waiting LED at each telephone flashes when a new General Message is recorded. After the extension user listens to the message, the Message Waiting LED goes out.

Park and Page

When an extension user is away from their telephone, Park and Page can let them know when they have a call waiting to be answered. The Personal Greeting and Park & Page options can have up to 200 total messages (note that the Park & Page feature uses two messages). To enable Park and Page, the user records a Personal Greeting along with an additional Paging announcement. Park and Page then answers an incoming call and plays the Personal Greeting to the caller. The caller then listens to Music on Hold (if available) while the system broadcasts the recorded Paging announcement. When the extension user hears the Page, they can go to any telephone and use Directed Call Pickup to intercept the call. Refer to [Call Forwarding – Park and Page on page 2-145](#), [Paging, External on page 2-769](#), [Paging, Internal on page 2-775](#), and [Park on page 2-779](#).

Automated Attendant (Operator Assistance)

Automated Attendant automatically answers outside calls, plays a recorded greeting and then lets the outside callers directly dial system extensions, Department Calling Groups and Voice Mail. Automated Attendant provides immediate answering and routing of outside calls without the need for an operator or dispatcher. Automated Attendant provides:

Single Digit Dialing

Single Digit Dialing allows Automated Attendant callers to press a single digit to dial extensions, Department Calling Groups, or Voice Mail. For example, your Automated Attendant can greet calls with, *“Thank you for calling. To place an order, dial 1. To check on an existing order, dial 2. To speak with an operator, dial 0.”* You can set up single digit dialing for each VRS Message programmed to answer outside calls via the Automated Attendant. This allows you to set up day/night/holiday greetings or unique greetings for each incoming trunk. (Keep in mind that, with a default system, if you assign destinations to digits 3, 4 and 5, outside callers cannot dial system extensions.)

Simultaneous Call Answering

With VRS installed, the Automated Attendant can answer up to 16 calls simultaneously.

Flexible Routing

The outside caller can directly dial any system extension, Department Calling Group, or Voice Mail. If the caller dials a busy extension, Automated Attendant allows them to dial another extension or wait for the busy extension to become free.

Automatic Overflow

Automatic Overflow automatically redirects a call if it cannot go through. This can happen if all VRS ports are busy, if the called extension does not answer, or if the caller misdials or waits too long to dial. (This occurs if the caller is using a dial pulse telephone.) When the call overflows, it rings a designated Ring Group or the Voice Mail system.

 Programmable Automated Attendant Greetings

You can record a different greeting for each trunk answered by the Automated Attendant. The greetings can be different in the day, at night or on holidays or weekends. You can also have a special greeting if the caller misdials. You record the greetings just the way you want. For example, *“Dial the 3-digit extension number you wish to reach, dial 500 for Sales or dial 600 for Customer Service.”* When assigning and recording Automated Attendant greetings, you can choose among the 48 VRS messages.

VRS Waiting Message

Using VRS Waiting Message, the system can automatically answer an incoming trunk call first (either a normal trunk or one designated for a department group) to let the outside caller hear a recorded message when the call is not answered in a programmed time. With this feature, the call keeps ringing at the same destination until it is answered or until other programming, takes affect.

This feature can use up to two messages for an incoming call and the duration between the messages is programmable. These messages are repeated and, between these messages, either ring back tone or Music on Hold can be played.

This feature has two different modes:

 Permanent Mode



This mode sets the feature using system programming and is available for the following calls.

- Normal Incoming Call
When the call is not answered or a user presses the VRS Waiting Message function key, this feature is initiated. The waiting message is played until another no-answer program (e.g. transfer to another incoming ring group or disconnect) takes affect.
- Designated Call for the Department Group
When a department group receives a call from a DID, DIL, DISA or E&M trunk and all terminals in the group are busy, the call is put in a queue and VRS Waiting Message is also initiated. The waiting message is played until another no-answer program (e.g. transfer to another incoming ring group or disconnect) takes affect or a terminal becomes available to receive the department call.

 Manual Mode

This mode can be programmed by pressing the VRS Waiting Message function key from a multiline terminal to set this feature for each incoming ring group. This mode can be used for normal incoming calls only.

The following programs define the VRS Waiting Message feature and the trunk overflow:

- 11-10-20: Service Code Setup (for System Administrator) – VRS - Record/Eraser Message
- 15-07: Programmable Function Keys
Automatic Answer with Delay Message Setup (Function Number 52)
 -  *Function Key 52 can be used to enable the VRS Waiting Message feature when Program 22-01-10 is set to 1 (Changed by Manual Operation).*
- Automatic Answer with Delay Message Start (Function Number 53)
 -  *Function Key 53 can be used to play the VRS Waiting Message immediately when Function Key 53 + the ringing Trunk Appearance Key are pressed.*
- 20-07-13: Class of Service Options (Administrator Level) – VRS Record (VRS Msg Operation)
- 20-15-11: Ring Cycle Setup – VRS Waiting Message Incoming Call
- 22-01-04: System Options for Incoming Calls – DIL No Answer Recall Time
- 22-01-08: System Options for Incoming Calls – DID Pilot Call No Answer Timer
- 22-01-10: System Options for Incoming Calls – VRS Waiting Message Operation
- 22-01-11: System Options for Incoming Calls – VRS Waiting Message Interval Time
- 22-08-01: DIL/IRG No Answer Destination
- 22-14-01~07: VRS Delayed Message for IRG
- 22-15-01~07: VRS Waiting Message for Department Group
- 25-07-02: System Timers for VRS/DISA – VRS/DISA No Answer Time
- 25-07-03: System Timers for VRS/DISA – Disconnect after VRS/DISA retransfer to IRG

Transfer to the VRS

Any extension user can Transfer outside call to the VRS. This lets their caller take advantage of the Automated Attendant extensive routing abilities. To Transfer the call, the user places the call on Hold, dials the unique VRS service code (set up in system programming : default 782), and hangs up.

Voice Prompting Messages

The VRS feature provides the system with Voice Prompting Messages. These Voice Prompting Messages tell the extension user the status or progress of their call. For example, if a user calls extension 300 when it is busy, they hear, “*Station 300 is unavailable, please dial a new station or dial 750 to wait.*”

The following table shows the available Voice Prompting Messages.

Table 2-46 Voice Prompting Messages

Message No.	Message	This message will play when . . .
1-00	This is station	A user dials 6 for the extension number.
1-01	Station	A user dials 6 for the extension number.
1-02	Is busy, for callback dial	A user is calling a busy extension.
1-03	All lines are busy, for callback dial	A user dials 9 or 704 (+ trunk group) and all trunks are busy.
1-04	Please do not disturb	A user calls an extension that has enabled Do Not Disturb.
1-05	Please hold on, all lines are busy, your call will be answered when a line becomes free.	ACD message – refer to the Electra Elite IPK II Automatic Call Distribution Manual.
1-06	Please hold on, your call is being rerouted	Call Forwarding Off-Premise is rerouting your call.
1-07	The lowest cost line is busy, please wait for the next one.	ARS tries to reroute the user's call and the least costly route is busy.
1-08	The number you have dialed is not in service.	User dials a Service Code that Class of Service prevents.
1-09	You have a message.	An extension user has a Message Waiting to which they have not responded.
1-10	You have a message.	An extension user has a Message Waiting to which they have not responded.
1-11	Your calls have been forwarded.	An extension user has forwarded their calls.
1-12	Vacant number	An extension user has dialed an extension that does not exist.
1-13	Is unavailable	An outside caller dials an extension through the Automated Attendant and the extension is busy.
1-14	Please dial a new station	
1-15	Or dial	
1-16	To wait	
1-17	To leave your number	
1-18	Dial # to call you back at	

Table 2-46 Voice Prompting Messages (Continued)

Message No.	Message	This message will play when . . .
1-19	Enter your area code and telephone number	An outside caller dials an extension through the Automated Attendant and the extension is busy.
1-20	Please enter your password	
1-21	Please enter an account code	A user tries to place a trunk call and Forced Account Codes are enabled.
1-22	Please start recording	A user dialed the code to record a VRS message.
1-23	Recording finished	A user is recording a VRS message and they have exceeded the maximum allowed recording length.
1-24	Audio file is full	No more space is available in the VRS for storing messages.
1-25	To listen dial	A user is trying to record a VRS message and the recording already exists.
1-26	To erase dial	
1-27	To re-record dial	
1-28	To save dial	
1-29	To leave a message	
1-30	Just a moment	
1-31	Hello	
1-32	Thank you	
1-33	Good-bye	
2-00	Oh	A user dials 6 for the extension number or 8 for the time.
2-01	Dial	
2-02	Star	
2-03	Pound	
2-04	Zero	

Table 2-46 Voice Prompting Messages (Continued)

Message No.	Message	This message will play when . . .
2-05	One	A user dials 6 for the extension number, 8 for the time and date or as part of a spoken code (e.g., 714).
2-06	Two	
2-07	Three	
2-08	Four	
2-09	Five	
2-10	Six	
2-11	Seven	
2-12	Eight	
2-13	Nine	
2-14	Ten	
2-15	Eleven	
2-16	Twelve	
2-17	Thirteen	
2-18	Fourteen	
2-19	Fifteen	
2-20	Sixteen	
2-21	Seventeen	
2-22	Eighteen	
2-23	Nineteen	
2-24	Twenty	
2-25	Thirty	
2-26	Forty	
2-27	Fifty	
2-28	Sixty	
2-29	Seventy	
2-30	Eighty	
2-31	Ninety	
2-32	Hundred	

Table 2-46 Voice Prompting Messages (Continued)

Message No.	Message	This message will play when . . .
2-33	Thousand	
2-43	Message	
2-44	Messages	
2-64	January	
2-65	February	
2-66	March	
2-67	April	
2-68	May	
2-69	June	
2-70	July	
2-71	August	
2-72	September	
2-73	October	
2-74	November	
2-75	December	
2-76	Sunday	A user dials 8 for the date.
2-77	Monday	
2-78	Tuesday	
2-79	Wednesday	
2-80	Thursday	
2-81	Friday	
2-82	Saturday	
2-83	The date is	A user dials 8 for the date.
3-04	The time is	A user dials 8 for the time.
3-05	AM	
3-06	PM	

900 Preamble

If the system has trunks that are part of a 900 (caller paid) service, the VRS can automatically play a recorded message when a user answers the call. This recorded message should describe the 900 service features and cost. The 900 Preamble ensures that the caller is always aware that they have accessed a 900 pay-per-call service. A system user cannot converse with the caller until the preamble message ends. If the caller hangs up before the message completes, they are not charged for the call. If the caller waits for the message to end, they can talk to a system user and call charging begins. The system answers as many 900 calls as there are available VRS ports. If a 900 call comes in when all VRS ports are busy, the call does not appear on an extension until a VRS port is available.

You can also use the 900 Preamble message to set up an *Auto-Answer with Greeting* application. When a receptionist answers a call, the VRS can play a preamble message such as, "Welcome to ABC Company. How can I help you?" When the caller replies, the receptionist answers, "One moment please," and quickly extends the call to the desired party. This ensures that all incoming calls are answered quickly, courteously and consistently.

Time, Date and Station Number Check

If the system has a DSP daughter board installed for VRS, any multiline terminal user can determine the time, date or the extension number while their telephone is idle (on-hook). The time and date check saves the user time since they do not have to look for a clock or calendar. Hearing the extension number conveniently identifies non-display multiline terminals. To find out their extension number, the user can press 6 (for **Number**). To listen to the time and date, the user can press 8 (for **Time/Date**).

Available with 64-Port Basic CPUII

The VRS feature is available with the 64-port Basic CPUII (no feature Upgrade PAL chip required).

The VRS feature requires a DSPII-U10 Unit attached to the CPUII with the optional VRS flash card installed. Although the DSPII-U10 Unit is recognized for this feature, it provides no additional tone resources (DTMF receivers, Caller ID receivers, or call progress tone detection).

Conditions

- VRS record time is fixed at two minutes and cannot be changed.
- The Automated Attendant (VRS) can answer up to 16 calls simultaneously.
- If Synchronous Ringing is enabled, the Preamble message cannot be used.
- Program 41-08-03: ACD Overflow Options – Delay Announcement Source Type is not supported in **Version 1600**.
- Program 41-19-01~Program 41-19-08: Voice Mail Delay Announcement, is not supported in **Version 1600**.

Default Setting

Disabled

System Availability

Terminals

Not applicable

Required Component(s)

DSPII-U10 Unit

Programming

- ➔ **11-10-20 Service Code Setup (for System Administrator) – VRS - Record/Erase Message**
Define the service code to use to record or erase a VRS message (default: 616).
- ➔ **11-10-21 Service Code Setup (for System Administrator) – VRS - General Message Playback**
Define the service code to use to playback the general message (default: 611).
- ➔ **11-10-22 Service Code Setup (for System Administrator) – VRS - Record or Erase General Message**
Define the service code to use to record or erase a general message on the VRS (default: 612).
- ➔ **11-12-54 Service Code Setup (for Service Access) – VRS Routing for ANI/DNIS**
Define the service code to use when setting up ANI/DNIS Routing to the VRS Automated Attendant. Using the Transfer feature, this also allows a call to be transferred to the VRS (default: 782).
- ➔ **15-07-01 Programmable Function Keys**
For the VRS Waiting Message feature, assign the VRS Incoming Call Queuing Setup key (code 52 + ring group #) to manually enable the feature.
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to extensions.
- ➔ **20-07-13 Class of Service Options (Administrator Level) – VRS Record (VRS Msg Operation)**
Enable (1)/Disable (0) the ability to record, erase and listen to VRS messages.
[default: 0 (COS 1~14) , 1 (COS 15)]

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- ➔ **20-07-14 Class of Service Options (Administrator Level) – VRS General Message Play**
Enable (1)/Disable (0) the ability to dial Service Code 611 and listen to their General Message.
[default: 0 (COS 1~14) , 1 (COS 15)]
 - ➔ **20-07-15 Class of Service Options (Administrator Level) – VRS General Message Record/Delete**
Enable (1)/Disable (0) the ability to dial Service Code 612 and record, listen to and erase the General Message.
[default: 0 (COS 1~14) , 1 (COS 15)]
 - ➔ **20-11-15 Class of Service Options (Hold/Transfer Service) – VRS Personal Greeting (Message Greeting)**
Enable (1)/Disable (0) the ability to dial Service Code 616 to record, listen to or erase a Personal Greeting. This option also affects Park and Page.
(default: 1)
 - ➔ **20-13-23 Class of Service Options (Supplementary Service) – Display the Reason for Transfer**
Enable (1)/Disable (0) the ability to display the reason (Call Forward, Busy, No Answer or DND) a VRS, DID, DISA, or ISDN call is being transferred to their extension.
(default: 0)
 - ➔ **20-15-11 Ring Cycle Setup – VRS Waiting Message Incoming Call**
Set the ring cycle callers hear when the VRS Waiting Message feature is used.
(default: 6)
 - ➔ **21-01-02 System Options for Outgoing Calls – Intercom Interdigit Time**
Automated Attendant callers must dial digits in this time (0~64800 seconds). If the caller dials too slowly, they hear busy tone and must hang up and dial again
(default: 10)
 - ➔ **22-01-10 System Options for Incoming Calls – VRS Waiting Message Operation**
Determine whether the VRS Waiting Message is automatically (0) or manually (1) set.
(default: 0)
 - ➔ **22-01-11 System Options for Incoming Calls – VRS Waiting Message Interval Time**
For VRS Waiting Message, determine the time (0~64800 seconds) between the VRS messages.
(default: 20)
 - ➔ **22-02-01 Incoming Call Trunk Setup**
For each Night Service mode, enter 1 if trunk should be automatically answered by VRS Automated Attendant.
[default: 0 (Normal)]
 - ➔ **22-04-01 Incoming Extension Ring Group Assignment**
Assign extensions to the Ring Group programmed in PRG 25-03 and PRG 25-04 that should receive the rerouted calls.
(default: Extensions 101~108 answer Ring Group 1 Calls, No other extensions ring these calls)
 - ➔ **22-14-01 VRS Delayed Message for IRG – 1st Waiting Message Start Time**
For each Ring Group, set the time the system waits before playing the first message (0~64800 seconds). This time is also used for VRS Waiting Message.
(default: 0)
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- ➔ **22-14-02 VRS Delayed Message for IRG – 1st Waiting Message Number**
For each Ring Group, select the message number to be played as the first message (0~48). This program is also used for VRS Waiting Message.
[default: 0 (No Message)]
 - ➔ **22-14-03 VRS Delayed Message for IRG – 1st Waiting Message Sending Count**
For each Ring Group, set the number of times the first message is played (0~255). This program is also used for VRS Waiting Message.
(default: 0)
 - ➔ **22-14-04 VRS Delayed Message for IRG – 2nd Waiting Message Number**
For each Ring Group, select the message number to be played as the second message (0~48). This program is also used for VRS Waiting Message.
[default: 0 (No Message)]
 - ➔ **22-14-05 VRS Delayed Message for IRG – 2nd Waiting Message Sending Count**
For each Ring Group, set the number of times the first message is played (0~255). This program is also used for VRS Waiting Message.
(default: 0)
 - ➔ **22-14-06 VRS Delayed Message for IRG – Tone Kind at Message Interval**
For each Ring Group, determine what the caller hears between messages (0 = Ringback Tone, 1 = MOH, 2 = BGM). This program is also used for VRS Waiting Message.
(default: 0)
 - ➔ **22-14-07 VRS Delayed Message for IRG – Disconnect Time After the end of VRS Waiting Message**
For each Ring Group, set the time (0~64800 seconds) the system waits after playing the VRS message before disconnecting the call. To prevent the call from disconnecting, set this option to 0. This program is also used for VRS Waiting Message.
(default: 60)
 - ➔ **22-15-01 VRS Waiting Message for Department Group – 1st Waiting Message Start Time**
For each Department Group, set the time (0~64800 seconds) the system waits before playing the first message . This program is also used for VRS Waiting Message.
(default: 0)
 - ➔ **22-15-02 VRS Waiting Message for Department Group – 1st Waiting Message Number**
For each Department Group, select the message number to be played as the first message (0~48). This program is also used for VRS Waiting Message.
[default: 0 (No Message)]
 - ➔ **22-15-03 VRS Waiting Message for Department Group – 1st Waiting Message Sending Count**
For each Department Group, set the number of times the first message is played (0~255). This program is also used for VRS Waiting Message.
(default: 0)
 - ➔ **22-15-04 VRS Waiting Message for Department Group – 2nd Waiting Message Number**
For each Department Group, select the message number to be played as the second message (0~48). This program is also used for VRS Waiting Message.
[default: 0 (No Message)]
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- ➔ **22-15-05 VRS Waiting Message for Department Group – 2nd Waiting Message Sending Count**
For each Department Group, set the number of times the first message is played (0~255). This program is also used for VRS Waiting Message.
(default: 0)
 - ➔ **22-15-06 VRS Waiting Message for Department Group – Tone Kind at Message Interval**
For each Department Group, determine what the caller hears between messages (0 = Ringback Tone, 1 = MOH, 2 = BGM). This program is also used for VRS Waiting Message.
(default: 0)
 - ➔ **22-15-07 VRS Waiting Message for Department Group – Disconnect Time After the End of VRS Waiting Message**
For each Department Group, set the time (0~64800 seconds) the system waits after playing the VRS message before disconnecting the call . To prevent the call from disconnecting, set this option to 0. This program is also used for VRS Waiting Message.
(default: 60)
 - ➔ **24-02-03 System Options for Transfer – Delayed Call Forwarding Time**
Set the time (0~64800 seconds) a telephone rings before the call reroutes to the programmed destination.
(default: 10)
 - ➔ **25-01-02 VRS/DISA Line Basic Data Setup – DISA User ID**
 - ➔ **25-02-01 DID/DISA VRS Message**
For each Night Service mode, enter 1 at the “Talkie” prompt if trunk should be automatically answered by VRS and the message number the caller should hear (1~48).
(default: 1)
 - ➔ **25-03-01 VRS/DISA Transfer Ring Group With Incorrect Dialing**
Set the destination that Automated Attendant (OPA) calls ring if the OPA caller dials an incorrect extension number. This also sets the options for DISA calls. The system allows Ring Groups or Ring Groups: 1~100
[default: 0 (disconnect)]
 - ➔ **25-04-01 VRS/DISA Transfer Ring Group With No Answer/Busy**
Set the destination that Automated Attendant (OPA) calls ring if the OPA caller dials an extension that does not answer or is busy. This also sets the options for DISA calls. The system allows Ring Groups or Disconnect = 0.
Ring Groups: 1~100
[default: 0 (disconnect)]
 - ➔ **25-05-01 VRS/DISA Error Message Assignment**
For each trunk that is answered by the VRS, enter the VRS message (1~48) the outside caller hears if they dial incorrectly after answer. If you enter 0, the call reroutes according to Program 25-03 and Program 25-04. Make one entry for each Night Service mode.
[default: 0 (No Setting)]

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- ➔ **25-06-01 VRS/DISA One-Digit Code Attendant Setup – Next Attendant Message Number**
Set up single digit dialing for Automated Attendant callers. For each VRS Message programmed to answer outside calls (see Program 25-02-01), specify the digit the Automated Attendant caller dials (1~9, 0, *, #). (Keep in mind that if you assign destinations to digits 3 and 4, outside callers cannot dial system extensions.)
[default: 0 (No Setting)]
 - ➔ **25-06-02 VRS/DISA One-Digit Code Attendant Setup – Destination Number**
Set up single digit dialing for Automated Attendant callers. For each VRS Message programmed to answer outside calls (see Program 25-02-01), specify the destination reached (four digits maximum) when the caller dials the single digit code.
 - ➔ **25-07-02 System Timers for VRS/DISA – VRS/DISA No Answer Time**
If an Automated Attendant caller dials an extension that does not answer, the call waits this time (0~64800 seconds) before rerouting to the Ring Group specified in PRG 25-03 and PRG 25-04. This setting also affects unanswered DISA calls.
(default: 0)
 - ➔ **25-07-03 System Timers for VRS/DISA – Disconnect after VRS/DISA retransfer to IRG**
Set the time (0~64800 seconds) for disconnecting a call after it is re-transferred to a ring group by VRS/DISA.
(default: 60)
 - ➔ **25-08-01 DISA User ID Setup – Password**
Set up password (six digits).
(default: No Setting)
 - ➔ **25-13-01 System Option for DISA – VRS Message Access Password**
Enter the password DISA callers must dial before the system allows them to record, listen to or erase VRS messages.
(default: No Setting)
 - ➔ **31-02-01 Internal Paging Group Assignment – Internal Paging Group Number**
Assign extensions to Internal Paging Groups (0~64).
[default: 0 (No Setting) for IP Station, 1 for TDM Station]
 - ➔ **31-02-02 Internal Paging Group Assignment – Internal All Call Paging Receiving**
Allow (1)/Prevent (0) All Call Internal Paging for each extension.
(default: 0)
 - ➔ **31-04-01 External Paging Zone Group – Paging Group Number**
(default: Speakers 1~8 = Groups 1~8, 9 (CPU1) = Group 1)
 - ➔ **31-07-01 Combined Paging Assignments**
Assign an external paging Group (0~8) to an Internal Paging Zone (0~64) for combined paging. A combined Page simultaneously broadcasts to both the Internal and External Zones.
(default: 1)
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- ➔ **40-07-01 Voice Prompt Language Assignment for VRS**
Select the language to be used for the VRS. Although the system allows this option to be changed in programming, the language changes only if the DSPII-U10 Unit has the firmware which provides the newly selected language.
(default: 1, English)
 - ➔ **40-08-01 Voice Prompt Language Assignment for Mailboxes**
(default: 1, English)
 - ➔ **40-10-01 Voice Announcement Service Option – VRS Fixed Message**
Enable (1)/Disable (0) the system ability to play the fixed VRS messages such as “You have a message.”
(default: 0)
 - ➔ **40-10-02 Voice Announcement Service Option – General Message Number**
Enter the number of the VRS message you want to use for the General Message (01~48). The message you select should not be used as a VRS message.
[default: 0 (No Message)]
 - ➔ **40-10-03 Voice Announcement Service Option – VRS No Answer Destination**
When all VRS ports are busy, incoming DILs and DISA calls wait for the VRS No-Answer Time (Program 40-10-04) and then ring the VRS No Answer Destination Ring Group.
 - ➔ **40-10-04 Voice Announcement Service Option – VRS No Answer Time**
If an extension has Personal Greeting enabled and all VRS ports are busy, a DIL or DISA Call to the extension waits for this time (0~64800 seconds) for a VRS port to become free.
(default: 0)
 - ➔ **40-10-05 Voice Announcement Service Option – Park and Page Repeat Timer (VRS Msg Resend)**
If a Park and Page is not picked up during this time (0~64800 seconds), the paging announcement repeats.
(default: 0)
 - ➔ **40-10-06 Voice Announcement Service Option – Set VRS Message for Private Call Refuse (VRS Msg Private Call)**
Assign the VRS Message number (0~49) used as Private Call Refuse.
[default: 0 (No Message)]
 - ➔ **40-10-07 Voice Announcement Service Option – Set VRS Message for Caller ID Refuse (VRS Msg CID)**
Assign the VRS Message number used as Caller ID Refuse.
[default: 0 (No Message)]
 - ➔ **40-11-01 Preamble Message Assignment**
For each trunk that should have the 900 Preamble option, enter the number of the VRS message (0~48) that is your recorded Preamble Message.
[default: 0 (No Service)].

➔ **47-03-02 IPK II In-Mail Master Mailbox Options – Master Mailbox Number (Version 1600 or higher required)**

The Master Mailbox Number is the same as the Department Group master (pilot) number. Select the Department Group master (pilot) number (up to seven digits using 0~9), associated with the Master Mailbox you are programming.

[default: 0 (No Setting)]

Related Features

Transfer

Operation


VRS Messages

To record a VRS message:

1. Press **Speaker** or lift the handset.
- OR -
At a single line telephone, lift the handset.
2. Dial **616**.
3. Dial **7 (Record)**.
4. Dial the VRS message number you want to record (01~48).
5. When you hear, "Please start recording" followed by a beep, record your message.
6. Press **#** to end recording
- OR -
Hang up to save the message.

To listen to a previously recorded VRS message:

1. Press **Speaker** or lift the handset.
- OR -
At a single line telephone, lift the handset.
2. Dial **616**.
3. Dial **5 (Listen)**.
4. Dial the VRS message number to which you want to listen (01~48).

 You hear the previously recorded message. If you hear a beep instead, no previous message is recorded.

5. Press **#** to hear the message again.

- OR -

To hear another message, dial 5 and then enter the message number (01~48).

- OR -

Hang up.

To erase a previously recorded VRS message:

1. Press **Speaker** or lift the handset.

- OR -

At a single line telephone, lift the handset.

2. Dial **616**.
3. Dial **3** (Erase).
4. Dial the number of the VRS message you want to erase (01~48).
5. Press **Hold** (multiline terminal only) to cancel the procedure without erasing (and return to step 3).

- OR -

Hang up to erase the message.

To record, listen to or erase a VRS message if you call in using DISA:

1. Place call to the system.

 You hear dial tone.

2. After the system answers, dial the DISA password (normally 000000).

 You hear dial tone.

3. Dial **616** and the VRS password.


4. Dial the function you want.


7 = Record

5 = Listen

3 = Erase

5. Dial the message number (01~48), record the message and press **#** to end recording.

 If you dialed 7 to record, you can dial **#** to listen to the message you just recorded.



 If you dialed 5 to listen, you can dial 5 and the message number to hear it again or if you want to Record, listen to or erase another message, go back to step 4.

General Message

To listen to the General Message:





Multiline Terminal Only

Your Message Waiting LED flashes when there is a new General Message. A voice message periodically reminds you.

1. Do not lift the handset or press **Speaker**.
2. Dial **4 (General)**.
- OR -
 1. Lift the handset and dial **611**.
 -  You will hear the General Message.
 -  Normally, your MW LED goes out. If it continues to flash, you have unanswered Message Waiting requests or new messages in your Voice Mail mailbox.

To record, listen to or erase the General Message:

1. Press **Speaker** or lift the handset.
- OR -

At single line telephone, lift the handset.
2. Dial **612**.
3. Dial the function you want.
7 = Record
5 = Listen
3 = Erase
 -  If you dialed 7 to record, press **#** to end the recording.
 -  If you dialed 5 to listen, you can dial 5 to listen to the message again.
 -  To Record the General Message again, go back to step 1.
 -  If you dialed 3 to erase the General Message, you must go to step 4 (hang up). To cancel without erasing on a multiline terminal, press **HOLD** instead and go back to step 1.
4. Hang up when you are done.

Time, Date and Station Number Check


To check the extension number of any multiline terminal:

1. Do not lift the handset or press **Speaker**.
2. Dial **6** for extension number.

To check the system time and date from any multiline terminal extension:

1. Do not lift the handset or press **Speaker**.
2. Dial **8** for time and date.

900 Preamble**To answer a 900 Preamble call:**

1. Answer the ringing call.
 -  *The line key or Call Appearance (CAP) key turns solid red as the system plays the preamble to the caller.*
2. When you hear two beeps and the line key turns green, converse with the caller.

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Volume Controls

Description

Each multiline terminal user can control the volume of incoming ringing, splash tone, Paging, Background Music, Handsfree and your handset. Multiline terminals consolidate all adjustments into the volume buttons. Press the VOLUME ▲ or VOLUME ▼ to adjust the volume level for whichever feature is active (outside call, ICM, ICM ringing, paging, etc.). Press these keys when the telephone is idle to adjust the contrast level of the telephone display. The users should set the volumes for their most comfortable levels.

Conditions

- The contrast is not adjustable when the telephone has background music enabled.
- Multiline terminal users can press Speaker and dial Code 729 to further increase station ring volume.
- LCD contrast, off-hook ringing volume, station ringing volume, and speaker volume adjustments are determined by PRG 15-02-27.
- The LCD of the Electra Elite IPK terminals provide a volume bar indication while adjusting the following volumes or controls:
 - Speaker Volume
 - Handset/Headset Volume
 - Background Music (BGM) Volume
 - Ring Volume/Off-Hook Ring Volume
 - LCD Contrast



Default Setting

Enabled

System Availability

Terminals

All Multiline Terminals

Required Component(s)

None

Related Features

Electra Elite IPK Terminals

Electra Elite Terminal Migration

Handsfree and Monitor

Off-Hook Signaling

Programming

- ➔ **15-02-27 Multiline Telephone Basic Data Setup – Handset Volume**
This option determines whether or not a multiline terminal handset volume changes back to the system (0) or whether it is retained at the user's setting (1) after hanging up the handset. This command also controls LCD Contrast, Off-Hook Ringing Volume, Station Ringing Volume and Speaker Volume.
(default: 0)

Operation

To adjust the volume of incoming ringing and splash tone:

1. If the telephone is idle, press **Speaker** and dial **729**. If the telephone is ringing, skip to Step 2.
2. Press VOLUME ▲ or VOLUME ▼.

To adjust the volume of ringing incoming Paging announcements, Handsfree, the handset or Background Music:

1. Press VOLUME ▲ or VOLUME ▼.

 *The feature must be active to change the volume. Press the volume keys when the telephone is idle to adjust the display contrast.*

Warning Tone For Long Conversation

Description

The system can broadcast warning tones to a trunk caller, warning the caller that he has been on the call too long. If he chooses, the caller can disregard the tones and continue talking. The outside caller does not hear the warning tones. In addition, warning tones do not occur for Intercom calls and most incoming trunk calls. DISA trunks can also have warning tones. Warning tones are not available to analog single line telephone (SLT) users.

There are two types of warning tones: Alarm Tone 1 and Alarm Tone 2. Alarm Tone 1 is the first set of tones that occur after the user initially places a trunk call. Alarm Tone 2 broadcasts periodically after Alarm Tone 1 as a continued reminder. Each alarm tone consists of three short beeps.

If programmed, DISA calls are disconnected unless the continue code is entered by the user. With the Long Conversation Cutoff feature, incoming or outgoing central office calls can also be disconnected.

Warning Tone for DISA Callers

For DISA callers, with this feature enabled, the warning tone timer begins when an incoming DISA call places an outgoing call and either the inter-digit time expires or the outgoing call is answered.

If an outside call is transferred to forwarded off-premise using an outside trunk, the warning tone timer begins immediately. This occurs only if either trunk involved in the call is programmed for this feature (Program 14-01-17). When transferring a trunk call off-premise, Program 14-01-13 must be enabled (set to 1).

Conditions

- Warning Tone for Long Conversation does not occur for incoming trunk calls.
- Warning Tone for Long Conversation occurs for all outgoing trunk calls, regardless of how they are placed or other outgoing restrictions.
- Warning Tone for Long Conversation can be enabled for DISA calls.
- Warning Tone for Long Conversation does not occur for Intercom calls.
- Warning Tone for Long Conversation can be used with the Long Conversation Cutoff feature for outgoing calls.
- Warning Tone is presented to a single line telephone in the ear piece.



Default Setting

Disabled

Programming

- ➔ **14-01-17 Basic Trunk Data Setup – Trunk to Trunk Warning Tone for Long Conversation Alarm**
Enable (1)/Disable (0) whether or not DISA callers should hear the Warning Tone for Long Conversations.
(default: 0)
- ➔ **14-01-25 Basic Trunk Data Setup – Continued/Discontinued Trunk-to-Trunk Conversation**
Enable (1)/Disable (0) whether or not a user should be able to use the continue/discontinued code.
(default: 0)
- ➔ **20-06-01 Class of Service for Extensions**
Assign a Class of Service (1~15) to an extension.
(default: 101 to Class 15 – All others to Class 1)
- ➔ **20-13-01 Class of Service Options (Supplementary Service) – Long Conversation Alarm**
Enable (1)/Disable (0) Warning Tone for Long Conversation.
(default: 0)
- ➔ **20-21-01 System Options for Long Conversation – Long Conversation Alarm 1**
After a user places a trunk call, the system sends the first warning tone to their extension after this time (0~64800 seconds).
(default: 170)
- ➔ **20-21-02 System Options for Long Conversation – Long Conversation Alarm 2**
After hearing the first warning tone, the system sends additional warning tones after this time (0~64800 seconds). The warning tones continue, spaced by this interval, until the user hangs up.
(default: 180)
- ➔ **20-28-01 Trunk to Trunk Conversation – Conversation Continue Code**
Enter a single digit Continue Code for the DISA call to use to immediately disconnect or continue their outside call (0~9, * or # are accepted entries).
(default: No Setting)
- ➔ **20-28-02 Trunk to Trunk Conversation – Conversation Disconnect Code**
Enter a single digit Disconnect for the DISA call to use to immediately disconnect or continue their outside call (0~9, * or # are accepted entries).
(default: No Setting)
- ➔ **20-28-03 Trunk to Trunk Conversation – Conversation Continue Time**
When Program 14-01-25 is enabled, determine the time (0~64800 seconds) a call is extended when the user dials the Continue code (defined in 20-28-01).
(default: 0)

- **21-01-01 System Options for Outgoing Calls – Seizure Trunk Line Mode**
Set the time the (0~64800 seconds) system should wait before starting the Warning Tone Timer.
(default: 0)
- **21-01-03 System Options for Outgoing Calls – Trunk Interdigit Time (External)**
Set the time (0~64800 seconds) the system waits to expire before putting a call in talk state.
(default: 5)
- **25-07-07 System Timers for VRS/DISA – Conversation Warning Tone Time**
Set the time (0~64800 seconds) the system should wait before the Warning Tone is heard by DISA callers . If an outside call is transferred or forwarded off-premise using an outside trunk, this time starts immediately. This occurs if either trunk involved in the call is programmed for the Warning Tone (Program 14-01-17).
(default: 3600)
- **25-07-08 System Timers for VRS/DISA – Long Conversation Disconnect**
Set the time (0~64800) after the Warning Tone is heard the system waits before disconnecting DISA calls – unless the Continue code is entered (Program 20-28-01).
(default: 10)

Related Features

Central Office Calls, Answering

Central Office Calls, Placing

Direct Inward System Access (DISA)

Intercom

Long Conversation Cutoff

Single Line Telephones, Analog 500/2500 Sets

Toll Restriction

Operation

Warning Tone for Long Conversation is automatic if programmed.

Warning Tone for Long Conversation for DISA Callers:

1. A DISA caller dials into the system and places a call.
2. After the Warning Tone is heard, ***To continue the call***, the DISA caller must press the programmed Continue Code.

- OR -

To disconnect the call, the DISA caller presses the programmed Disconnect Code.

Wireless - DECT

Description

The Wireless – DECT (Digital Enhanced Cordless Telecommunication) system allows using 2.4 GHz Wireless – DECT telephones. These telephones provide the freedom and convenience of a wireless telephone but also allow access to features provided by the Electra Elite IPK II system.

The BSU(4M)-U20 ETU interfaces the Electra Elite IPK KSU with four Base Stations (BS) and can be expanded to 16 base stations with two BSU(6S)-U20s.

The Electra Elite IPK II Expanded system supports 248 Wireless – DECT telephones.

The Electra Elite IPK II Basic system supports 56 Wireless – DECT telephones.

Components of the Wireless – DECT system include the following:

Handset

The handset has the following features:

- Alphanumeric Display with Backlight

The backlight can be turned On/Off in the profile setup.

- LED Indication for Incoming and Unanswered Calls

- Telephone Book with 80 Number Memory Capacity

While idle, dial the number to be stored, then press > and OK. Enter the name associated with the number using the dial pad, and press OK.

- Built-in Vibrator

The vibrator can be turned On/Off using the > and < keys in the profile setup.

- Auto Log-in (auto switch between four systems)

The handset can be subscribed to four different systems. When Auto log-in is selected from the handset menu, the handset automatically selects the closest system. The selected system is marked with A.

- Silent Mode (mute all sounds)

To set/cancel Silent Mode, press the Menu key and dial #.

- Redial Function (last 10 numbers)

Press < and continue to press < to scroll through the numbers. Press Hook key to dial a number.

Programming Pause

A long press on * adds a pause to predial or phone book numbers.

 Programming of two Different Setups

Each handset can program two profiles to control ring tone and vibrator. One can be an indoor setting; the other, an outdoor setting.

 Adjustable Volume

Ring volume can be adjusted using > and < in the profile setup.

 Key Lock

Press Menu and * to lock the dial pad.

 Nine Different Ring Tones

Ring tones can be selected using > and <.

 Microphone Mute

Press OK while the telephone is off-hook to mute the microphone.

 Caller ID Presentation Headset Connection Automatic Off-Hook

B-Answer can be turned On/Off using > and < in the profile setup. When set to On, the telephone automatically goes off-hook when it rings.

 R-Key for Transfer and Special Services

When off-hook, press R to Recall, transfer.

Base Station

The Base Station provides the link between the Wireless – DECT telephone and the Electra Elite IPK II system. Base Stations are connected to the BSU()-U20 ETU using standard two wire (twisted pair) telephone cable, CAT 4 or CAT 5. The maximum distance from the BSU()-U20 ETU to the Base Station is 3,280 feet. Local power is not required because the Base Station receives power from the IPK II system. Up to 16 Base Stations can be connected to the system. Each Base Station supports four simultaneous IPK II Wireless – DECT traffic channels at 32 Kbs.

Repeater

The Repeater allows extended coverage for low traffic areas not covered by a Base Station. Sufficient coverage for the main traffic area should be provided by the Base Station. An external antenna can also be connected to extend the coverage area. Local power is required for the Repeater (within six feet) and must be synchronized with a Base Station in the zone providing coverage. The Repeater is synchronized with the Base Station using the Repeater Programming Kit that provides an RS232 cable to connect the Repeater to a PC. A Windows application is used to define the Base Station that should be synchronized with the Repeater. Each Repeater can support two simultaneous Wireless – DECT traffic channels at 32 Kbs. As the Repeater is paired with the Base Station, these are not additional channels, but are available to handle calls from the Base Station as the user moves out of range from the Base Station to the Repeater area. Repeaters should be placed a minimum of 75 feet line of sight between each other to prevent problems with the channels in use.

Conditions

- The CLKG-U() Unit is not required for IPK II Wireless – DECT.
- Four simultaneous calls can be made per Base Station.
- Sixteen simultaneous calls can be made per BSU (Master) ETU.
- Forty-eight simultaneous calls can be made per IPK II system.
- Two simultaneous calls can be made per Repeater.
- Repeaters expand the coverage area of a Base Station without using a BSU port.
- Each Base Station can have up to three Repeaters, and they can be used to form a linear line (chain) or a non-linear line. Three repeaters can be assigned to one base.
- When the Wireless – DECT telephone does not respond to an incoming call within 12 seconds because it is out of area, the originator hears a busy tone.
- The Out of Area Timer is fixed at 12 seconds (Program 20-22-05).
- The Call Forward – Busy/No Answer feature is available when the CPU detects out of area.
- Repeaters require an AC Adapter for power.
- The maximum distance between the KSU and Base Station is 3,280 feet.
- The maximum number of Wireless – DECT handsets is 248.
- The maximum number of Base stations is 16.
- The maximum number of BSU ETUs that can be installed in the IPK II KSU is three: one master and two slaves.
- When installing BSU ETUs, a maximum of two slave ETUs can be stacked on the Master ETU.

This limits the combinations to those shown in the table below:

No.	Slot N-2	Slot N-1	Slot N	No. of Base Stations	Note
1	None	None	BSU(4M) ETU	4	
2	None	BSU(2S) ETU	BSU(4M) ETU	6	
3	None	BSU(6S) ETU	BSU(4M) ETU	10	
4	BSU(2S) ETU	BSU(2S) ETU	BSU(4M) ETU	8	
5	BSU(6S) ETU	BSU(2S) ETU	BSU(4M) ETU	12	
6	BSU(2S) ETU	BSU(6S) ETU	BSU(4M) ETU	12	
7	BSU(6S) ETU	BSU(6S) ETU	BSU(4M) ETU	16	

- Wireless – DECT is not supported with ACD.
- Off-Hook signaling is not supported for Wireless – DECT telephones.

Default Setting

None

System Availability

Terminals

Wireless – DECT Telephones

Required Component(s)

- BSU(4M)-U20 ETU (Master)
- BSU(2S)-U20 ETU (Slave)
- BSU(6S)-U20 ETU (Slave)
- Base Station (BS)
- Tool for Wireless Line ETU (Windows Application) included with the BSU(4M)-U20 ETU.

Related Features


None

Programming


- ➔ 10-02-05 Location Setup – Trunk Access Code
- ➔ 15-14-01 Programmable One-Touch Keys – Dial Data
- ➔ 15-15-05 Wireless – DECT Terminal Basic Data Setup – Voice Mail Answering When Out of Range
- ➔ 15-15-06 Wireless – DECT Terminal Basic Data Setup – Call Transferring When Out of Range
- ➔ 15-15-07 Wireless – DECT Terminal Basic Data Setup – DECT Transmit Level
- ➔ 15-15-08 Wireless – DECT Terminal Basic Data Setup – DECT Receive Level
- ➔ 20-22-05 System Options for Wireless – DECT Service – Out of Area Judging Time
- ➔ 83-12-01 Wireless – DECT Measurement
- ➔ 90-27-01 Wireless – DECT System ID
- ➔ 91-06-01 Wireless – DECT Subscription, New
- ➔ 91-07-01 Wireless – DECT Subscription, Delete

Operation

Placing an outside call:

1. Press the  key (On-Hook/Off-Hook).
2. Dial **9**.
3. Dial the outside number.

Placing an outside call with pre-dial:

1. Dial **9**.
2. Press and hold the asterisk * key to insert a pause (–), if necessary.
3. Dial the outside number.
4. Press the  key (On-Hook/Off-Hook).



Codes Tables

SECTION 1 ABOUT THIS CHAPTER

The charts in this chapter provide a list of the Service Codes, Function Key Codes, and System Number Plan/Capacities. The service codes and function codes are listed by number and by feature in separate charts for ease of use.

SECTION 2 SIMPLIFYING MULTILINE TERMINAL OPERATIONS WITH ONE-TOUCH KEY OPERATION

A multiline terminal user can access many features through Service Codes (e.g., Service Code **#9** to access a specific trunk). To streamline the operation of their telephone, a multiline terminal user can store these codes under One-Touch Keys. This provides one-button operation for almost any feature. To find out more, turn to the One-Touch Calling feature.

When reading an instruction using programmable keys, you see a notation similar to (**PRG 15-07 or SC 7nn**). This means the key requires function code nnn, and you can program this code through Program 15-07 or by dialing Service Code 751 or 752. Refer to the Programmable Function Keys feature for more information.

SECTION 3 USING HANDSFREE

The manual assumes each extension has Automatic Handsfree. This lets a user just press a line key or Speaker Key to answer or place a call. For extensions without Automatic Handsfree, the user must:

- Lift the handset or press **Speaker** for intercom dial tone.
- Lift the handset or press **Speaker**, then press a line key for trunk dial tone.

Table 3-1 Post Dialing Service Codes – Single Digit Post Dialing Codes

Code	For this feature. . .	When you . . .
1	Handsfree Answerback / Forced intercom Ringing	Change the signaling mode of your outgoing Intercom call
2	Department Step Calling	Cycle to the next member of a Department Calling Group
3~5	Not used	
6	Voice Over	Send a Voice Over to a busy extension after hearing Busy/Ring tone
7	Barge-In	Barge into another station active call
8	Voice Mail	Leave a message in a co-worker's mailbox after calling their busy or unanswered extension
0	Message Waiting	Leave a Message Waiting at a co-worker's busy or unanswered extension
#	Call Waiting / Camp-On / Callback / Trunk Queuing	Use Call Waiting / Camp-On / Callback / Trunk Queuing
*	Off-Hook Signaling	Send off-hook signal tones to a busy extension

Table 3-2 Service Codes by Number

Dial this Service Code. . . ¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
* + Enter Account Code + *	Enter an Account Code	Account Codes	-
**	Pick up a call ringing or waiting at another extension	Directed Call Pickup Voice Response System (VRS)	-
**#	Pick up a call ringing an extension in your own pickup group (except Ring Group calls)	Group Call Pickup	24
*0	Answer a Message Waiting request	Message Waiting	38
*06	Set the Automatic Transfer for each trunk line	Transfer	-
*07	Cancel the Automatic Transfer for each trunk line	Transfer	-
*08	Set the Destination for Automatic Trunk Transfer	Transfer	-
*1	Make a Combined Page	Paging	-
+ 0	Cancel Call Forwarding	Call Forwarding	16
*2 + 1 + Type (2~4)	Activate Personal Answering Machine Emulation	Voice Mail (Personal Answering Machine Emulation)	16
*2 + 2 + Destination + Type (2~4)	Activate Call Forwarding when Busy/Not Answered	Call Forwarding	16
*2 + 3 + Destination + Type (2~4)	Activate Call Forward Follow Me at the destination extension	Call Forwarding with Follow Me	16
*2 + 4 + Destination + Type (2~4)	Activate Call Forwarding Immediate	Call Forwarding	16
*2 + 6 + Destination + Type (2~4)	Activate Call Forwarding when Unanswered (delayed)	Call Forwarding	16
*2 + 7 + Destination + Type (2~4)	Activate Call Forwarding (Both Ringing)	Call Forwarding	16
*3 (After + 001~200 + busy)	Disconnect a call in progress on a trunk	Forced Trunk Disconnect	-
*4 + 3 + Message (01~20), or + 3 + Hang up to cancel	Activate or cancel Selectable Display Messaging	Selectable Display Messaging	17

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
*4 + 6 + Trunk access code + Outside number, or + 6 + Hold + Hang up to cancel	Forward your calls to an off- premise telephone number	Call Forwarding Off-Premise	17
+ 7 + Record message + # + Condition (2, 4, 6 or 7) + Destination + Type (2 or 3) or + 7 + 3 to cancel	Record, listen to or erase a Personal Greeting or Park and Page	Voice Response System (VRS) (Personal Greeting)	17
*5	Log out of or in to an ACD group	Automatic Call Distribution (ACD)	*10
*6 + Orbit (01~64)	Pick up a call parked in a system Park orbit (01~64)	Park	*04 + orbit
*7	Split (switching) between calls on a single line telephone	Call Waiting / Camp-On	-
*8	Call your mailbox	Voice Mail	67
# * # *	Enter the system programming mode	System Programming Password Protection	-
# * # 9	Back up system data	Maintenance	-
Hookflash + ## + Enter Account Code + Hookflash	Enter an Account Code at a single line telephone	Account Codes	-
#0	Use Universal Answer Code to pick up a call ringing over the paging system	Central Office Calls, Answering	-
Hookflash + #1 + extension + hookflash twice	Activate Conference from a Single Line (500/2500) set	Conference	-
#2 + bin	Dial a Common Abbreviated Dialing number	Abbreviated Dialing	27
#3	Flash a trunk from an single line telephone	Flash	-
#4 + bin	Dial a group Abbreviated Dialing number	Abbreviated Dialing	28
#5	Use Last Number Redial	Last Number Redial	-
#6 + orbit (01~64)	Park a call in a system Park orbit (1~8, 01~32 or 01~64)	Park	*04 + orbit (1~64)
#7	Use Personal Abbreviated dialing	Abbreviated Dialing	-

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
#8	Set up an Unsupervised Conference	Tandem Trunking (Unsupervised Conference)	-
#9 + 001 -200	Place a call over a specific trunk	Central Office Calls, Placing	*01 + trunk (001~200)
0 (Off-Hook)	Leave a Message Waiting at a co-worker's busy or unanswered extension	Message Waiting	35
1 (Off-Hook)	Change the signaling mode of your outgoing Intercom call	Handsfree Answerback/Forced Intercom Ringing	-
4 (On-Hook)	Listen to the General Message	Voice Response System (VRS)	-
6 (On-Hook)	Check an extension number	Voice Response System (VRS)	-
8 (On-Hook)	Listen for the time	Voice Response System (VRS)	-
9	Place a call using ARS or Trunk Group Routing	Automatic Route Selection Trunk Group Routing	*02
100 + code + 0	Use Dial Block	Toll Restriction, Dial Block	-
101 + code + 0	Are a supervisor using Dial Block	Toll Restriction, Dial Block	-
+ Group number (1~8 or 01~64)	Set Automatic Transfer Setup for each extension group	Transfer	-
102 + Group number (1~8 or 01~64)	Cancel Automatic Transfer Setup	Transfer	-
103 + Group number (1~8 or 01~64) + mode + extension	Set the destination for Automatic Transfer Setup for each extension group	Transfer	-
104 + Group number (1~8 or 01~64)	Set Delayed Transfer for each extension group	Transfer	-
105 + Group number (1~8 or 01~64)	Cancel Delayed Transfer	Transfer	-
106 + Group number (1~8 or 01~64)	Set up DND for each extension group	Transfer	-

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
107 + Group number (1~8 or 01~64)	Cancel DND for each extension group	Transfer	-
111	Are an SLT Listening to the General Message	Voice Response System (VRS)	-
112 + 3 to erase, 5 to listen or 7 to record	Record, listen to or erase the General Message	Voice Response System (VRS)	-
116 + 3 to erase, 5 to listen or 7 to record	Record, listen to or erase a VRS Message	Voice Response System (VRS)	-
118	Use Night Mode Switching for other group	Night Answer	-
120	Use Common Cancelling Service Code	TBD	-
121	Print the SMDR Extension Accumulated printout	Station Message Detail Recording (SMDR)	-
122	Print the SMDR Group Accumulated printout	Station Message Detail Recording (SMDR)	-
123	Print the SMDR Account Code Accumulated printout	Station Message Detail Recording (SMDR)	-
782	Transfer a call to the VRS. This can also be used for routing ANI/DNIS to the VRS.	Transfer	-
627	Enable DND at a room telephone	Hotel/Motel (Do Not Disturb)	-
628	Cancel DND at a room telephone	Hotel/Motel (Do Not Disturb)	-
629	Enable DND for another room telephone	Hotel/Motel (Do Not Disturb)	-
630	Cancel DND at another room telephone	Hotel/Motel (Wake Up Call)	-
631	Set up a Wake Up call for your own room telephone	Hotel/Motel (Wake Up Call)	-
632	Cancel a Wake Up Call for your own room telephone	Hotel/Motel (Wake Up Call)	-
633	Set a Wake Up Call for another guest's room telephone	Hotel/Motel (Wake Up Call)	-

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
634	Cancel a Wake Up Call for another guest's room telephone	Hotel/Motel (Wake Up Call)	-
635	Enable Room to Room Call Restriction for a guest's room telephone	Hotel/Motel (Room to Room Call Restriction)	-
636	Disable Room to Room Call Restriction for a guest's room telephone	Hotel/Motel (Room to Room Call Restriction)	-
637	Change a room telephone Toll Restriction (When Checked In) level	Hotel/Motel (Toll Restriction When Checked In)	-
638	Set a room as checked in	Hotel/Motel (Room Status)	-
639	Set a room as checked out	Hotel/Motel (Room Status)	-
641	Set a room status from another telephone	Hotel/Motel (Room Status)	-
642	Request a Room Status Printout	Hotel/Motel (Room Status Printouts)	-
145 + trunk # + 1 (block) 145 + trunk # + 0 (enable)	Block/busy out outbound usage on a trunk with Trunk Port Disable.	Central Office Calls, Placing	-
150 + 0 (install) or 1 (remove)	Log in (0) or log out (1) for your Department Calling Group	Department Calling	
654	Enable Conversation Record at a single line telephone	Voice Mail	-
155	Log out of or in to an ACD Group from a single line telephone	Automatic Call Distribution (ACD)	-
156	Activate Work Time from a single line telephone	Automatic Call Distribution (ACD)	*17
157	Cancel Work Time from a single line telephone	Automatic Call Distribution (ACD)	*17
158	Activate Rest Mode from a single line telephone	Automatic Call Distribution (ACD)	*13
159	Cancel Rest Mode from a single line telephone	Automatic Call Distribution (ACD)	*13

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
Hookflash + 160	Use ACD Recording for a single line telephone	Automatic Call Distribution (ACD)	-
163 + 6-digit code + line + telephone number	Override Toll Restriction	Toll Restriction	-
			-
167	Log an agent into their ACD Group	Automatic Call Distribution (ACD)	-
168	Log an agent out of their ACD Group	Automatic Call Distribution (ACD)	-
169	Are a Supervisor assigning an agent into another ACD Group or changing an agent's status	Automatic Call Distribution (ACD)	-
170 + ACD Group	Change your ACD Group assignment	Automatic Call Distribution (ACD)	-
172 + Line number (001~200)	Answer a call on a specific trunk	Central Office Calls, Answering Hold	
675	Monitor a room telephone	Hotel/Motel (Room Monitor)	-
177	Change the COS of another extension. Must be allowed in Program 20-13-28.	Class of Service	-
678 + 0~9	Change the language of a display telephone.	Alphanumeric Display / Maintenance	-
179 + 1 (set) or 0 (cancel)	Change the ability for a second call with DID/DISA/DIL	Central Office Calls, Answering	-
189	Transfer an Electra Elite IPK II Wireless call when out of range	Electra Elite IPK II Wireless	-
700 + extension # + enter name + Hold	Program extension names	Name Storing	55
701 + zone (1~9 or 01~64) 801 + zone (0 or 00)	Make an Internal Zone Page Make an All Call Internal Page	Paging, Internal	21 + zone 22
702 + Door Box (1~4 or 1~8)	Place a call to a Door Box	Door Box	-
703 + zone (1~4 or 1~8) 803 + zone (0)	Make an External Zone page Make an External All Call page	External Paging	19 + zone 20

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
704 + trunk group (1~8 or 1~9 or 001~200)	Place an outside call over a trunk group	Central Office Calls, Placing	*02 + group
707	Override Do Not Disturb or Call Forwarding	Call Forwarding Do Not Disturb	37
708	Step through a Department Group	Department Step Calling	36
709	Send a Call Waiting tone to a busy extension	Call Waiting	33
710	Break into another extension call	Barge-In	-
711 + 1 (ICM) or 2 (TRK) + tone (1~8)	Listen to the incoming ring choices	Selectable Ring Tones	-
712	Change the signal type for calling an extension	Intercom	-
715	Save a number (from SLT) or dial a saved number	Save Number Dialed	30
718 + 1 718 + 2 718 + 3 718 + 4 718 + 5 718 + 6 718 + 7 718 + 8	Activate Day 1 Mode Activate Night 1 Mode Activate Midnight 1 Mode Activate Rest 1 Mode Activate Day 2 Mode Activate Night 2 Mode Activate Midnight 2 Mode Activate Rest 2 Mode	Night Service	09 + 1 09 + 2 09 + 3 09 + 4 09 + 5 09 + 6 09 + 7 09 + 8
720 + 1 (ICM) or 2 (TRK) + tone (1~8)	Change your extension incoming ring tones	Selectable Ring Tones	-
721	Enable Handsfree Answerback for incoming Intercom calls	Handsfree Answerback/Forced Intercom Ringing	-
722	Call off-premise with a Door Box	Call Forwarding, Off-Premise Door Box	54
723	Enable Forced Ringing for incoming Intercom calls	Handsfree Answerback/Forced Intercom ringing	-
724	Enable/disable Dial Pad Confirmation Tone	Dialing Pad Confirmation tone	-
725	Turn Background Music on or off	Background Music	04

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
727 + 1 or 2 + time, or 727 + 1 or 2 + 9999 to cancel	Check, set or cancel an alarm	Alarm	-
728 + hour + minutes	Set the system Time	Time and Date Clock/Calendar Display	-
729	Check or change ring volume	Volume Control	-
730	Use Remote maintenance	-	-
732	Place a call on Group Hold	Hold	-
740	Use System alarm message delete	-	-
747 + 0 (Cancel) 1 (Trk calls) 2 (Paging, ICM, Call Forward and transfers) 3 (All calls) 4 (Call Forwards)	Activate Do Not Disturb	Do Not Disturb	-
749	Place a call on Exclusive Hold at a SLT set.	Hold	-
750	Camp-On to an extension when calling into the system through the VRS	Voice Response System (VRS)	35
751 + key + code	Change the function of a programmable key using 751 service codes	Programmable Function Keys	-
752 + key + code	Change the function of a programmable key using 752 service codes	One-Touch Serial Operation	-
753 + bin + number + Hold + Name + Hold to store	Store Common Abbreviated Dialing numbers	Abbreviated Dialing	-
754 + bin + number + Hold + Name + Hold to store	Store Group Abbreviated Dialing numbers	Abbreviated Dialing	-
755 + One Touch key + code	Program a One-Touch Key or Personal Speed Dial	One-Touch Dialing	-
756	Answer a call ringing a telephone in your pickup group (except Ring Group calls)	Group Call Pickup	-
757	Park a call or pick up a parked call at an extension	Park	-

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
759	Retrieve a call from Exclusive Hold at a SLT set.	Hold	-
760	Use DID ACD Access Code	Automatic Call Distribution (ACD)	-
762	Pick up a call from Group Hold	Hold	-
763	Join a Meet Me Conference or Meet Me Page on an Internal Paging Zone (if your extension is in the group called)	Meet Me Conference Meet Me Paging	23 or 32
764 + zone paged (0~9 or 00~64)	Join a Meet Me Conference or Meet Me Page if your extension is not in the group paged	Meet Me Paging	23 or 32
765 + zone (0~8)	Join a Meet Me Conference or Meet Me Page on an External Paging Zone	Meet Me Conference Meet Me Paging	23 or 32
768 + pickup group (1~8 or 1~9 or 01~64)	Answer a call ringing a telephone in another pickup group (except Ring Group calls)	Group Call Pickup	26 + group
769	Answer a call ringing a telephone in another pickup group if you do not know the group's number (except Ring Group Calls)	Group Call Pickup	25
770	Cancel a Callback request	Callback	-
771 + ext	Cancel Messages Waiting you have left at a specific extension	Message Waiting	-
773	Cancel all Messages Waiting you have left at other extensions	Message Waiting	-
775 + pswd (0000) + place outside call	Temporarily override an extension Toll Restriction	Toll Restriction Override	-
776	Clear a number saved by Last Number Redial	Last Number Redial	-
780 + Relay (0~8)	Use the General Purpose Relay	Paging, External Night Service	51
781 + 00 (no tone), 01 (general) or 02 (holiday)	Change the Music on Hold Tone	Music on Hold	-

Table 3-2 Service Codes by Number (Continued)

Dial this Service Code. . .¹	When you. . .	For this feature. . .	Also see Function Key. . .
¹ Except where indicated, dial Service Code from Intercom dial tone (e.g., press idle Speaker key first).			
782	Route ANI/DNIS to the VRS. It can also be used to transfer to VRS.	Transfer Voice Response Service (VRS)	
783	Enable the data communication auto-answer mode	Data Communications	-
784	Access the VRS	Voice Response Service (VRS)	-
785	Clear the number saved by Save Number Redial	Save Number Redial	-
790	Use Voice Over after calling a busy extension	Voice Over	48
794	Split between two calls on a single line telephone	Call Waiting	
799	Test Callback operation for a single line telephone	Callback	-

Table 3-3 Service Codes by Feature

For this feature...	Dial this Service Code...¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Speed Dialing	753 + bin + number + Hold + Name + Hold to store	Store System Speed Dialing numbers	-
	754 + bin + number + Hold + Name + Hold to store	Store Group Speed Dialing numbers	-
	#2 + bin	Dial a System Speed Dialing number	27
	#4 + bin	Dial a Group Speed Dialing number	28
	#7 + bin	Use Personal Speed Dialing	-
Account Codes	* + Enter Account code + *	Enter an Account Code	-
	Hookflash + ## + Enter account code + Hookflash	Enter an Account Code at a single line telephone	-
Alarm	727 + 1 or 2 + time, or 727 + 1 or 2 + 9999 to cancel	Check, set or cancel an alarm	-
Alphanumeric Display	678 + 0~9	Select the language to be used on display multiline terminals.	-
Wireless – DECT	689	Transfer a Wireless – DECT call when out of range	-

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code...¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Automatic Call Distribution (ACD)	*5	Log out of or in to an ACD Group	*10
	655	Log out of or in to an ACD Group from a single line telephone	
	656	Activate Work Time from a single line telephone	*17
	657	Cancel Work Time from a single line telephone	*17
	658	Activate Rest Mode from a single line telephone	*13
	659	Cancel Rest Mode from a single line telephone	*13
	Hookflash + 660	Use ACD Recording for SLT	-
	667	Allow ACD Agent to log into a group	
	668	Allow ACD Agent to log out of a group	
	669	Allow supervisor to change agent's status	
	670 + ACD Group	Change your ACD Group assignment	-
Automatic Route Selection or Trunk Group Routing	9	Place an call using Trunk Group Routing or Automatic Route Selection	*02
Background Music	725	Turn Background Music on or off	04
Call Forwarding	745	Set/Cancel Call Forwarding (Both Ringing)	
	742	Set/Cancel Call Forwarding when Busy	
	744	Set/Cancel Call Forwarding when Busy/ No Answer	
	743	Set/Cancel Call Forwarding No Answer	
	746	Set/Cancel Call Forwarding Follow Me	
	741	Set/Cancel Call Forwarding Immediate	

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code... ¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Call Forwarding, Off-Premise Door Box	722	Call off-premise with a Door Box	54
Call Forwarding/Do Not Disturb Override	707	Override an extension Call Forward or DND setting.	37
Call Waiting / Camp-On	794	Split (switching) between calls on a single line telephone	-
	770	Cancel a Callback request	-
	799	Test Callback operation for SLT	-
Callback / Camp-On / Trunk Queuing	#	Camp-On or leave a Callback for a busy extension or trunk	35
	770	Cancel a Callback request	-
	799	Test Callback operation for single line telephone	-
Central Office Calls, Answering / Hold	#0	Use Universal Answer to pick up a call ringing over the paging system	-
	672 + Line number (001~200)	Answer a call on a specific trunk	-
	679 + 1 (set) or 0 (cancel)	Change the ability for a second call with DID/DISA/DIL	
Central Office Calls, Placing	#9 + 001~200	Place a call over a specific trunk	*01 + trunk (100-200)
	645 + trunk # + 1 (block) 145 + trunk # + 0 (enable)	Block/busy out outbound usage on a trunk with Trunk Port Disable.	-
	704 + trunk group (1~9 or 001~200)	Place an outside call over a trunk group	*02 + group
Class of Service	677	Change the COS of another extension. Must be allowed in Program 20-13-28.	-
Conference	Hookflash + #1 + extension + hookflash twice	Activate Conference from a Single Line (500/2500) set	1016
Data Communications	783	Enable the data connection auto-answer mode	-
	784	Disconnect an active data call	-

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code... ¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Department Calling	650 + 0 (install) or 1 (remove)	Log in (0) or log out (1) for your Department Calling Group	46
Department Step Calling	#	Step Call through a Department Group	36
Dial Pad Confirmation Tone	724	Enable/disable Dial Pad Confirmation Tone	-
Directed Call Pickup	** + ext.	Pick up a call ringing or waiting at an extension	-
Do Not Disturb	747 + 0 (Cancel) 1 (Trk calls) 2 (Paging, ICM, Call Forwards, and Transfers) 3 (All calls) 4 (Call Forwards)	Activate Do Not Disturb	-
Door Box	702 + Door Box (1~4 or 1~8)	Place a call to a door Box	-
	722	Forward a Door Box off-premise	-
E911	786	Turn off the E911 alarm	-
Flash	#3	Flash a trunk from a single line telephone	-
Forced Trunk Disconnect	* 3 (after #9 + 1~8 or 001~200 + busy)	Disconnect a call in progress on a trunk	-
Group Call Pickup	*#	Pick up a call ringing an extension in your own pickup group (except Ring Group calls)	24
	768 + pickup group (1~8 or 1~9 or 01~64)	Answer a call ringing a telephone in another pickup group	26 + group
	769	Answer a call ringing a telephone in another pickup group if you do not know the group number (except Ring Group calls)	25
Handsfree Answerback/Forced Intercom Ringing	1 (Off-Hook)	Change the signaling mode of your outgoing Intercom call	-
	721	Enable Handsfree Answerback for incoming Intercom calls	-
	723	Enable Forced Ringing for incoming Intercom calls	-

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code...¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Hold	732	Place a call on Group Hold	-
	749	Place a call on Exclusive Hold at a SLT set	-
	759	Retrieve a call from Exclusive Hold at a 2-Button telephone	-
	762	Pick up a call from Group Hold	-
			-
Hotel/Motel (Do Not Disturb)	627	Enable DND at a room telephone	-
Hotel/Motel (Do Not Disturb)	628	Cancel DND at a room telephone	-
Hotel/Motel (Do Not Disturb)	629	Enable DND for another room telephone	-
Hotel/Motel (Do Not disturb)	630	Cancel DND at another room telephone	-
Hotel/Motel	675	Monitor a room's telephone	-
Hotel/Motel (Wake Up Call)	631	Set a Wake Up Call or your own room telephone	-
Hotel/Motel (Wake Up Call)	632	Cancel a Wake Up Call for your own room telephone	-
Hotel/Motel (Wake Up Call)	633	Set a Wake Up Call for another guest's room telephone	-
Hotel/Motel (Wake Up Call)	634	Cancel a wake Up Call for another guest's room telephone	-
Hotel/Motel (Room to Room Call Restriction)	635	Enable Room to Room Call Restriction for a guest's room telephone	-
Hotel/Motel (Room to Room Call Restriction)	636	Disable Room to Room Call Restriction for a guest's room	-
Hotel/Motel (Toll restriction [When Checked In])	637	Change a room's telephone Toll Restriction (When Checked In) level	-
Hotel/Motel (Room Status)	638	Set a room as checked in	-

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code...¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Hotel/Motel (Room Status)	639	Set room as checked out	-
			-
Hotel/Motel (Room Status)	641	Set a room as available (clean) from another telephone	-
Hotel/Motel (Room Status Printouts)	642	Request a Room Status Printout	-
Last Number Redial	#5	Use Last Number Redial	-
	776	Clear number saved by Last Number Redial	-
Maintenance	#*# 9	Back up system data	-
	678 + 0~9	Display the language the telephone is using	-
Meet Me Conference Meet Me Paging	763	Join a Meet Me Conference or Meet Me Page on an Internal Paging Zone (if your extension is in the group called)	23 (Meet Me Paging) or 32 (Meet Me Conference)
	764 + zone paged (0~9 or 00~64)	Join a Meet Me Conference or Meet Me Page if your extension is not in the group paged	-
	765 + zone (0~8)	Join a Meet Me conference or Meet Me Page on an External Paging Zone	-
Message Waiting	0 (Off-Hook)	Leave a Message Waiting at a co-worker's busy or unanswered extension	38
	*0	Answer a Message Waiting request	38
	771 + ext	Cancel Messages Waiting you have left at a specific extension	-
	773	Cancel all Messages Waiting you have left at other extensions	-
Music on Hold	781 + 00 (no tone), 01 (general) or 02 (holiday)	Change the Music on Hold Tone	-
Name Storing	700 + enter name + Hold	Program extension names	55

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code... ¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Night Service	618	Use Night Mode Switching for other group	
	718 + 1	Activate Day 1 Mode	09 + 1
	718 + 2	Activate Night 1 Mode	09 + 2
	718 + 3	Activate Midnight 1 Mode	09 + 3
	718 + 4	Activate Rest 1 Mode	09 + 4
	718 + 5	Activate Day 2 Mode	09 + 5
	718 + 6	Activate Night 2 Mode	09 + 6
	718 + 7 718 + 8	Activate Midnight 2 Mode Activate Rest 2 Mode	09 + 7 09 + 8
Off-Hook Signaling	* (Off-Hook) or 709	Send off-hook signal tones to a busy extension	33
One-Touch Dialing	755 + One-Touch key + code	Program a One-Touch Key or Personal Speed Dial	-
Paging, Combined	*1 + Zone (1~8)	Make a combined zone page.	19 + zone
	*1 + Zone (0)	Make a combined All Call page.	20
Paging, External	703 + zone (1~8)	Make an external zone page	19 + zone
	703 + zone (0)	Make an external All Call page	20
Paging, External Night Service	780 + relay number (0~8)	Activate the general purpose relay.	51
Paging, Internal	701 + zone (1~8, 1~9 or 01~64)	Make an Internal Zone Page	21 + zone or
	701 + zone (0 or 00)	Make an internal All Call Page	22
Park	#6 + orbit (01~64)	Park a call in a system Park orbit (01~64)	*04 + orbit
	*6 + orbit (01~64)	Pick up a call parked in a system Park orbit (01~64)	*04 + orbit
	757	Park a call or picking up a parked call at an extension	-
Programmable Function Keys	751 + key + code	Change the function of a programmable key using 751 service codes	-
	752 + key + code	Change the function of a programmable key using 752 service codes	-
Save Number Dialed	715	Save a number (from SLT) or dialing a saved number	30
	785	Clear the number saved by Save Number Redial number	-

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code... ¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Selectable Display Messaging	*4 + 3 + message (01~20), or *4 + 3 + Hang up to cancel	Activate and Canceling Selectable Display Messaging	17
Selectable Ring Tones	711 + 1 (ICM) or 2 (Trk) + tone (1~8)	Listen to the incoming ring choices	-
	720 + 1 (ICM) + 2 (Trk) + tone (1~8)	Change your extension incoming ring tones	-
System Programming Password Protection	# * # *	Enter the system programming mode	-
Station Message Detail Recording	621	Print the SMDR Extension Accumulated printout	-
	622	Print the SMDR Group Accumulated printout	-
	623	Print the SMDR Account Code Accumulated printout	-
Tandem Trunking (Unsupervised Conference)	#8	Set up an Unsupervised Conference	-
Time and Date Clock/Calendar Display	728 + hour + minutes	Set the system Time	-
Toll Restriction, Dial Block	600 + code + 0	Use Dial Block	-
	600 + code + 0	Are a supervisor using Dial Block	-
Toll Restriction Override	775 + pswd (0000) + place outside call	Temporarily override an extension Toll Restriction	-
	663 + digit code + line + telephone number	Override Toll Restriction	-

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code... ¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Transfer	*06	Set the Automatic Transfer for each trunk line	
	*07	Cancel the Automatic Transfer for each trunk line	
	*08	Set the Destination for Automatic Trunk Transfer	
	602 + Group number (1~8 or 01~64)	Set Automatic Transfer Setup for each extension group	-
	603+ Group number (1~8 or 01~64)	Cancel Automatic Transfer Setup	-
	604 + Group number (1~8 or 01~64) + mode + extension	Set the destination for Automatic Transfer Setup for each extension group	-
	605 + Group number (1~8 or 01~64)	Set Delayed Transfer for each extension group	-
Transfer (continued)	606 + Group number (1~8 or 01~64)	Cancel Delayed Transfer	-
	607 + Group number (1~8 or 01~64)	Set up DND for each extension group	-
	608 + Group number (1~8 or 01~64)	Cancel DND for each extension group	-
	624 + Extension number	Transfer a call into an existing call.	-
	782	Transfer a call to the VRS. This can also be used to route ANI/DNIS to the VRS.	
Trunk Group Routing or Automatic Route Selection	9	Place a call using Trunk Group Routing or Automatic Route Selection	*02
Trunk Queuing	*8	Call your mailbox	67
	654	Enable Conversation Record at SLT set	-
	# (Off-Hook)	Camp-On to or leave a Callback at a busy trunk	35

Table 3-3 Service Codes by Feature (Continued)

For this feature...	Dial this Service Code... ¹	When you...	Also see Function Key...
¹ Except where indicated, dial Service Code from intercom dial tone (e.g., press idle Speaker key first).			
Voice Mail	8 (Off-Hook)	Leave a message in a co-worker's mailbox after callback their busy or unanswered extension	-
	*8	Call your mailbox	67
	654	Enable Conversation Record at SLT set	-
Voice Over	6 (Off-Hook) or 790	Send a Voice Over to a busy extension after hearing Busy/Ring tone	48
Voice Response System (VRS)	** + ringing ext.	Pick up a call ringing another extension for Directed Call Pickup or VRS Park and Page.	-
	*4 + 7 + Record message + # + Condition (2, 4,6 or 7) + Destination + Type (2 or 3) or *4 + 7 + 3 to cancel	Record, listen to or erase a Personal Greeting or Park and Page	17
	4 (On-Hook)	Listen to the General Message	-
	6 (On-Hook)	Check an extension number	-
	8 (On-Hook)	Listen for the time	-
	611	Use SLT listening to the General Message	-
	612 + 3 to erase, 5 to listen or 7 to record	Record, listen to or erase the General Message	-
	616 + 3 to erase, 5 to listen or 7 to record	Record, listen to or erase a VRS Message	-
	750	Camp-On to an extension when calling into the system through the VRS	-
	782	Transfer a call to the VRS. This can also be used for routing ANI/DNIS to the VRS.	-
784	Access the VRS	-	
Volume Control	729	Check or change ring volume	-
Common Canceling Service Code	620	Use Common Canceling Service Code	-

Table 3-4 Function Key Codes by Feature

To program a key, press the Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
For this feature...	Use this key...	When you...	Key Lamp Status	Also See Srvc Code
Speed Dialing	Code: 27 Operation: Press key + bin + Line or Speaker key	Dial a stored System Speed Dialing number	None	+ bin
	Code: 28 Operation: Press key + bin + Line or Speaker key	Dial a stored Group Speed Dialing number	None	# 4 + bin
Account Codes	Code: 50 Operation: Press key + Dial Account Code	Enter Account Codes	None	*
Automatic Call Distribution (ACD)	Code: *10 Operation: Press key to log in Press key + 1 to log out or 0 to cancel	Basic Operation Log in or out of an ACD Group	On red when logged in Off when logged out	
	Code: *12 Operation: Press key	Emergency Call Place or receive an Emergency Call	On while calling your supervisor or after being answered by your supervisor Flashing fast at the supervisor while ringing	-
	Code: *13 Operation: Press key	Rest Mode Enable/disable Rest Mode	On red when Rest Mode enabled Off when Rest Mode disabled	-
	Code: *14 Operation: Press key + Press 1 (Yes) or 2 (No)	Out of Service Take an ACD Group out of Service (for Group Supervisors only), or Taking all ACD Groups out of service (for System Supervisors only)	On red when the group is out of service.	-
	Code: *15 Operation: Call busy ACD agent + Press key	Terminal Monitor Monitor an ACD Agent's conversation	On red while monitoring Off when not monitoring	-

Table 3-4 Function Key Codes by Feature (Continued)

To program a key, press the Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
For this feature...	Use this key...	When you...	Key Lamp Status	Also See Srvc Code
Automatic Call Distribution (ACD) (cont.)	Code: *16 Operation: Press key to put agent on hold. Press key again + 1 to hang up agent or 0 to bring agent back into call.	Switch (split) between an ACD Agent and their outside caller after answering an emergency call	On red while the agent is on hold	-
	Code: *17 Operation: Press key	Working Time Enable/Disable Work Time	On when Work Time enabled, Flashing while on a call if Auto Work Time enabled Off when Work Time disabled	-
	Code: *18 + ACD Group Operation: Press key	ACD Overflow Control Overflow ACD calls to another group	On when enabled, Slow flash when disabled	-
	Code: *19 Operation: Press key while on-hook + Vol Up or Vol Down to scroll	Queue Status Check When in an ACD group, check the status of the queue groups	None	-
Background Music	Code: 04 Operation: Press key	Turn Background Music on or off	None	725
Barge-In	Code: 34 Operation: Call ext + Press key	Barge-In on a co-worker's conversation	None	710
Call Arrival (CAR) Key	Code: *03 + ext. Operation: Press key	Place or answer a call to your co-worker's extension	Slow Flash red when ringing, On red when busy	-
Call Forwarding, Both Ring	Code: 14 Operation: Press key + Dest. Extension	Call Forward Both Ring to extension	Slowly flashes red	745
Call Forwarding, Busy	Code: 11 Operation: Press key + Dest. Extension	Call Forward Busy to extension or Voice Mail	Slowly flashes red	742
Call Forwarding, Busy/No Answer	Code: 13 Operation: Press key + Dest. Extension	Call Forward Busy/No Answer to extension or Voice Mail	Slowly flashes red	744
Call Forwarding, External by Door Box	Code: 54 Operation: Press key + Dest. Number	Externally Call Forward Door Box calls	Slowly flashes red	722
Call Forwarding, Follow Me	Code: 15 Operation: Press key + Dest. Extension	Call Forward Follow Me to extension or Voice Mail	Slowly flashes red	746

Table 3-4 Function Key Codes by Feature (Continued)

To program a key, press the Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
For this feature...	Use this key...	When you...	Key Lamp Status	Also See Srvc Code
Call Forwarding, Immediate	Code: 10 Operation: Press key + Dest. Extension	Call Forward Immediate to extension or Voice Mail	Slowly flashes red	741
Call Forwarding, No Answer	Code: 12 Operation: Press key + Dest. Extension	Call Forward No Answer to extension or Voice Mail	Slowly flashes red	743
Call Forwarding / Do Not Disturb Override	Code: 37 Operation: Call extension + Press key	Override an extension Call Forward or Do Not Disturb	None	-
Callback / Camp-On/ Trunk Queuing	Code: 35 Operation: Call busy extension or access busy trunk + Press key	Leave a Call back request at a busy extension, Camp-On to a busy extension, or Queue for a busy trunk	On red when activated	#
Call Redirect	Code: 49 + extension or voice mail Operation: Press key	Redirect a ringing call to the predefined destination	On red when activated	-
Central Office Calls	Code: *01 + Trunk number (001~200) Operation: Press key	Press a line key to place or answer a trunk call (where trunks are 001~200)	On green when seized, on red when in use (by other party), Slow Flash green when ringing, Hold flash when on Hold	#9
Conference	Code: 07 Operation: Set up call + Press key + set up call to add + Press key twice	Set up a Conference or a Meet Me Conference	On red during setup	#1
Data Communications	Code: 66 Operation: Press key + ext or outside number	Place a data call	On red when call set up	-
Department Calling	Code: 46 Operation: Press Key	Log in or log out of your Department Calling Group	On when removed, Off when installed	650
Department Step Calling	Code: 36 Operation: Dial busy ext + Press key	Step Call through a Department Group for an idle member	None	2
Direct Station Selection / One-Touch Calling	Code: 01 Operation: Press key + dest. ext. or outside tel. # + Hold	Call an extension or outside number using a DSS key	Off = extension idle On = extension busy Flashing = DND	-
Do Not Disturb	Code: 03 Operation: Press key + code (0~4)	Set your telephone in DND	DND key on red	747

Table 3-4 Function Key Codes by Feature (Continued)

To program a key, press the Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
For this feature...	Use this key...	When you...	Key Lamp Status	Also See Svc Code
Do Not Disturb/Call Forward Override	Code: 37 Operation: Press key	Call an extension which is in DND or Call Forwarded	None	-
General Purpose Relay	Code: 51 + relay number Operation: Press key	Activate the general purpose relay	On when active	780
Group Call Pickup	Code: 24 Operation: Speaker key + Press key	Answer a call ringing another telephone in your Pickup Group	None	**#
	Code: 25 Operation: Speaker key + Press key	Answer a call ringing a telephone in another Pickup Group – if you do not know the group number	None	769
	Code: 26 + Pickup Group (1~8 or 1~9 or 01~64) Operation: Speaker key + Press key + Pkup Group	Answer a call ringing a telephone in a specific Pickup Group	None	768
Handset Cutoff	Code: 40 Operation: Press key	Cut off the handset transmission while on a call	On when feature active (no transmission on handset)	-
Hotline	Code: 01 + dest. ext Operation: Press key	Place a call to your Hotline partner	Full BLF (red) for covered ext.	-
Headset Operation	Code: 05 Operation: Press key	Enable or disable Headset Operation	On red when activated	834
Hold	Code: 44 Operation: Place or answer call + Press key	Put a call on System Hold (if your telephone Hold key is reassigned)	None	-
	Code: 45 Operation: Place or answer call + Press key	Put a call on Exclusive Hold	None	-
Meet Me Conference (Also see Conference)	Code: 32 Operation: Press key	Join a Meet Me Conference	None	763 or 764

Table 3-4 Function Key Codes by Feature (Continued)

To program a key, press the Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
For this feature...	Use this key...	When you...	Key Lamp Status	Also See Srvc Code
Memo Dial	Code: 31 Operation: <u>Store:</u> While on call, press key + number to store <u>Use:</u> Press Key + Call or line <u>Erase:</u> Speaker key + Press key	Store, use or check a Memo dial number	None	-
Message Waiting	Code: 38 Operation: Leave message: Call ext + Press key OR Answer message: Press key	Answer/Leave a Message Waiting	None	*0
Microphone Cutoff	Code: 02 Operation: Set up call + Press key	Use Microphone Cutoff	On red when activated	-
Call Arrival (CAR) Keys	Code: *03 + ext. or dept group Operation: Press key	Place or answer a call to your virtual (phantom) extension	Slow Flash red when ringing, On red when busy	-
Name Storing	Code: 55 Operation: Press key + ext ## + name + Hold	Enter a name for the extension to be displayed on telephones	None	700
Networking	Code: *06 + network (01~50) Operation: Press key	Access a networked trunk	None	-
Night Service	Code: 09 + mode (1~4 or 1~8) Operation: Press key	Activate the Day/Night Mode	On red when activated	718 + 0
Off-Hook Signaling	Code: 33 Operation: Call ext. and receive busy + Press key	Signal a busy extension	None	7
Paging, External	Code: 19 + zone (1~8) Operation: Press Key	Make an external zone page	On red when activated	703 + zone
	Code: 20 Operation: Press key	Make an external All Call page	On red when activated	703 + 0
Paging, Internal	Code: 21 + zone (1~8, 1~9 or 01~64) Operation: Press key	Broadcast to an Internal Paging Zone	On red when activated	701 + zone
	Code: 2 Operation: Press key	Broadcast to all Internal Paging zones	On red when activated	701 + 0 or 00

Table 3-4 Function Key Codes by Feature (Continued)

To program a key, press the Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
For this feature...	Use this key...	When you...	Key Lamp Status	Also See Svc Code
Park	Code: *04 + orbit (1~9 or 01~64) Operation: Place or answer call + Press key	Place a call into or retrieve a call from a Park Orbit	Fast flash when orbit is busy (green at originator, red at others)	#6 (Park) *6 (pickup)
Repeat Redial	Code: 29 Operation: Place call and press key	Activate Repeat Redial while on a call	Fast Flash while system waits to redial	-
Reverse Voice Over	Code: 47 + dest. ext. Operation: Press and hold key	Initiate Reverse Voice Over	Full BLF red	-
Room Monitor	Code: 39 Operation: Press key at destination & source + ext	Activate Room Monitor	Dest. Fast Flash red, Source Hold Flash red	-
Save Number Dialed	Code: 30 Operation: <u>Save:</u> Place call + Press key <u>Redial:</u> Line or Speaker key + Press key	Save, redial or check a saved number	None	-
Secretary Call (Buzzer)	Code: 41 + sec. ext Operation: Press key	Call your secretary (using the buzzer)	On red at source Fast Flash red at destination	-
Secretary Call Pickup	Code: 42 + boss ext Operation: Press key	Are a secretary picking up a call ringing your boss's extension	On red when activated	-
Selectable Display Messaging	Code: 18 Operation: Press key + additional data if needed	Set up Call Forwarding Off-Premise, Selectable Display Messaging, VRS Park and Page and VRS Personal Greeting	Flashes red when activated	*4
Serial Call	Code: 43 Operation: Trk call + Hold + ext + Press key	Place a Serial Call to a co-worker	None	-
Step Call	Code: 36 Operation: Press key	Step through a department group	None	#
Transfer	Code: 06 Operation: Establish call + Hold + Ext + Press key	Transfer a call	None	-
Trunk Group Routing	Code: *05 Operation: Press key	Access a trunk using Trunk Group Routing	On red when active	9

Table 3-4 Function Key Codes by Feature (Continued)

To program a key, press the Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
For this feature...	Use this key...	When you...	Key Lamp Status	Also See Srvc Code
Trunk Groups	Code: *02 + TRK group (1~9 or 001~200) Operation: Press key	Use a trunk group key to access a Trunk Group	On red when active	704
Trunk Queuing	Code: 35 Operation: Hear busy tone for Trk + Press key	Camp-On or Queue for a trunk	On red while camped on	
Voice Response System (VRS) (Park and Page) (Personal Greeting)	Code: 17 Operation: Press key + device type code + requested data (depends on device selected).	Set up Call Forwarding Off-Premise, Selectable Display Messaging, VRS Park and Page and VRS Personal Greeting	Flashes red	*4
Voice Mail	Code: 83 + code (0~4) Operation: Press key	Use Voice Mail Service	Flashes slowly when monitoring	-
	Code: 77 + extension or Message Center number Operation: Press key	Call Voice Mail or leave a message	Flashes green on your key for your messages or flashes red for the Message Center	*8 or 8
	Code: 78 + 0 Operation: Set up call + Press key	Use Voice Mail Record	Slow Flash red when active	-
Voice Over	Code: 48 Operation: Hear Off-Hook Signaling tone + Press key	Initiate or respond to Voice Over	On red when responding Hold Flash red when listening	6

Table 3-5 Function Key Codes by Number

To program a key, press Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
Use this key...	For this feature...	When you...	Key Lamp Status	Also see Svc Code
Code: 01 + dest. ext. or outside tel # + Hold Operation: Press key	Direct Station Selection, Hotline, One-Touch Calling	Call an extension or outside number using a DSS key	Off = extension idle On = extension busy Flashing = DND	-
Code: 02 Operation: Set up call + Press key	Microphone Cutoff	Use Microphone Cutoff	On red when activated	-
Code: 03 Operation: Press key	Do Not Disturb	Activate DND	On red when activated	-
Code: 04 Operation: Press key	Background Music	Turn BGM on or off	On red when activated	725
Code: 05 Operation: Press key	Headset Operation	Enable or disable Headset Operation	On red when activated	734
Code: 06 Operation: Establish call + Hold + Ext + Press key	Transfer	Transfer a call	None	-
Code: 07 Operation: Set up call + Press key + set up call to add + Press key twice	Conference	Set up a conference or a Meet Me Conference	On red during setup	#1
Code: 08 Operation: Press key	Incoming Caller ID List	List incoming caller ID to extension	Flashing when new log created On in call log	-
Code: 09 + mode (1~4 or 1~8) Operation: Press key	Night Service	Activate the Day/Night Mode	On red when activated	718 + mode (1~4 or 1~8)
Code: 10 Operation: Press key + Dest. Ext.	Call Forwarding, Immediate	Call Forward to extension or Voice Mail	Slowly flashes red	741
Code: 11 Operation: Press key + Dest. Ext.	Call Forwarding, Busy	Call Forward to extension or Voice Mail	Slowly flashes red	742

Table 3-5 Function Key Codes by Number (Continued)

To program a key, press Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
Use this key...	For this feature...	When you...	Key Lamp Status	Also see Srvc Code
Code: 12 Operation: Press key + Dest. Ext.	Call Forwarding, No Answer	Call Forwarding to extension or Voice Mail	Slowly flashes red	743
Code: 13 Operation: Press key + Dest. Ext.	Call Forwarding, Busy/No Answer	Call Forwarding to extension or Voice Mail	Slowly flashes red	744
Code: 14 Operation: Press key + Dest. Ext.	Call Forwarding, Both Ring	Call Forwarding to extension	Slowly flashes red	745
Code: 15 Operation: Press key + Dest. Ext.	Call Forwarding, Follow Me	Call Forwarding to extension or Voice Mail	Slowly flashes red	746
Code: 19 + zone (1~8) Operation: Press key	Paging, External	Broadcasting to an External Paging Zone	On red when activated	703 + zone
Code: 20 Operation: Press key	Paging, External	Broadcasting to all External Paging Zones	On red when activated	703 + 0
Code: 21 + zone (1~8, 1~9 or 01~32) Operation: Press Key	Paging, Internal	Broadcasting to an Internal Paging Zone	On red when activated	701 + zone
Code: 22 Operation: Press key	Paging, Internal	Broadcasting to all Internal Paging Zone	On red when activated	701 + 0 or 00
Code: 23 Operation: Press key	Meet Me Paging	Join a Meet Me Page	None	763, 764, or 765
Code: 24 Operation: Speaker key + Press Key	Group Call Pickup	Answer a call ringing another telephone in your Pickup Group	None	**
Code: 25 Operation: Speaker key + Press key	Group Call Pickup	Answer a call ringing a telephone in another Pickup Group – if you do not know the group number	None	769
Code: 26 + Pickup Group (1~8 or 1~9 or 01~64) Operation: Speaker key + Press key + Pickup Group	Group Call Pickup	Answer a call ringing a telephone in a specific Pickup Group	None	768

Table 3-5 Function Key Codes by Number (Continued)

To program a key, press Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
Use this key...	For this feature...	When you...	Key Lamp Status	Also see Svc Code
Code: 27 Operation: Press key + bin + Line or Speaker key	Speed Dialing	Dial a stored System Speed Dialing number	None	#2 + bin
Code: 28 Operation: Press key + bin + Line or Speaker key		Dial a stored Group Speed Dialing number	None	#4 + bin
Code: 29 Operation: Place call + Press key	Repeat Redial	Activate repeat redial while on a call	Fast Flash while system waits to redial	-
Code: 30 Operation: <u>Save:</u> Place call + Press key <u>Redial:</u> Line or Speaker key + Press key	Save Number Dialed	Save, redial or check a saved number	None	715
Code: 31 Operation: <u>Store:</u> While on call, Press key + number to store <u>Use:</u> Press key + Speaker key or line <u>Erase:</u> Speaker key + Press key	Memo Dial	Store, use or check a Memo Dial number	None	-
Code: 33 Operation: Call ext. and receive busy + Press key	Off-Hook Signaling	Signal a busy extension	None	709
Code: 34 Operation: Call ext + Press key	Barge-In	Barge-In on a co-worker's conversation	None	710
Code: 35 Operation: Call busy extension or access busy trunk + Press key	Callback / Camp-On / Trunk Queuing	Leave a Callback request at a busy extension, Camping On to a busy extension, Queuing for a busy trunk	On red when activated	750
Code: 36 Operation: Dial busy ext + Press key	Department Step Calling	Step Call through a Department Group for an idle member	None	708
Code: 37 Operation: Call extension + Press key	Call Forwarding / Do Not Disturb Override	Override an extension Call Forwarding or Do Not Disturb	On red when activated	707

Table 3-5 Function Key Codes by Number (Continued)

To program a key, press Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
Use this key...	For this feature...	When you...	Key Lamp Status	Also see Srvc Code
Code: 38 Operation: Leave message: Call ext + Press key OR Answer message: Press key	Message Waiting	Answer/Leave a Message Waiting	None	*0 or 0
Code: 39 Operation: Press key at destination and source + ext	Room Monitor	Activate Room Monitor	Fast Flash red at destination, Hold Flash red at source	-
Code: 40 Operation: Press key	Handset Cutoff	Cut off the handset transmission while on a call	On when feature active (no transmission on handset)	-
Code: 41 + sec. ext. Operation: Press key	Secretary Call	Call your secretary (using the buzzer)	On red at source Fast Flash red at destination	-
Code: 42 + boss ext. Operation: Press key		Are a secretary picking up a call ringing your boss's extension.	On red when activated	-
Code: 43 Operation: TRK call + Hold + ext + Press key	Serial Call	Place a Serial Call to a co-worker	None	-
Code: 44 Operation: Place or answer call + Press key	Hold	Put a call on System Hold (if hold key is reassigned)	None	-
Code: 45 Operation: Place or answer call + Press key		Put a call on Exclusive Hold	None	-
Code: 46 Operation: Press key	Department Calling	Log in or log out of your Department Calling Group	On when removed, Off when installed	150
Code: 47 + dest. ext. Operation: Press and hold key	Reverse Voice Over	Initiate Reverse Voice Over	Full BLF red	-
Code: 48 Operation: Hear Off-Hook Signaling tones + Press key	Voice Over	Initiate or respond to Voice Over	On red when responding Hold Flash red when listening	790

Table 3-5 Function Key Codes by Number (Continued)

To program a key, press Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
Use this key...	For this feature...	When you...	Key Lamp Status	Also see Svc Code
Code: 49 + ext or voice mail number Operation: Press key	Call Redirect	Redirect an incoming call to an extension or voice mail	On red when activated Flashes when in DND/Call Forward	-
Code: 50 Operation: Press key	Account Codes	Enter Account Codes	None	* or ##
Code: 51 + relay number Operation: Press key	General Purpose Relay	Activate the general purpose relay	On when active	780
Code: 55 Operation: Do not Lift the handset + Press key + Enter extension number + Enter name + Press Hold	Name Storing	Change the name displayed on your display telephone	None	700
Code: 66 Operation: Press key + ext or outside number	Data Communications	Set up a Data Call	On red when call set up	-
Code: 83 + 0~4 Operation: Press key	Voice Mail	Use Voice Mail Service	Flashes slowly when monitoring	-
Code: 77 + extension or Message Center number Operation: Press key		Call Voice Mail or leave a message	Flashes green on your key for your messages or flashes red for the Message Center	*8 or 8
Code: 78 + Conversation Record Operation: Press key		Use Conversation Record	Flashes red when recording	-
Code: *01 + Trunk number (001~200) Operation: Press key	Central Office Calls	Press a line key to place or answer a trunk call (where trunks are 001~200)	On green when seized, on red when in use (by other party), Slow Flash green when ringing, Hold flash when on Hold	#9

Table 3-5 Function Key Codes by Number (Continued)

To program a key, press Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
Use this key...	For this feature...	When you...	Key Lamp Status	Also see Srvc Code
Code: *02 + TRK group (1~9 or 001~200) Operation: Press key	Trunk Groups	Use a trunk group key to access a Trunk Group	On red when active	704
Code: *03 + ext. or department group Operation: Press key	Call Arrival (CAR) Keys	Call Arrival (CAR) Key: Place or answer a call from your virtual (phantom) extension or Call Arrival (CAR) Key: Place or answer a call to your co-worker's extension	Slow Flash red when ringing, On red when busy	-
Code: *04 + orbit (01~64) Operation: Place or answer call + Press key	Park	Place a call into or retrieve a call from a Park Orbit	Fast Flash when orbit is busy (green at originator, red at others)	#6 (Park) *6 (pickup)
Code: *06 + Network number (1~50) Operation: Press key	Networking	Access a networked trunk	None	-

Table 3-5 Function Key Codes by Number (Continued)

To program a key, press Speaker key, dial 751 (for 2-digit codes) or 752 (for 3-digit codes), press the key and enter the code (e.g., 48 for Voice Over).				
Use this key...	For this feature...	When you...	Key Lamp Status	Also see Svc Code
Code: *10 Operation: Press key to log in Press key + 1 to log out or 0 to cancel	Automatic Call Distribution (ACD)	Basic Operation Log in or out of an ACD Group	On red when logged in Off when logged out	*5
Code: *12 Operation: Press key		Emergency Call Place or receive an Emergency Call	On while calling your supervisor or after being answered by your supervisor Flashing fast at the supervisor while ringing	-
Code: *13 Operation: Press key		Rest Mode Enable/Disable Rest Mode	On red when Rest Mode enabled Off when Rest Mode disabled	-
Code: *14 Operation: Press key + Press 1 (Yes) or 2 (No)		Out of Service Take an ACD Group out of service (for Group Supervisors only), or Take all ACD Groups out of service (for System Supervisors only)	On red when a group is out of service	-
Code: *15 Operation: Call busy ACD agent + Press key		Terminal Monitor Monitor an ACD Agent's conversation	On red while monitoring, Off when not monitoring	-
Code: *16 Operation: Press key to put agent on hold. Press key again + 1 to hang up agent or 0 to bring agent back into call.		Supervisor Split Switch (split) between an ACD Agent and their outside caller after answering an emergency call	On while agent is on hold	-
Code: *17 Operation: Press Key		Work Time Enable/disable Work Time	On when Work Time enabled, Flashing (while on a call) if Auto Work time enabled Off when Work Time disabled	-
Code: *18 + ACD Group Number Operation: Press key		Enable ACD overflow	On red when activated Slowly flashes red when disabled	-
Code: *19 Operation: Press key while on-hook + Vol Up or Vol Down to scroll		Queue Status Check View the ACD Queue Status of each ACD group	None	-



Feature Availability by Software Revision

SECTION 1 FEATURE AVAILABILITY CHART

This chapter provides an alphabetical listing of the features that are available with each software revision.

Note: the following table provides a breakout of the availability of each feature by revision, see [Table 4-1 Feature Availability by Software Revision](#).

S = Supported Feature

N/A = Feature not supported for this software release

E = Supported and Enhanced

Table 4-1 Feature Availability by Software Revision

IPK II Feature Name	Ver. 1000	Ver. 1100	Ver. 1500	Ver. 1600	Ver. 2000	Ver. 2100
Account Code Entry	S	S	S	S	S	S
Account Code – Forced/Verified/Unverified	S	S	S	S	S	S
Alarm	S	S	S	S	S	S
Alarm Reports	N/A	N/A	E	S	S	S
Alphanumeric Display	S	S	S	S	S	S
Analog Communications Interface (ACI)	S	S	S	S	S	S
Ancillary Device Connection	S	S	S	S	S	S
Answer Hold	S	S	S	S	S	S
Answer Key	S	S	S	S	S	S
Attendant Call Queuing	S	S	S	S	S	S
Automatic Call Distribution (ACD)	S	E	S	E	E	S
Automatic Release	S	S	S	S	S	S
Automatic Route Selection	S	S	S	S	S	S
Background Music	S	S	S	S	S	S
Barge-In	S	S	S	S	S	S
Battery Backup – System Memory	S	S	S	S	S	S
Battery Backup – System Power	S	S	S	S	S	S
Call Appearance (CAP) Keys	S	S	S	S	S	S
Call Arrival (CAR) Keys	S	S	S	S	S	S
Call Duration Timer	S	S	S	S	S	S
Call Forwarding	S	S	E	S	S	S
Call Forwarding – Centrex	N/A	N/A	E	E	S	E
Call Forwarding/ Do Not Disturb Override	S	S	S	S	S	S
Call Forwarding Park and Page	S	S	S	S	S	S
Call Forwarding with Follow Me	S	S	S	S	S	S
Call Forwarding, Off-Premise	S	S	S	S	S	S
Call Monitoring	S	S	E	S	S	S

Table 4-1 Feature Availability by Software Revision (Continued)

IPK II Feature Name	Ver. 1000	Ver. 1100	Ver. 1500	Ver. 1600	Ver. 2000	Ver. 2100
Call Redirect	S	S	S	S	S	S
Call Waiting/Camp-On	S	S	S	S	S	S
Callback	S	S	S	S	S	S
Caller ID	S	S	S	E	S	E
Caller ID Call Return	S	S	S	S	S	S
Central Office Calls, Answering	S	S	S	S	S	S
Central Office Calls, Placing	S	S	S	S	S	S
Class of Service	S	S	S	E	E	E
Clock/Calendar Display	S	S	S	E	S	E
CO Message Waiting Indication	S	S	S	S	S	S
Code Restriction	S	S	S	S	S	S
Code Restriction Override	S	S	S	S	S	S
Code Restriction, Dial Block	S	S	S	S	S	S
Computer Telephony Integration (CTI) Applications	S	S	S	S	S	S
Conference	S	S	S	E	S	S
Conference, Voice Call/Privacy Release	S	S	S	S	S	S
Continued Dialing	S	S	S	S	S	S
Cordless Telephone Connection	S	S	S	S	S	S
Data Line Security	S	S	S	S	S	S
Delayed Ringing	S	S	S	S	S	S
Department Calling	S	S	S	S	S	S
Department Step Calling	S	S	S	S	S	S
Dialing Number Preview	S	S	S	S	S	S
Dial Pad Confirmation Tone	S	S	S	S	S	S
Dial Tone Detection	S	S	S	S	S	S
Digital Trunk Clocking	N/A	N/A	N/A	S	E	S
Digital Voice Mail	S	S	S	E	S	S
Direct Inward Dialing (DID)	S	S	E	S	S	S

Table 4-1 Feature Availability by Software Revision (Continued)

IPK II Feature Name	Ver. 1000	Ver. 1100	Ver. 1500	Ver. 1600	Ver. 2000	Ver. 2100
Direct Inward Line (DIL)	S	S	S	S	S	S
Direct Inward System Access (DISA)	S	S	S	S	S	S
Direct Station Selection (DSS) Console	S	S	E	E	E	S
Directed Call Pickup	S	S	S	S	S	S
Directory Dialing	S	S	E	E	S	S
Distinctive Ringing, Tones and Flash Patterns	S	S	E	E	S	S
Do Not Disturb	S	S	E	S	S	S
Door Box	S	S	S	S	S	S
Drop Key	S	S	S	S	S	S
<i>D^{term}</i> Cordless II Terminal	S	S	S	S	S	S
<i>D^{term}</i> Cordless Lite II Terminal	S	S	S	S	S	S
<i>D^{term}</i> Handset Cordless	S	S	S	S	S	S
<i>D^{term}</i> IP Gateway System	S	S	S	S	S	S
E911 Compatibility	S	S	S	S	S	S
Electra Elite IPK Terminals	S	S	S	S	S	S
Electra Elite Terminal Migration	S	S	S	S	S	S
Elite CallAnalyst	S	S	S	S	S	S
EliteApps – Interactive Voice Response	S	S	S	S	S	S
Facsimile CO Branch Connection	S	S	S	S	S	S
Flash	S	S	S	S	S	S
Flexible System Numbering	S	S	S	E	E	S
Flexible Timeouts	S	S	S	S	S	S
Forced Trunk Disconnect	S	S	S	S	S	S
Group Call Pickup	S	S	S	S	S	S
Group Listen	S	S	S	S	S	S
Handset Mute	S	S	S	S	S	S
Handsfree and Monitor	S	S	S	S	S	S

Table 4-1 Feature Availability by Software Revision (Continued)

IPK II Feature Name	Ver. 1000	Ver. 1100	Ver. 1500	Ver. 1600	Ver. 2000	Ver. 2100
Handsfree Answerback/Forced Intercom Ringing	S	S	S	S	S	S
Headset Operation	S	S	S	S	S	S
Hold	S	S	S	S	S	S
Hotel/Motel	S	S	E	E	E	S
Hot Key-Pad	N/A	N/A	S	S	S	S
Hotline	S	S	S	S	S	S
Howler Tone Service	S	S	S	S	S	S
Intercom	S	S	S	S	S	S
Internal Hub	S	S	S	S	S	S
IP Extenders/Mobile ConneX	N/A	N/A	N/A	N/A	S	S
IP Station (MEGACO) – IAD Integrated Access Device	N/A	S	E	S	S	S
IP Station (MEGACO) – MG 16	S	S	E	E	E	E
IP Station (SIP) – MG16	N/A	N/A	E	E	E	E
IP Trunk – H.323 Protocol	S	S	S	S	S	S
IP Trunk – (SIP) Session Initiation Protocol	S	S	S	E	S	S
IP Trunk (SIP) – MG16	N/A	N/A	E	S	S	S
IPK II In-Mail	S	S	E	E	E	S
IPK II – PC Assistant	S	S	E	E	S	S
IPK II – PC Attendant	S	S	E	E	S	S
IPK II VoIP Management System	N/A	N/A	S	S	S	S
ISDN Compatibility	S	S	E	S	S	S
K-CCIS – IP with IAD	S	S	S	E	E	E
K-CCIS – IP with PVA	N/A	S	E	E	E	E
K-CCIS – T1	S	S	S	E	E	S
Last Number Redial	S	S	S	S	S	S
Licensing	N/A	N/A	E	E	S	S
Line Preference	S	S	S	S	S	S

Table 4-1 Feature Availability by Software Revision (Continued)

IPK II Feature Name	Ver. 1000	Ver. 1100	Ver. 1500	Ver. 1600	Ver. 2000	Ver. 2100
Long Conversation Cutoff	S	S	S	S	S	S
Meet Me Conference	S	S	S	S	S	S
Meet Me Paging	S	S	S	S	S	S
Meet Me Paging Transfer	S	S	S	S	S	S
Memo Dial	S	S	S	S	S	S
Message Waiting	S	S	S	E	S	S
Microphone Cutoff	S	S	S	S	S	S
Multiline Conference Bridge	S	S	S	S	S	S
Multimedia Conference Bridge	N/A	S	S	S	S	S
Multiple Trunk Types	S	S	S	S	S	S
Music on Hold	S	S	S	S	S	S
Name Storing	S	S	S	E	S	S
Night Service	S	S	S	S	S	S
Off-Hook Signaling	S	S	S	S	S	S
One-Digit Dial Option	N/A	N/A	N/A	S	S	S
One-Touch Calling	S	S	S	S	S	S
Operator	S	S	E	S	S	S
(OPX) Off-Premise Extension	S	S	S	S	S	S
Paging, External	S	S	S	S	S	S
Paging, Internal	S	S	S	S	S	S
Park	S	S	S	E	E	S
PBX Compatibly	S	S	S	S	S	S
PC Programming	S	E	S	E	S	E
Power Failure Transfer	S	S	S	S	S	S
Prime Line Selection	S	S	S	S	S	S
Private Line	S	S	S	S	S	S
Programmable Function Keys	S	S	E	E	S	S
Programming from Multiline Terminal	S	E	S	S	S	S

Table 4-1 Feature Availability by Software Revision (Continued)

IPK II Feature Name	Ver. 1000	Ver. 1100	Ver. 1500	Ver. 1600	Ver. 2000	Ver. 2100
Pulse to Tone Conversion	S	S	S	S	S	S
Quick Transfer to Voice Mail	S	S	S	E	S	S
Redial Key	S	S	S	S	S	S
Remote (System) Upgrade	S	S	S	S	S	S
Repeat Redial	S	S	S	S	S	S
Resident System Program	S	S	S	S	S	S
Reverse Voice Over	S	S	S	S	S	S
Ring Groups	S	S	S	E	S	S
Ringdown Extension, Internal/External	S	S	S	S	S	S
Room Monitor	S	S	S	S	S	S
Save Number Dialed	S	S	S	S	S	S
Secondary Incoming Extension	S	S	S	S	S	S
Secretary Call (Buzzer)	S	S	S	S	S	S
Secretary Call Pickup	S	S	S	S	S	S
Selectable Display Messaging	S	S	S	S	S	S
Selectable Ring Tones	S	S	S	S	S	S
Serial Call	S	S	S	S	S	S
Single Line Telephones, Analog 500/2500 Sets	S	S	S	S	S	S
SLT Adapter	S	S	S	S	S	S
SNMP Simple Network Management Protocol	N/A	N/A	N/A	S	S	S
Softkeys	S	S	S	S	S	S
Speed Dial – System/Group/Station	S	S	S	S	S	S
Station Add-On Console	S	S	E	S	S	S
Station Hunt	S	S	S	S	S	S
Station Message Detail Recording	S	S	S	S	S	S
Station Name Assignment User Programmable	S	S	S	S	S	S
Station Relocation	S	S	S	S	S	S

Table 4-1 Feature Availability by Software Revision (Continued)

IPK II Feature Name	Ver. 1000	Ver. 1100	Ver. 1500	Ver. 1600	Ver. 2000	Ver. 2100
Synchronous Ringing	S	S	S	S	S	S
T1 Trunking (with ANI/DNIS Compatibility)	N/A	S	E	S	S	S
Tandem Ringing	S	S	S	S	S	S
Tandem Trunking (Unsupervised Conference)	S	S	S	S	S	S
TAPI Compatibility	S	S	E	E	S	S
Tone Override	S	S	S	S	S	S
Traffic Reports	S	S	S	S	S	S
Transfer	S	S	E	S	S	S
Trunk Group Routing	S	S	S	S	S	S
Trunk Groups	S	S	S	S	S	S
Trunk Queuing/Camp-On	S	S	S	S	S	S
Unified Messaging	S	S	S	S	S	S
Uniform Call Distribution (UCD)	S	S	S	S	S	S
Uniform Numbering Network	S	S	S	S	S	S
Universal Slots	S	S	E	S	S	S
User Programming Ability	S	S	S	S	S	S
Virtual Extensions	S	S	S	E	S	S
Voice Mail Integration (Analog)	S	S	E	E	S	E
Voice Mail Message Indication on Line Keys	S	S	S	S	S	S
Voice Over	S	S	S	S	S	S
Voice Over Internet Protocol (VoIP)	S	S	S	S	S	S
Voice Response System (VRS)	S	S	S	E	E	S
Volume Controls	S	S	S	S	S	S
Warning Tone For Long Conversation	S	S	S	S	S	S
Wireless – DECT	S	S	S	S	S	S

NEC

Electra **Elite** IPK II
FEATURES AND SPECIFICATIONS
MANUAL