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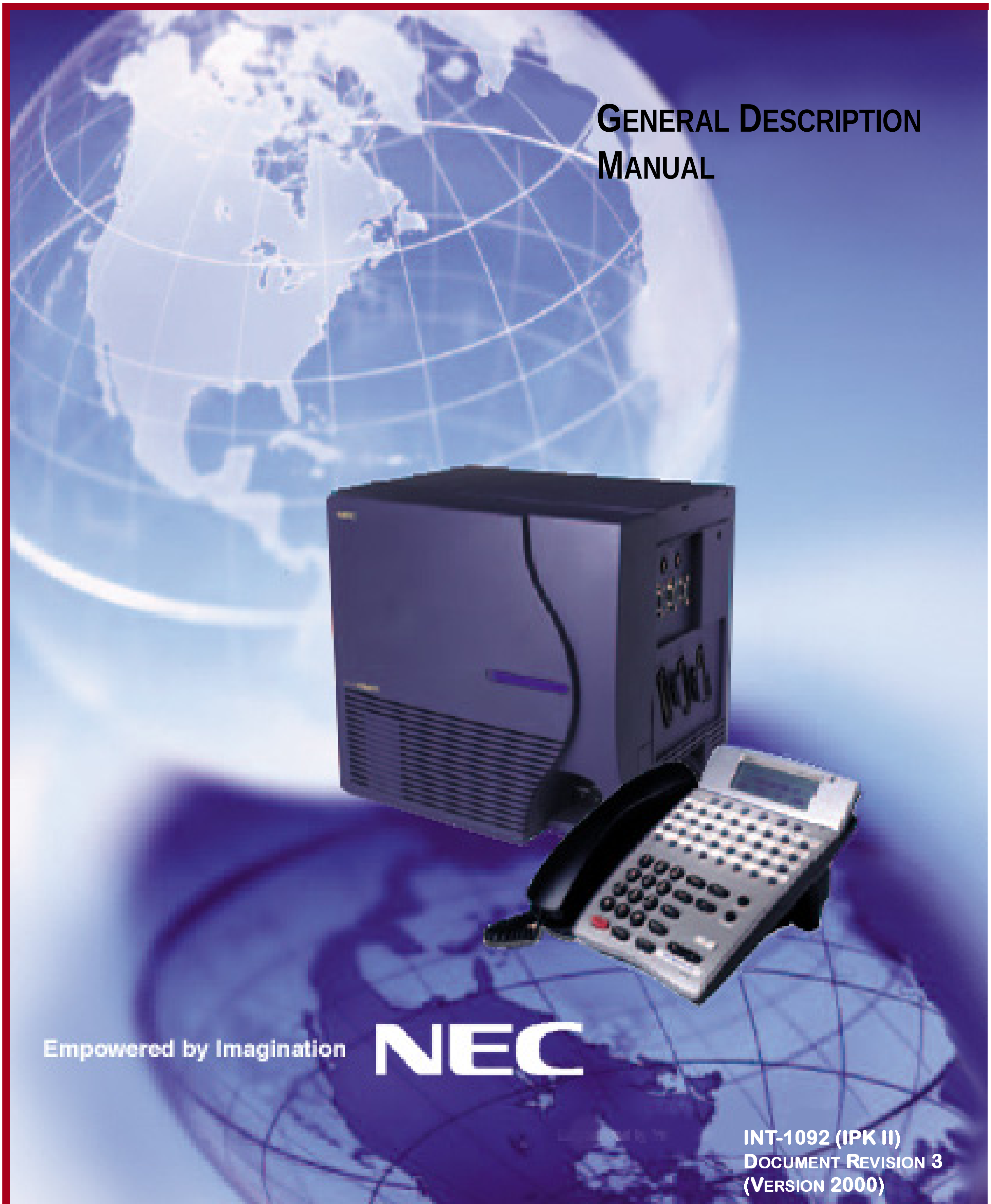
Electra Elite IPK II



General Description Manual

Empowered by Imagination

NEC

INT-1092 (IPK II)
Document Revision 3
(Version 2000)

NEC Infrontia, Inc.
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Irving, TX 75039-2402

Technology Development

Preface

GENERAL INFORMATION

The Electra Elite IPK II system is a feature-rich key system that provides over 170 features including Computer Telephony Integration, Automatic Call Distribution, Uniform Call Distribution, ISDN-BRI Voice Trunks, ISDN-PRI Voice Trunks, IP Telephony, Voice over Internet Protocol (VoIP) trunks and stations, and many others.

The Electra Elite IPK II system provides what 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. The manuals are described in this preface.

THIS 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 is useful when presenting information to potential customers.

Chapter 1 – Introduction

This chapter provides an overview and a brief description of the system.

Chapter 2 – Features

This chapter provides a list of features that are available with the system. Each feature is briefly described.

Chapter 3 – Equipment

This chapter provides a list and brief description of the equipment that is available with the system.

Chapter 4 – Installation, Programming, and Maintenance Overview

This chapter briefly describes the installation, programming functions, and maintenance of the system.

Chapter 5 – Hardware Specifications

This chapter provides requirements and specifications relating to the system hardware. This chapter is helpful to those that install the system.

SUPPORTING DOCUMENTS

Electra Elite IPK II Features and Specifications Manual

This manual provides detailed information concerning every feature available in the system.

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.

Electra Elite IPK II PC Programming Manual

This manual describes the operation of the PCPro program for the .Electra Elite IPK II key telephone system. This program is a user-friendly Windows application that allows the user to program and configure features of the Electra Elite IPK II KTS from the PC environment.

Regulatory

GENERAL INFORMATION

Established Federal Communications Commission (FCC) rules permit this telephone system to be directly connected to the telephone network. A jack is provided by the telephone company. Jacks for this type of customer provided equipment are not provided on party lines or coin lines.

The telephone company may make changes in its technical operations and procedures. When such changes affect the compatibility or use of the Electra Elite IPK II system, the telephone company is required to give adequate notice of the changes.

COMPANY NOTIFICATION

Before connecting this telephone system to the telephone network, the following information must be provided to the telephone company:

1. Your telephone number.
2. FCC registration number:

- ❑ When the system is to be installed as a Key Function system (no dial access to Trunk Groups/ Route Advance Blocks), use the following number:

  NIFMUL-43074-KF-E

- ❑ When the system is to be installed as a Multifunction system, use the following number:

  NIFMUL-43076-MF-E

- ❑ When the system is to be installed as a PBX Function system, use the following number:

  NIFMUL-43075-PF-E

- ❑ Ringer Equivalence Number (REN): **2.0B**
- ❑ USOC jacks required: **RJ21X** and **RJ2GX**

The following table lists the Facility Interface Codes (FIC), Ringer Equivalent Numbers (REN), Service Order Codes (SOC), and Jack Types for the interface ETUs.

Table 1 FIC, REN, SOC, and Jack Types for Electra Elite IPK II System ETUs

Trunk/Station ETU Type	FIC	REN	SOC	Jack
BRT(4)-U( ) ETU	02IS5	N/A	6.0F	N/A
CAMA Trunk	02RV-O	0.7A	9.0F	RJ21X
COI(4)-U( ) ETU (Loop Start)	02LS2	0.7A	9.0F	RJ21X
COI(8)-U( ) ETU (Loop Start)	02LS2	0.7A	9.0F	RJ21X
COI(8)-U( ) ETU (Ground Start)	02GS2	0.7A	9.0F	RJ21X
COIB(4)-U(10) ETU ETU for COI/ COID Mode (Loop Start)	02GS2	0.7A	9.0F	RJ21X
COIB(4)-U(10) ETU ETU for COI Mode (Ground Start)	02GS2	0.7A	9.0F	RJ21X
COIB(4)-U(20) ETU ETU for COID/COI Mode (Loop Start)	02LS2	0.7A	9.0F	RJ21X
COIB(8)-U( ) ETU for COI/COID Mode (Loop Start)	02LS2	0.7A	9.0F	RJ21X
DID(4)-U( ) ETU	02RV2T	N/A	9.0F	RJ21X
DTI-U( ) ETU	04DU9-BN 04DU9-DN 04DU9-1KN 04DU9-1SN	N/A	6.0P	N/A
OPX(2)-U( ) ETU	0L13C	N/A	9.0F	RJ21X
TLI(2)-U( ) ETU	TL31M	N/A	9.0F	RJ21X

INCIDENCE OF HARM

When the system is malfunctioning, it may also be causing harm to the telephone network. The telephone system should be disconnected until the source of the problem can be determined and until repair has been made. When this is not done, the telephone company may temporarily disconnect service.

RADIO FREQUENCY INTERFERENCE

In compliance with FCC Part 15 rules, the following statement is provided:

IMPORTANT NOTE

“This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the System Hardware Manual, may cause interference to radio communications. This equipment has been tested and approved for compliance with the limits for a Class B (except as noted below) computing device pursuant to subpart J of Part 15 of FCC Rules, that are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this telephone system in a residential area is likely to cause interference, in which case, the user, at his or her own expense, is required to take whatever measures may be required to correct the interference.”

✎ *When equipped with the B64-U30 KSU and P64-U20 PSU, the Electra Elite IPK II can be operated as a Class B device except when using one of the ETUs in the following table. The system then becomes a Class A device that may not be used in a residential area.*

CCH(4)-U-10	CMS(2)/(4)-U30	FMS(2)/(4)/(8)-U30
HUB(8)-U10	VMS(2)/(4)/(8)-U30	

HEARING AID COMPATIBILITY

The NEC Multiline Terminals and NEC Single Line Telephones that are provided for this system are hearing aid compatible. The manufacturer of other Single Line Telephones for use with the system must provide notice of hearing aid compatibility to comply with FCC rules that now prohibit the use of non-hearing aid compatible telephones.

DIRECT INWARD DIALING

Operating this equipment without providing proper answer supervision is a violation of Part 68 of the FCC rules.

Proper Answer Supervision occurs when:

This equipment returns answer supervision to the Public Switched Telephone Network (PSTN) when Direct Inward Dialing (DID) calls are:

- ❑ Answered by the called station.
- ❑ Answered by the Attendant.
- ❑ Routed to a recorded announcement that can be administered by the Customer Premise Equipment (CPE) user.
- ❑ Routed to a dial prompt.

This equipment returns answer supervision on all DID calls forwarded to the Public Switched Telephone Network (PSTN). Permissible exceptions are:

- A call is unanswered.
- A busy tone is received.
- A reorder tone is received.

VOICE ANNOUNCEMENT/ MONITORING OVER DID LINES

CAUTION

Using the Voice Announcement feature to eavesdrop or record sound activities at the other end of the telephone line may be illegal under certain circumstances and laws. Consult a legal advisor before implementing any practice to monitor or record a telephone conversation. Some federal and state laws require a party monitoring or recording a telephone to use a beep-tone(s), notify all parties to the telephone conversation and/or obtain consent of all parties to the telephone conversation. In monitoring or recording sound activities at the other end of the telephone line using the Voice Announcement feature, the sound of the alert tone at the beginning of the Voice Announcement may <u>or may not</u> be considered sufficient under applicable laws. Some of the applicable laws provide for strict penalties for illegal monitoring or recording of telephone conversations.

MUSIC ON HOLD

IMPORTANT NOTE

"In accordance with U.S. Copyright Law, a license may be required from the American Society of Composers, Authors and Publishers, or other similar organization, if radio or TV broadcasts are transmitted through the Music On Hold feature of this telecommunication system. NEC Unified Solutions, Inc., hereby disclaims any liability arising out of the failure to obtain such a license."

SERVICE REQUIREMENTS

If equipment malfunctions, all repairs must be performed by an authorized agent of NEC Unified Solutions, Inc. or by NEC Unified Solutions, Inc. The user requiring service is responsible for reporting the need for service to an NEC Unified Solutions, Inc. authorized agent or to NEC Unified Solutions, Inc.

UL REGULATORY INFORMATION

This equipment has been listed by Underwriters Laboratories and found to comply with all applicable requirements of the standard for telephone equipment UL 1459.

INDUSTRY CANADA REQUIREMENTS

Industry Canada has established rules that permit this telephone system to be directly connected to the telephone network. Prior to the connection or disconnection of this telephone system to or from the telephone network, the telephone company must be provided with the following information.

1. Your telephone number.

2. IC registration number: **140 7942 A**

3. Ringer Equivalence Number (REN) of the equipment: **2.1**

The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements as prescribed in the applicable Terminal Equipment Technical Requirements document(s). The Department does not guarantee that equipment operates to the user satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, When present, are connected together. This precaution may be particularly important in rural areas.

CAUTION

Users should not attempt to make such connections themselves, but should contact the applicable electric inspection authority or electrician.

The Ringer Equivalence Number (REN) assigned to each terminal device provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The termination of an interface may consist of any combination of devices subject only to the requirement that the sum of the Ringer Equivalent Numbers of all the devices does not exceed five.

This equipment has been listed by the Canadian Standards Association and found to comply with all applicable requirements of the standard for telephone equipment **C 22.2 No. 225**.

This equipment meets IC requirements **CS03**.

This digital apparatus does not exceed the Class A limits for radio noise emissions from digital apparatus as set out in the radio interference regulations of Industry Canada.

Le present appareil numerique n’emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques de Classe A prescrites dans le reglement sur le brouillage radioelectrique edicte par Industrie Canada.

BATTERY DISPOSAL

The Electra Elite IPK II system includes the batteries listed below. When disposing of these batteries, KSUs, and/or ETUs, you must comply with applicable federal and state regulations regarding proper disposal procedures.

Table 2 Battery Types and Quantities for KSUs and ETUs

Unit Name	Type of Battery	Quantity
B64-U20 KSU	Lead Acid	2
CPUII( )-U10 ETU	Lithium	1
CTI/VP(4)/(8)/(12)/(16)-U( ) ETU	Lithium	1
DTP-1HM-1 TEL DTP-1HM-2 TEL	Lithium	1
DTP-16HC-1 TEL	Nickel-Cadmium	1
DTR-1HM-1 TEL	Lithium	1
DTR-4R-1 TEL	Nickel-Cadmium	1
DTU-4R-1 TEL	Lead Acid	1
FMS(2)/(4)/(8)-U( ) ETU	Nickel-Cadmium	1
VMS(2)/(4)/(8)-U( ) ETU	Lithium	1

The Electra Elite IPK II CPUII( )-U10 ETU provides memory backup for approximately three years. The Lithium battery should be replaced every two years.

IMPORTANT SAFEGUARDS FOR BATTERY DISPOSAL

DO NOT PLACE USED BATTERIES IN YOUR REGULAR TRASH! THE PRODUCT YOU PURCHASED CONTAINS LITHIUM, NICKEL-CADMIUM OR SEALED LEAD BATTERY. LITHIUM, NICKEL-CADMIUM OR SEALED LEAD BATTERIES MUST BE COLLECTED, RECYCLED, OR DISPOSED OF IN AN ENVIRONMENTALLY SOUND MANNER.

The incineration, landfilling or mixing of nickel-cadmium or sealed lead batteries with the municipal solid waste stream is PROHIBITED BY LAW in most areas. Contact your local solid waste management officials for other information regarding the environmentally sound collection, recycling, and disposal of the battery.

Nickel-Cadmium (or sealed lead) batteries must be returned to a federal or state approved nickel-cadmium (or sealed lead) battery recycler. This may be where the batteries were originally sold or a local seller of automotive batteries. Contact your local waste management officials for other information regarding the environmentally sound collection, recycling and disposal of the battery contained in this product. For Ni-Cd batteries, you can also call 1-800-8-BATTERY℠ when further information is required.

The packaging for the Electra Elite IPK II system contains the following labels regarding proper disposal.

PRODUCT PACKAGE LABELING

Ni-Cd

CONTAINS NICKEL-CADMIUM BATTERY. BATTERY MUST BE RECYCLED OR DISPOSED OF PROPERLY. MUST NOT BE DISPOSED OF IN MUNICIPAL WASTE.

Pb

CONTAINS SEALED LEAD BATTERY. BATTERY MUST BE RECYCLED. MUST NOT BE DISPOSED OF IN MUNICIPAL WASTE.

Ni-MH

CONTAINS NICKEL-METAL HYDRIDE BATTERY. BATTERY MUST BE RECYCLED OR DISPOSED OF PROPERLY. MUST NOT BE DISPOSED OF IN MUNICIPAL WASTE.

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Introduction

SECTION 1 SYSTEM OVERVIEW

The Electra Elite IPK II system is a complete communications system that enhances productivity and controls costs. Its objectives are based on four Es – Easy to Install, Easy to Maintain, Easy to Expand, and Easy to Use – all at a reasonable price. The Electra Elite IPK II, like all NEC communications products, is user-friendly, reliable, and cost-effective.

- ❑ Easy to Install

  With the Electra Elite IPK II system, NEC has reduced the number of hardware components, making the system easier to install. Only 1-pair wire is required to connect telephones. This system provides Windows 98 SE or higher PC programming, with a menu-driven guide, to both simplify and speed installation. All programming information and station labels can be printed as completed. For further convenience and versatility, end-user programming is provided for up to approximately 35 features.

- ❑ Easy to Maintain

  When system memory failure occurs, PC Programming software can be used locally or from a remote location to upload/download all system data. Each Electronic Telephone Unit (ETU) except those required to sustain system operation (*e.g.*, CPU, VMS) can be installed or removed (hot swap) without shutting down the system. Other considerations for easy maintenance include:

  - ❍ Standard Amphenol Connectors
  - ❍ Built-in RS-232 connectors for all communication needs
  - ❍ Standard Station wiring for DTR and DTH Multiline Terminals
  - ❍ Compact KSU
  - ❍ Flash ROM for software upgrades
  - ❍ Flash ROM upgrade using PC programming

- Easy to Expand

  The Electra Elite IPK II system offers a single cabinet that is used for the Basic and two Expansion KSUs to provide easy and cost effective growth using universal slots to enhance system configuration.

  The Electra Elite IPK II B64-U( ) KSU offers eight 16-port interface slots (or 128 ports). The system can be expanded to a maximum of 416 ports by adding two expansion cabinets. The first expansion cabinet provides an additional 128 ports. A second expansion cabinet provides another 128 ports for a maximum of 384 ports. There are 32 common ports to bring the overall total to 416 ports.

- Easy to Use

  The Electra Elite IPK II system is Centrex compatible to allow maximum flexibility and ease of use. One-Touch key access can be programmed for most features, including Centrex options and Speed Dial abilities. A voice prompt can be provided to help a user make calls. Voice Mail integration, Automated Attendant, and personalized messaging all give the system that personal touch so important in a well-run business. Most communication equipment can be connected to this system including facsimile machines and modems. The user-friendly, cost-effective programs can be updated with future enhanced system upgrades, minimizing confusion about software levels, documentation, and configuration requirements.

- Unique Design

  The Electra Elite IPK II system is a powerful key system that can meet the ever changing communications demands of current businesses. Its unique compact design allows the system to be easily and quickly installed.

  The Electra Elite IPK II system can grow with your business. You can easily and economically add slots when necessary. Two expansion units can be added to provide a total capacity of 24 interface slots.

The feature-rich Electra Elite IPK II system provides the telephone functions and supports advance features such as:

- Automatic Number Indication (ANI)/Caller ID
- Automatic Call Distribution (ACD)
- Automatic Route Selection
- Caller ID Call Return
- Centralized Voice Mail
- Computer Telephony Integration (CTI)
- Dialed Number Indication Service (DNIS)

- *D*^{term} Cordless II Terminal
- *D*^{term} Handset Cordless
- *D*^{term} Headset Cordless
- *D*^{term} Cordless II Lite Terminal
- E911 Compatibility
- Integrated Digital Voice Mail
- ISDN-BRI and ISDN-PRI Voice Trunks
- K-CCIS Common Channel Interoffice Signaling
- Live Monitoring
- Live Record
- Multiline Conference Bridge
- Multilingual LCD Indication
- Multiple Music on Hold Interface
- PC Attendant Console
- Unified Messaging
- Universal Slots
- Voice over Internet Protocol (VoIP) and stations
- Wireless
- IP Telephony
- Optional 33.6 kbps Modem for Remote Programming and Maintenance

SECTION 2 MULTILINE TELEPHONES

The Electra Elite IPK II system offers a variety of Multiline Terminals that are compatible with the system, available in 8-line, 16-line, or 32-line capacity, and offered as display or non-display terminals. A 2-line non-display terminal and 60-line Attendant Console are also available.

A customer with existing Electra Elite terminals can be easily connected to the Electra Elite IPK II system, providing inexpensive migration. Most Electra Elite IPK II system features are available with the Electra Elite Terminals.

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

- ❑ Electra Elite IPK Terminals

  The Electra Elite IPK Terminals (DTH/ITH telephones) offer a variety of colors, display and non-display and line sizes:

  - ❍ Terminals are available in black or white.
  - ❍ Terminals are available with or without an LCD display. The large Liquid Crystal Display (LCD) on the display terminal provides call status data and programming information.
  - ❍ Line size includes: 4-line, 8-line, 16-line, or 32-line.
  - ❍ IP terminals are available in 4-line, 8-line, or 16-line (with LCD).
  - ❍ Speakerphones with full handsfree operation and headset jacks are standard.
  - ❍ All (except IP telephones) are compatible with the AD(A)-R, AP(A)-R, AP(R)-R, CT(A)-R Unit and CT(U)-R Unit adapters. The AP(R)-R Unit requires an AC-R Unit to supply AC power.
  - ❍ An Attendant Add-On DCR-60-1 CONSOLE is available with 60 station, outside line, and or function key assignments.

- ❑ Electra Elite IPK II Terminal Feature Access Keys

  - ❍ Feature Access Keys.

    Depending on the type, a Multiline Terminal can have 2, 4, 8, 16, or 32 line keys. These highly-flexible keys can be used for station DSS/BLF or Programmable Feature Keys.

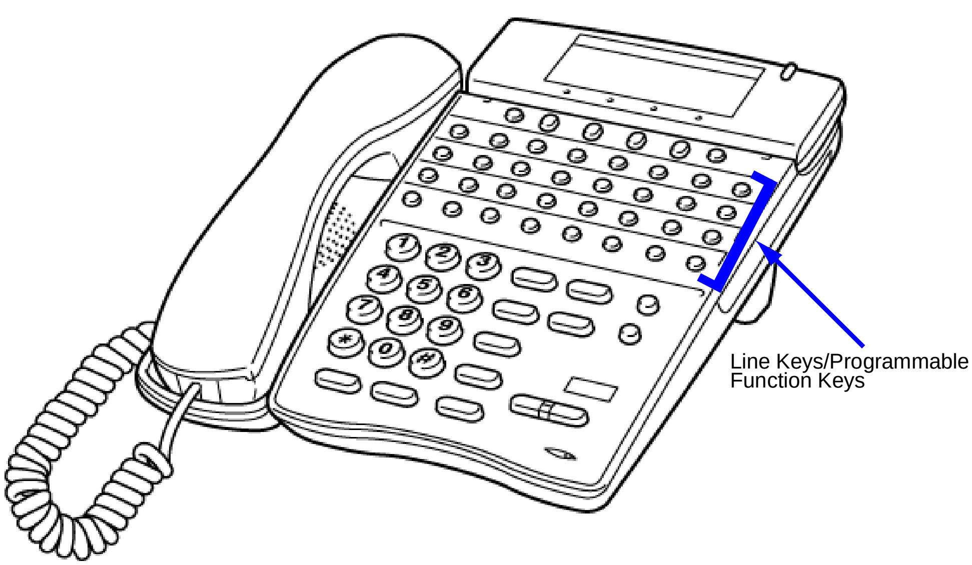


Figure 1-1 Key Assignment Example

- ❑ D^{term} Series i Terminals

The D^{term} Series i Terminals (DTR telephones) offer a variety of colors and line sizes:

- ❍ Terminals are available in black or white.
- ❍ Terminals with or without LCD are available. The large Liquid Crystal Display (LCD) on the display terminals provides call status data and programming information.
- ❍ Line size includes: 4-line, 8-line, 16-line or 32-line.
- ❍ The DTR-2DT-1 has only 2-lines.
- ❍ Speakerphones with full handsfree operation and headset jacks are standard (except on the DTR-2DT-1).
- ❍ All but the DTR-2DT-1 and DTR-4D-1 are compatible with the AD(A)-R, AP(A)-R, AP(R)-R and CT(A)-R Unit adapters. The AP(R)-R Unit requires an AC-R Unit to supply AC power. For Attendant Positions, an Attendant Add-On DCR-60-1 CONSOLE is available with 60 station, outside line, and or function key assignments.
- ❍ The DTR-2DT-1 is 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) without ringer.
- ❍ The Electra Elite IPK Single Line Terminal is 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 functions. The Electra Elite IPK Single Line Terminals come standard with a Message Waiting Indicator that also functions as an Incoming Call Indicator. 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, or Loud), and three ring patterns (Slow, Medium, or 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 keys (maximum 21 digits each). The DTR-1HM-1 also has Hold and Monitor Function keys.

- ❑ Electra Elite and D^{term} Series E Terminals

The Electra Elite Terminals (DTU telephones) and D^{term} Series E terminals (DTP telephones) are available in a variety of colors and line sizes:

- ❍ Terminals are available in black or white.
- ❍ Terminals are available with or without an LCD. The large Liquid Crystal Display (LCD) on the display terminal provides call status data and programming information.
- ❍ Line sizes include: 8-line, 16-line and 32-line.
- ❍ Speakerphones with full handsfree operation and headset jacks are standard.
- ❍ The D^{term} Handset Cordless terminal is a 16-button phone (display only).
- ❍ An Attendant Add-On DCR-60-1 CONSOLE is available with 60 station, outside line, and or function key assignments.
- ❍ An SLT Adapter can be used in place of a digital terminal for connecting Single Line Telephones, or similar devices.

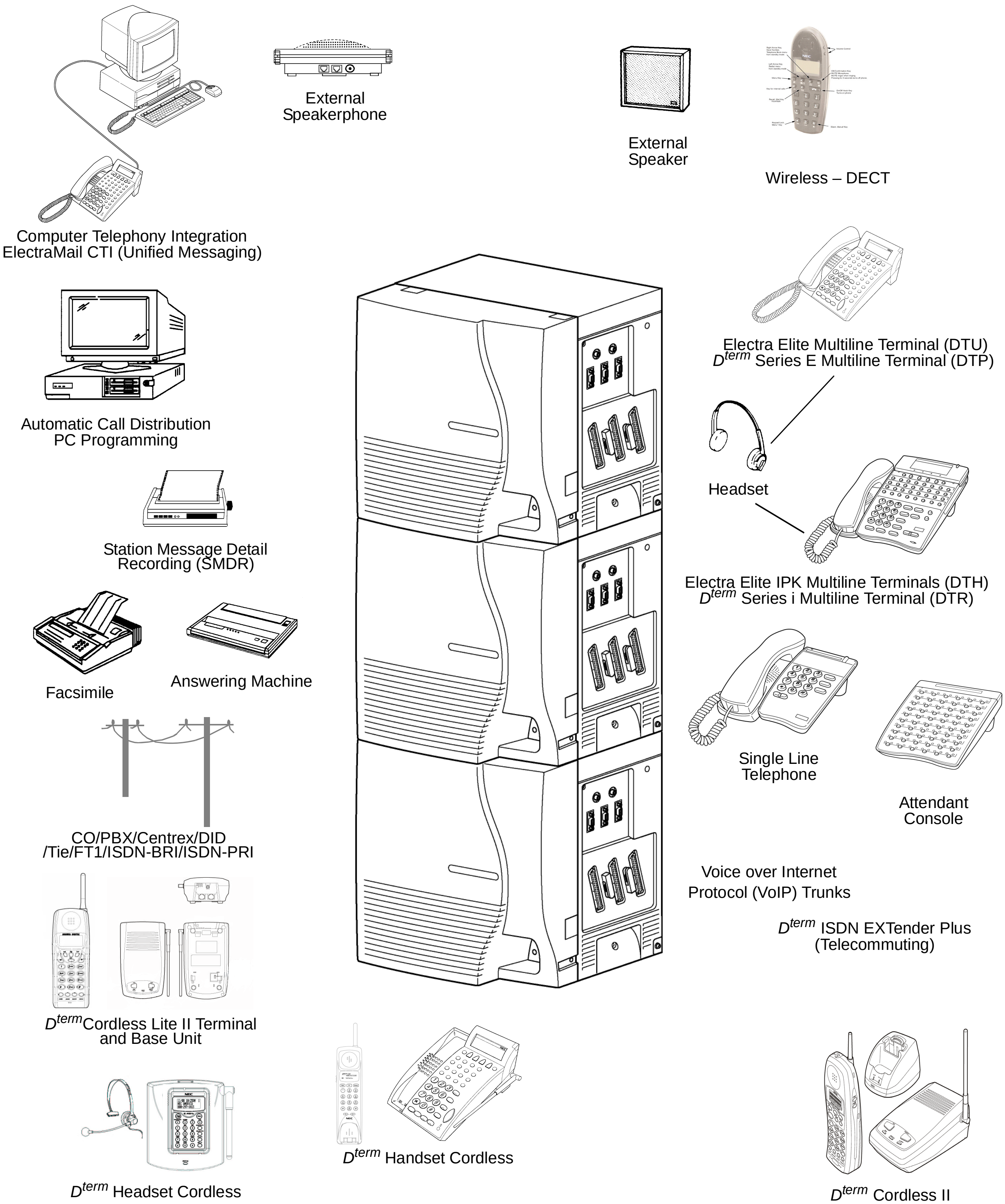


Figure 1-2 System Configuration Example

SECTION 3 SYSTEM DESCRIPTION

The Electra Elite IPK II uses a universal port concept. These ports support telephones, outside lines and other circuits and devices. The assignment of ports is flexible, but the system configuration determines the number of ETUs that can be installed. The maximum number of devices that can be supported by the system are shown in Table 3-1 Maximum System Capacities for Station Interface ETUs on page 3-1, Table 3-2 Maximum System Capacities for Trunk Interface ETUs on page 3-8, Table 3-3 Maximum System Capacities for Application Interface ETUs on page 3-12. The universal port technique provides flexibility for meeting various customer requirements by allowing a wide range of configurations.

Design Technologies

- ❑ Non-blocking time division switching for Multiline Terminals
- ❑ Stored program control
- ❑ Distributed processing based on the use of microprocessors

Design Goals

- ❑ Modular Growth
- ❑ Universal Slots
- ❑ Variety of Terminals
- ❑ Ease of Operation
- ❑ Networking Ability
- ❑ Computer Telephony Integration
- ❑ IP Converged Technology

The Electra Elite IPK II system is a 32-bit microprocessor based, stored program controlled, digital communication system using Pulse Code Modulation (PCM).

The system has central equipment cabinets and telephones located throughout the installation site. The central equipment cabinets contain the Key Service Unit (KSU). A maximum of three Electra Elite IPK II KSUs can be installed to accommodate customer requirements.

The KSU is built for modular growth. The Electra Elite IPK II KSUs are stacked vertically for quick interconnection. Printed circuit boards, called Electronic Telephone Units (ETUs), provide common control and interface to equipment that is external to the KSU.

Interface ETUs are installed in the KSU to support the various telephones, outside lines, and other devices or features. The same ETUs are used for both the basic and expansion port packages.

The universal slot design minimizes the hardware required for a system and provides greater flexibility in the number and type of devices that can be installed. Refer to Figure 1-3 ETU Slot Design.

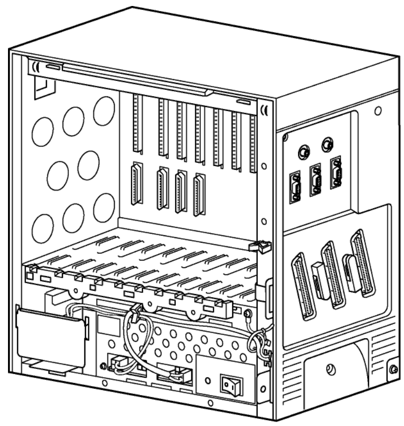


Figure 1-3 ETU Slot Design

The Electra Elite IPK II KSU contains an application (AP) slot (not used), a CPU/EXP ETU slot, and eight interface slots. A power supply and battery backup area complete the KSU.

The Electra Elite IPK II system allows connection of a variety of telephones. The different needs of the customer may require various types of telephones.

The Elite IPK (DTH), *D*^{term} Series i (DTR), Electra Elite (DTU), and *D*^{term} Series E (DTP) telephones are briefly described below.

DTH/ITH/ITR Multiline Terminals (Elite IPK)

- 4-Line *D*^{term} IP terminal with display, ITR-4D-3 TEL
- 8-line terminal without display, DTH-8-1 TEL
- 8-line terminal with display, DTH-8D-1 TEL or ITH/ITR-8D-2/3 TEL
- 16-line terminal without display, DTH-16-1 TEL
- 16-line terminal with display, DTH-16D-1 TEL or ITH/ITR-16D-2/3 TEL
- 16-line terminal with display and 16 programmable One-Touch keys, DTH-32D-1 TEL
- 16-line terminal with display, DTH-16LD-1 TEL. This telephone is equipped with two additional LCDs. These can be programmed to identify the line key designations.

DTR Multiline Terminals (*D*^{term} Series i)

- 2-line terminal without display, DTR-2DT-1 TEL
- 4-line display, DTR-4D-1 TEL
- 8-line terminal without display, DTR-8-1 TEL
- 8-line terminal with display, DTR-8D-1 TEL
- 16-line terminal without display, DTR-16-1 TEL
- 16-line terminal with display, DTR-16D-1 TEL
- 32-line terminal with display, DTR-32D-1 TEL
- Attendant Console, DCR-60-1 console

Comparison of DTH/ITH/ITR/DTR and DTP/DTU Terminals

- DTR-1-1 Single Line Telephone has 6-level receive volume control, 4-level ring volume control, and 3-tone ring pitch. DTP-1-1 has only 3-level receive volume control, and 2-level ring volume control.
- DTR-1HM-1 Single Line Hotel-Motel Telephone has 6-level receive volume control, 4-level ring volume control, 3-tone ring pitch, and monitor dialing. DTP-1HM-1 has only 3-level receive volume control, and 2-level ring volume control.
- DTR-2DT-1 has one more function key (nine) than DTP-2DT-1 and an outgoing only Analog SLT port.
- ITH-4D-3 TEL is an IP terminal with four programmable line keys and four softkeys.

- ❑ DTH-8-1/DTR-8-1, DTH-8D-1, ITH/ITR-8D-2/3, DTR-8D-1 [with three additional fixed keys for message (MSG) microphone (MIC), and directory (DIR)], DTP-8-1, DTU-8-1, DTP-8D-1, and DTU-8D-2 have the same line capacity.
- ❑ DTH-16D-1, ITH/ITR-16D-2/3, DTR-16D-1, DTP-16D-1, DTU-16D-2 [with three additional fixed keys for message (MSG) microphone (MIC), and directory (DIR)], and DTP-16HC-1 have the same line capacity.
- ❑ DTH-32D-1/DTR-32D-1 [with three additional fixed keys for message (MSG) microphone (MIC), and directory (DIR)], DTP-32D-1, and DTU-32D-2 have the same line capacity.
- ❑ DTP terminal supports handset cordless, DTP-16HC-1 model.
- ❑ DTH/ITH/ITR/DTR terminals are compatible with AD(A)-R, AP(A)-R, AP(R)-R, or CT(A)-R Unit adapters. DTP/DTU terminals except for DTP-2DT-1, DTR-4D-1, and DTP-16HC-1, or cordless terminals are compatible with ADA-U, APA-U, APR-U, CTA-U, CTU(S)-U, or HFU-U.

DTP or DTU Terminals

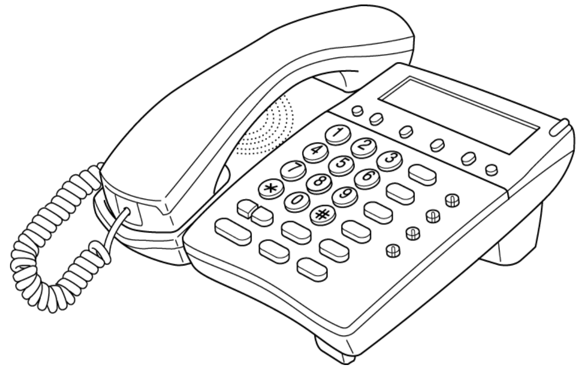
- ❑ 2-line Multiline Terminal without display, DTP-2DT-1 TEL
- ❑ 8-line Multiline Terminal without display, DTP-8-1 or DTU-8-1 TEL
- ❑ 8-line Multiline Terminal with display, DTP-8D-1 or DTU-8D-2 TEL
- ❑ 16-line Handset Cordless Terminal, DTP-16HC-1 TEL
- ❑ 16-line Multiline Terminal without display, DTP-16-1 or DTU-16-1TEL
- ❑ 16-line Multiline Terminal with display, DTP-16D-1 or DTU-16D-2 TEL
- ❑ 32-line Multiline Terminal without display, DTP-32-1 or DTU-32-1 TEL
- ❑ 32-line Multiline Terminal with display, DTP-32D-1 or DTU-32D-2 TEL

Comparison of DTP and DTU Terminals

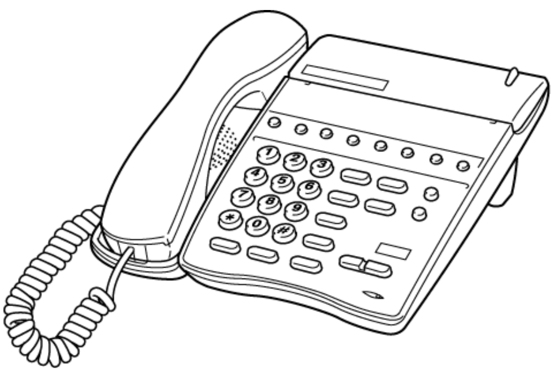
- ❑ DTP terminals are feature comparable to DTU and are supported by the Electra Elite and NEAX family of products.
- ❑ DTU Multiline terminals are supported by the Electra Elite family of products.

Refer to Figure 1-4 Elite IPK (DTH/ITH/ITR) / Dterm Series i (DTR) Multiline Terminals, Figure 1-5 Electra Elite Multiline Terminals, Figure 1-6 DTR Single Line Telephones, Figure 1-7 Attendant Consoles, Figure 1-8 D^{term} Cordless II Terminal, Figure 1-9 D^{term} Cordless Lite II Terminal, Figure 1-10 D^{term} Cordless Lite II Base Unit, Figure 1-11 D^{term} Handset Cordless Terminal, Figure 1-12 D^{term} Headset Cordless, and Figure 1-13 Wireless – DECT.

DTR-2DT-1 TEL
2-Line Non-display

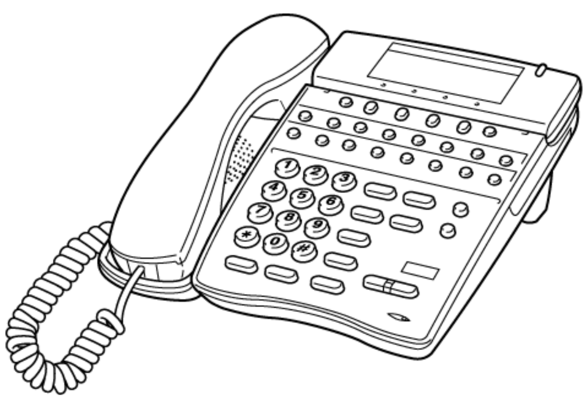


DTR-4D-1 TEL
4-Line Display

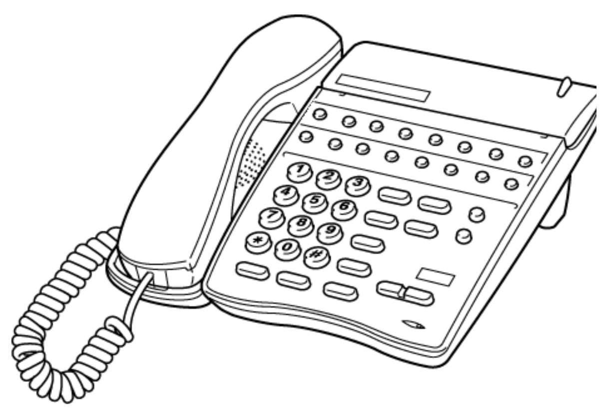


DTH-8-1 TEL/DTR-8-1 TEL
8-Line Non-display

DTH-8D-1/ITH-8D-2/3/DTR-8D-1 TEL
8-Line Display



DTH-16D-1/ITH-16D-2/3/ DTR-16D-1 TEL
16-Line Display

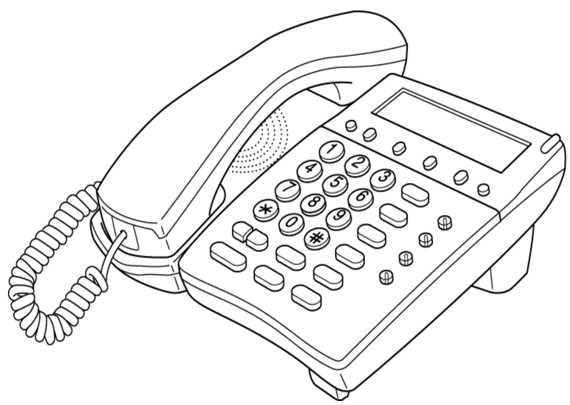


DTH-16-1 TEL/DTR-16-1 TEL
16-Line Non-display

Figure 1-4 Elite IPK (DTH/ITH/ITR) / *D*[term] Series i (DTR) Multiline Terminals

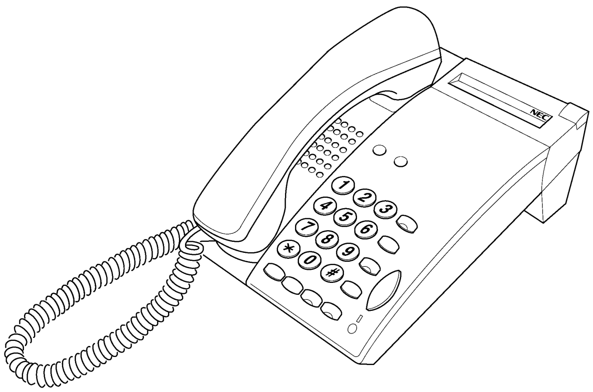
DTH-16LD-1 TEL / DTR-16LD-1 TEL
3 Displays

DTH-32D-1 TEL / DTR-32D-1 TEL
16-Line Display with 16 Programmable One-Touch Keys

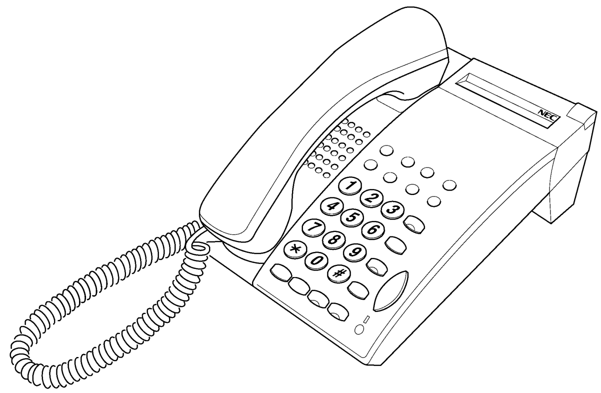


ITR-4D-3 TEL
IP terminal with 4 Programmable Line keys

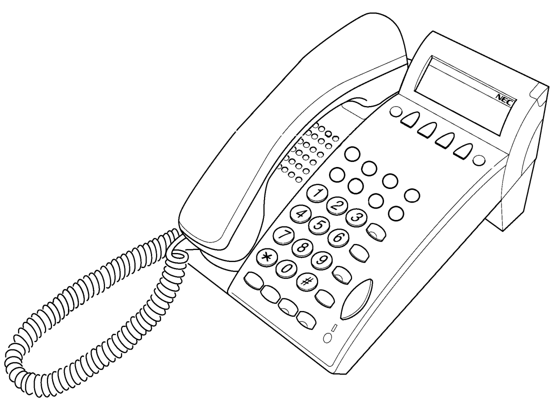
Figure 1-4 Elite IPK (DTH/ITH/ITR) / *D*^{term} Series i (DTR) Multiline Terminals (continued)



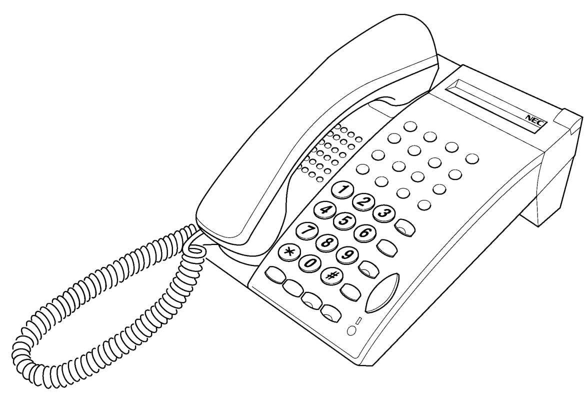
DTP-2DT-1 TEL
2-Line Non-display



DTU-8-1 TEL
8-Line Non-display

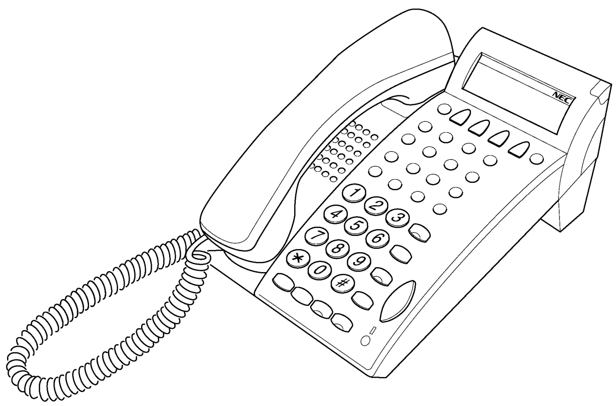


DTU-8D-2 TEL
8-Line Display



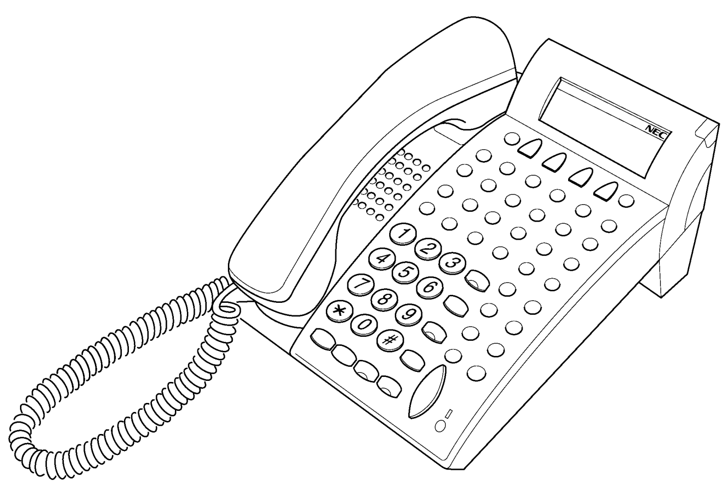
DTU-16-1 TEL
16-Line Display

Figure 1-5 Electra Elite Multiline Terminals



DTU-16D-2 TEL
16-Line Display

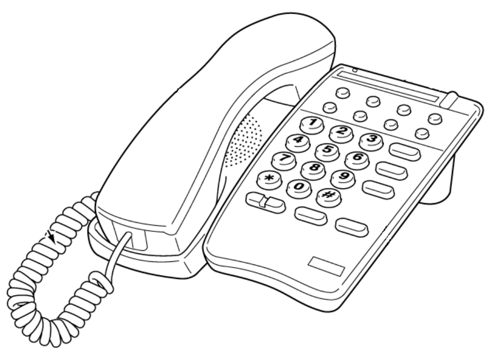
DTU-32-1 TEL
16-Line Non-display with
16 Programmable One-Touch Keys



DTU-32D-2 TEL
16-Line Display with 16 Programmable
One-Touch Keys

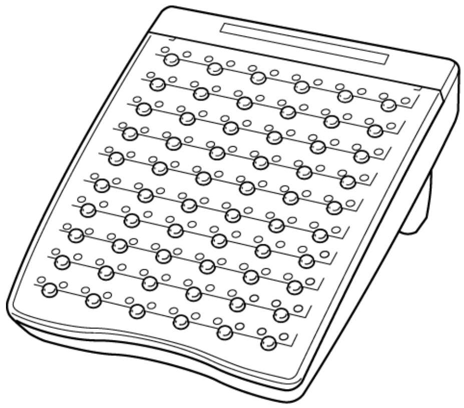
Figure 1-5 Electra Elite Multiline Terminals (continued)

DTR-1-1 TEL

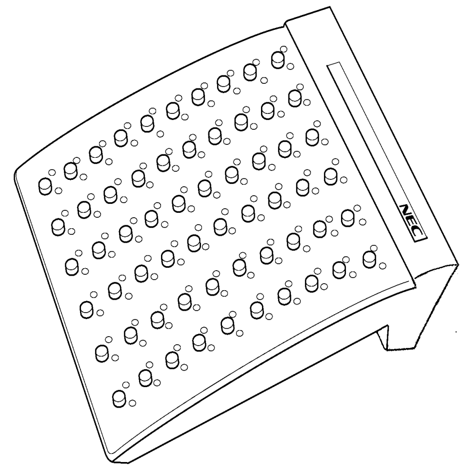


DTR-1HM-1 TEL

Figure 1-6 DTR Single Line Telephones

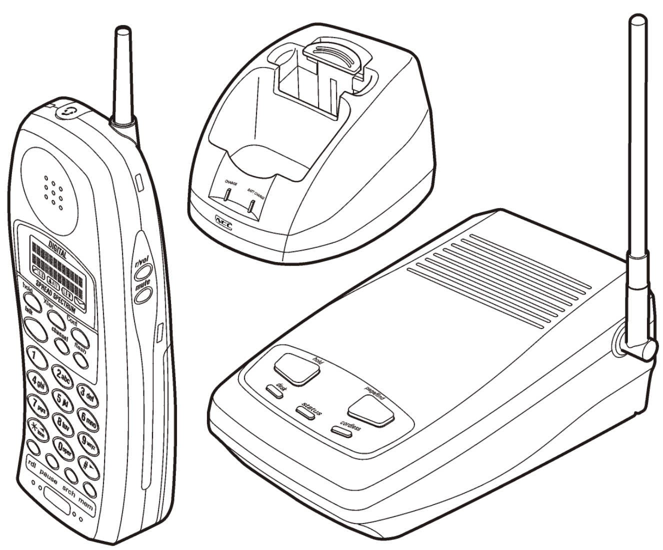


DCR-60-1 Console
(D^{term} Series i / Electra Elite IPK)



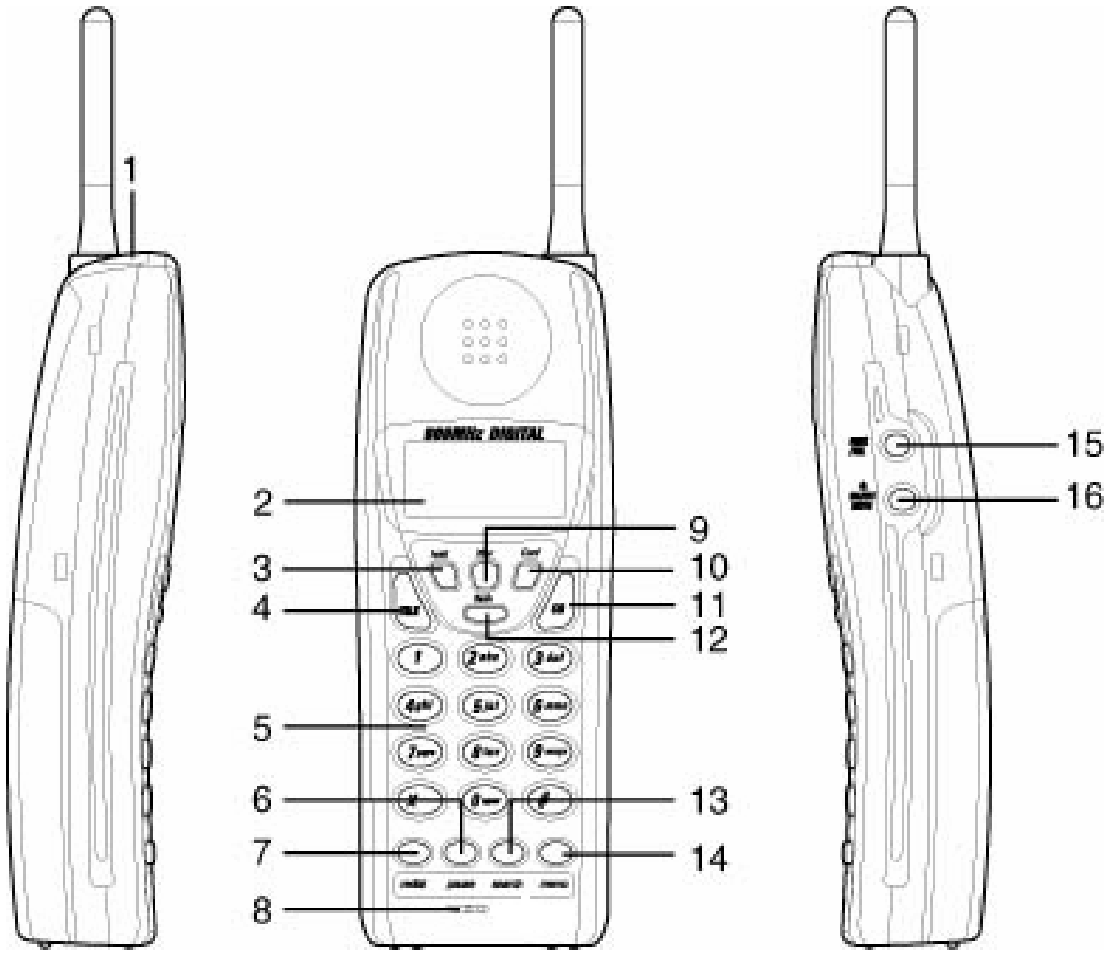
DCU-60-1 Console
(Electra Elite IPK)

Figure 1-7 Attendant Consoles



DTR-4R-1 TEL

Figure 1-8 ***D***^{term} **Cordless II Terminal**

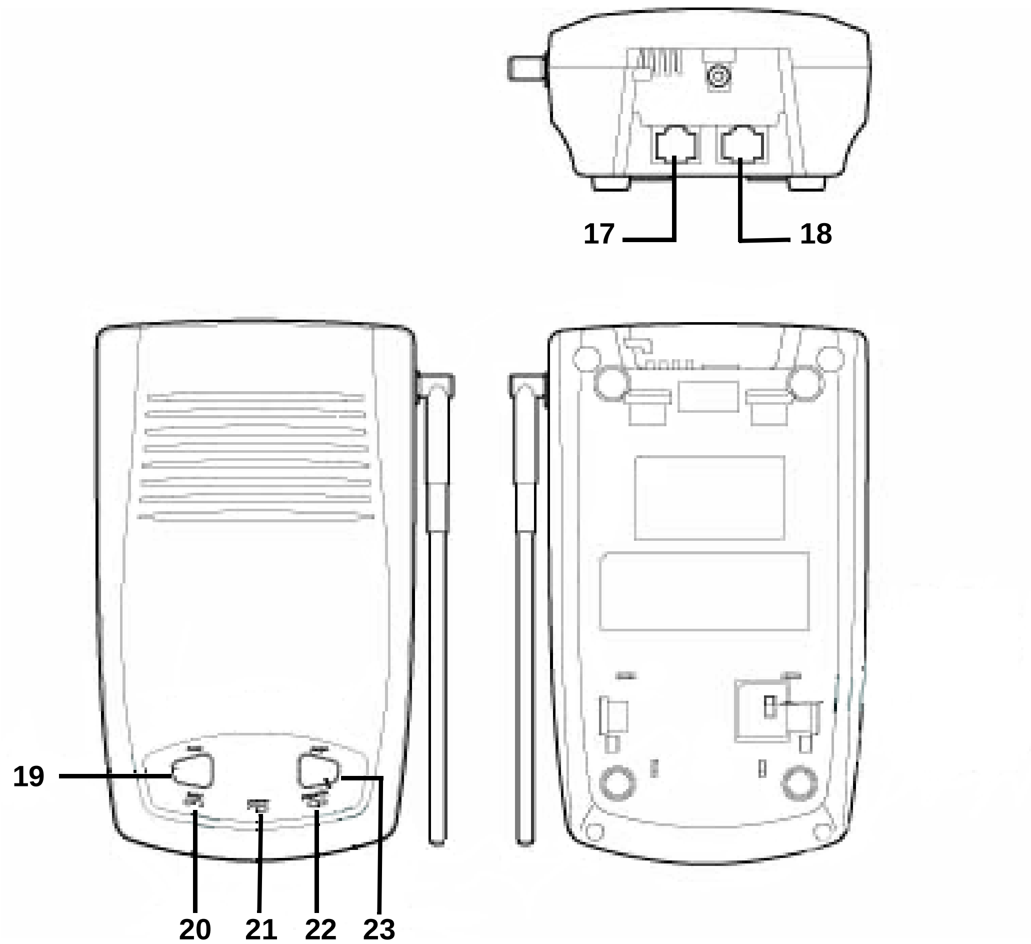


DTH-4R-1 Handset

Figure 1-9 *D*^{term} Cordless Lite II Terminal

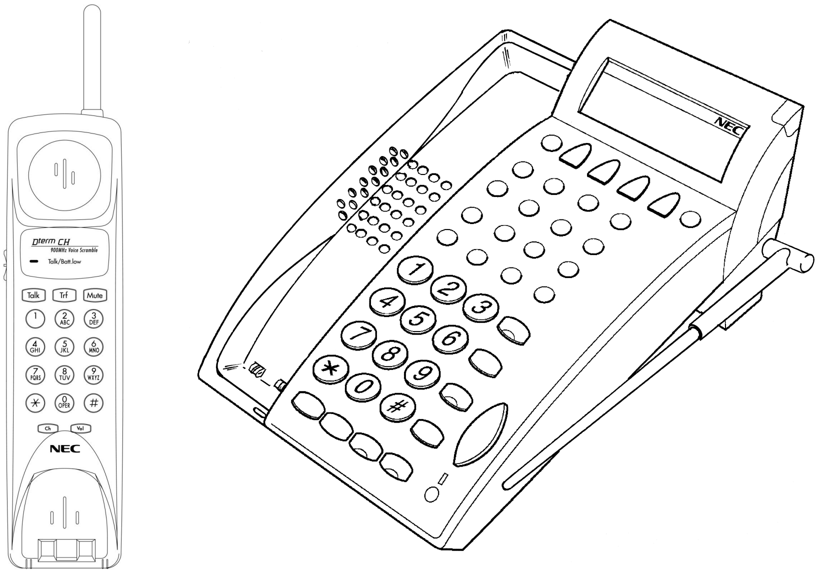
***D*^{term} Cordless Lite II Terminal Controls**

1 Headset Jack	5 Numeric Key Pad	9 CONF (Conference) Key	13 F3
2 LCD Message Display	6 F2	10 TRANSFER Key	14 F4
3 HOLD Key	7 F1	11 CH (Channel) Key	15 Ring/Vol Key
4 TALK Key	8 Microphone	12 REDIAL or Desk/Cordless Softkey Switch Key	16 MUTE Key



17. Line Out	21. Power
18. Line In	22. Desk LED
19. Cordless	23. Desk
20. Cordless LED	

Figure 1-10 *D*^{term} Cordless Lite II Base Unit



DTP-16HC-1 TEL

Figure 1-11 *D*^{term} Handset Cordless Terminal

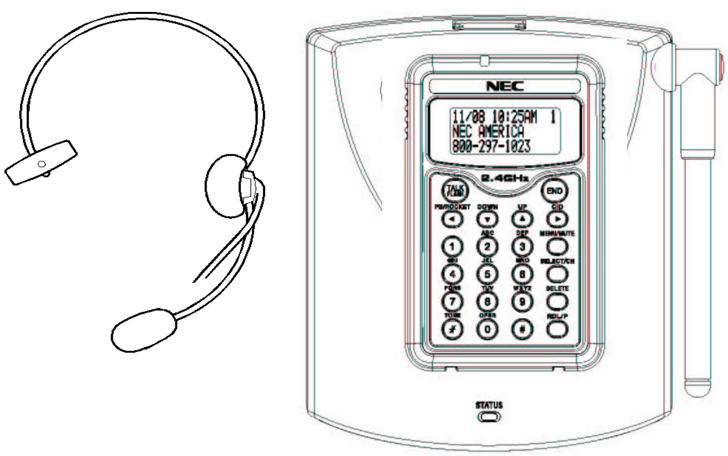


Figure 1-12 *D*^{term} Headset Cordless

Figure 1-13 Wireless – DECT

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Features

SECTION 1 OVERVIEW

This chapter provides a feature comparison list, which compares the IPK feature names to the IPK II feature names.

The remainder of the chapter provide a list of all of the IPK II features and a brief description. For a more detailed description of the feature, refer to the Electra Elite IPK II Features and Specifications Manual.

SECTION 2 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 – Forced/Verified/Unverified	Account Code – Forced/Verified/Unverified
Account Code Entry	Account Code Entry
Add-On Conference	Conference
All Call Page	Paging, Internal
Alphanumeric Display	Alphanumeric Display
Analog Line Extender (D^{term} Analog EXT)	*Not Supported*
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

IPK Feature Name	IPK II Feature Name
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
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
Call Forward – Off-Premise	Call Forwarding, Off-Premise

IPK Feature Name	IPK II Feature Name
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
Callback Request	Message Waiting
Caller ID Call Return	Caller ID Call Return
Caller ID Indication (Analog Trunks)	Caller ID
Cascade CPU	*Not Supported*
Centralized Voice Mail (with E&M Tie Lines)	*Not Supported*
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
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)
Digit Insertion	Automatic Route Selection
Digital Line Extender (*D*^{term} ISDN EXTender Plus)	*Not Supported*

IPK Feature Name	IPK II Feature Name
Not Supported	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
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
D^{term} Analog Cordless Terminal	*Not Supported*
D^{term} Cordless II Terminal	*D*^{term} Cordless II Terminal
D^{term} Cordless Lite II Terminal	*D*^{term} Cordless Lite II Terminal
D^{term} Handset Cordless	*D*^{term} Handset Cordless
D^{term} IP Gateway System	*D*^{term} 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	*Not Supported*
Electronic Volume Control	Volume Controls
Elite ACD Plus	*Not Supported*
Elite CallAnalyst	Elite CallAnalyst
Elite Q-Master	*Not Supported*

IPK Feature Name	IPK II Feature Name
EliteApps – Interactive Voice Response	EliteApps – Interactive Voice Response
EliteApps – PC Attendant	IPK II – PC Attendant
Emergency 911 – Cut Through	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
Flexible Numbering Plan	Flexible System Numbering
Flexible Ringing Assignment	Ring Groups
Flexible Timeouts	Flexible Timeouts
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
Hot Key Pad	Hot Key-Pad
Hot Line	Hotline
Howler Tone Service	Howler Tone Service
I-Hold Indication	Distinctive Ringing, Tones and Flash Patterns

IPK Feature Name	IPK II Feature Name
Incoming Call Identification	Caller ID
Incoming Trunk Name or Number Display	Name Storing
Internal Hub	Internal Hub
Internal Voice/Tone Signaling	Handsfree Answerback/Forced Intercom Ringing
Internal Zone Paging (Meet-Me)	Paging, Internal
IP CPU & Media Gateway	*Not Supported*
IP Station (MEGACO)	IP Station (MEGACO) – IAD Integrated Access Device
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
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
Not Supported	Licensing
Live Monitoring	Digital Voice Mail
Loop Start Trunks	Multiple Trunk Types
Message Display Board	*Not Supported*
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	*Not Supported*
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 (Actual name)
Not Supported	One-Digit Dial Option
One-Touch Feature Access	Programmable Function Keys
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 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
Resident System Program	Resident System Program
Restriction (Outgoing)	Code Restriction
Ring Tone Variation	Distinctive Ringing, Tones and Flash Patterns

IPK Feature Name	IPK II Feature Name
Ringing Line Preference	Line Preference
Route Advance Block	Trunk Group Routing
Save and Repeat	Save Number Dialed
Scrolling Directories	Directory Dialing
Secondary Incoming Extension	Secondary Incoming Extension
Seized Trunk Name/Number Display	Name Storing
Simplified Call Distribution	Department Calling
Single Line Telephone Access	Single Line Telephones, Analog 500/2500 Sets
SLT Adapter	SLT Adapter
SLT Timed Alarm	Alarm
Not Supported	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 Programmable	Station Name Assignment User Programmable
Station Outgoing Lockout	Code Restriction, Dial Block
Station Relocation	Station Relocation
Station Transfer	Transfer
Step Call	Department Step Calling
Store and Repeat	Memo Dial
Stored Hookflash	Speed Dial – System/Group/Station
Synchronous Ringing	Synchronous Ringing

IPK Feature Name	IPK II Feature Name
System Data Up/Down Load	PC Programming
T1 Connection	T1 Trunking (with ANI/DNIS Compatibility)
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
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
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	*Not Supported*
Wireless – DEC	Wireless – DECT

SECTION 3 FEATURES DESCRIPTIONS

Account Code Entry

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 Code – Forced Verified/Unverified

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

❑ Verified Account Codes

Alarm

Alarm lets any station extension work like an Alarm clock. An extension user can have an 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)

Alarm Reports

Enhancements

Version 1500 or higher is required to support this feature.

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.

Alphanumeric Display

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.

Analog Communications Interface (ACI)

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(8)-U20.

Ancillary Device Connection

Enhancements

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 on ports 193~256.

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.

Answer Hold

Answer Hold allows a multiline terminal user to press the flashing Answer key 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 setting in Program 15-02-06.

Answer Key

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, the Answer key is used to pick up calls. The Answer key continues flashing until the last unanswered call is answered. Press the Answer key during a call to hold the current call and allow the next call to be answered.

Attendant Call Queuing

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 would 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.

Automatic Call Distribution (ACD)

Enhancements

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

Version 2000 or higher:

- ❑ Delay Announcements using In-Mail are supported and 1-Key dial out options are supported when using VRS Delay Announcements.
- ❑ ACD calls which overflow and are answered outside the original group are counted as an overflowed call in the MIS software. Prior to this release, an overflowed call was counted as an Abandoned call.
- ❑ A caller in ACD queue is able to 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.

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 representative 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 Release

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.

Automatic Route Selection

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 today’s telecommunications world 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 extension number, just an extension number, or by using a trunk access code.

Background Music

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

Barge-In

Barge-In permits an extension user to break into another extension user 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 conversation but cannot participate. With Speech Mode, the caller Barging In can listen and join another user 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.

Battery Backup – System Memory

The CPUII( )-U( ) ETU battery retains the Clock/Calender 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 about three years.

The System Programmed memory (Customer Database) is stored in non-volatile Memory and can be erased only by a First Initialization. After power is restored, the system returns to normal operation.

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

Battery Backup – System Power

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 capacity of the batteries.

Call Appearance (CAP) Keys

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.

Once 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.

Call Arrival (CAR) Keys

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 when in the ringing state and are not used during the talking state.

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

Call Duration Timer

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.

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/Now Answer, and Immediate) will light solid instead of flashing when the key's forward type is set.
- ❑ DSS/BLF displays when a telephone is off-hook (Busy) when Call Forward Immediate is set.

Call Forwarding permits an extension user to redirect their calls to another extension or an off-premise number. Call Forwarding ensures that the user calls are covered when a user is 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 2-17 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.

Call Forwarding – Centrex

Enhancements

This feature added with **Version 1500**.

The Call Forwarding – Centrex feature allows a station user 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 Forwarding – Immediate

Call Forwarding – Busy

Call Forwarding – No Answer

Call Forwarding – Busy/No Answer

Call Forwarding/Do Not Disturb Override

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.

Call Forwarding, Off-Premise

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.

Call Forwarding – Park and Page

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 prerecorded Paging announcement. When the extension user hears the Page, they can go to any telephone and use Directed Call Pickup to intercept the call.

Call Forwarding with Follow Me

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

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

Call Monitoring

Enhancements

Version 1500 or higher supports Call Monitoring with coaching.

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 Redirect

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. By pressing the Call Redirect key, the call is transferred, allowing you to continue with your current call.

This feature works with the following types of calls:

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

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

- ❏ ACD
- ❏ Transferred
- ❏ Department Group (all ring mode)

- ❑ Door Box
- ❑ Virtual Extension

Call Waiting/Camp-On

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.

Callback

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.

Caller ID

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 is available as a pre-answer display so the user can preview the telephone number before answering the call.

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

Caller ID Call Return

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.

Central Office Calls, Answering

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.

Central Office Calls, Placing

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 individual needs. A user can place a call by:

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

There are 200 available trunks.

Class of Service

Enhancements

Using Version 2000 or higher, when a Ring Group call rings a station, a BLF Indication for this station shows idle. Prior to this release, the BLF indicator showed busy.

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.

Clock/Calendar Display

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 on a nightly basis. This feature allows you to make adjustments should the system cabinet regularly lose or gain time.

CO Message Waiting Indication

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 shows a user that a message is present in their voice mail box. When VMWI is provided, the Electra Elite IPK II flashes the 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

Code Restriction

Code Restriction limits the numbers an extension user may dial. By allowing users 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 of an extension. The system allows for up to 15 Code Restriction Classes and 416 extensions.

Code Restriction-Dial Block

Code Restriction-Dial Block lets a user temporarily block an extension Code Restriction. This lets a user block their telephone from being used by another person while they are away from their desk. A user must enter a 4-digit personal code to enable/disable this feature.

Dial Block can also be set by using the supervisor access code. If Dial Block has already been set by an extension user, the supervisor cannot release it. Additionally, if Dial Block has been set using the supervisor 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.

Code Restriction Override

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 the **Speaker** key, dial a service code and enter their override code.
- ❑ Press the **Speaker** key 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 21-06-xx.

Computer Telephony Integration (CTI) Applications

Computer Telephony Integration (CTI) applications automate your office with Telephony Applications Programming Interface (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 will allow your PC to operate your telephone. The TAPI software driver provides all TAPI Basic Services and a host of TAPI Supplemental Services. See TAPI Compatibility on page 2-88 for more information.

Conference

Enhancements

Conference Call functionality is available for Virtual Extensions with **Version 2000 or higher**.

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 circuits, 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 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 circuits are all busy.

Conference, Voice Call/Privacy Release

Voice Call Conference lets an extension user in the same work area join in a trunk Conference. To initiate a Voice Call Conference, an extension user 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 to indicate 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.

Privacy Mode Toggle Option

The Privacy Mode Toggle option allows an extension user to quickly change an outside call from the non-private mode to the private mode. If the outside call is on a line key, the user presses 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 presses 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.

Continued Dialing

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 the Class of Service for an extension, a multiline terminal user may be able to 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.

NOTICE

Continued Dialing may make the system more susceptible to toll fraud.

Cordless Telephone Connection

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 support cordless telephones, but this feature refers to Multiline Terminal cordless connection.

Data Line Security

Data Line Security protects any station port from receiving audible tones (such as Camp-On or Override) and denies a station from barging in while busy to prevent disruption of data transmission when using a modem or facsimile machine.

Delayed Ringing

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.

Department Calling

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 would let a caller dial the Sales department just by knowing the Sales department pilot number. The caller would not have to know any Sales department extension number.

There are two types of routing 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.

Department Step Calling

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.

Dialing Number Preview

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.

Dial Pad Confirmation Tone

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.

Dial Tone Detection

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 through a feature like Last Number Redial), the system can drop the trunk or optionally skip to the next trunk in the group.

Digital Trunk Clocking

Enhancements

This feature was added to the documentation with **Version 2000**. This feature is available with Version **1000 or higher**.

The IPK II CPU is equipped with a built-in clock source for all digital trunks 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 CPU card is the timing source for all digital trunk cards in the system.

The PLO is responsible for synchronizing 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 within the network. The source clock signals are extracted from digital trunk cards and are supplied to the PLO.

Digital Voice Mail

Enhancements

Version 2000 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 also requires Linux 2.0 voice mail software version 2.0.3.x or higher.

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 company receptionists and secretaries for more productive work.

Automated Attendant automatically answers the incoming system calls. After listening to a customized message, an outside caller can dial a system extension or use Voice Mail.

Direct Inward Dialing (DID)

Enhancements

Version 1500 or higher is required for the DID Call by Time Schedule feature.

Direct Inward Dialing (DID) lets outside callers directly dial a system extension. DID saves time for callers who know the extension number they want 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.*

DID Call by Time Schedule allows for 100 pre-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 of the Time Patterns can reference any one of the 2000 different Dial-In Conversion table entries in PRG 22-11-01.

Direct Inward Line (DIL)

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 to the Technical Service department.

Direct Inward System Access (DISA)

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 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.

Direct Station Selection (DSS) Console

Enhancements

In **Version 1500 or higher** DSS/One-Touch keys can be used to one-touch transfer without having to use the transfer key.

Version 2000 or higher:

- ❑ 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.
- ❑ 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 would show busy.

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.

Directed Call Pickup

Directed Call Pickup permits an extension user to intercept a call ringing another extension. This allows a user to conveniently answer a co-worker call 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

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 2000 or higher**, extensions (including Virtual Extensions) can be removed from the Extension Directory list based on a new Class of Service option (20-13-51). Prior to this release, installed extensions and Virtual Extensions with a Name Assigned were always displayed in the Extension Directory list.

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

Distinctive Ringing, Tones and Flash Patterns

Enhancements

Version 2000 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 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 (PGM 15-02-35)
  - ❍ Message Waiting Lamp Cycle for Called Extension (PGM 15-02-36)
  - ❍ Voice Mail Message Wait Lamp Color (PGM 15-02-37)
  - ❍ Voice Mail Message Wait Lamp Cycle (PGM 15-02-38)

  This provides more distinction between incoming calls, Message Waiting (Set/Received) and a VM Message Waiting indication.

Distinctive Ringing, Tones, and Flash Patterns provide extension users with audible and visual call status signals. This lets users tell the type of call 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 calls are 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.

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 DND is set.
- ❑ DSS/BLF displays when a telephone is off-hook (Busy) when DND is set.

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 while 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:*

- ❑ Incoming trunk calls blocked
- ❑ Paging, incoming Intercom, Call Forwards and transferred trunk calls blocked
- ❑ All calls blocked
- ❑ Incoming Call Forwards blocked
- ❑ Do Not Disturb canceled

Door Box

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 connect 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	50mOhms

The system can have up to eight Door Boxes. Six chime tones are available.

Drop Key

The Drop Key abandons a call while retaining the PBX/Centrex line to originate another call. The Drop Key is provided by programming a 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 with an access code.

Dterm Cordless II Terminal

The NEC *Dterm* 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	*Dterm* 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

D^{term} Cordless Lite II Terminal

The NEC *D^{term}* Cordless II Lite Terminal may be used with the Electra Elite IPK II KTS. The DTH-4R-( ) 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	*D^{term}* 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 in the same frequency spectrum.

D^{term} Handset Cordless

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 semiautomatic 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
- ❑ Auto Talk™ 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

D^term^ IP Gateway System

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 10/100 (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 10/100 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

E-911 Compatibility

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 in 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 an asterisk in a PBX access code if the Account Code feature is used.*** With the Account Code feature enabled, if an asterisk 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 system 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
- ❑ Emergency Routing
- ❑ Compatibility with Customer Provided E911 Equipment
- ❑ Calling Party Identification

Electra Elite IPK Terminals

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.

Electra Elite Terminal Migration

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.

EliteApps – Interactive Voice Response

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 (*e.g.*,movie schedules)

The IVR application uses prerecorded 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.

Elite CallAnalyst

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.

Facsimile CO Branch Connection

The Electra Elite IPK II system provided 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 share 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.

Flash

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.

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, and 10,000s group for 7-digit station numbers.

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. In addition, factory technicians can make comprehensive changes to your system number plan. You can have factory technicians:

- Set the number of digits in internal (Intercom) functions. For example, extension numbers can be up to eight digits long.
- 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.

Flexible Timeouts

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.

Forced Trunk Disconnect

Forced Trunk Disconnect allows an extension user to disconnect (release) another extension's 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.

Group Call Pickup

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 others 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 types of calls:

- ❑ A call ringing the user 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.

Group Listen

Group Listen permits a multiline terminal user to talk on the handset and have their voice broadcast over the telephone speaker. This lets the multiline terminal user co-workers listen to the conversation. Group Listen turns off the multiline terminal handsfree microphone so the caller does not pick the co-worker voices during a Group Listen.

Handset Mute

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 the MIC button to mute the transmit speech path. The station user can still hear the outside (or intercom) voice.

Handsfree and Monitor

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.

There are three variations of Handsfree.

- ❑ Handsfree

  The user can place and answer calls by pressing the Speaker key 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 the Speaker key. An extension can have Automatic Handsfree for just 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.

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.

Headset Operation

A multiline terminal user can use a customer-provided headset in place of the handset. Like using Handsfree, using the headset frees up hands for other work. Headset Operation also 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 an option to use the handset, headset or the speakerphone for calls.

Hold

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 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. There are four types of Hold:

- 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 others calls.
- Intercom Hold
  A user can place an Intercom call on Hold. The Intercom call on Hold does not indicate at any other extension.

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 can support 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.

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
- ❑ Single Digit Dialing
- ❑ A Department Calling Group
- ❑ Message Waiting
- ❑ PMS Integration
- ❑ Room to Room Calling Restriction
- ❑ Toll Restriction (When Checked In)
- ❑ Room Status
- ❑ Room Status Printouts
- ❑ DSS Console Monitoring

- Do Not Disturb
- Flexible Numbering Plan

Hot Key-Pad

Enhancements

Version 1500 or higher is required for this feature.

The Hot Key-Pad feature allows the user to place a call without lifting the handset or pressing the Speaker key. When the user dials another extension number on an idle telephone with Hot Key-Pad enabled, the Speaker key lights and the internal call is made. When the user dials the trunk access code from a telephone with Hot Key-Pad enabled the Speaker key lights, a trunk is seized and the outgoing call is made.

Hotline

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.

The Hotline feature has two applications.

- Hotline (Hotline partner)
- Ringdown Extension, Internal/External (Refer to Ringdown Extension, Internal/External on page 2-76.)

In addition, the Hotline key shows the status of the partner 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.

Howler Tone Service

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.

Intercom

Intercom gives extension users access to other extensions. This 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.

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 busy trunk 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 they should use the Barge-In feature for the extension or trunk. This information automatically displays for a multiline terminal when programmed.

Internal Hub

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 referred to as 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
- ❑ VLAN Configuration
- ❑ Quality of Service
- ❑ Port Mirroring

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.

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 10/100 Base-T 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, supports industry standard MEGACO (H.248) communication on the WAN side, and interfaces with the Electra Elite IPK as a regular Electronic Station Interface [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
- ❑ 10/100 Base-T LAN interface
- ❑ Echo Canceller

- ❑ Remote configuration and maintenance
- ❑ NAT Beater - Network Address Translation

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 utilizing the override login function. Users have the flexibility of logging into their IP Station in the office or remotely at home.
- ❑ 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 the use of a file server, thus reducing time and cost required to update each Terminal individually. The Center Download feature, when configured completely, allows the MG16 the ability to communicate with the IP Terminals via DRS listening port 161, gathering the IP Terminal's Firmware information. The MG16 will then send an "Initial Setting Request" to the Elite IPK II CPU. The CPU will compare 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 CPU sends a download instruction.
- ❑ The MG16 web server will have 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).

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, and 10,000s group for 7-digit station numbers.

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 10/100 Base-T 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. The 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. 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 IP 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 [84-05-03] Master/Slave determination. The Media Gateway Controller performs the duties of interpreting 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-System IP Phone (e.g., POT, System Phone), 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.

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 uLaw - Highest Bandwidth
- ❏ G.729 - Mid-Range Bandwidth

IP Station (SIP) – MG16

Enhancements

This feature requires **Version 1500 or higher**.

Session Initiation Protocol (SIP) is used for Voice over Internet Protocol. It is defined by the IETF (Internet Engineering Task Force) RFC3261. Commonly known as an SIP Station, this feature is used for IP Stations 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.

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 MG Application 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 Electra Elite IPK II.

The Electra Elite IPK II CPU 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 utilize 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 Processor, therefore KSU Programming [84-05-03] Master/Slave determination is **NOT** required.

IP Trunk – H.323 Protocol

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 the ability to communicate using standard H.323 (Normal and Fast Start) Protocol and allows connectivity to any H.323 standards compliant voice gateway and gatekeeper. This VoIP Trunk Card also allows Registration and Authentication Server (RAS) support 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 Mu 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 10/100 Base T Ethernet.
- The ETU operating mode can be configured per ETU, but not per port.

IP Trunk – (SIP) Session Initiation Protocol

The Electra Elite IPK II Voice over IP Trunk Card SIP 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.

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 Mu 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 10/100 Base T Ethernet
- ❑ The ETU operating mode can be configured per ETU, but not per port

IP Trunk (SIP) – MG 16

Enhancements

Version 1500 or higher is required to support this feature.

SIP (Session Initiation Protocol) is a protocol used for Voice over IP. It is described in the IETF (Internet Engineering Task Force) RFC3261. Commonly known as 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 a 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 – 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.

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 CPU 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 supports IP signaling for up to 16 (SIP Trunks and/or SIP Stations) and reduces the maximum capacity of IP stations in the system by 16.

A maximum of 16 PVA(X)-U( ) ETUs can be installed supporting the maximum of 200 SIP Trunks.

SSIP 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/Internet, the SIP Trunk can be configured to use any of the following voice compressions:

- ❑ G.711 Mu Law – Highest Bandwidth
- ❑ G.729 (a) – Most often used

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.

The Elite IPK II In-Mail is a low cost voice mail solution that mounts on the CPUII. Its programming is fully integrated with KSU programming. This system offers most voice mail system features customers expect.

Automated Attendant automatically answers the incoming system calls. After listening to a customized message, an outside caller can dial a system extension or use Voice Mail.

Up to four Elite IPKII In-Mail voice mail ports are 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
- ❑ Leaving a Message
- ❑ Transferring to Voice Mail
- ❑ Live Record
- ❑ Personal Answering Machine Emulation
- ❑ Voice Mail Overflow
- ❑ Message Center Mailbox
- ❑ Voice Mail Caller ID

IPK II – PC Assistant

Enhancements

Version 2000 supports:

- ❑ An updated 3rd party TAPI drivers to provide screen pops for incoming calls to Virtual extensions. Version 2000 system software and version 2.0 TAPI drivers are required.
- ❑ A Microsoft Outlook Add-in to provide screen pops and dial out capabilities without full PC Assistant installation.

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 only available 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 is a application which 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. The use of 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 that is 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.

IPK II – PC Attendant

Enhancements

Version 1500 supports updated 3rd party TAPI drivers to provide screen pops for incoming calls to Virtual Extensions. Version 2000 system software and version 2.0 TAPI drivers are required.

Version 2000:

- ❑ Supports phone Display Messages (from PC Attendant to phone). Version 2000 system software and version 2.0 TAPI drivers are required.
- ❑ Supports Speed Dial buttons (Programmable Speed Dial buttons on PC Attendant).

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 is able to 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 also be installed on multiple PCs for installations that need to support more than one attendant position (each attendant PC would require 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, called Quick Message. By installing the Quick Message client on individual PCs, the attendant is able to quickly send short messages to other employees, who can respond with a single keystroke.

The host PC to be 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.

IPK II VoIP Management System

Enhancements

Version 1500 or higher is required to support this feature.

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) with CCIS application can be added to the database, but the MoS scores are inaccurate. The PVA(X) with MG16 and associated terminals cannot be added to the database.*

ISDN Compatibility

Enhancements

The T1/PRI Interface ETU can be programmed as a 4/8/12/16/20/24-port Fractional T1/PRI. **V1500 or higher** (when 10-39-01 is set to Enabled) is required. **Version 1500 or higher** supports Called Party Number Enhancement so the IPK II can edit the Called Party Number Parameter using the Network Specified Parameter Table settings (26-12-01 and 26-12-02).

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. The Electra Elite IPK II supports it. 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
- ❑ Calling Line Identification Presentation (CLIP)
- ❑ Calling Party Number (CPN) Presentation from Station
- ❑ Enhanced 911 Service with ISDN - PRI
- ❑ Two B-Channel Transfers (ISDN - PRI only)
- ❑ SMDR Includes Dialed Number
- ❑ Display Shows Why Caller ID is Not Available

K-CCIS - IP with IAD

Enhancements

This feature was added with **Version 1100**.

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.
- ❑ 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.

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.

K-CCIS – IP with PVA

Enhancements

This feature was added with **Version 1100**.

Version 1500 or higher:

- ❑ A 4-port license registration provides users with the flexibility of adding CCISoIP Ports in increments of four (4, 8, 12, 16, 20 or 24).
- ❑ The CCISoIP web server will have a new HTTP page to display the network statistics based on values entered in PRG 84-06.

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.
- ❑ 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.

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.

K-CCIS - T1

Enhancements

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.
- ❑ 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.

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), and a Common Channel Handler (CCH) to receive or transmit common signaling data from/to a distant office. The system can provide a variety of interoffice service features 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.

Last Number Redial

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 a System Speed Dialing bin number to be dialed. Pressing the Redial key repeatedly scrolls through the last 10 numbers dialed.

Licensing

Enhancements

Version 1500 or higher license allows for the system to have SIP Trunks/Stations.
Version 2000 or higher supports the 30-day promo license.

Licenses are used to activate certain features and applications for the Electra Elite IPK II. The following are the licenses that the IPK II provides:

- ❑ CPU Licenses
  - ❍ InACD - This license allows the system to run the InACD feature.
  - ❍ SMDR - This license allows the system to print out SMDR reports.
  - ❍ Remote Software Upgrade - This license allows the system to be upgraded remotely.
  - ❍ Megaco Stations - The IPK II has 16 Megaco Station licenses when purchased, but more licenses can be purchased in increments of one, four, or eight with the maximum being 256.
  - ❍ SIP - This license allows for the system to have SIP Trunks/Stations. The licenses can be purchased in increments of one, four, or eight with the maximum being 256. (**V1500 or higher software is required**.)
  - ❍ Hotel/Motel - This license allows the system to run the Hotel/Motel feature.
  - ❍ CTI - This license allows the system to have a 3rd 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 stations, 256 Megaco stations, and SMDR license for up to 10 days.
- ❑ 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. (**V2000 or higher is required**.)

- ❑ Application License

  The IPK II system has extra applications that can be purchased. These extra applications are:

  - ❍ Elite Call Analyst
  - ❍ PC Assistant
  - ❍ PC Attendant
  - ❍ PC Programming (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 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.

Line Preference

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 lifting the handset. If a multiline terminal has Outgoing Trunk Line Preference, the user hears trunk dial tone when lifting 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 ringing calls, however, always have priority over calls ringing co-worker extensions.

Long Conversation Cutoff

For incoming and outgoing central office calls, each trunk can be programmed to disconnect after a defined time. The timer begins when the trunk is seized and disconnects the call 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.

Meet Me Conference

With Meet Me Conference, an extension user can set up a Conference with their current call and up to 32 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 CPU 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.

Meet Me Paging

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.

Meet Me Paging Transfer

If a user wants to Transfer a call to a co-worker but does 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.

Memo Dial

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 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.

Message Waiting

Enhancements

Version 2000 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 (PGM 15-02-35)
  - ❍ Message Waiting Lamp Cycle for Called Extension (PGM 15-02-36)
  - ❍ Voice Mail Message Wait Lamp Color (PGM 15-02-37)
  - ❍ Voice Mail Message Wait Lamp Cycle (PGM 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.

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 steadily lit MW lamp on the calling extension. Answering the Message Waiting automatically calls the extension that 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.

Microphone Cutoff

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 use work area. The user may turn off the microphone while their telephone is idle, busy, or ringing. The microphone stays off until the user turns it back on.

Multiline Conference Bridge

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.

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 toggle mute (on/off) by pressing 0 on the telephone dial pad.

❑ 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)-U10 ETU.

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.

Multiple Trunk Types

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.

Music on Hold

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 (e.g., 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 has been 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.*

Name Storing

Enhancements

With **Version 1500 or higher,** telephones that are not connected or uninstalled, do not show up in the Extension Directory.

Version 2000 or higher:

- ❑ Station Number and Name can be removed from the display when the phone is idle based on new Class of Service options (20-13-47 and 20-13-48). Prior to this release, the station Number and Name was 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 (20-13-51). Prior to this release, installed extensions and Virtual Extensions with a Name Assigned would always be 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.

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 that allows Name Storing – Speed Dial – System/Group/Station, One-Touch Keys, Extension Name, Trunk Naming.

Night Service

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.

Off-Hook Signaling

The signal is an off-hook ringing over the idle (second) line appearance. 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
- ❑ Automatic Signaling
- ❑ Manual Signaling
- ❑ Selectable Off-Hook Signaling Mode
- ❑ Off-Hook Ringing
- ❑ DID Call Waiting
- ❑ Block Manual Off-Hook Signals
- ❑ Block Camp On

One-Digit Dial Option

Enhancements

Version 2000 or higher is required.

When a caller to an ACD Group is being played a 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. (**V2000 or higher is required**.)

There is one one-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, and another ACD Group.

One-Touch Calling

One-Touch Calling gives a multiline terminal user one-button access to extensions, trunks, speed dial bins and selected system features. This saves 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 presses key 1 to call the company, then one of keys 2~5 to ring the employee to which they want to speak.

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.

Operator

Enhancements

Version 1500 or higher allows multiple Operator Groups.

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.

OPX (Off-Premise Extension)

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.

Paging, External

With External Paging, a user can broadcast announcements over paging equipment connected to external Paging zones. When a user pages one 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 (Zone #9). All other zones (#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. 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.

Paging, Internal

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.

Park

Enhancements

Version 2000 or higher allows the ability to perform Call Park - Step Call by pressing one key to find and park a call into an idle park location. The park location is displayed.

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 will 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 times 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 with the ability to use 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.

PBX Compatibility

You can connect your telephone system trunks to Centrex/PBX lines, rather than to Telco trunk circuits. This makes the trunk inputs into 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 for up to four groups of trunk access codes. The codes can have one or two digits, consisting of the digits 0~9, # and ✱. (You can 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 facilities 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, Last Number Redial, Repeat Redial and Save Number Dialed. This gives the connected PBX time to set up its trunk circuits.

PC Programming

Enhancements

Version 1100 or higher provides updated PCPro software that includes a database conversion tool and supports DESI. **Version 2000** PC Programming software allows the Terminal Type to be assigned and saved for DESI export.

IPK II introduces three different methods for programming. The first is via the handset, the second is by PCPro (SAT), and the 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 CPUII card of the KTS. No special installation program is required. A user programs the KTS using their standard web browser.

Power Failure Transfer

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 pre-selected single line telephone. The single line telephone can function in the system during normal operation or be used during a power failure.

Prime Line Selection

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 first press keys or dial codes. This simplifies handling calls on a frequently used trunk.

Prime Line Selection has the following two modes of operation:

❑ Outgoing Prime Line Preference

  Lifting the handset seizes the Prime Line. Outgoing Prime Line Preference would help 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).

- ❑ Incoming Prime Line Preference

  When the Prime Line rings the extension, lifting the handset answers the call. Incoming Prime Line Preference could benefit the Service Department dispatcher who must quickly answer customer service calls and then dispatch repair technicians. The dispatcher has the assurance that whenever a customer calls in, the dispatcher lifts the handset to get their call. (Incoming Prime Line Preference can optionally seize an idle line appearance.)

Private Line

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.

Programmable Function Keys

Enhancements

Version 1500 or higher required for:

- ❑ Allowing 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 of if the telephone is off-hook.

In **Version 2000 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.

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 programmable keys from each multiline terminal. Depending on your telephone style, you can have either 4, 8, 16 or 32 Programmable Function keys.

Programming from a Multiline Terminal

Enhancements

Version 1100 or higher provides a temporary license (10 days) for the following: ACD, CTI, firmware upgrade, Hotel/Motel, 256 Megaco stations and SMDR.

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.

Pulse to Tone Conversion

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 (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.

Quick Transfer to Voice Mail

Enhancements

Version 2000 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.

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.

Redial Key

Users can press the Redial Key to cycle through the last 10 outside numbers dialed. Pressing the # key redials the number displayed. Users can also press the Redial Key and dial a System Speed Dial bin number to access System Speed Dial.

Remote (System) Upgrade

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 CPU. The upgrade can be applied immediately, or at a scheduled date and time. Remote system upgrade can be done with a TCP/IP, Serial, or Modem connection.

Repeat Redial

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).

Resident System Program

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 or DSS Console]. This enables immediate operation, even before the system is programmed to accommodate the individual site requirements.

Reverse Voice Over

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 just presses 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 presses 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.*

Ring Groups

Enhancements

In **Version 2000 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.

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.

Ringdown Extension, Internal/External

With a Ringdown Extension, a user can call another extension, outside number, or Speed Dialing number by 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.

Room Monitor

Room Monitor lets an extension user listen to the sounds in a co-worker 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 only Monitor one extension at a time. 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.

Save Number Dialed

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.

Secondary Incoming Extension

Secondary Incoming Extensions (SIEs) 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.

Secretary Call (Buzzer)

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.

Secretary Call Pickup

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 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 is helpful to customer service representatives that must frequently cover other 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.

Selectable Display Messaging

An extension user can select a preprogrammed Selectable Display Message for their extension. Display multiline terminal callers see the selected message when they call the user 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 could then tell when the user was coming back from vacation. The system allows all telephones to use the Selectable Display Messaging feature at the same time.

All telephones are able to use Selectable Display Messaging at one time.

The default messages are:

Table 2-1 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	

Selectable Ring Tones

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.

Serial Call

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."

Single Line Telephones, Analog 500/2500 Sets

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 ETU 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. 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 timers in Program 20-03-04 and 23-03-07.

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 23-03-07 System Options for Single Line Telephones – Forced Dial Sending Start Time 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).

SLT Adapter

The SLT (Single Line Telephone) Adapter allows a port of an ESIB(8)-U( ) 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.

SNMP Simple Network Management Protocol

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 more 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 CPUII, since the Elite IPK II CPUII controls most of the circuit cards this provides a great advantage of capturing most of the circuit's card status, failure and alarm details.

Softkeys

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.

Speed Dial - System/Group/Station

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 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 having up to 24 digits.

When placing a 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 there are 2000 Speed Dialing bins available in the system, once programmed, these bins can currently only be dialed 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.)

The Speaker key 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 extension number under the other (e.g., #201). The DSS Console user presses the first key to call the company, waits for the Automated Attendant to answer, then presses the second key to call the client (extension 400).

The DSS Console user can also chain to a 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, a 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 out the stored number. This provides true one-touch calling via a telephone function keys.

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 distinquishes whether the telephone is set for DND/Call Forward All Calls of if the telephone is off-hook.

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.

Station Hunt

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.

Station Message Detail Recording (SMDR)

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 American, European or Japanese format. Set the format for your preference.

- **Transferred Call Tracking**

  SMDR shows each extension's share of a transferred call. If an outside call is transferred among four extensions, SMDR shows how long each caller stayed on the call.

- **Data Call Tracking**

  Data Call Tracking can log the system 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 you 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 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 CPUII (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 within 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 DIALED NO./CLI and ACCOUNT areas, 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.

Station Name Assignment – User Programmable

This feature allows a user to program the Station Name for their telephone extension or any extension in the system. The name is displayed on the multiline terminal LCD when an intercom or K-CCIS call is placed.

Station Relocation

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.

Synchronous Ringing

Synchronous Ringing synchronizes CO/PBX incoming ringing with the incoming ringing pattern from a Central Office.

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 Enabled. This allows the T1/PRI Interface ETU to be programmed as a 4/8/12/16/20/24 port Fractional T1/PRI.

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 additional 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 lines [1]

With **Version 1100 or lower**, the T1/PRI Interface ETU uses the first block of 24 consecutive trunks. For example, if you have an 8COIU 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 T1PRI. **Version 1500 or higher** is required *when* PRG 10-39-01 is set to Enabled.

1. 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:

- Selectable Receive Format

  You can set up the system for compatibility with any combination of ANI, DNIS and Dialed Number (Address) data provided by the Telco.

- 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.

Tandem Ringing

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.

A multiline terminal must be paired with 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.

Tandem Trunking (Unsupervised Conference)

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 ETU 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 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 to automatically set up a Conference/Tandem Trunking call on hanging up the telephone.

- ❏ Method D – Automatic Tandem Trunking Setup to Abbreviated 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 to automatically set up a Conference/Tandem Trunking call on 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.

TAPI Compatibility

Enhancements

Version 2000 provides new 1st party TAPI drivers for CTA or CTU adapters.

In versions prior to Version 2000, the 1st party TAPI drivers only supported single line mode. With Version 2000, the CTA or CTU drivers can be configured for single line mode or multi line mode. Multi line mode will allow 3rd party applications that support muti line mode to also screen pop on calls to virtual extensions on the station.

The system has Telephony Applications Programming Interface (TAPI) ability. TAPI ability provides:

- ❏ Reduced TAPI Feature set.
- ❏ Caller ID data to the PC for data base lookups and screen pops.
- ❏ Telephone control (off-hook, on-hook and dialing).

The CTA or CTU adapter 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.

Tone Override

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.

Traffic Reports

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 totalled is determined by system programming. The system outputs this data to the PC for the total period.

Transfer

Enhancements

Version 1500 or higher allows DSS/One-Touch keys to be used for one-touch transfer instead of using the Transfer key.

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 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 would have to 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 has an option that 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 to Barge-In to transfer a call to 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 would allow, 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 to Trunk Ring Group Available

It is possible to transfer a trunk call to the trunk 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 Incoming Ring Group (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 : Incoming 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 can also be used with the VRS. This allows the caller listening to the VRS message 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

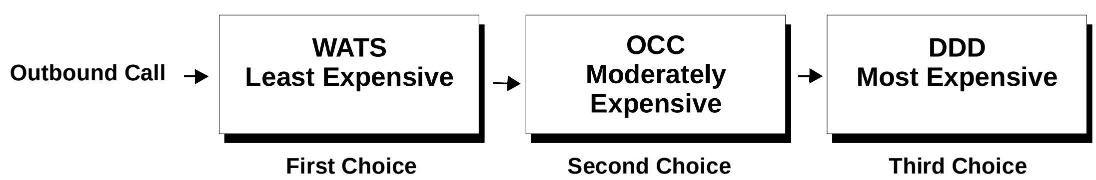
When the Transfer key is pressed while a user is on a call, the call is placed on hold.

Trunk Group Routing

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 will help 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.



Trunk Groups

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

Trunk Queuing/Camp-On

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.

Unified Messaging

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 CTI 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, View Call 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 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 board.

- ❑ 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

  Upon purchasing hospitality:

  - ❍ On VMP-U40, three languages are unlocked on the system and this is the limit.
  - ❍ On CTP-U10, five languages are unlocked - additional languages can be purchased up to a limit of 18.

- ❑ Text-to-Speech

  This converts e-mails 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

Uniform Call Distribution

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 Sales department extension numbers.

UCD uses Circular Routing. Each new call rings the extension that has been idle the longest in the group.

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 cycle 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 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 2-100 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

Uniform Numbering Network

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 has the ability to 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.

Universal Slots

Enhancements

With **Version 1500 and 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 and higher**, the DTI/PRI ETUs can be programmed as a 4/8/12/16/20/24-port Fractional T1/PRI (when 10-39-01 is set to be enabled).

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.

User Programming Ability

A station user can perform programming functions. Speed Group Dialing and Function Keys are just two features programmable from a station.

Virtual Extensions

Enhancements

Version 2000 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
  - ❍ Tone Override to a busy Virtual Extension
  - ❍ 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 will be displayed. Prior to this release, Caller ID would not be displayed.

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.

Voice Mail Integration (Analog)

Enhancements

Version 1500 or higher supports flexible Voice Mail integration to the Analog Voice Mail ports.

Version 2000 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 (PGM 15-02-35)
- ❑ Message Waiting Lamp Cycle for Called Extension (PGM 15-02-36)
- ❑ Voice Mail Message Wait Lamp Color (PGM 15-02-37)
- ❑ Voice Mail Message Wait Lamp Cycle (PGM 15-02-38)

  This provides more distinction between incoming calls, Message Waiting (Set/Received) and a VM Message Waiting indication.

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 company 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. 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 the user or a co-worker 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 calls trying to get to the voice mail are 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.

Voice Mail Message Indication on Line Keys

Voice Mail Message Indication on Line Keys indicates a new voice mail message on Line Keys or DSS/BLF keys.

Voice Over

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's conversation.

Voice Over could help a lawyer, for example, waiting for an urgent call. While on a call with another client, the lawyer 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

Voice over Internet Protocol (VoIP)

VoIP 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 using 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.[2]

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).

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).

The DSP daughter board provides the option for 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 prerecorded 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.

2. 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 IP-based remote voice solution. Therefore, NEC recommends connecting VoIP equipment through a local area network using a Private IP address.

- ❑ **ACD Messages** - provides announcement and overflow messages for ACD groups.
- ❑ **Transfer to the VRS** - any extension user can 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.

Volume Controls

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. Pressing the VOLUME ▲ or VOLUME ▼ adjusts the volume level for whichever feature is active (*e.g.,* outside call, ICM, ICM ringing, or paging). Pressing these keys when the telephone is idle adjusts the contrast level of the telephone display. The users should set the volumes for their most comfortable levels.

Warning Tone For Long Conversation

The system can broadcast warning tones to a trunk caller warning them that they have 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 is 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 time 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).

Wireless – DECT

The Wireless – DECT (Digital Enhanced Cordless Telecommunication) system allows the use of 2.4 GHz IPK 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. A 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 256 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 65 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 pre-dial or phone book numbers.

- ❑ Programming of 2 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 IPK 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 IPK 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.

Equipment

Chapter 3

SECTION 1 EQUIPMENT LIST

The tables below list all equipment used with the Electra Elite IPK II system. The equipment name, a description of the equipment, and the maximum capacity allowed for a Basic Port Package and an Expanded Port Package are given. The Equipment Name is listed alphabetically by category.

The maximum capacities available in the Electra Elite IPK II system are shown in Table 3-1 Maximum System Capacities for Station Interface ETUs, Table 3-2 Maximum System Capacities for Trunk Interface ETUs, and Table 3-3 Maximum System Capacities for Application Interface ETUs.

Table 3-1 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(8)-U( ) 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, 6

Table 3-1 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 3-1 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
CTP(4)/(8)-U( ) ETU CTP(12)/(16)-U( ) ETU	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: • 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 One DSP8-U10 ETU is required for 8-ports. Two DSP8-U10 ETUs are required for 12- or 16-ports.	1	1	Notes 1, 3, 4, 6
DSPII-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 CPUII( )-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	7	23	
ESIB(8)-U( ) ETU	This 8-port Electronic Station Interface ETU contains eight circuits. Each circuit can support any Attendant Console, Multiline Terminal, or Single Line Telephone adapter. 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.	7	23	Notes 1, 6

Table 3-1 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	These combined ETUs provide a 16-port Electronic Station Interface. The ESIE ETU is installed on the ESIB ETU. Each Electronic Station Interface ETU contains eight circuits. Each circuit can support any Attendant Console, Multiline Terminal, or Single Line Telephone adapter. 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.	3	15	Notes 1, 6
FMS(2)/(4)-U( ) ETU	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.	1	1	Notes 1~4, 6
FMS(8)-U( ) ETU	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.	1	1	Notes 1, 3, 4, 6
IVR Application [VMP(4)/(8)-U( ) ETU with IVR HDD Kit]	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.	1	1	Notes 1, 3, 4, 6

Table 3-1 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 3-1 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 combined ETUs provide 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
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, 6
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

Table 3-1 Maximum System Capacities for Station Interface ETUs (Continued)

Station Interface Units	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
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 ETU and the 4- or 8-port ETU 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 capacity 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, or 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: When the DSPII-U10 Unit with an In-Mail 2- or 4-port Compact Flash is installed on the CPUII( )-U10 ETU, it uses eight ports from the maximum station port capacity.

Table 3-2 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 8 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 can provide an E911 CAMA trunk. 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 can provide an E911 CAMA trunk. This ETU shares the total number of CO/PBX lines in the system.	7	23	Note 1

Table 3-2 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. This ETU can provide an E911 CAMA trunk. 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. This ETU can provide an E911 CAMA trunk. 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 3-2 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	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. 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	Notes 1,4
DTI-U40 ETU for DTI Function DTI-U40 ETU for PRI Function	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. 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. 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.	2	11	Notes 1, 5, 7.

Table 3-2 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 capacity 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 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 6 are assigned 16 Channels, and the 11th is assigned eight channels.

Table 3-3 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
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	

Table 3-3 Maximum System Capacities for Application Interface ETUs (Continued)

Application Interface ETUs	Description	Maximum Capacities		Notes
		Basic Port Package	Expanded Port Package	
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	
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 (MG16) DSP Resources in increments of four (4, 8, 12 and 16). **Version 2000 or higher** – Using the MG_CCIS Combo Package 4-port license registration provides user with the flexibility of adding CCISoIP ports in increments of four (4, 8, 12, 16, 20 or 24) or (MG16) DSP resources in increments of four (4, 8, 12, 16, 20 or 24).	2	11	Note 4

Note 1: Refer to the KSU Power-Based Calculator Chart.

Note 2: A maximum of three BSU(4M)/(2S)/(6S) Wireless – DECT ETUs can be installed.

Note 3: The IAD(8)-U( ) ETU is assigned as a CCISoIP ETU and counts as 8-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.

A maximum of one BSU(M) Master and two BSU(2S)/(6S) Slave ETUs can be installed.

SECTION 2 KSU POWER-BASED CALCULATOR CHART

The Card Calculator on the next page allows you to determine the maximum power consumption for the Power supply in each cabinet under the following conditions:

- ❑ Each basic cabinet can contain a maximum of 10 cards.
- ❑ Each expansion cabinet can contain a maximum of nine cards.
- ❑ The total point value cannot exceed 1000 points for +5V.

❑ The total point value cannot exceed 1000 points for -24V.

To calculate the two values (+5V and -24V) for a card:

1. Pick the card type in the chart below.
2. Calculate the +5V total point value by multiplying the number of cards by the +5V value in the chart.
3. Calculate the -24V total point value by multiplying the number of cards by the -24V value in the chart.

An example is shown below:

❑ +5V

Five ESIB(8)-U10 ETUs multiplied by a table value of 21 is 105 points toward a possible 1000 (895 points remaining).

❑ -24V

Five ESIB(8)-U10 ETUs multiplied by a table value of 83 is 415 points toward a possible 1000 (585 points remaining).

Table 3-4 KSU Power-Based Calculator Chart

Package Name	Power Consumption		Number of ETUs	Equivalent Total Power Point	
	+5V	-24V		+5V	-24V
Common					
CPUII( )-U10 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			
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			

Table 3-4 KSU Power-Based Calculator Chart (Continued)

Package Name	Power Consumption		Number of ETUs	Equivalent Total Power Point	
	+5V	-24V		+5V	-24V
DTI-U40 ETU	73	0			
IAD(8)-U( ) ETU	31	110			
TLI(2)-U( ) ETU	15	10			
Station					
CNF(16)-U20 ETU	17	52			
ESI(8)-U( ) ETU	21	83			
ESIB(8)-U10 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			
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			

Table 3-4 KSU Power-Based Calculator Chart (Continued)

Package Name	Power Consumption		Number of ETUs	Equivalent Total Power Point	
	+5V	-24V		+5V	-24V
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**

An example of KTS Configuration with a Basic and Expansion Cabinet using the Calculator Chart is shown below:

SAMPLE CALCULATION USING KSU POWER-BASED CALCULATOR CHART			
Package	Quantity	Total Power Consumption +5V value from Chart	Total Power Consumption -24V value from Chart
Basic Cabinet			
CPUII( )-U10 ETU	1	200	0
ESIB/E(8)-U10 ETU	3	96	498
ESIB(8)-U10 ETU	1	21	83
SLIB(4)-U10 ETU	3	90	51
SLIE(4)-U( ) ETU	3	51	36
CTP(8)-U10 ETU	1	120	288

Sample Calculation Using KSU Power-Based Calculator Chart			
Package	Quantity	Total Power Consumption +5V value from Chart	Total Power Consumption -24V value from Chart
Totals	**10**	**578**	**956**
Expansion Cabinet			
EXP-U10 ETU	1	6	0
DTI-U30 ETU	1	84	0
DTI-U40 ETU	1	73	0
CCH(4)-U10 ETU	1	50	0
ESIB/E(8)-U10 ETU	4	128	664
COIB(4)-U30 ETU	1	29	5
Totals	**9**	**370**	**669**

Installation, Programming, and Maintenance Overview

SECTION 1 INSTALLATION

Reduced Installation Time
The Electra Elite IPK II System uses modularity and connectivity throughout to reduce installation time and labor. The modular Key Service Units (KSUs) are installed vertically for the Electra Elite IPK II system. Most internal connections are made with plug and jack.

Reducing the labor required for installation, modularity and connectivity increases reliability. No wiring changes are made in the KSUs and all connectors are factory tested.

The power supply unit and the battery backup unit are installed in the KSU and allow easy connection to extra battery backup units. All circuits installed in the KSUs are located on printed circuit boards (ETUs) that plug into prewired connector slots. Connection for voice and data between the KSUs is provided by a single ribbon cable between the basic and expansion KSUs. Voice and data are transmitted between KSUs using an EXP-U( ) ETU in the Electra Elite IPK II system.

Connection to telephones, outside lines, and other external devices is made using telephone cable connectors. A music source for Music on Hold is connected by standard audio equipment plugs.

Universal Slots
Using Universal Slots maximizes flexibility by allowing installation of any ETU in any interface slot. Full use of each KSU, before adding another, reduces hardware requirements.

Resident System Program
A Resident System Program is provided when the system first receives power. The CPU scans the KSUs and recognizes the ETUs and Multiline Terminals that are connected to the system. Standard (default) values are assigned in the System Program for all system and device parameters to allow the system to operate immediately after initialization, before programming is done.

The assignments provided by the Resident System Program can be altered to fit the requirements of a particular installation. Changing programming assignments is the function of multiline terminals or a personal computer. When programming from a multiline terminal, Flexible Line keys and the dial pad are used to enter new values, and the display provides the necessary information for programming.

Multiline Terminals and Single Line Telephones
A variety of telephones can be connected to satisfy the requirements of a particular installation. All Multiline Terminals are fully modular and are powered from the central unit. Cabling is twisted 1-pair for proprietary multiline terminals and single line telephones.

SECTION 2 PROGRAMMING

From Multiline Terminals
Programming is done using DTH/DTR/DTP-8D-1 TEL, ITH/DTU-8D-2 TEL, DTH/DTR/DTP-16D-1 TEL, ITH/DTU-16D-2 TEL, DTH/DTR/DTP-32D-1 TEL, or DTU-32D-2 TEL multiline terminal.

When a programming multiline terminal is off-line in the Program Mode, the rest of the system continues to function. Most program changes can be entered anytime, but some changes take effect only when the affected stations and circuits are idle. This avoids disrupting calls in progress.

PC Programming
System data can be transferred to/from a disk for backup. The System Program End User software allows end users to program several features for their Multiline Terminals, such as: Line Key Assignment, Telephone Names, Zone Paging Groups, or various timers.

Battery Backup
The battery on the CPUII( )-U( ) is used to retain the Clock/Calendar and Last Number Redial (LNR) buffers for each station when the CPU encounters a power loss. When the battery is fully charged, the settings are retained for approximately three years. The Lithium (CR2032) battery should be replaced every two years. The system programmed memory (Customer Database) is stored in Non-Volatile Memory and can only be erased by a First Initialization.

The batteries, located in the KSUs, support system operation for up to 30 minutes during a power outage.

User Programmable Features
Multiline terminal users can also program the following features from their station:

- ❑ Ringing Line Preference
- ❑ Feature Access and/or One-Touch keys (*e.g.*, Speed Dial or Direct Station Selection)
- ❑ Speed Dial

Multiline terminals without programmable One-Touch keys and single line telephones can be used to program Station Speed Dial memories. Attendant Positions can be used to program System Speed Dial memories and the System Clock/Calendar.

SECTION 3 MAINTENANCE

Installing Interface ETUs without Disrupting Ongoing Calls
Each interface and optional ETU has an ON/OFF switch with an LED indication of power status. An interface ETU with this switch OFF can be removed or installed with the system power on.

The combination of status indication and ETU replacement with power on allows the maintenance technician to replace suspect circuits without disrupting ongoing calls.

Up/Down Load of Data
Using PC Programming, Station Speed Dial data, System Speed Dial data, and all System Data can be transferred from/to a PC. The Up/Down Load may be accomplished from a local or remote location.

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Hardware Specifications

SECTION 1 SYSTEM CAPACITY

The Electra Elite IPK II system consists primarily of the Basic KSU and the Expansion KSUs. Expansion KSUs can be stacked vertically on the Basic KSU to expand the system capacity. Two expansion KSUs can be added to expand to 24 interface slots. The B64-U20 KSU is used for the basic and both expansion KSUs.

1. Basic KSU: 8 interface slots
2. Basic KSU + Expansion KSU: 16 interface slots
3. Basic KSU + 2 Expansion KSUs: 24 interface slots

System capacities of the Electra Elite IPK II system are listed below in Table 5-1 System Capacities.

Table 5-1 System Capacities

Hardware	Maximum Capacities/System Slots
Basic Unit	8 interface slots
Basic + Expansion Unit	16 interface slots
Basic + 2 Expansion Units	24 interface slots

Refer to Section 3 System Description on page 1-7 in Chapter 1 Introduction for maximum system capacities.

SECTION 2 TRAFFIC CAPACITY

Table 5-2 Traffic Capacity provides information about the traffic capacity for the Basic Port Package and the Expanded Port Package.

Table 5-2 Traffic Capacity

Traffic Capacity	Basic Port Package	Expanded Port Package
Traffic Capacity (CPUII)	4800 BHCA	4800 BHCA

✎ 4800 Busy-Hour Call Attempts (BHCA) is based on a 176Trunk/240 Station configuration.

SECTION 3 CABLING REQUIREMENTS AND SPECIFICATIONS

This section provides cabling requirements and specifications for various equipment used in the Electra Elite IPK II system.

Figure 5-1 Connecting the ESI Using Twisted 2-Pair Cable provides a diagram of the KSU connected with each of the multiline terminals and single line telephones by a separate twisted 1-pair cable or 2-pair cable (only for Multiline Terminals).

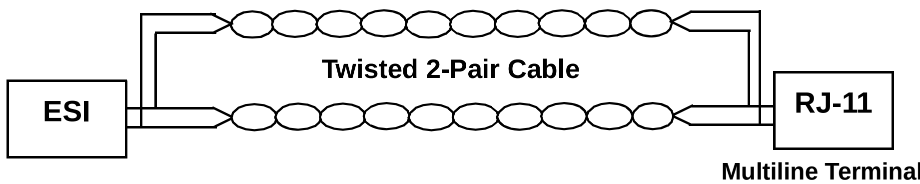


Figure 5-1 Connecting the ESI Using Twisted 2-Pair Cable

Refer to the following tables for cabling requirements and specifications.

- ❑ Table 5-3 D^{term} Series i or D^{term} IP Terminal Loop Resistance and Cable Length
- ❑ Table 5-4 Electra Elite/Dterm Series E Multiline Terminal Loop Resistance and Cable Length
- ❑ Table 5-5 Cable Connection Between the Analog Port and the Single Line Equipment
- ❑ Table 5-6 Cabling Requirements

Table 5-3 *D*term Series i or *D*term IP Terminal Loop Resistance and Cable Length

Terminal or Adapter	Maximum Loop Resistance (without AC Adapter) (Ohms)	By Twisted 1-Pair Cable (without AC Adapter) 24 AWG	By Twisted 2-Pair Cable (without AC Adapter) 24 AWG	Maximum Loop Resistance (with AC Adapter) (Ohms)	By Twisted 1-Pair Cable (with AC Adapter) 24 AWG	By Twisted 2-Pair Cable (with AC Adapter) 24 AWG
DTH-8-1 TEL DTR-8-1 TEL	37	700	1400	107	2000	2000
DTH-8D-1 TEL DTR-8D-1 TEL ITH-8D-2/3 TEL	37	700	1400	107	2000	2000
DTH-16-1 TEL DTR-16-1 TEL	35	660	1320	107	2000	2000
DTH-16D-1 TEL DTR-16D-1 TEL ITH-16D-2/3 TEL	35	660	1320	107	2000	2000
DTH-32D-1 TEL DTR-32D-1 TEL	26	500	1000	107	2000	2000
DTH-16LD-1 TEL	37	700	1400	107	2000	2000
DCR-60-1 Console✱	—	—	—	107	2000	2000

✱ An AC Adapter is required.

Table 5-4 Electra Elite/D^{term} Series E Multiline Terminal Loop Resistance and Cable Length

Terminal or Adapter	Maximum Loop Resistance (Ohms)	Maximum Feet by Twisted 1-Pair Cable 24 AWG	Maximum Feet by Twisted 2-Pair Cable 24 AWG
DTU-8-1 TEL DTP-8-1 TEL	35	600	1000
DTU-8D-2 TEL DTP-8D-1 TEL	35	600	1000
DTU-16-1 TEL DTP-16-1 TEL	26	450	900
DTU-16D-2 TEL DTP-16D-1 TEL	26	450	900
DTP-16HC-1 TEL✱	57	1083	
DTU-32-1 TEL DTP-32-1 TEL	21	360	720
DTU-32D-2 TEL DTP-32D-1 TEL	21	360	720
DTR-2DT-1 TEL	35	600	1000
DTR-4D-1 TEL	37	700	1400
DTR-4R-1/2 TEL	N/A	650	1000
DTH-4R-1/2 TEL	N/A	650	1000
SLTII(1)-U( ) ADP✱✱	35	600	1000
DP-D-1	20	410	820

✱ An AC Adapter is required for the DTP-16HC-1 TEL.

✱✱ The length for the specified SLT Adapter is between the SLT Adapter and the ESI.

Table 5-5 Cable Connection Between the Analog Port and the Single Line Equipment

Connected Equipment	Cable	Maximum Feet from Connected Equipment to Telephone
AD(A)-R or AD(A)-2R Unit	Twisted Pair	10 feet
ADA(2)-W Unit	Twisted Pair	10 feet
AP(A)-R or AP(R)-R Unit	Twisted Pair	50 feet
APA-U Unit or APR-U Unit	Twisted Pair	50 feet
OPX(2)-U( ) ETU	Twisted Pair	1,600 ohms
SLI(4)/(8)-U( ) ETU	Twisted Pair	300 ohms
SLTII(1)-U( ) ADP	Twisted Pair	50 feet

✎ Mixing digital and analog ports through the same 25-pair cable runs is not recommended.

Table 5-6 Cabling Requirements

Connected Equipment	Cable
Music on Hold and Background Music Sources	Hi-Fi Shielded Audio Cable
External Amplifier	Hi-Fi Shielded Audio Cable
ITH Cabling	Cat 5 Straight Data Network Cable - 100 meters maximum distance.

SECTION 4 POWER REQUIREMENTS

The power supply inputs and the power consumption specifications for the Electra Elite IPK II are listed below.

4.1 Power Supply Inputs

The AC input [P64-U( ) PSU] requirements for the Electra Elite IPK II are listed below:

- ❑ 117 Vac ± 10%
- ❑ 60 Hz ± 10%
- ❑ Single Phase
- ❑ 7.5A circuit
- ❑ A dedicated outlet, separately fused and grounded.

4.2 Power Supply Consumption

Table 5-7 Power Consumption

KSU	Maximum RMS Current	Watts Used (Idle)	Watts Used (Maximum)
Basic KSU – B64-U20 KSU	2.5 A	120	230
Basic KSU + Expansion KSU	5.0 A	240	460
Basic KSU + 2 Expansion KSUs	7.5 A	360	690

When replacing fuses, refer to the specifications in Table 5-8 Fuse Replacement.

Table 5-8 Fuse Replacement

Unit	Fuse Number	Specifications	Description	Dimensions
P64-U( ) PSU	F1	125V, 6.0A	AC Input	1/4” x 1 1/4”
P64-U( ) PSU	F101	250V, 10A	Battery Input	1/4” x 1 1/4”

✎ All fuses are normal blown glass tube.

Do not use slow blow fuses. Replace with a fuse of the same type and rating.

SECTION 5 ENVIRONMENTAL CONDITIONS

5.1 Temperature

- ❑ Operating: +32°F ~ +104°F (0°C ~ 40°C)
- ❑ Recommended Long Term: +50°F ~ +90°F (10°C ~ 32.2°C)

5.2 Humidity

- ❑ Operating: 10% ~ 90% noncondensing

5.3 Weights and Dimensions

Table 5-9 Weights and Dimensions indicates these values for the Units, ETUs and KSUs.

Table 5-9 Weights and Dimensions

Unit	Shipping Weight [1]	Height	Width	Depth
ACA-U Unit	22.5 oz (638 g)	3.4” (86 mm)	4.2” (107 mm)	5.2” (133 mm)
AD(A)-R Unit	4.0 oz (113 g)	2.25” (56.25 mm)	2.75” (68.75 mm)	5.5” (137.5 mm)
AD(A)-2R Unit	4.0 oz (113 g)	2.25” (56.25 mm)	2.75” (68.75 mm)	5.5” (137.5 mm)
AP(A)-R Unit	5.6 oz (158 g)	2.25” (56.25 mm)	2.75” (68.75 mm)	5.5” (137.5 mm)
AP(R)-R Unit	5.6 oz (158 g)	2.25” (56.25 mm)	2.75” (68.75 mm)	5.5” (137.5 mm)
B64-U20 KSU	460.8 oz (13063 g)	13.0” (328.7 mm)	14.0” (354 mm)	10.25” (259 mm)
BRT(4)-U10 ETU	14.6 oz (414 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
BRT(4)-U20 ETU	11.3 oz (320 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
BSU(4M)-U20 ETU	14.8 oz (419 g)	1.97” (50mm)	8.27” (210 mm)	11.47” (290 mm)
BSU(2S)-U20 ETU	13.4 oz (381 g)	1.97” (50mm)	8.27” (210 mm)	11.47” (290 mm)
BSU(6S)-U20 ETU	15 oz (423 g)	1.97” (50mm)	8.27” (210 mm)	11.47” (290 mm)
CCH(4)-U( ) ETU	12.0 oz (340 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
CMS(2)/(4)-U( ) ETU	102.4 oz (2903 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
CMS/FMS/VMS-U30 ( ) ETU	102.4 oz (2903 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
CNF(8)-U( ) ETU	12.0 oz (340 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
CNF(16)-U20 ETU	12.3 oz (349 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
COI(4)-U( ) ETU	13.6 oz (385 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
COI(8)-U( ) ETU	16.6 oz (471 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
COIB(4)-U( ) ETU	14.4 oz (408 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)

Table 5-9 Weights and Dimensions (Continued)

Unit	Shipping Weight [1]	Height	Width	Depth
COIB(8)-U( ) ETU	16.6 oz (471 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
COID(4)-U( ) ETU	14.4 oz (408 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
COID(8)-U( ) ETU	16.6 oz (471 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
CPUII( )-U10 ETU	13.4 oz (380 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
CT(A)-R Unit	4.0 oz (113 g)	2.25” (56.25 mm)	2.75” (68.75 mm)	5.5” (137.5 mm)
CTA-U Unit	4.3 oz (122 g)	2.4” (60 mm)	2.3” (59 mm)	4.8” (121 mm)
CTI/VP(4)/(8)/(12)/(16)-U( ) ETU	192 oz [2] (5443 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
CTP-U10 ETU	20 oz [3] (580 g)	0.8” (20 mm)	7.5” (190 mm)	7.5” (190 mm)
CTU(C)-U Unit	9.5 oz (270 g)	2.4” (60 mm)	4.3” (110 mm)	4.4” (112 mm)
CTU(S)-U Unit	9.5 oz (270 g)	2.4” (60 mm)	4.3” (110 mm)	4.4” (112 mm)
CT(U)-R Unit	8.4 oz (239 g)	2.25” (56.25 mm)	2.75” (68.75 mm)	5.5” (137 mm)
DCR-60-1 Console	53 oz (1503 g)	4.2” (107 mm)	12.8” (326 mm)	7.14” (182 mm)
DID(4)-U( ) ETU	15.5 oz (439 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
DP-D-1A Doorphone	8.4 oz (238 g)	1.5” (38 mm)	5.5” (140 mm)	4.6” (121 mm)
DTH-16D-1 TEL DTR-16D-1 TEL	43.5 oz (1233 g)	4.78” (122 mm)	10.2” (260 mm)	9.8” (250 mm)
DTH-8-1 TEL DTR-8-1 TEL	41.0 oz (1163 g)	4.78” (122 mm)	10.2” (260 mm)	9.8” (250 mm)
DTH-8D-1 TEL DTR-8D-1 TEL	43.5 oz (1233 g)	4.78” (122 mm)	10.2” (260 mm)	9.8” (250 mm)
DTH-32D-1 TEL DTR-32D-1 TEL	48 oz (1361 g)	4.78” (122 mm)	10.2” (260 mm)	9.8” (250 mm)
DTI-U( ) ETU	13.2 oz (374 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)

Table 5-9 Weights and Dimensions (Continued)

Unit	Shipping Weight [1]	Height	Width	Depth
DTI-U40 ETU	5.99 oz (170 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
DTP-1-1 TEL DTP-1-2 TEL DTP-1HM-1 TEL DTP-1HM-2 TEL	26.8 oz (760 g)	2.36” (60 mm)	6.22” (158 mm)	8.81” (224 mm)
DTU-16-1 TEL DTP-16-1 TEL	41 oz (1162 g)	4.8” (123 mm)	7.8” (197 mm)	9.3” (235 mm)
DTU-16D-2 TEL DTP-16D-1 TEL	43.5 oz (1233 g)	4.8” (123 mm)	7.8” (197 mm)	9.3” (235 mm)
DTP-16HC-1 TEL	53 oz (1503 g)	6.00” (152 mm)	9.08” (230 mm)	8.04” (204 mm)
DTP-2DT-1 TEL	41 oz (1163 g)	4.8” (123 mm)	7.8” (197 mm	9.3” (235 mm)
DTU-32-1 TEL DTP-32-1 TEL	46 oz (1304 g)	4.8” (123 mm)	8.7” (220 mm)	9.3” (235 mm)
DTU-32D-2 TEL DTP-32D-1 TEL	48 oz (1361 g)	4.8” (123 mm)	8.7” (220 mm)	9.3” (235 mm)
DTU-8-1 TEL DTP-8-1 TEL	41.0 oz (1163 g)	4.8” (123 mm)	7.8” (197 mm)	9.3” (235 mm)
DTU-8D-2 TEL DTP-8D-1 TEL	43.5 oz (1233 g)	4.8” (123 mm)	7.8” (197 mm)	9.3” (235 mm)
DTR-1-1 TEL DTR-1HM-1 TEL	26.8 oz (760 g)	2.47” (100 mm)	7.65” (195 mm)	9.54” (243 mm)
DTR-1R-1 TEL	14.4oz (408 g)	4.5” (114 mm)	6.1” (153 mm)	8.62” (218 mm)
DTR-2DT-1 TEL	41 oz (1163 g)	2.47” (100 mm)	7.65” (195 mm)	9.54” (243 mm)
DTR-4D-1 TEL	44 oz (1250 g)	5.98” (152 mm)	8.54” (217 mm)	9.65” (245 mm)
DTR-4R-1 TEL	15.4 oz (437 g)	2.25” (57 mm)	4.25” (108 mm)	7.5” (191 mm)
DTU-4R-1 TEL	15.4 oz (437 g)	2.25” (57 mm)	4.25” (108 mm)	7.5” (191 mm)
D16(LD)-R ADM	27 oz (770 g)	4.33” (110 mm)	10.24” (260 mm)	7.09” (180 mm)
ESI(8)-U( ) ETU	14.5 oz (411 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)

Table 5-9 Weights and Dimensions (Continued)

Unit	Shipping Weight [1]	Height	Width	Depth
ESIB(8)-U( ) ETU	11.1 oz (315 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
ESIE(8)-U( ) ETU	9.9 oz (280 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
EXP-U( ) ETU	14.6 oz (414 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
FMS(2)/(4)/(8)-U( ) ETU	102.4 oz [2] (2903 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
HF-R Unit	9.9 oz (280 g)	2.9” (74 mm)	4.2” (106 mm)	5.6” (141 mm)
HUB(8)-U( )ETU	10.4 oz (294 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
IAD(8)-U( ) ETU	8.11 oz (230 g)	7.5” (190 mm)	6.3” (160 mm)	0.87” (22 mm)
ITH-8D-2/3 TEL	50.92 oz (1445 g)	9.84” (250 mm)	10.31” (262 mm)	4.76” (121 mm)
ITH-16D-2/3 TEL	50.92 oz (1445 g)	9.84” (250 mm)	10.31” (262 mm)	4.76” (121 mm)
IVR Application VMP(4)/(8)-U( ) ETU with IVR HDD	14.6 oz (414 g)	1.75” (44 mm)	10.5” (266 mm)	8.62” (219 mm)
OPX(2)-U( ) ETU	13.4 oz (380 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
PRT(1)-U( ) ETU	13.2 oz (374 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
PVA(X)-U( ) ETU	8.0 oz (230g)	0.7” (18 mm)	6.5” (165 mm)	7.5” (190 mm)
RAK-U( ) Unit	320 oz (9072 g)	20” (507 mm)	15” (380 mm)	8.5” (216 mm)
SLI(4)-U( ) ETU	13.0 oz (370 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
SLI(8)-U( ) ETU	14.1 oz (400 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
SLIB(4)-U10 ETU	13.0 oz (370 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
SLIE(4)-U10 ETU	10.7 oz (303 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)

Table 5-9 Weights and Dimensions (Continued)

Unit	Shipping Weight [1]	Height	Width	Depth
SLTII(1)-U( ) ADP	9 oz (255 g)	1.8” (45 mm)	2.8” (70 mm)	4.8” (120 mm)
TLI(2)-U( ) ETU	13.8 oz (391 g)	1.97” (50 mm)	9.45” (240 mm)	7.68” (195 mm)
VMS(2)/(4)/(8)-U( ) ETU	102.4 oz [2] (2903 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)
VMP(2)/(4)/(8)-U( ) ETU	102.4 oz [2] (2903 g)	1.89” (48 mm)	11.47” (290 mm)	8.46” (214 mm)

1 Shipping weight includes the shipping carton.
2 Shipping weight includes the shipping carton and documentation.
3 Drive is shipped separately.

SECTION 6 AUDIBLE AND VISUAL INDICATION

6.1 Tone Patterns

Table 5-10 Tone Patterns lists the frequency and the pattern for the tones. Tones are used to inform Electra Elite IPK II station users of system functions such as dial tone, busy tone, or ringback tone.

6.2 Multiline Terminal LED Flash Patterns

The Electra Elite IPK II system has 2-color LEDs. Green is used primarily for I-Use conditions and for outside calls. Red is used primarily for Other Use conditions and internal calls. Refer to Table 5-11 Multiline Terminal LED Flash Patterns.

Table 5-10 Tone Patterns

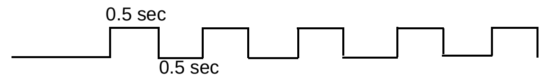
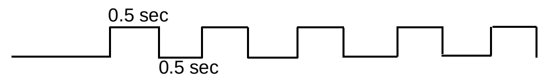
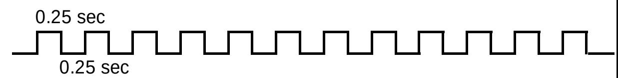
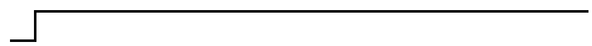
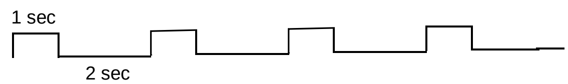
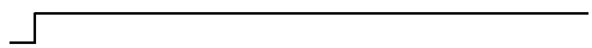
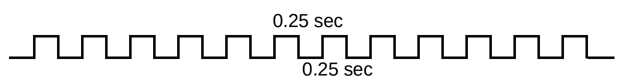
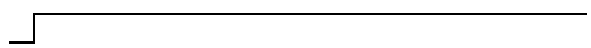
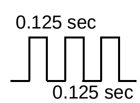
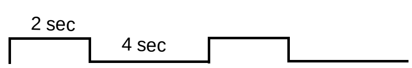
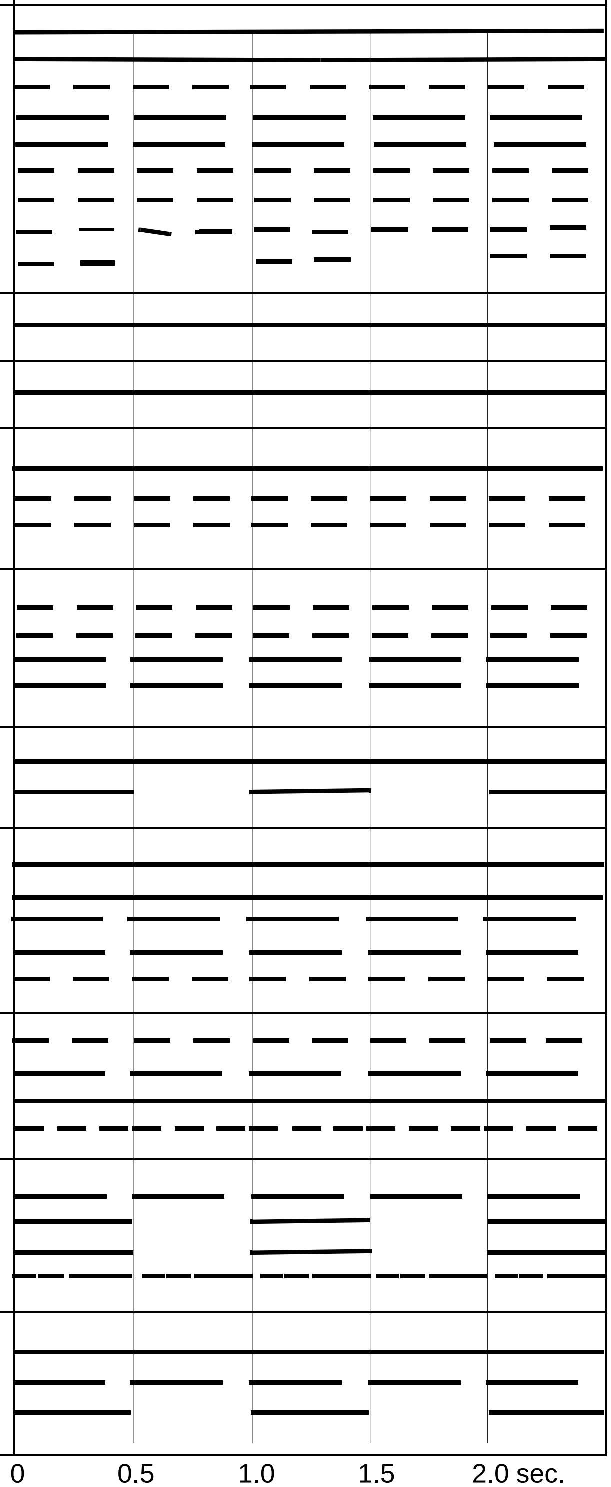
System Tone (Fixed)	Frequency (Hz) (Fixed)	Intermit (Default)	Cycle
Busy Tone	480/620	60 IPM	
Call Waiting Tone	440	60 IPM	
Second Dial Tone	350/440	120 IPM	
Howler Tone	2400 Modulation (16 Hz)	Continuous	
Internal Dial Tone	350/440	Continuous	
Internal Ringback Tone	440/480	1 sec On 2 sec Off	
LCR Dial Tone	440	Continuous	
Reorder Tone	480/620	120 IPM	
Service Set Tone	440	Continuous	
Special Dial Tone	440	240 IPM	
Tone Burst 1 Tone	440	Continuous	
Tone Burst 2 Tone	620	Continuous	
Tie/DID Ringback Tone	440/480	2 sec On 4 sec Off	
Camp-On Tone Call Alert Notification Attendant Tone Override	440	Continuous	
DIT Alert Tone	480/620	Continuous	
Call Forward Alert Tone Call Forward Configuration Tone	350/440	120 IPM	

Table 5-11 Multiline Terminal LED Flash Patterns

LED	Condition	Color	Flash Patterns
Line Key	I-Use Busy Incoming Call I-Hold Call Hold Hold Recall Transfer Recall Live Monitoring Mode Message Waiting on Line Key	Green Red Red Green Red Green Green Green Red	
Microphone	ON	Red	
Mic	ON (Series i)	Red	
ICM	I-Use ICM Incoming Call Voice Over Broker	Red Red Red	
Large LED	Incoming Internal Call Incoming Outside Call Message from Attendant Voice Mail Message	Red Green Green Red	
Speaker	ON System Data Entry	Red Red	
Conf	Conference in Progress/Barge In All Conference Circuits Used Hold Conference Call ICM Call Hold SPD Confirmation	Red Red Red Red Red	
Answer	Incoming Trunk Exclusive Hold User Ringing Line Preference Voice Over with Broker's Call	Red Green Red Green	
Feature	Callback Set Auto Repeat Set ON (to set function) Call FWD - All Calls Set	Red Red Red Red	
BLF or DSS Key	Use, Hold DND, Call FWD-All Calls Set Special Mode (while pressing Feature or going off-line)	Red Red Red	



SECTION 7 OUTSIDE LINE TYPES

The following outside lines can be used with the Electra Elite IPK II system.

- 2-wire, Loop Start or Ground Start Trunks
- 2-wire, 2-way DID Lines (Dial Pulse or DTMF)
- 4-wire, E&M Tie Lines (Type I or V, Dial Pulse, or DTMF)
- Digital Trunk T1/FT1 (Loop Start, Ground Start, Tie Line (E&M), or DID Signaling)
- ISDN-BRI Trunks
- ISDN-PRI Trunks
- VoIP Trunks (Internet Protocols)

SECTION 8 NETWORK AND CONTROL

8.1 Transmission, Network, and Control Specifications

8.1.1 Transmission

- Data Length:

  From Multiline Terminal to ESI(8)-U( ) ETU: 23 bits

  From ESI(8)-U( ) ETU to Multiline Terminal: 23 bits

- Data Transmission Rates:

  Between ESI(8)-U( ) ETU and Multiline Terminal: 184K bps (voice and signaling)

- Scanning Time for each Multiline Terminal: 32 ms.

8.1.2 Network

Time Division Multiplexing (TDM) allows transmission of data and voice simultaneously over one communications medium. The specifications that the Electra Elite IPK II system uses for switching, clock, data bus, and timeframe are shown below.

- TDM Switching: PCM (µ Law)
- TDM Clock: 2.048 MHz
- TDM Data Bus: 8 bit
- TDM Timeframe: 125 µs.

8.1.3 Control

This section indicates the speed or capacity:

- ❑ Control: Stored program with distributed processing
- ❑ Central Processor: 32-bit microprocessor
- ❑ Clock: 25 MHz
- ❑ Interface ETU: 8- or 16-bit microprocessor
- ❑ Optional ETUs: 16- or 32-bit microprocessor
- ❑ Multiline Terminal (TDM):   8-bit microprocessor
- ❑ Multiline Terminal (IP): 32-bit microprocessor
- ❑ IP Adapter: 32-bit microprocessor
- ❑ Attendant Console: 4-bit microprocessor
- ❑ SLT Adapter: 4-bit microprocessor

8.1.4 Electra Elite IPK Terminals and Equipment

The voltage, current, and ring signal for the Electra Elite IPK Multiline Terminals, Single Line Telephone equipment, and AP(A)-R/AP(R)-R Units are listed below:

- ❑ Multiline Terminal

  Voltage: -11 ~ -26 Vdc

  Maximum Current: 250 mA

  > Acoustical characteristics meet Electronic Industry Association (EIA) standard proposal SP-1286 and standard EIA RS-470.

- ❑ Single Line Telephone

  Standard 2500 Set: 500 type network

  Nominal Current: 35 mA

  Ring Signal: 56 Vac RMS @ 20 Hz

- ❑ SLTII(1)-U( ) ADP

  Standard 2500 Set: 500 type network

  Nominal Current: 30 mA

  Ring Signal: 56 Vac RMS @ 20 Hz

- ❑ AP(A)-R Unit

  Standard 2500 Set: 500 type network

  Nominal Current: 30 mA

- ❑ AP(R)-R Unit

  Standard 2500 Set: 500 type network

  Nominal Current: 30 mA

  Ring Signal: 56 Vac RMS @ 20 Hz

8.1.5 Series i Terminals

- ❑ The voltage and current for the *D*[term] Series i Multiline Terminals are listed below:

  Voltage: -11 ~ -48 Vdc

  Maximum Current: 250 mA

Acoustical characteristics meet Electronic Industry Association (EIA) standard proposal SP-1286 and standard EIA RS-470.

- ❑ Voltage, current, and ring signal information for Single Line Telephone equipment, AP(A)-R Unit, and AP(R)-R Unit are the same a those listed in the previous paragraph.

SECTION 9 DIALING SPECIFICATIONS

9.1 Dial Pulse Address Signaling

Dial Pulse Address Signaling uses dial pulses (regular momentary interruptions) to signal the equipment. The following Dial Pulse specifications are used In the Electra Elite IPK II system.

- ❑ Pulse Rate: 10 ± 0.5 pps/20 ± 1.0 pps
- ❑ Percent Break: 60 ± 1.5%
- ❑ Interdigit Interval: 0 pps/20 pps 770 ms. ~ 830 ms.

9.2 Dual-Tone Multifrequency (DTMF) Address Signaling

DTMF signaling includes push button or Touchtone dialing. When a key on a telephone is pushed, two tones (one high frequency and one low frequency) are provided. In the Electra Elite IPK II system, the following DTMF specifications are used.

- ❑ Frequencies

  Two sinusoidal frequencies are provided, one from the high frequency group and one from the low frequency group.

- ❑ Frequency Deviation: Less than ±1.0%
- ❑ Signal Level:

  Nominal level per frequency: -6 ~ -4 dBm

  Minimum level per frequency

  Low Group: -10 dBm

  High Group: -8 dBm

  Maximum level per frequency: 0 dBm

- ❑ Rise Time: Within 5 ms.
- ❑ Duration of Dual Frequency Signal:

  110 ms. default/60 ms. minimum

  Interdigital Time: 80 ms. default/70 ms. minimum

Nominal **High** Group Frequencies (Hz)

Nominal **Low** Group Frequencies (Hz)

	1209	**1336**	**1477**
697	1	2	3
770	4	5	6
852	7	8	9
941	✱	0	#

SECTION 10 EXTERNAL EQUIPMENT CONNECTION

10.1 Music Source for Music on Hold via KSU

- ❑ Auxiliary Input: 0.6V PPS Signal Level
- ❑ Input Impedance: 600 Ω

10.2 Music Source for Station Background Music via ACI

- ❑ Auxiliary Input: 0.6V PPS Signal Level
- ❑ Input Impedance: 600 Ω

10.3 External Paging (Audio)

- ❑ Output Power: –10 dBm Signal Level
- ❑ Output Impedance: 600 Ω
- ❑ Relay Contact Rating: 500 mA, 24 Vdc

10.4 External Tone Ringer/Night Chime Output

- ❑ Output Level: –10 dBm
- ❑ Output Impedance: 600 Ω
- ❑ Relay Contact Rating: 500 mA, 24 Vdc

10.5 SMDR Output

- ❑ Female Connector (System Output) Standard DB-9 (straight)

10.6 PC Connection

- ❑ Female Connector (System Output) Standard DB-9 (straight)

10.7 Relay Contact

- ❑ All Relay Contact Ratings: 500 mA, 24Vdc

SECTION 11 BATTERY BACKUP

The Electra Elite IPK II system has battery backup functions for system backup and for memory backup.

11.1 System Backup

During a power failure, the system is backed up using a rechargeable battery. This battery backup supports all system operations for approximately 30 minutes.

When a brownout or power failure occurs, and the battery backup circuit is not activated, Time and date, Terminal status (e.g., MIC), and SMDR data are reset. System data is not lost due to the battery backup circuit.

When a CPUII( )-U10 ETU is installed and the system or battery backup fails for any reason, the clock/calendar must be set. The fully charged battery retains memory for approximately three years.

11.2 Memory Backup

The CPUII( )-U10 ETU battery retains the Clock/Calender and Last Number redial (LNR) buffers for each station when the CPUII( )-U( ) ETU encounters a power loss. With a fully charged battery, the settings are retained for about three years. The System Programmed memory (Customer Database) is stored in non-volatile Memory and can be erased only by a First Initialization. After power is restored, the system returns to normal operation.

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Electra Elite IPK II

General Description Manual

NEC Unified Solutions, Inc.

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