Empowered by Innovation NEC

1. The Basics

2. The Components

3. Features

4. Specifications and Parts List

UX5000

Product Description

P/N 0913010
Rev 1, June 2008
Printed in U.S.A.
1.00

Technical Support Web Site:
http://www.necux5000.com

This manual has been developed by NEC Unified Solutions, Inc. It is intended for the use of its customers and service personnel, and should be read in its entirety before attempting to install or program the system. Any comments or suggestions for improving this manual would be appreciated. Forward your remarks to:

NEC Unified Solutions, Inc.
4 Forest Parkway
Shelton, CT 06484
necunifiedsolutions.com

Table of Contents

Table of Contents

Table of Contents

Ensure Your Business Success

In today's highly competitive business environment, effective and reliable communication is critical to the success of your business. Providing the latest Voice over Internet Protocol (VoIP) technology and comprehensive desktop solutions that deliver superior performance, efficiency, flexibility, and reliability when and where you need it, is key to survival and growth in today's information-driven business environment. With over 100 years of building powerful communication and technology solutions, NEC understands the numerous and ever changing demands and challenges that exist in today's growing market. NEC accepts these challenges and meets the customer's demands for a connected world – head on!

NEC leverages its strengths to bring forth innovations, and to integrate those innovations into new products that will provide seamless communications, customer satisfaction and manageability throughout your organization. This powerful combination of innovative design, high quality assurance, and maximum integrated solutions creates a converged business environment and a highly competitive advantage.

NEC offers a winning strategy for their customers, empowering them to take advantage of new opportunities, and trust that they have invested in a sound communication server for today and tomorrow.

Offer The Latest VoIP Technology and Improved Functionality

In today's technology driven market, your communication server and your personal computer are invaluable tools that are central to your business. Therefore, it is important to invest in the latest communications and technology solutions that will dramatically improve functionality and performance of these two essential tools and deliver increased productivity throughout your entire organization.

Enjoy Freedom of Choice with Investment Protection

The UX5000 is a comprehensive integrated solution designed to meet the unique challenges of both business telephony applications and VoIP. This top performance communication server supports pure peer-to-peer IP telephony connectivity, advanced networking, traditional digital switching, or a combination - all from one solution!

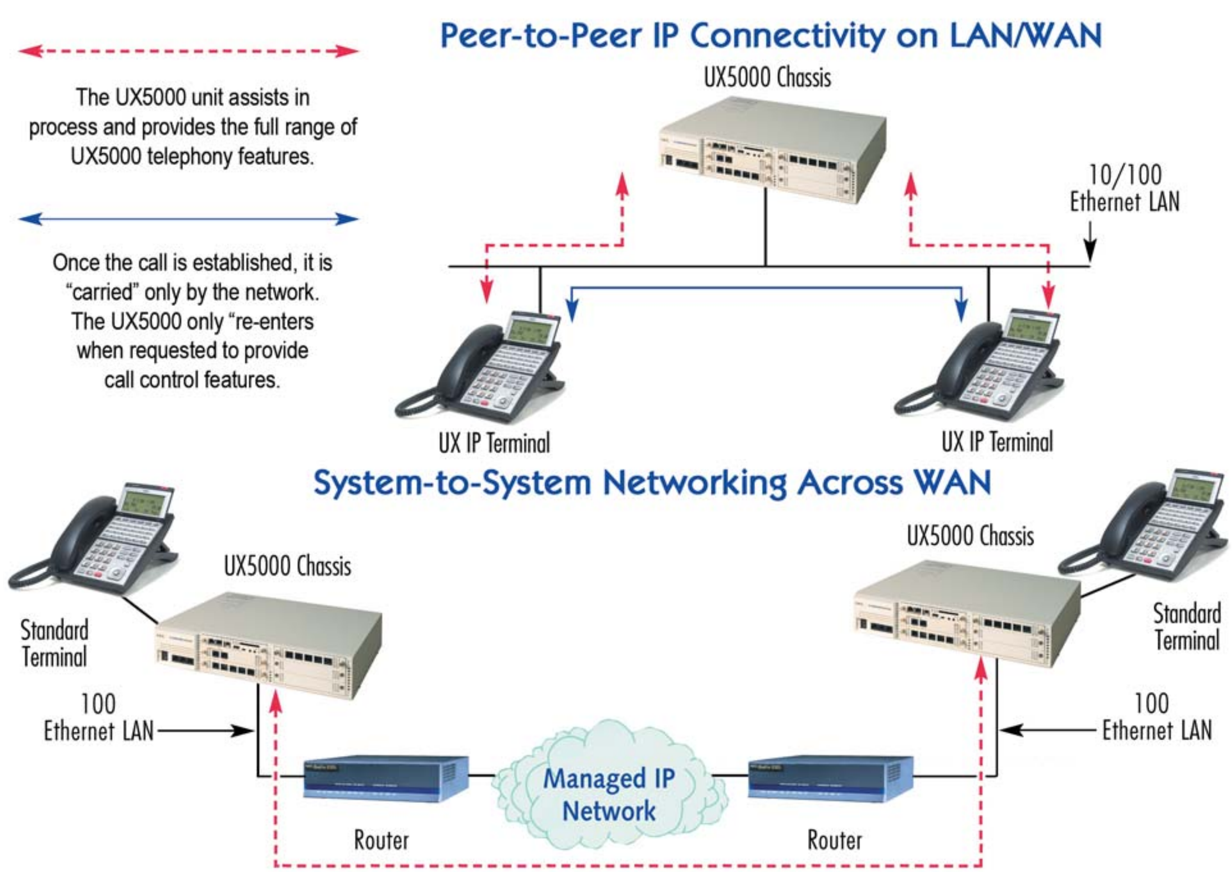
It allows your organization to converge your voice and data network and benefit from the cost-saving advantages, convenience, and ease of use afforded by networked communication servers. VoIP provides seamless internal and external communications and access to advanced data and productivity tools.

By integrating diverse hardware components and software applications, NEC brings control of telephony features and related call information right to the user's PC, and provides advanced Computer Telephony Integration (CTI) throughout your organization. Reduce costs and improve network efficiency by transparently sharing communication features and resources between branch or remote locations with CygniLink. Share voice mail and other applications for additional cost savings.

Even if you're not ready to migrate to 100% IP telephony immediately, UX5000 will work for you. You can deploy traditional circuit-switched technology, VoIP or a combination! You have the freedom to adopt VoIP when and where you need it without rendering your existing systems obsolete - providing a superior return on your investment.

Reduced Costs of Peer-to-Peer IP Connectivity

- Maintain one network rather than two
- Bypass the long distance carrier by sending voice calls over the data network
- Single cable termination to the desktop
- Reduced brick and mortar expenses by deploying main office features to remote personnel.



Customize The Best Communication Solution For Your Business

The extensive feature set and reliable call processing applications are mature, efficient and dependable - yet intuitive and easy to use.

The architecture and design of the UX5000 delivers high performance, optimal voice quality, and reliability. A compact yet powerful solution that is simple to deploy, administer and maintain.

The UX5000 can start small and can cost-effectively expand to 712 ports.

Connectivity . . .

- Network Efficiency utilizes a single network for both voice and data.
- Link multiple business locations to provide feature transparency between communication servers which improve employee collaboration and communication.
- Share resources such as trunks, operator services and voice mail.
- Automatic Fail-Over is provided when connecting communication servers with CygniLink.
- IP terminals communicate by Peer-to-Peer, which means that the IP terminals participating in a call are connected directly to each other over an IP network. The signals travel through the IP network, not through a telephone switch as in traditional telephony.

Manageability . . .

- Reduce Total Cost of Ownership – Reduce the expense of initial setup, moves, adds and changes with our intuitive programming interface.
    - ❖ UX5000 PCPro/WebPro – An instrumental programming and maintenance tool that empowers users to manage their own terminals and provides them with the functionality needed to simplify terminal setup and changes. Windows®-based PCPro provides centralized online HTML-based programming access. With this intuitive browser software and its easy-to-follow wizards, programming is simplified and the time needed to complete it is significantly reduced. Administrators can schedule automatic updates to the UX5000 software remotely.
- Reduce Future Cost of Ownership – Today's investment is protected for tomorrow. The UX5000 is designed to transition to new technologies as the need arises. No need to replace an entire system to accommodate changes in your business requirements. The UX5000 allows you to protect the company's investment with modular expansion and technology updates as your business grows.
- Easily maintain all communication servers on the network from a single location, connected to the network locally or remotely.
- Choice of Either IP or Digital Terminals – Whether your business communications are pure IP or any combination of IP and traditional circuit - switched technology, NEC provides a full line of terminals that will meet your needs. Because the user interface and the terminal functionality remain the same for IP or digital versions, employees can easily transition between models.

Usability . . .

- Whether you are in your home office or on the road, an IP terminal offers all the same feature-rich capabilities as your desktop terminal. Users can place, receive, or transfer calls as if sitting at a desk in the office.
- With the UX Softphone application, a computer becomes an IP terminal and all features of the office terminal are available with the click of a mouse. Mobile workers can place calls, receive calls or check voice mail while away from the office. Adding a web cam can deliver video between another camera equipped Softphone.
- Color Touch Screen, provided by the IP-CTS Terminal offers intuitive icon-based feature operation and graphical XML support for a superior user experience.
- Select terminal models provide paperless key labeling. Key labels automatically change as the button functions are customized.
- Customized terminal options are available for specialized applications including enlarged dial pad, selectable font size, labeling options and more . . .

Improve Customer Experience, Deliver Productivity and Versatility to Your Work Environment

Automatic Call Distribution (ACD) - Distributes calls evenly among member agents and provides initial and repeating announcements that encourage callers to remain on the line. Callers can leave a message if they choose to receive a callback from an agent. A client-based Supervisor position provides traffic management reporting and agent scheduling capability.

Multimedia Conference Server - Eliminates the ongoing cost of using outside vendors to host conference calls. The browser accessible Conference Server allows the user the ability to schedule, host or participate in a conference call with ease and efficiency. Users receive an Email with the telephone number and password to dial into the conference. Hosted video conference is supported for web cam equipped PCs.

UX5000 Desktop Suite - Is an integral part of the overall workstation; It is the combination of three complimentary software applications designed to help users become more mobile, productive and better informed. Components of the Desktop Suite are:

- PC Attendant - Call handling capability can be performed right from your PC. Display visuals let you know if an extension is in use, idle, call forwarded, or set for do not disturb. In addition, conversations can be recorded, saved, and forwarded as an Email attachment.

  Instant Messaging - The PC Attendant can initiate an Instant Message, and the users can then easily respond with displayed soft keys.

- PC Assistant - Provides management and operation of a desktop terminal from a PC - for easy Abbreviated Dialing, call management, contact lookup, and seamless CRM integration.

- Softphone - The UX Softphone application provides convenient, cost-effective mobility. A computer becomes a phone and all features of the office terminal are available with the click of a mouse. Add a web cam to deliver video between another camera equipped Softphone.

Video Conference in Collaboration with UX Softphone - Upon placing a call, the terminal can automatically identify whether that caller has a video enabled PC. With the push of a button, the video enabled PC can be activated to establish a video conference.

Call History - Saves information about incoming and outgoing calls and are accessible to the user. Logged calls can be redialed or saved to memory.

Bluetooth Hub Adapter - Users can synchronize peripheral equipment such as PDAs, mobile phones, headsets, conference units, and keyboards with the terminal.

XML Open Interface Support - Enables developers to create displayable and accessible applications via UX IP terminals. Applications such as calendar links, wallboards, directories, stock tickers, news reports, and more can be displayed.

Downloadable Ring Tones - Download from your favorite web site. Ring Tones can be programmed and assigned to people in a person's directory. When an incoming call arrives, a user hears an identifying distinctive ring tone and can immediately know who is calling.

Secure Mode - Offers three levels of protection - Personal, Corporate and Telephony modes. Each IP terminal can be locked to prevent access or use, thus ensuring privacy and security of your corporate directories and terminal data.

E911 Compatibility - Identifies the origination of a 911 call so emergency services can reach the specific extension location quickly.

SIP (Session Initiation Protocol) - Compatible with most SIP trunk providers and a variety of SIP terminals.

InRouter – The InRouter offers an intelligent, all in one networking and monitoring solution for NEC communications servers. A feature-rich blade that delivers reliability and performance by combining multiple voice and data features into a converged networking router. In addition, the InRouter includes security features, simplified troubleshooting and diagnostics for Quality of Service necessary for VoIP applications.

PoE Gigabit Switch – The UX5000 PoE Gigabit fully managed 8 port switch brings gigabit speed to your users while adding a whole new level of intelligence and security to your network.

Mobility Solutions

Select from a variety of Mobility Solutions to keep your customers and team connected - while providing access to all your communication server's advanced communication and voice messaging features.

Connectivity, Mobility and Convenience –
Reduce brick and mortar expenses by deploying main office operations at remote locations. Connect to the office communication server from a branch office, hotel room, customer site, or anywhere with broadband access. Users at home or virtually anywhere can place and receive calls transparently as if they were seated at a desk in the company office. Therefore, you are always ready to receive your customer's calls, and make the most of every business opportunity.

Bluetooth Cordless Handset - For mobility, efficiency and improved customer service. When using the Bluetooth cordless handset you have the ability to move about your personal workspace without being tethered to your desk.

Cordless Phones - Provide the freedom to move about in your workspace with multi-line call handling capability.

IP DECT Wireless Handsets - Make or receive important calls from anywhere in your workplace. Provides the flexibility to set your wireless handset to have the same extension as your desk terminal or operate separately.

WiFi Handset - When using NEC access points, delivers multi-line call handling capability with roaming throughout your workplace. If outside the workplace, the WiFi Handset can be used from any hot spot to access many UX5000 features.

Mobile Extension - Gives the ability to use your cell phone as a single line extension of the UX5000. Forward your desk terminal to your cell phone and easily receive calls, transfer incoming calls to other extensions, make intercom calls, access your voice mail, or perform any number of other call-handling communication server features – all from your cell phone.

Messaging Options

Manage your messages more effectively and enhance your communication server by delivering your information more quickly and efficiently wherever you may be.

Choose from a variety of capabilities to provide unified communications - including the ability to consolidate multiple sources such as Voice Mail, Fax Mail, and Email to your in box and PDA.

UX5000 Versatility Feature List

Usability

- 7 Color LED Status Indicator
- Application Sharing
- Backlit Display
- Built-In Headset Jack
- Call History
- Contrast Control
- File Transfer
- Illuminated Dial Pad
- Instant Messaging
- Last Number Redial List
- One Touch Feature Operation
- Tilt Display
- Video Conference
- Video Softphone
- Whiteboard

Flexibility

- Adjustable Height Telephone
- Flexible Numbering Plan
- Secure Station Relocation
- Universal Blade Slots
- Virtual Extension keys

Serviceability

- Alarm Notification
- Automatic Software Upload
- Fail Over
- Remote Programming
- Self Diagnostics
- Web-Based Programming

Versatility

- Analog Trunks and Stations
- Bluetooth Support
- Colored Face Mats
- CSTA/TAPI Support
- Digital Trunks and Terminals
- IP Trunks and Terminals
- Modular Terminal Components
- SIP
- WiFi Handsets
- XML Support
- Secure Mode
- Toll Restriction
- Walking Class of Service

Manageability

- Automatic Call Distribution
- Automatic Terminal Relocation
- Built-In Mini Gatekeeper
- Conference Scheduler
- PoE Gbit Switch
- QoS Router Blade

Scalability

- Application Processors
- CygniLink IP Network
- Distributed Processing
- Up to 712 Ports

Adaptability

- 19” Rack or Wall Mountable
- Adjustable Height Terminal
- IP and Digital Terminal Options
- Messaging Options
- Universal Blade Slots

The Terminals

✍ **Make a note. . .**
For your convenience. . . there is a Parts List located at the end of this guide. You should find this list helpful when selecting system equipment. More detailed tools are also available — ask your Account Representative for the specifics.

IP-CTS – P/N 0910080

The IP-CTS terminal is the system's premier IP terminal, featuring a backlit color touch screen, 11 fixed feature keys, and 4 interactive Soft Keys. By pressing the icons on the touch screen, the display changes to provide intuitive access to the system's most sophisticated features. Every IP-CTS terminal has a built-in speakerphone for full Handsfree operation. Handsfree Answerback and Intercom voice-announce capability is also standard.

The telephone's 32 programmable function keys can be customized by the user for one-button access to co-workers, features like Paging or Park or specific outside lines. The color touch screen displays 8 keys at a time. There are a total of 4 screens to access all 32 keys - the user selects which page is to be viewed. The programmable key will light/flash to help the user see which calls are for them and which features are active. Access to other commonly used features is simplified by 11 fixed feature keys.

In addition, the IP-CTS terminal provides a built-in wall-mount bracket, as well as adjustable legs which allow each terminal to be angled at a height which best suits the user.

This is available as an IP terminal.

- Color-LCD with Touch Panel
- XML Open Interface Support
- Full Duplex Handsfree
- Gigabit Ethernet
- Menu/Soft Key Operation
- Navigation Pad
- Information Protection/Lock Button
- Backlit Dial Pad
- Backlit Display
- Directory
- XML Application
- Enhancement of Incoming Ringing
- Bluetooth Interface
- Network management (voice quality improvement)
- Downloading
- NAT & Firewall Traversal
- IPv6 (future)
- VPN
- Security Enhancement of Auto Configuration
- Network Authentication
- Security (Summary)
- LLDP (future)
- Protection for Terminal Information

✍ **At a glance (some features require optional equipment)**

IP-CTS (*IP*) Terminal					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	✔
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	No
Half Duplex	-			BCH Adapter:	✔
Dual LEDs:	✔	12 Line Key Kit:	No	BHA Adapter:	✔
Backlit LCD:	✔	16-Button DLS	✔	PSA Adapter:	✔
Backlit Display	✔	Headset Jack:	✔	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	✔	Navigator Pad	✔

Enhanced: DESI-Less – P/Ns 0910056 & 0910058 (Digital) P/Ns 0910076 & 0910078 (IP)

This terminal is available as an IP terminal or as a digital keyset.

The 32-Button DESI-Less Display Terminal has 2 gray scale displays - one for call control and one for line keys/programmable function keys. The call control display provides 4-lines and 28-characters with four interactive soft keys for intuitive feature access. The line key/programmable function key display shows 8 keys at a time. There are a total of 4 screens to access all 32 keys - the user selects which page is to be viewed. The dual LED for the key will light/flash to help the user see which calls are for them and which features are active. These user-programmable function keys provide for one-button access to co-workers, features and outside lines. Access to other commonly used features is simplified by 13 fixed feature keys.

The DESI-Less Terminal has a built-in speakerphone and can accept an optional adapter. You can also assign/connect 60-Button DSS Consoles or connect 16-Button DLS Consoles to these terminals. The DESI-Less Terminal provides Handsfree Answerback, Intercom voice-announcements. In addition, the terminals provide a built-in wall-mount bracket, as well as adjustable legs which allow each terminal to be angled at a height which best suits the user.

The DESI-Less Enhanced terminal features:

- Full-Dot Gray Scale Backlit Double LCD
- DESI-Less Line Key
- Full Duplex Handsfree
- Gigabit Ethernet
- Menu/Soft Key Operation
- Navigation Pad
- Information Protection Button
- Backlit Dial Pad
- Backlit Display

The IP terminal also supports the following features:

- Directory
- XML Application
- Enhancement of Incoming Ringing
- Bluetooth Interface
- Network management (voice quality improvement)
- Downloading
- NAT & Firewall Traversal
- IPv6 (future)
- VPN
- Security Enhancement of Auto Configuration
- Network Authentication
- Security (Summary)
- LLDP (future)
- Protection for Terminal Information

✍ **At a glance**

Enhanced DESI-Less (*Digital*) Terminal **(some features require optional equipment)**					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	✔
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	✔
Half Duplex	-			BCH Adapter:	✔
Dual LEDs:	✔	12 Line Key Kit:	✔	BHA Adapter:	✔
Backlit LCD:	✔	16-Button DLS	✔	PSA Adapter:	✔
Backlit Display	✔	Headset Jack:	✔	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	No	Navigator Pad	✔

✍ **At a glance**

Enhanced DESI-Less (*IP*) Terminal **(some features require optional equipment)**					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	✔
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	No
Half Duplex	-			BCH Adapter:	✔
Dual LEDs:	✔	12 Line Key Kit:	✔	BHA Adapter:	✔
Backlit LCD:	✔	16-Button DLS	✔	PSA Adapter:	✔
Backlit Display	✔	Headset Jack:	✔	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	✔	Navigator Pad	✔

Enhanced: 12-Button Display– P/Ns 0910044 & 0910046 (Digital) P/Ns 0910064 & 0910066 (IP)

This terminal is available as an IP terminal or as a digital keyset.

The 12-Button Display Terminal features a 4-line, 28-character display with 4 interactive soft keys for intuitive feature access, in addition to 12 function keys with Dual LEDs. The function keys are user-programmable and can provide 1-button access to co-workers, features and outside lines. The terminal additionally provides12 additional fixed feature keys.

The 12-Button Display Terminal has a built-in speakerphone, provides Handsfree Answerback, Intercom voice-announcements. In addition, the 12-Button Display provides a built-in wall-mount bracket, as well as adjustable legs which allow each terminal to be angled at a height which best suits the user.

The terminal can accept an optional adapter as well as a 60-Button DSS Consoles or 16-Button DLS.

These Enhanced terminals feature:

- Full-Dot Gray Scale Backlit LCD
- Full Duplex Handsfree
- Gigabit Ethernet
- Menu/Soft Key Operation
- Navigation Pad
- Information Protection Button
- Backlit Dial Pad
- Backlit Display (standard on IP terminals and ***optional*** on digital terminals)

The IP terminal also supports the following features:

- Directory
- XML Application
- Enhancement of Incoming Ringing
- Bluetooth Interface
- Network management (voice quality improvement)
- Downloading
- NAT & Firewall Traversal
- IPv6 (future)
- VPN
- Security Enhancement of Auto Configuration
- Network Authentication
- Security (Summary)
- LLDP (future)
- Protection for Terminal Information

✍ **At a glance**

Enhanced (*Digital*) Terminal **(some features require optional equipment)**					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	✔	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	✔
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	✔
Half Duplex	-			BCH Adapter:	✔
Dual LEDs:	✔	12 Line Key Kit:	✔	BHA Adapter:	✔
Backlit LCD:	✔	16-Button DLS	✔	PSA Adapter:	✔
Backlit Display	✔	Headset Jack:	✔	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	No	Navigator Pad	✔

✍ **At a glance**

Enhanced (*IP*) Terminal **(some features require optional equipment)**					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	✔
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	✔	APR Adapter:	No
Half Duplex	-			BCH Adapter:	✔
Dual LEDs:	✔	12 Line Key Kit:	✔	BHA Adapter:	✔
Backlit LCD:	✔	16-Button DLS	✔	PSA Adapter:	✔
Backlit Display	✔	Headset Jack:	✔	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	✔	Navigator Pad	✔

Enhanced: 24-Button Display – P/Ns 0910048 & 0910050 (Digital) P/Ns 0910068 & 0910070 (IP)

This terminal is available as an IP terminal or as a digital keyset.

The 24-Button Display Terminal features a 4-line, 28-character display with 4 interactive soft keys for intuitive feature access, in addition to 24 function keys with Dual LEDs. The function keys are user-programmable and can provide 1-button access to co-workers, features and outside lines. The terminal additionally provides12 additional fixed feature keys.

The 24-Button Display Terminal has a built-in speakerphone, provides Handsfree Answerback, Intercom voice-announcements. In addition, the 24-Button Display provides a built-in wall-mount bracket, as well as adjustable legs which allow each terminal to be angled at a height which best suits the user.

The terminal can accept an optional adapter as well as a 60-Button DSS Consoles or 16-Button DLS.

These Enhanced terminals feature:

- Full-Dot Gray Scale Backlit LCD
- Full Duplex Handsfree
- Gigabit Ethernet
- Menu/Soft Key Operation
- Navigation Pad
- Information Protection Button
- Backlit Dial Pad
- Backlit Display (standard on IP terminals and ***optional*** on digital terminals)

The IP terminal also supports the following features:

- Directory
- XML Application
- Enhancement of Incoming Ringing
- Bluetooth Interface
- Network management (voice quality improvement)
- Downloading
- NAT & Firewall Traversal
- IPv6 (future)
- VPN
- Security Enhancement of Auto Configuration
- Network Authentication
- Security (Summary)
- LLDP (future)
- Protection for Terminal Information

✍ **At a glance**

Enhanced (*Digital*) Terminal **(some features require optional equipment)**					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	✔	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	✔
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	✔
Half Duplex	-			BCH Adapter:	✔
Dual LEDs:	✔	12 Line Key Kit:	✔	BHA Adapter:	✔
Backlit LCD:	✔	16-Button DLS	✔	PSA Adapter:	✔
Backlit Display	✔	Headset Jack:	✔	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	No	Navigator Pad	✔

✍ **At a glance**

Enhanced (*IP*) Terminal **(some features require optional equipment)**					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	✔
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	✔	APR Adapter:	No
Half Duplex	-			BCH Adapter:	✔
Dual LEDs:	✔	12 Line Key Kit:	✔	BHA Adapter:	✔
Backlit LCD:	✔	16-Button DLS	✔	PSA Adapter:	✔
Backlit Display	✔	Headset Jack:	✔	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	✔	Navigator Pad	✔

Value: 6-Button Display – P/N 0910042 (Digital)
P/N 0910062 (IP)

This terminal is available as an IP terminal or as a digital keyset.

The 6-Button Display Terminal features a 3-line, 28-character display with 4 interactive soft keys for intuitive feature access, in addition to 6 function keys with Dual LEDs. The function keys are user-programmable and can provide 1-button access to co-workers, features and outside lines. The terminal additionally provides12 additional fixed feature keys.

The 6-Button Display Terminal has a built-in speakerphone, provides Handsfree Answerback, Intercom voice-announcements. In addition, the 6-Button Display provides a built-in wall-mount bracket, as well as adjustable legs which allow each terminal to be angled at a height which best suits the user.

2

The digital terminal can accept a 60-Button DSS Console.

These Value terminals feature:

- Full-Dot Black and White LCD
- 6 Line Button with LCD

The IP terminal also supports the following features:

- Directory
- Enhancement of Incoming Ringing
- Network management (voice quality improvement)
- Downloading
- NAT & Firewall Traversal
- IPv6 (future)
- VPN
- Security Enhancement of Auto Configuration
- Network Authentication
- Security (Summary)
- LLDP (future)
- Protection for Terminal Information

✍ **At a glance (some features require optional equipment)**

Value (*Digital*) Terminal					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	No
Full Duplex	-	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	No
Half Duplex	✔			BCH Adapter:	No
Dual LEDs:	✔	12 Line Key Kit:	No	BHA Adapter:	No
Backlit LCD:	No	16-Button DLS	No	PSA Adapter:	No
Backlit Display	No	Headset Jack:	No	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	No	Navigator Pad	✔

✍ **At a glance (some features require optional equipment)**

Value (*IP*) Terminal					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console (though could be associated via digital port):	No
Handsfree (Speakerphone):				ADA Adapter:	No
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	No
Half Duplex	-			BCH Adapter:	No
Dual LEDs:	✔	12 Line Key Kit:	No	BHA Adapter:	No
Backlit LCD:	No	16-Button DLS	No	PSA Adapter:	No
Backlit Display	No	Headset Jack:	No	Retro Dial Pad	✔
Soft Keys:	✔	Lock Button:	✔	Navigator Pad	✔

Value: 2-Button w/o LCD – P/N 0910040 (Digital) P/N 0910060 (IP)

This terminal is available as an IP terminal or as a digital keyset.

The 2-Button Non-Display Terminal features 2 function keys with Dual LEDs. The function keys are user-programmable and can provide 1-button access to co-workers, features and outside lines. The terminal additionally 10 additional fixed feature keys.

The 2-Button Non-Display Terminal has a built-in speaker-phone, provides Handsfree Answerback, Intercom voice-announcements. In addition, the 2-Button Non-Display provides a built-in wall-mount bracket, as well as adjustable legs which allow each terminal to be angled at a height which best suits the user.

The terminal can accept a 60-Button DSS Console.

These Value terminals feature:

- 2 Line Buttons without LCD

The IP terminal also supports the following features:

- Directory
- Enhancement of Incoming Ringing
- Network management (voice quality improvement)
- Downloading
- NAT & Firewall Traversal
- IPv6 (future)
- VPN
- Security Enhancement of Auto Configuration
- Network Authentication
- Security (Summary)
- LLDP (future)
- Protection for Terminal Information

✍ **At a glance (some features require optional equipment)**

Value (*Digital*) Terminal					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console:	✔
Handsfree (Speakerphone):				ADA Adapter:	No
Full Duplex	-	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	No
Half Duplex	✔			BCH Adapter:	No
Dual LEDs:	✔	12 Line Key Kit:	No	BHA Adapter:	No
Backlit LCD:	No	16-Button DLS	No	PSA Adapter:	No
Backlit Display	No	Headset Jack:	No	Retro Dial Pad	✔
Soft Keys:	No	Lock Button:	No	Navigator Pad	No

2

✍ **At a glance (some features require optional equipment)**

Value (*IP*) Terminal					
Function Keys:	✔	Digital DESI-Less Line Key/LCD Unit:	No	60-Button DSS Console (though could be associated via digital port):	No
Handsfree (Speakerphone):				ADA Adapter:	No
Full Duplex	✔	IP DESI-Less Line Key/ LCD Unit:	No	APR Adapter:	No
Half Duplex	-			BCH Adapter:	No
Dual LEDs:	✔	12 Line Key Kit:	No	BHA Adapter:	No
Backlit LCD:	No	16-Button DLS	No	PSA Adapter:	No
Backlit Display	No	Headset Jack:	No	Retro Dial Pad	✔
Soft Keys:	No	Lock Button:	✔	Navigator Pad	No

Cordless Telephone — P/N 730088 & 730087

The UX5000 supports the Cordless Telephone. The DTR-4R-2 Cordless II (P/N 730088) is a 900 MHz spread-spectrum digital cordless telephone that provides mobility, flexibility and convenience for those who spend much of the workday away from their desk. Fully integrated with the telephone system, the DTR-4R-2 Cordless II offers many standard features such as Park, Do Not Disturb, Hotline, Voice Over and Voice Mail. Normally paired with a companion keyset for improved 1-button call coverage capabilities, the DTR-4R-2 Cordless II will also work as a stand-alone telephone.

Where users require greater range on the cordless phones and 3 or less cordless phones are being used at a specific site, the DTR-4R-2 Cordless II phone is recommended. Complemented by 4 fully programmable function keys (with LEDs), the DTR-4R-2 Cordless II achieves a whole new level of convenience and mobility. An easy-to-read LCD display, volume controls, a rechargeable nickel-cadmium battery pack and a handy belt clip round out the elegant and affordable DTR-4R-2 Cordless II Phone. This phone provides 10 channels.

A second Cordless Phone is also available: the Cordless Lite II (P/N 730087). The Cordless Lite II offers the same features as the DTR-4R-2 Cordless II except that it uses a NiMH battery and has FM modulation (single channel) instead of the spread spectrum modulation.

Where more than 3 cordless phones are to be used at one specific site, the Cordless Lite II phone is recommended. This phone provides 30 channels.

Cordless DECT Telephone - P/N 730095

This cordless telephone is adapted for the UX5000. It is designed for use in the office environment and provides a variety of features. These features are listed below.

- 1.9G (1920~1930 MHz)
- 5 Channels
- Display: 2 lines, 16-digit LCD
- 8 Programmable Keys: 4 Programmable Keys and 4 One Touch Keys
- Headset Jack
- Mute Control
- AutoStandby ®
- Separate Charging Stand with Spare Battery Charging Capability
- Handset/Headset/Handsfree Volume Control
- Adjustable Ringer Volume Control
- 6 Selectable Ring Tones
- Vibrating Ringer
- Out of Range Detection and Alarm Tone
- LCD Backlight
- Key Backlight
- Automatic Channel Selection
- Single Key Access to: Conference, Hold, Transfer and Redial features
- Low Battery Protection System
- Wall Mountable Separate Base Unit

- Wall Mountable Separate Charging Unit
- Easy Installation
- Compact Handset Design
- Use with an NEC Digital Keyset

The Base Unit allows users to switch between the Cordless DECT phone and the wired (desk) phone by using the Desk/Cordless buttons on the unit.

Cordless DECT Repeater - P/N 730639

The Cordless DECT Repeater lets you extend the coverage area of your Cordless DECT telephone in all directions, including up and down. If the repeaters are installed so their coverage area overlaps the coverage area of the base, the base can hand-off calls to the repeaters as the user moves from one coverage area to another. When connected to the repeater, the mobile handset operates the exact same way as it does when connected to the base, and the hand-off from the base to the repeater can be completely invisible to the end user, even during an active call.

Features of the Repeater include:
- Automatic registration to the base
- Up to six repeaters per base station
- Up to three repeaters in a sequential or daisy-chain layout
- Two internal antennas to support two simultaneous calls
- Repeater connection verification tone
- Low power consumption

DECT Wireless Access Point (DAP) - P/N 750104

The implementation of SIP DECT in the UX5000 is a stand-alone DECT system that is connected to the UX5000 via a TCP/IP connection using Session Initiation Protocol (SIP). This means that in the UX5000, the DECT extensions must be assigned as SIP extensions. From the UX5000 perspective, there is no difference between a SIP extension and a SIP DECT extension.

A DECT Access Point (DAP) is the actual transceiver. This unit supports up to 12 simultaneous calls and is for indoor applications. The DAP is equipped with internal antennas, no external antennas are necessary. In addition to the DAP unit, DAP Manager software is provided for programming the unit.

Power is obtained via the Ethernet connection using a data switch with 802.3af PoE, a PoE adapter or from an AC adapter (this AC adapter is not offered by NEC but available through an outside vendor).

The C124 SIP DECT Handset, P/N 750611, is used with this unit.

C124 SIP DECT Basic Handset - P/N 750611

The C124 DECT handset is a compact business handset for the user that simply wants to make and receive wireless calls while in the office. Features include a graphic display, internal directory, menu programming and handsfree operation. The design pays special attention to ergonomic considerations, to ensure optimal comfort and ease of use. The illuminated graphic LCD display allows the use in poorly lit environments. The internal loudspeaker provides hands-free operation with excellent sound quality. Advanced call-logging features enable incoming calls to be managed efficiently, without losing a single call. The applied DECT technology ensures crystal clear speech and powerful encryption techniques for secure communication.

Feature list:

- Crystal clear voice quality, seamless handover and powerful encryption
- Light slim-line ergonomic handset
- Illuminated graphic LCD
- Loud speaking
- Personal phone book (40 names)
- 10-entry call logging
- Redial list
- 5-step volume
- 13 languages supported
- Message Waiting Indication
- 200 hours stand-by time
- Up to 20 hours talk time
- 8 hours charging time

Included with this handset is a charger and AC adapter.

A 3rd Party/NEC SIP Client license is required.

DTH-1-1 (BK) Single Line Telephone - P/N 780034

This analog telephone provides the basic functions that a user may require. The telephone is only available in black.

This telephone provides:

- T/P (DTMF/Pulse) switch
- Volume control for the receiver and speaker (The volume control provides 6 different levels of adjustment to best suit the user.)
- Selectable ringing pitch and ringing volume
- Flash (to access certain public telephone exchange features)
- Redial
- Function to temporarily switch over of the dialing mode (switch from pulse to tone)
- 4 Memory buttons for one-touch speed dialing
- Message Waiting Indicator
- Handset Mic Mute key
- Wall-mountable

2

DTR-1-1 Single Line Telephone - P/N 780020 & 780021

When a simple analog telephone is required, the DTR-1-1 Single Line Telephone may be the one to select. The telephone is available in black or white.

This telephone provides an MF (DTMF)/DP switch, volume control for the receiver and speaker, selectable ringing pitch and ringing volume, flash (to access certain public telephone exchange features), redial, and a function to temporarily switch over of the dialing mode (switch from pulse to tone).

The volume control provides 6 different levels of adjustment to best suit the user.

DTR-1HM-1 Enhanced Single Line Telephone - P/N 780025 & 780026

This enhanced version of an analog telephone is similar to the DTR-1-1 Single Line Telephone, but provides some additional feature. The telephone is available in black or white.

This telephone provides:

- MF (DTMF)/DP switch
- Volume control for the receiver and speaker (providing 6 different levels of adjustment to best suit the user)
- Selectable ringing pitch and ringing volume
- Flash (to access certain public telephone exchange features)
- Redial
- Function to temporarily switch over of the dialing mode (switch from pulse to tone)
- 8 Speed Dial buttons for one-touch calling
- Monitor (to make and receive calls without picking up the handset)
- Hold

Keyset Adapters

The UX5000 digital keysets and IP terminals provide 3 types of optional interfaces/adapters: handset units, adapters for the bottom of the terminals, and line key kits.



Each UX5000 keyset/terminal may have multiple optional adapters installed, depending on the type of keyset/terminal and the adapter used. These adapters provide the keyset different capabilities, depending on the adapters installed.

Only the ADA and PSA adapters can be used on the IP phones.

- 16-Button DLS (16LK)
- ADA - Call Recording
- APR - Analog Port Adapter with Ringer
- BCH - Bluetooth Cordless Handset
- BHA - Bluetooth Hub Adapter
- PSA - Keyset/IP Terminal Power Failure

If Aspire keysets are connected to the UX5000 system, then these phones will support the use of the CTA (RS-232 Serial Interface) and CTU (USB) adapters as well. However, only for reports and SMDR - TAPI is not available with these units on the UX5000 system.

When installing or removing the adapters, ***the keyset should first be unplugged from the system***.

You can install one of these. . .

Handset Options - *Handset Device*

- *Standard Handset with PSA Built-In Cradle for Power Failure (P/N 0910088 & 0910090)*
  Transfers an IP terminal to a pre-connected outside trunk when commercial power is down or the user manually activates the switch. (No AC adapter is required)

- *BCH Bluetooth Handset Adapter (P/N 0910092)*
  Consists of: Bluetooth Cordless Handset (BTH) with Cordless Handset RF Cradle (BTC)
  The handset and cradle provide Bluetooth cordless handset operation (range: 50 meters). Both the Handset Option connection and the Bottom of the Terminal connection are used (preventing any voice/data devices from being installed). The handset provides a 20-character/2-line display, 8 Programmable Function Keys, a Navigator Key and Dial Keys. The handset will charge while in the cradle.

Bottom of the Terminal Adapters - *Voice/Data Device*

- *BHA Bluetooth Hub Adapter (P/N 0910086)*
  This adapter provides interface for a user-provided Bluetooth cordless headset.

- *ADA Call Recording Adapter (P/N 0910084)*
  Using the ADA Adapter provides connections for audio input and output. Connections are stereo mini audio jacks. This allows for a connection from a terminal to an external tape recorder or speaker. (No AC adapter is required)

- *APR Analog Port Adapter with Ringing (P/N 0910082)*
  The APR provides an analog interface for a keyset. The APR provides ringing which allows the connected device to be used for incoming and outgoing calls. It can use the same extension number as the keyset (B1 channel) or it can have its own extension number (B2 channel). The APR does not support reverse polarity, message waiting lamping, or Caller ID. (No AC adapter is required)
  The APR for the B1 channel consumes no ports, however, the B2 channel consumes 1 port, ranging from 512 to 1 in descending order.

Side Connector - *Dedicated Data Devices*

- *16-Button DLS (P/N 0910098 & 0910100)*
  Refer to the "16-Button DLS Console" on **16-Button DLS – 0910098 & 0910100** (page 25) for more details.

- 60-Button DSS (P/N 0910098 & 0910100)
  Refer to the "60-Button DSS Console" on **60-Button DSS Console – P/N 0910094 & 0910096** (page 26) for more details.

The following chart indicates the compatibility of the adapters with the different models of terminals. Select the terminal in the column and then select the adapter in the row to see if the adapter will work with the desired terminal.

Option Compatibility							
Options		Terminals:					
		IP			Digital		
		IP-CTS	Enhanced	Value	Enhanced	Value
Common	Retro Ten Key Dial Pad Kit	✔	✔	✔	✔	✔
	12 Line Key Kit	NA (Built in)	✔	NA	✔	NA
	16-Button DLS	✔	✔	NA	✔	NA
	BCH: Bluetooth Cordless Handset with Hub (Class 1)	✔	✔	NA	✔	NA
	BHA: Bluetooth HUB Adapter (Class 2)	✔	✔	NA	✔	NA
	ADA: Analog Recording Adapter	✔	✔	NA	✔	NA
	PSA: PSTN Adapter for Analog	✔	✔	NA	✔	NA
	DSS: 60-Button DSS Console	✔ (connects to side option slot)	✔ (connects to side option slot)	NA	✔ (connects to system chassis)	✔ (connects to system chassis)
Digital	APR: Analog Port Adapter with Ringer	NA	NA	NA	✔	NA
	DESI-Less Line Key/LCD Unit (8LK)	NA	NA	NA	✔	NA
	Backlit LCD	NA	NA	NA	✔	NA
IP	DESI-Less Line Key/LCD Unit (8LK)	NA (Built in)	✔	NA	NA	NA

When installing the adapters, ***the keyset should first be unplugged from the system***. Also note that the adapters may have an AC/DC power jack. Power is not required for all the adapters. You should refer to the information for the specific adapter to determine whether a power source is needed.

When wall-mounting terminals with ADA, APR, BHA adapters, an additional wall-mount bracket is required to accommodate these items. When using a 60-Button DSS Consoles with a terminal which has one of these adapters installed, the console also requires a special wall-mount bracket.

Optional Equipment

2PGDAD Module — P/N 0891027

The 2PGDAD module provides two circuits which allow connection to external terminals such as:

- Door Box
  (8 max. per system)
- External Speaker with Amplifier
  (8 maximum with 2PGDAD modules, 1 on the CCPU)
- External Music Source (external MOH)
  (96 maximum per system)
- External Recording System
  (96 maximum per system)
- External Ringing

The UX5000 supports a total of 368 2PGDAD modules on a non-networked system or 512 with a networked system. These modules can be used within the feature maximums indicated above.

The 2PGDAD module also provides multi-purpose controls. These control relays can be used for controlling the external amplifier, external music source and door lock control with the use of a Door Box. The UX5000 system allows for up 8 general purpose relays with the 2PGDAD modules (4 relays on each 2PGDAD) and 1 on the CCPU for a maximum of 9.

The 2PGDAD module connects to any available digital extension port. The terminal connections made within the PGDAD module and the jumper settings determine what features are used for each circuit.

16-Button DLS – 0910098 & 0910100

This optional unit connects to the side connector to the right of the terminal. This unit provides 16 additional keys which gives a keyset user a Busy Lamp Field (BLF) and one-button access to extensions, trunks and system features. Digital keysets with a 16-Button DLS installed can also have a 60-Button DSS Console connected. However, IP terminals can have either a 16-Button DLS or a 60-Button DSS Console - it cannot have both units installed.

Once defined in system programming, these keys are used the same as the line, loop, or Programmable Function keys on the digital keyset or IP terminal.

This module is connected to the keyset/terminal using a special cable. This cable is provided with the module.

Keep the following in mind when installing 16-Button DLS units or DSS Consoles:

- A 16-Button DLS does ***not*** require a separate digital station port.
- ***An IP terminal*** can have **only** a 16-Button DLS OR a 60-Button DSS Console - it cannot have both items attached to the terminal.
  A digital keyset can have a 16-Button DLS installed as well as a 60-Button DSS Console. The 60-Button DSS Console is installed to the right of the 16-Button DLS.

Option Compatibility				
IP Terminal			**Digital Keyset**	
IP-CTS	**Enhanced**	**Value**	**Enhanced**	**Value**
✔	✔	NA	✔	NA

60-Button DSS Console – P/N 0910094 & 0910096

The DSS Console gives a keyset user a Busy Lamp Field (BLF) and one-button access to extensions, trunks and system features. The 60-Button DSS Console provides an additional 60 programmable keys, while the 16-Button DLS provides 16 programmable keys. Keep the following in mind when installing DSS Consoles:

- A 60-Button DSS Console requires a separate digital station port when pairing with a digital keyset. For IP terminals, the console is connected to the side option slot using a special cable which is included with the DSS Console. Using the DSS Console on an IP terminal also requires an AC power source (AC-2R, P/N 780135).
- ***An IP terminal*** can have **only** a 16-Button DLS OR one 60-Button DSS Console - it cannot have both items attached to the terminal. (However, multiple 60-Button DSS Consoles using digital ports can be assigned to an IP terminal.)
  A digital keyset can have a 16-Button DLS installed as well as up to 32 60-Button DSS Consoles assigned. The 60-Button DSS Console is installed to the right of the 16-Button DLS.
- A 60-Button DSS Console attached to an IP terminal can provide up to 113 keys. A Page Switch key (Program 30-03-01, code 95) must be defined on the bottom row of DSS keys (keys 55-60). This will allow access to the additional keys. However, note that on the second level of DSS keys, the bottom row (keys 55-60) will remain on the first level of keys.

Option Compatibility				
IP Terminal			**Digital Keyset**	
IP-CTS	**Enhanced**	**Value**	**Enhanced**	**Value**
✔	✔	NA	✔	✔

AC Adapter (AC-2R) — P/N 780135

The AC Adapter is required for the IP adapter or Aspire IPhone if external power is needed. Also required for APR, CTU, and Speakerphone optional keyset adapters.

Audio Emcee Kit — P/N 750316

The Audio Emcee message on hold system provides the ability to use a CD-based on-hold messaging system connected to the UX5000 through the music on hold (MOH) port. When business calls are placed on hold, callers will hear advertisements that can help enhance the company image, cross sell products and services and reduce caller hangs-ups by keeping callers entertained and informed. Callers will hear programmed selections and this cycle will repeat until the call is answered.

The kit includes:
- MOH Unit
- Infrared Remote
- RCA Connecting Cable
- Starter CD with 24 Message/Music Tracks
  (8 music tracks, 8 courtesy tracks, 8 holiday tracks)
- Certificate Good for 8 Customized Messages
- Operating Guide
- 1 Year License

Conference Max — P/N 750073

The Conference Max provides an expandable tabletop conferencing solution with premium, full-duplex audio. A single unit can be used in small conference rooms, or for larger rooms, link up to 4 phone units for unrivaled microphone and loudspeaker coverage. The unit offers:
- Distributed Echo Cancellation® effectively eliminates echo
- Noise cancellation removes background noises from fans or HVAC systems
- Full-duplex sound enabling participants to speak and listen at the same time without cutting in and out
- Automatic level controls to keep participants' audio balanced and consistent
- First-mic priority eliminating hollow "tunnel" sound by activating only the microphone closest to the person speaking
- 3 microphones providing 360º audio pickup
- Stand-alone conference phone solution or link up to 4 complete phones - not just microphones
- Simple installation
- 10 Speed Dial numbers
- Recording output on base unit can connect to a recording device to capture both sides of the conversation

Conference Max Plus — P/N 750074

Similar to the Conference Max above, except the Conference Max Plus is a wireless unit. You are able to have up to 15 working units within a range of 150'. The recommended minimum distance between units is 1'.

This 2.4 GHz unit provides 12 hours of talk time and 36 hours of stand-by time.

Desktop Suite

The Desktop Suite is an integral part of the overall workstation. It is the combination of three complimentary software applications designed to help users become more mobile, productive and better informed. Components of the Desktop Suite are:

- PC Attendant - Significantly improves call management by enabling an attendant to easily perform call handling capabilities right from their PC.
- PC Assistant - Provides management and operation of a desktop terminal from a PC - for easy dialing, call management, contact lookup, and seamless CRM integration.
- Softphone - The UX Softphone application provides convenient, cost-effective mobility. A computer becomes a phone and all features of the office terminal are available with the click of a mouse. Add a web cam to deliver video between another camera-equipped Softphone.

These applications use a CTI (Computer Telephony Integration) service that is provided with the product in order to integrate with the UX5000 communications server. Through the CTI service, an ethernet connection over the company LAN, and a device for audio (UX000 terminal, USB handset/headset, PC microphone) for audio, the application is able to control the actions of the phone, including placing calls.

Multiple clients within a company can use the different applications (based on the application's license) providing the ease-of-use the application(s) provide to anyone on the network.

The host PC to be used as the CTI Server requires Windows XP, Windows 2000, or Windows Vista and an interface to the UX5000 through the 3rd-party CTI link to monitor and control the telephone activity.

Door Box — P/N 922450

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. The Door Box is weather-tight, but where possible, should have some coverage from the weather. A 2PGDAD Module (P/N 0891027) is required for this feature.

Each 2PGDAD module audio output can optionally support two analog Door Boxes. In addition, you can connect each circuit's control relay to an electric door strike. This allows an extension user to remotely activate the door strike while talking to a visitor at the Door Box. The control relays are normally open. The CCPU also provides 1 relay. This relay is defined as relay '0' in programming. The relays on the 2PGDAD modules are numbered 1-8. The system can have up to eight Door Boxes.

Headsets

A keyset user can utilize a customer-provided headset in place of the handset. Like using Handsfree, using the headset frees up the user's hands for other work. However, Headset Operation provides privacy not available from Handsfree.

An extension with a headset has two options for when it appears busy to incoming callers. The headset extension can be:

- Busy to incoming callers when only one extension appearance is busy (i.e., Off-Hook Signaling prevented)
  OR
- Busy to incoming callers only when both extension appearances are busy (i.e., Off Hook Signaling allowed)

As the headset plugs into a separate jack on the bottom of the terminal, the handset can still be connected to the terminal. This provides you with the option to use the handset, headset or the speakerphone for calls. Depending on the method used to activate the headset mode, you can either use the headset on a call-by-call basis or any calls placed/answered will automatically be in headset mode:

- Call-by-Call Basis: Press only the Programmable Function Key to answer/disconnect each call [defined in Program 15-07-01, code 05])

  With this option, the handset and speakerphone can also be used at any time to answer/place calls.

- Automatic Headset Mode: Enable Headset mode by entering the Headset service code defined in (11-11-65).

  With this option, the handset can also be used at any time to answer/place calls. The speakerphone will not function except for handsfree answerback on incoming intercom calls.

2

All Plantronics Polaris headsets should be compatible. Examples are:

Plantronics Polaris Headsets:	NEC Part Number
Polaris SupraPlus/NC-M (monaural with noise canceling transmitter)	750643
Polaris SupraPlus/NC-B (binaural with noise canceling transmitter)	750645
Polaris SupraPlus/VT-M (monaural with voice tube transmitter)	750644
Polaris Encore/VT-M (monaural with voice tube transmitter)	750634
Polaris Encore/NC-B (binaural with noise canceling transmitter)	750635
Polaris Tristar/VT-M (monaural with noise canceling transmitter)	750630
Polaris Mirage/VT-M (monaural with voice tube transmitter)	750631

Headset Operation for SLT Headset Operation

The ability for single line terminal users to use the Headset feature is available. When a single line terminal with a headset receives an incoming call, the UX5000 can let the SLT user know of the incoming call by a notification tone in their headset.

Notes:

- This feature is only applicable when the SLT is set in headset mode and the UX5000 allows the headset ringing with their class of service.
- If a SLT with headset ringing condition goes on hook, the SLT will ring normally.
- A SLT user with a headset should not be allowed to automatically seize a trunk (Line Preference) on an off-hook condition as the terminal would never go idle.
- If a SLT is set for the Ringdown feature, the headset mode is NOT available. But, if the Ringdown call start time is equal to or more than the headset ringing start time, the Headset feature, and not the Ringdown feature, is available.
- Caller ID should be disabled for the SLT when then Headset feature is to be used (Program 15-03-09 and 15-03-10 set to "0"). Otherwise, the terminal will only beep twice with an incoming call. In addition, the terminal will not receive the Caller ID information when the Headset feature is used.

CT-12 Cordless Headset Telephone (P/N 730094)

The CT-12 is a 2.4GHz cordless headset which connects to an analog port or an analog telephone line as a stand-alone unit or to an analog port adapter (APR, P/N 0910082). When the APR is set up as the same extension of the telephone, you can use the headset to answer and make calls using the cordless headset. The CT-12 offers Caller ID, but only if it is connected to an analog port on an analog station card. The CT-12 will not receive Caller ID if it is connected to an APR adapter (this adapter does not output Caller ID).

The number of units which can be used on the system is greatly affected by the environment. The closer or smaller the area, the smaller the number of units which can be used. It is recommended to start with 3 or less. If there are no conflicts between the telephones, you can try adding additional units (up to 5 would be the recommended maximum).

When using wireless LAN, keep in mind that although there should not be a problem with interference from WLAN's, 802.11b and 802.11g both share the same frequency as the CT-12 telephone. In theory, the CT-12 is a narrow band high power device where as the 802.11b and 802.11g are both wide band low power technologies. Therefore, the higher power CT-12 could disrupt the low power device and slow the data network. There are, however, many exceptions to this (for example, if the WLAN uses highly directional antennas, higher power relays between buildings, etc.). The CT-12 can not lock down channels, unlike the 802.11b and 802.11g.

The CT-12 features include:

- 2.4 GHz Cordless Headset Phone
- Range of Up to 150'
- 6 Hours of Talk Time, 80 Hours Standby Time
- Audible Low Battery Indicator
- Single Line Operation
- Ultra-Compact Remote Unit with Belt Clip
- Variable Range Volume Control
- 10 Speed Dial Numbers
- Page/Find Feature
- Redial/Flash
- Mute with Audible Reminder
- Talk/Charge/Power Indicator Lights
- Built-in Headset Stand

Dterm Headset Cordless II (P/N 730091)

The Dterm Headset Cordless II is a cordless headset telephone. It is basically a single-line telephone, but it has no dial pad, so it does not provide all of the functions of a single-line telephone. As such, it generally would not be connected directly to a telephone line or an analog station port. It is specifically designed to interface with UX5000 terminals which have an APR Adapter connected.

The number of units which can be used on the system can range from 15-50. The maximum number is greatly affected by the environment. The closer or smaller the area, the smaller the number of units which can be used. The Dterm Headset Cordless II uses frequency hopping to automatically change channels if interference is present. If the unit detects wideband interference (multiple users in a small area), the range will be decreased until an adequate link is formed. The present of other 900 MHz devices will cause interference and may need to be removed.

The Dterm Headset Cordless II features include:

- 8 hours of talk time
- 902-928MHz DECT Wireless Technology
- Up to 300' roaming distance from an office phone
- Can be converted from "over-the-head' style to "over-the-ear' style
- The base station is equipped with:
  - 4 audio adjustments
  - 3 status LEDs (Power, Talk, and Charging)
  - Handset Lifter switch (not used on the NEC version)
  - Charging cradle for the headset
  - AC adapter
  - 1 connecting cord
  - 1 DTR-1C-2 switch cord

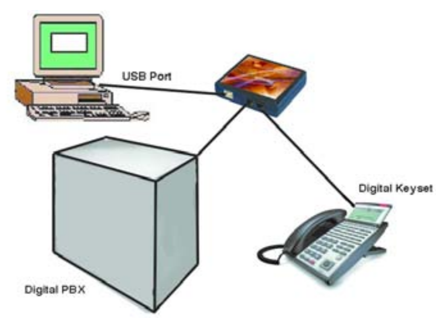
- The headset is equipped with:
  - Noise canceling microphone
  - Talk button (answer or disconnect a call)
  - Talk indicator LED
  - Listening volume control/mute switch
  - 2 styles of headbands with 2 types of foam ear rests
  - 3 different size ear loops

SLT Adapter — P/N 0891026

The SLT Adapter converts a digital port from an ESIU PCB into an analog port which can be used for connecting on-premise 2500 type single line devices (i.e., telephones, fax machines, modems, etc.). Caller ID is supported by these adapters. The SLT Adapter provides the ring generator circuit used by the analog device. The unit provides constant current which is fixed at 47 mA. Each SLT Adapter requires its own digital port.

Refer to *Single Line Telephones* in the Software Manual for more details.

VSR Adapter — P/N 780275



The Dterm® Voice Security Recorder (P/N 780275) is a USB device that taps across the digital extension pair of the NEC communications server allowing digital recording of the keyset user's conversation. The file created is saved either to the local PC or to a network location, depending on the application's setup. This adapter is for use with digital keysets. It cannot be used with analog or VoIP phones.

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.

PC Compatibility

The Dterm® Voice Security Recorder application supports Microsoft operating systems which support USB devices such as Windows 98SE, Windows ME, Windows 2000, and Windows XP. Note that Windows 95 and below, Windows NT and Macintosh operating systems are not supported.

4-Port Digital Call Logging Unit — P/N 780273

The 4-Port Digital Call Logging Unit is similar to the function of the Dterm® Voice Security Recorder (P/N 780275), however, this unit connects up to 4 UX5000 digital terminals. The system connection for these terminals is made at the MDF with the station cables, while the PC connection is USB. A maximum of 48 stations/12 USB connections per PC is permitted.

This device, as well as the single-port VSR, use the VSR Manager Software (P/N 780274) to manage multiple units, as well as the recordings.

UX5000 Common Equipment

9.5” 3-Blade Base Chassis — P/N 0910000

The Base chassis is the UX5000’s control center. It houses the Power Supply, has 3 blade slots, an expansion blade slot and provides for connection to trunks and extensions. Also included with this chassis are 2 blank blade slot covers, 1 expansion slot cover, and brackets for wall-mounting/desk-mounting the chassis (used for only the Base or the Base and Expansion chassis combined). The first slot in the Controlling Base Chassis is dedicated to the CCPU. The next two slots are universal. They can be used for any combination of Common, Trunk or Station blades. With any additional chassis connections (up to 3 expanded chassis), all slots can be universal slots. The chassis can be desk, wall or rack mounted.

9.5” 3-Blade Expansion Chassis — P/N 0910002

The Expansion chassis is used to provide additional slots for the Base chassis, expanding the capabilities of the UX5000. It provides an expansion blade slot, 3 blank blade slot covers, and 3 universal blade slots. These can be used for any combination of Common, Trunk or Station blades. The chassis can be floor, wall or rack mounted when combined with the Base chassis.

19” 6-Blade 2U Chassis — P/N 0910004

The 6-blade 2U chassis is the UX5000’s control center. It houses the Power Supply, has 6 blade slots, an expansion blade slot and provides for connection to trunks and extensions. Also included with this chassis are 5 blank blade slot covers and 1 expansion slot cover.

The first slot in the Controlling Base Chassis is dedicated to the CCPU. The next five slots are universal. They can be used for any combination of Common, Trunk or Station blades. With any additional chassis connections (up to 3 expanded chassis), all slots can be universal slots. The chassis can be floor, wall or rack mounted.

The power supply and fan in this chassis are replaceable (power supply = P/N 670018, fan = P/N 670507).

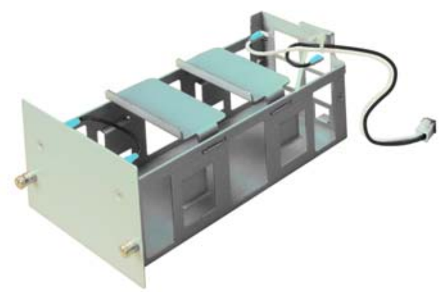
Short-Term Battery Box for 9.5” 3-Slot Chassis — P/N 0910006

If the power fails, connecting the power supply unit from each 9.5” base chassis to a battery box will allow the UX5000 to continue to function.

The short-term battery box will power the UX5000 for approximately 10 minutes.

This Battery Box requires 2 12V 0.8Ah batteries (P/N 670511).

Short-Term Battery Box Kit for 19” 6-Slot Chassis — P/N 670601



This short-term battery box kit is used for the 19” 2U chassis. Each 19” 6-blade chassis would have its own battery box installed.

If the power fails, connecting the power supply unit from the 9.5” base chassis to a battery box will allow the UX5000 to continue to function.

The short-term battery box will allow the UX5000 to continue to function for approximately 10 minutes.

This kit includes the Battery Mount (P/N 670509), cable (P/N 670530), and 2 12V 0.8Ah batteries (P/N 670533).

Long-Term Battery Box for 9.5” 3-Slot or 19” 6-Slot Chassis — P/N 0910007

This long-term battery box is installed adjacent to the 9.5” 3-Blade Controlling Base Chassis or beneath the 19” 6-Blade Chassis. One Long-Term Battery Box can be connected to each chassis (for up to 3 hours of back-up time) or one Battery Box can be connected for all 4 chassis (for up to 45 minutes of back-up time).

This unit does not provide surge protection or power conditioning as the UX5000 chassis still plugs into an AC outlet (the battery box is connected to a dedicated connector on the 19” chassis).

This Battery Box requires 3 large Battery Sets (P/N 670512) containing the 12V 0.8Ah batteries.

19” Rack-Mount Bracket Set for 9.5” 3-Blade Chassis — P/N 0910008

This Rack-Mount Bracket is required to install the 9.5” 3-Blade Chassis to a 19” rack mounting system.

19” Rack-Mount Bracket Set for 19” 6-Blade Chassis — P/N 670508

The 19” Rack-Mount Bracket is required to install the 19” 6-Blade Chassis to a 19” rack mounting system.

19” Wall-Mount Bracket Set for 19” 6-Blade Chassis — P/N 670510

Wall-mounting the 19” 6-Blade Chassis can be accomplished using this kit, which provides the required brackets.

19” Floor-Mount Bracket Set for a Combined Base/Expansion 9.5” 3-Blade Chassis or 19” 6-Blade Chassis — P/N 670019

Using this Floor-Mounting Bracket for either a combined 9.5” Base/Expansion chassis or a 19” chassis will allow for a secure installation when the chassis are mounted to the floor.

Note that a single 9.5” Base chassis should not be floor-mounted.

Central Processing Unit (CCPU) Blade — P/N 0911001

The CCPU controls all the functions and operations of the UX5000 system using the system software loaded into the CCPU memory. The system software can be upgraded as new software becomes available.

One CCPU blade must be installed in the CCPU slot in the Controlling chassis. In a networked system using CygniLink and an IP connection, a second CCPU can be used for a CCPU Fail Over feature (primary/secondary CPUs) as a backup in case of a hardware or power failure. (The CCPUs must be programmed for primary/secondary operation.)

The CCPU functions are:

- Music on Hold tone Circuit, External Source Control circuit
- Voice Mail Daughter Board Interface Circuit
- System Program and System Data Storing Memory Circuit
- USB Interface Circuit
- Ethernet Interface Circuit
- EXIFU-A1 Interface Circuit
- Main Processing 32-Bit CPU Circuit (MPC8248 @ 266 MHz)
- Time Switch, Optional Blade Control Circuit
- Backboard Interface Circuit

The CCPU without a MEMDB attached supports:

- 64 Ports for IP/Digital Trunks/ Extensions
- Expansion Chassis
- VOIPDB
- 8 IntraMail/VRS Ports
- BRI S-Bus/T-Bus
- PRI Trunks
- T1 Trunks
- Tie Line Trunks
- (1) 082U Blade
- ACD/inDepth
- Third-Party CTI/TAPI 2

The CCPU with a MEMDB attached and limited port licensing supports:

- 256 Ports for IP/Digital Trunks (200 max.) / Extensions (256 max.)
- Expansion Chassis
- VOIPDB
- 16 IntraMail/VRS Ports
- BRI S-Bus/T-Bus
- PRI Trunks
- T1 Trunks
- Tie Line Trunks
- Multiple 082U Blades (1 per chassis max.)
- ACD/inDepth
- Networking (CygniLink and AspireNet
- Recovery Point
- Remote Maintenance
- Third-Party CTI/TAPI 2

The CCPU with a MEMDB attached and full port licensing supports:

- 712 Ports for IP/Digital Trunks (200 max.) / Extensions (512 max.)
- Expansion Chassis
- VOIPDB
- 16 IntraMail/VRS Ports
- BRI S-Bus/T-Bus
- PRI Trunks
- T1 Trunks
- Tie Line Trunks
- Multiple 082U Blades (1 per chassis max.)
- ACD/inDepth
- Networking (CygniLink and AspireNet
- Recovery Point
- Remote Maintenance
- Third-Party CTI/TAPI 2

2

The CCPU provides:

- 200 trunk ports maximum (with MEMDB installed and licensed expansion)
- 512 extension ports maximum (with MEMDB installed and licensed expansion)
  - 512 ports digital/IP extensions maximum
  - 320 analog ports maximum
- 256 virtual extensions
- Connection for 32/64/128 VoIP Daughter Board
- Connection for Voice Mail Daughter Board
- Connection for Expanded Memory (MEMDB-A1)
- Supports TAPI 1.x
- 1 Green Status LED and 4 Red Status LEDs

  During normal operation, the "RUN" LED will be flashing and the remaining LEDs will be off.
- Tone Generator
- Tone Processing DSP
- Connection for Memory Module
- DTMF Tone Sender
- DTMF Tone Receiver
- System Tone Sender
- MFC Tone Sender
- MF Signal Sender (Sends caller information to CO for E911)
- Call Progress Tone Detection
- C-Channel Control
- Conference: 64 Channels (32 channels x 2 circuits)
- Caller ID Receiver
- Caller ID Generator

- A load switch which is used for initial system startup, resetting the system, or when upgrading system software
- One USB Port - USB 1.1/2.0
- One Gbit Ethernet Port for VOIPDB
- One Fast Ethernet Port (10/100 Base-T), auto negotiation and VLAN
- One CompactFlash Blade Slot
- Background Music/EXSP Control Port
- Status LED
- One EXIFU Interface Connector
- Two Audio Input/Output Terminals (sub-mini jacks)
- One General Purpose Control Terminal
- Call Control Server (ex: Conference Bridge Server, Voice Mail Server, SIP Server, RTP Forwarding, VoCoder Conversion)
- One lithium battery (Sony CR2032 or equivalent) which provides battery back-up of system data

The CCPU functions provided are:

- Call Control Server
- Conference Bridge Server
- Voice Mail Server (voice mail requires a compact flash card)
- SIP Server
- RTP Forwarding
- VoCoder Conversion

! IMPORTANT!
After removing a previously installed CCPU, handle the blade, carefully, from the edges. If certain solder points/resistors are touched on the back of the blade, some RAM/temporary memory may be lost (ex: time, date, user-defined settings, etc.)

Memory Expansion Daughter Board (MEMDB) — P/N 0911060

The Memory Expansion daughter board (MEMDB) provides additional memory for use with increased ports when using an expanded license, expanded system networking, remote software updates. This daughter board is also required if you wish to install more than one 082U blade into the communications server. It also provides expanded port capacity for the system. This daughter board is mounted on the CCPU and provides:

Description	Memory Capacity	Equipped Memory
SDRAM	128 MB	256 MB / 16 bit x 4 pcs
Flash Memory	32 MB	256 MB / 16 bit x 1 pc
SRAM	1 MB	4 MB / 16 bit x 2 pcs

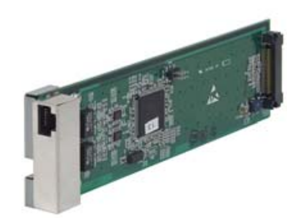
System	Maximum Ports
UX5000 - **No** MEMDB installed ● No recovery point created for database ● No remote update from flash memory, but possible when USB flash drive is installed ● 8 ports maximum for IntraMail/VRS ● One 082U Blade per system maximum	64 (Trunks or Extensions)
UX5000 - MEMDB installed and limited port license ● Networking (CygniLink and AspireNet) ● No remote update from flash memory, but possible when USB flash drive is installed ● 16 ports maximum for IntraMail/VRS ● Up to 3 recovery dates can be created ● Multiple 082U Blades (1 per chassis max.)	256 (Trunks: 200 max. Extensions: 256 max.)
UX5000 - MEMDB installed and maximum port license) ● Networking (CygniLink and AspireNet) ● No remote update from flash memory, but possible when USB flash drive is installed ● 16 ports maximum for IntraMail/VRS ● Up to 3 recovery dates can be created ● Multiple 082U Blades (1 per chassis max.)	712 (Trunks: 200 max. Extensions: 512 max.)

! IMPORTANT!
When installing a MEMDB into a chassis which has been previously installed and programmed, the CCPU must be cold-started due to database compatibility issues.

Expansion Blade for Base Chassis (EXIFU-B1) — P/N 0911020

and

Expansion Blade for Expansion Chassis (EXIFU-E1) — P/N 0911022



The EXIFU blade provides a connection from the Controlling Chassis-B to the Base Chassis-B expansion unit. This connection is required with any multiple-chassis setup. This blade allows the CCPU to transmit/receive data as required to the additional chassis.

The EXIFU-**B1** blade is installed in the EXIFU slot of the Controlling Chassis-B which is equipped with a CCPU blade. The EXIFU-**E1** blade is installed in the EXIFU slot of the Base Chassis-B, which does not have a CCPU.

The EXIFU cable is used to connect the Controlling chassis and its EXIFU-**B1** interface to the second, third, and fourth Base chassis EXIFU-**E1** interface.

Use only the CAT 5 cables provided by NEC to make the connections between the Base chassis and Expansion chassis.

The EXIFU provides:

- Communication Processor Interface for data handling through Cch (18 slots max.)
- 64 Channels for Telephony Resource (ex: DTMF Tone Receiver, Call Progress Tone Detector, MFC Tone Receiver, Caller ID Receiver, Caller ID Signal Sender)
- DSP Resource Management

VoIP Resource Daughter Board (VOIPDB) — 32VOIPDB: P/N 0911030, 64VOIPDB: P/N 0911032, 128VOIPDB: P/N 0911034

The 32VOIPDB, 64VOIPDB and 128VOIPDB daughter boards are used for converting the RTP (Real Time Transfer Protocol) packets via the IP network and PCM highway. The daughter board is installed on the CCPU. The IP terminals are connected directly to the IP bus. When IP terminals need to be connected to a conventional PCM-based digital circuit, this daughter board converts the IP packet signal into a PCM signal format and connects to the PCM time division switch.

The VOIPDB daughter board is required in order for IP terminals to communicate with non-VoIP UX5000 terminals, as well as to place or receive outside calls.

The VoIP daughter board provides the voice (RTP/RTCP) processing function. The call control function is mounted on the CCPU. Only one version of the VOIPDB (32, 64, or 128) can be installed on the CCPU at a time.

The VOIPDB daughter board provides:

- 32 (32VOIPDB), 64 (64VOIPDB) or 128 (128VOIPDB) channels
- **64VOIPDB and 128VOIPDB Only:** Layer 2 Switch
- **64VOIPDB and 128VOIPDB Only:** 10/100/1000 Gigabit Ethernet Connection
- **64VOIPDB and 128VOIPDB Only:** RTP/RTCP Packet Transmitted/Received Directly
  32VOIPDB: RTP and RTCP Packeted Transmitted/Received by Fast Ethernet LAN Interface on CCPU
- **64VOIPDB and 128VOIPDB Only:** 4 Blade Status LEDs
- Support for CODECs: G722, G726, iLBC

When installing a VoIP daughter board, the system will not allocate trunk ports for the daughter board. This must be defined in Program 10-40-02. Ports are allocated in groups of 4. For IP extensions, the system will only allocate extension ports once an IP terminal is connected. At that time, the system will reserve a group of 4 extension ports. Once the 4 ports are used, the system will then reserve another group of 4 when another IP extension is connected.

UX5000 Trunk Blades

4-Port CO Loop/Ground Start Trunk Blade (4COIU-LG1) — P/N 0911072

The 4COIU is used to provide 4 ports for loop/ground start trunks. The blade can accept a analog trunk daughter board (4COIDB) to provide an additional 4 ports.

The COIU blade provides:

- 4 analog loop start/ground start trunk circuits
- 1 trunk status LED
- 1 blade status LED
- 4 Caller ID Circuits
- 2 Power Failure Transfer Circuits
- Connection for COIDB Daughter Board

The 4COIU consumes 4 trunk ports ranging between ports 001-200. The CN2 connector provides connection to 4 analog trunk ports. The ground start ports ***are polarity sensitive (tip to tip, ring to ring)*** - the loop start trunks are not. The power failure circuits (CN3), however, are not polarity sensitive.

4-Port CO Loop/Ground Start Trunk Daughter Board (4COIDB-LG1) — P/N 0911074

The 4COIDB is used to provide 4 additional ports for loop/ground start trunks. The blade is connected to a 4COIU blade to provide a total of 8 ports for the combined blade.

This daughter board can also be installed on the Digital/SLT Combination blade (082U). This combined blade will provide 8 digital extension ports, 2 analog extension ports, and 4 trunk ports.

The analog trunk daughter board provides:

- 4 analog loop start/ground start trunk circuits
- 4 Caller ID circuits
- Connector for COIU or 082U blade

The 4COIDB consumes 4 trunk ports ranging between ports 001-200. The CN2 connector provides connection to 4 analog trunk ports. The ground start ports ***are polarity sensitive (tip to tip, ring to ring)*** - the loop start trunks are not. The power failure circuits (CN3), however, are not polarity sensitive.

T1/PRI Interface Blade (1PRIU) — P/N 0911052

The T1/PRI blade provides an interface for DS1, T1 and ISDN Primary Rate Interface (PRI) applications. This blade has a single 24-channel 64Kb/s digital signal circuit which can be configured for either T1 trunks or PRI. Each blade connects to the network via an NTI Network Termination.

If set for T1, the T1/PRI blade gives the system 24 trunks in a single universal slot. These trunks can be one of the following:

- Loop Start
- Ground Start
- DID
- E&M Trunks
- ANI/DNIS E&M Trunks

T1 gives the system the advantages of advanced digital trunking as well as conserving universal slots. For example, a system with 12 loop start trunks, two tie lines and six DID trunks would use up five universal slots. With T1 all these trunks would be available in a single universal slot, freeing up four additional universal slots for other uses.

If set for PRI, each T1/PRI blade provides 24 PRI (23 B& 1 D) channels running at 1.544Mbps with 64Kb/s clear channel. The blade supports the following PRI services:

- Basic PRI Call Control (BCC)
- Display of incoming caller's name and number (when allowed by the telco)
- Speech and 3.1 KHz audio

By default, the system programming has an installed T1/PRI blade defined as a PRI blade in system programming. System programming is also used to define whether the connection is T-Point or S-Point.

When installed, the T1/PRI Interface blade uses the first block of 24 consecutive trunk ports. For example, if you have an COIU blade installed for trunks 1-8, the T1/PRI Interface blade will automatically use trunks 9-32. If you have COIU blades installed for trunks 1-8 and 17-24, the T1/PRI blade will use trunks 25-48. The T1/PRI Interface blade cannot use trunks 9-16 (even if available) since they are not part of a consecutive block of 24 trunks. Each T1/PRI blade requires that 24 ports be available in the system, even if not all the ports will be used, otherwise the blade will not function.

A maximum of 2 PRIU blades can be installed in the main chassis with the CCPU. A maximum of 5 PRIU blades (for T-Point or S-Point) can be installed in a 19" chassis with a CCPU. The T1/PRI blade requires one universal slot and provides:

- 1.5M or T1 Function

4-Port DID/OPX (4DIOPU) Blade — P/N 0911054

The 4DIOPU blade supports the analog DID and single line telephone interface functions (such as Off-Premise Extension). The function type is assigned in programming for each port. The circuit types, however, should be grouped together. For example, with 3 DID circuits and 1 OPX circuit, they should be grouped as DID, DID, DID and OPX and not DID, DID, OPX and DID.

The DIOPU blade provides:

- 4 (4DIOPU) DID trunk circuits
- 4 (4DIOPU) DID trunk status LEDs
- 2 blade status LEDs

The DIOPU blade can be installed in any universal slot.

The CN2 connector each provides connection to 4 analog DID trunk ports, ***which are polarity sensitive (tip to tip, ring to ring)***. The OPX circuits, however, are not polarity sensitive. The DIOPU requires one universal slot.

4-Port E&M Tie Line Blade (4TLIU) — P/N 0911056

The 4TLIU Tie Line blade is an out band dial type analog tie line interface blade. The blade supports system connections to either 2-wire (four lead, tip/ring) or 4-wire (eight lead, tip/ring/tip 1/ring 1) E&M signalling tie lines (determined in system programming). System programming is also used to select the connection types with Type 1 or Type V.

Each blade requires one universal slot and provides:

- 4 analog 4-circuit tie line interfaces
- 2 blade status LEDs

2-Channel BRI Blade (2BRIU) — P/N 0911048

The BRI blade provides:

- 2 (2BRIU) 2-Channel Circuits (2B + D) configured as T-Bus or S-Bus
- 64 Kb/s Clear B-Channel and 16 Kb/s D-Channel
- 2 blade status LEDs
- Connector for 2BRIDB

These trunk circuits can be connected to either ISDN trunks or ISDN telephones, depending on the switch setting within system programming. All ISDN telephone circuits (#1-2 and #3-4 [with the BRI daughter board]) are supplied DC power from the UX5000 system.

When used for S-Bus, a maximum of 8 ISDN terminals can be connected to each circuit.

The BRI Interface blade uses a single universal slot. Each blade connects to the network via an NTI Network Termination.

In order to block new calls on the blade, system programming (Program 90-05-01, Menu 3) must be used. This program will prevent new calls from being established on the blade, but it will not terminate any existing calls.

2 Channel BRI Daughter Board (2BRIDB) — P/N 0911050

The BRI daughter board provides 2 BRI circuits. This daughter board is installed on the 2BRIU or 082U blade and provides:

- 2 (2BRIU) 2-Channel Circuits (2B + D) configured as T-Bus or S-Bus
- 64 Kb/s Clear B-Channel and 16 Kb/s D-Channel
- Connection point for 2BRIU or 082U

These trunk circuits can be connected to ISDN trunks or ISDN telephones, depending on the switch setting within system programming. All ISDN telephone circuits (#1-2 [BRI blade] and #3-4 [with the BRI daughter board]) are supplied DC power from the UX5000 system.

In order to block new calls on the blade, system programming (Program 90-05-01, Menu 3) must be used. This program will prevent new calls from being established on the blade, but it will not terminate any existing calls.

UX5000 Station Blades

16-Port Digital Blade (16ESIU) — P/N 0911038

The 16ESIU blade provides:

- 16 digital extension circuits (used for digital telephones, DSS consoles, SLTAD adapters, 2PGDAD adapters)
  These ports provide -48V feeding.
- 2 Blade status LEDs - 1 Live LED, 1 Busy/Idle LED

The ESIU can be installed in any universal slot in the system and up to a maximum of 20 ESIU blades installed per 4-chassis system, providing up to 320 digital ports. A maximum of 16 2B channels per chassis is possible. With CygniLink and a 7-chassis system, the maximum number of blades is 32.

2

Per 19” chassis, there is a maximum of 80 digital or analog station ports allowed.

8-Port Digital Blade (8ESIU) — P/N 0911036

The 8ESIU blade provides:

- 8 digital extension circuits (used for digital terminals, DSS consoles, SLTAD adapters, 2PGDAD adapters)
  These ports provide -48V feeding.
- 2 Blade status LEDs - 1 Live LED, 1 Busy/Idle LED

The ESIU can be installed in any universal slot in the system and up to a maximum of 20 ESIU blades installed per 4-chassis system, providing up to 160 digital ports. A maximum of 16 2B channels per chassis is possible. With CygniLink and a 7-chassis system, the maximum number of blades is 32, providing 256 digital ports. This blade provides compatibility with most Aspire telephones (all digital keysets and IP terminals, as well as CTA and CTU keyset adapters).

Per 19” chassis, there is a maximum of 80 digital or analog station ports allowed.

8-Port Analog Station Blade (8SLIU) — P/N 0911044

The 8SLIU blade provides 8 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems).

The 8SLIU is not rated for OPX use. It is recommended that a 4DIOPU blade be used instead (it supports the analog DID and single line telephone interface functions, such as Off-Premise Extensions).

- 1 Extension Status LED
- 1 Blade Status LED
- Constant Current Type Battery Feeding (25mA / -28Vdc)
- Feeding Polarity Reverse Ability
- Connector for 4/8SLIDB Daughter Boards
- Ring Generator (16Hz/20Hz/25Hz, 75Vrms -28Vdc)
- Caller ID Sending Ability
- Message Wait Lamping Ability (-110Vdc)

Per 19” chassis, there is a maximum of 80 digital or analog station ports allowed.

The SLIU can be installed in any universal slot and up to a maximum of 20 SLIU blades installed per 4-chassis system. With CygniLink and a 7-chassis system, the maximum number of blades is 32.

The SLIU and SLIDB are categorized as TNV2. With this designation, off-premise wiring is not acceptable. Any cabling to the SLIDB blade must be within the building - no outside cabling is permitted.

4-Port Analog Station Blade (4SLIU) — P/N 0911040

The 4SLIU blade provides 4 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems).

The 4SLIU is not rated for OPX use. It is recommended that a 4DIOPU blade be used instead (it supports the analog DID and single line telephone interface functions, such as Off-Premise Extensions).

- 1 Extension Status LED
- 1 Blade Status LED
- Constant Current Type Battery Feeding (25mA / -28Vdc)
- Feeding Polarity Reverse Ability
- Connector for 4/8SLIDB Daughter Boards
- Ring Generator (16Hz/20Hz/25Hz, 75Vrms -28Vdc)
- Caller ID Sending Ability
- Message Wait Lamping Ability (-110Vdc)

Per 19” chassis, there is a maximum of 80 digital or analog station ports allowed.

The SLIU can be installed in any universal slot and up to a maximum of 20 SLIU blades installed per 4-chassis system. With CygniLink and a 7-chassis system, the maximum number of blades is 32.

The SLIU and SLIDB are categorized as TNV2. With this designation, off-premise wiring is not acceptable. Any cabling to the SLIDB blade must be within the building - no outside cabling is permitted.

8-Port Analog Station Expansion Daughter Board (8SLIDB) — P/N 0911046

The 8SLIDB daughter board provides:

- 8 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems)
  The 8SLIDB is not rated for OPX use. It is recommended that a 4DIOPU blade be used instead (it supports the analog DID and single line telephone interface functions, such as Off-Premise Extensions).
- Connector for 4/8SLIU blade
- Caller ID Sending Ability
- Message Wait Lamping Ability (110Vdc)
- Constant current type battery feeding (25mA / -28Vdc)
- Feeding Polarity Reverse Ability

These daughter boards can be installed on the 4SLIU or 8SLIU blades.

The SLIU and SLIDB are categorized as TNV2. With this designation, off-premise wiring is not acceptable. Any cabling to the SLIDB blade must be within the building - no outside cabling is permitted.

The CN2 and CN3 connectors each provide connection to 4 analog station ports and are not polarity sensitive. The SLIDB consumes extension ports ranging between ports 001-256.

4-Port Analog Station Expansion Daughter Board (4SLIDB) — P/N 0911042

The 4SLIDB daughter board provides:

- 4 analog extension ports (used for on-premise analog telephones, fax machines, and analog modems)
  The 4SLIDB is not rated for OPX use. It is recommended that a 4DIOPU blade be used instead (it supports the analog DID and single line telephone interface functions, such as Off-Premise Extensions).
- Connector for 4/8SLIU blade
- Caller ID Sending Ability
- Message Wait Lamping Ability (-110Vdc)
- Constant current type battery feeding (25mA / -28Vdc)
- Feeding Polarity Reverse Ability

These daughter boards can be installed on the 4SLIU or 8SLIU blades.

The SLIU and SLIDB are categorized as TNV2. With this designation, off-premise wiring is not acceptable. Any cabling to the SLIDB blade must be within the building - no outside cabling is permitted.

The CN2 connector provides connection to 4 analog station ports and are not polarity sensitive. The SLIDB consumes extension ports ranging between ports 001-256.

8-Port Digital Station / 2-Port Analog Station "Combo" Blade (082U) — P/N 0911060

The Digital/SLT Combination blade provides 8 digital ports and 2 analog ports. This blade allows for either a 4COIU analog trunk daughter board or 2BRI daughter board to be installed.

The blade is installed in a universal slot and provides:

- 8 digital ports
- 2 analog ports
- 2 status LEDs

Without a MEMDB installed, only 1 082U blade can be installed in the system. With a MEMDB, multiple 082U blades can be installed in the communications server - however, only one 082U blade can be installed in a 9.5" base chassis or 1 per 19" chassis (for a maximum of 4 082U blades with a 4-chassis 19" system). When connecting this blade, the system uses 8 digital port and 2 analog ports.

Conference Bridge

8-Port Multimedia Conference Bridge (PVAU) Blade - P/N 0911070

The PVAU blade is a conference bridge system designed for the UX5000. The blade is installed in the chassis and allows up to 8 parties to take part in a conference call. The blade uses 16 system ports when installed. The participants dial a pre-assigned phone number at the determined time, optionally enter a password, and are prompted to speak their name which will be announced to the other conference participants.

The Multimedia Conference Bridge Application functionalities include:

- **Preset Conference**
  The preset conference configuration is also called ‘always on conference’. There is no stipulated time for these conferences to occur.
  - Number of Preset conference should be determined by the number of hardware resources (PVA ports) that will be used for the conference;
  - These ports will be reserved at all time for preset conference.
  - Preset conference password length may set from 1 ~ 5 digits
  - Password protection for each conference

- **Advanced Mode**
  - Password protection is provided for each conference.
  - Applicable voice messages and announcements (e.g., Entry, password request, exit) are available.
  - Early Entry: When using this option, conference participants are allowed to enter the conference call earlier (by the specified number of minutes) than the scheduled conference time.
  - Email Invitation: When a conference is created with all the details (including the email IDs of the participants of the conference), an email is sent to all the parties who are expected to attend the conference. The email contains the schedule details, the conference bridge number that participants dial to join the conference, the conference pass key, etc. The interface user can also specify a customized message that will be conveyed in the invitation email.
    The Multimedia Conference Bridge Application E-mail configuration supports SMTP mail Server ONLY.
  - Host Required: When enabled, the host/organizer is required to be logged into the conference before any other participant can enter. This option is selectable when setting up new conferences.
  - Admission Control, when enabled, requires the organizer to dial a digit allowing each participant to enter the conference.
  - One customer greeting can be recorded for each Multimedia Conference Bridge Application. Predefined password is necessary to record personal greetings.
  - Password protection option for each conference.
  - Remote conference programming with conference scheduler (via a Web User Interface).
  - Programmable gain adjustments.
  - Support for DTMF detection for manual setup options (Telephone User Interface).
  - HTTP Interface for conference schedule management and conference blade administration.
  - Conference Mode: There are two types of conference mode; Lecturer Mode and Discussion Mode.
  - Lecturer Mode – When the conference starts, all conference participants are placed in mute and remain muted for the duration of the conference. Only the participant, designated as the Lecturer, is not muted.
  - Discussion Mode – All participants can be heard when this mode is selected.

Router / Hub Blades

Router Blade with 8-Port License (RTU-B1) - P/N 0911062

This blade is an in-skin router for the UX5000. Using an RJ45, a connection can be made directly to a PC. This blade combines multiple voice and data features into an in-skin converged networking route. The Router blade includes a single T1 WAN interface and an integrated 4-port managed Ethernet switch with VLAN support. In addition, an 8-port license is provided with this blade to allow for QoS on 8 simultaneous SIP calls. (Any calls over the 8th call will go through, however, it will not have any priority queuing.)

The Router blade provides:

- High Speed Forwarding (Routing). IPsec forwarding ability is about 100 Mbps.
- Supported Interface:
  - T1 WAN Port
  - 10/100 Base-TX Ethernet WAN Port
  - 4 Switched LAN Ethernet Ports (10/100/1000 BaseT) with VLAN and Port Mirroring Capabilities
- VPN/Firewall
- 15 VPN Tunnels
- VoIP Application Layer Gateway (SIP, MGCP and H.323) Resolves NAT/Firewall Traversal Issues
- QoS: Priority Queuing, Traffic Shaping, Diffserv Marking/Policing
- VoIP Call Admission Control
- VoIP Survivability
- Passive Call Quality Monitoring (MOS, Jitter, Latency and More)
- DHCP Server
- NAT/PAT
- Management: HTTP, HTTPS, SSH, Telnet, SNMPv1 and V3
- Ethernet/LAN: Auto-Sensing, Full or Half Duplex

The VOIPDB daughter board, which is required in order for IP telephones to communicate with non-VoIP UX5000 terminals, as well as to place or receive outside calls, must be connected to either a switching hub or to the Router blade.

8-Port Switching Hub Base Blade with PoE (GSWU-B1) - P/N 0911066

The GSWU blade is a managed 8-port gigabit ethernet PoE switch. The GSWU blade provides:

- 8 Gigabit Ethernet (10/100/1000) Ports
- Per Port Status LED Indicating Link, Speed and Activity
- 802.3af PoE on All Ports Providing up to 15.4W of Power
  - Selectable level per port via web-based management interface
- Auto-MDI/MDI-X Auto Crossover (when auto-negotiation is available)
- Layer 2 Switching
- QoS

- 802.1Q VLANs
- 802.1p Priority Queuing
- Port Mirroring
- 802.3x Flow Control
- Independent VLAN Learning Support
- TCP/IP Networking Stack
- Multi-Unit Stacking (multiple blades in a system are managed from the same user interface)
- Dynamic PoE Control (allows setting the proper PoE classifications for each port to stay within the system power budget)
- Switch Management Through Web-Based GUI
- Software Upgrades Via TFTP

Ports 1 and 8 are the default “uplink” ports. All the user management and stacking is based on this setup.

Multiple-Unit Stacking allows the user the ability to manage multiple GSWU blades in a system as one switch, instead of individual units and IP addresses. For example, a set of 3 blades would appear to the user interface as a 24-port switch, instead of 3 8-port switches. Stacking works by assigning a Main Management Blade, which will provide all the GUI information for all blades in the same stack. The CCPU assigns the Main blade by issuing an IP address to the Main blade via PAW/PRW during initialization sequence. All other GSWU blades detected in the system will not be assigned an IP address, thereby signifying them as “add-on” blades.

A single UX5000 system can have up to 12 GSWU blades per system. However, only 3 GSWU blades can be grouped together to form a single 20-port switch. When more than 3 GSWU blades are present within a system, the blades not grouped together will not have any of the software feature of the stacked blades. They will behave as an unmanaged gigabit ethernet switch (only eight 10/100/1000 ethernet ports and PoE Class 3 [lowest power class]). The 2 GSWU blades can be categorized into one “Main” blade, with 2 additional “Add-on” blades.

When a Main blade is initialized, it assigns the first blade ports 1-8. When subsequent add-on blades are recognized, the system assigns port numbers on a first-come-first-serve basis. If you wish to have sequential port numbers, insert each add-on blade, one at a time, into the system without adding any other type of blade. When a blade is removed, the port numbers are not automatically removed. They can be removed by accessing the Main blade GUI.

The IP address, subnet mask, and gateway address of the Main blade will be assigned in Program 10-55.

Stacking 3 GSWU blades to form a 20-port switch is restricted to a single system location. The grouping is not allowed across a network/CygniLink.

The VOIPDB daughter board, which is required in order for IP telephones to communicate with non-VoIP UX5000 phones, as well as to place or receive outside calls, must be connected to either a switching hub or to the Router blade.

Voice Mail

16-Port UX IntraMail Daughter Board (16VM) — P/N 0911026

The UX IntraMail is a plug-in "in-skin" full-featured, integrated Voice Mail with Automated Attendant. This daughter board is installed on the CCPU and requires a 16-hour or 32-hour IntraMail CompactFlash card.

The IntraMail Automated Attendant answers incoming calls and routes them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Soft Keys guide the display telephone user through the extensive IntraMail feature set.

16-Hour UX IntraMail CompactFlash Card — P/N 0910505

Used with the UX IntraMail Daughter board (P/N 0911026), this 256M CompactFlash card will provide 16 hours of voice storage. The voice storage capacity is with the default 3 languages installed (U.S. English (Numeric & Mnemonic), Spanish, and French).

This card will also provide VRS features, including ACD announcements.

32-Hour UX IntraMail CompactFlash Card — P/N 0910506

Used with the UX IntraMail Daughter board (P/N 0911026), this 512M CompactFlash card will provide 32 hours of voice storage. The voice storage capacity is with the default 3 languages installed (U.S. English (Numeric & Mnemonic), Spanish, and French).

This card will also provide VRS features, including ACD announcements.

APSU / UX Mail Blade (APSU) — P/N 0911064

The UX Mail is a fully integrated, blade-based “in-skin” Voice Mail with Automated Attendant. Its robust feature set rivals the capabilities of stand-alone products on a single, plug-in PCB. This blade requires a 124-hour or 550-hour UX Mail CompactFlash card.

The IntraMail Automated Attendant answers incoming calls and routes them quickly and efficiently. Integrated Voice Mail features include Conversation Record, Answering Machine Emulation, and Caller ID with Return Call. Interactive Soft Keys guide the display telephone user through the extensive IntraMail feature set.

125-Hour UX Mail CompactFlash Card — P/N 0910530

Used with the UX Mail APSU blade (P/N 0911067), this 2G CompactFlash card will provide 125 hours of voice storage. The voice storage capacity is with the default 4 languages installed (U.S. English Numeric, U.S. English Mnemonic, Spanish, and French).

This card will also provide VRS features, including ACD announcements.

550-Hour UX Mail CompactFlash Card — P/N 0910531

Used with the UX Mail APSU blade (P/N 0911067), this 8G CompactFlash card will provide 550 hours of voice storage. The voice storage capacity is with the default 4 languages installed (U.S. English Numeric, U.S. English Mnemonic, Spanish, and French).

This card will also provide VRS features, including ACD announcements.

Abbreviated Dialing

Feature Availability
• Available.
• 2000 bins available (0000-1999) for Common and Group Abbreviated Dialing. Up to 64 Abbreviated Dialing Groups available.
• 10 bins available (1-9, 0) for Personal Abbreviated Dialing.

Description

Abbreviated Dialing gives an extension user quick access to frequently called numbers. This saves time, for example, when calling a client with whom they deal with often. Instead of dialing a long telephone number, the extension user just dials the Abbreviated Dialing code.

There are three types of Abbreviated Dialing: Common, Group and Personal. All co-workers can share the Common Abbreviated Dialing numbers. All co-worker's in the same Abbreviated Dialing Group can share the Group Abbreviated Dialing numbers. Personal Abbreviated Dialing numbers are available only at a user's own extension. The UX5000 has 2000 Abbreviated Dialing bins that you can allocate between Common and Group Abbreviated Dialing. (The Group bins are assigned in groups of 10.) Personal Abbreviated Dialing allows up to 10 bins in which you can store numbers or functions.

Each Abbreviated Dialing bin can store a number up to 24 digits long.

Common and Group Abbreviated Dialing

When placing an Abbreviated Dialing call, the UX5000 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 UX5000 can optionally force Common Abbreviated Dialing numbers to route over a specific Trunk Group. User pre-selection always overrides the UX5000 routing.

Common Bins Limited to 1000 with Dial Key or #2 Service Code

Though there are 2000 Abbreviated Dialing bins available in the UX5000, once programmed, these bins can currently only be dialed using the Directory Dial feature (Press Directory Dialing Soft Key + ABB Soft Key + Use arrow keys to locate number or enter the Abbreviated Dial bin name + CALL or SPK to place call.)

The DIAL key and service code #2 operations are not available for any 4-digit Abbreviated Dial common bin number.

DSS Key Chaining

DSS key chaining allows an extension user with DSS keys to chain to an Abbreviated Dialing number stored under a DSS 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 key (e.g., #200) and the client's extension number under the other (e.g., 400). The 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). See Programming below for additional details.

The user can also chain to an Abbreviated Dialing number dialed manually, from a Programmable Function Key.

Storing a Flash

To enhance compatibility with connected Centrex and PBX lines, an Abbreviated Dialing bin can have a stored Flash command. For example, storing 9 Flash 926 5400 will cause the UX5000 to dial 9, flash the line and then dial 926 5400. The Flash can be stored by the user from their terminal or by the system administrator during UX5000 programming.

Using a Programmable Function Key

To streamline frequently-called numbers, an Abbreviated Dialing Programmable Function Key can also store an Abbreviated Dialing bin number. When the extension user presses the key, the terminal automatically dials out the stored number. This provides true one-touch calling via a terminal's function keys.

Personal Abbreviated Dialing

Personal Abbreviated Dialing gives a keyset user quick access to extensions, trunks and selected UX5000 features. This saves users time when accessing co-workers, clients and features they use most often. When used in combination with a Programmable Function Key, a user can have one-button access to stored telephone numbers or features. An extension user can define Personal Abbreviated Dialing bins for:

- **Direct Station Selection** - quick access to extensions
- **Abbreviated Dialing** - quick access to stored Common/Group Abbreviated Dialing numbers or to locally stored numbers (up to 24 digits long)
- **Trunk Calling** - quick access to trunks or trunk groups
- **Service Codes** - quick access to specific Service Codes

When using Programmable Function Keys for Personal Abbreviated Dialing, an extension user can chain dial. For example, a user can store the number for a company's Automated Attendant in key 13 and employee extension numbers in keys 14-18. The user presses key 13 to call the company, then one of keys 14-18 to ring the employee to which they want to speak.

An extension user or system administrator can optionally store a Flash command with a Personal Abbreviated Dial bin number. The stored Flash may be helpful to access features of the connected telco, PBX or Centrex.

Account Codes

Feature Availability
• Available.

Description

Account Codes are user-dialed codes that help the system administrator categorize and/or restrict trunk calls. The UX5000 has three types of Account Codes:

- **Optional Account Codes**
  Optional Account Codes allow a user to enter an Account Code while placing a trunk call or anytime while on a call. This type of Account Code is optional; the UX5000 ***does not*** require the user to enter it.
- **Forced Account Codes**
  Forced Account Codes ***require*** an extension user to enter an Account Code every time they place a trunk call. If the user doesn't enter the code, the UX5000 prevents the call. As with Optional Account Codes, the extension user can elect to enter an Account Code for an incoming call. However, the UX5000 does not require it. ***Forced Account Codes does not block 1-800, 1-888 and emergency assistance (911) calls***.

  Once set up in UX5000 programming, you can enable Forced Account Codes on a trunk-by-trunk basis. In addition, Forced Account Codes can apply to all outside calls or just long distance calls. Forced Account Codes for toll calls restricts calls according to the following chart:

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

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

<u>**Timer Allows for Forced Account Code Interdigit Timer**</u>
Programming allows the adjustment of the interdigit timer used for Forced Account Codes. The UX5000 will use the time set in **Program 21-01-04 : System Options for Outgoing Calls - Dial Tone Detection Time** for the interval the UX5000 will wait for a user to enter a Forced Account Code when using a trunk access code (ex: 9) or F-Routing. By default, this option is set to 5 (Entries: 0-64800 seconds).

- **Verified Account Codes**
  With Verified Account Codes, the UX5000 compares the Account Code the user dials to a list of up to 2000 pre-programmed codes. If the Account Code is in the list, the call goes through. If the code dialed is not in the list, the UX5000 prevents the call. Verified Account Codes can be from 3-16 digits long using the characters 0-9 and #. During programming, you can use "wild cards" to streamline entering codes into UX5000 memory. For example, the entry 123W lets users dial Verified Account Codes from 1230 through 1239.

Operator Notification
To prevent Account Code abuse, the UX5000 can notify the operator each time an Account Code violation occurs. This can happen if the user fails to enter an Account Code (if Forced) or enters a Verified Account Code that is not in the list. The notification is an automatic Intercom call to the attendant and a "*RESTRICT*" message in the operator's display. (If the attendant fails to enter a valid Account Code, the UX5000 drops the call.)

Account Codes for Incoming Calls
The UX5000 can control the ability of extension users to enter Account Codes for incoming calls. When this option is enabled, a user can dial * while on an incoming call, enter an Account Code, and then dial * to return to their caller. If the option is disabled, any digits the user dials after answering an incoming call outdial on the connected trunk.

Hiding Account Codes
Account Codes can be optionally hidden from a terminal's display. This would prevent, for example, an unauthorized co-worker from obtaining a Verified Account Code by watching the display and making note of the digits that dial out. When hidden, the Account Code digits show as the character "*" on the terminal's display.

Account Code Capacity
Account Codes print along with the other call data on the SMDR record after the call completes. Account Codes can be 1-16 digits in length using 0-9 and #. Verified Account Codes can be from 3-16 digits long.

Redialed Numbers Do Not Contain Account Codes
When using the Last Number Redial, Save or Repeat Dial features, the UX5000 will not retain Account Code information. Any number redialed with these features, the user will need to reenter an Account Code.

Note:
If a user enters *12345*203 926 5400*67890*, if the Last Number Redial feature is used, the UX5000 dials the number as 203 926 5400*67890*. The *67890* is not treated as an Account Code.

Alarm

Feature Availability
• Available.

Description

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

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

Alarm Reports

Refer to the Maintenance (page 131) **for information on this feature.**

Alphanumeric Display

Feature Availability
• Available.

Description

Multibutton display terminals have an alphanumeric display that provides various feature status messages. The size of the display varies depending on the keyset model and the display size selected (normal or double height). For example, the Sophisticated IP terminal can provide a 4-line, 28 character per line or a 3-line, 28 character per line display at the double-height setting. These messages help the display terminal user process calls, identify callers and customize features. Refer to the table below for a listing of the various display types available:

Terminal	Number of Rows	Number of Columns
IP-CTS	4 rows (normal) / 3 rows (double)	28
Enhanced IP	4 rows (normal) / 3 rows (double)	28
Value IP	3 rows (normal)	24
Enhanced Digital	4 rows (normal)	24
Value Digital	3 rows (normal)	24

General Purpose LED

If required, General Purpose LED Operation keys can be assigned. These Programmable Function Keys can be used to light an LED on a keyset (for example, in a doctor's office, to let the doctor know a patient is ready to be seen). Code 56 is programmed on the keyset which should set the status, and code 57 is programmed on the keyset which should display the lit LED. A key can display as light solid red or solid green, or by pressing once it's lit solid red and pressing the key again it's lit solid green.

3

Analog Communications Interface (ACI)

Feature Availability
• Available - 96 ACI software ports (48 PGDAD modules max. when used for ACI ports) and 16 ACI Department Groups.

Description

The Analog Communications Interface (ACI) feature uses a PGDAD module to provide two analog ports (with associated relays) for Music on Hold, External Paging or auxiliary devices such as tape recorders and loud bells. The UX5000 allows up to 48 PGDAD modules (when used for ACI ports), for a maximum of 96 analog ports. Each PGDAD module requires an unused port on an ESIU blade.

- **Music on Hold**
  You can connect up to two customer-provided Music on Hold music sources to a PGDAD module. This lets you add additional music sources if the external source on the CPU blade or the internal source are not adequate. By using PGDAD modules, you could even have a different music source for each trunk.

  When the UX5000 switches the ACI analog port to a trunk on Hold, the PGDAD relay associated with the ACI analog port closes. You can use this capability to switch on the music source, if desired.

  Extension users can dial the ACI analog port extension number and listen to the connected music source. The PGDAD relay associated with the port closes when the call goes through.

  For Music on Hold, connect the music source to the PGDAD module. Connect the music source control leads to the CTL (control relay) jack. Refer to the Hardware Manual for additional details.

- **External Paging**
  An ACI analog port can also be an External Page output. When connected to customer-provided External Paging equipment, the ACI port provides External Paging independent of the UX5000 CCPU external paging input. To use the External Paging, an extension user just dials the ACI analog port extension number and makes the announcement. The UX5000 broadcasts the announcement from the ACI analog port and simultaneously closes the associated PGDAD relay. You can use the relay closure to control the External Paging amplifier, if required.

  This external paging zone is not included in external all call paging or combination paging (internal and external).

  For External Paging, connect the Paging amplifier to the PGDAD jack. Connect the amplifier control leads to the CTL (control relay) jack. Refer to the Hardware Manual for additional details.

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

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

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

- **ACI Call Recording**
  ACI Call Recording allows you to use a recording device connected to a PGDAD module to manually or automatically record calls. The recording device is typically a customer-provided tape recorder. You can set up ACI Call Recording to output to a single ACI port/recording device or to a pool of ACI ports/devices. With a single device, all calls are stored in a centralized location. With a pool of devices, you'll be sure to have a port available for recording - even in peak traffic periods. You can set up automatic recording on a per trunk or manually on a per extension basis.

  When set up for manual recording, the user presses the ACI Conversation Record key (Service Code 851 + 69 + 0) to begin recording the call from that point. When set up for automatic recording, ACI Call Recording starts automatically as soon as the user places or answers their call. The UX5000 can be programmed to record all ***incoming*** trunk calls which ring an extension. This includes the following trunk types:
  - Central Office calls programmed to ring the extension.
  - Direct Inward Dialing (DID)
  - Direct Inward Line (DIL)
  - Direct Inward System Access (DISA)
  - Tie lines

  The UX5000 can also be programmed to record ***outgoing*** trunk calls, however, this is only possible using E&M tie lines, PRI or BRI trunks.

  ACI Call Recording is not available for intercom calls, transferred calls, or calls placed on hold and answered by an extension with Call Recording enabled. To manually record any type of call (transferred, ICM, outgoing CO trunk, etc.), use the Voice Mail Conversation Record key (Service Code 851 + 78).

Physical Ports and Software Ports

Each PGDAD module consists of a physical port for connection to the UX5000 and two logical ports. For programming purposes, the ports are also called software ports. The physical port connects to a station position on a ESIU blade. During installation, the first PGDAD module you set up is physical port 1; the second PGDAD module is physical port 2, etc. Each PGDAD module has two software ports, which are numbered independently of the physical ports. Normally, the first PGDAD module set up has software ports 1-2; the second PGDAD module has software ports 3-4, etc. There are a total of 96 software ports (48 PGDAD modules x 2 ports each). During programming, you assign ACI extension numbers and Department Group options to PGDAD software ports, not physical ports. During installation, you connect equipment to the jacks on the PGDAD module that correspond to the software port. Refer to the UX5000 Hardware Manual for more installation details.

Aspire Telephones

Feature Availability
• Available.

Description

With the UX5000, Aspire digital keysets, 24-Button and 110-Button DSS Consoles, and some of the phone adapters (ex: APR, APA, etc.) can be used on the UX5000 communications server.

Each Aspire keyset, console or adapter would have the same load factor as when used on the Aspire system. Refer to the Aspire Hardware Manual (P/N 0893100) for a chart with the load factor information.

The following Aspire hardware is supported on the UX5000 communications server:

Keysets/Consoles:

- 2-Button Telephone - Black (P/N 0890047)
- 2-Button Telephone - White (P/N 0890048)
- 22-Button Handsfree Non-Display Telephone - Black (P/N 0890041)
- 22-Button Handsfree Non-Display Telephone - White (P/N 0890042)
- 22-Button Handsfree Display Telephone - Black (P/N 0890043)
- 22-Button Handsfree Display Telephone - White (P/N 0890044)
- 34-Button Handsfree Display Telephone - Black (P/N 0890045)
- 34-Button Handsfree Display Telephone - White (P/N 0890046)
- 34-Button Super Display Telephone - Black (P/N 0890049)
- 34-Button Super Display Telephone - White (P/N 0890050)
- 24-Button DLS Console - Black (P/N 0890053)
- 24-Button DLS Console - White (P/N 0890054)
- 110-Button DSS Console - Black (P/N 0890051)
- 110-Button DSS Console - White (P/N 0890052)

Cordless Telephones and Options:

- Analog Cordless Phone SLT DTR-1R-2 (BK) (5.8 GHz) (P/N 730093)
- NEC Cordless II Telephone (P/N 730088)
- NEC Cordless Lite II Telephone (P/N 730087)

Peripherals

- APR (Analog Interface with Ringing) Adapter (P/N 0890056)
- APA (Analog Interface without Ringing) Adapter (P/N 0890057)
- 2PGDAD Module (for Door Box/Page/ACI) (P/N 0891027)
- Door Box (P/N 922450)

The following Aspire hardware is **NOT** supported on the UX5000 communications server:

- Aspire Wireless Base Station (P/N 780136)
- Aspire Wireless Handset with Battery (P/N 780004)
- 4-Button Aspire IP Phone - Black (P/N 0890072)
- 34-Button Aspire IPhone - Black (P/N 0890065)
- 34-Button Aspire IPhone - Black (P/N 0890073)
- IP Adapter (IP) (P/N 0890060)
- In-Line Power Adapter (ILPA) (P/N 780122)
- Power Failure Adapter for IP Telephone (PSA) (P/N 0890067)
- Recording Adapter for IP Telephone (ADA2) (P/N 0890066)

Attendant Call Queuing

Feature Availability
• Available.

Description

Attendant extensions can have up to 32 incoming calls queued before additional callers hear busy tone. This helps minimize call congestion in UX5000s 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 doesn’t have an appearance for the queued call, it waits in line on a CALL key. 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 UX5000 feature.

Automatic Call Distribution (ACD)

Feature Availability
• Available with ACD Software License - 64 ACD Groups and 512 ACD Agents. • For more information, refer to the ACD Manual (P/N 0913202).

Description

Automatic Call Distribution (ACD) uniformly distributes calls among member agents of a programmed ACD Group. When a call rings into an ACD Group, the UX5000 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 UX5000 allows up to 64 ACD Groups and 512 ACD agents.

You can put any agent in any group. In addition, an agent can be in more than one group. This allows, for example, a Technical Service representative to answer Customer Service calls at lunch time when many of the Customer Service reps 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 within the normal extension range.

Automatic Call Distribution operation is further enhanced by:

- **ACD Call Queuing**
  When all agents in an ACD Group are unavailable, an incoming call will queue and cause the Queue Status Display to occur on the ACD Group Supervisor’s display. The display helps the supervisor keep track of the traffic load within their group. The Queue Status Displays shows:
  - The number of calls queued for an available agent in the group.
  - The trunk that has been waiting the longest, and how long it has been waiting.

  For each ACD Group, you can set the following conditions:
  - The number of trunks that can wait in queue before the Queue Status Display occurs.
  - How often the time in queue portion of the display reoccurs.
  - If the supervisor should hear a Queue Alarm whenever the time in queue portion reoccurs. This alarm is a single beep tone that reminds the supervisor to check the condition of the queue.

- **ACD Overflow**

  ACD Group as Overflow Destination
  The UX5000 can transfer an overflow call to a specific ACD Group, to voice mail using Program 41-09, an off-premise number (using a programmed Abbreviated Dial number) or incoming ring group. When **Program 41-08-02 : ACD Overflow Destination has the ACD Overflow Destination** set to '65', the UX5000 will overflow the call to the ACD Group programmed in Program 41-09. (The UX5000 will not allow you to program an ACD group with that ACD group as the overflow.) If, while the call is ringing, the extension to which the call was transferred becomes available, both the extension and the overflow ACD group will ring.

  Notes:
  - When using Off-Premise Overflow transfer and if all trunks are busy, the UX5000 will set the ACD Overflow Timer again and wait to retry. When this occurs, the queue information of this call will be set to the oldest queue.
  - Once the Overflow call has successfully transferred to the Off-Premise or ring group, the call is removed from the retry queue.
  - Once the Overflow call has successfully transferred to the Off-Premise or ring group, the UX5000 will notify the ACD-MIS that the call was abandoned by the caller.
  - If the user programmed Overflow Transfer to the ring group which is used by the other ACD group, the UX5000 will notify ACD-MIS with new call information. Therefore, the ACD-MIS will count the transferred call as new call.

  Option for Counting Overflow Calls in ACD MIS Software
  An option is available for use with ACD MIS (such as inDepth Lite/inDepth/inDepth+) software to count overflow calls. When the number of queued ACD overflow calls exceeds the limit and a busy tone is sent to the caller, the UX5000 programming will determine if the call is counted in the ACD MIS software.

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

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

  With the use of ACD Overflow and a VRS, you also have the option of providing a Queue Depth Announcement. With this option enabled, your callers will know where they stand in line while waiting for an available agent.

  ACD Announcements can be used with IntraMail when the “quiet hang up” option is used. This will allow the message to play, but will suppress the “good bye” heard with other mailboxes.

  The voice mail can also provide the ACD overflow announcements in UX5000s that do not have VRS available. When a caller queues for an available agent, designated Voice Mail ACD Announcement Mailboxes provide the overflow messages.

- **Escape from Queue**
  This option requires a VRS or voice mail.
  An option is available for ACD callers when the VRS or voice mail provides the announcements for an ACD queue. When a caller is waiting in the ACD queue for an available agent, the UX5000 can allow the user to dial a single digit code to exit the queue and be transferred to a defined destination. This option can be set to allow the user to dial out during the delay announcement or within a set time after the announcement finishes. After listening to this type of announcement, they can either wait in queue or dial a digit for an alternate destination. The destination is typically the operator, a mailbox or an extension. In order for this option to work, a VRS must be installed in the UX5000 and it must be providing the ACD announcements or the voice mail’s Call Routing Mailbox must be used to provide the ACD announcements.

  Notes:
  - VRS or voice mail’s Call routing Mailbox must be used to provide the queue announcements.
  - This feature is not available with ACI announcements. When internal voice mail is used, the delay announcements will be provided, but the caller will not be able to escape the queue.
  - If there are no DTMF receivers available at the time the caller presses the single digit code, the UX5000 will retry after the next delay message.
  - The Escape From Queue feature is disabled if Programs 41-14-14 and 41-14-15 are both set to "0".
  - This feature will only allow the Escape From Queue single digit code to be dialed. Any other numbers dialed will be ignored and the call will stay in queue.

- **Agent Log In and Log Out Services**
An ACD Agent can log in and log out of their ACD Group. While logged in, the agent is available to receive ACD Group calls. When logged out, the agent is excluded from the group's calls. The programmable keys and alphanumeric display on an agent's terminal show at a glance when they are logged in or logged out.

- **Agent Identity Code (AIC)**
An Agent Identity Code (AIC) allows ACD agents to log in any extension without setting Program 41-02 (AIC Log In). Using AIC, ACD agents can also log in to multiple ACD groups at the same time (up to 64 ACD Groups). The UX5000 will also allow all extensions (up to the communications server maximum) to log in using the same AIC code. AIC and ACD groups for each work period (mode pattern number) can be set in Program 41-18 as shown in the following example.

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

With this example, ACD will work as follows:
Example 1: Log In with AIC 789

- During Mode Pattern 1, ACD agents will belong to ACD groups 1, 2, and 16 at the same time.
- During Mode Pattern 2, ACD agents will belong to only ACD group 1.
- During Mode Pattern 3-8, ACD agents will not belong to any ACD group and the ACD extensions will work as normal extensions.

Example 2: Log In with AIC 567

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

Example 3: Log In with AIC 678

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

AIC log on/log off operations are supported. The UX5000 P Commands will indicate which ACD Group is being logged onto when an AIC code is entered. Also when an agent logs off, there will be a multiple log of events, one log off event for each previous ACD log on event.

- **Multiple Agent Log In**
ACD agents can log their extension in using multiple AICs (up to 3). The UX5000 will also allow all extensions (up to the UX5000 maximum) to log in using the same AIC code. For example, even if ACD agent "A" logs in extension 350 with AIC 789, ACD agent "B" can also log in to extension 351 with AIC 789 at the same time. Using the example setup above, ACD will work as follows:

Example 1: Log In with AIC 789 and 568

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

Example 2: Log In with AIC 789, 568 and 678

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

In addition to an agent logging in with multiple AIC codes, ACD agents can log in to multiple ACD groups at the same time (up to 64 ACD Groups). AIC and ACD groups for each work period (mode pattern number) can be set in Program 41-18.

Some conditions with Multiple Agent Log In:

- ACD agents cannot log in to the system supervisor or group supervisor's extension.
- In order to log in with AIC, the extension should be set to AIC Log In mode in Program 41-17.
- If the extension is set to AIC log in mode in Program 41-17, the UX5000 will ignore the setting of Program 41-02 for the extension.
- A supervisor cannot log out an agent logged in by an AIC code.

3

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

- **Enhanced DSS Operation**
  A programmed extension user can use their DSS Console to monitor the status of the ACD Agents within a group. The DSS Console is an essential tool for supervisors. The console key flash rates tell the supervisor at a glance which of the group's agents are:
  - Logged onto the group (i.e., in service)
  - Logged out of the group (i.e., out of service)
  - Busy on a call
  - Placing an Emergency Call to the supervisor
  - Not available or installed

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

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

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

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

- **Rest Mode**
  Rest Mode temporarily logs-out an ACD agent's terminal. There are two types of Rest Mode:-
  - **Manual Rest Mode**
    An ACD Agent can enable Manual Rest Mode anytime they want to temporarily log out of the ACD Group. They might want to do this if they go to a meeting or get called away from their work area. While logged out, calls to the ACD Group will not ring the agent's terminal.
  - **Automatic Rest Mode**
    When an ACD Group has Automatic Rest Mode, the UX5000 will automatically put an agent's terminal in Rest Mode if it is not answered. This ensures callers won't have to wait while ACD rings an extension that won't be answered. For keysets, the UX5000 enables Automatic Rest Mode for all terminals with Rest Mode keys. For SLTs, you must set an option in programming to enable Automatic Rest Mode. If an agent's terminal is placed into Rest Mode because a call is not answered, the agent will need to manually cancel Rest Mode in order to log back into the ACD group.

  With a Rest Mode key programmed on an ACD agent's terminal, when the agent is in rest mode, the key will be lit. If the Rest Mode key is pressed while an agent is on a call, the key will flash to indicate a pre-Rest Mode status. When the current call is finished, the agent's terminal will be in rest mode. The agent can place intercom calls or receive direct incoming calls while in Rest Mode. The ability to receive incoming intercom calls is defined in UX5000 programming for each ACD group. Note that an ACD System Supervisor cannot be placed in Rest Mode.

- **Supervisor, ACD Group**
  You can designate an extension in an ACD Group to be the group's supervisor. Once assigned as an ACD Group Supervisor, the user can:
  - Take the entire ACD Group out of service.
  - Check the log out status of each agent after the group taken down.
  - Restore the ACD Group to service.

  During programming, you can choose one of three modes of operation for each ACD Group supervisor:
  - Supervisor's extension cannot receive calls to the ACD Group.
  - Supervisor's extension can only receive ACD Group calls during overflow conditions.
  - Supervisor's extension receives calls just like any other ACD Group agent (mode 2).

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

- **Supervisor, ACD System**
  You can designate an extension as an ACD System Supervisor. Once assigned as an ACD System Supervisor, the user can:
  - Take the all the UX5000's ACD Groups out of service simultaneously

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

The UX5000 can have only one ACD System Supervisor.

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

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

  - **Automatic Work Time**
    The UX5000 implements Automatic Work Time for the agent as soon as they hang up their current call. This is helpful in applications (such as Tech Service groups) where follow-up paperwork is a requirement for every call. When the agent is done with their work, they manually return themselves to the ACD Group.

- **ACD Group Call Coverage Keys**
  To help cover calls during peak periods, a keyset can have Call Coverage keys for ACD Groups by assigning the ACD master number to the Call Coverage key. When a call rings into a covered ACD Group, it rings the appropriate ACD Group Call Coverage key, allowing users to pick up incoming ACD calls. The key can ring immediately, after a delay or just flash. The Call Coverage key also facilitates one-button Transfer for an ACD Group. The covering extension does not have to be a member of the ACD Group.

ACD Call Coverage Key LED Pattern	Status
Off	There is no incoming call to the ACD group.
Flashing Red	Incoming call(s) are ringing the ACD group.

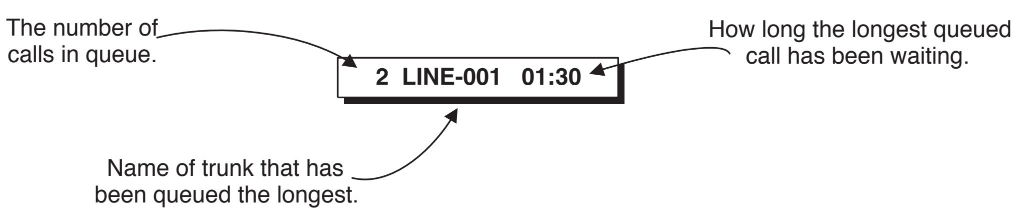
- **Hotline Key Shows Agent Status**
  An extension's Hotline keys provide the "normal" Busy Lamp Field (BLF) for co-workers and a unique BLF for ACD Agents. Similarly to the supervisor's DSS Console BLF, the unique BLF shows when the covered agent is in service, out of service or busy on a call. This enhanced BLF gives a department manager, for example, ACD Group monitoring capabilities without having to become a supervisor with a DSS Console.
  Hotline gives a keyset user one-button calling and Transfer to another extension (the Hotline partner). Hotline helps co-workers that work closely together. The Hotline partners can call or Transfer calls to each other just by pressing a single key. Enhanced for ACD applications, Hotline provides a unique Busy Lamp Field for ACD agents as well as a BLF for co-workers that are not ACD agents. The charts below show both sets of BLF indications.

BLF For ACD Agents	
When the key is . . .	**The ACD Agent is . . .**
Off	Idle and is not an ACD Agent
On	Busy
Double Wink Off	Making an Emergency Call
Wink Off	Logged off or not installed
Double Wink On	Logged on

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

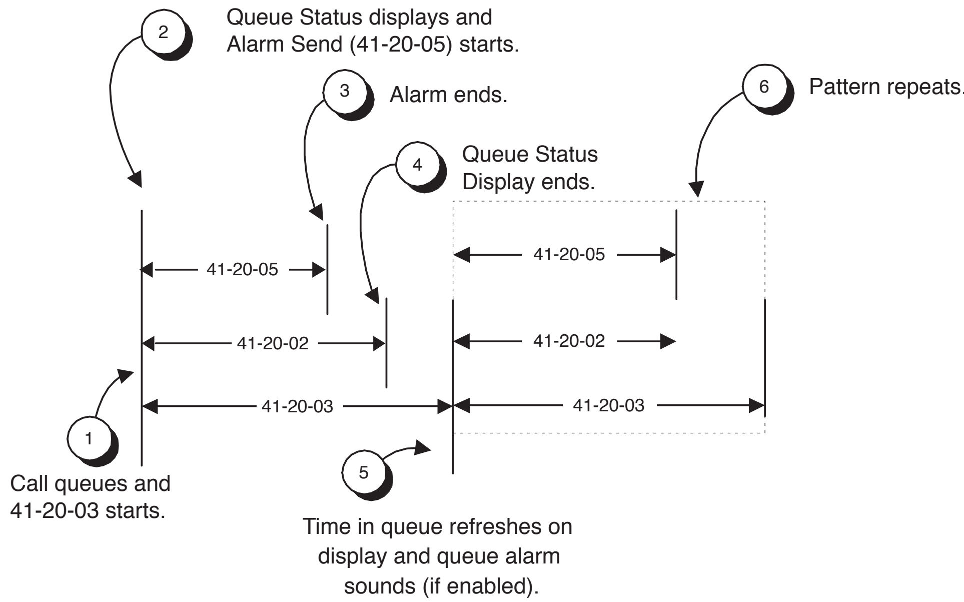
- **Enhanced Supervisor Options**
  An ACD supervisor can individually assign extensions to ACD Groups, and set an agent's status once assigned. This provides the supervisor with tremendous flexibility to reassign agents as work loads vary.

- **Queue Status Display with Scrolling**
  When all agents in an ACD Group are unavailable, an incoming call will queue and cause the Queue Status Display to occur on the ACD Group Supervisor and/or agent's display (based on the Class of Service). The display helps the supervisor keep track of the traffic load within their group. In addition, any display keyset can have a Queue Status Display Check programmable function key. The keyset user can press this key any time while idle, and using the VOL ▲ and VOL ▼, scroll through the Queue Status Displays of all the ACD Groups. The Queue Status Displays shows (see the Queue Status Display illustration below):
  - The number of calls queued for an available agent in the group.
  - The trunk that has been waiting the longest, and how long it has been waiting.



For each ACD Group, you can set the following conditions:
- The number of trunks that can wait in queue before the Queue Status Display occurs.
- How often the time in queue portion of the display reoccurs (see the Queue Status display Timing illustration below).
- Queue Status Display holding time.
- Queue Status Alarm enable/disable.

●Queue Status Alarm sending time.



When Logged Out of ACD Group

When ACD agents are logged out and a call is placed into the ACD queue, the terminals of the logged out agents will display the Queue Status and hear the alarm according to the settings defined in UX5000 programming. Pressing the Queue Status Display Programmable Function key will return the terminal to idle until the timer in Program 41-20-03 expires again.

Notes:

(A.) Do not use both 41-15 and 41-20 to set the ACD queue alarm. Select either one or the other for the UX5000 to follow.

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

(B.) If a terminal is not idle, it cannot use the Queue Status Display Programmable Function key.

(C.) The Queue Status Display is not shown and the Queue Alarm is not heard by ACD agents those in Off-Duty mode.

(D.) In order to scroll through the ACD groups queue status, the Queue Status Display

Programmable Function key must be used. You cannot scroll when the Queue Status Display appears due to an alarm condition.

(E.) If the Queue Status display and alarm are active and the queued called is answered/ disconnected, the display and alarm will continue until the timers in Program 41-02-02 and 41-20-05 expire.

(F.) When an overflowed call is in queue, the call will be included in its original ACD group's queue and not in the group's queue to which it overflowed.

(G.) The Queue Status is not displayed on a supervisor's terminal based on the settings in Programs 41-02-xx. The supervisor must use the Queue Status Display Programmable Function key to view the queue.

- **Programmable Wrap-up Timer**
  When an agent finishes their call, the UX5000 automatically starts a wrap-up timer and blocks any ACD calls to the agent. This gives them time to complete important logs and records before a new call comes in. When the timer expires, the UX5000 returns the agent to the ACD Group to handle new callers.

- **inDepth Lite, inDepth and inDepth+**
  inDepth Lite, inDepth and inDepth+ are Windows-based Management Information Systems that work with the UX5000's built-in ACD. These ACD/MIS systems enhance the UX5000 with real time statistics and reports on ACD Group traffic patterns and usage. Refer to the *inDepth Lite, inDepth and inDepth+* feature for more details.

Using CVM and Multiple Sources for Announcements

Each delay message can separately be assigned to use the ACI, VRS, local voice mail, or centralized voice mail. The voice mail sources can be from an external voice mail, the UX5000 Mail or IntraMail.

Note the following conditions with this feature:

- When using the CVM for announcement message, if the voice mail is located in the local UX5000, the programming should still be entered as for a CVM and not a local voice mail.
- If the message source for the first announcement is not set and all the ACD agents are busy, a caller will continue to hear ring back tone even after the message start time.
- If the message source for the second announcement is not set and all the ACD agents are busy, the following will occur:
  With Program 41-08-08 set to VMI or CVM and Program 41-08-09 set to ACI or VRS
  After the first message is sent, the defined interval tone (RBT, MOH, BGM) for the first announcement message is heard. This tone will continue to be heard until an ACD terminal becomes free.
- If the message source for the second announcement is not set and all the ACD agents are busy, the following will occur:
  With Program 41-08-08 set to VMI and Program 41-08-09 set to CVM
  OR
  Program 41-08-08 set to CVM and Program 41-08-09 set to VMI
  After the 1st message is sent, the second message is sent from the source of the first message.

The following table indicates the required programs when setting Delay Announcements using different sources for the first and second announcements.

2nd Announcement Source	1st Announcement Source			
	ACI	VRS	Local Voice Mail	CVM
ACI	41-10	41-11: 01, 02, 03, 06 22-01-11 41-10: 02, 05	41-19: 01, 02, 03, 06, 08 41-10: 02, 05	41-19: 01, 02, 03, 06, 08 41-10: 02, 05
VRS	41-10: 01, 03 41-11: 04, 05, 06, 07 22-01-11	41-11 22-01-11	41-19: 01, 02, 03, 06, 08 41-11: 04, 05, 06, 07 22-01-11	41-19: 01, 02, 03, 06, 08 41-11: 04, 05, 06, 07 22-01-11
VMI	41-10: 01, 03 41-19: 01, 04, 05, 06, 08	41-11: 01, 02, 03, 06 22-01-11 41-19: 01, 04, 05, 06, 08	41-19	41-19
CVM	41-10: 01, 03 41-19: 01, 04, 05, 06, 07, 08	41-11: 01, 02, 03, 06 22-01-11 41-19: 01, 04, 05, 06, 08	41-19	41-19

For more on Automatic Call Distribution, refer to the ACD Manual (P/N 0913202).

3

Automatic Release

Feature Availability
• Available.

Description

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

Automatic Route Selection

Feature Availability
• Available.

Description

Automatic Route Selection (ARS) provides call routing and call restriction based on the digits a user dials. ARS gives the UX5000 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 UX5000 options) from a display terminal. ARS accommodates 400 call routing choices - without a custom-ordered rate structure database. With ARS, you can modify the UX5000’s 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 pre-determined levels. When UX5000s are networked together by a tie line, the networked systems can be called by a system number and a user's extension number, just an extension number, or by using a trunk access code.

ARS Feature Summary

ARS provides:

- **Call Routing**
  ARS can apply up to 24-digit analysis to every number dialed. For programming, ARS provides separate 8-digit and 24-digit tables. Each table can have up to 250 numbers.
- **Dialing Translation (Special Dialing Instructions)**
  ARS can automatically execute stored dialing instructions (called Dial Treatments) when it chooses a route for a call. The UX5000 allows up to 15 Dial Treatments. The Dial Treatments can:
  - Automatically insert or delete a leading 1
  - Insert or delete an area code (NPA)
  - Add digits (such as a dial-up OCC number), pauses and waits to the dialing sequence
  - Require the user to enter an authorization code when placing a call (refer to Program 44-03).
- **Time of Day Selection**
  For routing purposes, ARS provides ten different day selections (called Time Schedule Patterns). Each Time Schedule Pattern can provide up to 20 time intervals which are assigned to one of the eight day/night modes. The Time Schedule Patterns are then assigned to a day of the week (Monday-Friday, Saturday, Sunday or Holiday).
- **Hierarchical Class of Service Control**
  ARS allows or denies call route choices based on an extension's ARS Class of Service. This allows lower Classes of Service (e.g., 1) to access routes unavailable to higher Classes of Service (e.g., 16). The UX5000 provides up to 16 (0=unrestricted, 1~16) ARS Classes of Service.
- **Separate Routing for Selected Call Types**
  To provide unique control, you can program separate routing instructions for:
  - Directory assistance (411, 1411 and 555) calls
  - Emergency (911) calls
- **Separate Routing for Equal Access (1010XXX) Calls**
  Choose different routing for directly-dialed (1010XXX + 1) and operator-assisted (1010XXX + 0) Equal Access calls.

Basic ARS Operation

When a user places an outside call, ARS analyzes the digits dialed and assigns one of 64 Selection Numbers to the call. The Selection Number chosen depends on which digits the user dialed. ARS then checks the time of day, the day of week and the extension's ARS Class of Service. Based on these call routing options, ARS selects a trunk group for the call and imposes the Dial Treatment instructions (if any).

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

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

Class of Service Matching

With the ARS Class of Service Match Access feature, you can determine whether the UX5000 should allow a call based on the COS assigned to the Dial Analysis Table (Program 26-02). This change can be used to create a tenant-like application. It will then use the trunk group defined in the Additional Entry in Program 26-02-03 to place the outgoing call.

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

For this feature the UX5000 option, **Program 26-01-06 : Automatic Route Selection Service, COS Match Access**, is used.

The examples below use the following UX5000 programming:

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

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

Program 12-02 for Automatic Night Service Patterns as:

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

Program 26-04 for ARS Class of Service as:

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

Program 26-01-03 for ARS Misdialed Number Handling as: 1 (Warning Tone)

With Program 26-01-06: ARS COS Match Access disabled (set to '0'):

- If at 9:00 AM, each extension dialed '9+(203)926-5400'
  All Extension would use Trunk Group 3
- If at 9:00 AM, each extension dialed '9+(214)262-2000'
  All Extension would use Trunk Group 1
- If at 6:00 PM, each extension dialed '9+(203)926-5400'
  All Extension would use Trunk Group 3

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

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

- If at 9:00 AM, each extension dialed '9+(203)926-5400'
  Extension 301 would use Trunk Group 1
  Extension 302 would use Trunk Group 2
  Extension 401, 402 would use Trunk Group 3

- If at 9:00 AM, each extension dialed '9+(214)262-2000'
  Extension 301 would use Trunk Group 1
  Extension 302 would use Trunk Group 2
  Extension 401, 402 would use Trunk Group 3

- If at 6:00 PM, each extension dialed '9+(203)926-5400'
  Extension 301 would use Trunk Group 1
  Extension 302 would use Trunk Group 2
  Extension 401, 402 would use Trunk Group 3

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

Alternate Carrier Access for ISDN Trunks

An option is available which allows the UX5000 to provide a Transit Network Selection information element for ARS calls using ISDN trunks. This information element identifies a requested transit network. This function is valid only for outbound calls by ISDN trunk.

Local calls do not need Network Selection information since they will be handled by the local exchange carrier (ILEC). If this is the case, ARS is able to distinguish between local and long distance calls and add the Transit Network Selection information element when required.

This option would apply to both PRI and BRI ISDN trunks. If the trunk used in **Program 26-02 : Dial Analysis Table for ARS/LCR** is not an ISDN trunk, this code in the dial treatment will be ignored.

The examples below use the following UX5000 programming:

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

Table No.	Program 26-02-01 Dial	Program 26-02-02 Service Type	Program 26-02-03 Add Data	Program 26-02-04 ARS COS	Program 26-02-05 Dial Treatment
1	203@@@@@@@	1:TRG	1(Group 1)	0	1
2	@@@@@@@@@@	1:TRG	1(Group 1)	0	0

- **Program 26-03-01 for ARS Dial Treatments set as:**
  Table No. 1 - Dial Treatment: A1RE
- **Program 26-11-01 for Transit Network ID Table set as:**
  Table No. 1 - Transit Network ID: 0288

Example: If an extension dialed '9+(203)925-5400', the setup message will contain the Transit Network Selection IE with Network ID 0288.

Example: If an extension dialed '9+(214)262-2000', the setup message will not contain the Transit Network Selection IE.

Background Music

Feature Availability
• Available.

Description

Background Music (BGM) sends music from a customer-provided music source to speakers in keysets. If an extension user activates it, BGM plays whenever the user's extension is idle.

Barge In

Feature Availability
• Available.

Description

Barge In permits an extension user to break into another extension user's established call, including Conference calls. This sets up a Conference-type conversation between the intruding extension and the parties on the initial call. With Barge In, an extension user can get a message through to a busy co-worker right away. If allowed in an extension's Class of Service, multiple users can barge into the same call (up to 32 callers maximum).

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

Silent Monitor on Barge In to Conference

A system-wide option is available which can allow users barging into a conference call to perform a Silent Monitor. The Barge In feature must be programmed for the user, including Program 20-13-10. This program is only available through terminal programming and it is recommended not to change this option unless required.

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 Coverage

Please refer to the Multiple Directory Numbers / Call Coverage (page 144) for information on this feature.

Call Duration Timer

Feature Availability
• Available.

Description

Call Duration Timer lets a keyset user time their trunk calls on the terminal display. This helps users that must keep track of their time on the terminal. For incoming trunk calls, the Call Timer begins as soon as the user answers the call. For outgoing trunk calls, the Call Timer starts about 5 seconds after the user dials the last digit.

Call Forwarding

Feature Availability
• Available.

Description

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

- **Call Forwarding when Busy or Not Answered**
  Calls to the extension forward when busy or not answered.

- **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** (page 81) for more.
- **Personal Answering Machine Emulation**
  Allows the extension to emulate an answering machine. Refer to “Voice Mail” for more.

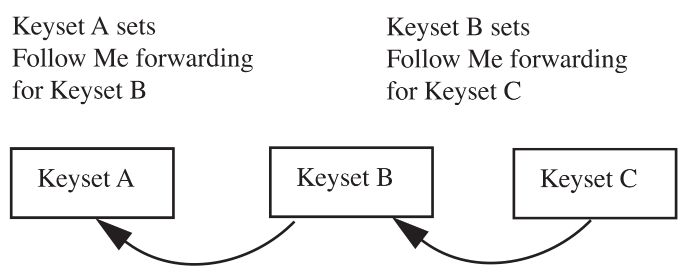
Call Forwarding will reroute calls ringing an extension, including calls transferred from another extension. The extension user must enable Call Forwarding from their terminal. To redirect calls while a user is at another terminal, use “Call Forwarding with Follow Me”.

Activating Call Forwarding While On a Call

A keyset user can activate or deactivate Call Forwarding while on a call if the terminal has a Call Forward Programmable Function Key programmed (15-07-01 or SC: 851 16). Activating Call Forwarding while on a call works only for Call Forward to Station key (Code: 16). Other Call Forward keys (Code: 10-15, 17) do not work while on a call. This option cannot be enabled while the user is hearing a confirmation or warning tone.

This option is not available for single line terminals.

When Call Forward Follow Me is set up in a chain, forwarding cannot be canceled by the middle keyset (as shown in the example below) while on a call.



Call Forwarding, Centrex

Feature Availability
• Available.

Description

With this feature, ***when the UX5000 is using all centrex trunks,*** an extension can be forwarded to a destination using a Centrex/PBX line. (*With this feature set, all forwarded calls will have the line flashed, so the UX5000 must use only Centrex trunks*.) Forwarding can be set for a keyset, SLT or virtual extension user. The Call Forward can be set manually using a service code (Program 11-11-61) or through UX5000 programming (Program 24-08-01, 24-08-02, and 24-08-03). Only Call Forward Immediate, Busy, No Answer, or Busy/No Answer can be used with Centrex lines.

Incoming calls from the COIU blade must be set to DIL (Program 22-02-01) to the extension or a call can be transferred to the extension. Analog lease lines, analog and digital/PRI DID trunks, and SIP trunks cannot be used with this feature.

Direct Inward Line (DIL)

Trunks which are set for DIL to an extension can follow the Call Forwarding with Centrex setting. With no answer, the timer in Program 24-02-03 is followed to determine when the call should be forwarded. When the timer expires, the UX5000 answers the call and sends a hookflash to the trunk. After the timer in Program 21-01-06 expires, the UX5000 dials the forwarded number. Then, after the timer in Program 21-01-06 expires again, the trunk goes on hook, completing the forward.

Transfer

If an extension user transfers a trunk call to an extension with Call Forwarding with Centrex set, a hookflash and the destination digits are set to the Centrex trunk. The transferring extension using will hear ringback tone from the CO and must hang up in order to complete the transfer. A call transferred over a Centrex trunk cannot be retrieved. The UX5000 will follow the timers in Program 24-02-03 and 21-01-06 for when to transfer and the pause to be inserted before dialing.

When transferring a call to an extension which has set Call Forward with Centrex, forward will not complete until the transferring extension goes on hook. This means that with an Automated Attendant transferring a call, Unscreened Transfer must be used.

When a call is transferred to an extension which has the Call Forwarding with Centrex set, only Immediate and Busy forwarding is available and the transfer must be an unscreened transfer in order to follow the forwarding.

Call Forwarding, Fixed

Feature Availability
• Available.

Description

Fixed Call Forwarding is a type of forwarding that is *permanently* in force at an extension. Calls to an extension with Fixed Call Forwarding enabled automatically reroute - without any user action. Unlike normal Call Forwarding (which is turned on and off by extension users), Fixed Call Forwarding is set by the administrator in UX5000 programming. Fixed Call Forwarding complements Voice Mail, for example. The administrator can program Fixed Call Forwarding to send a user's unanswered calls to their Voice Mail mailbox. Each individual user no longer has to manually set this operation.

In UX5000 programming, the administrator can set the Fixed Call Forwarding destination and type for each extension and virtual extension. The forwarding destination can be an on- or off-premise extension or Voice Mail. The Fixed Call Forwarding types are:

- Fixed Call Forwarding with Both Ringing (Program 24-06 Option 1)
- Fixed Call Forwarding when Unanswered (Program 24-06 Option 2)
- Fixed Call Forwarding Immediate (Program 24-06 Option 3)
- Fixed Call Forwarding when Busy or Unanswered (Program 24-06 Option 4)
- Fixed Call Forwarding Off-Premise (Program 24-07)

Fixed Call Forwarding reroutes the following types of incoming calls:

- Ringing intercom calls from co-worker's extensions
- Calls routed from the Voice Mail
- Direct Inward Lines
- DISA, DID and tie line calls to the forwarded extension
- Transferred calls

Fixed Call Forwarding Chaining

Fixed Call Forward Chaining allows Fixed Call Forwards to loop from one extension to the next. For example, you could have the chain 301 ☞ 302 ☞ 303 ☞ 304 set up for Fixed Call Forwarding when Busy. If extension 301 is busy, calls to 301 route to 302. If 302 is also busy, the calls route to 303 and so on. Chaining allows you to set up very basic hunting between co-workers.

Keep the following in mind when setting up Fixed Call Forwarding Chaining:

- If Fixed Call Forwarding Chaining forms a complete Call Forwarding loop (i.e., 301 ☞ 302 ☞ 303 ☞ 301), the UX5000 rings the last extension in the chain (303). It does not complete the loop.
- If Fixed Call Forwarding Chaining finds an extension with user-implemented Call Forwarding in the middle of a chain, it rings that extension. It does not continue routing to the other extensions in the chain.
- If one of the extensions in a Fixed Call Forwarding chain has its fixed option set for Both Ringing (1), the UX5000 rings that extension. It does not continue routing to the other extensions in the chain.
- If **Program 24-07-01 : Fixed Call Forwarding Off-Premise** is used to set the off-premise forwarding, Call Forward Chaining will not work to call the outside number. The option must be set manually (*4 + 6 + trunk access code + outside number + HOLD). It will, however, ring the extension which is forwarded off-premise. For example, Extension 302 is forwarded off-premise (using Fixed Call Forward program 24-07-01) and Extension 305 is forwarded to extension 302. Any calls to 305 will ring extension 302, but not call the outside number programmed in 24-07-01 for the extension.

  This prevents chain forwarding for virtual extensions which are forwarded off-premise as they cannot be programmed manually.
- The receiving extension's display shows:

STA AAA	AAA is the extension that initially placed the call.
TRANSFER<< STA BBB	BBB is the first extension in the Fixed Call Forwarding chain.

3

Call Forwarding, Off-Premise

Feature Availability
• Available.
• DSL sets can be used.

Description

Off-Premise (OPX) Call Forwarding allows an extension user to forward their calls to an off-site location. By enabling OPX Call Forwarding, the user can stay in touch by having the UX5000 forward their calls while they are away from the office. The forwarding destination can be any phone number the user enters, such as a car phone, home office, hotel or meeting room. Off-Premise Call Forwarding can route the off-site phone number over a specific trunk or through a trunk group, Automatic Route Selection or Trunk Group Routing.

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

- Ringing intercom calls from co-worker's extensions
- Calls routed from the Voice Mail [1]
- Direct Inward Lines [1]
- DISA, DID and tie line calls to the forwarded extension [1]
- Transferred calls [1]

OPX Call Forwarding does not reroute Call Coverage keys, Multiple Directory Number keys, or Ring Group calls (i.e., trunk ringing according to Ring Group assignments made in Programs 22-04 and 22-05). Ring Group calls may be forwarded off-premise using voice mail's Call Routing, a dial action table can be created to forward calls to an outside number or Abbreviated Dial number (enter #2001PP with 001 being the # to which the call is forwarded) on timeout using UTRF.

Off-Premise Call Forward for Door Boxes

Off-Premise Call Forwarding allows Door Box callers to be transferred automatically to the pre-programmed external party. The destination telephone number is stored in the Common Abbreviated Dial area. This feature may be used in case a co-worker is out of the office. All incoming calls for their extension will be automatically transferred to their external number (example: cell phone). Off-Premise Call Forward for Door Boxes can be transferred to the external party through ***ISDN lines only***.

Trunk-to-Trunk Off-Premise Call Forwarding

Use Trunk-to-Trunk Forwarding to automatically forward an incoming trunk call to an outside location. The forwarding destination can be stored in an Abbreviated Dial bin. This feature can be used for trunks which are defined as normal (0) or DID (3) in Program 22-02 : Incoming Call Trunk Setup.

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

Call Forwarding with Follow Me

Feature Availability
• Available.

Description

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

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

Call Forwarding/Do Not Disturb Override

Feature Availability
• Available.

Description

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

Call Pickup Group

Please refer to Group Call Pickup (page 114) for information on this feature.

Call Redirect

Feature Availability
• Available.

Description

Call Redirect allows a keyset user to transfer a call to a pre-defined 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 (ringing calls only)

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

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

Call Waiting / Camp On

Feature Availability
• Available.

Description

With Call Waiting, an extension user may call a busy extension and wait in line (Camp-On) without hanging up. When the user Camps-On, the UX5000 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 line for a busy extension without being forgotten.

Callback

Feature Availability
• Available.

Description

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

The UX5000 processes Callback requests as follows:

1. Caller at extension A leaves a Callback at extension B.

   Caller can place or answer additional calls in the mean time.

2. When extension B becomes idle, the UX5000 rings extension A. This is the Callback ring.

3. Once caller A answers the Callback ring, the UX5000 rings (formerly busy) extension B.
   If caller A doesn't answer the Callback ring, the UX5000 cancels the Callback.
4. As soon as caller B answers, the UX5000 sets up an Intercom call between A and B.

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

When leaving a Callback request for an extension which does not answer, the Callback will not be processed until the user at the destination extension lifts/replaces their handset.

Caller ID

Feature Availability
• Available - 64 resources available on the CPU for Caller ID (also used for DTMF receivers and Call Progress Tone Detection.

3

Description

Caller ID allows a display keyset to show an incoming caller's telephone number (called the Directory Number or DN) and optional name. The Caller ID information is available as pre-answer display. With the pre-answer display, the user previews the caller's number before picking up the ringing line. If Caller ID is enabled in the UX5000, a user can check the Caller ID for a parked call using their Park key and possibly the Flash key (depending on programming). Refer to the table on the following page for the available Caller ID displays.

Second Call Display

While busy on a call, the terminal display can show the identity of an incoming trunk or Intercom call. For incoming trunk calls, the display will show the Caller ID or ANI data or the trunk's name if Caller ID or ANI are not installed. (See the T1 Trunking feature on page 175 for more on ANI compatibility.) For incoming Intercom calls, the display will show the calling extension's name.

Caller ID supports the telco's Called Number Identification (CNI) and Called Number Delivery (CND) service, when available. These services provide the Caller ID information (i.e., messages) between the first and second ring burst of an incoming call. There are two types of Caller ID message formats currently available: Single Message Format and Multiple Message Format. With Single Message Format, the telco sends only the caller's phone number (DN). The DN is either 7 or 10 digits long. In Multiple Message Format, the telco sends the DN and the caller's name. The DN for this format is also 7 or 10 digits long, and the name provided consists of up to 15 ASCII characters.

Terminal's display can show up to 12 Caller ID digits (for non-ACD calls).

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

- Ring Group calls
- Calls transferred from another extension
- Calls transferred from Voice Mail (unscreened)
- Direct Inward Lines (DILs)

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

Temporary Memory/Caller ID History

An unanswered call will cause the Call History key (PGM 15-07 or SC 851: 08) to flash, indicating a new call has been placed in the temporary memory. If enabled in programming, the terminal's display will show "CHECK LIST".

This Caller ID data from the temporary memory can be saved in Abbreviated Dial bins, One-Touch bin numbers, or in the Central Telephone Book making them available for placing future calls.

The UX5000 provides the ability to add or delete Caller ID digits to a number within the Caller ID History. This feature is available to any display keyset user which has Caller ID enabled (Programs 15-02-15 and 20-09-02) and has a Caller ID Log History key (Program 15-07-01, code 08).

In addition, this feature can be set up to provide an Add/Delete option for area codes within the Caller ID History.

Caller ID Displays

Abbreviation	Description
Absence code	Absence Reason Code P displays as PRIVATE Absence Reason Code O displays as NO CALLER INFO
CID-num CID-name Trunk name	Caller ID number (Provided by telco) Caller ID name (Provided by telco) Trunk name provided by UX5000 (Program 14-01-01)
NN:NN:NN HH:MM:SS YY:MM:DD	UX5000's Call Timer display UX5000 Time UX5000 Date

Conditions	Row	Pre-Answer Display	Post-Answer Display	Display When Reviewing
With Caller ID name and number	1	Trunk name	Trunk name	CID-name
	2	CID-num CID-name	CID-num CID-name	YY:MM:DD HH:MM:SS
With Caller ID number Without Caller ID name With name absence code	1	Trunk name	Trunk name	CID-num
	2	CID-num	CID-num	YY:MM:DD HH:MM:SS
Without Caller ID number With Caller ID name With number absence code	1	Trunk name	Trunk name	CID-name
	2	CID-name	CID-name	YY:MM:DD HH:MM:SS
Without Caller ID number Without Caller ID name With number & name absence codes	1	Trunk name	Trunk name	No Entry
	2	Name Absence Code	Name Absence Code	
Without Caller ID number Without Caller ID name Without number absence code	1	Trunk name	Trunk name	No Entry
	2	Ringing	Ringing	

Outputting Caller ID Data

The UX5000 includes the Caller ID data on the SMDR report. The report provides the incoming call's DN in the DIALED NUMBER field. The CLASS field shows PIN (just like all other incoming calls).

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

Display Reason for No Caller ID Information
With Caller ID enabled, the UX5000 will provide information for analog calls that do not detect the Caller ID information. If the Caller ID information is restricted, the terminal display will show “PRIVATE”. If the UX5000 is not able to provide Caller ID information because telco information is not detected, then the display will show “NO CALLER INFO”.

Calling Party Number Information
When using the IP DECT terminal, the UX5000 can provide the Caller ID information for an external call if it is provided by the telco.

Option to Enable Caller ID Name for SLT
UX5000 programming provides an option for single line terminals to display Caller ID.

Caller ID Sender Queuing
The UX5000 can provide Caller ID (calling party number) to a single line terminal which has a display. The UX5000 can queue the incoming call to the single line terminal if the UX5000 Caller ID sender resources are busy using Program 20-19-05.

While an incoming call is waiting in queue, if the SLT user lifts their handset, they will hear silence (no dial tone) and can not dial out. When the SLT user goes back on hook, the UX5000 immediately sends the queued call to the SLT without Caller ID.

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

Flexible Ringing by Caller ID
The Flexible Ringing by Caller ID feature provides several different options for rerouting calls based on the Caller ID received.

- **Reject/Reroute "Private" Caller ID Calls**
  When an analog or ISDN trunk call is received with “Private” Caller ID information, the UX5000 can reject the call by playing a VRS message or it can route the call to an alternative extension or incoming ring group programmed in Program 22-18-01.

- **Reject/Reroute Based on Entry in ABB Table**
  When an analog, ISDN or IP trunk call is received with regular Caller ID information, the UX5000 can reject the call by playing a VRS message if the Caller ID number matches the ABB group number programmed in Program 22-16-01 and ABB entry in Programs 13-02-01 and 13-04-01.

  The analog, ISDN or IP trunk call can also be routed to an alternative extension or incoming ring group if the Caller ID number matches the common ABB table (Program 13-04).

This option can block calls on all trunks or it can be set on a per-trunk basis.

Refer to **Central Office Calls, Answering** (page 86) for the required programming for this feature.

Central Office Calls, Answering

Feature Availability
• Available - Capacity depends on the number of blades installed and the system port licensing (max: = 200 trunks).

Description

The UX5000 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 UX5000 extensions. For additional information on making trunk ring, refer to the Ring Group feature.

Delayed Ringing

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

Clear Down / Automatic Release

When on a speakerphone call only, when an outside caller hangs up, the UX5000 will return the keyset to an idle condition. The user does not have to press the SPK key to hang up. This feature is automatic and no programming is required. Refer to the **Automatic Release** (page 71) feature for details.

Codec Filter Data Setup Program

When **Program 81-07-01 : Codec Filter Setup for Analog Trunk Ports** is set to "4 - Specified Data", the UX5000 will use the settings in **Program 81-09 : COIU Codec Filter Data Setup**.

These values should not be changed from their default settings unless directed by NEC'S Technical Service department.

The side tone of the COIU is adjusted using all 16 values, however, special software is required in order to compute these values. The setting is not proportional to the gain level. To change these values, contact NEC's Technical Service department for assistance.

Display Reason for Transfer

When incoming DID, DISA, DIL or ISDN calls are transferred to another extension or ring group due to a Call Forward or DND setting, the reason for the transfer can be displayed on the terminal receiving the transferred call. The extension user can then recognize why they are receiving the call. This feature requires a display terminal in order to view the message.

Flexible Ringing by Caller ID

The Flexible Ringing by Caller ID feature provides several different options for rerouting calls based on the Caller ID received.

- **Reject/Reroute "Private" Caller ID Calls**
  When an analog or ISDN trunk call is received with "Private" Caller ID information, the UX5000 can reject the call by playing a VRS message or it can route the call to an alternative extension or incoming ring group programmed in Program 22-18-01.

- **Reject/Reroute Based on Entry in ABB Table**
  When an analog, ISDN or IP trunk call is received with regular Caller ID information, the UX5000 can reject the call by playing a VRS message if the Caller ID number matches the

ABB group number programmed in Program 22-16-01 and ABB entry in Programs 13-02-01 and 13-04-01.

The analog, ISDN or IP trunk call can also be routed to an alternative extension or incoming ring group if the Caller ID number matches the common or group ABB table (Program 13-04).

This option can block calls on all trunks or it can be set on a per-trunk basis.

Programming Examples for Flexible Ringing by Caller ID:

- To refuse the "Private" Caller ID incoming call:
  Program 14-01-27: 1 (reject)
  Program 20-07-24: 1 (Enable for COS)
  Program 22-18-01: 0 (no transfer)
  Program 40-10-06: 2 (VRS message 2)
  Then turn on the Private Call Refuse mode using the service code (Program 11-10-32) or Programmable Function Key (code 86).

- To transfer the "Private" Caller ID incoming call to extension 301 as ring pattern 2:
  Program 14-01-27: 1 (reject)
  Program 22-18-01: 1 (extension number)
  Program 22-18-02: 301 (extension 301)
  Program 22-18-03: 2 (ring pattern 2)
  Then turn on the Private Call Refuse mode using the service code (Program 11-10-32) or Programmable Function Key (code 86).

- To transfer the "Private" Caller ID incoming call to incoming ring group 2 as ring pattern 3:
  Program 14-01-27: 1 (reject)
  Program 22-18-01: 2 (incoming ring group)
  Program 22-18-02: 2 (group 2)
  Program 22-18-03: 3 (ring pattern 3)
  Then turn on the Private Call Refuse mode using the service code (Program 11-10-32) or Programmable Function Key (code 86).

- To reject the call with "2039261111" Caller ID incoming call:
  Program 14-01-27: 1 (reject)
  Program 20-07-25: 1 (Enable for COS)
  Program 22-16: 64 (ABB group 64)
  Program 13-02; Group 64: 1000 - 1099
  Program 13-04-01; Table 1000: 2039261111
  Then turn on the Caller ID Refuse mode using the service code (Program 11-10-34) or Programmable Function Key (code 87).

- To transfer the call with "2039261111" Caller ID incoming call to extension 301 as ring pattern 1:
  Program 13-04-01: 2039261111
  Program 13-04-03: 1 (extension number)
  Program 13-04-04: 301 (extension 301)
  Program 13-04-05: 1 (tone pattern 1)

- To transfer the call with "2039261111" Caller ID incoming call to incoming ring group 2 as ring pattern 2:
  Program 13-04-01: 2039261111
  Program 13-04-03: 2 (incoming ring group)
  Program 13-04-04: 2 (group 2)
  Program 13-04-05: 2 (tone pattern 2)

3

Notes:

- Caller ID Matching
  The UX5000 compares the Caller ID and programmed ABB dial and allows/denies as indicated below.

Caller ID	ABB Dial	Result
2039261111	2039261111	Matched
2039261111	20392611119	Matched
2039261111	203	Matched
9261111	2039261111	Unmatched
2039261111	9261111	Unmatched

- The ABB dial table is searched from the starting number and the first match result is used.
- The maximum number of VRS message channels that can be used simultaneously is 16. These channels are also shared with the voice mail.
- With ISDN trunks, the unrestricted digital call can not use this feature.
- This feature does not work with incoming trunk calls via networking (from the other system). In this case, the refuse/routing program must be programmed in the system that has those trunks. Routing to the other system's extension is available.
- When Program 13-04 is used, it will override the setting in Program 22-02-01 : Incoming Call Trunk Setup.
- Program 13-04 will follow Common or Group Abbreviated Dial numbers.

Sidetone Volume Setup

An option is available to allow the change of the keyset side tone volume. There are two levels, based on whether the connected trunk is a digital trunk or analog trunk.

Universal Answer

Universal Answer allows an employee to answer a call by going to any keyset and dialing a unique Universal Answer code. The employee doesn't have to know the trunk number or dial any other codes to pick up the ringing trunk. You'll normally set up Universal Answer along with Universal Night Answer (see "Night Service"). When a Universal Night Answer call rings the External Paging, an employee can answer the call from the first available terminal. You might also want to use Universal Answer in a noisy warehouse or machine shop where the volume of normal terminal ringing is not adequate. After hearing the ringing over the Paging, an employee can then easily pick up the call from a shop terminal. See "Night Service" for more on Universal Night Answer.

The Automatic Answer of Universal Answer Calls option (Program 20-10-07) determines whether or not the extension has the Auto Answer feature for ringing calls. This option allows a user to simply lift the handset to answer a ringing call; they no longer need to dial the service code.

Central Office Calls, Placing

Feature Availability
• Available - Capacity depends on the number of blades installed and the system port licensing (max: = 200 trunks).

Description

The UX5000 provides flexibility in the way each extension user can place outgoing trunk calls. This lets you customize the call placing options to meet site requirements and each individual's needs. A user can place a call by:

- Pressing Line Keys or "Loop Keys"
- Pressing a Trunk Group (i.e., loop) 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 (804 + group number)
- Dialing a code for Trunk Group Routing or ARS (9)
- Dialing an Alternate Trunk Route Access Code (which you must define)

Trunk Port Disable

The UX5000 provides a service code (Default: 145) which can be used by an extension user to block a trunk for outgoing calls. The user which busied out the trunk will still have access to it. All other users will be blocked from seizing it to place an outgoing call. The trunk, however, can still be answered by any users programmed with the trunk access.

Clear Down / Automatic Release

When on a speakerphone call only, when an outside caller hangs up, the UX5000 will return the keyset to an idle condition. The user does not have to press the SPK key to hang up. This feature is automatic and no programming is required. Refer to the **Automatic Release** (page 71) feature for details.

Codec Filter Data Setup Program

When **Program 81-07-01 : Codec Filter Setup for Analog Trunk Ports** is set to "4 - Specified Data", the UX5000 will use the settings in **Program 81-09 : COIU Codec Filter Data Setup**.

These values should not be changed from their default settings unless directed by NEC'S Technical Service department.

The side tone of the COIU is adjusted using all 16 values, however, special software is required in order to compute these values. The setting is not proportional to the gain level. To change these values, contact NEC's Technical Service department for assistance.

Sidetone Volume Setup

An option is available to allow the change of the keyset side tone volume. There are two levels, based on whether the connected trunk is a digital trunk or analog trunk.

3

Central Telephone Book

Feature Availability
• Available.

Description

The Central Telephone Book feature provides each display keyset user up to two personal telephone books which can be used to store numbers. The UX5000 allows multiple extensions to be assigned the same telephone book - this allows users to share commonly used numbers.

- Up to 100 books of 300 entries can be stored.
- The Telephone Books can be password protected.
- Numbers can be stored using Last Number Dialed and Caller ID

The following key operation can be used with the Telephone Book:

- HOLD button - User can enter the data. (same as the Abbreviated Dial input)
- DIAL button - User can input the character: @ (same as the Abbreviated Dial input)
- MIC button - User can input the character: P (same as the Abbreviated Dial input)
- FLASH button - User can change the input mode or input the character: R. (same as the Abbreviated Dial input)
- EXIT button - User can finish the telephone book operation.
- CONF button - User can delete the characters (same as the Abbreviated Dial input) or the user returns back one page.
- VOL UP button/Navigation Pad VOL UP - User can change the edit item or page OR the user can start a search.
- VOL DOWN button/Navigation Pad VOL DOWN - User can change the edit item or page OR the user can start a search.
- SOFT KEY button - User can select the function.
- Dial pad - User can input the data or select list.
- Navigation Pad Right Arrow - User can access the menu for Central Telephone Book or move the cursor to the right.
- Navigation Pad Left Arrow - User can move the cursor to the left.

Class of Service

Feature Availability
• Available - 15 Classes of Service.

Description

Class of Service (COS) sets various features and dialing options (called items) for extensions. The UX5000 allows any number of extensions to share the same Class of Service. An extension can have a different Class of Service for each of the Night Service modes. This lets you program a different set of dialing options for daytime operation, nighttime operation and even during lunch breaks. An extension's Class of Service can be changed in UX5000 programming or via a Service Code (normally 177).

A Class of Service program is available for some of the timers which affect only the extensions and trunks which are assigned to use the timer Class of Service. Each timer Class of Service has 15 different classes (01-15), plus the option to use the system-wide timer (0).

Computer Telephony Integration (CTI) Applications

Feature Availability
• Available.

Description

Computer Telephony Integration (CTI) applications automate your office with TAPI compatibility and external PC control. CTI puts your communications server on the cutting edge of modern office productivity with:

Database Lookup
Provided through Caller ID and TAPI-compatible third-party software (such as ACT!), Database Lookup displays your caller's account information before their call is even answered.

3

Conference

Feature Availability
• Available - CCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.

Description

Conference lets an extension user add additional inside and outside callers to their conversation. With Conference, a user may set up a multiple-party telephone meeting without leaving the office. The UX5000 CCPU provides 2 blocks of 32 Conference circuits, allowing each block to have any number of internal or external parties conferenced up to the block's limit of 32. This means that one extension can Conference up to 31 internal and/or external parties together (the originator would be the 32nd party reaching the maximum of 32). While this Conference call is active, another user can use the second block of Conference circuits to make the same type of call.

Each block of Conference circuits can have multiple Conference calls, providing there are Conference circuits available. It is not restricted to one Conference per block.

With the Conference feature, a user can:
- Add another party while in an active Conference call.
- Break up a Conference call by pressing Hold.
- Drop a trunk or trunks from the Conference.

Split (From Conference)

Split allows a user to alternate (i.e., switch) between their callers in Conference. This will allow a dispatcher, for example, to control a telephone meeting between themselves, a customer and a service technician. The dispatcher can meet together with all parties, privately set up a service strategy with the technician and then meet again to set the schedule.

Split cycles through the Conference in the same order in which the Conference was initially set up. If a user places an outside call, conferences extension 302 followed by extension 303, Split will cycle from the trunk, to 302 and finally to 303. The Split cycle then repeats.

Barge Into Conference

If a user's extension has Barge In capability enabled, they can also Barge In on an established Conference. This permits, for example, an attendant or supervisor to join a Conference in an emergency. It also allows a co-worker to leave a conference -- and then rejoin the telephone meeting when it is convenient to do so.

Automatic Conference on Hang Up

A Class of Service option is available which will allow or deny an extension user from automatically setting up a Conference/Tandem Trunking call upon hanging up the terminal.

Transfer Call Into Conference

An extension with Barge In capability can Transfer a call into an existing Conference. 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.

Change CONF to a Transfer Key

An option is available which allows an extension's CONF key to be programmed for Conference or for Transfer. When set for Transfer, the user places a call on hold, dials the extension to which it should be transferred, the user presses the CONF key. The call is then transferred. When set for Conference, with an active call, the user presses the CONF key, places a second call, then presses the CONF key twice. All the calls are then connected.

Conference, Remote

Feature Availability
• Available - CCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.

Description

The Remote Conference feature enhances the Conference feature by allowing outside parties to dial a Remote Conference pilot number and a Conference Group number to connect to a Conference call.

The conference circuits on the CCPU are used to join each party to the conference. A maximum of 32 conference participants is possible for one Conference. However, the conference call cannot be split over the CCPU's conference blocks. This could limit the number of participants if other conference circuits are in use.

A maximum of 4 simultaneous Remote Conference calls is possible. The conference call is password protected so that any user joining the conference would be required to enter a password before being connected.

- One terminal or trunk needs one conference channel to participate in the conference.
- The conference call cannot be split over the CPU's two conference blocks, limiting the number of conference participants to 32.

Conference, Voice Call/Privacy Release

Feature Availability
• Available - CCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.

Description

Voice Call Conference lets extension user's in the same work area join in a trunk Conference. To initiate a Voice Call Conference, an extension user just presses the Voice Call Conference key and tells their co-workers to join the call. The UX5000 releases the privacy on the trunk, and other users can just press the trunk's line key to join the call.

Voice Call Conference does not use the UX5000 features to announce the call. The person initiating the Voice Call Conference just announces it "through the air."

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. This would help a workgroup supervisor, for example, that needed to quickly monitor any group member's call. If the supervisor wanted to make a "secure" call, however, they could quickly switch the line's mode and be assured that their call would not be monitored. If the outside call is on a line key, the user just presses the line key to switch modes. If the call is on a loop key, the user presses their Privacy Release function key instead.

For UX5000s using the Privacy Mode Toggle option, trunks initially have the privacy released. If privacy is desired for a trunk, use the toggle option or press the Privacy Release function key to switch modes.

Continued Dialing

Feature Availability
• Available.

Description

Continued Dialing allows an extension user to dial a call, wait for the called party to answer and then dial additional digits. This helps users that need services like Voice Mail, automatic banking and Other Common Carriers (OCCs).

There are two types of Continued Dialing:

- **Continued Dialing for Intercom Calls**
  Depending on an extension's Class of Service, a keyset user may be able to dial additional digits after their Intercom call connects. In UX5000s 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 UX5000's Toll Restriction will cut off the call after a specific number of dialed digits. See Programming below for additional information.

NOTICE
Continued Dialing may make the UX5000 more susceptible to toll fraud.

Cordless DECT Terminals

Feature Availability
• Available.

Description

The UX5000 supports a Cordless DECT terminal. The DTL-8R-1 Cordless DECT terminal is a 1.9G DECT 6.0 digital cordless terminal that provides mobility, flexibility and convenience for those who spend much of the work day away from their desk. Supporting up to 6 Repeaters allows for further increase of the range and mobility of the cordless handset. Fully integrated with the communications server, the DTL-8R-1 Cordless DECT offers many standard features such as Park, Do Not Disturb, Hotline, Voice Over and Voice Mail. Normally paired with a companion keyset for improved 1-button call coverage capabilities, the DTL-8R-1 Cordless DECT will also work as a stand-alone terminal.

This terminal provides 5 channels which will change automatically to find the best reception for a call.

Complemented by 4 fully programmable function keys (with LEDs) and 4 one-touch keys, the DTL-8R-1 Cordless DECT achieves a whole new level of convenience and mobility. An easy-to-read LCD display, volume controls, a rechargeable nickel-metal hydride battery pack and a handy belt clip round out the elegant and affordable DTL-8R-1 Cordless DECT Phone.

The base unit provides a “Cordless” and “Desk” button which allows the one extension port to switch from the Cordless DECT to an attached UX5000 keyset. The terminals must be in an idle state when the button is pressed, otherwise any active call will be disconnected.

Privacy

Cordless telephones are radio devices. Communications between the handset and base of the cordless telephone are accomplished by means of radio waves which are broadcast over the open airways. Because of the inherent physical properties of radio waves, communication can be received by radio receiving devices other than your own telephone unit, consequently, any communications using the cordless telephone may not be private.

Refer to the Dterm Cordless DECT Owner’s Manual (P/N INT-2070 (DECT)) for installation and setup information.

3

Department Calling

Feature Availability
• Available - 64 Department Groups.

Description

With Department Calling, an extension user can call an idle extension within a preprogrammed Department Group by dialing the group's pilot number. The call would ring the first available extension in the group. For example, this would let a caller dial the Sales department just by knowing the Sales department's pilot number. The caller would not have to know any of the Sales department's extension numbers.

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 (with circular routing, a new call will ring the extension which has been idle the longest in the group). The ringing order is based on an extension's priority. In a Department Group with extensions 310 (Priority 1), 311 (Priority 2) and 312 (Priority 3)

- The first call rings 310.
- The second call rings 311.
- The third call rings 312.
- The fourth call rings 310 and the cycle repeats.

Note: When programming, the high priority extensions have low priority numbers. For example, priority 1 has a higher priority than priority 10.

Enhanced Hunting

Department Calling provides expanded hunting capabilities. Hunting sets the conditions under which calls to a Department Group pilot number will cycle through the members of the group. The hunting choices are:

- **Busy**
  A call to the pilot number will hunt past a busy group member to the first available extension.
- **Not answered**
  A call to the pilot number will cycle through the idle members of a Department Calling group. The call will continue to cycle until it is answered or the calling party hangs up. If the Department Group has Priority Routing enabled, and the highest priority member is busy, the call will not route.
- **Busy or not answered**
  A call to the pilot number will cycle through the idle members of a Department Calling group. The call will continue to cycle until it is answered or the calling party hangs up. Calls into groups with Priority Routing and Circular Routing route identically.
- **Simultaneous ringing**
  All idle members of the Department Group ring simultaneously. Calls do not cycle between group members.

If all members of the Department Group are busy, an incoming or transferred call to the group's pilot number will queue for an available member. Each group has a queue that can hold any number of waiting calls. If a display terminal is waiting in queue, the user will see: *WAITING (group name)*. If a transferred call in queue is an outside call, and the UX5000 has DSP daughter board installed with the VRS, the queued caller will hear, *"Please hold on. All lines are busy. Your call will be answered when a line becomes free."*

The UX5000 prevents hunting to a Department Group extension if it is:

- Busy on a call
- In Do Not Disturb
- Call Forwarded

Overflow Routing

Department Calling also provides overflow routing for extensions within the group. If a user directly dials a busy extension within a Department Group, the UX5000 can optionally route the call to the first available group member.

User Log Out/Log In

An extension user can log out and log in to a Department Calling Group. By logging out, the user removes their extension from the group. Once logged out, Department Calling bypasses their extension. When they log back in, Department Calling routes to their extension normally. All users can dial a code to log in or log out of their Department Calling Group. A keyset can optionally have a function key programmed for one-button log in and log out operation.

3

Call Restriction Between Groups Added

An option is available to prevent calling between certain Department Groups. This restricts calls to the extension numbers as well as the Department Group number. After calling a restricted extension number, the user will hear a warning tone or a message "The number you have dialed is not in service." if the VRS is installed.

An extension user in one Department Group, however, can use an extension in a restricted Department Group as the destination extension when using the Call Forwarding feature.

This restriction option does not apply to secondary Department Groups (defined in Program 16-03-01).

Department Step Calling

Feature Availability
• Available.

Description

After calling a busy Department Calling Group member, an extension user can have Department Step Calling quickly call another member in the group. The caller does not have to hang up and place another Intercom call if the first extension called is unavailable. Department Step Calling also allows an extension user to cycle through the members of a Department Group.

Dial Number Preview

Feature Availability
• Available.

Description

Dialing Number Preview lets a display keyset user dial and review a number before the UX5000 dials it out. Dialing Number Preview helps the user avoid dialing errors.

Dial Pad Confirmation Tone

Feature Availability
• Available.

Description

For an extension with Dial Pad Confirmation Tone enabled, the user hears a beep each time they press a key. This is helpful for Intercom calls and Dial Pulse trunk calls, since these calls provide no Call Progress tones.

Dial Tone Detection

Feature Availability
• Available.

Description

If a trunk has Dial Tone Detection enabled, the UX5000 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's number), the UX5000 will drop the trunk if dial tone does not occur. If the user access the trunk via a Trunk Group (by dialing a trunk group code or automatically through a feature like Last Number Redial), the UX5000 can drop the trunk or optionally skip to the next trunk in the group. Refer to the chart under Programming below for more.

Dial Tone Detection is available for the following features:

- Automatic Route Selection
- Abbreviated Dialing
- Central Office Calls, Placing
- Last Number Redial
- Loop Keys (outbound)
- Save Number Dialed

- T1 Trunking (with ANI/DNIS Compatibility)
- Tie Lines
- Trunk Group Routing
- Trunk Groups

Direct Inward Dialing (DID)

Feature Availability
• Available - 20 DID Translation Tables can be divided between 2000 entries.

Description

Direct Inward Dialing (DID) lets outside callers directly dial UX5000 extensions. DID saves time for callers who know the extension number they wish to reach. To place a DID call, the outside caller dials the local exchange (NNX) and additional digits to ring the UX5000 terminal 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.

Note: Direct Inward Dialing requires DID service from telco. DID trunks are used for incoming calls only and are not used for outgoing calls.

In addition to direct dialing of UX5000 extensions, DID provides:

- DID Dialed Number Translation
- Flexible DID Service Compatibility
- DID Intercept
- DID Camp-On
- DID Routing through the VRS
- Delayed DID
- DID Intercept Destination for each DID Number

DID Dialed Number Translation

DID allows different tables for DID number translation. This gives you more flexibility when buying DID service from telco. If you can't buy the exact block of numbers you need (e.g., 301-556), use the translation tables to convert the digits received. For example, a translation table could convert digits 501-756 to extension numbers 301-556.

The UX5000 has 2000 DID Translation Table entries that you can allocate among the 20 DID Translation Tables. There is one translation made in each entry. For a simple installation, you can put all 2000 entries in the same table. For more flexibility, you can optionally distribute the 2000 entries among the 20 tables.

In addition to number conversion, each DID Translation Table entry can have a name assigned to it. When the DID call rings the destination extension, the programmed name displays.

Flexible DID Service Compatibility

You can program the UX5000 to be compatible with three and four digit DID service. With four digit service, the telco sends four digits to the UX5000 for translation. With three digit service, the telco sends three digits to the UX5000 for translation. Be sure to program your UX5000 for compatibility with the provided telco service. For example, if the telco sends four digits, make sure you set up the translation tables to accept the four digits.

The UX5000 is compatible with Dial Pulse (DP) and DTMF DID signaling. DID trunks can be either wink start or immediate start.

DID Intercept

DID Intercept automatically reroutes DID calls under certain conditions. There are three types of DID Intercept:

- **Vacant Number Intercept**
  If a caller dials an extension that does not exist or mis-dials, Vacant Number Intercept can reroute the call to the programmed DID Intercept extension ring group or Voice Mail. Without Vacant Number Intercept, the caller hears error tone after mis-dialing.

- **Busy Intercept**
  Busy Intercept determines DID routing when a DID caller dials a busy extension. If Busy Intercept is enabled, the call immediately routes to the programmed DID Intercept extension ring group or Voice Mail. If Busy Intercept is disabled, the call follows DID Camp-On programming (see below).

- **Ring-No-Answer Intercept**
  Ring-No-Answer Intercept sets the routing options for DID calls that ring unanswered at the destination extension. With Ring-No-Answer Intercept enabled, the unanswered call reroutes to the DID Intercept extension ring group or Voice Mail after the DID Ring-No-Answer Time interval. If Ring-No-Answer Intercept is disabled, the unanswered call rings the destination until the outside caller hangs up.

DID Camp-On

DID Camp-On sets what happens to DID calls to busy extensions when you have Busy Intercept disabled. With DID Camp-On enabled, a call to a busy extension camps-on for the DID Ring No Answer Time interval. It then diverts to the programmed DID Intercept extension ring group or Voice Mail. Without DID Camp-On, the caller to the busy extension just hears busy tone.

Delayed DID

Delayed DID allows a user a pre-programmed amount of time to answer a call. If the call is not answered within this time period, the UX5000 automatically answers the call. An outside party will hear music, or dial tone according to the following conditions:

- If a customer-provided audio system (example: tape recorder) is connected, an error message or music can be played for the caller.
- If there is no equipment connected for an announcement, the UX5000 sends a unique dial tone to the outside caller.

This feature is not available for the normal incoming call on ISDN trunks.

DID Intercept Destination for Each DID Number

With this feature the UX5000 allows you to program a DID Intercept destination for a DID number which receives no answer or busy call. The UX5000 can be programmed to use a trunk ring group, or the voice mail as the programmed destination. Each vacant number intercept for a DID number can have two destinations. The first destination is for an invalid DID number, busy or no answer extension. The second destination is for a no answer trunk ring group.

For busy or no answer intercept calls, a third destination can be defined in Program 22-12. If the first and third destinations are programmed but the second destination is not, the incoming call goes to the third destination after the first destination. If the first and second destinations are not programmed, but the third destination is, the call goes directly to the third destination.

This feature works for DID trunks with a trunk service type 1 in Program 22-02. Other types of trunks may use the DID table, but the DID intercept feature is not supported.

With the DID Intercept for each DID number feature, when the primary destination (Program 22-11-05) is set to Voice Mail, the Voice Mail protocol is:

1. Busy Intercept = Forward Busy
2. Ring-No-Answer Intercept = Forward RNA

When the secondary destination (Program 22-11-06) is set to Voice Mail, the Voice Mail protocol is based on the first destination's routing. When the incoming call is forwarded to the first destination by a busy intercept, the Voice Mail protocol will be that it forwards busy calls. When the incoming call is routed to the first destination by a ring-no-answer intercept, the protocol will be that it forwards ring-no-answer. The Voice Mail will transfer the calls to the mailbox number defined in Program 22-11-02.

Note: Any valid DID number must be entered in the DID table (Program 22-11). If a valid DID number is not entered, there will be no ring destination for any incoming calls to that number (the calls will not ring any extension in the UX5000).

Call by Time Schedule Feature Added

For every DID number programmed, the DID Call by Time Schedule feature allows each of the 8 different time patterns (defined in Program 22-17) to be assigned a different destination, following the DID conversation table in Program 22-11. This time pattern can also be selected manually, using a service code. The time pattern used for this option is separate from the 8 day/night time modes defined in Program 12-02.

Notes:

- When the time pattern is changed manually, the following conditions apply:
  - Manually selecting the time pattern temporarily overrides the entries in Program 22-17. However, when a time pattern changes with the time schedules in Program 22-17, the pattern applied by the manual change is cancelled.
  - If the time pattern is manually selected and the UX5000 is reset, once the UX5000 restarts, the manually selected time pattern is still in affect.
- When a time pattern is set from 00:00 to 00:00 and Program 22-11 is defined, the UX5000 will always follow the setting in Program 22-11. To prevent the UX5000 from using Program 22-11, it must be undefined.
- If the destination defined in Program 22-17 is not found, the UX5000 then checks the setting in Program 22-11.
- If an incoming DID call is unanswered/busy, it will follow the entries in Program 22-12 and 22-13 for the trunk group based on the time the no answer/busy indication was received.

SMDR Includes Dialed Number

The SMDR report can optionally print the trunk’s name (entered in UX5000 programming) or the number the incoming caller dialed (i.e., the dialed DID digits). This gives you the option of analyzing the SMDR report based on the number your callers dial. (This option also applies to an ISDN trunk as well.)

Federal Communications Commission DID Requirements

Allowing this equipment to operate in a manner that does not provide proper answer supervision signaling is in violation of Part 68 rules.

This equipment returns answer supervision to the Public Switched Telephone Network when the DID trunk is:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user
- Routed to a dial prompt

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

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

Direct Inward Line (DIL)

Feature Availability
• Available - Capacity depends on the number of blades installed and the system port licensing. (max. 200 trunks, 64 Department Groups, 256 extensions, and 256 virtual extensions)

Description

A Direct Inward Line (DIL) is a trunk that rings an extension, virtual extension or Department Group directly. Since DILs only ring one extension or group (i.e., the DIL destination), employees always know which calls are for them. For example, a company operator can have a Direct Inward Line for International Sales Information. When outside callers dial the DIL's phone number, the call rings the operator on the International Sales line key. The DIL does not ring other extensions.

DIL Delayed Ringing

Extensions in a Ring Group can have delayed ringing for another extension's 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 a Sales Support department. If the Sales Support calls are not answered, they ring into the Technical Service department.

Direct Inward System Access (DISA)

Feature Availability
• Available - 15 users, 15 DISA Classes of Service.

Description

DISA permits outside callers to directly dial extensions, trunks and selected features. This could help an employee away from the office that wants to directly dial co-workers or use the company's trunks for long distance calls. To use DISA, the employee:

- Dials the telephone number that rings the DISA trunk
- Waits for the DISA trunk to automatically answer with a unique dial tone
- Dials the 6-digit DISA password (access code)

- Waits for a second unique dial tone
- Accesses a trunk, uses a selected feature or dials a extension

DISA calls ring 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 call rings an idle CALL key.

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 Class of Service

DISA Class of Service provides features and dialing restrictions for DISA callers. This allows you to control the capabilities of the DISA callers dialing into your UX5000. When a DISA caller first accesses the UX5000, they must enter a DISA password before proceeding. The UX5000 associates the password entered with a specific user number, which in turn has a Class of Service. If the Class of Service allows the action (such as making outgoing trunk calls), the call goes through. If the DISA Class of Service doesn't allow the action, the UX5000 prevents the call. The DISA Class of Service options are:

- **Trunk Group Routing/ARS Access**
  When a DISA caller dials into the UX5000, they may be able to dial 9 and place outside calls. Any toll charges are incurred by the UX5000. The call follows the UX5000's Trunk Group Access or Automatic Route Selection - whichever is enabled.
- **Trunk Group Access**
  DISA callers may be able to access a specific trunk group for outgoing calls through the UX5000. To access a Trunk Group, the user dials Service Code 804 followed by the Trunk Group number (e.g., 1). This allows the DISA caller to place an outgoing call over the selected group. Trunk Group Access bypasses the UX5000's Trunk Group Routing/ARS/Trunk Access Maps. As with dial 9 access, any toll charges are incurred by the UX5000. Also see Direct Trunk Access below.
- **Common Abbreviated Dialing**
  The UX5000's Common Abbreviated Dialing bins may be available to DISA callers. This could save the DISA caller time when dialing.
- **Operator Calling**
  A DISA caller may be able to dial 0 for the UX5000's operator.
- **Paging**
  Internal and External Paging may be available to DISA callers. This allows co-workers in adjacent facilities, for example, to broadcast announcements to each other.
- **Direct Trunk Access**
  DISA callers may be able to select a specific trunk for outgoing calls through the UX5000. To directly access a trunk, the user dials Service Code #9 followed by the trunk's number (e.g., 001). This allows the DISA caller to place an outgoing call over the selected trunk. Direct Trunk Access bypasses the UX5000's Trunk Group Routing/ARS/Trunk Access Maps. As with dial 9 access, any toll charges are incurred by the UX5000. Also see Trunk Group Access above.
- **Call Forward**
  DISA callers may be able to set Call Forwarding to redirect an extension's calls to another extension. Call Forwarding ensures that the user's calls are covered when they are away from their work area.
- **DISA/Tie Trunk Barge In**
  The DISA/Tie Trunk Barge In option allows a tie line caller to break into another extension's established call. This sets up a three-way conversation between the intruding party and the two parties on the initial call.

DISA Toll Restriction

The digits a DISA caller dials for an outgoing call may be subject to the UX5000's Toll Restriction. For example, Toll Restriction can prevent users from dialing a 1-900 service. When an incoming DISA caller tries to use UX5000 trunks to dial 1-900, Toll Restriction will deny the call.

DISA Operating Modes

The DISA Operating Modes determine what happens when a DISA caller forgets to dial, calls a busy or unanswered extension or dials incorrectly. The UX5000 can either drop the call or send it to a preset Ring Group (called a the DISA Transfer Destination).

Warning Tone for Long DISA Calls

You can set up the UX5000 to provide a warning tone to DISA callers that have been on a call too long. The warning tone can be just a reminder (which the caller can ignore) or can be followed by a forced disconnect of the call. When the DISA caller hears the warning tone, they have the option of dialing a code to continue the conversation or disconnect.

Trunk Disconnect Continue/Disconnect Codes

With **Program 24-02-07 : System Options for Transfer - Forced Release for Trunk-to-Trunk Transfer** or **Program 25-07-08 : System Timers for VRS/DISA - DISA Long Conversation Disconnect** enabled, users can be provided with the option to use a Continue or Disconnect service code. The Continue service code will extend the conversation a programmed length of time. If the user enters the Disconnect service code, the call will be disconnected immediately.

Example:

The following example indicates how a call will be handled with the UX5000 programmed as follows:

- Program 14-01-25: 1
- Program 20-28-01: #
- Program 20-28-02: No setting
- Program 20-28-03: 180
- Program 24-02-07: 600 (Only used with Tandem Trunking)
- Program 24-02-10: 30 (Only used with Tandem Trunking)
- Program 25-07-07: 600 (Only used with DISA)
- Program 25-07-08: 30 (Only used with DISA)

1. An external call connects to an external number (either by transferring with Tandem Trunking or by DISA caller).
2. After 10 minutes (Tandem Trunking = Program 24-02-07 or DISA = Program 25-07-07), a warning tone is heard and the user dials "#" (Program 20-28-01) to extend the conversation.
3. After 3 minutes (Program 20-28-03), the warning tone is heard again. After 30 seconds (Tandem Trunking = Program 24-02-10 or DISA = Program 25-07-08), the call is disconnected.

Remote Feature Setup with DTMF

An option may be available which can be used to remotely set various UX5000 functions for the specified extension by dialing the extension number and service code using a DISA line. This option is available for keysets, single line terminals, and IP terminals.

The following features can be set using service codes with this option:

Function Name	Default Service Code	Description
Day / Night Mode Switching for own Night Group (Program 11-10-01)	818	Change the Operation Mode for each Night Group
Setting the Automatic Trunk Transfer for each Trunk (Program 11-10-06)	833	Set the Automatic Trunk Transfer for each Trunk
Canceling the Automatic Trunk Transfer for each Trunk (Program 11-10-07)	834	Cancel the Automatic Trunk Transfer for each Trunk
Setting the destination for Automatic Trunk Transfer (Program 11-10-08)	835	Register the destination telephone number for Automatic Trunk Transfer. With the Remote Feature Setup, after dialing the destination telephone number, *# must be entered (ex: 2035551234 *#).
VRS - Record / Erase Message (Program 11-10-20)	116	Record / Playback / Erase VRS Messages
VRS - General Message Playback (Program 11-10-21)	111	Playback General Message
VRS - Record / Erase General Message (Program 11-10-22)	112	Record / Playback / Erase General Message
Call Forward - Immediate (Program 11-11-01)	-	Set/Cancel Call Forward Immediate *(Service Code + 1 <Set> / 0 <Cancel> + Transferred Destination Extension Number)*
Call Forward - Busy (Program 11-11-02)	-	Set/Cancel Call Forward when Busy *(Service Code + 1 <Set> / 0 <Cancel> + Transferred Destination Extension Number)*
Call Forward - No Answer (Program 11-11-03)	-	Set/Cancel Call Forward when No Answer during pre-assigned period *(Service Code + 1 <Set> / 0 <Cancel> + Transferred Destination Extension Number)*
Call Forward - Busy/No Answer (Program 11-11-04)	-	Set/Cancel Call Forward when No Answer during pre-assigned period *(Service Code + 1 <Set> / 0 <Cancel> + Transferred Destination Extension Number)*
Call Forward - Both Ring (Program 11-11-05)	-	Set/Cancel Call Forward Both Ring *(Service Code + 1 <Set> / 0 <Cancel> + Transferred Destination Extension Number)*
Call Forward - Follow Me (Program 11-11-07)	-	Set/Cancel Call Forward - Follow Me *(Service Code + 1 <Set> / 0 <Cancel> + Appropriate Extension Number)*
DND (Do Not Disturb) (Program 11-11-08)	847	Set/Cancel DND *(Service Code + 0 <Cancel> / 1 <External Call> / 2 <Internal Call> / 3 <All Call> / 4 <CFW Transferred Call>)*

The DISA feature must be enabled for this function.

Notes:

- While the outside caller is setting the function via DISA, no one can use the extension which is being set.
- The outsider caller can not set/cancel a function via DISA when the selected extension is being used except during incoming ringing (including incoming ACD calls). If the extension is busy, the call will be terminated.
- The VRS is required to send the fixed messages heard during the feature setup, but the feature can be used without the prompts.

Direct Station Selection (DSS) Console

Feature Availability
• Available - 32 60-Button or Aspire 110-Button Consoles combined maximum. A digital extension can have up to 32 UX5000 60-Button DSS Consoles installed, while an IP terminal can have only 1 UX5000 60-Button DSS Console installed (however additional DSS Consoles connected to digital ports can be assigned to an IP terminal).

Description

The 60-Button DSS Consoles (P/N 0910094 or 0910096) give a keyset user a Busy Lamp Field (BLF) and one-button access to extensions, trunks and UX5000 features. This saves time for users that do a lot of call processing (e.g., operators or dispatchers). The DSS Console simplifies:

- Calling extensions and Door Boxes

  Consoles can be used to allow a non-ACD DSS console to lamp indicating the status of both non-ACD agents as well as ACD agents.
- 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 keyset user with a DSS Console quickly reroute their calls to a co-worker. When the user places their console off-duty (by pressing the ALT. key), their calls route automatically to the programmed co-worker. Transferred and dial “0” calls ring both DSS Consoles. Central office calls ring both consoles.

You can also program the DSS Console keys to store Service Codes codes (up to 29 digits long). This provides the DSS Console user with many of the features available on Programmable Feature Keys. The DSS Console keys can optionally store additional associated digits after the Service Code. For example, storing 8141 under a DSS Console key accesses Trunk Group 1 when the console user presses the key.

The maximum number of DSS Consoles allowed per UX5000 is 32 (these can be either 60-Button or Aspire 110-Buttons Consoles or any combination of both). The consoles can be installed on digital keysets or IP keysets. ***When installing the 60-Button DSS Console for use with a digital keyset, the console is connected to a digital port. When installing the console for use with an IP keyset, it is connected to the bottom of the IP keyset.***

Each UX5000 60-Button DSS Console can be set to display two different ranges of extensions using the Page Switching key (Program 30-03-01, code: 95).

- Except when in Hotel/Motel mode, the fixed keys (ALT, Night, Day, Break, Nite2, Page, Group, Door, Ext.1, Ext.2) of a 110-button DSS console are not used and will be ignored if pressed.
- When displaying the Extension range 2 on an Aspire 110-button DSS console, if the Page, Group, or Door key is pressed in Hotel/Motel mode, the console will display the Extension Range 1 keys for Message Waiting, Wake Up Call, and Room Status.
- There is no Extension range 2 for the Hotel/Motel modes: Message Waiting, Wake Up Call, and Room Status. If the range is switched, it will change to display the Extension range 2 ICM indications.
- An IP terminal can have a 16DLS attached as well as one 60-Button DSS Console, which would be physically wired to the terminal. However additional DSS Consoles connected to digital ports can be assigned to an IP terminal). A digital terminal can have a 16DLS attached as well as any number of 60-Button DSS Consoles (up to the system maximum).

Directed Call Pickup

Feature Availability
• Available.

Description

Directed Call Pickup permits an extension user to intercept a call ringing another extension. This allows a user to conveniently answer a co-worker's call from their own terminal. 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

Feature Availability
• Available.

Description

Directory Dialing allows a display keyset user to select a co-worker or outside call from a list of names, rather than dialing the phone number. There are four types of Directory Dialing:

- **Abbreviated Dialing** (includes a sub-menu for **ABBc - Common** and **ABBg - Group**)
- **EXT.** - Co-worker's extensions
- **STA** - Personal Abbreviated Dialing (One-Touch Keys)
- **TELBK** - Central Telephone Book

Display Messaging, Selectable

Please refer to Selectable Display Messaging (page 166) for information on this feature.

Distinctive Ringing, Tones and Flash Patterns

Feature Availability
• Available.

Description

Distinctive Ringing, Tones and Flash Patterns provide extension users with audible and visual call status signals. This lets users tell the types of calls by listening to the ringing/tones and watching the keys. It also helps users monitor the progress of their calls. In addition, Distinctive Ringing lets keyset 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 keysets to ring at different pitches, the co-workers can always tell which calls are for them.

You can also customize the tones the UX5000 uses for splash tone, confirmation tone, trunk ring tone, Intercom ring tone and Alarm ring tone. Refer to the chart below and the Programming section for more details.

Distinctive Ringing, Tones and Flash Patterns	
Program	**Description**
80-01 : Service Tone Setup	Set the frequency of the UX5000's splash tone. This is the tone the UX5000 uses, for example, to alert the user of an incoming voice-announced Intercom call.
82-01 : Incoming Ring Tone	Set the trunk ring tones, which are the tones a user hears when a trunk rings an extension.
82-01 : Intercom and Alarm Ring Tone	Set the Intercom and External Alarm Sensor ring tones.
82-03 : DSS Console LED Pattern Setup and 30-05 : DSS Console Lamp Table	Set the DSS and Hotline key flash rates for busy, idle, DND, ACD Agent status, and hotel options.

Do Not Disturb

Feature Availability
• Available.

Description

Do Not Disturb blocks incoming calls and Paging announcements. DND permits an extension user to work by the terminal undisturbed by incoming calls and announcements. The user can activate DND while their terminal is idle or while on a call. Once activated, incoming trunk calls still flash the line keys. The user may use the terminal in the normal manner for placing and processing calls.

There are five Do Not Disturb options available at each extension:
1 = Incoming trunk calls blocked
2 = Paging, incoming Intercom, Call Forwards and transferred trunk calls blocked
3 = All calls blocked
4 = Incoming Call Forwards blocked
0 = Do Not Disturbed canceled

Keyset users are able to activate or deactivate Do Not Disturb while on a call. This option is not available for single line terminals.

Door Box

Feature Availability
• Available - 8 Door Boxes and 6 Chime Tones.

Description

The Door Box is a self-contained Intercom unit typically used to monitor an entrance door. A visitor at the door can press the Door Box call button (like a door bell). The Door Box then sends chime tones to all extensions programmed to receive chimes. To answer the chime, the called extension user just 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 just answer the Door Box chimes instead. Any number of UX5000 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 keyset user can press FLASH to activate the Door Box contacts. This in turn releases the electric strike on the entrance door.

The communications server can have up to eight Door Boxes.

Dual Line Appearance

Feature Availability
• Available.

Description

Each keyset has two line appearance keys (CALL1 and CALL2) for placing and answering calls. These line appearance keys, assigned to the extension's number, simplify operations for busy users. For example, the user can easily process a new call on one appearance with a call in progress on the other.

E911 Compatibility

Feature Availability
• Available. • IP terminals using the Lock button must be programmed to allow emergency numbers (such as 911) to be dialed. Refer to the Operation section of this feature for details.

Description

IMPORTANT - PLEASE NOTE THE FOLLOWING!

1. ***All local trunks or the trunk connected to external 911 equipment must be put into the E911 route.*** By placing all the local trunks into the E911 route, this assures that any user may make a call to 911.
2. ***When ARS is NOT enabled and the UX5000 allows trunk access by dialing '9'***, single line terminals will disregard Program 20-03-03 - System Options for Single Lines Terminals - SLT DTMF Dial to Trunk Lines. This will prevent the UX5000 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 will require additional programming changes.
3. ***When using external E911 equipment***, do not allow analog single line terminals to directly access trunks. When an analog SLT directly accesses a trunk (#9 xxx) and dials 911, the UX5000 will not follow the 911 routing. If your UX5000 is connected to external E911 equipment, the UX5000 will not route the call to that equipment.
4. ***Do not use an asterisk within a PBX access code if the Account Code feature is used.*** With the Account Code feature enabled, if an asterisk is used within the access code, the trunk stops sending digits to the central office after the * is sent.
5. ***Finally, but most important, 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 will 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 terminal, lifts the handset and dials 911. The UX5000's built-in E911 compatibility places the emergency call even if the user forgets to dial an access code or press a line key. The E911 capabilities include:

- **Attendant Notification**
  The attendant receives a notification each time a co-worker dials an emergency 911 call. This notification is the co-worker's name and number display optionally accompanied by an audible alarm. Notification occurs regardless of whether the attendant is idle or busy on a call. You can optionally extend this capability to other supervisory extensions as well.

- **Emergency Routing**
  When an extension user dials 911, the UX5000 can automatically find a trunk for the call. The UX5000 can choose a route to which the user normally does not have access. If all normal routes are busy, the UX5000 can even disconnect an active call and place the emergency call. E911 Compatibility uses the flexibility of the Automatic Route Selection Call Route Options to route 911 emergency calls (even in UX5000s in which ARS is not enabled).
  - **E911 Outgoing Dialing**
    The E911 calls follow the trunk group route programming. It is possible to use the flexibility of the Automatic Route Selection Call Route Options for additional routing options.
  - **Forced Disconnect Follows Timer to Disconnect Call**
    When all lines in the programmed route are busy and the UX5000 must drop a call in order to place a 911 call, the UX5000 will wait the amount of time set in Program 81-01 before disconnecting the call.

- **Compatibility with Customer Provided E911 Equipment**
  The UX5000 can automatically send a 911 call to customer-provided E911 equipment (such as the Proctor 911 ANI-LINK System II). The E911 equipment will dial emergency service and provide the caller's extension number to the emergency personnel. When using this type of equipment, the UX5000 must be programmed to send E911 calls to the trunk connected to the E911 box. This is done by assigning the trunk to the E911 trunk group and using ARS to route all 911 calls to that port. The Dial Treatment must then be set to send the extension number and '911' to that port. This is usually accomplished using XRE in the Dial Treatment, but check the requirements of your E911 box.
- **Calling Party Identification**
  With ISDN installed, the UX5000 can provide the calling party's telephone number and extension number. No additional customer-provided 911 equipment is required.

Uninstalled Trunks in Trunk Group Prevent Call from Dialing Out

By UX5000 default, all trunks in **Program 14-05-01 : Trunk Group** are in group 1. When placing a 911 call, the UX5000 will try to access the trunks defined in the group. If the trunks do not exist, the call will not dial out. In order for E911 to function correctly, remove any uninstalled trunks from the trunk group.

The UX5000 software will be changed in a future release to ignore trunks which have no voltage present on the blade.

If **Program 21-01-12 : System Options for Outgoing Calls, Dial 911 Routing Without Trunk Access** is set to "0" (trunk access code required), when using the Dial Number Preview feature and dialing 9+911, if all trunks are busy, the user will hear a busy signal and the call will not dial out.

If option 21-01-12 is set to "1" (trunk access code not required) and using Dial Number Preview, 911 is dialed, the UX5000 will disconnect a trunk and dial the call.

3

Dial Number Preview is when a telephone number is first dialed (previewing the number in the display) then pressing the CALL or line key to place the call.

Networked IP Terminals Can Follow ARS Class of Service Matching to Call Local Authorities
When using IP terminals at a remote site (Site B) which are registered to the main system (Site A), you can use the ARS Class of Service Matching feature to route 911 calls to the local authorities at a remote location. Without this programming, since the terminals are registered to the main site, when 911 calls are placed by IP terminals, the local authorities at the main site (Site A) would be called.

Flash

Feature Availability
• Available.

Description

Flash allows an extension user to access certain CO and PBX features by interrupting 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

Feature Availability
• Available.

Description

Flexible System Numbering lets you reassign the UX5000's 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 UX5000's 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 UX5000's Service Code numbers
- Assign single digit access to selected Service Codes

You can also use Flexible System Numbering to change the UX5000's Trunk Group Routing code. Although the default code of 9 is suitable for most applications, you can alter the code if needed.

For more information on the UX5000s standard numbering, refer to Tables 1, 2, 3, 4 and 6 at the beginning of this section.

The UX5000 provides a completely flexible system numbering plan. Refer to the chart below and the Programming section for more details.

Flexible System Numbering	
Program	**Description**
11-01 : System Numbering	Set the UX5000's internal (Intercom) numbering plan. The numbering plan includes the digits an extension user must dial to access features and other extensions.
11-09-01 : Trunk Access Code	Assign the single-digit trunk access code (normally 9). This is the code users dial to access Automatic Route Selection or Trunk Group Routing.
11-10 : Service Code Setup (for System Administrator) 11-11 : Service Code Setup (for Setup/Entry Operation) 11-12 : Service Code Setup (for Service Access) 11-13 : Service Code Setup (for ACD) 11-14 : Service Code Setup (for Hotel) 11-15 : Service Code Setup, Administrative (for Hotel)	Customize the Service Codes.
11-16 : Single Digit Service Code Setup	Assign the Single Digit Service Codes. these are the post-dialing codes a user can dial after placing an Intercom call to a co-worker.

Forced Trunk Disconnect

Feature Availability
• Available.

Description

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

Feature Availability
• Available - 64 Call Pickup Groups.

Description

Group Call Pickup allows an extension user to answer a call ringing an extension in a Pickup Group. This permits co-workers in the same work area to easily answer each other's calls. The user can intercept the ringing call by dialing a code or pressing a programmed Group Call Pickup key. If several extensions within the group are ringing at the same time, Group Call Pickup intercepts the call based on the extension's priority within the Pickup Group.

With Group Call Pickup, a user can intercept the following types of calls:

- A call ringing the user's own pickup group
- A call ringing another pickup group when the user knows the group number
- A call ringing another pickup group when the user doesn't know the group number

Group Listen

Feature Availability
• Available.

Description

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

Handsfree and Monitor

Feature Availability
• Available.

Description

Handsfree allows a keyset user to process calls using the speaker and microphone in the terminal (instead of the handset). Handsfree is a convenience for workers who don't 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 terminal.

The UX5000 provides three variations of Handsfree operation:

Handsfree	User can place and answer calls by pressing SPK instead of using the handset.
Automatic Handsfree	User can press a line or line appearance key without first lifting the handset or pressing SPK. An extension can have Automatic Handsfree for just outgoing calls or both outgoing calls and incoming line/loop key calls. Automatic Handsfree can also be used with the Call Coverage or Park features. Normally, extensions without speakerphones should have Automatic Handsfree for outgoing calls only.
Monitor	User can place a call without lifting the handset, but must lift the handset to speak.

Handsfree Answerback/Forced Intercom Ringing

Feature Availability
• Available.

Description

Handsfree Answerback permits an extension user to respond to a voice-announced Intercom call by speaking toward the terminal, without lifting the handset. Like Handsfree, this is a convenience for workers who don't have a free hand to pick up the handset.

Headset Operation

Feature Availability
• Available.

Description

A keyset user can utilize a customer-provided headset in place of the handset. Like using Handsfree, using the headset frees up the user's hands for other work. However, Headset Operation provides privacy not available from Handsfree.

An extension with a headset has two options for when it appears busy to incoming callers. The headset extension can be:

- Busy to incoming callers when only one extension appearance is busy (i.e., Off-Hook Signaling prevented)
  OR
- Busy to incoming callers only when both extension appearances are busy (i.e., Off Hook Signaling allowed)

As the headset plugs into a separate jack on the bottom of the terminal, the handset can still be connected to the terminal. This provides you with the option to use the handset, headset or the speakerphone for calls. Depending on the method used to activate the headset mode, you can either use the headset on a call-by-call basis or any calls placed/answered will automatically be in headset mode:

- Call-by-Call Basis: Press only the Programmable Function Key to answer/disconnect each call [defined in Program 15-07-01, code 05])

  With this option, the handset and speakerphone can also be used at any time to answer/place calls.

- Automatic Headset Mode: Enable Headset mode by entering the Headset service code defined in Program 11-11-65.

  With this option, the handset can also be used at any time to answer/place calls. The speakerphone will not function except for handsfree answerback on incoming intercom calls.

All Plantronics Polaris headsets should be compatible. Examples are:

Plantronics Polaris Headsets:	NEC Part Number
Polaris SupraPlus/NC-M (monaural with noise canceling transmitter)	750643
Polaris SupraPlus/NC-B (binaural with noise canceling transmitter)	750645
Polaris SupraPlus/VT-M (monaural with voice tube transmitter)	750644
Polaris Encore/VT-M (monaural with voice tube transmitter)	750634
Polaris Encore/NC-B (binaural with noise canceling transmitter)	750635
Polaris Tristar/VT-M (monaural with noise canceling transmitter)	750630
Polaris Mirage/VT-M (monaural with voice tube transmitter)	750631

Notes:

- While talking with an outside party while on speakerphone or off-hook with the handset, ringing for an incoming call will not be heard in the headset.
- Going off-hook with the handset while an incoming call is ringing can provide an inconsistent dial tone level.
- When the user is set up for ringing through the headset, the handset should not be used. It is recommended to only use the headset in this case.

Headset Operation for SLT Headset Operation

The ability for single line terminal users to use the Headset feature is available. When a single line terminal with a headset receives an incoming call, the UX5000 can let the SLT user know of the incoming call by a notification tone in their headset.

Notes:

- This feature is only applicable when the SLT is set in headset mode and the UX5000 allows the headset ringing with their class of service.
- If a SLT with headset ringing condition goes on hook, the SLT will ring normally.
- A SLT user with a headset should not be allowed to automatically seize a trunk (Line Preference) on an off-hook condition as the terminal would never go idle.
- If a SLT is set for the Ringdown feature, the headset mode is NOT available. But, if the Ringdown call start time is equal to or more than the headset ringing start time, the Headset feature, and not the Ringdown feature, is available.
- Caller ID should be disabled for the SLT when then Headset feature is to be used (Program 15-03-09 and 15-03-10 set to "0"). Otherwise, the terminal will only beep twice with an incoming call. In addition, the terminal will not receive the Caller ID information when the Headset feature is used.

Hold

Feature Availability
• Available.

Description

Hold lets an extension user put a call in a temporary waiting state. The caller on Hold hears silence or Music on Hold, not conversation in the extension user's work area. While the call waits on Hold, the extension user may process calls or use a UX5000 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 keysets. Any keyset 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 keysets that have a key for the trunk. Exclusive hold is important if a user doesn't want a co-worker picking up their call on Hold.
- **Group Hold**
  If a user places a call on Group Hold, another user in the Department Group can dial a code to pick up the call. This lets members of a department easily pick up each other's calls.
- **Intercom Hold**
  A user can place an Intercom call on Hold. The Intercom call on Hold does not indicate at any other extension.

With Automatic Hold enabled (Program 15-02-07), when the user is on an ***outside call using the handset***, the user can press a flashing line/loop key to answer an incoming call without disconnecting their first call. The first caller is automatically placed on hold. This feature does not work using handsfree or when the user is on an ICM and presses a flashing line/loop key (the ICM call is disconnected).

Hold Recall to Operator

Hold Recall to Operator enhances how the UX5000 handles calls that have been left on hold too long. With Hold Recall to Operator:

- A trunk call recalls the extension that placed it on Hold after the Hold/Exclusive Hold Recall time.
- The recalling trunk will ring the extension that placed it on Hold for the Hold/Exclusive Hold Recall Callback Time.
- After the Hold/Exclusive Hold Recall Callback Time, the trunk call will ring the operator.

Hold Recall to Operator applies to trunk calls placed on System Hold, Exclusive Hold and Group Hold. It does not apply to Intercom calls.

3

Hot Keypad

Feature Availability
• Available.

Description

This software is enhanced with the Hot Keypad feature. This feature allows a user simply start dialing a desired number on the keypad of an idle keyset. There is no need to press the Speaker key first when this option is enabled.

This feature can be used for ICM, outgoing calls, service codes, etc., as long as 2 or more digits are dialed. This option can be used with F-Routing or ARS as well.

The dialing operation with this option enabled is the same as previous operation - except there is no need to press SPK first. (Outside calls would require a trunk access code to be dialed prior to the telephone number.)

Hotel/Motel

Feature Availability
• Available.

Description

The UX5000 can provide comprehensive hotel/motel services in addition to the features normally available to business users. Hotel/Motel features include:

- **Do Not Disturb**
  A guest can enable and disable Do Not Disturb for their room terminal. In addition, a hotel/motel employee with a keyset can enable and disable Do Not Disturb for a specific room terminal. Do Not Disturb (DND) blocks the room terminal's incoming calls and Paging announcements.

- **DSS Console Monitoring**
  Your DSS Console provides unique one-touch room monitoring capabilities. Just press a button on your DSS Console to check a room's status. Or, see at a glance which rooms have Wake Up Calls set or messages waiting. In addition, you can still use your console for business mode features.

- **Flexible Numbering Plan**
  To simplify dialing guests and services in your facility, customize your UX5000 to have room numbers match each terminal's extension number. For example, if the rooms on the first floor are numbered 100-120, the corresponding room extensions can also be 100-120.

- **Message Waiting**
  A hotel/motel employee with a keyset can send a Message Waiting to a room terminal. The message lamp on the room terminal flashes until the guest answers the Message Waiting. (The DSS Console can show all the rooms that have messages waiting.)

- **PMS Integration**
  The UX5000, in combination with a PMS Interface Box (PMS-U10), can support third-party PMS applications. The PMS-U10 serves as a gateway between the PMS applications and compatible voice mails. Through the exchange of room status and guest services messages, PMS Integration automates many lodging management tasks, such as check-in, check-out, room status and room Roll Restriction.

- **Room-to-Room Calling Restriction**
  Prevent guests in one room from calling guests in another — a hotel/motel employee with a keyset can enable and disable room- to-room calling.

- **Room Status**
  To better manage room usage, an employee with a keyset can change the status of a room terminal, including:
  - Checked In
  - Checked Out
  - Maid Required
  - Maid in Room

- **Room Status with Printout**
  An employee's DSS Console can indicate the status of the hotel/motel rooms. Optionally, a printer connected to a LAN can print out room status reports to provide more detailed information:
  - Room Status (occupied, available, ready and to be cleaned)
  - Room Telephone Call and Toll Restriction Information
  - Do Not Disturb and Clean Up Extension List
  - Message Waiting Report
  - Wake-up Call No-Answer Report

- **Single Digit Dialing**
  Single Digit Dialing gives guests one-touch access to important Hotel/Motel services. They can just lift the handset and press a single key for:
  - Extensions such as the front desk, reservation services, housekeeping or the maitre d' of the restaurant.
  - Feature Access Codes for one button access to selected features and outside lines.
  - Voice Mail, so guests can leave requests even when service providers are unavailable.
  - A Department Calling Group allowing, for example, guests to reach the first available agent in the reservation desk group.

- **Toll Restriction (When Checked In)**
  An employee can change the Toll Restriction for a guest's terminal. For example, the receptionist can enable long distance calling for each room terminal as the guests check in.

- **Wake Up Call**
  A guest can set or cancel a wake-up call request. A hotel/motel employee with a keyset can also set or cancel a wake-up call for a room terminal.

3

PMS Integration Available

With the use of a PMS Interface Box (PMS-U10), the UX5000 can support third-party PMS applications. The PMS-U10 serves as a gateway between the PMS applications the compatible voice mails. Through the exchange of room status and guest services messages, PMS Integration automates many lodging management tasks such as check-in, check-out, room status and room Toll Restriction.

In addition any voice mail used must be licensed for the Hotel feature and have PMS enabled. Refer to the appropriate voice mail installation manual for information on configuring the voice mail.

Hardware Requirements

- UX5000 with a Hospitality License
  After activating the license in WebPro or PCPro, the UX5000 must be reset.
- PMS Interface Box (PMS-U10)
- Compatible Voice Mail
- 3rd-Party Hospitality System
  Compatible with hospitality systems based on Hitachi PMS protocol. Note that some functionality implemented or implied by the Hitachi protocol may not be supported.

For additional information on Hotel/Motel features, refer to the Hotel/Motel Manual (P/N 0913208).

Hotline

Feature Availability
• Available - 512 Internal Hotline extensions.

Description

Hotline gives a keyset 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.

In addition, the Hotline key shows the status of the partner's extension

When the key is . . .	The extension is . . .
Off	Idle
On	Busy or ringing
Fast Flash	DND - All calls (option 3) or Intercom calls (option 2)

Hotline, External

Refer to the Ringdown Extension, Internal/External (page 164) for information on this feature.

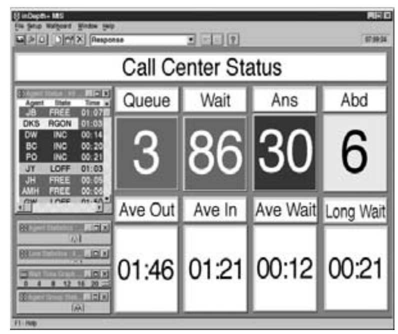
InDepth Lite, inDepth and inDepth+

Feature Availability
• Available.

Description

The inDepth applications are Windows-based Management Information Systems that work in conjunction with the UX5000 ACD. These ACD/MIS systems enhance the UX5000 ACD with real time statistics and reports on ACD group traffic patterns and usage. The inDepth Lite, inDepth and inDepth+ applications have an extensive set of user-configurable Real Time Windows and Reporter subsystems.

InDepth+ is the more comprehensive and capable of the ACD/MIS systems and offers:



- **Real Time Status Window**
  This window displays ACD agent status, state and connection state.
- **Real Time Statistics Window**
  The statistics window provides a visual performance summary for lines, agents and ACD Groups.
- **Call Queue and Wait Time Windows**
  These windows show the number of calls in queue, the longest wait time, as well as the number of calls answered and abandoned.
- **Wallboard Template**
  Use the wallboard template display to motivate and inform ACD agents through a dynamic display of real time statistics and messages.
- **Reporter**
  ACD administrators can create fully-configurable reports for display and printing.

Similar in many respects to inDepth+, inDepth is streamlined for more modest ACD applications. InDepth provides a single real time screen template, up to seven reports and can track report data for up to one full month. InDepth includes ACD/MIS features like Report View/Print and Audible/Visual Alarms, but excludes the Sub-Supervisor Positions and the Wallboard Support.

The inDepth Lite software provides small or startup organizations the ability to choose from six different management reports and four different status options to appear on-screen in the display window. Only a single screen template can be created and saved.

Hardware, Software and System Requirements

- Pentium 3 700 MHz or equivalent, 256 RAM
- Windows XP, Windows 2000 Professional SR-3, Windows 2000 Server, Windows 2003 Server
- SVGA mode (800 x 600)
- 2 GB Hard Drive Space
- Monitor
- CD-Rom Drive
- Network card - 10 Mbps
- Sound card

3

- 3 Serial Ports (depending on configuration)
- 1 Parallel Port
- Internet Explorer 6

For more information, refer to the inDepth/inDepth+ Manual (P/N 0893230) for the specifics.

Intercom

Feature Availability
• Available.

Description

Intercom gives extension users access to other extensions. This provides the UX5000 with complete internal calling capability.

Handsfree Answerback/Forced Intercom Ringing

Handsfree Answerback permits an extension user to respond to a voice-announced Intercom call by speaking toward the terminal, without lifting the handset. Like Handsfree, this is a convenience for workers who don't have a free hand to pick up the handset. Refer to **Handsfree Answerback/ Forced Intercom Ringing** (page 115) for more.

Busy Status Display

When a display keyset user places an Intercom call to a busy extension, the details of the busy status (who is talking to the extension or which line is in use by the extension) can be displayed. The details of the trunk's busy status (the extension using the line) can be displayed after trying to access the trunk. This feature provides a user information which can determine whether they should use the Barge-In feature for the extension or trunk. This information automatically displays for a keyset once programmed.

Intercom Abandoned Call Display

Feature Availability
• Available.

Description

Intercom Abandoned Call Display shows a display keyset user a list of Intercom calls placed to them that they did not answer. This is a convenience if a user has to temporarily leave their desk. When they return, they can display the list to find out who called while they were out.

IP DECT Terminal

Feature Availability
• Available - 120 wireless terminals maximum.

Description

The IP DECT Handset provides:

- Single Line Telephone Operation
- Unified Design with Desktop Terminal
- Fast Speed WLAN (802.11b/g)
- Authentication (EAP-TLS/PEAP)
- Encryption (WEP/TKIP)
- 8 Function Buttons
- Menu / Soft Key Operation
- Full-Dot Monochrome LCD with Backlit
- Ten-Key with Backlit
- MW Lamp

Refer the IP DECT Terminal manual and user guide for complete installation and setup information.

ISDN Compatibility

Feature Availability
• PRI Available.

Description

!! Important !!
Always check with your NEC's Technical Service Representative before setting up your ISDN application. Working together will ensure maximum compatibility and reliable ISDN performance.

Primary Rate Interface (PRI)

Flexible Transfer Feature

This combination tie line/DISA-type function provides the ability for an extension user to:

- call out on a virtual S-Bus extension and loop back into the UX5000 on a virtual T-Bus trunk without the need for a PRI blade.
  OR
- call out on a virtual T-Bus trunk and loop back into the UX5000 on a virtual S-Bus extension without the need for a PRI blade.

The Flexible Transfer feature provides a virtual ISDN trunk or ISDN station. When this feature is enabled, virtual trunk ports (ISDN trunk) and virtual extensions ports (ISDN station) are created. No PRI blade or open slot in the chassis is required to use this virtual PRI loop back ability. Therefore, the user can save costs for 2 PRIU blades and save 2 blade slots in the UX5000.

The user interface for the virtual ISDN trunks and virtual ISDN extensions ports are the same as real trunks and extensions.

This feature can be used, for example, to allow for transferring a call across a network to an ACD pilot number. It can also be used to allow voice mail to transfer calls to ring groups - locally or across the network.

The following table compares program options with Program 10-03 (with a PRI blade) and Program 10-42 (without a PRI blade using the Flexible Transfer feature).

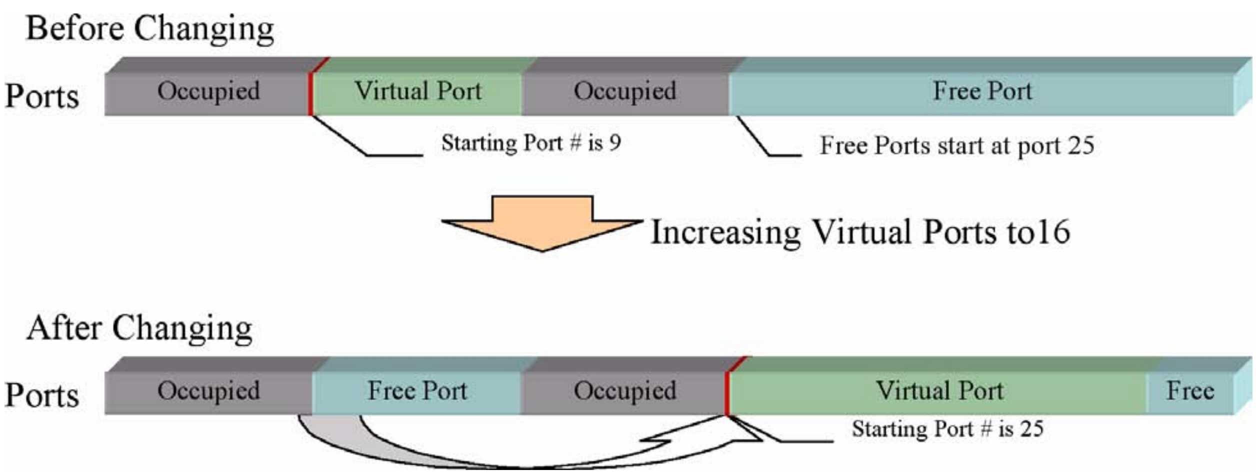
Item	Program 10-03 for PRI	Program 10-42 for Flexible Transfer
01	ISDN Line Mode	N/A
02	Logical Port No.	Trunk: Program 10-42-02 Station: Program 10-42-03
03	CRC Multi-Frame(CRC4)	N/A
04	Layer 3 Timer Type	Program 10-42-04
05	Calling Party Number	Program 10-42-05
06	Transmit Pulse Shape	N/A
07	S-Point DID Digits	Program 10-42-06
08	Dial Sending Mode	Overlap Sending always used.
09	Dial Information Element	Called Party Number always used.
10	Master/Slave System	N/A
11	Networking System No.	N/A
12	Short/Long-Haul	N/A
13	Loss-Of-Signal Detection Limit	N/A
14	Service Protocol for S-Point	N/A (This setting should be available for connecting S-Bus terminal)
15	Call Busy Mode for S-Point	Program 10-42-07
16	Two B-Channel Transfer for PRI Service	N/A
17	ISDN Line Ringback Tone	Ringback tone will always be local tone.
18	Type of Number	This setting is for outgoing calls to the Network. It is not needed with the Flexible Transfer feature.
19	Numbering Plan Identification	This setting is for outgoing call to Network. It is not needed with the Flexible Transfer feature.

Port Assignment

Once the number of virtual loop back ports is defined in Program 10-42-01, the trunk and extension ports are assigned automatically. The UX5000 takes the next available group of consecutive ports and these are in reserved in sets of 2. This could result in a dead/unused port if the number of ports defined in Program 10-42-01 is an odd number. For example, if 5 ports are programmed in Program 10-42-01, the UX5000 actually reserves 6 ports.

The assigned ports (trunk and extension) will be freed when Program 10-42-01 is set to 0 or reduced. If there is an established call(s) using a virtual PRI loop back when this program is changed, these calls will automatically be dropped.

In the following case, the starting port number may be changed after additional ports are entered in Program 10-42-01.



Operation

The same operation and appearance (number / name display, LED indications on terminals) are the same as the loop back operation using a physical PRIU blade and connection. The only exception is with the VRS Queue Messaging during Alerting status.

With the VRS Queue Messaging during Alerting status, when using a physical PRIU blade and connection, a CONNECT message is sent when starting a VAU message when an ISDN trunk has not sent a CONNECT message.

With the Flexible Transfer feature and virtual loop back, at this point a PROGRESS message will be sent instead of the CONNECT message. A CONNECT message will be sent when the extension users answer a call.

Using the Flexible Transfer feature, the UX5000 supports many features, such as:

- Internal SMDR
- Internal Toll Restriction
- Internal ACD
- Internal (DID) Routing Table
- Internal Night Service
- Internal Ring Groups
- Above features available via networking
- in addition to many other features

Establish a Call

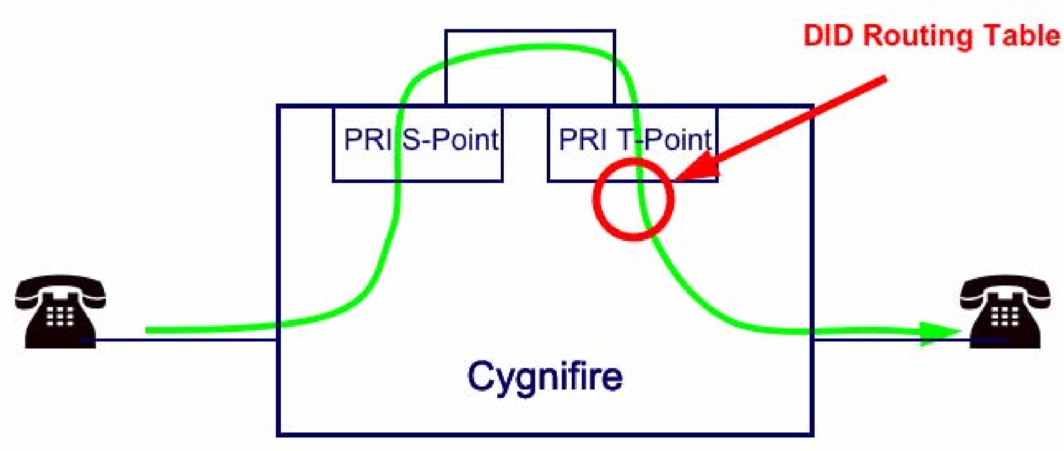
- S-Point Ports can be called with
  - S-Point Port Number (+DID Digits)
  - S-Point Department Group Number (+DID Digits)

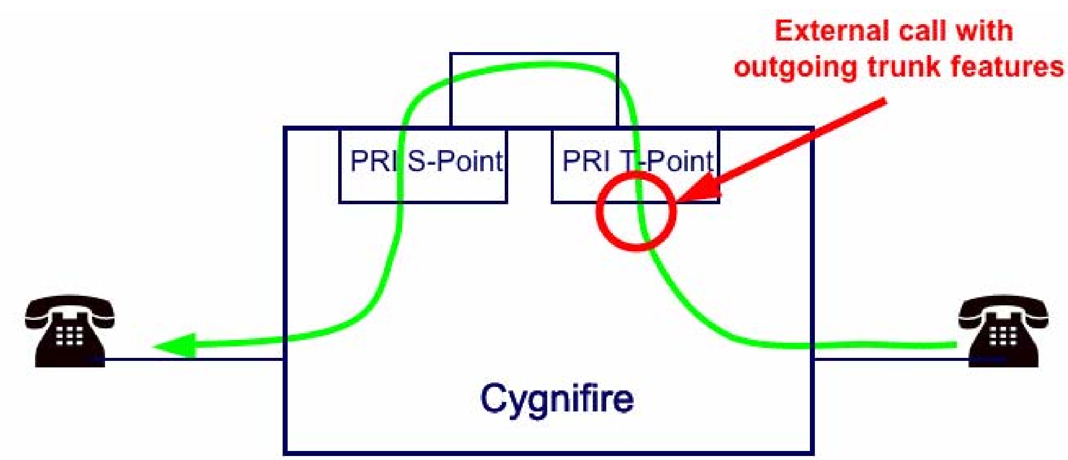
- T-Point Ports can be seized with
  - Direct Trunk selection
  - Trunk Group Access
  - Trunk Group Routing
  - F-Routing
  - Any other Trunk access procedure

Combination with Other Features

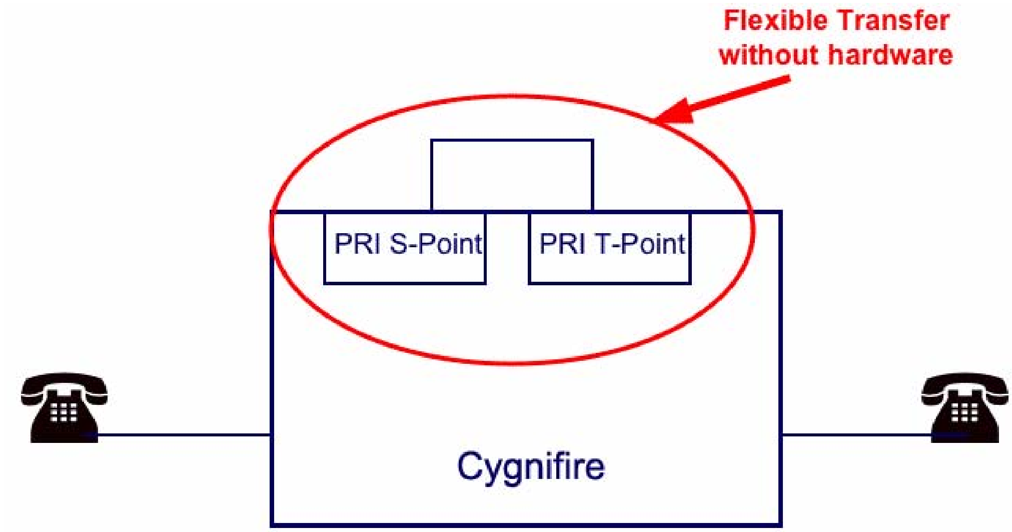
- SMDR
  - Internal Calls through the Loop are Reported
  - Seize a virtual Trunk and dial any internal Number
  - Call a S-Point Port that is routed via a Trunk to a target
- Toll Restriction / Automatic Trunk Seizure
  - Set automatic Trunk seizure (loop-back Trunks)
  - Prohibit any internal Calls in the Class of Service
  - Use the Toll restriction for any Call (also internal)
- Call Forward
  - Set a CFW to the S-Point Number + DID
  - Use the DID Conversion Table to point to the Target
- DID
  - Point a DID target to an S-Point loop-back Port
  - Use the full flexibility of the DID Routing Table after DID
    - Overflow Busy / No Answer
    - Ring Groups
    - External Targets
  - SMDR will report the DID Selection
  - Chain different DID Trees
- Night Service
  - Night Service can use different DID Conversion Tables
  - Consider to use Night Service Groups for the Virtual loop-back
- F-Routing
  - F-Routing is the most flexible way to:
    - Access Trunks
    - Modify Dial Informations
- ACD
  - Place a Call to a S-Point Port
  - The DID Conversion allows to place the call into ACD
  - Internal Calls can ring ACD Groups

- Ring Groups / Queue Messages
  - Place a call to a S-Point Port
  - Use Ring Groups as Target in the Conversion Table
  - Use Queue Messages for those Ring Groups





3



With this feature, the following conditions apply:

- The feature will not be activated if the port assignment procedure fails to allocate the number of consecutive trunk ports and extension ports.
- The user cannot change UX5000 data in Program10-03. S-point DID digits for Virtual S-point ports can be defined in Program 10-42. The other setting in Program 10-03 will be run with default value.
- Virtual Loop Back PRI ports cannot be blocked. These ports are always available.
- Assigned port numbers can be confirmed in Program 10-42-02 and 10-42-03. PCPro will indicate this information in the slot configuration as well, however, with WebPro, the only indication will be in 10-42-02/03.
- Though assigned virtual ports are the same as PRI ports in the UX5000, these ports can be used in the non-licensed UX5000. Because these ports do not require the hardware.
- Program 90-39 can be used to reset all of the Virtual Loop Back ports. All connected calls will be dropped when this program is executed.

Last Number Redial

Feature Availability
• Available.

Description

Last Number Redial allows an extension user to quickly redial the last number dialed. For example, a user may quickly recall a busy or unanswered number without manually dialing the digits.

Last Number Redial saves in UX5000 memory the last 24 digits a user dials. The number can be any combination of digits 0-9, # and *. The UX5000 remembers the digits regardless of whether the call was answered, unanswered or busy. The UX5000 normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

Licensing

Feature Availability
• Available.

Description

Certain features are allowed based on licensing. Contact UBSD's Customer Support for complete details on the required licenses and their associated part numbers.

Line Preference

Feature Availability
• Available.

Description

Line Preference determines how a keyset 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 keyset user answers calls. When a call rings the keyset, 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 keyset user places calls. If a keyset has Outgoing Intercom Line Preference, the user hears Intercom dial tone when they lift the handset. If a keyset has Outgoing Trunk Line Preference, the user hears trunk dial tone when they lift the handset. Outgoing Line Preference also determines what happens at extensions with Idle Line Preference. The user hears either trunk ("dial 9") or Intercom dial tone.

Auto-Answer of Non-Ringing Lines

With Auto-Answer of Non-Ringing Lines, an extension user can automatically answer trunk calls that ring other extensions (not their own). This would help a user that has to answer calls for co-workers that are away from their desks. When the user lifts the handset, they automatically answer the ringing calls based on Trunk Group Routing programming. The extension user's own ringing calls, however, always have priority over calls ringing other co-worker's extensions.

Long Conversation Cutoff

Feature Availability
• Available.

Description

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

When used with the Warning Tone for Long Conversation feature, the UX5000 can provide a warning tone on outgoing trunks calls before the call is disconnected. This tone is not available to analog single line terminal users.

Loop Keys

Feature Availability
• Available.

Description

Loop keys are uniquely programmed function keys that simplify placing and answering trunk calls. There are three types of loop keys: Incoming Only, Outgoing Only and Both Ways.

- **Incoming Only Loop Keys**
  Incoming Only loop keys are for answering trunk calls. An extension can have an incoming loop key for a specific trunk group (fixed) or a "catch all" loop key for any trunk group (switched). Fixed loop keys allow an extension user to tell the type of call by the ringing key. Switched loop keys are ideal for an extension with a large number of feature keys. In addition, switched loop keys are a destination for any trunk not on a line key or fixed loop key. Without a switched loop key, calls not appearing on a line key or fixed loop key will ring only the CALL key. Incoming Only loop keys also receive Transferred trunk calls.

- **Outgoing Only Loop Keys**
  Outgoing Only loop keys are for placing trunk calls. An extension can have outgoing loop keys for a specific trunk group or for ARS access. When a user presses the loop key, they get dial tone from the first available trunk in the group (or from ARS if programmed). Outgoing Only loop keys help ensure that an extension will always have a key available for placing calls.

- **Both Ways Loop Keys**
  Both Ways loop keys combine the functions of both Incoming Only and Outgoing Only loop keys. Both Ways loop keys work well for extension users that handle a moderate amount of calls and don't separate keys for incoming and outgoing calls. Both Ways loop keys also receive Transferred trunk calls.

An extension can have many loop keys - of any type. You can program an operator, for example, with four loop keys for incoming calls and four for outgoing calls.

Once a loop 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 will ring the first available loop key, beginning with the lowest numbered key. If keys 1-3 are loop keys, for example, the first incoming call rings key 1. If key 1 is busy, the next call rings key 2. If keys 1 and 2 are busy, the next call rings key 3. If all three keys are busy, additional incoming calls queue for the first available key. The terminal display will show “WAITING - LOOP KEY” if the user presses a loop key when there are additional calls waiting.

Maintenance

Feature Availability
• Available.

3

Description

The UX5000 provides several features to help maintain the UX5000 database, for backup, and system analysis.

- System Backup/Restore Files
- PCPro / WebPro / UserPro
- Remote Maintenance
- Alarm Reports
- System Log/DIM
- Self Test
- Traffic Reports
- Blade Software Block

System Backup/Recovery

UX5000 programming allows up to 5 backup/restore points to be saved. These files are saved in the flash memory on the CCPU. This data is used for recovering data if required.

Database Maintenance Using PCPro/WebPro

In addition to the TelPro (keyset) programming, the UX5000 provides the ability to use a PC to access UX5000 programming. The Windows-based PCPro and the HTML-based WebPro allows you to:

- Edit the terminal programming options from a remote location.

  The PCPro application requires changes to be uploaded to the UX5000 before they take affect. The WebPro application applies the changes as soon as the APPLY or OK icon is clicked.
- Access UX5000 maintenance functions (like reports and tests) as well as:
  - Slot Control (possible with terminal programming or WebPro access)
    Allows the blade slots to be deleted or reset

- Trunk Control (possible with terminal or PCPro programming or WebPro access)
  Allows the trunks to be blocked so no new additional calls can be made on the blade
- Extension Control (possible with terminal programming or WebPro access)
  Allows hard or soft reset for each extension
- System Reset (possible with terminal programming or WebPro access)
  Allows the UX5000 to be reset

In addition, PCPro allows you to:

- Save your programming to your PC's hard disk - then upload it via a LAN (Local Area Network) or USB connection.
- Download the existing programming in the UX5000 via a LAN or USB connection - and save it to your PC's hard disk.
- Set up a default database with the settings you use most often.
- Create a unique database for each UX5000 you have installed. Since you save the site-specific data to your PC's hard disk, you can easily retrieve a customer's programming if something goes wrong.

Programming access may be restricted based on the type of program entry used and if other users are connected to the UX5000 for programming purposes.

- PC Pro: Only one user allowed access to the UX5000 programming at a time.
- WebPro: Up to 4 WebPro or TelPro users can be connected at the same time.
- TelPro: Up to 4 TelPro or WebPro users can be connected at the same time.

Refer to the UX5000 Software Programming Manual, P/N 0913202, for details on accessing the UX5000 using TelPro or, with PCPro or WebPro, check the PCPro/WebPro Installation Manual for additional details.

UserPro: End-User Programming with Web Browser

A programming option is available to allow a user to adjust only the user level UX5000 programs. Similar to the WebPro application, a user would connect with an internet browser to the UX5000. Two password levels are supported, based on the user name and password used at the sign-in, which determine the programs which can be accessed.

Mode 1 is the UA (User Programming Administrator) mode. This level allows all of the user-programmable options to be changed. The password access for this level is set up in **Program 90-02-01 : Setting the Programming Password**.

Mode 2 is the UB (User Programming) mode. This level allows a user to change only the UX5000 data pertaining to their extension. The password access for this level is set up in **Program 90-28-01 : User Programming Password Setup** and is defined for each extension user required.

Remote Maintenance

This feature requires the installation of an MEMDB on the CCPU.

The Remote Maintenance feature allows an administrator to update a remote UX5000 using either software downloaded from a connected server or from a PC connected via PCPro.

This can either be performed manually or by a UX5000 timer.

When updating the software for remote UX5000s, the main UX5000 software must first be updated. The remote UX5000s can then be updated using the LAN connection to the main UX5000 or it can be updated using the PCPro application.

Extension Data Swap Function Added

The UX5000 provides the ability to swap an extension's programming to another extension number. The following extension-based programs will be swapped:

Refer to Secure Set Relocation feature below for additional options.

Program No.	Program Name
11-02	Extension Numbering
15-01	Extension Basic Data Setup (include Virtual Extension)
15-02	Multi-Line Terminal Basic Data Setup
15-03	Single Line Terminal Basic Data Setup
15-04	PHS Terminal Basic Data Setup
15-06	Trunk Access Map for Extension
15-07	Programmable Function Key
15-08	Incoming Virtual Extension Ring Tone Setup
15-09	Virtual Extension Ring Assignment
15-10	Incoming Virtual Extension Ring Tone Order Setup
15-11	Virtual Extension Delayed Ring Assignment
15-12	Conversation Recording Destination for Extension
20-06	Class of Service for Extension
21-02	Trunk Group Routing for Extensions
21-04	Toll Restriction Class for Extensions
21-07	Toll Restriction Override Password Setup
21-11	Hotline Assignment
23-02	Call Pickup Group
23-03	Ringing Line Preference
23-04	Ringing Line Preference for Virtual Extensions
24-03	Park Group Assignment
31-02	Internal Paging Group Assignment

Keep the following items in mind when using the Extension Data Swap function:

- Any user-defined programming stored in the SRAM will not be swapped (for example, Call Forward set up, Selectable Display Messaging, etc.).
- ***The extensions to be swapped must be idle*** while the swap is performed, or an "Invalid" error message will be received.
- Data for virtual extension's cannot be swapped.
- When a swap is performed, the following actions are executed for the swapped extensions.
  - Camp On Clear (Program 11-12-05)
  - Common Cancel (Program 11-12-37)
  - Last Number Redial Clear (Program 11-12-17)

3

- Saved Number Clear (Program 11-12-18)
- Incoming History data is deleted.

- ***Using Program 92-04-01 will also swap the order in which these extensions are displayed in all extension-related programs***. This means that the UX5000 will no longer display all the extension numbers from low to high. For example, if port 2 and 6 were swapped, when viewing the extensions in 15-02-01, the extensions will display in the following order: 301, 306, 303, 304, 305, 302.
- When swapping IP extensions, the terminals will automatically reset once swapped.

Extension Data Swap by Service Code / Secure Set Relocation

The UX5000 provides the ability to swap an extension's programming to another extension number using **Program 92-04-01 : Extension Data Swap**.

The extension data can be swapped using a service code as well. With this option, the user must enter a 4-digit password (fixed at 4 digits) in order to complete the swap.

The following extension-based programs will be swapped:

Program No.	Program Name
11-02	Extension Numbering
12-05	Night Mode Group Assignment for Extensions
13-03	Abbreviated Dialing Group Assignment for Extensions
15-01	Extension Basic Data Setup (include Virtual Extension)
15-02	Multi-Line Terminal Basic Data Setup
15-03	Single Line Terminal Basic Data Setup
15-06	Trunk Access Map for Extension
15-07	Programmable Function Key
15-08	Incoming Virtual Extension Ring Tone Setup
15-09	Virtual Extension Ring Assignment
15-10	Incoming Virtual Extension Ring Tone Order Setup
15-11	Virtual Extension Delayed Ring Assignment
15-12	Conversation Recording Destination for Extension
15-13	Loop Keys
15-14	Programmable One-Touch Keys
16-02	Department Group Assignment for Extensions
20-06	Class of Service for Extension
21-02	Trunk Group Routing for Extensions
21-04	Toll Restriction Class for Extensions
21-07	Toll Restriction Override Password Setup
21-10	Dial Block Restriction Class Per Extensions
21-11	Hotline Assignment
21-13	ISDN Calling Party Number Setup for Extensions
21-15	Individual Trunk Group Routing for Extensions
21-18	IP Trunk Calling Party Number Setup for Extensions

Program No.	Program Name
21-19	IP Trunk (SIP) Calling Party Number Setup for Extensions
21-20	SIP Trunk Call Discernment Setup for Extensions
22-04	Incoming Extension Ring Group Assignment
22-06	Normal Incoming Ring Mode
23-02	Call Pickup Group
23-03	Ringing Line Preference
23-04	Ringing Line Preference for Virtual Extensions
24-03	Park Group Assignment
24-06	Fixed Call Forwarding
24-07	Fixed Call Forwarding Off-Premise
26-04	ARS Class of Service
26-07	Not used in U.S.
31-02	Internal Paging Group Assignment
41-02	ACD Group and Agent Assignments
41-17	ACD Login Mode Setup
42-02	Hotel/Motel Terminal Setup
92-05	Password for Extension Data Swap

Secure Set Relocation Conditions

- Any user-defined programming stored in the SRAM will not be swapped (for example, Call Forward set up, Selectable Display Messaging, etc.).
- The extensions to be swapped must be idle while the swap is performed, or an "Invalid" error message will be received.
- Data for virtual extension's cannot be swapped.
- When a swap is performed, the following actions are executed for the swapped extensions.
  - Camp On Clear (Program 11-12-05)
  - Common Cancel (Program 11-12-37)
  - Last Number Redial Clear (Program 11-12-17)
  - Saved Number Clear (Program 11-12-18)
  - Incoming History data is deleted.
- Using this option will also swap the order in which these extensions are displayed in all extension-related programs. This means that the UX5000 will no longer display all the extension numbers from low to high. For example, if port 2 and 6 were swapped, when viewing the extensions in 15-02-01, the extensions will display in the following order: 301, 306, 303, 304, 305, 302.
- When swapping IP extensions, the terminals will automatically reset once swapped.

Alarm Reports

The UX5000 logs various errors and information about the operation which can be used to determine the cause of a problem (up to 100 individual alarms are stored, then oldest data is deleted to allow for new information to be stored). The UX5000 can indicate the errors using one of the following methods:

- On a keyset's display
- Output information to a PC via the COM port, the RS-232C port using a CTA Adapter (the CTA Adapter must be connected to an Aspire keyset) or PC terminal connected using Telnet
- Send the data via EMail
- PCPro's Alarm Report
- UX5000-SNMP Manager
- Save the information to a USB thumb drive
- CCPU LED lights when alarm generated

When an alarm is displayed using the CygniLink feature, as the alarm information is shared between UX5000s, the display will indicate the CygniLink system ID as part of the alarm notification. Each linked communications server can have a display terminal defined (for a maximum of 50). For example:

```
SID:XX-###  MM/DD  HH:MM
SMDR # full
```

SID:XX = System ID xx
XXX = Alarm Number
MM/DD HH:MM = Month, Date, Hour and Minute When Alarm Generated

Sample Report Through RS-232C / CTA Port:

<< Alarm Report >> 08/20/2007 14:12 PAGE 001

LVL	NO	STAT	DATE	TIME	ITEM	UNIT	SLT	PRT	PARAMETER
MIN	0002	ERR	2007/8/20	11:34	PKG Installation	VOIPU	6	0	
MAJ	8	ERR	2007/8/20	13:47	Memory Backup Bat	none	0	0	
MIN	0051	WAR	2007/8/20	15:34	System Data Change	none	0	0	

Meet Me Conference

Feature Availability
• Available - CCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number of internal or external parties conferenced up to the block's limit of 32.

Description

With Meet Me Conference, an extension user can set up a Conference with their current call and up to 31 other internal or external parties, for a total of 32 participants. 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.

Meet Me Paging

Feature Availability
• Available.

Description

Meet Me Paging allows an extension user to Page a co-worker and privately meet with them on a Page zone. The Paging zone is busy to other users while the meeting takes place. While the co-workers meet on the zone, no one else can hear the conversation, join in or make an announcement using that zone. Meet Me Paging is a good way to talk to a co-worker when their location is unknown. If the co-worker can hear the Page, they can join in the conversation.

Meet Me Paging Transfer

Feature Availability
• Available.

Description

If a user wants to Transfer a call to a co-worker but they don't 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

Feature Availability
• Available.

Description

While on an outside call, Memo Dial lets a display keyset user store an important number for easy redialing later on. The terminal can be like a notepad. For example, a user could dial Directory Assistance and ask for a client's telephone number. When Directory Assistance plays back the requested number, the caller can use Memo Dial to jot the number down in the terminal's 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

Feature Availability
• Available.

Description

An extension user can leave a Message Waiting indication at a busy or unanswered extension requesting a return call. The indication is a flashing MW lamp at the called extension and a steadily lit MW lamp on the calling extension. Answering the Message Waiting automatically calls the extension which left the indication. Message Waiting ensures that a user will not have to recall an unanswered extension. It also ensures that a user will 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 keyset 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. **Single Line Terminal MW Indication Option**
An option is available for analog single line terminals which provide a display. When a user leaves a Message Waiting for a SLT which has a display, **Program 15-03-13 : Single Line Terminal Basic Data Setup - MW Signal Type** is used to determine whether the SLT user will see a MW LED indication or if the Caller ID will be used to display the call. In addition, if a Message Waiting or Voice Mail message is left at a SLT, **Program 15-03-17 : Single Line Terminal Basic Data Setup - Dial Ton Select** can be used to provide an initial stutter dial tone (three beeps then normal dial tone) when the SLT handset is lifted.

MH250 Wireless IP DECT Terminal

Feature Availability
• Available.

Description

MH250 is a wireless IP terminal set that complies with IEEE 802.11b specifications. MH250 has the following features:

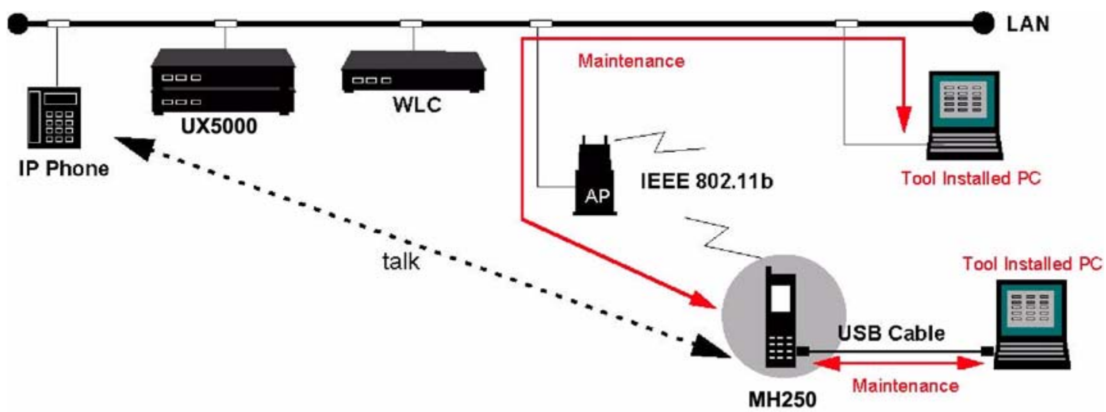
- Single line terminal PCS/PHS equivalent features by connecting with UX5000 and WLAN.

- MH250 dedicated features such as Short Message, Presence (Note), Phone Book, Call Log, Speed Dial, and Courtesy Mode.

- Easy maintenance by using a suite of GUI based tools are also available. For example, the firmware of MH250 can be updated by using one of the tools - Update Tool. You can run the tools on your PC, which is connected to the terminal via a USB cable or via LAN.

Note: While the MH250 terminal can be configured to support Presence functionality, it is important that the Presence server also supports NEC Extended SIP. Please refer to your specific Presence Server Documentation.

The following shows a system configuration that is composed of MH250 and the related equipment. To use MH250, the related equipment must be installed and configured properly in advance. For easy maintenance, a suite of useful maintenance tools are also available. Note that the tools can be used either via WLAN or via USB.



Refer the Wireless IP Terminal manual and user guide for complete installation and setup information.

Microphone Cutoff

Feature Availability
• Available.

Description

Microphone Cutoff lets a keyset user turn off their terminal's handsfree or handset microphone at any time. When activated, Microphone Mute prevents the caller from hearing conversations in the user's work area. The user may turn off the microphone while their terminal is idle, busy on a call or ringing. The microphone stays off until the user turns it back on.

Mobile Extension

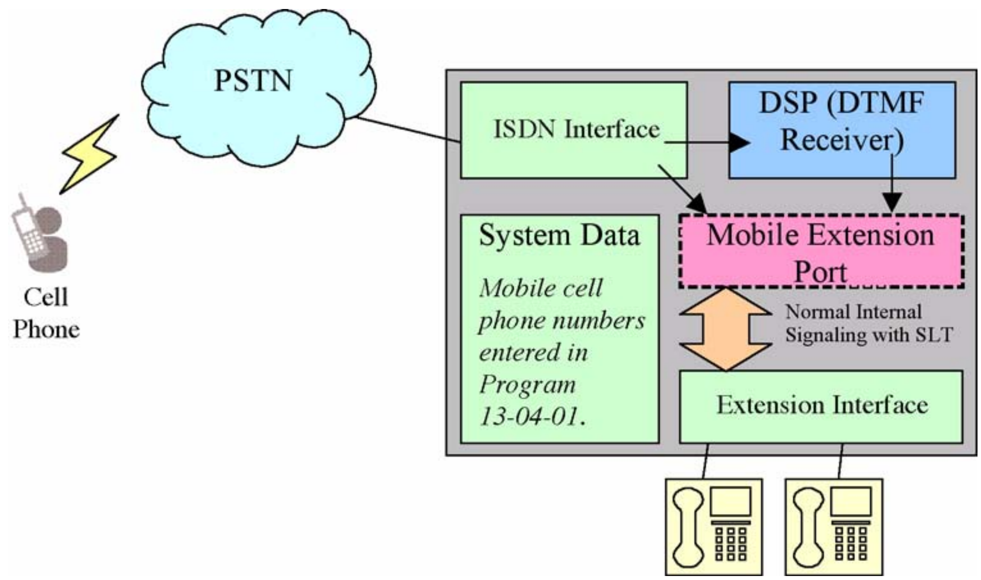
Feature Availability
• Available.

Description

Currently, this feature is only released using PRI or SIP trunks.
Analog trunk usage may be available in a future release.

A mobile extension is an external terminal (preferably a mobile phone) linked to the UX5000 via a Proxy Port in order to operate as an internal SLT extension. The extension sends DTMF signals to the UX5000 allowing access to the UX5000 features. A registered Mobile Extension uses 1 analog port (ports are reserved in groups of 4), however, **no** blade support (analog or digital) is required. In addition, the Mobile Extension port must be an unequipped extension port on the UX5000 - no physical keyset is required on the UX5000.

Note: A mobile extension cannot be used as a voice mail port.



The number of Mobile Extensions per UX5000 is based on the number of extension ports in the UX5000. For smaller systems, it is important to remember that the number of Mobile Extension ports = 25% of physical ports (8 ports allows for 2 Mobile Extension entries).

This restriction is based on the number of ports which could be required to call the mobile extension (for example: 1 port for an UX5000 keyset, 1 port for the Mobile Extension, and 1 or 2 trunk ports - depending on whether the call being sent to the Mobile Extension is an intercom call or an outside caller).

With larger systems (64 ports and higher), the limitation is basically limited only by available unequipped extension ports.

In the event of the limit being reached, the Mobile Extensions will be able to be added in programming, but will give the indication of an invalid dial entry when called.

This feature can currently be used with ISDN PRI trunks or SIP trunks.

Note: ***In order to provide a proper disconnect, Disconnect Supervision is required for the trunks used for this feature.***

The Mobile extension's internal extension number (Proxy Port) is linked to an abbreviated dial bin to provide integration.

Note: If all external trunks are busy when a call is made to the mobile extension, ringback tone is presented giving the impression the phone is ringing.

A DID is directed to the Mobile Extension's internal extension number (Proxy Port) and, in order to provide internal dial tone to the Mobile Extension, the incoming CLI of the Mobile Extension must match the number in the Abbreviated Dial bin. Once internal dial tone is presented, the operation is similar to an SLT lifting its handset.

In the absence of DID's, the VRS can be used to transfer the Mobile Extension call to the Mobile Extension extension number. This will provide internal dial tone providing the CLI is presented and matches the number in the associated Abbreviated Dial bin.

Alternatively, if CLI routing is enabled, the relevant Abbreviated Dial bin could be 'transferred' to the Mobile Extension proxy port which would then provide internal dial tone.

Features

The features available from a Mobile Extension are listed below. As the Mobile Extension is based on an SLT port, the service codes used are as per an SLT port. Any feature not listed should be assumed to be not supported:

- Hold
- Transfer
- Incoming Ring Group member
- Department Group member
- DID
- Toll Restriction
- Class of Service
- DSS Keys

  Though DSS keys are available for the Mobile Extension, they cannot provide an exact indication of busy status if, for example, the Mobile Extension is active on a call not linked to the UX5000.

The following service codes are supported:

Type Incoming Feature	**Program**	**Code**	**Set By Mobile Extension**	**Set to Mobile Extension**
Day/Night Mode Switching	11-10-01	818	**Yes**	
Night Mode Switching (other group)	11-10-12	118	**Yes**	
Call Forward - Immediate	11-11-01	No Setting	**Yes**	**Yes**
Call Forward - Busy	11-11-02	No Setting	**Yes**	**Yes**
Call Forward - No Answer	11-11-03	No Setting	**Yes**	**Yes**
Call Forward - Busy/No Answer	11-11-04	No Setting	**Yes**	**Yes**
Call Forward - Both Ring	11-11-05	No Setting	**Yes**	**Yes**

Type Incoming Feature	Program	Code	Set By Mobile Extension	Set to Mobile Extension
Call Forward - Select Option (Answering Machine Emulation not available)	11-11-06	*2	Yes	
Call Forward - Follow Me	11-11-07	No Setting	Yes	Yes
DND	11-11-08	847	Yes	
Message Waiting - Answer	11-11-09	*0	Yes	
Message Waiting - Cancel All	11-11-10	873	Yes	
Automatic Transfer Setup per Extension Group	11-11-25	102	Yes	
Automatic Transfer Cancellation per Extension Group	11-11-26	103	Yes	
Delayed Transfer per Extension Group	11-11-28	105	Yes	
Delayed Transfer Cancellation per Extension Group	11-11-29	106	Yes	
DND Setup per Extension Group	11-11-30	107	Yes	
DND Cancellation per Extension Group	11-11-31	108	Yes	
Pilot Group Withdrawing<	11-11-35	150	Yes	
One Touch Dial Number Entry (dial *0 to hang up)	11-11-39	855	Yes	
VAU/Off-Premise Call Forwarding (common cancelling code (120) required to cancel)	11-11-40	*4	Yes	
Automated Attendant (DSPDB) - not used in the U.S.	11-11-44	No Setting	Yes	
DND/Call Forward Override Call (Bypass Call)	11-12-01	807	Yes	Yes
Conference	11-12-02	#1	Yes	
Override Off-hook Signalling	11-12-03	809	Yes	
Enable Camp-on	11-12-04	850	Yes	Yes
Disable Camp-on	11-12-05	870	Yes	Yes
Voice Call & Signal Call Switching	11-12-06	812	Yes	
Step Call	11-12-07	808	Yes	Yes
Barge-In	11-12-08	810	Yes	Yes
Enable Extension Group to All Ring	11-12-09	No Setting	Yes	
Common/Station Speed Dialling	11-12-10	#2	Yes	
Group Speed Dialling	11-12-11	#4	Yes	
Trunk Group Access	11-12-14	804	Yes	
Specified Trunk Access	11-12-15	805	Yes	
Trunk Access via Networking	11-12-16	No Setting	Yes	
Internal Group Paging (Mobile Extension cannot be a member of a paging group)	11-12-19	801	Yes	
External Paging	11-12-20	803	Yes	
Meet-Me Answer to Specified Internal Paging Group	11-12-21	864	Yes	
Meet-Me Answer to External Paging	11-12-22	865	Yes	
Meet-Me Answer in Same Paging Group (although Mobile Extension cannot be paged)	11-12-23	863	Yes	Yes
Combined Paging	11-12-24	*1	Yes	
Direct Call Pickup own group	11-12-25	856	Yes	Yes
Call Pickup for specified group	11-12-26	868	Yes	Yes
Call Pickup	11-12-27	*#	Yes	Yes
Call Pickup for another group	11-12-28	869	Yes	Yes

Type Incoming Feature	Program	Code	Set By Mobile Extension	Set to Mobile Extension
Direct Extension Call Pickup	11-12-29	**	Yes	
Park	11-12-31	#6	Yes	
Answer Park	11-12-32	*6	Yes	
Group Hold	11-12-33	832	Yes	
Answer Group Hold	11-12-34	862	Yes	
Personal (Extension) Park	11-12-35	857	Yes	
Door Box Access (Door Box can also ring the Mobile Extension. *# operates relay)	11-12-36	802	Yes	
Common Cancel Service Code	11-12-37	120	Yes	
General Purpose Indication	11-12-38	883	Yes	
Voice Mail Center Access - Not used in the U.S.	11-12-39	884	Yes	
Personal Abbreviated Dialing	11-12-40	#7	Yes	
Voice Over	11-12-41	890	Yes	
Flash on Trunk lines	11-12-42	#3	Yes	
Enable SLT On-hook when Holding	11-12-45	849	Yes	
Answer SLT On-hook when Holding	11-12-46	859	Yes	
Call Waiting Answer/Split Answer for SLT	11-12-47	894	Yes	
Account Code	11-12-48	##	Yes	
General Purpose Relay	11-12-50	880	Yes	
Call Own Mailbox (In-skin VM)	11-12-51	*8	Yes	
SLT Live Recording	11-12-53	154	Yes	
ANI/DNIS Routing to VAU	11-12-54	882	Yes	
Tandem Trunking	11-12-57	#8	Yes	
Transfer into Conference	11-12-58	124	Yes	
Enable DND for other extensions	11-14-03	129	Yes	Yes
Disable DND for other extensions	11-14-04	130	Yes	Yes
Enable Wake-up Call for own extension	11-14-05	131	Yes	
Disable Wake-up Call for own extension	11-14-06	132	Yes	
Enable Wake-up Call for other extensions	11-14-07	133	Yes	Yes
Disable Wake-up Call for other extensions	11-14-08	134	Yes	Yes
Enable Room to Room Call Restriction	11-14-09	135	Yes	Yes
Disable Room to Room Call Restriction (Hotel)	11-14-10	136	Yes	Yes
Set Toll Restriction Class for other extensions	11-14-11	137	Yes	Yes
Check-in	11-14-12	138	Yes	Yes
Check-out	11-14-13	139	Yes	Yes
Set Room Status for own extension	11-14-14	140	Yes	
Set Room Status for other extensions	11-14-15	141	Yes	Yes
Room Status Output	11-14-16	142	Yes	
Hotel Room Monitor	11-14-17	175	Yes	Yes

Although some features may be available to the Mobile Extension, it may be advisable to disable them in Class of Service. There are also features that should be disabled in any case.

The features ***to be disabled/not used*** for Mobile Extension include:

- ACD
- TAPI (including applications such as PC Attendant, PC Assistant, etc.)
- H.323 Trunks
- Analog Trunks
- Port Swap
- Hotline
- General Message
- Message Waiting
- Headset Mode for SLT
- Flexible Transfer/Virtual Loop Back
- Tandem Ringing
- Park over CygniLink
- Virtual extension key as Call Coverage Key for mobile extension
- Automatic Conversation Record for trunks

Multiple Directory Numbers / Call Coverage

Feature Availability
• Available - Virtual ports 1-256.

Description

Multiple Directory Numbers let a keyset have more than one extension number. Calls can route to the keyset's installed number or to the keyset's "virtual extension" Multiple Directory Number key. This helps users identify incoming calls. For example, an extension installed at 304 (Sales) could have a virtual extension for 460 (Service). Calls to 304 ring the extension normally. Calls to 460 ring the Multiple Directory Number key. This lets the user at extension 304 differentiate Sales calls from Service calls. After answering a call, based on extension or system-defined options, the call can remain on the Multiple Directory Number key or it can switch to a CALL key, line key or loop key.

Call Coverage

A keyset can have Multiple Directory Number keys set up as Call Coverage keys for co-worker's extensions. The Call Coverage key lights when the co-worker's extension is busy and flashes slowly when the co-worker has an incoming call. The Call Coverage key can ring immediately when a call comes into the covered extension, ring after a delay or not ring at all. In addition, the keyset user can press the Call Coverage key to intercept their co-worker's incoming call. The user can also go off hook and press the Call Coverage key to call the covered extension.

If the covered extension is busy and they receive a second call, the covering extension's Call Coverage key will flash. The user just presses the flashing key to pick up the call.

The Call Coverage keys follow the extension's Do Not Disturb and Off-Hook Signaling programming. These keys do not, however, indicate the lamping for extensions in DND. If this is required, a Hotline key can be used instead.

Place and Receive Calls on Call Coverage/Multiple Directory Number Keys
Multiple Directory Number keys/Call Coverage keys can be used three separate ways, depending on how the key is defined in Program 15-02-21.

- a DSS key to the extension and for receiving incoming calls
- answering incoming calls with the ability to place outgoing ICM or CO calls
  OR
- just for receiving incoming calls

A keyset can have Multiple Directory Number/Call Coverage keys for many different extensions and virtual extensions. In addition, co-workers can share the same Multiple Directory Numbers. For example, everyone in the Service Department could have a key for the Sales Department's virtual extension.

Auto Off-Hook Answer and Ringing Line Preference for Call Coverage Keys
An extension's Call Coverage Keys can be programmed to allow the user to simply pick up the handset to answer a ringing call. So as not to interfere with ringing trunk or Intercom calls, the UX5000 automatically assigns Call Coverage Key ringing with the lowest answering priority. If multiple Call Coverage Keys are ringing, answering priority is set first by the assigned ring pattern and then by the key position.

Virtual Extension vs. Ring Groups
As the UX5000 does not allow voice mail calls to ring Ring Groups, a virtual extension can be created which will allow you to direct more than one call to the extension. The UX5000 will allow up to 10 calls to be queued. When you program a Call Coverage Key for that extension to ring, the next call can then be answered.

This could allow a voice mail caller to dial the virtual extension and have all the extensions with the same virtual extension key ring. The UX5000 can be programmed as follows:

- Program 11-04, 15-01-01: Assign a virtual extension number and name (example: virtual port 1, extension 5400, Sales, virtual port 2, extension 5401, Customer Service, etc.).
- Program 11-07: Assign a Department Group Pilot number for the virtual extension (example: Department Group Number 2 = 5555).
- Program 15-07: Assign a Call Coverage key (*03) to an extension for the virtual extension number assigned.
- Program 16-02: Department Group Assignment for Virtual Extensions (example: virtual extension 5400 - group 2, virtual extension 5401 - group 2).

The end user can then simply transfer the call to the virtual extension number (example: 5555). The call is in placed in queue and will be answered in turn as soon as the extension is available.

Call Forward Off-Premise From CO Trunk/Voice Mail Transfer Possible
The UX5000 allows virtual extensions to use Fixed Call Forward Off-Premise with normal central office trunks. This allows a call transferred by the voice mail to a virtual extension to be forwarded off-premise.

Music on Hold

Feature Availability
• Available.

Description

Music on Hold (MOH) sends music to calls on Hold and parked calls. The music lets the caller know that his call is waiting, not forgotten. Without Music on Hold, the UX5000 provides silence to these types of calls. The Music on Hold source can be a system tone, internal (synthesized) music, or from a customer-provided music source (i.e., tape deck, receiver, etc.). The customer-provided source can connect to a PGDAD module analog port or to a connector on the CCPU blade.

Note: 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 communication servers. NEC America hereby disclaims any liability arising out of the failure to obtain such a license.

Name Storing

Feature Availability
• Available.

Description

Extensions and trunks can have names instead of just circuit numbers. These names show on a keyset's 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 be up to 12 digits long, consisting of alphanumeric characters, punctuation marks and spaces.

To enter a space or accept an entry, the # key is used. However, this key will only work when performing user programming (such as Name Storing for an extension, service code 800). When in UX5000 programming, the right arrow soft key must be used to advance the cursor.

Networking - AspireNet

Feature Availability
• Available - Feature requires a MEMDB be installed on the CCPU. Up to 50 nodes possible.

Description

Use the AspireNet networking feature to integrate multiple phone systems (UX5000 and Aspire) into a single "virtual" UX5000 communications server. Interconnected with ISDN PRI or BRI or

VoIP, each phone system becomes a node on the network that can communicate with any other phone system node.

AspireNet provides seamless feature support for all features connected to the network.

- **Centralized Network Attendant**
  Centralized Network Attendant allows multiple networked systems to share a single centralized attendant. This centralized attendant can receive calls from and transfer calls to any
  destination in any network node. Unanswered calls recall and route as if they were part of a single, much larger system.

- **Flexible Network Routing**
  Use network routes to set up ISDN and VoIP networking between many separate systems - or use mixed (ISDN or VoIP) networking per system for greater network performance. Data tables in the system program define the routing for 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 simply dialing an extension number. The communications server 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 48 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 decides which trunk group to access and any modified dialed data if required.

  When a UX5000 is connected via a VoIP connection to an Aspire system in a tie-line type setup, in order to transfer calls from the UX5000 to the Aspire, in addition to the VoIP programs specified in the VoIP feature (page 191), set up the Flexible Routing Tables as follows:

  - **Program 44-05-01 : ARS/F-Route Table** ; Table Number 1 = 9 (Trunk Group for UX5000 VoIP Trunk)
  - **Program 10-23-02 : H.323 System Interconnection, IP Address** ; System Number 3 = 172.16.9.10 (IP Address for i-Series System)
  - **Program 10-23-04 : H.323 System Interconnection, Alias Address** ; System Number 3 = 4 (For Dial 4 Calls)

  With this programming, the UX5000 will wait for the Trunk Interdigit Timer to expire before dialing out after an i-Series extension (4xxx) is dialed.

  If the F-Routing is set up with Program 44-05-01; Table Number 1 set to 103 (Networking) and Program 10-27-01; System ID 3 = 172.16.9.10 (IP Address for i-Series), though the i-Series system will be able to transfer calls to the UX5000, the UX5000 system will not be able to transfer to the i -Series.

3

PRI AspireNet With Two Local Voice Mails, Masters Must Have Different Numbering

When programming a PRI network with each UX5000 having their own local voice mail, the master numbers for the voice mails must be defined in different series in Program 11-01. The second digit of the extension number can not be the same. For example, 700 and 701 will not work, however 700 and 710 can be used.

With Two Local Voice Mails, Network ID Must Match in Programming

With a networked system, with each system having their own voice mail, in order for users to properly connect to the correct voice mail when leaving a message, the programming must be set as described below.

<u>PRI AspireNet</u>
Each node on a networked UX5000 is defined in **Program 10-03-11 : Blade Setup - Networking System Number** for the desired PRI slot. In the local system (Site 1), **Program 11-01-01 : System Numbering** has a digit defined for networking (example: 7x=8 (network)).

The ID entry in Program 10-03-11 must match the ID set in Program 11-01-01. Otherwise, callers could not press "8" to leave a message at a networked user's voice mail. For example:

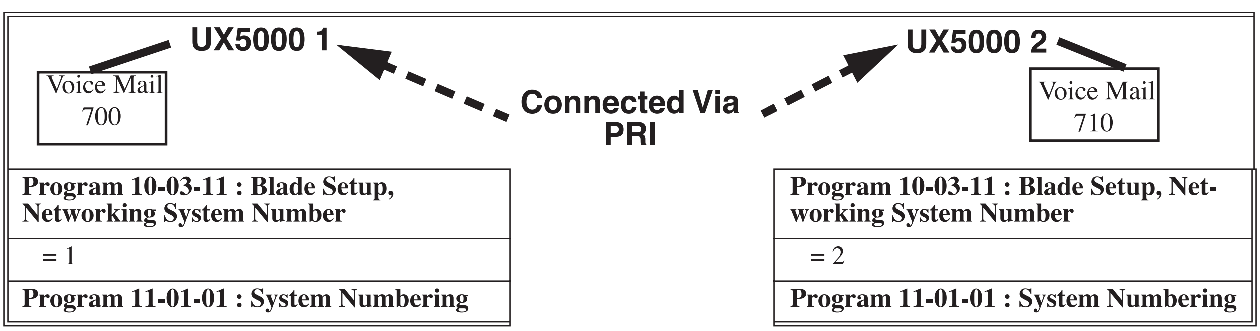
- In Site 1 in **Program 10-03-11 = 1**, **Program 11-01-01 7X=8, ID = 2**.
  When Site 1's extension 301 calls Site 2's extension 401, then presses digit "8", the user would hear a reorder tone.
- In Site 1 in **Program 10-03-11 = 1**, **Program 11-01-01 7X=8, ID = 1**.
  When Site 1's extension 301 calls Site 2's extension 401, then presses digit "8", the user will hear the voice mail message for extension 401.

Note that in order to assign a system ID, the "type" must temporarily be set to "8" (networking). Once the system ID has been assigned, you can change the type to the required entry (2).

Programming

1. Set Program 11-01-01 to "8" (networking) for the local and remote voice mail master numbers and define the ID number.
2. Set the local system ID in Program 10-03-11 to the same entry defined in Program 11-01-01.
3. Change Program 11-01-01 for the local voice mail master number back to "2" (extension number).

Refer to the example below:



Digit 7: 71 = 3 Digits Type 8 (Networking) System ID: 1 70 = 3 Digits Type 2 (Extension) System ID: 1 (must temporarily change to Type 8 to be able to enter the system ID)	Digit 7: 71 = 3 Digits Type 2 (Extension) System ID: 2 (must temporarily change to Type 8 to be able to enter the system ID) 70 = 3 Digits Type 8 (Networking) System ID: 2

IP AspireNet

Each node on an IP networked UX5000 is defined in **Program 10-27-01 : IP System ID**. For example, Site A has ID 1 and Site B system has ID 2 in Program 10-27-01. ***This must be the same in every node.***

In **Program 11-01-01 : System Numbering**, if Site 1 system (system ID 1) has 700 as the master number, the entry for "70" would be "2".
Site 2 system (system ID 2) has 710 as the master number, the entry for "71" would also be "2".

Note that in order to assign a system ID, the "type" must temporarily be set to "8" (networking). Once the system ID has been assigned, you can change the type to the required entry (2).

The system ID defined in Program 11-01-01 is where the voice mail master number resides. Refer to the example below:

UX5000 1	UX5000 2
Program 10-23-01 : H.323 System Interconnection	**Program 10-23-01 : H.323 System Interconnection**
System 1: Interconnect=Yes (1) IP Address: System 1	System 1: Interconnect=Yes (1) IP Address: System 1
System 2: Interconnect=Yes (1) IP Address: System 2	System 2: Interconnect=Yes (1) IP Address: System 2
Program 11-01-01 : System Numbering	**Program 11-01-01 : System Numbering**
Digit 7: 71 = 3 Digits Type 8 (Networking) System ID: 2 70 = 3 Digits Type 2 (Extension) System ID: 2 (must temporarily change to Type 8 to be able to enter the system ID)	Digit 7: 71 = 3 Digits Type 2 (Extension) System ID: 2 (must temporarily change to Type 8 to be able to enter the system ID) 70 = 3 Digits Type 8 (Networking) System ID: 1

Multiple Site AspireNet Available

IP AspireNet allows for a maximum of 50 nodes. With ISDN AspireNet, the maximum number of nodes depends on the type of trunks used - PRI, BRI or a mixture of both.

The recommended connection for multiple system networking is to interconnect all the UX5000s using hardware (required with ISDN AspireNet) and programming. This also allows such features as Park to be used in a network.

With VoIP AspireNet, the system IDs for each networked system are defined in **Program 10-27 : IP System ID**. An IP address is defined for each node, and using the Numbering Plan (Program 11-01xx), the dialed digits are analyzed, the system ID is determined, and the call is routed to the networked system.
With ISDN AspireNet, interconnecting the systems is accomplished by defining a master blade and slave blade between each system (**Program 10-03-10 : Blade Setup - Master/Slave System**). Each system must have an ISDN blade for each other system in the network.

The networked systems can be interconnected using a combination of PRI and VoIP trunks. In this type of setup, the IP resources will received priority over the PRI resources. The PRI will be used when the IP resources are busy. The following two graphics indicate correct networking layouts. Each one provides connection between all three systems within the network.

ISDN AspireNet Clock Source
When using ISDN AspireNet (PRI or BRI) on the UX5000, the option selected in **Program 10-03-01 : Blade Setup - ISDN Line Mode** determines the clock source for the networked connection. The following information indicates how, with each option, the clock source is obtained for the networked systems.

<u>Option 3: Network Mode (Leased Line)</u>
Telco sends the clock to the Master System
Telco sends the clock to the Slave System

<u>Option 4: Network Mode (Interconnected Line)</u>
Master System sends the clock to the Telco (or direct connection without telco) which then sends the clock to the Slave System

<u>Option 5: (Interconnected Line, Fixed Layer 1=NT)</u>
Master System sends the clock to the Telco
Slave System sends the clock to the Telco

Keep Alive Programs
Two options in Program 10-31 for use with the AspireNet feature are available to define the interval of Keep Alive and how many times the system resends Keep Alive.

PRI Channel Limitation
The system provides an option which can be used to assign the number of B-channels to be used for each ISDN blade. This allows for fractional PRIs when used with multiple site networking. If this program is limited to less than "23" on one side of the network, then it also limits both inbound and outbound network calls. This also applies on the other side of the network as well.

Transfer Network Trunk to Local Voice Mail Using Voice Mail Key
IP or PRI network sites that have their own voice mail can transfer a call into voice mail using the following steps if the inbound call originated in another site (for example, a call comes in to Site A and it translates to an extension at Site B):

- HOLD + the Voice Mail Programmable Function Key (Program 15-07 or SC 851 + 77) + the extension number.

Keep the following in mind when using this option:

- ***Note that if you have a local and central voice mail, you can not have the same mailbox number at each node.*** If both sites have the same mailbox, when transferring a call, it will be transferred into the local voice mail and not across the network.
- As this software change allows transferring to a general message by a Single Line Set, the following two operations have different results between Networking Call Transfer and Internal Call Transfer.
  - Hold + Voice Mail Master Dial then hang up.
  - Hold + Service Code (Own Mailbox Access - *8) then hang up.

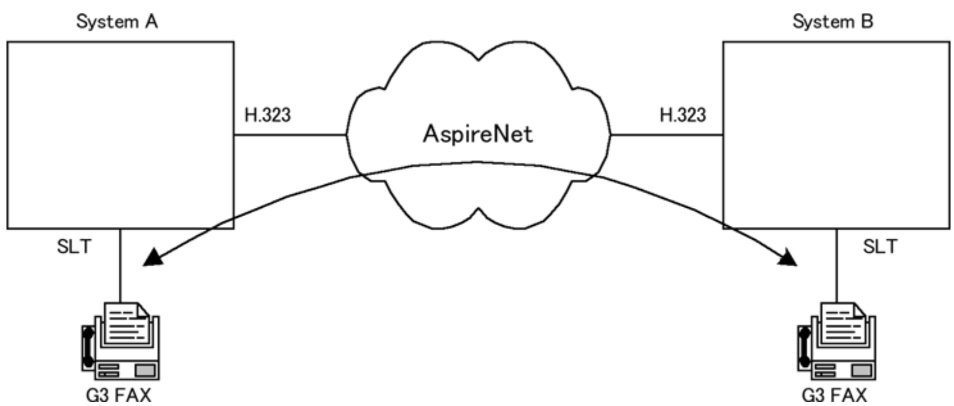
Fax Over AspireNet

The purpose of AspireNet is to be able to connect several systems and have them appear to operate as one system. However, some restrictions apply. With Fax Over AspireNet using H.323 trunks, if a resource is busy, the operation may not be performed efficiently. Although the operation continues if there is no G.711 compression, there is no resending procedure with RTP. The software enhances this operation to provide better performance.

With IP AspireNet, the modem signal of the fax relay uses H.245. This enhancement only applies to G3.

Examples:

FAX Relay with SLT Extension (G3)

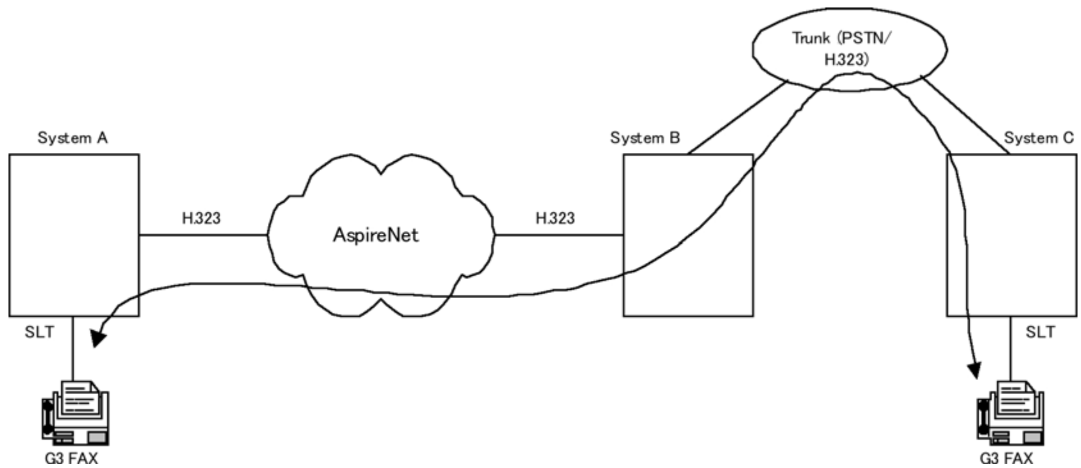


Setup:

- System A
  Program15-03-03 = G3 Fax 1 (special)
  Program 84-12-32 = 2 (Each Port Mode)

- System B
  Program 15-03-03 = G3 Fax 1 (special)
  Program 84-12-32 = 2 (Each Port Mode)

FAX Relay with Trunks



Setup:

- System A
  Program15-03-03 = G3 Fax 1 (special)
  Program 84-12-32 = 2 Each Port Mode

- System B
  Program 84-01-59 = 2 special
  Program 84-12-32 = 2 Each Port Mode

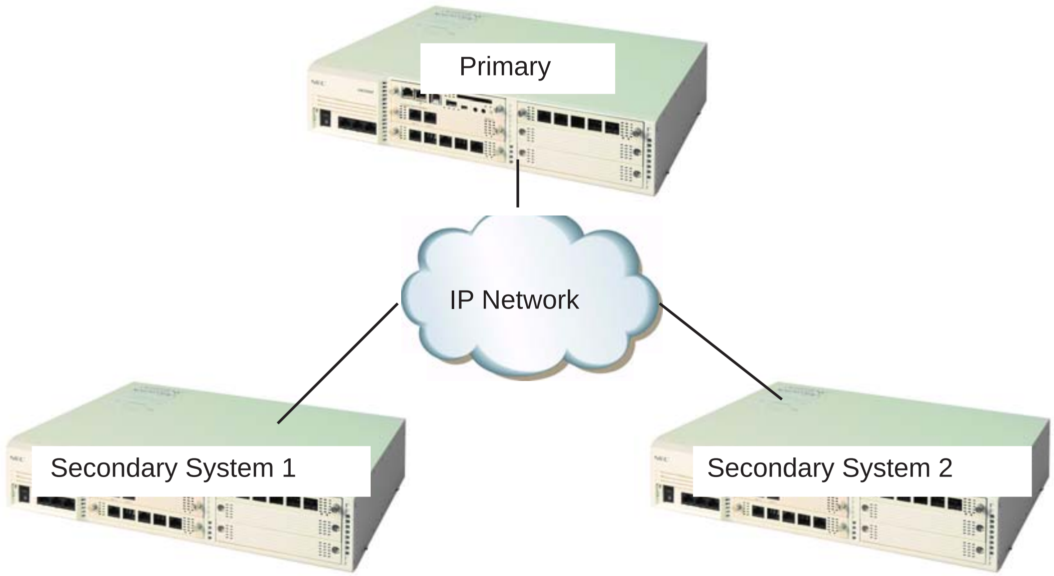
- System C
  Program15-03-03 = G3 Fax 1 (special)
  Program 84-01-59 = 2 Each Port Mode

For additional information on networking, refer to the Aspire Networking Guide (P/N 0893207).

Networking - CygniLink

Feature Availability
• Available - Feature requires a VOIPDB and a MEMDB be installed on the CCPUs. Up to 16 nodes possible.

Description



Use the built in CygniLink networking feature to integrate multiple communication servers into a single "virtual" communications system. Interconnected with ISDN PRI or BRI or VoIP, each communications server becomes a node on the network that can communicate with any other communications server node.

With CygniLink, the Fail-Over ability will allow the UX5000 to switch to another CCPU defined in programming, should there be a power outage/failure with the main CCPU. The networked communication servers will reboot (this takes approximately 30 seconds) and when they come back up are linked to the new Primary System.

CygniLink provides seamless feature support for all features connected to the network.

For additional information on networking, refer to the UX5000 CygniLink Networking Guide (P/N 0913207).

Night Service

Feature Availability
• Available - 32 Night Service Patterns/Groups and 8 time modes.

Description

Night Service lets UX5000 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 UX5000 also provides external contacts to enable Night Service.

There are eight Night Service modes:

- Day 1 / Day 2 Modes - for normal working hours
- Night 1 / Night 2 Modes - after hours (usually evening)
- Midnight 1 / Midnight 2 Modes - late at night to early in the morning
- Rest 1 / Rest 2 Modes - interval usually used for lunch

Assigned Night Answer (ANA)
With Assigned Night Answer, Night Service has calls ring extensions directly. Assigned Night Answer provides an answering point for Night Service calls. For certain applications, this may be more appropriate than Universal Night Answer. For example, you could program trunks to ring the security station terminal during off hours.

Universal Night Answer (UNA)
Universal Night Answer makes incoming calls ring over the External Paging speakers. With UNA, an employee can go to a terminal and press the flashing line key or use “Universal Answer” to pick up the call. Only ring groups calls can be used with Universal Night Answer. For more on setting up Universal Answer, turn to the “Central Office Calls, Answering” feature.

You may also be able to use Transfer to UNA. An extension user can Transfer their call to UNA (i.e., External Paging at night). Once transferred, the call will ring the External Paging speakers like any other UNA call and can be picked up at any extension.

Automatic Night Service
The UX5000 will allow or deny Automatic Night Service based on the extension’s class of service programming. If allowed, the calls will then route according to the service patterns programmed. The Night Service programming is stored in the RAM memory. This means that if the UX5000 is not using the Automatic Night Service, in the case of a power failure while in night mode, when the power is restored, the UX5000 will continue to be in night mode.

Programmable Function Key Can Toggle Night Modes
The software allows a Night Service Programmable Function Key (PGM 15-07 or SC 851: 09 + 0) to toggle night modes. You can determine in programming how many modes through which the user will toggle. Note that the additional data for the Programmable Function Key must be set to "0" for the toggle function to work.

Off Hook Signaling

Feature Availability
• Available.

Description

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 UX5000 provides the following Off Hook Signaling options:

- **Called Extension Block**
  The called extension’s Class of Service may block incoming Off Hook Signaling attempts. This is beneficial to users that don’t want interruptions while on a call.

- **Automatic Signaling**
  Calling a busy extension automatically initiates Off Hook Signaling. This option is useful to receptionists, operators and others that must quickly process calls. This is set in the called extension's Class of Service.
- **Manual Signaling**
  After reaching a busy extension, manual signaling gives the caller the choice of using Off Hook Signaling or activating other features. Extension's without automatic signaling have manual signaling. The users can dial a service code or press a Programmable Function Key to send Off Hook Signaling to the called terminal.
- **Selectable Off Hook Signaling Mode**
  The Off Hook Signal can be muted ringing, no off hook ringing or a beep in the handset - based on the caller's programming.
- **Off Hook Ringing**
  Use this option to enable or disable an extension's Off Hook Signaling for incoming calls. If enabled, Off Hook Signaling occurs normally. If disabled, calls queue behind the extension's busy line appearance and the user gets no Off Hook Signaling indication. The second line appearance stays idle. The caller hears ringback tone while their call waits. This is set in the called extension's Class of Service.
- **DID Call Waiting**
  An extension can optionally have DID calls camp on with Off Hook/Call Wait signaling, without Off Hook/Call Wait signaling or no signaling. This is set in the called extension's Class of Service.
- **Block Manual Off Hook Signals**
  This Class of Service option enables/disables a busy extension's ability to block off hook signals manually sent from a co-worker. If disabled (not blocked), callers can dial 7 at busy or busy/ring to signal the extension. If enabled (blocked), nothing happens when the caller dials 7 to off hook signal.
- **Block Camp On**
  If an extension has Block Camp On enabled, callers to the extension cannot dial 2 to Camp On after hearing busy or busy/ring. If the extension has Block Camp On disabled, callers are not prevented from dialing 2 to Camp on after hearing busy or busy/ring. This is set in the called extension's Class of Service.

One-Touch Calling

Feature Availability
• Available.

Description

One-Touch Calling, using a Programmable Function Key, gives a keyset user one button access to extensions, trunks and selected UX5000 features. This saves users time when accessing co-workers, clients and features they use most often. Instead of dialing a series of codes, the user need only press the One-Touch Key. An extension user can have One-Touch Keys programmed for:

- **Direct Station Selection** - one button access to extensions
- **Personal Abbreviated Dialing** - one button access to stored numbers (up to 24 digits long)
- **Abbreviated Dialing** - one button access to stored Abbreviated 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 Programmable Function Keys. For example, a user can store the number for a company's Automated Attendant in key 13 and employee extension numbers in keys 14-18. The user presses key 13 to call the company, then one of keys 14-18 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 Personal Speed Dial bins. The stored Flash may be helpful to access features of the connected telco, PBX or Centrex.

Operator

Feature Availability
• Available - up to 8 operators.

Description

When an extension user dials "0", calls are routed to a main UX5000 operator. The operator can answer and route outside calls or locate employees using the Page feature.

Paging, External

Feature Availability
• Available - 9 External Paging zones.

Description

With External Paging, a user can broadcast announcements over paging equipment connected to external Paging zones. When a user pages on of these external zones, the UX5000 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 UX5000 allows up to nine External Paging zones, with the additional zone (#9) provided on the CCPU. All other zones (#1-8) requires a port on a 2PGDAD module, with a maximum of two external paging circuits per module. You must have four 2PGDAD modules 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. Refer to the UX5000 hardware manual for additional details.

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's warehouse and outside loading dock at the same time. Combined Paging is available for zones 1-8 and All Call. Refer to **Paging, Internal** (page 157) for more on setting up Combined Paging.

Remove Paging Information from Display Terminals
A Class of Service option is available in UX5000 programming to prevent display terminals from showing incoming paging information. This allows the UX5000 to save processor time and speed up communications server operation.

Paging, Internal

Feature Availability
• Available - 64 Internal Paging Groups (Zones).

Description

Internal Paging lets extension users broadcast announcements to other keyset 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's 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.

Remove Paging Information from Display Terminals
A Class of Service option is available in UX5000 programming to prevent display terminals from showing incoming internal paging information. This allows the UX5000 to save processor time and speed up UX5000 operation.

Paging, Privacy Release

Please refer to Conference, Voice Call/Privacy Release (page 93) for information on this feature.

Park

Feature Availability
• Available - 64 System Park Orbits.

Description

Park places a call in a waiting state (called a Park Orbit) so that an extension user may pick it up. There are two types of Park: System and Personal. Use System Park when you want to have the call wait in a system orbit. Personal Park allows a user to Park a call at their extension so a co-worker can pick it up. After parking a call in orbit, a user can Page the person receiving the call and hang up. The paged party dials a code or presses a programmed Park key to pick up the call. With Park, it is not necessary to locate a person to handle their calls. A call parked for too long will recall the extension that initially parked it, however the call remains in the park orbit until it's answered. There are 64 Park Orbits (1-64) available for use.

If Caller ID is enabled in the UX5000, a user can check the Caller ID for a parked call using their Park key and possibly the Flash key (depending on programming).

Splitting Between Parked Calls

A keyset user can retrieve two calls from Park Orbit (for which they don't have line appearances) and easily split (alternate) between them. The split operation brings the calls to the user's terminal and frees up the Park Orbits.

Extended Park

An extension's Class of Service determines whether it will use the normal Park Orbit Recall time or the Extended Park Orbit Recall time. The timers are set up in UX5000 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's answered.

Automatic Park Search for Idle Park Orbit Feature

The Park feature provides an Automatic Park Search feature for use with ***display*** terminals. Without this option, the user needs to be aware of which Park orbit is available. With the Automatic Park Search feature enabled, the UX5000 will automatically Park a call in the first available orbit (the UX5000 searches in ascending order for an idle orbit). The user's display will temporarily indicate which orbit the call was parked.

Notes:

- If all Park orbits are busy, the user will hear a busy signal and the call will not be parked.
- The Automatic Park Search is within the local UX5000 only. It cannot check networked communication servers.
- Single line terminals, non-display keysets, and cordless terminal users should not use this feature as they can not determine which orbit the call was parked.

Personal Park Options

Programmable Function Key and Service Code Available for Personal Park

The Personal Park feature provides the ability to use a Programmable Function Key (15-07-01, code *07) or service code (3-digit or 1-digit) (11-16-11) to place a call in Personal Park. This option is available for keysets, single line sets, and IP DECT terminals and can be used for analog or ISDN trunks.

With this feature, the following conditions apply:

- An extension can have only one Personal Park key.
- When the terminal that has a call in Personal Park is unplugged, the Personal Park will be released and the held caller will receive busy tone.

The following table indicates what condition the service codes and Programmable Function key can be used.

Status	Using 3-Digit Service Code	Using 1-Digit Service Code	Using Personal Park Key
Speaking	Not Available	Not Available	Available
ICM Dial Tone or Busy Tone	Available	Not Available	Available
Calling Another Extension	Available	Available (with outside call on hold and when called extension does not answer)	Available
Receiving a Personal Park Recall	Available	Not Available	Available

Personal Park at a Co-Worker’s Extension

The Personal Park feature allows an extension user to place an outside call, which is on hold, on Personal Park at a co-worker’s extension after placing an intercom call. This feature is available for keysets, SLTs, IP terminals and IP DECT terminals.

Notes:

- If the called extension has Call Forwarding enabled, the outside call is parked at the originator’s extension instead.
- This feature is not available when calling a Department Group’s pilot number.
- If an extension user has a call in Personal Park and the terminal is unplugged, the Personal Park will be cancelled and the held caller will hear a busy tone.
- This feature will not work when calling a Networked or virtual extension.
- If an extension has currently placed a trunk in another extension’s Personal Park, the feature cannot be used for a new trunk until the initial trunk has been picked up.
- A Personal Park Programmable Function Key or the Soft Key must be used to park the call in a co-worker’s park. This operation cannot be done using a service code.

PBX Compatibility

Feature Availability
• Available.

Description

You can connect your communications server trunks to Centrex/PBX lines, rather than to telco trunk circuits. This makes the trunk inputs into the UX5000 500/2500 type compatible Centrex/ PBX extensions, rather than telco circuits. PBX Compatibility lets the UX5000 be a node (i.e., satellite) in a larger private telephone network. To place outside calls when the UX5000 is behind a PBX, UX5000 users must first dial the PBX's trunk access code (usually 9).

The UX5000 provides the following PBX Compatibility options:

- **PBX Trunk Access Code Screening**
  The UX5000 can monitor the numbers users dial and screen for PBX trunk access codes. The UX5000 can screen for up to 4 groups of trunk access codes. The codes can be one or two digits long, consisting of the digits 0-9, # and *. (You use Line Key 1 as a wild card entry.)

- **PBX Trunk Toll Restriction**
  The UX5000 can provide the Toll Restriction for the PBX trunk, or restriction can be handled solely by the connected PBX. If the UX5000 provides the restriction, it restricts the digits dialed after the PBX access code.

- **PBX Call Restriction**
  When the UX5000 does the Toll Restriction, it can further restrict users from dialing PBX extensions. In this case, the only valid numbers are those dialed after the PBX trunk access code. The only PBX facility UX5000 users can access are the PBX's outside trunks.

- **Automatic Pause**
  The UX5000 automatically pauses when it sees a PBX trunk access code during manual dialing, Abbreviated Dialing, Last Number Redial, Repeat Redial and Save Number Dialed. This gives the connected PBX time to set up its trunk circuits.

Prime Line Selection

Feature Availability
• Available.

Description

Prime Line Selection allows a keyset user to place or answer a call over a specific trunk by just 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 -- see Programming below.)

- **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's service calls and then dispatch repair technicians. The dispatcher would have the assurance than whenever a customer calls in, the dispatcher just lifts the handset get their call. (Incoming Prime Line Preference can optionally seize an idle line appearance -- see Programming below.)

Privacy Release

Please refer to Conference, Voice Call/Privacy Release (page 93) for information on this feature.

Private Line

Feature Availability
• Available.

Description

A Private Line is a trunk reserved for a keyset 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 UX5000.

- **Incoming only**
  The keyset has a Private Line only for incoming calls. The user cannot place calls on the Private Line.

- **Outgoing only**
  The keyset has a Private Line only for outgoing calls. The Private Line does not ring for incoming calls.

- **Both ways**
  The keyset has a Private Line for both incoming and outgoing calls.

Programmable Function Keys

Feature Availability
• Available.

Description

Each keyset has Programmable Function Keys. Programmable Function Keys simplify placing calls, answering calls and using certain features. You can customize the function of a keyset's programmable keys from your administration terminal, or the extension user can do it themselves. The number of Programmable Function Keys depends on your terminal style.

Refer to Tables 4 and 5 in the beginning of this section for the Programmable Function Key functions.

Pulse to Tone Conversion

Feature Availability
• Available.

Description

An extension can use Pulse to Tone Conversion on trunk calls. Pulse to Tone Conversion lets a user change their extension's dialing mode while placing a call. For UX5000s 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 UX5000 dials the digits after the conversion as DTMF.

Repeat Redial

Feature Availability
• Available.

Description

If a keyset user places a trunk call that is busy or unanswered, they can have Repeat Redial try it again later on. The user doesn't continually have to try the number again -- hoping it will go through. Repeat Redial automatically retries it until the called party answers (the number of retries is based on UX5000 programming).

Reverse Voice Over

Feature Availability
• Available.

Description

While on a handset call, Reverse Voice Over lets a busy keyset user make a private Intercom call to an idle co-worker. The idle co-worker can be at a keyset or 500/2500 set. The busy user just presses and holds down 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 releases 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 keyset is idle, the Reverse Voice Over key functions the same as a Hotline key. A keyset's 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

Note: When the keyset 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.

Ring Groups

Feature Availability
• Available - 100 Ring Groups.

Description

Ring Groups determine how trunks ring extensions. Generally, trunks ring extension's 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's Ring Group programming, assign ringing for the trunk.

Any number of extensions and trunks can be in a specific group. The UX5000 allows Ring Groups=1-100, In-Skin Voice Mail (102), or Centralized Voice Mail (103).

If an extension has a line key for the trunk, Ring Group calls ring the line key. If the extension doesn't 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

Feature Availability
• Available - 512 extensions/virtual extensions and 512 Hotline assignments.

Description

With a Ringdown Extension, a user can call another extension, outside number, or Abbreviated Dialing number by just lifting the handset. The call automatically goes through - there is no need for the user to dial digits or press additional keys. Ringdown Extensions are frequently used for lobby terminals, 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

Feature Availability
• Available.

Description

Room Monitor lets an extension user listen to the sounds in a co-workers area. For example, the receptionist could listen for sounds in the warehouse when it's left unattended. To use Room Monitor, the initiating extension ***and*** the receiving extension must activate it.

When using keysets for monitoring, an extension user can only Monitor one extension at a time. However, many extensions can Monitor the same extension at the same time.

With single line terminals, multiple SALT's can be programmed to be monitored by the same SLT, however, only one SLT can monitor another SLT at a time.

Room Monitor for Single Lines

This option enables you to monitor the room status through your single line terminal. This can be used with the Hotel/Motel feature as well. Between keysets, the monitored room status is picked up by the terminal's microphone and the activity is heard through the speaker of the monitoring keyset. Between single line terminals, a user goes off hook on the monitored terminal and, from another single line terminal, dials a service code and the extension number. The activity of the area where the monitored terminal is placed can then be heard at the monitoring terminal. This service is available until the handset of the monitored terminal 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

Feature Availability
• Available.

Description

Save Number Dialed permits an extension user to save their last outside number and easily redial it later on. For example, an extension user can recall a busy or unanswered number without manually dialing the digits. The UX5000 retains the saved number until the user stores a new one in its place.

Save Number Dialed saves in memory a dialed number up to 24 digits. The number can be any combination of digits 0-9, # and *. The UX5000 remembers the digits regardless of whether the call was answered, unanswered or busy. The UX5000 normally uses the same trunk group as for the initial call. However, the extension user can preselect a specific trunk if desired.

Secretary Call (Buzzer)

Feature Availability
• Available.

Description

Secretary Call lets two co-workers alert each other without disturbing their work. To have Secretary Call, both co-workers must have keysets with Secretary Call buzzer keys. When a user presses their buzzer key, the UX5000 alerts the called extension by sending a splash tone and flashing the called extension's buzzer key. The called user can respond by placing an Intercom call to the calling party. The called extension's 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's 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

Feature Availability
• Available.

Description

Secretary Call Pickup lets a keyset user easily reroute calls intended for a co-worker to themselves. By pressing a Secretary Call Pickup key, the user can have all calls to a co-worker's terminal ring or voice-announce theirs instead. Secretary Call Pickup is a simplified type of Call Forward with Follow Me for employees that work closely together. This feature could be helpful to customer service representatives that must frequently cover each other's clients. When a representative leaves their desk, an associate could press the Secretary Call Pickup key to intercept all their calls.

An extension can have Secretary Call Pickup keys for any number of extensions, limited only by the available number of programmable keys.

Secure Set Relocation

Please refer to Maintenance (page 131) for information on this feature.

Selectable Display Messaging

Feature Availability
• Available - all terminals are able to use Selectable Display Messaging at one time.

Description

An extension user can select a preprogrammed Selectable Display Message for their extension. Display keyset callers see the selected message when they call the user's extension. Selectable Display Messaging provides personalized messaging. For example, an extension user could select the message "GONE FOR THE DAY". Any display keyset 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 phone number after messages 1-8 and 10 (up to 24 characters). For example, an extension user could select the message "ON VACATION UNTIL" and then enter the date. Callers see the original message followed by the appended date. They would then be able to tell when the user was coming back from vacation. The UX5000 allows all terminals to use the Selectable Display Messaging feature at the same time.

The default messages are:

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 (phone 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

3

Feature Availability
• Available.

Description

An extension user can change the way trunks or ICM calls ring their terminal. Selectable Ring Tones allows an extension user to set up unique ringing for their calls. This is important in a crowded work area where several terminals are close together. Because their terminal has a characteristic ring, the user always can tell when it's their terminal ringing.

Serial Call

Feature Availability
• Available.

Description

Serial Call is a method of transferring a call so it automatically returns to the transferring extension. Serial Calling saves transferring steps between users. For example, a Customer Service Representative (CSR) has a client on the terminal who needs technical advice. The CSR wants to send the call to Tech Service, but needs to advise the client of certain costs when Tech 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 terminal. I need to talk to him again. Just hang up when you're done and I'll get him back."

Single Line Terminals, Analog 500/2500 Sets

Feature Availability
• Available. Capacity based on blades installed and port licensing (max. 001-512).

Description

The UX5000 is compatible with 500 type (Dial Pulse) and 2500 type (DTMF) analog single line terminals (SLTs). You can install single line terminals as On-Premise or Off-Premise extensions. Single line terminal users can dial codes to access many of the features available to keyset users. With Single Line Terminals, you can have your UX5000 simulate PBX type operation.

When installing single line terminals, you must have:

- A port on an SLIU blade for each single line terminal installed.
- (If you have 2500 sets) At least one block reserved on the CCPU for analog extension DTMF reception.

Codec Filter Data Setup Program

When **Program 82-07-01 : Codec Filter Setup for Analog Station Ports** is set to "4 - Specified Data", the UX5000 will use the settings in **Program 82-09 : SLIU Codec Filter Data Setup**.

These values should not be changed from their default settings unless directed by NEC'S Technical Service department.

The side tone of the SLIU is adjusted using all 16 values, however, special software is required in order to compute these values. The setting is not proportional to the gain level. To change these values, contact NEC's Technical Service department for assistance.

DTMF Dial Out Timer

A program is available for DTMF dialing, **Program 20-03-07 : System Options for Single Line Terminals**. When **Program 20-03-03 : System Options for Single Line Terminals - SLT DTMF Dial to Trunk Lines** is set to "0" (receive all digits before sending), the UX5000 will following the timers in Program 20-03-04 and 23-03-07.

The timer in **Program 20-03-04 : System Options for Single Line Terminals - Dial Sending Start Time for SLT or ARS** will reset when the user dials another digit.

The timer in **Program 23-03-07 : System Options for Single Line Terminals - Forced Dial Sending Start Time** will not reset when a digit is dialed. The user must finish dialing all the digits before this timer expires (Entries: 0-64800 seconds, Default: 0).

Protocol for External Analog Devices (ex: Fax Server) Supported

An analog station port can support the sending of DTMF tones and disconnect signal to support an external fax server or other similar products.
With a digital voice mail installed, this option allows the UX5000 to send protocol digits to an analog SLT port (this allows an external fax server to be used as well).

Note that this feature is created for SLT ports and not for a voice mail group. If an SLT port is assigned to a voice mail group, the existing analog voice mail protocol will be sent.

- DTMF digits can be sent only when a call is forwarded. The forwarding extension can be a keyset, single line terminal, or virtual extension.

- The extension can be set for Call Forwarding options: All, Busy, No Answer, or Busy/No Answer.
- The forwarding destination can be a SLT or Department Group.
- This feature is not available for networked ports.

Soft Keys

Feature Availability
• Available.

Description

Each display terminal provides interactive soft keys for intuitive feature access. It is no longer necessary to remember feature codes to access the terminal's advanced features because the function of the soft keys change as the user processes calls. For example, just press a soft key to Page, Park a call, leave a message or Camp On to a busy co-worker.

Station Message Detail Recording

Feature Availability
• Available. This feature requires a connection to the UX5000 using the LAN port on the UX5000 CCPU (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.), or a CTA or CTU adapter connected to an Aspire keyset.

Description

Station Message Detail Recording (SMDR) provides a record of the UX5000's 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 UX5000 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's application.

- **Customized Date Format**
The SMDR header can show the report date in one of three formats: American, European or Japanese. Set the format for your preference.

- **Transferred Call Tracking**
SMDR shows each extension's share of a transferred call. If an outside call is transferred among four extensions, SMDR shows how long each of the callers stayed on the call.

- **Data Call Tracking**
Data Call Tracking can log the UX5000's internal data calls. Since SMDR normally logs external (trunk) data calls, Data Call Tracking lets you get a complete picture of data terminal activity.

- **Digit Counting**
With Digit Counting, SMDR can selectively keep track of toll calls. For example, if the digit count is nine, SMDR won't include toll calls within the home area code. Digit Counting permits SMDR to include only the types of calls you want to monitor.

- **Digit Masking**
Digit Masking lets you "X" out portions of the number dialed on the SMDR report. A digit mask of seven, for example, masks out all exchange codes (NNXs) and local addresses. Digit Masking makes it easier to keep track of calling patterns, without having to interpret each individual number. You can also use Digit Masking to block out access and security codes.

- **Duration Monitoring**
SMDR can include calls of any duration, or only those that last longer than the interval you specify. If you want to keep track of all trunk activity, use a short duration. To keep track of only significant usage, use a longer duration.

- **Extension Exclusion**
You can selectively exclude extensions from the SMDR report. This ensures privacy for high-profile callers. For example, the company attorney negotiating a merger may not want his calls to show up on an in-house report.

- **PBX Call Reporting**
If your UX5000 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.

- **USB SMDR Communication**
SMDR communication to the UX5000 requires a LAN connection. However, when using an Aspire keyset on the UX5000, you can attach a CTA adapter to the Aspire keyset allowing for serial or USB output. The UX5000 is compatible with USB SMDR devices. You can output the SMDR report to a high speed printer or send it to disk through a PC's 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's name or extension number. Choose the method that makes it easier for you to track call usage.

SMDR with Caller ID

The SMDR output can include up to 16 or 24 characters of the Caller ID name information (depending on the view option selected in Program 35-02-18). You can select to display the Caller ID number or name or the DID number. If you wish to display the Caller Name in the "DIALLED NO./CLI" and "ACCOUNT" area, select "2" in the updated Program 35-02-15 and "1" in Program 35-02-17.

If the Caller ID name is not received, the area for Caller ID Name is left blank.

Sample SMDR Report

For example, with Program 35-01-09 = 0 (Format for NA) and Program 35-02-17 = 1 (Caller ID Name), if a call is received with the Caller ID Name of "NECinfrontia Corporation" (24 characters), the following SMDR record is displayed:

CLASS	TIME	DATE	LINE	DURATION	STATION	DIALLED No./CLI	ACCOUNT
POT	10:52	12/09	002	00:00:10	2001	2142623801	08754
PIN	10:52	12/09	001	00:00:20	2017	2142623802	NECinfrontia Cor
PIN	10:53	12/09	002			2142623801	NO ANSWER

If Program 35-02-18 = 1 (Caller ID Name Output Method) is set to line feed, the SMDR will display as follows:

CLASS	TIME	DATE	LINE	DURATION	STATION	DIALLED No./CLI	ACCOUNT
POT	10:52	12/09	002	00:00:10	2001	2142623801	08754
PIN	10:52	12/09	001	00:00:20	2017	2142623802	NECinfrontia Cor
NEXT NECinfrontia Corp.							
PIN	10:53	12/09	002			2142623801	NO ANSWER

External Call Forward Setting Display

A program option (Program 35-02-20) for the SMDR reports is available which determines which information is displayed in the "STATION" area for a transferred call when the extension has Call Forward set with an Abbreviated Dial number as the destination. You can choose to display the extension number which **called** the extension with external Call Forward set or display the extension number which **has the external Call Forward set**.

This option only applies when Call Forward is set using a service code (Program 11-11-01~11-11-07) and the destination uses an Abbreviated Dial bin. It does not include Off-Premise or Centrex transfers.

Definitions	
Call Record Number	SMDR record number (consecutive)
CLASS	Type of call (see Class Definitions below)
TIME	Time call placed or answered. (For Transferred calls, shows time user picked up Transfer.)
DATE	Date the call was made
LINE	Trunk number used for call. This field can show both the trunk number (or trunk name) and the DID number based on the setting in Program 35-02-16.
DURATION	How long call lasted. (For Transferred calls, shows how long user was on call after answering the Transfer.)
STATION	Extension number of call “owner” (i.e., extension that first placed or answered call) (For Transferred calls, there can be more than one owner - depending on how many extensions shared the call.)
DIALLED No./CLI	For outgoing calls, the number dialed or, for incoming calls, the Caller ID information. The "DIALLED No./CLI" field can print up to 20 digits. Programming determines whether the report shows the first 20 digits or the last 20 digits.
COST	For UX5000 with ARS, indicates the call cost
OR	
ACCOUNT	Account Code number entered by extension user
Class Definitions	
POT	Outgoing trunk call
POTA	Outgoing trunk call placed using Toll Restriction Override
PIN	Incoming trunk calls
ALB	All lines in group are busy (group number follows TIME field)
BRD	Call blocked due to Toll Restriction
PTRS	Transferred call
IVIN	BRI trunk call

SMDR Report Format with Program 35-02-14 Set to “0”	
Character Position	**Field Definition**
Header Line 1	
1-60	Spaces
61-70	MM/DD/YYYY
71	Space
72-75	PAGE
76	Space
77-79	Report page number (e.g., 001)
CR & LF	Carriage return and line feed
Header Line 2	
1-5	CLASS
6	Space

SMDR Report Format with Program 35-02-14 Set to "0"	
Character Position	**Field Definition**
7-10	TIME
11-14	Spaces
15-18	LINE
19-22	Spaces
23-30	DURATION
31-32	Spaces
33-39	STATION
40-44	Spaces
45-51	DIALLED
52	Space
53-59	No./CLI
60-63	Spaces
64-70	ACCOUNT
CR & LF	Carriage return and line feed
LF	Line feed
SMDR Record	
1-4	Call type (e.g., POT for outgoing)
5	Space
6-10	Time in 24 hour clock (HH:MM)
11	Space
12-21	LINE
22	Space
23-30	Call Duration (HH:MM:SS)
31	Space
32-41	Station number or name
42	Space
43-62	Number dialed (20 digits maximum)
63	Space
64-79	Account number or NO ANSWER

SMDR Report Format with Program 35-02-14 Set to "1"	
Character Position	**Field Definition**
Header Line 1	
1-60	Spaces
61-70	MM/DD/YYYY
71	Space
72-75	PAGE
76	Space
77-79	Report page number (e.g., 001)
CR & LF	Carriage return and line feed
Header Line 2	
1-5	CLASS
6	Space

3

SMDR Report Format with Program 35-02-14 Set to "1"	
Character Position	**Field Definition**
7-10	TIME
11	Spaces
12-15	DATE
16-17	Spaces
18-21	LINE
22	Space
23-30	DURATION
31-32	Spaces
33-39	STATION
40-44	Spaces
45-51	DIALLED
52	Space
53-59	No./CLI
60-63	Spaces
64-70	ACCOUNT
CR & LF	Carriage return and line feed
LF	Line feed
SMDR Record	
1-4	Call type (e.g., POT for outgoing)
5	Space
6-10	Time in 24 hour clock (HH:MM)
11	Space
12-16	DATE
17	Space
18-21	LINE
22	Space
23-30	Call Duration (HH:MM:SS)
31	Space
32-41	Station number or name
42	Space
43-62	Number dialed (20 digits maximum)
63	Space
64-79	Account number or NO ANSWER

Summary Reports

```
OUTGOING CALL/COST SUMMARY
FOR DAY OF nn/nn/nn

    TOTAL NO. OF OUTGOING PSTN CALLS:0
    TOTAL NO. OF OUTGOING ISDN CALLS:0
    NO. OF OUTGOING PSTN CALLS COSTED:0    COST:0
    NO. OF OUTGOING ISDN CALLS COSTED:0    COST:0

OUTGOING CALL/COST
SUMMARY FOR WEEK ENDING nn/nn/nn
    TOTAL NO. OF OUTGOING PSTN CALLS:49
    TOTAL NO. OF OUTGOING ISDN CALLS:0
    NO. OF OUTGOING PSTN CALLS COSTED:49   COST:0
    NO. OF OUTGOING ISDN CALLS COSTED:0    COST:0

OUTGOING CALL/COST SUMMARY
FOR MONTH ENDING nn/nn/nn
    TOTAL NO. OF OUTGOING PSTN CALLS:49
    TOTAL NO. OF OUTGOING ISDN CALLS:0
    NO. OF OUTGOING PSTN CALLS COSTED:49   COST:0
    NO. OF OUTGOING ISDN CALLS COSTED:0    COST:0
```


Synchronous Ringing

Feature Availability
• Available.

Description

Synchronous Ringing synchronizes CO/PBX incoming ringing with the incoming ringing pattern from a Central Office. This feature can be used for keysets or single line terminals (but not SLT's connected to an APR module or IP terminals).

T1 Trunking (with ANI/DNIS Compatibility)

Feature Availability
• Available - ANI/DNIS Compatibility is available.

Description

The T1/PRI Interface blade gives the UX5000 T1 trunking capability. This blade uses a single universal slot and provides up to 24 trunk circuits. In additional to providing digital-quality trunking, the T1/PRI Interface blade allows you to have maximum trunking capability with fewer blades. This in turn makes more universal slots available for other functions.

You can program each T1/PRI blade for any combination of the following trunks:

- CO loop start
- CO ground start
- Direct Inward Dialing
- Tie lines [1]

Tandem Ringing

Feature Availability
• Available.

Description

Tandem Ringing allows an extension user to have two terminals with one phone number. For example, extension 305 (the master terminal) sets Tandem Ringing with extension 306. When extension 305 receives an incoming call, both extension 305 and 306 ring. Callers would dial the master extension number (extension 305 in this example). When either the master terminal or slave terminal is in use, the other terminal cannot be used for outgoing calls - incoming calls, however, will ring the available terminal.

The keyset must be paired with either an analog single line terminal. It cannot be paired with another keyset, IP terminal, or Cordless terminal.

Tandem Trunking (Unsupervised Conference)

Feature Availability
• Available - CCPU provides 2 blocks of 32 conference circuits, allowing each block to have any number conferences with any number of internal or external parties conferenced as long as the total number of conference channels used does not exceed the block's limit of 32.

Description

Tandem Trunking allows an extension user to join two or more 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 reenter the conversation or end the tandem call at any time.

The number of simultaneous Conference calls is limited by the number of Conference circuits in the UX5000. Due to this fact, the maximum number of Conference calls cannot exceed the limits defined in the above table.

1. Two-wire (four-lead) type 1 tie lines (FIC TL11M) only.

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 at any time.

There are four methods for Tandem Trunking:

- **Method A - Set Up Without Transfer Key**
  An extension user can set up Tandem Trunking (Unsupervised Conference) with 2 or more outside calls by dialing a two-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 2 or more calls they have placed on Hold. It uses a uniquely programmed Transfer key to set up a tandem call.
  Note: The operation for connecting multiple outside callers is different than when only connecting 2 outside callers - the call can be on hold and, after calling the second party, the Transfer key can be pressed. With more than 2 outside calls, the CONF key must be pressed in order to connect the callers.
- **Method C - Automatic Tandem Trunking on Hang Up**
  This method allows an extension user to easily set up a 2-party Unsupervised Conference without having to place the conference call on Hold. ***This option cannot be used for Multiple Tandem Trunking calls (containing more than 2 outside parties).*** A Class of Service option is available which will allow or deny an extension user from automatically setting up a Conference/Tandem Trunking call upon hanging up the terminal.
- **Method D - Automatic Tandem Trunking Setup to Abbreviated Dial Number**
  This method allows an extension user to easily set up a 2-party Unsupervised Conference with a call they have placed on Hold. ***This option cannot be used for Multiple Tandem Trunking calls (containing more than 2 outside parties).*** A Class of Service option is available which will allow or deny an extension user from automatically setting up a Conference/Tandem Trunking call upon hanging up the terminal.

Trunk Disconnect Continue/Disconnect Codes

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 will extend the conversation a programmed length of time. If the user enters the Disconnect service code, the call will be disconnected immediately.

Example:

The following example indicates how a call will be handled with the UX5000 programmed as follows:

- Program 14-01-25: 1
- Program 20-28-01: #
- Program 20-28-02: No setting
- Program 20-28-03: 180
- Program 24-02-07: 600 (Only used with Tandem Trunking)
- Program 24-02-10: 30 (Only used with Tandem Trunking)
- Program 25-07-07: 600 (Only used with DISA)
- Program 25-07-08: 30 (Only used with DISA)

3

1. An external call connects to an external number (either by transferring with Tandem Trunking or by DISA caller).
2. After 10 minutes (Tandem Trunking = Program 24-02-07 or DISA = Program 25-07-07), a warning tone is heard and the user dials "#" (Program 20-28-01) to extend the conversation.
3. After 3 minutes (Program 20-28-03), the warning tone is heard again. After 30 seconds (Tandem Trunking = Program 24-02-10 or DISA = Program 25-07-08), the call is disconnected.

Tie Lines

Feature Availability
• Available.

Description

Tie lines directly link a local communications server with one or more remote systems. The link is independent of the telco's switched network. When a local system user seizes a tie line, they hear Intercom dial tone from the remote system. The user may then be able to:

- Dial extensions in the remote system
- Use the remote system's trunks for outgoing calls
- Access Common Abbreviated Dialing bins in the remote system
- Use the remote system's Internal and External Paging

The UX5000 provides connection for 2-wire (four lead, tip/ring) or 4-wire (eight lead, tip/ring/tip 1/ring 1) type tie line circuits. Using UX5000 programming, each circuit type can be set as Type I or V.

Tie Line Class of Service

Tie Line Class of Service provides features and dialing restrictions for incoming tie lines. This allows you to control the capabilities of callers dialing into your UX5000. The tie line Class of Service options are:

- **First Digit Absorption**
  A tie line can ignore (absorb) the first digit received, which helps when setting up a tie line network. For example, your UX5000 can have tie lines to two other systems with the same extension numbering plan. Use the first digit to differentiate between the systems. Tie line callers can dial 3200-3456 for the first system's extensions and 4200-4456 for the second system's extensions. The receiving system ignores the first digit and routes calls correctly to the extension dialed (i.e., 4301 is received as 301).

- **Trunk Group Routing/ARS Access**
  When a tie line user calls the remote system, they may be able to dial 9 and place outside calls through the remote system. Any toll charges are incurred by the remote system. The call follows the remote system's Trunk Group Access or Automatic Route Selection - whichever is enabled.

- **Trunk Group Access**
  Tie line callers may be able to access trunk groups in the remote system by dialing Programmable Function Key *02 and the trunk group number. This allows the callers to select a specific trunk group for an outgoing call. Trunk Group Access bypasses the remote system's

Trunk Group Routing/ARS. As with dial 9 access, any toll charges are incurred by the remote system.

- **Common Abbreviated Dialing**
The remote system's Common Abbreviated Dialing bins may be available to tie line callers. Use this capability to set up centralized Abbreviated Dialing control - or just save time when dialing.
- **Operator Calling**
A tie line caller may be able to dial 0 for the remote system's operator.
- **Paging**
Internal and External Paging may be available to tie line callers. This allows co-workers in adjacent facilities connected by tie lines, for example, to broadcast announcements to each other.
- **Direct Trunk Access**
This option allows tie line callers to directly access a trunk for an outside call by dialing #9 and the trunk's number. Like Trunk Group Access, this bypasses the remote system's Trunk Group Routing/ARS. Any toll charges are incurred by the remote system.
- **Forced Trunk Disconnect**
The Forced Trunk Disconnect option allows a tie line caller to disconnect (release) another extension's active outside call. The tie line caller can then place a call on the released trunk. Tie line callers should use Forced Trunk Disconnect only in an emergency, when no other trunks are available.
- **DISA/Tie Trunk Barge In**
The DISA/Tie Trunk Barge In option allows a tie line caller to break into another extension's established call. This sets up a three-way conversation between the intruding party and the two parties on the initial call.

Tie Line Outgoing Call Restriction

You can selectively deny incoming tie lines access to your UX5000's outgoing trunk groups. Incoming tie line callers could be able to access your outgoing WATS lines, for example, but not your DID trunks. The UX5000 allows you to set up a restriction matrix for each of your incoming tie lines - for each of your outgoing trunk groups.

Tie Line Toll Restriction Class

Incoming tie lines can have a Toll Restriction Class and be subject to the UX5000's toll restriction. For example, Toll Restriction can prevent users from dialing 1-900 calls. When an incoming tie line caller tries to use UX5000 trunks to dial a 1-900 service, Toll Restriction will deny the call.

Flexible Tie Line Service Compatibility

You can individually program tie lines for Dial Pulse (DP) or DTMF incoming or outgoing signaling. Outgoing tie lines can be either wink start or immediate start.

Wink Start Mode Does Not Require a DTMF Receiver

E&M trunks in wink start mode do not require a DTMF receiver for dial tone detection. It is detected with the wink. This frees the DTMF receivers for other use.

Time and Date

Feature Availability
• Available.

Description

The UX5000 uses Time and Date for:

- Central Office Calls (Access Maps)
- Class of Service (Class)
- Direct Inward Lines
- Display Terminals
- Fax Machine Compatibility
- 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 UX5000 should automatically adjust the UX5000 time for daylight savings time/standard time changes.

Clock Adjustment

The UX5000 can be programmed to automatically adjust the UX5000 clock on a nightly basis. This feature allows you to make adjustments should the UX5000 chassis regularly lose or gain time.

Toll Restriction

Feature Availability
• Available - 15 Toll Restriction Classes. Capacity depends on the number of blades installed and the system port licensing.

Description

Toll Restriction limits the numbers an extension user may dial. By allowing extensions to place only certain types of calls, you can better control long distance costs. The UX5000 applies Toll Restriction according to an extension's Toll Restriction Class. The UX5000 allows for up to 15 Toll Restriction Classes.

Toll Restriction offers the following capabilities:

- **Common Permit Code Table**
  Use the Common Permit Code Table when you have numbers you want all Toll Restriction Classes to dial. To let all users dial 911, for example, put 911 in the Common Permit Code Table. The Common Permit Code Table overrides the Restrict Code and Common Restrict Code Tables. The UX5000 provides 10 tables, with 10 entries in each table. Each code is 4 digits max., using 0-9, #, * and FLASH (as a wild card).

- **Common Restrict Code Table**
The Common Restrict Code Table lets you globally restrict certain numbers for all Toll Restriction Classes. To prevent all users from dialing directory assistance (411), for example, put 411 in the Common Restrict Code Table. Be sure you don't allow the codes you want to restrict in the Permit Code Table or the Common Permit Code Table. The UX5000 provides 10 tables, with 10 entries in each table. Each code is 4 digits max., using 0-9, #, * and FLASH (as a wild card).

- **Restrict Code Table**
When you want Toll Restriction to allow most calls and restrict only selected calls, use the Restrict Code Table. To block only 1-900 calls, for example, enter 1900 in the Restrict Code Table. (If the same Toll Restriction Class has both Permit and Restrict Code Tables, the UX5000 restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The UX5000 provides 4 tables, with 60 entries (restricted codes) in each table. A restricted code is 12 digits maximum, using 0-9, #, * and FLASH (as a wild card).

- **Permit Code Table**
The Permit Code Table lets you set up Toll Restriction so that users can dial only selected (permitted) telephone numbers. Use this table when you want to restrict most calls. To allow all users to dial only area code 203, for example, enter 1203 in the Permit Code Table. 1 + 203 + NNX + nnnn are the only numbers users can dial. (If the same Toll Restriction Class has both Permit and Restrict Code Tables, the UX5000 restricts calls that you enter only in the Restrict Code Table. Calls entered in both tables are not restricted.) The UX5000 provides 4 tables, with 200 entries (permitted codes) in each table. A permitted code is 12 digits maximum, using 0-9, #, * and FLASH (as a wild card).
- **International Call Restriction**
International Call Restriction lets you limit the international calls an extension user may dial. You can build a restrict table to prevent only certain calls, or you can build a permit table to allow only certain calls. To allow most international calls, use the *International Call Restrict Table*. To prevent most international calls, use the *International Call Allow Table*. The UX5000 provides 10 International Call Restrict tables with up to 4 digits in each table entry and 20 International Call Allow tables, with up to 6 digits in each table entry. Valid entries are 0-9, #,* and FLASH (for a wild card).

- **Toll Restriction for Abbreviated Dialing**
Abbreviated Dialing can bypass or follow Toll Restriction. If you allow many users to program Abbreviated Dialing, consider Toll Restricting the numbers they dial. If only administrators can program Abbreviated Dialing, Toll Restriction may not be necessary. You can separately restrict Group and Common Abbreviated Dialing.

- **Call Digit Counting**
Use Call Digit Counting to limit the number of digits local callers can dial. You can use this option to prevent users from accessing local dial-up services. For example, set the Maximum Number of Digits in Local Calls to 7 to limit local callers to dialing the exchange code (NNX) and local address (nnnn) only. The UX5000 provides 4 tables in which you can make entries for this option. The range is 4-30 digits.

- **Toll Call Digit Counting**
With Toll Call Digit Counting, you can limit the number of digits long distance callers can dial. This lets you prevent callers from dialing extensively into long distance dial-up services. You can make four entries (4-30 digits).

- **Toll Free Trunks**
  Certain trunks can be completely unrestricted, such as the company president's Private Line. Users can place calls on Toll Free Trunks anytime -- to anywhere, without inadvertently being toll restricted.

- **PBX Call Restriction**
  Toll Restriction programming lets you enable/disable PBX Call Restriction and enter PBX access codes. You only need to do this if your UX5000 is behind a PBX and you have trunks programmed for behind PBX operation. Refer to PBX Compatibility feature for the specifics.

- **Tie Line Toll Restriction**
  With **Program 34-01-05 : E&M Tie Line Basic Setup - System Toll Restriction** set to '0', the UX5000 will follow the setting in **21-05-13 : Toll Restriction Class - Restriction of Tie Line Calls** to determine whether or not the toll restriction setting in Program 34-08 is to be followed (only 20 entries are allowed in this option). If this option is set to '1', the UX5000 will follow the UX5000 toll restriction settings defined in Program 21-05-01 through 21-05-13.

Toll Restriction Level for Trunks

The UX5000 allows a trunk to be assigned a Toll Restriction level. When both an extension and a trunk have a Toll Restriction level assigned, the higher class will apply for outgoing calls. For example:

- When a trunk is set to class 1 and an extension is class 02, Toll Restriction class 02 is applied to the outgoing call.
- When a trunk is set to class 15 and an extension is class 03, Toll Restriction class 15 is applied to the outgoing call.

This feature can be used for any type of extension (real or virtual) and using any type of terminal (keyset, SLT, etc.). When virtual extensions are to be used, Program 15-02-21 must be set to "1" to allow outgoing calls on a virtual/Call Coverage key.

When DISA and Tie Line trunks are used, the restriction class for the incoming trunk is compared to the restriction class of the outgoing trunk.

When a trunk makes an outgoing call, the restriction class of the incoming trunk (Program 21-21-01) is compared to the restriction class of the outgoing trunk. The higher class will be used for outgoing calls.
DISA Trunk (22-02-01 is set to "2") - Program 25-11-01 is compared to 21-21-01.
Tie Line Trunk (22-02-01 is set to "5") - Program 34-04-01 is compared to 21-21-01.

Networking does not support Trunk Toll Restriction.

Toll Restriction Override

Feature Availability
• Available.

Description

Toll Restriction Override lets a user temporarily bypass an extension's Toll Restriction. This helps a user that must place an important call that Toll Restriction normally prevents. For example, you could set up Toll Restriction to block 900 calls and then provide a Toll Restriction Override code to your attendant and executives. When the attendant or executive needs to place a 900 call, they just:

- Press CALL1, dial a service code and enter their override code.
- Press a line key or dial a trunk access code (e.g., 9 or #9 002).
- Place the 900 call without restriction.

You can assign a different Toll Restriction Override code to each extension. Or, extensions can share the same override code.

Toll Restriction Override will override ***all*** Toll Restriction programming. Walking Toll Restriction allows you to assign a Toll Restriction level for each user. When a call is placed using Walking Toll Restriction, the restriction for the call is based on the Toll Restriction level defined in Programs 21-05-xx and 21-06-xx.

Toll Restriction, Dial Block

Feature Availability
• Available.

3

Description

Toll Restriction Dial Block lets a user temporarily block an extension's Toll Restriction. This helps a user block his or her terminal from being used by another person while they are away from their desk. A user would need to enter a 4-digit personal code to enable/disable this feature.

Dial Block can also be set by the system administrator. If Dial Block has already been set by an extension user, the supervisor can not release it. Additionally, if Dial Block has been set by the supervisor, an extension user can not 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.

Traffic Reports

Feature Availability
• Available.

Description

The UX5000 provides the ability to send data to a PC connected to the UX5000. The terminal call traffic data for each extension is captured for use with the SMDR feature.

Transfer

Feature Availability
• Available.

Description

Transfer permits an extension user to send (i.e., extend) an active Intercom or outside call to any other extension in the UX5000. 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 UX5000 can optionally play ringback tone or Music on Hold to the caller.

The UX5000 allows the following types of transfers:

- **Screened Transfer**
  The transferring user announces the call to the destination before hanging up
- **Unscreened Transfer**
  The transferring party extends the call without an announcement.
- **Extension (Department) Groups Transfer**
  The Transferring party sends the call to a Department instead of an extension.
- **Transfer Without Holding**
  A user presses a busy line key and waits for the call to complete. The UX5000 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 304 can answer a trunk, press HOLD, dial 305 and hang up. The UX5000 extends the call to extension 305. Without Automatic On-Hook Transfer, the call would stay on Hold at extension 304 when the user hangs up. To extend the call, the user at extension 304 would have to press CONF or a Transfer function key 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 will allow 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 keyset or single line terminal user with Barge In capability the ability to transfer a call into an existing call. This call can be a 2-party call, a Conference call, or a Barge In Conference. The UX5000 allows Intercom, analog trunk, ISDN trunk and H.323 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.
This feature is not supported across a network or with S-Bus ports (this includes transferring an S-Bus call into a conference or transferring a call into a conference which includes an S-Bus port).

Transfer to Trunk Ring Group Available

Software allows a user to transfer a DID or trunk call to the trunk's defined ring group (defined in **Program 22-05-01 : Incoming Trunk Ring Group Assignment**). The trunk will then ring 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 terminal.

To enable this feature, the UX5000 has a program option, **Program 11-15-09 : Service Code Setup Administrative (for Special Access) - Transfer to Trunk Ring Group Code**. When a call is transferred using this service code, it's 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 will ring all extensions programmed for Ring Group 4 or ring the External Paging Group for Ring Group 4, depending on how the UX5000 is programmed.

Program 22-04-01 : Extension Ring Group Assignment and **Program 22-05-01 : Incoming Trunk Ring Group Assignment** must be programmed to allow an extension access to the ring groups. If the call is not answered, it can overflow to the destination defined in **Program 22-08-01 : DIL/IRG No Answer Destination**.

This service code can also be used with the VRS. This provides the caller listening to the VRS message with the ability to transfer their call and have it ring the external page. The code the caller would dial is defined in **Program 25-06-02 : VRS/DISA One-Digit Code Attendant Setup**.

Step Transfer for Automatic Trunk-to-Trunk Transfer Feature

With the Automatic Trunk-to-Trunk Transfer feature, if the destination to which a trunk has been transferred receives no answer (following the timer set in Program 24-02-12), the UX5000 can then automatically Step Transfer the call to a new destination. Up to 8 different destination numbers can be defined (Program 14-01-26).

Notes:

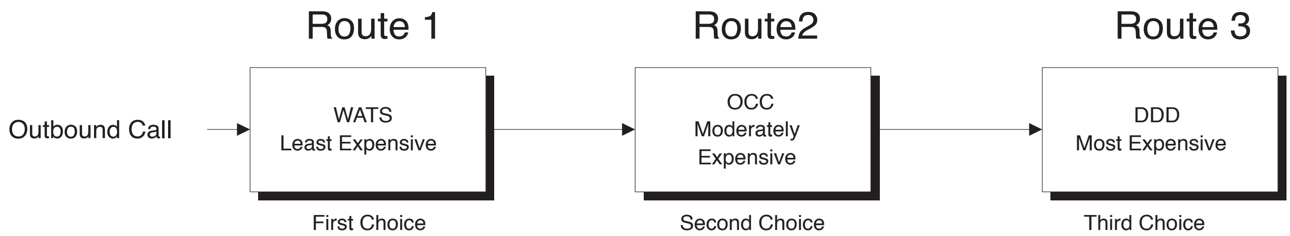
- With DID trunks, if the DID Transfer Destination (Program 22-11-04) is allowed, if the Automatic Trunk-to-Trunk Transfer programming is defined, the DID trunks will follow the Step Transfer.
- If the Step Transfer reaches the last defined destination for a call and there is still no answer, the call will continue to ring the last number - it will not restart dialing the first destination.

Trunk Group Routing

Feature Availability
• Available - 100 trunk groups and 100 routes.

Description

Trunk Group Routing sets outbound call routing options for users that dial the Trunk Group Routing code (9) for trunk calls. Trunk Group Routing routes calls in the order specified by UX5000 programming. If a user dials 9 and all trunks in the first group are busy, the UX5000 may route the call to another group. When you're setting up your UX5000, Trunk Group Routing will help you minimize the expense of toll calls. For example, if your UX5000 has outbound WATS lines, OCC lines and DDD lines, use Trunk Group Routing to route calls to the WATS lines first.



Trunk Groups

Feature Availability
• Available - 100 trunk groups.

Description

Trunk Groups let you optimize trunk usage for incoming and outgoing calls. With Trunk Groups, users can have loop (rotary) keys for trunk calls. Incoming trunk group calls ring these loop keys. For outgoing calls, the user presses a loop key to access the first available trunk within the group. You set the access order in trunk group programming.

Loop keys give an extension user more available function keys, since the user doesn't need a separate line key for each trunk. The user only needs one loop key for each trunk group. This simplifies placing and answering calls.

Like Trunk Group Routing, Trunk Groups help you minimize the expense of toll calls. For example, if your UX5000 has outbound WATS lines, OCC lines and DID 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

Feature Availability
• Available.

Description

Trunk Queuing permits an extension user to queue (wait in line) on hook for a busy trunk or trunk group to become free. The UX5000 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 UX5000 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 UX5000 connects the extensions in the order that the requests were left.

Ultra CallAnalyst

Ultra CallAnalyst is an easy to use, graphically oriented software package that allows you to monitor and analyze phone calls, understand phone usage and cut costs. CallAnalyst tracks both incoming and outgoing calls accurately as well as the date and time of each call. If you need to track the incoming phone call with name and/or telephone numbers, CallAnalyst requires Caller ID services from the local phone company.

Ultra CallAnalyst increases productivity, facilitates billing and helps detect toll fraud and phone abuse. It also has powerful tabular (text) and graphical report generating capabilities. Reports include extension/line summaries, date/time and department summaries, longest/most expensive calls, most frequently called numbers, and other commonly used summaries. Ultra CallAnalyst also has the ability to automatically generate and Email reports to managers on set schedules. These reports can be used to analyze your telephone as a critical business communication tool, thereby, improving its effectiveness and helping you to reduce your telephone related costs.

CallAnalyst keeps track of:

- The date and time calls were made or received.
- The duration of each call.
- Which extension made or received the call.
- The CID/ANI, DNIS of the caller.
- The trunk or line numbers which handled the call.
- Account codes and authorization codes used for the call.

As an example; a report can be generated showing calling patterns by volume or duration on a color coded map of the United States. This can help a Customer Support, Sales Order, or Telemarketing business become more focused, more productive and cost effective.

Ultra CallAnalyst can be networked to allow for collecting and generating telephone usage reports for multiple telephone systems on a network. It is a scalable call accounting solution for the small to mid-size business.

Ultra CallAnalyst is an easy to use, graphically oriented software package that allows you to monitor and analyze phone calls, understand telephone usage and cut costs. Ultra CallAnalyst tracks both incoming and outgoing calls accurately.

The application supports connection of heterogeneous (different type) telephone systems and supports real time consolidation and reporting of call accounting data. If you need to track the incoming telephone traffic with calling name and/or telephone numbers, Ultra CallAnalyst requires Caller ID services from the local phone company.

Main Server Minimum PC Requirements
Hardware

- PC with Pentium 4 Processor
- 512 MB RAM
- SVGA Monitor with 1024 x 768 resolution
- 2 GB of free hard drive space
- CD-ROM drive (for software installation)
- Network Interface Card (NIC)
- Available serial port and RS-232 cable (if required)
- Printer (if required to print reports)

Software

- Windows 2000 Professional w/SP4, XP Professional w/ SP2, 2003 Server w/SP2, or Vista Business Edition
- Database Server:
  - Microsoft SQL Server 2005
  - Microsoft SQL Server Express 2005 (SP2 or later required)
  - MS-SQL Server 2000 (SP4 or later required)
  - MSDE (Microsoft Database Engine) for the database
    Microsoft does not recommend or support MSDE on Vista platforms.
- Microsoft Internet Explorer 6.0 or higher
- Internet Information Services (IIS) (if you intend to install Web Reports)

Network Client / Remote Site Reporting Client Minimum PC Requirements
Hardware

- PC with Pentium III Processor
- 256 MB RAM
- 1 GB of free hard drive space
- CD-ROM drive (for software installation)
- Network Interface Card (NIC)
- Available serial port and RS-232 cable (if required)

Software

- Windows Vista Business Edition, XP Professional w/ SP2 or later, 2003 Server w/SP2 or later, or 2000 Professional w/ SP4 or later
- Microsoft Internet Explorer 6.0 or higher

Universal Answer

Please refer to Central Office Calls, Answering (page 86) for information on this feature.

Voice Mail

Feature Availability
• Available

Description

Voice Mail ends the frustration and cost of missed calls, inaccurate written messages and telephone tag. This frees a company's busy receptionists and secretaries for more productive work.

For complete details on the voice mail options, refer to the following manuals:

- UX5000 IntraMail Manual (P/N 0913240)
- UX5000 UX Mail Manual (P/N 0913250)
- UX5000 Multibutton Feature Handbook (P/N 0913400)
- UX5000 Multibutton Quick Reference Guide (P/N 0913401)

Voice Over

Feature Availability
• Available.

Description

- Important -
Do not use Voice Over to a user on speakerphone as the conversation may be heard by the outside party.

Voice Over lets a user interrupt a keyset extension user busy on another call. With Voice Over, the busy keyset extension user hears an alert tone followed by the voice of the interrupting party. The keyset extension user can respond to the interrupting party without being heard by the original caller. If desired, the keyset extension 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's paralegal could announce the urgent call as soon as it comes in. The lawyer could then give the paralegal instructions how to handle the situation - all without the original client hearing the conversation.

Either a keyset or 500/2500 set user can initiate a Voice Over, but only a keyset user can receive a Voice Over.

To enable Voice Over, a keyset should 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	Listening to the interrupting party
On	Responding to the interrupting party

Voice Response System (VRS)

Feature Availability
• Available - 16 Channels (shared with voice mail).

Description

The IntraMail daughter board provides the option for Voice Response System (VRS) which gives the UX5000 voice recording and playback capability. The VRS CompactFlash card provides up to 100 system messages (General Message, Automated Attendant greetings, ACD messages, and the 900 Preamble). In addition, the Personal Greeting and Park & Page options can have messages (note that the Park & Page feature uses 2 messages). This enhances the UX5000 with:

- **General Message** - provides a prerecorded message to which any user can listen
- **Personal Greeting** - lets an extension user record a message and forward their calls. Callers to the extension hear the recorded message and are then redirected.
- **Park and Page** - parks a call at an extension and automatically pages the user to pick it up
- **Automated Attendant (Operator Assistance)** - answers incoming calls, plays a greeting to the caller and then lets the caller directly dial a system extension
- **ACD Messages** - provides announcement and overflow messages for ACD groups
- **Transfer to the VRS -** 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 keyset extension user quickly hear a recording for the time, date, or the extension’s number.

VRS Messages

The VRS allows you to record up to 100 VRS messages. You allocate these messages for Automated Attendant greetings, the General Message, ACD messages and the 900 Preamble message.

Any on-premise extension caller can listen, record and erase VRS Messages (unless restricted in programming). DISA and DID callers can listen and record VRS messages (unless restricted in programming).

VoIP

Feature Availability
• Available - 512 IP extensions maximum, 128 IP trunks maximum. • The UX5000 provides a maximum of 32, 64, or 128 VoIP channels (depending on the VOIPDB used). • IP terminals using the Lock button must be programmed to allow emergency numbers (such as 911) to be dialed. Refer to the 911 section of this feature for details.

Description

VoIP (voice over Internet protocol or voice over IP) allows the delivery of voice information using the Internet protocol (sending data over the Internet using an IP address). This means that voice information, in a digital form, 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 within the company are received and routed to other parts of the company's intranet (local area or wide area network) or they can be sent over the Internet using CO lines to another gateway.[1]

The UX5000 VoIP supports H.323 trunks and compressions of G.711, G.722, G.723.1, G.726, G.729AB, and iLBC. The UX5000 is IEEE 802.3af compliant for PoE.

Note: G.723 is not supported with SIP multiline terminals.

Program Available for Gain Setup of VOIPDB Daughter Board

The UX5000 software provides an option to adjust the gain setting for the VOIPDB daughter board.

Calling Party Number Setup for Trunks and Extensions

The UX5000 provides two programs which allow the programmed entry of the Calling Party Number. These entries determine the information displayed when VoIP trunks and extensions are used.

Using LANs

Using a LAN setup (local area network) with the UX5000 complies with the ethernet standard (10Base-T/100Base-TX).

To connect a terminal to a LAN connection, the UX5000 allows the use of an UX5000 digital IP keyset, SIP terminal.

If connecting a LAN to a WAN (wide area network), follow the instructions included with the ADSL modem or gateway device.

1. The voice quality of VoIP is dependent 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's control, it cannot guarantee the performance of the user's IP-based remote voice solution. Therefore, NEC recommends connecting VoIP equipment through a local area network using a Private IP address.

IP Address

Equipment/devices used in the UX5000 LAN setup must have an IP address assignment. An IP address assigns a unique address for each device. There are two types of IP addresses: Private and Global. A Private IP address is not accessible through the internet - a Global IP address can be accessed through the internet.

With a Private IP address, with equipment that does not access the internet directly, addresses can be assigned to the equipment within Class A, B or C by assigning a number within the class's range of numbers.

Class	Allowed IP Address	Recommended Environment
A	10.0.0.0 ---10.22.255.255	Large Scale Network
B	172.16.0.0 --- 172.31.255.255	Mid Scale Network
C	192.168.0.0 --- 192.168.255.255	Small Scale Network

With a Global IP Address, connected equipment can be accessed through the internet, so each address must be unique. To avoid a conflict, the addresses are controlled by ARIN (American Registry for Internet Numbers). To obtain a Global IP Address, contact ARIN or apply with your local ISP (internet service provider).

The first one to three groups of numbers (depending on the subnet mask) identify the network on which your computer is located. The remaining group(s) of numbers identify your computer on that network.

Subnet Mask

As the IP Address includes information to identify both the network and the final destination, the Subnet Mask is used to set apart the network and destination information.

The default subnet masks are:

Class	Default Subnet Mask
A	255.0.0.0
B	255.255.0.0
C	255.255.255.0

In the above table, you'll see that the Subnet Mask is made up of four groups of numbers. When a group contains the number '255', this is telling the router to ignore or mask that group of numbers in the IP address as it is defining the network location of the final destination. So, for example, if the IP Address were: 172.16.0.10 and the Subnet Mask used was Class B (255.255.0.0), the first two groups of numbers (172.16) would be ignored once they reached the proper network location. The next two groups (0.10) would be the final destination within the LAN to which the connection is to be made.

DHCP

DHCP (Dynamic Host Configuration Protocol) is a protocol which assigns a dynamic IP Address. Network control may be easier with DHCP as there is no need to assign and program individual IP Addresses for the LAN equipment. To use a dynamic IP Address, a DHCP server must be provided - either the DHCP server in the UX5000 can be used or an external server.

When equipment which is connected to the LAN (the DHCP client) is requesting an IP Address, it searches the DHCP server. When the request for an address is recognized, the DHCP server assigns an IP Address, Subnet definition, and the IP Address of the router, etc., based upon the UX5000 programming.

When the UX5000 is used as the DHCP server, note the following:

- The DHCP client must have a direct connection to the same network - a router cannot be used between the UX5000 and the terminal.
- No other DHCP servers can be on the network.
- No UX5000 reset is required when enabling/disabling DHCP.
- IPv6 is not supported with DHCP. It is unnecessary as an IP address is automatically prepared in IPv6 by the exchange between the terminal and a router.
- It's possible that an IP address may be repeated prior to the lease time expiring. When an IP address is leased and the maintenance data of the DHCP is cleared, duplication may occur of the IP address assigned until the lease time expires.

Note that the CPU must always have a static IP address. This address is set in ***Program 10-12-01 : CPU (FEC1) Network Setup - IP Address*** (default: 172.16.0.10).

Gatekeeper

Whenever an H.323 terminal activates, a check is made of the network to see if there are any gatekeepers available. When a gatekeeper is present, it provides users with:

- Address Translation
  Users typically do not know the IP addresses of other terminals. When a user makes a call, the gatekeeper translates an alias address (name or number) to the destination address.
- Admissions Control
  Users will not all be able to access the network at the same time because of limited shared resources. Gatekeepers may restrict network access based on call authorization, bandwidth usage, or some other criteria. It is important to note that Admissions Control is a way to preserve the integrity of the calls (provide QoS guarantees) that are already up and operating when a user requests access.
- Bandwidth Control
  Besides network access control, the gatekeeper offers network managers the ability to restrict or assign bandwidth to different applications along certain protocol conventions. This is another place network managers can enforce QoS guarantees and other enterprise-wide usage policies.

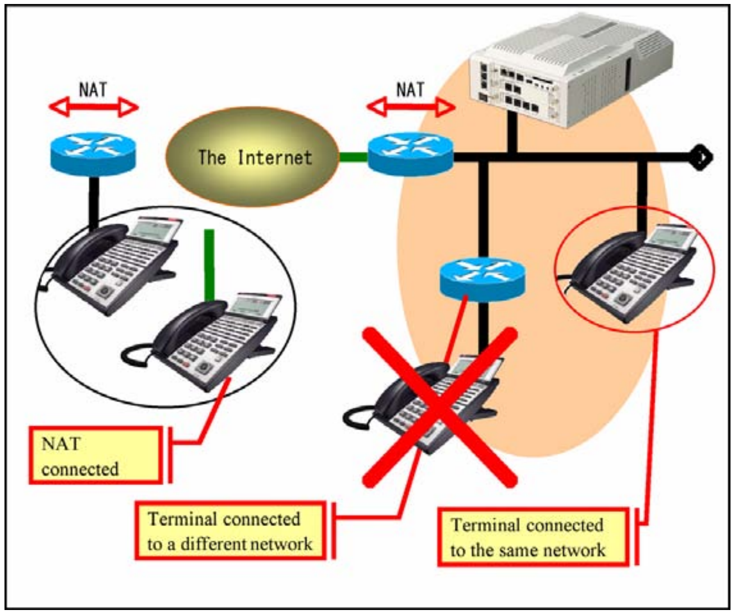
3

Routers

When purchasing a router for use with the UX5000 IP feature, the minimum requirements would be that it provide VPN and QoS. Note that a router which supports 'VPN Pass Through' requires a VPN server. Current VoIP protocols for the UX5000, NGT terminals can communicate over NAT, however multiple NAT routers on a network are not supported. The following NAT connections are possible:

- UX5000 (with local IP address) ⟶ NAT Router ⟶ Internet ⟶ NAT Router ⟶ IP terminal (with local IP address)
- UX5000 (with global IP address) ⟶ Internet ⟶ NAT Router ⟶ IP terminal (with local IP address)
- UX5000 (with local IP address) ⟶ NAT Router ⟶ Internet ⟶ IP terminal (with global IP address)

When the UX5000 is connected to a NAT router, the terminals connected to the LAN must be on the same network domain.



The priority control feature is required to prevent RTP packet loss. If a WAN is used for VoIP only and the bandwidth is wide enough for the VoIP channel, then the QoS feature might not be required.

The following are available routers which provide VPN and QoS:

- NEC IX1000 / 2000 Series
- Yamaha RT105, RTX1000 / 2000
- Cisco 800 Series
- Furukawa FITELnet - F40
- Fujitsu SiR-170, SiR-150

The following routers provide VPN but no priority control (QoS):

- Linksys BEFSX41, DEFVP41
- OMRON MR104DV, MR104FH
- Allied Telesis AR410

VoIP Bandwidth Calculation

A digital communications server converts an analog voice signal into a stream of bits expressed in K bits per second (where K is used to mean one thousand). For standard PCM digital encoding, this stream of bits is 64 K bits per second. This is 64 K bps in each direction (transmit and receive).

To improve transmission efficiency, this bit stream is compressed using a standard compression algorithm such as G.729. The result is still a bit stream, but with fewer bits per second. For example, G.729 will reduce the 64 K bits per second to a bit stream of 8 K bits per second.

This bit stream is then divided into chunks (called Voice Samples or Voice Frames) that can be placed in packets for transmission over a data network.

This reduced bit stream is examined repeatedly in fixed time intervals. This examination time is called the Voice Frame Interval. This is the time used to collect the bits for one Voice Frame. The

Voice Frame Interval is expressed in milliseconds (ms). A millisecond is one thousandth of a second.

To help determine the bandwidth requirements for the UX5000, the NEC Technical Support web site (http://ws1.necii.com) provides a bandwidth calculator. This web site requires registration with the NEC Sales Support. Contact them by phone (1-800-365-1928) or EMail (ubsdsupport@necinfrontia.com) in order to register. It is important to remember that the bandwidth calculator is based on a single voice channel. It takes two voice channels (send and receive) for each telephone conversation.

IP Hardware

Blades:

- **CCPU** - Signals the gateway with VoIP communication
- **32VOIPDB-A14** - VoIP daughter board provides a 32-channel voice packet gateway unit and works as a media gateway for VoIP communication. A VoIP daughter board is required for VoIP trunks extensions. The 32VOIPDB requires 32 trunk ports.
- **64VOIPDB-A14** - VoIP daughter board provides a 64-channel voice packet gateway unit and works as a media gateway for VoIP communication. A VoIP daughter board is required for VoIP trunks extensions. The 64VOIPDB requires 64 trunk ports.
- **128VOIPDB-A14** - VoIP daughter board provides a 128-channel voice packet gateway unit and works as a media gateway for VoIP communication. A VoIP daughter board is required for VoIP trunks extensions. The 128VOIPDB requires 128 trunk ports.

The number of VoIP channels may be restricted based on the codec and encryption used. Refer to Codec Selection for additional information.

When installing a VoIP daughter board, the UX5000 allocates the maximum number of trunk ports for the daughter board being installed. For instance, the 4VOIPDB daughter board requires 4 ports, while the 16VOIPDB daughter board requires 16 ports. If the daughter board is not going to be used for trunks, the logical trunk ports can be set to ‘0’ in **Program 10-03-01 : Blade Setup**, but the physical trunk ports are still assigned to the daughter board and cannot be used for any other blade. If the trunk ports will not be used and the trunk port usage is a concern, plug the VOIPDB daughter boards into the UX5000 last. The UX5000 will allow extension ports to be assigned even if there are no trunk ports available. A maximum of 512 IP extensions or 200 IP trunks are possible with the UX5000. The VoIP daughter board is installed onto the UX5000’s CCPU.

As long as at least one VOIPDB daughter board is installed, the UX5000 will allow up to 512 IP keysets to be registered.

Terminals:

- SIP Multi-Line Terminal (MLT) (supports standard SIP [RFC3261] and NECi protocol)
  SIP MLTs consist of the Value, Enhanced, DESI-Less, IP-CTS terminals, as well as the Softphone and WiFi phones.
- DTerm IP (Protims)
- Standard SIP Terminal
- H.323 Terminal

Trunks:

- SIP Trunk
- H.323 Trunk

Other Possible VoIP Connections:

- Telephone Call Between UX5000s Using CygniLink
- Networked Trunks/Extensions With H.323

Power must be supplied to the IP terminals using either a local or central power supply. If there is a power outage, the VoIP terminals will not work unless the terminals are plugged into a UPS (uninterruptible power supply).

When the first IP terminal is plugged in, the UX5000 automatically assigns the next three consecutive station ports available as IP ports. The next three IP terminals installed will use this group of ports. When the fifth IP terminal is connected, the next 3 consecutive station ports available will be assigned as IP ports.

Switches/Hubs for PoE

Using a central power supply or the Switch Hub (GSWU) blade, power over category 5 network cables can be provided. This eliminates the need of installing separate power adapters for each IP terminal and it allows for centralized power backup.

If PoE (power over ethernet) is to be used to eliminate the separate power adapters, due to the power requirements, a separate power source is suggested. It is recommended that you use one of the central power supplies below:

- 24-port power supply PoE-managed switch (NEC BlueFire 200/24)
  - PoE (Power Over Ethernet) to UX5000 IP/H.323 Terminals
  - Spare Pair (4/5, 7/8) / Signal Pair (1/2, 3/6) Selection

    For communication servers which require layer 2 switching capability and PoE, the NEC BlueFire 200/24 switch is recommended. This unit provides layer 2 switch capability in addition to being able to supply ethernet power to 24 NEC IP terminals.
- Cisco Data Switch - CDP Supported

Other manufacturer central power supplies may be usable, but the above items have been tested for compatibility with the UX5000 equipment.

Note: Each IP terminal consumes one port. When automatically selected, the port number ranges from 257-512. The port number can be manually assigned using the port range 001-512.

Use of SIP Protocol Available

SIP (Session Initiation Protocol) is a protocol used for Voice over IP. It is defined by the IETF (Internet Engineering Task Force) in RFC3261. SIP trunking is the term used for linking a PBX, like the UX5000, to the public telephone network by means of VoIP. This provides the possibility for users to place and receive communications and services from any location and for networks to identify the users wherever they are located.

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.

The UX5000 can support the following:

- IP Trunk : SIP Trunks
- IP Extension : Dterm IP / Standard SIP Extensions / SIP Multi-Line Terminal (MLT) (supports standard SIP [RFC3261] and NECi protocol)

  SIP MLTs consist of he Value, Enhanced, Desi-Less, IP-CTS terminals, as well as the Softphone and WiFi phones.
- IP Networking : Networking over SIP

  This protocol can be used simultaneously.

With the UX5000, 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. SIP trunks can be used for making a simple IP networking.

If a common carrier supports SIP, then the UX5000 can connect the SIP Carrier and outgoing calls to the PSTN network and the common IP network using an UX5000 SIP trunk.

Refer to the UX5000 SIP Trunking Manual, P/N 0913214, for details on programming this feature.

<u>***Conditions - SIP Trunking***</u>

3

- UX5000 does not support a simultaneous using of a SIP trunk inter-connection and a SIP Trunk Carrier connection.
- UX5000 supports a "100rel" option and "Session Timer" option.
- UX5000 supports a "DNS resolution access" and a "IP address direct access" for SIP server.
- UX5000 restricts an outgoing call under the following conditions:
  - SIP configuration failed.
  - SIP registration failed.
  - CPU/MBU/VOIPDB link down.
  - Lack of VOIPDB DSP resource.
  - Lack of a VoIP bandwidth.
- UX5000 can connect a SIP server over NAPT router by one static global IP address.
- UX5000 supports the sub-address feature with SIP trunk inter-connection.
- Support the 401 response for the Initial Invite.
  If 401 message is sent for the Initial Invite.
- Support the 401/407 response for the Invite of Session Timer.
  If 401/407 response is sent for the invite of Session Timer with Authentication header.
- SIP RFC3261 Supported Completely by SIP Trunks (UX5000 SIP trunks or SIP Carrier trunks) and SIP Extensions

Conditions - SIP Extensions

- SIP extensions support G.711, G.729 codec 20ms.
- SIP MLT extensions do not support G.723 codec.
- SIP RFC3261 Supported Completely by SIP Trunks (UX5000 SIP trunks or SIP Carrier trunks) and SIP Extensions
- **Hold/Transfer**

  The UX5000 will only support SIP extensions which comply with the two standards for the Hold and Transfer features as described in the Internet Engineering Task Force (IETF) documents:
  - draft-ietf-sipping-service-examples-09txt
  - draft-ietf-sipping-cc-transfer-05.txt
  These documents can be obtained at the IEFT web site: www.ietf.org

  - Call Hold
  - Transfer - Screened

Note: If a SIP terminal does a semi-attended Transfer to a trunk, SIP extension and voice mail, the UX5000 calls back to the SIP terminal immediately.

Volume Controls

Feature Availability
• Available.

Description

Each keyset user can control the volume of incoming ringing, splash tone, Paging, Background Music, Handsfree and your handset. Keysets consolidate all adjustments into the volume buttons. Pressing the Navigation Pad up or down will adjust the volume level for whichever feature is active (outside call, ICM, ICM ringing, paging, etc.). Pressing these keys when the terminal is idle will adjust the contrast level of the terminal's display. The users should set the volumes for their most comfortable levels.

Within UX5000 programming, an option is available which will allow the terminal to retain the user's settings for the handset and speakerphone volume, or it can default back to the UX5000 setting.

Warning Tone For Long Conversation

Feature Availability
• Available.

Description

The UX5000 can broadcast warning tones to a trunk user warning them that they have been on the call too long. The tones are just a reminder -- the user may be able to disregard the tones and continue talking if they choose. In addition, warning tones do not occur for Intercom calls.

The UX5000 can be programmed to use warning tones for outgoing or incoming trunks as well as DISA trunks. When warning tones are used for outgoing calls, the outside caller does not hear the warning tones. Using warning tones on incoming calls, however, both parties will hear the tone.

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.

If programmed, DISA calls will be disconnected unless the “continue” code is entered by the user. With the Long Conversation Cutoff feature, incoming or outgoing central office calls can also be disconnected.

Warning Tone for DISA Callers

For DISA callers, with this feature enabled, the warning tone timer begins when an incoming DISA call places an outgoing call and either the inter-digit timer 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 will occur 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’).

System Number Plan/Capacities

System Number Plan/Capacities		
	UX5000 Capacity	
System		
Analog Caller ID Detector	64	
Classes of Service	15	
Conference Bridge Groups	4	
Day/Night Mode Numbers	8	
Day/Night Service Patterns	32	
Dial Tone Detector DTMF Receiver	48 or 64 w/EXIFU-B1 Mounted	
Network Nodes: • CygniLink • AspireNet	 16 50	
System Ports (trunks and analog/digital/IP extensions)	200 trunks and 512 extensions * Chassis must be networked to reach max.	
Toll Restriction Classes	15	
Verifiable Account Code Table	2000	
Trunk		
Trunk Port Number	1-200 * A CCPU without a MEMDB, the trunks count toward the total number of allowed hardware ports (64).	
Trunk Ports (Total) • Analog Trunks • BRI Trunk Ports • T1/PRI Trunk Ports • E&M Analog Trunk Ports • DID Analog Trunk Ports • VoIP Trunk Ports	**19" Chassis x 4** 184 184 200 92 92 128	**Networked Chassis** 200 200 200 200 200 128
BRIU Logical Ports	T-Bus: 1-200 S-Bus: 1-256	
COIU: • Physical Ports • Logical Ports	 01-08 0-200	

System Number Plan/Capacities	
	UX5000 Capacity
DIOPU: • Physical Ports • Logical Ports	01-04 LD Trunk: 0-200 OPX: 0-256
PRIU Logical Ports	T-Bus: 1-200 S-Bus: 1-256
TLIU: • Physical Ports • Logical Ports	01-04 0-200
VOIPDB: • Physical Ports • Logical Ports	001-128 0-200
DID Translation Tables	20
DID Translation Table Entries	2000
DISA • Classes of Service • Users	15 1-15
Ring Groups	1-100
Tie Line Classes of Service	15
Tie Line Toll Restriction Classes	15
Trunk Access Maps	1-200
Trunk Group Numbers	1-100
Trunk Routes	1-100
Extension	
Telephone Extension Port Numbers • Keysets • Single Line Phones/Analog Devices • VoIP Extensions • IP DECT	1-384 (1-384) (1-384) (1-512) [5] 001-512 (manual select) [5] 385-512 (auto select) [5]
	* A CCPU without a MEMDB, the trunks count toward the total number of allowed hardware ports (64).

System Number Plan/Capacities	
	UX5000 Capacity
ESIU • Physical Ports • Logical Ports -Tone Ringer (2PGDAD) -Door Box (2PGDAD) -Analog I/F (2PGDAD) -ACI (2PGDAD) -APR for B2 Mode	01-16 1-8 1-8 1-96 1-96 193-512 (descending order)
SLIU • Physical Ports • Logical Ports	01-16 1-256
Telephone Extension Number Range	
Virtual Extension Ports	256
Virtual Extension Port Numbers	001-256
Virtual Extension Number Range	Undefined
2PGDAD Modules	512
ADA (Recording Jack) Adapters	512 (104 max. with digital terminals/ 512 max with IP terminals)
Door Boxes	8
Door Box Numbers	1-8
DSS Consoles Numbers • 16-Button DLS Consoles, Maximum Installed • 60-Button DSS Consoles, Maximum Installed	8 512 (384 max. with digital terminals / 512 max. with IP terminals) 32
Operator Access Number	0
Operator Extension	1-8
Ringdown Assignments	512
SLT Adapters	• 32 (9.5" Chassis) • 80 (19" Chassis) • 96 (19" Chassis x 2) • 368 (19" Chassis x 4) • 512 (Networked)
Voice Mail Master Numbers	

4

System Number Plan/Capacities	
	UX5000 Capacity
Abbreviated Dialing	
Abbreviated Dialing Groups	64
Abbreviated Dialing Bins	0-1999
Abbreviated Dialing Table-Common	1000
ACD	
ACD Groups	64
ACD Agent Extensions	512
ACI	
ACI Groups	16
ACI Ports	96
Automated Attendant	
VRS Message Numbers	1-100
Bluetooth Adapters	
BCH - Bluetooth Cordless Handset	16
BHA - Bluetooth Hub Adapter	16
Conference	
Conference Circuits	64 - maximum (32 Parties Per Conference)
Data Communication Interfaces	
APR Software Port Numbers	193-512
APA Adapters-Aspire Version	192 (only on Aspire phones)
APR Adapters-UX5000 Version	32
CTA or CTU Adapters-Aspire Version	128 (only on Aspire phones)
CTE	128
Module Extension Number Range	
Department and Pickup Groups	
Department (Extension) Group Numbers	1-64
Department (Extension) Group Number Range	
Call Pickup Group Numbers	1-64

System Number Plan/Capacities	
	UX5000 Capacity
Hotline	
Internal Hotline	512
External Hotline	512
Paging and Park	
Internal Page Group Numbers	0, 1-9 or 01-64
External Page Group Numbers	0, 1-8
External Speakers • CCPU • PGDAD Module	9 (1) (1-8)
Park Group Numbers	1-64
Park Orbits	1-64
Power Failure Adapters	
PSA (Power Failure) Adapters	• 16 (9.5” Chassis) • 40 (19” Chassis) • 88 (19” Chassis x 2) • 184 (19” Chassis x 4) • 200 (Networked)
SMDR	
SMDR Ports	1-8
VRS	
VRS (on DSP Daughter Board)	1
VRS Channels	16 (shared with IntraMail voice mail)
VRS Attendant Messages	3
VRS Recordable Messages	100
Voice Mail	
Ports for UX IntraMail	4-16
Ports for UX Mail	4-16

System Number Plan/Capacities	
	UX5000 Capacity
VoIP	
VoIP Extensions	512
Gigabit Adapters	512
IP Phones	512
RAS Unicast Ports	0-65535
Call Signaling Ports	0-65535
NGT Signal Receive Ports	0-65535
IP Call Procedure Port	0-65535
H.323 Alias Addresses	1-6
Note:	
Extension numbers can be three or four digits long. See Flexible System Numbering.	

UX5000 System Specifications

UX5000 System Capacities						
	9.5" Chassis	19" Chassis	19" Chassis x 2	19" Chassis x4	Networked System Maximum	Notes
Trunk Ports - Analog (CO/PBX lines) - IP (SIP) - PRI - 1.5M/2M - BRI Channels	16 128 48 16	40 128 96 40	88 128 192 88	184 128 200 184	200 128 200 200	Full license required for maximum number of ports. * Maximum number of simultaneous calls (128) is limited by the IP Pad Channels available.
Extension Ports - Digital/Analog [-24V] - IP Terminals - Analog [-48V]	32 512 * 8	80 512 * 20	160 512 * 44	320 512 * 92	512 512 * 512	Full license required for maximum number of ports. * 512 is peer-to-peer; maximum independent of chassis configuration.
Softphone	128					Limited by 1st party CTI max. License required
TDM Timeslots/ Channels	48	104	208	416	712	
VoIP Channels: - With sRTP - Without sRTP	96 Channels 128 Channels					
Voice Mail Ports	16 Ports					
V.34bis (33.6 kbps) Modem	1 Channel					
* ***NOTE:*** Maximum capacities above are determined by maximum blade per function. Additional factors may limit these quantities (such as load factor, available bandwidth, VIF and compression).						

4

UX5000 System Capacities						
	9.5" Chassis	19" Chassis w/CCPU	19" Chassis w/o CCPU	19" Chassis x4	Networked Chassis	Notes
Conference Circuits	64	64	64	64	64	32-parties max per Conference
SLT Adapter	32	80	96	368	512	Limited by load factor
2PGDAD Modules	32	80	96	368	512	Limited by load factor Max. Channels: 96 ACI 8 Door Boxes 8 Pages
Door Box/Door Unlock Contacts Internal Page Zones External Page Zones	8 64 9	8 64 9	8 64 9	8 64 9	8 64 9	
Universal Blade Slots	3 (1 required for CCPU)	6 (1 required for CCPU)	6	24 (1 required for CCPU)	1200 (1 CCPU required for each site)	

* ***NOTE:*** Maximum capacities above are determined by maximum blade configuration allowed. When installing single line sets, DISA, or tie lines, CCPU circuits must be allocated for DTMF receivers. To install single line sets with CO/PBX line access, or when installing immediate-start tie lines, CCPU circuits must be allocated for dial tone detection.

UX5000 Blade Capacities						
	9.5" Chassis	19" Chassis w/CCPU	19" Chassis w/o CCPU	19" Chassis x4	System Max. w/ Networking	Max. Trunks/Ext/ Channels
Chassis: 9.5" Chassis-B1 - Base Chassis with power supply (3 slots)	1	1	1	4	CygniLink: 64 AspireNet: 200 (4 per site)	-
9.5" Chassis-E1 - Expansion Chassis (3 slots)	0	1	1	4	CygniLink: 64 AspireNet: 200 (4 per site)	-
19" Chassis-A1 - Base or Expansion Chassis (6 slots)	0	1	1	4	CygniLink: 64 AspireNet: 200 (4 per site)	-
CCPU Central Processing Unit	1 (installed in CCPU slot)	1 (installed in CCPU slot)	0	1 (installed in CCPU slot)	16: CygniLink 50: AspireNet (1 per site) (installed in CCPU slot)	-
EXIFU Expansion Blade: EXIFU-B (for Base Chassis)	0	1	0	1	50 (1 per site)	-
EXIFU-E (for Expansion Chassis)	0	0	1	3	CygniLink: 48 AspireNet: 150 (3 per site)	-
External Battery Box	1	1	1	1	CygniLink: 16 AspireNet: 50 (1 per site)	-
VMDB-A1/B1 (installs on CCPU)	1	1	0	1	CygniLink: 16 AspireNet: 50 (1 per site)	Channels: 16

UX5000 Blade Capacities						
	9.5" Chassis	19" Chassis w/CCPU	19" Chassis w/o CCPU	19" Chassis x4	System Max. w/ Networking	Max. Trunks/Ext/ Channels
MEMDB-A1 (installs on CCPU)	1	1	0	1	CygniLink: 16 AspireNet: 50 (1 per site)	-
VOIPDB-A1 VoIP Media Gateway (installs on CCPU)	1	1	0	1	CygniLink: 16 AspireNet: 50 (1 per site)	-
8ESIU 8 Digital Stations **OR** 16ESIU 16 Digital Stations	2	5	5	20	32	Extensions: 512 (16x32)
4SLIU 4 Analog Stations **OR** 8SLIU 8 Analog Stations	2	5	5	20	32	Extensions: 256 (8x32)
4SLIDB 4 Analog Stations Daughter Board **OR** 8SLIDB 8 Analog Stations Daughter Board (installs on SLIU)	2	5	5	20	32	Extensions: 256 (8x32)
4COIU-LG1 4 Analog/Loop Start Trunks (with ground start)	2	5	6	23	25	Trunks: 200 (8x25)
4COIDB-LG1 8 Analog/Loop Start Trunks (with ground start) (installs on 4COIU-LG1)	2	5	6	23	25	Trunks: 200 (8x25)
2BRIU-A1 2 Two-Channel BRI Circuits	2	5	6	23	T-Bus (TRK): 25 S-Bus (STA): 64	T-Bus Trunks: 50 (2x25) Channels: 100 (2Bx50) S-Bus Extensions: 128 (2x64) Channels: 256 (2Bx128)

UX5000 Blade Capacities						
	9.5" Chassis	19" Chassis w/CCPU	19" Chassis w/o CCPU	19" Chassis x4	System Max. w/ Networking	Max. Trunks/Ext/ Channels
2BRIDB-A1 2 Two-Channel BRI Circuits (installs on 2BRIU-A1)	2	5	6	23	T-Bus (TRK): 25 S-Bus (STA): 64	T-Bus Trunks: 50 (2x25) Channels: 100 (2Bx50) S-Bus Extensions: 128 (2x64) Channels: 256 (2Bx128)
4TLIU 4 E&M Tie Line Trunks	2	5	6	23	50	Trunks: 200 (4x50)
4DIOPU 4 DID/OPX Trunks	2	5	6	23	TRK: 50 STA: 128	Trunks: 200 (4x50) Extensions: 512 (4x128)
082U-A1 Digital/SLT Combo Blade (8 Digital Plus 2 SLT Ports)	1	Without MEMDB: 1 With MEMDB: 1 Per Chassis	Without MEMDB: 1 With MEMDB: 1 Per Chassis	Without MEMDB: 1 With MEMDB:1 Per Chassis	Without MEMDB: 1 With MEMDB: 1 Per Chassis	
APSU-A1	2	2	2	2	CygniLink: 32 AspireNet: 100 (2 per site)	-
6SHUBU 6 Switch Hub with PoE	2	2	3	12	240 (6x240 sites)	1440 (6x240 sites)
8CNF 8 Circuit Conference Blade	2	5	6	23	32	-
16CNF 16 Circuit Conference Blade	2	5	6	23	32	-
RTU-B1 Router	2	2	2	8	CygniLink: 32 AspireNet: 100 (2 per site)	-
PVAU	2	5	6	23		

UX5000 Blade Capacities						
	9.5" Chassis	19" Chassis w/CCPU	19" Chassis w/o CCPU	19" Chassis x4	System Max. w/ Networking	Max. Trunks/Ext/ Channels
GSWU	2	3	3	12		

* ***NOTE:*** Maximum capacities above are determined by maximum blade per function. Additional factors may limit these quantities (such as load factor, available bandwidth, VIF and compression).

Environmental Requirements
Meeting established environmental standards maximizes the life of the system. Refer to the Standard Practices Manual for further information. Be sure that the site is not: 1. In direct sunlight or in hot, cold or humid places. 2. In dusty areas or in areas where sulfuric gases are produced. 3. In places where shocks or vibrations are frequent or strong. 4. In places where water or other fluids comes in contact with the main equipment. 5. In areas near high-frequency machines or electric welders. 6. Near computers, telexes, microwaves, air conditioners, etc. 7. Near radio antennas (including shortwave).

Power Requirements
A dedicated 110 VAC 60 Hz circuit located within seven feet of the chassis is required. You should install a separate dedicated outlet for each chassis.

Site Requirements
The system can be floor-, wall- or rack-mounted. Brackets secure each chassis to a wall. These mounting brackets also provide for a desktop placement.

Environmental Specifications
Chassis, Key Telephones, BCH, BHA, 16LK, Console, ADA, APR. PSA/PSTN Temperature: 0°C - 40°C (32 - 104°F) Humidity: 10-90% RH (non-condensing) ***Storage:*** Temperature: -20°C - 60°C (-4 - 140°F) Humidity: 10-90% RH Blades - EXIFU, VMDB, MEMDB, ESIU, 1PRIU Temperature: 0°C - 40°C (32 - 104°F) Humidity: 10-90% RH (non-condensing) ***Storage:*** Temperature: -20°C - 60°C (-4 - 140°F) Humidity: 10-90% RH Blades - SLIU, SLIDB, COIU, COIDB Temperature: 0°C - 40°C (32 - 104°F) Humidity: 20-90% RH (non-condensing) ***Storage:*** Temperature: -20°C - 60°C (-4 - 140°F) Humidity: 20-90% RH Door Box Temperature: -20°C - 60°C (-4 - 140°F) Humidity: 20-80% (non-condensing) UX5000 Power Supply ***Operating:*** Temperature: 0°C - +40°C (32 - 104°F) Humidity: 20-90% RH

UX5000 Electrical Specifications
Power Supply AC Power Supply Dedicated 15 Amp circuit Power Requirements: 120 VAC @ 15A Controlling/Base Chassis Input Voltage: 100VAC to 240VAC (Rated Voltage: 115VAC / 120VAC) Input Current: 3A (at 100VAC) to 1.5A (at 240VAC) per power supply Output Voltage Type Supplied: +3.3VDC / +5VDC / -24VDC / -48VDC / -27.3VDC (Charging Voltage for Battery Backup Frequency: 45Hz - 66Hz (Rated frequency: 50/60Hz) Phase and Wire: Single Phase 2 Line Type Grounding Requirements: No. 14 AWG copper wire Feeding Voltage Digital/OPX/DID: -48V SLT: 25mA / -28V **3-Slot Chassis or 6-Slot Chassis:** **With input voltage of 120 VAC, 50/60Hz, 2.19A and with full load conditions:** Output Power: Base Chassis=136W, Expansion 1 Chassis=272W, Expansion 2 Chassis=408W, Expansion Chassis 3=544W AC Input I: Base Chassis=2.55A, Expansion 1 Chassis=5.10A, Expansion 2 Chassis=7.65A, Expansion Chassis 3=10.20A VA @ 120V: Base Chassis=306VA, Expansion 1 Chassis=612VA, Expansion 2 Chassis=918VA, Expansion Chassis 3=1224VA KWh @ AC Input I x 120V/1000: Base Chassis=.306KWH, Expansion 1 Chassis=.612KWH, Expansion 2 Chassis=.918KWH, Expansion Chassis 3=1.224KWH BTU (KWh x 3413): Base Chassis=1004 btu, Expansion 1 Chassis=2008 btu, Expansion 2 Chassis=3012 btu, Expansion Chassis 3=4016 btu
ADA/APR Adapters, 16-Button DLS, 60-Button DSS, Digital and IP Terminals: Electrical Standard: FCC Part15 Class B, FCC Part68 Safety Standard: UL/CSA 60950
BCH/BHA Adapters: Electrical Standard: FCC Part15, Sub Part C, FCC Part68 Safety Standard: UL/CSA 60950

Mechanical Specifications				
Equipment	**Width**	**Depth**	**Height**	**Weight**
UX5000 9.5 Chassis - Base	8 1/2”	• 14 1/2” with no short- term battery back-up • 17 5/8” with short-term battery backup	4 9/16”	5 lbs 13 oz fully equipped
- Expansion	8 3/4”	• 14 3/8” with no short- term battery back-up • 17 3/8” with short-term battery backup	4 9/16”	3 lb 13 oz fully equipped
- Combined Chassis	16 7/8”	• 14 1/2” with no short- term battery back-up • 17 5/8” with short-term battery	4 9/16”	9 lb 9 oz fully equipped
UX5000 19” Chassis-A1	16.9” (430mm)	14.6” (36mm)	3.6” (88mm)	approx. 8 kg (when all slots occupied)
Value Digital Telephone: DG-2v (2-Button Non-Display) DG-6v (6-Button Display)	 179mm 179mm	 225mm 225mm	 112mm 112mm	 0.9 kg 0.9 kg
Value IP Telephone: IP-2v (2-Button Non-Display) IP-6v (6-Button Display)	 179mm 179mm	 225mm 225mm	 112mm 112mm	 0.9 kg 0.9 kg
Enhanced Digital Telephone: DG-12e (12-Button Display) DG-24e (24-Button Display) DG-32e (DESI-Less)	 179mm 179mm 179mm	 258mm 258mm 272mm	 112mm 112mm 112mm	 1 kg 1 kg 1.1 kg
Enhanced IP Telephone: IP-12e (12-Button Display) IP-24e (24-Button Display) IP-32e (DESI-Less)	 179mm 179mm 179mm	 258mm 258mm 272mm	 112mm 112mm 112mm	 1 kg 1 kg 1.1 kg
Enhanced IP-CTS Telephone	227mm	250mm	112mm	1.2 kg
60-Button DSS	224mm	137mm	57mm	600g

16-Button DLS (16LK)	46.3mm	224mm	57mm	300g
ADA	65mm	82mm	25mm	5.3 oz.
APR	65mm	82mm	25mm	5.3 oz.
BCH	64mm	223mm	112mm	10.6 oz.
BHA	65mm	82mm	25mm	5.3 oz.
BTH	51mm	180mm	33mm	7.1 oz.
Door Box	3 7/8”	1”	5 1/8”	6.5 oz

IP-CTS Telephone		
LCD		5.7" TFT QVGA 16-bit Color with Backlit Touch Panel 2 Size Display - Double height characters can be displayed Maximum screen size is 28 characters (double byte character 14 characters) x 4 rows
Lock Button		Terminal information is locked by the Lock button. Status is indicated by a red LED.
Line Key		LCD Displayed Key
Fixed Function Keys		Ten keys (0-9, *, #) - Backlit, Cursor and Function Keys (changeable for vertical markets)
Soft Key		4 Buttons plus CHECK and CLEAR
Headset Interface		Yes
Option	**Bottom - 1 slot**	BCH (Bluetooth Cordless Handset) - uses the Bottom and Handset slots BHA (Bluetooth Hub) ADA (Recording Adapter)
	Handset - 1 slot	BCH (Bluetooth Cordless Handset) - uses the Bottom and Handset slots PSA (PSTN Adapter for Analog)
	Side - 2 slots	16 Line Key (16LK) Module
		60-Button DSS Console
Module		Ten Key Dial Pad Kit
CODEC		Narrow Band - 3.4kHz Audio Bandwidth: G.711, G.711 Appendix II, G.729A, G.729aB Wide Band - 7kHz Audio Bandwidth G.722 - 64kbps
Handset Bandwidth		Wide Band (300~7000Hz) with Hearing Aid Compatibility (HAC)
MIC		Omni Directional
Speakerphone		Full Duplex Wideband (300-7000Hz)
WAV Ring Tone		Downloadable WAV Music Support (8/16kHz Sampling)
LAN		2 Ports 10Base-T (IEEE802.3) 100Base-TX (IEEE802.3u) 1000Base-T (IEEE802.3ab)
Power Feed		Center Power Feed: IEEE802.3aF PoE (need ILPA for CISCO PoE) Local Power Feed: AC Adapter (27V 750mA)
Power Consumption		Approx. 10.2W without options

4

IP-CTS Telephone	
Message/Incoming Call LED	7 Color
Tilt	Adjustable (main body and LCD)
Wall Mount	Built-In
RoHS Compliant	Yes

IP-CTS Telephone		
Item		**Explanation**
Telephony Function	Call Protocol	NEC Enhanced SIP / KTS Enhanced SIP RTP (Audio)
	CODEC	Narrow Band - 3.4kHz Audio Bandwidth G.711, G.711 Appendix II, G.729a, G.729aB Wide Band - 7kHz Audio Bandwidth G.722 - 64kbps
	Improved Voice Quality / QoS	Acoustic Echo Canceller Adaptive Jitter Buffer Packet Loss Concealment (G.711 Appendix I) IEEE802.1p Priority ToS Value (IP Precedence, Diffserve (DSCP))

IP-CTS Telephone		
Network Function	Protocol	TCP/IP (IPv4, TCP, UDP) TFTP FTP HTTP/HTTPS DNS
	Plug & Play	DHCP (RFC-1532/1533 compatible) DHCP (Vendor Extensions) (RFC-2131/2132 compatible) Auto Config (NEC proprietary)
	Authentication Protocol	IEEE802.1X Supplicant EAPOL Forwarding (for PC port's supplicant)
	LAN I/F Setting	Speed (Auto, 100M, 10M) Duplex (Auto, Full, Half) Auto-MDI/MDIX (When both Speed and Duplex are set to Auto) Tag VLAN (IEEE802.1Q) Port VLAN (PC Port only)
	NAT	SupPorts SIP-ALG
	Remote Connection	VPN Client (IPsec)
	Security	Authentication (Call control, XML browser, etc.) Encryption (Call control, RTP, etc.)
	Management Function	SIP, CDP QoS Trouble information notification (NEC proprietary) Self-test can be performed with the telephone or with the IP Phone Manager. Loopback test of voice media can be performed by call control server
Maintenance Function	Configuration	IP Phone Manager (Windows based) Handset Programming WebPro
	Firmware Update	Manual and Automatic Update
Security	Protection of Personal Information	Protection for Terminal information Password Protection Remote Lock
Standard Application (only LCD phone)	Browser	XML Browser
Ringer/MOH Tone	Downloadable	Ringer : Yes Music on Hold: WAV file stored in IP terminal - 3 to 5 files can be stored (8kHz WAV File - 32 seconds max / 16kHz WAV File - 16 seconds max)

4

IP-CTS Telephone		
Other	Presence Indication	Supported
	Instant Message	Supported. Max. 300 characters, Max. 8 destinations simultaneously. New message indicated by MW LED or chime.
	Menu	Menu / Shortcut Menu
	Desktop	Desktop Icon Wallpaper Screen Saver

Enhanced IP Telephones		
LCD	IP-32e (DESI-Less)	2 LCD Displays - One for Call Control, One for Line Keys 224x96 Dot Matrix Gray Scale LCD with Backlit 28x6 (8x16 Font) or 34x8 (6x12 Font) Line: Alphanumeric 14x6 (16x16 Font) or 18x8 (12x12 Font) Line: Japanese Kanji 2 Size Display - Double height characters can be displayed Maximum screen size is 28 characters (double byte character 14 characters) x 4 rows
	IP-24e, IP-12e	224x96 Dot Matrix Gray Scale LCD with Backlit 28x6 (8x16 Font) or 34x8 (6x12 Font) Line: Alphanumeric 14x6 (16x16 Font) or 18x8 (12x12 Font) Line: Japanese Kanji 2 Size Display - Double height characters can be displayed Maximum screen size is 28 characters (double byte character 14 characters) x 4 rows
Line Key	IP-32e (DESI-Less)	8-Button with Red & Green LEDs 4 Page (8 linex3 + 8 One Touch)
	IP-24e, IP-12e	12/24-Button with Red & Green LEDs
Lock Button		Terminal information is locked by the Lock button. Status is indicated by an LED.
Fixed Function Keys		Ten keys (0-9, *, #) - Backlit, Cursor and Function Keys (changeable for vertical markets)
Soft Key		4 Buttons plus CHECK and CLEAR
Headset Interface		Yes

Enhanced IP Telephones		
Option	Bottom - 1 slot	ADA (Recording Adapter) BHA (Bluetooth Hub) BCH (Bluetooth Cordless Handset) - uses the Bottom and Handset slots
	Handset - 1 slot	BCH (Bluetooth Cordless Handset) - uses the Bottom and Handset slots PSA (PSTN Adapter for Analog)
	Side - 2 slots	16-Button DLS Module
		60-Button DSS Console
Module		DESI-Less Line Key/LCD Unit for IP 12 Line Key (12LK) Kit Ten Key Dial Pad Kit
CODEC		Narrow Band - 3.4kHz Audio Bandwidth G.711, G.711 Appendix II, G.729a, G.729aB Wideband - 7kHz Audio Bandwidth G.722 - 64kbps
Handset Bandwidth		300~7000Hz
MIC		Omni Directional
Speakerphone		Full Duplex Wideband (300-7000Hz)
WAV Ring Tone		Downloadable WAV Music Support (8/16kHz Sampling)
LAN		2 Ports 10Base-T (IEEE802.3) 100Base-TX (IEEE802.3u) 1000Base-T (IEEE802.3ab)
Power Feed		Center Power Feed: IEEE802.3aF PoE (need ILPA for CISCO PoE) Local Power Feed: AC Adapter (27V 750mA)
Power Consumption		**12-Button Display/24-Button Display:** Approx. 8.4W without options **DESI-Less:** Approx. 8.2W without options
Message LED		7 Color
Tilt		Adjustable (main body and LCD)
Wall Mount		Built-In
RoHS Compliant		Yes

4

<table>
<tr><th colspan="3">Enhanced IP Telephones</th></tr>
<tr><th colspan="2">Item</th><th>Explanation</th></tr>
<tr><td rowspan="3">Telephony Function</td><td>Call Protocol</td><td>NEC Enhanced SIP / KTS Enhanced SIP
RTP (Audio)</td></tr>
<tr><td>CODEC</td><td>Narrow Band - 3.4kHz Audio Bandwidth
G.711, G.711 Appendix II, G.729a, G.729aB
Wide Band - 7kHz Audio Bandwidth
G.722 - 64kbps</td></tr>
<tr><td>Improved Voice Quality / QoS</td><td>Acoustic Echo Canceller
Adaptive Jitter Buffer
Packet Loss Concealment (G.711 Appendix I)
IEEE802.1p Priority
ToS Value (IP Precedence, Diffserve (DSCP))</td></tr>
<tr><td rowspan="9">Network Function</td><td>Protocol</td><td>TCP/IP (IPv4, TCP, UDP)
TFTP
FTP
HTTP/HTTPS
DNS</td></tr>
<tr><td>Plug & Play</td><td>DHCP (RFC-1532/1533 compatible)
DHCP (Vendor Extensions)
(RFC-2131/2132 compatible)
Auto Config (NEC proprietary)</td></tr>
<tr><td>Authentication Protocol</td><td>IEEE802.1X Supplicant
EAPOL Forwarding (for PC port's supplicant)</td></tr>
<tr><td>LAN I/F Setting</td><td>Speed (Auto, 100M, 10M)
Duplex (Auto, Full, Half)
Auto-MDI/MDIX (When both Speed and Duplex are set to Auto)
Tag VLAN (IEEE802.1Q)
Port VLAN (PC Port only)</td></tr>
<tr><td>NAT</td><td>SupPorts SIP-ALG</td></tr>
<tr><td>Remote Connection</td><td>VPN Client (IPsec)</td></tr>
<tr><td>Security</td><td>Authentication (Call control, XML browser, etc.)
Encryption (Call control, RTP, etc.)</td></tr>
<tr><td>Management Function</td><td>SIP, CDP
QoS
Trouble information notification (NEC proprietary)
Loopback test of voice media can be performed by call control server</td></tr>
<tr><td colspan="2" style="display:none"></td></tr>
<tr><td rowspan="2">Maintenance Function</td><td>Configuration</td><td>IP Phone Manager (Windows based)
Handset Programming
WebPro</td></tr>
<tr><td>Firmware Update</td><td>Manual and Automatic Update</td></tr>
</table>

Enhanced IP Telephones		
Security	Protection of Personal Information	Protection for Terminal information Password Protection Remote Lock
Standard Application (only LCD phone)	Browser	XML Browser
Ringer/MOH Tone	Downloadable	Ringer : Yes Music on Hold: WAV file stored in IP terminal - 3 to 5 files can be stored (8kHz WAV File - 32 seconds max / 16kHz WAV File - 16 seconds max)
Other	Presence Indication	Supported
	Instant Message	Supported
	Menu	Menu / Shortcut Menu
	Desktop	Desktop Icon Screen Saver

Value IP Telephones		
LCD	with LCD	168x41 Full-Dot Matrix Gray Scale LCD *without* Backlit 28 characters x 3 row
	Non-Display	None
Line Key	with LCD	6-Button with Red & Green LEDs
	Non-Display	2-Button with Red & Green LEDs
Fixed Function Keys	with LCD	Ten keys (0-9, *, #), Cursor and Function Keys (changeable for vertical markets)
	Non-Display	Supports Keys Provided
Soft Key	with LCD	4 Buttons plus CHECK and CLEAR
	Non-Display	No
Headset Interface		No
Option		No Adapters
Module		Ten Key Dial Pad Kit
CODEC		Narrow Band - 3.4kHz Audio Bandwidth G.711, G.711 Appendix II, G.729a, G.729aB
Handset Bandwidth		Narrow Band with Hearing Aid Compatibility (HAC)
MIC		Omni Directional
Speakerphone		Full Duplex Narrow Band
WAV Ring Tone		No

Value IP Telephones	
LAN	2 Ports 10Base-T (IEEE802.3) 100Base-TX (IEEE802.3u)
Power Feed	Center Power Feed: IEEE802.3aF PoE (need ILPA for CISCO PoE) Local Power Feed: AC Adapter (27V 750mA)
Power Consumption	**2-Button Non-Display:** Approx. 4.0W **6-Button Display:** Approx. 4.4W with LCD
Message LED	3 Color
Tilt	Adjustable (main body and LCD)
Wall Mount	Built-In
RoHS Compliant	Yes

Value IP Telephones		
Item		**Explanation**
Telephony Function	Call Protocol	NEC Enhanced SIP / KTS Enhanced SIP RTP (Audio)
	CODEC	Narrow Band - 3.4kHz Audio Bandwidth G.711, G.711 Appendix II, G.729a, G.729aB
	Improved Voice Quality / QoS	Acoustic Echo Canceller Adaptive Jitter Buffer Packet Loss Concealment (G.711 Appendix I) IEEE802.1p Priority ToS Value (IP Precedence, Diffserve (DSCP))

Value IP Telephones		
Network Function	Protocol	TCP/IP (IPv4, TCP, UDP) TFTP FTP HTTP/HTTPS DNS
	Plug & Play	DHCP (RFC-1532/1533 compatible) DHCP (Vendor Extensions) (RFC-2131/2132 compatible) Auto Config (NEC proprietary)
	Authentication Protocol	IEEE802.1X Supplicant EAPOL Forwarding (for PC port's supplicant)
	LAN I/F Setting	Speed (Auto, 100M, 10M) Duplex (Auto, Full, Half) Auto-MDI/MDIX (When both Speed and Duplex are set to Auto) Tag VLAN (IEEE802.1Q) Port VLAN (PC Port only)
	NAT	SupPorts SIP-ALG
	Remote Connection	VPN Client (IPsec)
	Security	Authentication (Call control, XML browser, etc.) Encryption (Call control, RTP, etc.)
	Management Function	Trouble information notification (NEC proprietary)
Maintenance Function	Configuration	IP Phone Manager (Windows based) Handset Programming
	Firmware Update	Manual and Automatic Update
Security	Protection of Personal Information	Protection for Terminal information Password Protection Remote Lock
Standard Application (only LCD phone)	Browser	XML Browser
Ringer/MOH Tone	Downloadable	Ringer : No Music on Hold: WAV file stored in IP terminal
Other	Presence Indication	Not Supported
	Instant Message	Not Supported

Enhanced Digital Telephones		
LCD	DS-32e (DESI-Less)	1. Call Control Area 168x55 Dot Matrix Black & White LCD with Backlit 28 characters x 4 rows (6x12 Font) : Alphanumeric 14 characters x 4 rows (12x12 Font) : Japanese Kanji 2. Line Key Area 12 Characters and Icon: Alphanumeric 5 Characters and Icon: Japanese Kanji
	DS-12e, DG-24e	168*55 Dot Matrix Black & White LCD *without* Backlit 28 characters x 4 rows (6x12 Font) : Alphanumeric 14 characters x 4 row s(12x12 Font) : Japanese Kanji
Line Key	DS-32e (DESI-Less)	8-Button with Red & Green LEDs 4 Page (8 linex3 + 8 One Touch)
	DS-12e, DG-24e	12/24-Button with Red & Green LEDs
Fixed Function Keys		Ten keys (0-9, *, #), Cursor and Function Keys (changeable for vertical markets)
Soft Key		4 Buttons plus CHECK and CLEAR
Headset Interface		Yes
Handset Bandwidth		300~3400Hz
MIC		Omni Directional
Speakerphone		Wideband (300-3400Hz)
Option	**Bottom - 1 slot**	ADA (Recording Adapter) APR (Analog Port Adapter with Ringer) BHA (Bluetooth Hub) BCH (Bluetooth Cordless Handset) - uses the Bottom and Handset slots
	Handset - 1 slot	BCH (Bluetooth Cordless Handset) - uses the Bottom and Handset slots PSA (PSTN Adapter for Analog)
	Side - 1 slot	16-Button DLS) Module
Module		12 Line Key (12LK) Kit Backlit LCD for Digital Value Terminal Ten Key Dial Pad Kit DESI-Less Line Key/LCD Unit for Digital
Speakerphone		Full Duplex
Input Voltage		DC 12 - 53 V (terminal end)
Power Consumption		**24-Button Display:** Approx. 2.0W without options **DG-32e (DESI-Less):** Approx. 2.2W without options
Message LED		3 Color
Tilt		Adjustable (main body and LCD)
Wall Mount		Built-In
RoHS Compliant		Yes

Value Digital Telephones		
LCD	with LCD	24 characters x 3 row LCD *without* Backlit
	Non-Display	None
Line Key	with LCD	6-Button with Red & Green LEDs
	Non-Display	2-Button with Red & Green LEDs
Fixed Function Keys		Ten keys (0-9, *, #), Cursor and Function Keys (changeable for vertical markets)
Soft Key	with LCD	4 Buttons plus CHECK and CLEAR
	Non-Display	No
Headset Interface		No
Handset Bandwidth		300~3400Hz
MIC		Omni Directional
Speakerphone		Wideband (300-3400Hz)
Option		No
Module		Ten Key Dial Pad Kit
Speakerphone		Half Duplex
Input Voltage		DC 12 - 53 V (terminal end)
Power Consumption		**2-Button Non-Display:** Approx. 1.0W **6-Button Display:** Approx. 1.1W with LCD
Message LED		3 Color
Tilt		Adjustable (main body but not LCD)
Wall Mount		Built-In
RoHS Compliant		Yes

082U Combination Blade
Analog Ports: Constant Current System (20mA / -28Vdc) Resistance: 600 Ohm (loop) For Polarity Reversing, Number Display, and Message Waiting: -110VDC Ringer Signal: 16Hz, 35Vrms, -28Vdc

4

2PGDAD Module/NTCPU Input/Output	
Audio/Music Input Input Impedance: External Amplifier Input:	 47 KOhm @ 1KHz 12dBV 5KOhm
Audio/Paging Output **2PGDAD:** Output Impedance: Maximum Output: **CCPU:** Output Impedance: Maximum Output:	 600 Ohms @ 1 KHz +8 dBm 600 Ohms @ 1 KHz -3 dBm
Relay Contacts Configuration: Maximum Contact Ratings:	 Normally Open 24 VDC, 0.5A 120 VAC, 0.25A
Night Mode Relay Connection, Input Break: Make:	 48 VDC 7 mA

10 Key Dial Pad Kit, Dedicated Double-Contact	
Purpose	Added When PSA Used to Provide Dedicated Double-Contact Dial Pad *Without* Backlit
Target Phone	IP-CTS and Enhanced IP Phones Enhanced Digital Phones
Install To	Dial Pad
AC Adapter	Not Required

12LK Kit - Line Key Kit	
Purpose	Provide 12 Additional Line Keys
Target Phone	DG-12e or IP-12e (Enhanced 12-Button Displays)
Install Location	Line Key Area
Number of Keys	12 Keys
LED Color	Red and Green
AC Adapter	Not Required

16-Button DLS Module	
Purpose	Provide Additional Line/Loop/Programmable Function Keys
Target Phone	IP-CTS Enhanced IP-12e, IP-24e, IP-32e, DG-12e, DG-24e, DG-32e (12-Button Display, 24-Button Display, DESI-Less Phone)
Install To	Left of Terminal (special connector)
Number of Keys	16 Keys with LEDs
LED Color	Red and Green
AC Adapter	Not Required
Power Feed	Digital Keyset: Received from 2-Wire Line IP Terminal: Received from 3-Wire Line
Power Consumption	Approx. 0.5W
RoHS Compliant	Yes

AC-2R Input/Output	
AC Input Output Tip	100 ~ 240VAC 50/60Hz 800mA 27VDC 750MA Negative

ADA - Recording Adapter	
Purpose	Recording of Handset/Headset/Speakerphone Speech
Target Phone	IP-CTS and Enhanced IP Phones Enhanced Digital Phones
Install Slot	Bottom Slot
Interface to Recorder	For Tape Recorder: Stereo Mini-Jack (3.5 mm diameter) For ACD: Barrel Terminal (AMP)
Jacks	2 - Input and Output
Recording Playback	Stereo
Playback Path	When voice recording to a PC using a CTI application, it is possible to playback through the telephone.
Feature	Voice Sent to Recorder Hookswitch Information
Power Consumption	0.8W approximately
Power Feed / AC Adapter	Supplied by telephone - no separate AC adapter required (unless over 900m line length from system chassis).
Terminal Impedance	600 ohm or 30 ohm
Confirmation Tone Output	Confirmation tone to far end possible if recording device has confirmation tone generation
Control for Recording	Recording started/stopped by off-hook or VOR (synchronized with voice level) of the recording machine. When the terminal is idle, no recording is done.
RoHS Compliant	Yes
Retrofit	Can connect to UX5000 telephone on an Aspire system.

APR - Analog Port with Ringer Adapter	
Purpose	Connecting Analog Device (fax/modem/cordless phone)
Target Phone	Enhanced Digital Phones
Install Slot	Bottom Slot
Interface	RJ11
Feature	MF Signal Relay to Chassis Ringer Generation Voice Path Connection Not Polarity Sensitive Change of B-Channel (B1/B2) Possible by System
AC Adapter	48v Power Required (when over 600 meters in line length)
Power Feed	Supplied by telephone - no separate AC adapter required (unless over 900m line length from system chassis).
Power Consumption	Approx. 1.2W
Terminal Impedance	600 ohm or complex (selectable)
Ring Output Voltage	50Vrms
Distance (APR-SLT)	Max. 15m (49 feet) - Less than loop resistance 5.4 ohm
Supply Current to Analog Line	Less than 26mA (no branch connection)
RoHS Compliant	Yes
Retrofit	Can connect to UX5000 telephone on an Aspire system.

BGM/MOH Music Source Input	
Input Impedance: Input Level: Maximum Input:	47KOhm / 1Khz Nominal 250 mV (-10 dBm) 1V RMS
Inputs for MOH and BGM are located on the CCPU blade. The 2PGDAD also provides MOH inputs.	

BHA - Bluetooth Hub Adapter	
Purpose	Provide Bluetooth Hub Adapter Using Common Bluetooth Technology
Target Peripheral	Bluetooth Headset
Install Slot	Bottom Slot
Wireless Class	1 (1mW100mW)
Distance	2 Meters
Maximum Units Installed	16 within 100 meters (outside) or 50 meters (inside). No 2 devices should be closer than 1 meter.
Bluetooth Version	2.0 Class 1
AC Adapter	IP Phone: Not Required Digital Phone: Not Required
Power Consumption	0.8W
Power Feed	From digital or IP telephone
RoHS Compliant	Yes
Retrofit	**Cannot** connect to Aspire keyset.
Voice Call	Handsfree with BTC Speaker Call: IP Terminal: Wide band 7 kHz audio bandwidth Digital Terminal: Narrow band 3.4 kHz audio bandwidth Bluetooth Headset Call: Bluetooth headset only supPorts narrow band audio BTH and Bluetooth headset cannot be used simultaneously

4

BCH - Bluetooth Cordless Handset with Hub	
Purpose	Provide Cordless Handset Terminal Using Common Bluetooth Technology
Cordless Handset Display	Black and White LCD 120x82 Dot Matrix Grayscale LCD *with* Backlit 20 digit x 2 line (5x12 dot / font): Alphanumeric 10 digit x 2 line (12x12 dot/font): Japanese Kanji Remaining 2 lines for displaying icons
Cordless Handset Keys	8 2-Color Line Keys with LED 6 Function keys (Talk, Disconnect, Hold, etc.) 10 Key Dial Pad (-9, *, #) - Backlit Cursor Keys (4 Direction keys and 1 Enter key)
Target Phone	IP-CTS, IP Enhanced, Digital Enhanced
Install Slot	Charge Cradle Connects to Handset Slot (left of terminal - handset cradle needs to be changed) and Interface Cable Connects to Bottom Slot
Speaker	For Ringing
Soft Keys	No
Options	No Adapters
Headset Jack	Yes
Wireless Class	1 (1mW100mW)
Distance	100 Meters Outside, 50 Meters Indoors
Maximum Units Installed	16 within above distance. No 2 devices should be closer than 1 meter.
Bluetooth Version	2.0 Class 1
AC Adapter	IP Phone: Not Required (power is fed from IP Phone) Digital Phone: 48v Power Required (when over 400 meters in line length)
Power Consumption	1.5W (with BTH which is approx. 0.5W)
Talk Time	8 hours on full charge
Charge Time	16 hours for full charge
Stand-By Time	30 hours (full charge, interval on/off) 24 hours (full charge, continuously on)
Battery	Li-Ion Battery installed on BTH
Power Feed	From digital or IP telephone
RoHS Compliant	Yes
Retrofit	**Cannot** connect to Aspire keyset.

BCH - Bluetooth Cordless Handset with Hub	
Voice Call	Handsfree with BTC Speaker Call: IP Terminal: Wide band 7 kHz audio bandwidth Digital Terminal: Narrow band 3.4 kHz audio bandwidth Bluetooth Headset Call: Bluetooth headset only supPorts narrow band audio BTH and Bluetooth headset cannot be used simultaneously

DESI-Less LK/LCD Unit for IP Phone	
Purpose	Provide Additional LCD Line Keys
Target Phone	Enhanced IP Phones
LCD	2 LCD Displays - One for Call Control, One for Line Keys 224*96 Dot Matrix Gray Scale LCD with Backlit 28*6 (8* 16 Font) or 34*8 (6* 12 Font) Line: Alphanumeric 14*6 (16* 16 Font) or 18*8 (12* 12 Font) Line: Japanese Kanji
Install To	Line Key Area
Number of Keys	8 Keys
LED Color	Red and Green
AC Adapter	Not Required

DESI-Less LK/LCD Unit for Digital Phone	
Purpose	Provide Additional LCD Line Keys
Target Phone	Enhanced Digital Phones
LCD	1. Call Control Area 168*58 Dot Matrix Black & White LCD with Backlit 28*4 (6* 12 Font) Line: Alphanumeric 14*4 (12* 12 Font) Line: Japanese Kanji 2. Line Key Area 12 Characters and Icon: Alphanumeric 5 Characters and Icon: Japanese Kanji
Install To	Line Key Area
Number of Keys	8 Keys
LED Color	Red and Green
AC Adapter	Not Required

DSS Console - 60-Button Direct Station Selection	
Purpose	Provide Additional Function Keys
Install Slot	IP Phone: Right of Terminal (connected to the side option slot using a special cable) Digital Phone: Direct Connection to System
Number of Keys	60 Keys with LEDS Multiple Page Support (2 pages) for Keys 1-54
LED Color	Red
AC Adapter	IP Phone: Required Digital Phone: Not Required
Power Feed	For IP Terminal: 27V 750mA (option) For Digital Keysets: Received from 2-Wire Line
Power Consumption	Approx. 1.0W
RoHS Compliant	Yes

Door Box/External Paging	
Output Impedance: Output Level: Maximum Output: Configuration:	600 Ohm Nominal 250 mV (-10 dBm) 400 mV RMS Normally open

PSA - PSTN Adapter for Analog Telephone	
Purpose	Provide Functionality In Case of Power Failure or Down Network Capability for PSTN & PBX Connection in Remote Office
Target Peripheral	IP-CTS and Enhanced IP Phones Enhanced Digital Phones
Install Slot	Handset Slot (left of terminal - handset cradle needs to be changed)
Ten Key Dial Pad	Must Change Dial Pad Kit on Telephone To Dedicated Double-Contact Dial Pad Key Without Backlit
PSTN & PBX Selection Key	Yes (surface on the cradle)
PSTN Type	Analog PSTN
Dial Method	MF/DP 20pps/10pps)
AC Adapter	Not Required
Public Standard	UL/CSA 60950, FCC Part. 15 Class B, FCC Part 68, RoHS

SLT Adapter
Constant Current Circuit: Current fixed at 20mA / -48Vdc Resistance (Between SLT Adapter and Telephone): 500 Ohm (loop) **Signal Method** On-Hook Condition: 48VDC Ringer Signal: 180 Vp-p, 16Hz

SLIU BLADE / SLIDB
Signal Method On-Hook Condition: -46VDC +- 3VDC Message Waiting Signal: -110VDC +- 3VDC or FSK Ringer Signal: 16Hz/20Hz/25Hz, 75Vrms +-1Vrms (no load condition), -28Vdc Constant Current Type Battery Feeding: (25mA / -28Vdc)

4

UL Listed System
A label will be affixed to the product with the letters UL inside a circle which is the symbol used by UL to indicate that a product is UL Listed. If you see a small “c” outside the symbol, then the product also meets the requirements for Canada.

FCC Registration Information	
Responsible Party: Manufacturer: FCC Part 15 Registration:	NEC Infrontia, Inc. NEC Infrontia Thai Ltd. Class A
Model: FCC Registration Number: (Refer to the label on the Controlling/Base chassis for the FCC Registration Number.	UX5000 (Cygnifire) 9.5” Plastic Chassis: KF: US: NIFKF07BCYGNFIRE MF: US: NIFMF07BCYGNFIRE PF: US: NIFPF07BCYGNFIRE 19” Metal Chassis KF: US:NIFKF07BSN1759 MF: US:NIFMF07BSN1759 PF: US:NIFPF07BSN1759

Reg. Status	**Facility Interface Code (FIC)**	**Mfrs. Port Identifier**	**Ringer Eq. Number**	**Service Order Code (SOC)**	**Network Jacks**
Original	02GS2	4COIU-LG1	0.7B	9.0F (Analog)	RJ21X
Original	02GS2	4COIUDB-LG1	0.7B	9.0F (Analog)	RJ21X
Original	02RV2-T	4DIOPU	---	AS.2 (DID)	RJ21X

Reg. Status	**Analog Private Line Interfaces**	**Mfrs. Port Identifier**		**Service Order Code (SOC)**	**Network Jacks**
Original	TL11M	4TLIU	---	9.0F (Analog)	RJ21EX

* The SOC for digital services is 6.0P and 6.0N.

Loop Resistance for COIU Blades	
DC Loop Resistance	Less than 60 Ohms
AC Impedance	600 Ohms

FCC DID Requirements	
Federal Communications Commission DID Requirements This equipment must operate in a manner that is not in violation of Part 68 rules. This equipment returns answer supervision to the Public Switched Network when the DID trunk is: (1) answered by the called station; (2) answered by the attendant; (3) routed to a recorded announcement that can be administered by the CPE user; (4) routed to a dial prompt. The equipment returns answer supervision on all DID calls forwarded back to the Public Switched Telephone Network except when: (1) a call is unanswered; (2) a busy tone is received; (3) a reorder tone is received. When ordering DID Service, provide the telco with the following information:	
UX5000 FCC Registration Number:	9.5" Plastic Chassis: KF: US: NIFKF07BCYGNFIRE MF: US: NIFMF07BCYGNFIRE PF: US: NIFPF07BCYGNFIRE 19" Metal Chassis KF: US:NIFKF07BSN1759 MF: US:NIFMF07BSN1759 PF: US:NIFPF07BSN1759
DID Facility Interface Code: DID Service Order Code: DID Answer Supervision Code DID USOC Jack Type:	02RV2-T 9.0F A S.2 RJ21X
Please note the following: 1. DID services must be purchased from the local telephone company. 2. Refer to the Software Manual for detailed DID description, conditions, and programming instructions.	

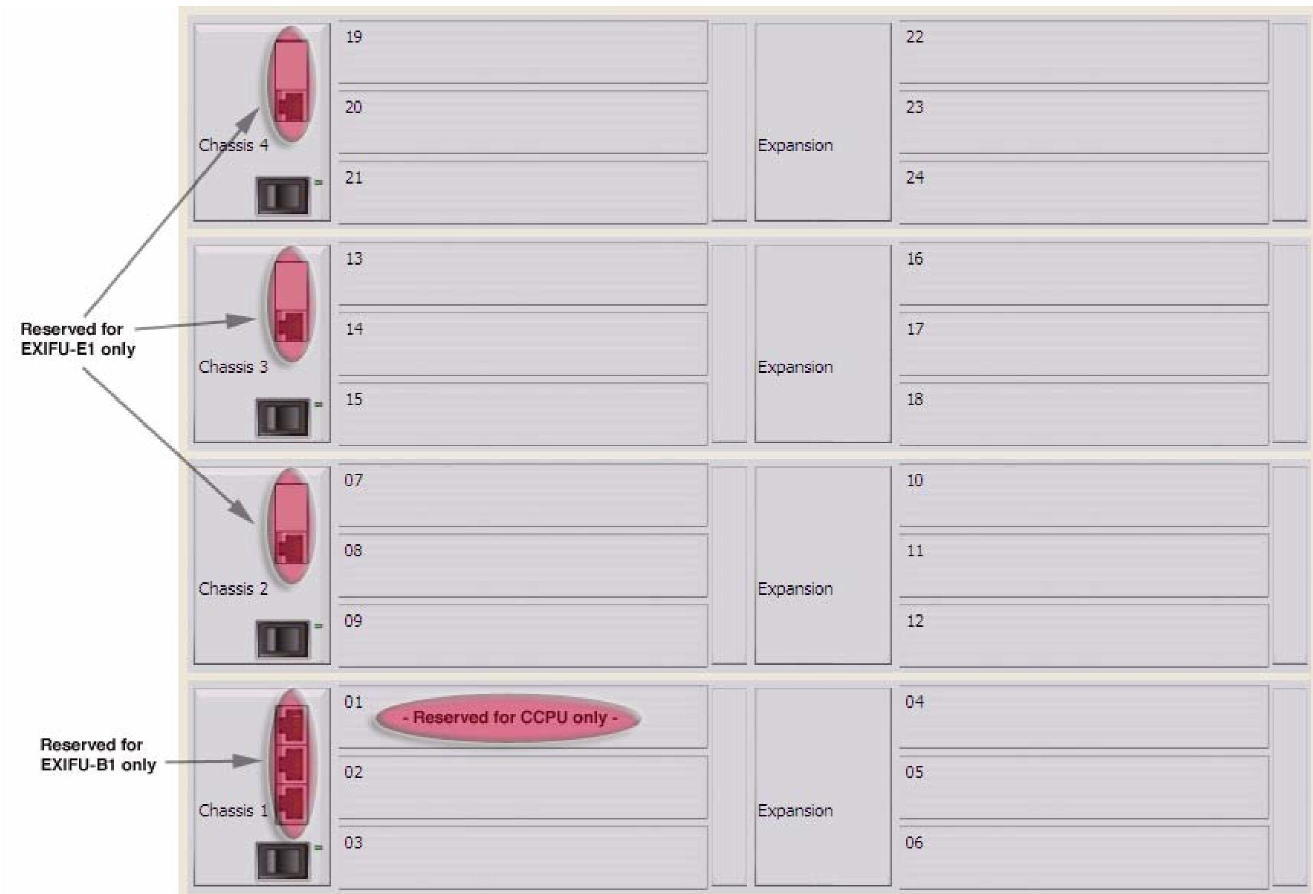
Cabling Requirements
1. Do not run station cable in parallel with the AC source, telex or computer, etc. If the cables are near cable runs to those devices, use shielded cable with grounded shields or install the cable in conduit. 2. When cables must be run on the floor, use cable protectors. 3. Cable runs for key terminals, single line telephones, Door Boxes, and 2PGDAD Modules must be a dedicated, isolated cable pair. 4. The Telco RJ21X and cross-connect blocks should install to the right of the Controlling /Base chassis. Extension blocks and cross-connect blocks should be installed to the left of the Controlling/Base chassis.

4

UX5000 Cable Requirements				
Device	**Cable Type**		**Cable Run Length (ft)**	**Notes**
Key Telephone, DSS Console:	CAT5	2-wire 26 AWG	• 1312	
		2-wire 24 AWG	• 2790 (no AC adapter or options) • 3940 (with AC adapter, no options)	
		2-wire 22 AWG	• 2624	
Single Line Telephone, Analog Terminals:	CAT5	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	3280 4921 8202	Constant Current: 20mA Loop Resistance: Below 600 Ohm (Includes resistance of telephone)
SLT Adapter: SLTAD to SLT:	CAT5	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG 2-wire 26 AWG	1148 1968 2624 2953	500 Ohms resistance between SLT adaptor and terminal (including phone's DC termination)
PSA Adapter:	CAT5	2-wire	Loop Resistance: Below 1700 Ohm	Internal resistance of PSA: 350 Ohms (20mA) (includes resistance of PSA)
2PGDAD Adapter:	CAT5	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	1312 1968 2624	
Using Under Carpet Cable: Use stranded wire under carpet cable. The use of this cable decreases the cable run length by 50%.				
CCPU to PC:	LAN UTP 10Base-T/ 100Base-TX	4-wire	328	
NTCPU to Hub:	LAN UTP 10Base-T/ 100Base-TX	10Base-T: 4--wire 100Base-TX: 8-wire	328	
CCPU to Printer:	LAN UTP 10Base-T/ 100Base-TX	10Base-T: 4--wire 100Base-TX: 8-wire	328	

UX5000 Cable Requirements				
Device	**Cable Type**		**Cable Run Length (ft)**	**Notes**
LAN Equipment (IP Telephone, Hub, etc.)	LAN (UTP)	LAN (UTP) cable with category 3 or more for 10Base-T, and 5 or more for 100 Base-TX	328.08 (100 meters)	Regardless of whether it receives power from the AC adapter or switching hub.
				
DID Trunks:	CAT5		Analog DID = 2,500 ohms OPX = 1,500 ohms	Includes SLT or exchange
Tie Line Trunks:	CAT5	2-wire 26 AWG 2-wire 24 AWG 2-wire 22 AWG	13,123 20,997 34,777	Limitation depends on the connecting Router, Multiplexer or Exchange. If the UX5000 is connected to another UX5000 directly, up to 1,500 ohms loop resistance (including system).
2/4/8BRIU to ISDN Terminals:	CAT5	4-wire 24 AWG	Point-Multipoint short connection: 329 Point-Multipoint long connection: 094 Point-Point connection: 164-	
1PRIU to ISDN Terminal:	CAT5	4-wire 24 AWG	164	From main device to IP terminal and within 328’.
ISDN Terminal:	CAT5	4-wire 24 AWG	Point-Multipoint Short Connection: 328 Point-Multipoint Extension: 984 Point-Point: 1640	

UX5000 Configuration Guide



Parts List

Common Equipment	
Description	**Part Number**
9.5” Controlling/Base Chassis - 3 blade chassis (IP3NA-3KSU-B1)	0910000
9.5” Expansion Chassis - 3 blade expansion chassis (IP3WW-3KSU-E1)	0910002
19” Chassis - 6 blade chassis (IP3NA-6KSU-A1)	0910004
Upper Joint Bracket Set (IP3WW-UPPER JOINT BRACKET SET)	0910011
Blank Slot Cover Set for 9.5” or 19” Chassis (IP3WW-BLANK SLOT COVER SET)	0910012
AC Power Cable (IP3NE- AC Power Cable -US)	670529
9.5” Chassis Battery Box, short term (10 minutes) (IP3WW-SMALL BATT BOX) (requires Battery Set, P/N 670511)	0910006
9.5” or 19” Chassis Battery Box, long term (3 hours) (IP3WW-LARGE BATT BOX) (requires Battery Set, P/N 670512)	0910007
19” Chassis Battery Box, short term (10 minutes) (2.3AH-12V BATT)	0910009
Long Term Battery Box Rack Mount Bracket Set (IP3WW-L BATT BOX RACK MOUNT BRACKET)	0910014
Battery Mount for 6-Slot Chassis	670509
Short Term Battery Set (12V 0.8Ah Battery) for 9.5” 3-Slot Chassis (IP3WW-SMALL BATT SET) : Includes 2 Batteries	670511
Short Term Battery Set for 19 6-Slot Chassis	670533
Long Term Battery Set (12V 0.8Ah) (IP3WW-LARGE BATT SET) : Includes 3 Batteries	670512
Battery Cable for Internal Battery 2U Chassis (IP3WW -BATT CABLE -INT BATT)	670530
Battery Cable for External Battery 2U Chassis (IP3WW -BATT CABLE -EXT BATT)	670531
Floor Mount Bracket Set for 9.5” Chassis (IP3WW-STD BRACKET SET)	0910013
Floor Mount set for all Chassis when using Long Term Battery Box (3 Blade, 6 Blade and CPU	670019
blade for IPS) (IP3NE-FLOOR MOUNT SET)	670508
Rack Mount Set for 6-Slot Chassis (IP3WW-STD BRACKET SET)	670510
Wall Mount Set for 6-Slot/CPU Blade Chassis	670513
Rack Mount Set for 3-Slot Chassis (IP3-STD BRACKET SET)	0910008

Blades	
Description	**Part Number**
Common	
CPU Main Processor (IP3NA-CCPU-A1)	0911001
Expansion Interface for Controlling Chassis (IP3WW-EXIFU-B1)	0911020
Expansion Interface for Additional Chassis-Bs (IP3WW-EXIFU-E1)	0911022
Voice Mail Daughter Board on CCPU - 16 Ports (IP3WW-16VMDB-A1)	0911024
Voice Mail Daughter Board w/Modem on CCPU - 16 Ports (IP3WW-16VMDB-B1)	0911026
Memory Expansion Board on CCPU (IP3WW-MEMDB-A1)	0911060
32-Port VoIP Daughter Board on CCPU (IP3WW-32VOIPDB-A1)	0911030
64-Port VoIP Daughter Board on CCPU (IP3WW-64VOIPDB-A1)	0911032
128-Port VoIP Daughter Board on CCPU (IP3WW-128VOIPDB-A1)	0911034
In-Skin Router Blade (IP3WW-RTU-B1)	0911062
Application Server/Voice Mail (IP3WW-APSU-A1)	0911064
Switching HUB Base (IP3WW-GSWU-B1)	0911066
Switching HUB Exp. (IP3WW-GSWU-E1)	0911068
Switching Hub Blade (SWHUBU)	
Conference Bridge (IP3WW-PVAU-A1)	0911070

Blades	
Description	**Part Number**
Trunk Interfaces	
4-Port Analog Loop/Ground Start Trunk Blade ((IP3WW-4COIU-LG1)	0911072
4 Port Analog Loop/Ground Start Trunk Daughter Board for Combination Blade or COIU Blade (IP3WW-4COIDB-LG1)	0911074
2-Port BRI Blade (IP3WW-2BRIU-A1)	0911048
2-Port BRI Daughter Board (IP3WW-2BRIDB-A1)	0911050
1-Port T1/PRI Blade (IP3WW-1PRIU-A1)	0911052
4-Port DID/OPX Blade (IP3WW-4DIOPU-A1)	0911054
4-Port E&M Tie Line Blade (IP3WW-4TILU-A1)	0911056
Station Interfaces	
8-Port Digital Station Blade (IP3WW-8ESIU-A1)	0911036
16-Port Digital Station Blade (IP3WW-16ESIU-A1)	0911038
8-Port Digital Station Daughter Board	0911076
4-Port SLT Blade with Message Waiting (IP3WW-4SLIU-A1)	0911040
8-Port SLT Blade with Message Waiting (IP3WW-8SLIU-A1)	0911044
4-Port SLT Daughter Board with Message Waiting on 4/8SLIU-A1 (IP3WW-4SLIDB-A1)	0911042
8-Port SLT Daughter Board with Message Waiting on 4/8SLIU-A1 (IP3WW-8SLIDB-A1)	0911046
Combination Blade, 8 digital/2 analog extensions (IP3WW-082U-A1)	0911058

Station Equipment	
Description	**Part Number**
Value Digital 2-Button (DG-2v) Terminal w/o LCD (BK) (IP3NA-2TH TEL(BK))	0910040
Value Digital 6-Button (DG-6v) Terminal w/ LCD (BK) (IP3NA-6TXH TEL(BK))	0910042
Enhanced Digital 12-Button (DG-12e) Terminal w/ LCD (BK) (IP3NA-12TXH TEL(BK))	0910044
Enhanced Digital 12-Button (DG-12e) Terminal w/ LCD (WH) (IP3NA-12TXH TEL(WH))	0910046
Enhanced Digital 24-Button (DG-24e) Terminal w/ LCD (BK) (IP3NA-24TXH TEL(BK))	0910048
Enhanced Digital 24-Button (DG-24e) Terminal w/ LCD (WH) (IP3NA-24TXH TEL(WH))	0910050
Enhanced Digital 12-Button Display Terminal + BCH (BK) (IP3NA-12TXH(BT) TEL(BK))	0910052
Enhanced Digital 12-Button Display Terminal + PSA (BK) (IP3NA-12TXH(PA) TEL(BK))	0910054
Enhanced Digital DESI-Less (DG-32e) (BK) (IP3NA-8LTXH TEL(BK))	0910056
Enhanced Digital DESI-Less (DG-32e) (WH) (IP3NA-8LTXH TEL(WH))	0910058
Value IP 2-Button (IP-2v) Terminal w/o LCD (BK) (IP3NA-2TIH TEL(BK))	0910060
Value IP 6-Button (IP-6v) Terminal w/ LCD (BK) (IP3NA-6TIXH TEL(BK))	0910062
Enhanced IP 12-Button (IP-12e) Terminal w/ LCD (BK) (IP3NA-12TIXH TEL(BK))	0910064
Enhanced IP 12-Button (IP-12e) Terminal w/ LCD (WH) (IP3NA-12TIXH TEL(WH))	0910066
Enhanced IP 24-Button (IP-24e) Terminal w/ LCD (BK) (IP3NA-24TIXH TEL(BK))	0910068
Enhanced IP 24-Button (IP-24e) Terminal w/ LCD (WH) (IP3NA-24TIXH TEL(WH))	0910070
Enhanced IP 12-Button Display Terminal + PSA (BK) (IP3NA-12TIXH(PA) TEL(BK))	0910072
Enhanced IP 12-Button Display Terminal + BCH (BK) (IP3NA-12TIXH(BT) TEL(BK))	0910074
Enhanced IP DESI-Less (IP-32e) (BK) (IP3NA-8LTIXH TEL(BK))	0910076
Enhanced IP DESI-Less (IP-32e) (WH) (IP3NA-8LTIXH TEL(WH))	0910078
IP-CTS Terminal (BK) (IP3NA-320TISXH TEL(BK))	0910080
Retrofit Keypad Assembly (BK) (IP3NA-BS-Retro(BK))	0910102
Retrofit Keypad Assembly (WH) (IP3NA-BS-Retro(WH))	0910103
12LK Kit (BK) (IP3NA-12LK-L(BK))	0910104
12LK Kit (WH) (IP3NA-12LK-L(WH))	0910106
DESI-Less LK/LCD Unit for Digital Terminal (BK) (IP3NA-8LKD-L(BK))	0910108
DESI-Less LK/LCD Unit for Digital Terminal (WH) (IP3NA-8LKD-L(WH))	0910110
DESI-Less LK/LCD Unit for IP Terminal (BK) (IP3NA-8LKI-LD(BK))	0910112
DESI-Less LK/LCD Unit for IP Terminal (WH) (IP3NA-8LKI-LD(WH))	0910114

Corded Headsets	
Description	**Part Number**
Plantronics Polaris Headsets:	
Polaris SupraPlus/NC-M (monaural with noise canceling transmitter)	750643
Polaris SupraPlus/NC-B (binaural with noise canceling transmitter)	750645
Polaris SupraPlus/VT-M (monaural with voice tube transmitter)	750644
Polaris Encore/VT-M (monaural with voice tube transmitter)	750634
Polaris Encore/NC-B (binaural with noise canceling transmitter)	750635
Polaris Tristar/VT-M (monaural with noise canceling transmitter)	750630
Polaris Mirage/VT-M (monaural with voice tube transmitter)	750631
Accessories/Replacement Parts for Polaris Headsets:	
Polaris Extension Cable-10 Ft (for all Polaris models)	750655
Ear Cushion (Pkg of 2) (for Supra & Encore models)	750656
Clothing Clip (Qty 1) (for Mirage, Tristar & encore models)	750657
Wind Screen (Qty 1) (for Supra NC models - not SupraPlus)	750650
Clear Voice Tube (for Encore & Tristar models)	750652
Clear Voice Tube (for Mirage & Supra models)	750651
Rainbow Voice Tube Pack (Pkg of 6) (for Encore & Tristar models)	750654
Rainbow Voice Tube Pack (Pkg of 6) (for Mirage & Supra models)	750653
Corded Headset 2-Page Brochure (Pkg of 25)	750629

Peripheral Station Equipment	
Description	**Part Number**
Analog Port Adapter w/Ringer (APR) (IP3NA-APR ADAPTER(BK))	0910082
Analog Recording Adapter (ADA) - black (IP3WW-ADA ADAPTER(BK))	0910084
Bluetooth Hub Adapter (Class 2) (BHA) - black (IP3NA-BHA ADAPTER(BK))	0910086
PSTN Adapter for Analog (PSA) - black (IP3NA-PSA ADAPTER(BK))	0910088
PSTN Adapter for Analog (PSA) - white (IP3NA-PSA ADAPTER(WH))	0910090
Bluetooth Cordless Handset with Hub (Class 1) (BCH) - black (IP3NA-BCH ADAPTER(BK))	0910092
60-Button Console (BK) (IP3WW-60D DSS(BK))	0910094
60-Button Console (WH) (IP3WW-60D DSS(WH))	0910096
16-Button DLS (16LK Unit) (BK) (IP3WW-16DL DLS(BK))	0910098
16-Button DLS (16LK Unit) (WH) (IP3WW-16DL DLS(WH))	0910100
2PGDAD Module (for Door Box/Page/ACI) (IP1WW-PGDAD)	0910015
SLT Adapter (IP1NA-1SLTAD)	0910010
Door Box	92245
In-Line Power Adapter (ILPA) for IP Terminals	780122
AC Adapter (AC-2R)	780135
Braille Dial Designation Sticker (for all UX5000 Terminals	690612
Wall-Mount for Use With Adapters	0910120
Wall-Mount for 60-Button DSS Console	680754

Equipment	
Description	**Part Number**
Compact Flash Media for IVR Application (AKS IVR APP CF) Q-master External Application Server (AKS Qmaster APP Server) XML Application (Common Key, IPS, UBSD) (AKSU XML APP CD) Hotel/Motel PMS Application (AKSU PMS APP UNIT) Kentrox Satellite 932 CSU	670839 670841 670842 859451
MISC. Compact Flash Media 4 Pt/16 Hr 256M for IntraMail with VRS (AU IntraMail - 256M APP CF) Compact Flash Media 4 Pt/32 Hr 512M for Intramail (AU IntraMail - 512M APP CF) Compact Flash Media 2G for UX Mail - 125 hour (AU UM - 2G APP CF) Compact Flash Media 8G for UX Mail - 550 Hour (AU UM - 8G APP CF) Compact Flash Media for Multi-Media Conference Bridge (AU CONF BRIDGE APP CF) Compact Flash Media for IVR Application (AU IVR APP CF) Desktop Application CD for User Desktop (AU DESKTOP PC APP CD) consists of: - UX5000 Desktop Suite (Softphone, PC Assistant, PC Attendant) - Interactive UG - End-User Programming - UX5000 inDepth System Application CD (AU SYSTEM PC APP CD) consists of: - UX5000 CallAnalyst - UX5000 PCPro - UX5000 inDepth MIS	 0910505 0910506 0910530 0910531 0910532 0910533 0910527 0910528

4

DESI Labels	
Description	**Part Number**
DESI Sheet for 2-Button IP/Digital Value Terminal (3parts/sheet) (DESI IP3NA-2T (25 PKG))	0910700
DESI Sheet for 6-Button IP/Digital Value Terminal (3parts/sheet) (DESI IP3NA-6T (25 PKG))	0910701
DESI Sheet for 12-Button IP/Digital Enhanced Terminal (2parts/sheet) (DESI IP3NA-12T (25 PKG))	0910702
DESI Sheet for 24-Button IP/Digital Enhanced Terminal (2parts/sheet) (DESI IP3NA-24T (25 PKG))	0910703
DESI Sheet for Standard Dial/FNC (DESI IP3NA-NewUI DIAL (25 PKG))	0910704
DESI Sheet for CTS Terminal Dial/FNC (DESI IP3NA-SOPHI DIAL (25 PKG))	0910705
DESI Sheet for Retro Dial/FNC (DESI IP3NA-Retro DIAL (25 PKG))	0910706
DESI Sheet for 16-Button DLS (16LK) (3parts/sheet) (DESI IP3NA 16LK (25 PKG))	0910707
DESI Sheet for 60-Button DSS Console (1parts/sheet) (DESI IP3WW-60D (25 PKG))	0910708

Spare Parts	
Description	**Part Number**
LCD Unit W/ backlit (BK) (IP3NA-LBU-LCD(BK))	0910116
LCD Unit W/ backlit (WH) (IP3NA-LBU-LCD(WH))	0910118
Wall Mount Unit (IP3NA-WM)	0910120
Spare Narrow Band Handset (BK) (IP3NA-HANDSET Narrow (BK))	0912001
Spare Narrow Band Handset (WH) (IP3NA-HANDSET Narrow (WH)	0912002
Spare Wide Band Handset (BK) (IP3NA-HANDSET Wide (BK))	0912003
Spare Wide Band Handset (WH) (IP3NA-HANDSET Wide (WH))	0912004
Spare Handset Cord 12 FT (BK) (IP3WW-Handset Cord 12 FT (BK) SET)	0912005
Spare Handset Cord 12 FT (HW) (IP3WW-Handset Cord 12 FT (WH) SET)	0912006
Spare Handset Cord 25 FT (BK) (IP3WW-Handset Cord 25 FT (BK) SET)	0912007
Spare Handset Cord 25 FT (BK) (IP3WW-Handset Cord 25 FT (WH) SET)	0912008
Spare Handset Hanger (BK) (IP3WW-Handset Hanger (BK) SET)	0912009
Spare Handset Hanger (WH) (IP3WW-Handset Hanger (WH) SET)	0912010
Spare Line Cord (BK) (IP3WW-Line Cord (BK) SET)	0912011
Push to Mute Handset (IP3NA PTM Handset (BK))	0912012
Push to Talk Handset (IP3NA PTT Handset (BK))	0912013
Spare Plastic Cover Kit (2 BTN) (IP3WW-LKPANEL 2BTN SET)	0912014
Spare Plastic Cover Kit (6 BTN) (IP3WW-LKPANEL 6BTN SET)	0912015
Spare Plastic Cover Kit (12 BTN) (IP3WW-LKPANEL 12BTN SET)	0912016
Spare Plastic Cover Kit (24 BTN) (IP3WW-LKPANEL 24BTN SET)	0912017
Spare Plastic Cover Kit (16 BTN) (IP3WW-LKPANEL 16BTN SET)	0912018
Spare Plastic Cover Kit (60 BTN) (IP3WW-LKPANEL 60BTN SET)	0912019
Digital Base (IP3NA-Digital Value Base UNIT)	0912020
Digital Standard LCD (BK) (IP3NA-DLU-LCD(STD) (BK))	0912021
Digital Standard LCD (WH) (IP3NA-DLU-LCD(STD) (WH))	0912022
IP Base (IP3NA-IP Value Base UNIT)	0912023
IP LCD w/ Backlit (BK) (IP3NA-ILU-LCD(STD) (BK))	0912024
IP LCD w/ Backlit (WH) (IP3NA-ILU-LCD(STD) (WH))	0912025
Ten Key Dial Pad Kit (IP3NA-TENKEY(STD) SET)	0912026
Standard Function Key (BK) (IP3NA-FNCKEY(STD) (BK) SET)	0912027
Standard Function Key (WH) (IP3NA-FNCKEY(STD) (WH) SET)	0912028
12-Line Key w/ Soft Key (IP3WW-12LKSoft SET)	0912029
12-Line Key (IP3WW-12LK SET)	0912030
Cradle (BK) (IP3NA-Cradle(STD) (BK))	0912032
Cradle (WH) (IP3NA-Cradle(STD) (WH))	0912033
Enhanced Telephone Directory Card Unit	0912040
Value Telephone Directory Card Unit	0912041
IP3NA-8WV AC CHARGER	0910121
IP3NA-8WV AC ADAPTER	0910122
TYX3588-010197 (Headset Adapter)	0910123

NEC

NEC Unified Solutions, Inc.
4 Forest Parkway
Shelton, CT 06484
www.necux5000.com

Important Telephone Numbers	
Sales Support and General Information	800-365-1928
Technical Service	
Current Products	203-925-8801
Current Products After Hours (Emergencies Only)	203-929-7920
Discontinued Products	900-990-2541
Customer Service	203-926-5444
Technical Training	203-926-5430

Have any comments, suggestions or corrections for this guide?

Forward your comments, suggestions or corrections to the above address or web site.

NEC

NEC Unified Solutions, Inc.
4 Forest Parkway
Shelton, CT 06484

www.necux5000.com

(See inside back cover for contact information.)

0913010
June 9, 2008, Rev 1

Printed in U.S.A.