# *UNIVERGE* NEAX 2000 IPS

## *Family of Products*

## General Description

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**NEC Unified Solutions, Inc.**

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The *UNIVERGE* NEAX 2000 IPS General Description Guide has been developed to provide information on the *UNIVERGE* NEAX 2000 IPS. The information provided has been compiled from a variety of available documentation and has been consolidated into a single manual.

Information concerning questions not covered in this guide, corrections and/or comments are most welcome and should be sent to:

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# Regulatory Information

## Regulatory Requirements

The Federal Communications Commission (FCC) has established rules that permit the PBX to be directly connected to the telephone network. A jack is provided on party lines or coin lines.

The telephone company may make changes in its technical operations and procedures. If such changes affect the compatibility or use of the PBX, the telephone company must provide adequate notice of the changes.

This equipment complies with the requirements in Part 15 of FCC Rules for a Class A computing device. Operation of this equipment in a residential area may cause unacceptable interference to radio and TV reception requiring the operator to take whatever steps are necessary to correct this interference.

## FCC Part 15 Requirements

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

**Warning:** *This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the instruction manual, may cause interference to radio communications. It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference in which case the user at his own expense will be required to take whatever measures may be required to correct the interference.*

# FCC Part 68 Registration

## Company Notification

Before installing the PBX to the telephone network, the telephone company must be provided with the following:

- Your telephone number
- The FCC registration numbers:

| | JAPAN | USA |
|---|---|---|
| PBX | AY5JPN-20542-PF-E | AY5USA-21582-PF-E |
| Hybrid | AY5JPN-20543-MF-E | AY5USA-21583-MF-E |
| Key System | AY5JPN-20586-KF-E | AY5USA-21584-KF-E |

The Ringer Equivalence Number is 1.6B; the required USOC jacks are RJ21X, RJ2EX, RJ2GX, and RJ49C.

**Note:** *Limitations on features exist if the system is registered as a KF system. Refer to Features and Specifications for details.*

## Location of FCC Compliance Labels

Labels stating the *UNIVERGE* NEAX 2000 IPS FCC registration number and compliance with FCC Parts 15 and 68 are attached on the inside of the system's front cover. Label examples are as follows:

"This equipment complies with the requirements in Part 15 of FCC Rules for a Class A computing device. Operation of this equipment in a residential area may cause unacceptable interference to radio and TV reception requiring the operator to take whatever steps are necessary to correct the interference."

*UNIVERGE* NEAX 2000 IPS
Complies With Part 68 FCC Rules
FCC Registration Numbers
Ringer Equivalence: 1.6B
NEC Unified Solutions, Inc. MADE IN USA

## FCC Requirements for Private Line Operations

In order to connect this system to the private line network, provide the telephone company with:

- The quantities and USOC numbers of the required jacks (See the following table.)
- The sequence in which the trunks are to be connected
- The facility interface codes by position
- The Ringer Equivalence Number or service order code, as applicable, by position

| Mfg's Port ID | Facility Interface Code | Network Jacks | Service Order Code |
|---|---|---|---|
| PN-8COTS | 02GS2 | RJ21X | |
| PN-8COTQ | 02LS2 | RJ21X | |
| PN-4DIDA | 02RV2-T | RJ21X | |
| PN-AUCA | 02RV2-T | RJ21X | |
| PN-AUCA | 0L13A, 0L13B, 0L13C | RJ21X | 9.0F |
| PZ-8PFTA | 02LS2 | RJ21X | |
| PN-40DTA | TL11M | RJ2EX | 9.0F |
| PN-40DTA | TL31M | RJ2GX | 9.0F |
| PN-24DTA-C | 04DU9-BN | N/A | 6.0P |
| PN-24DTA-C | 04DU9-DN | N/A | 6.0P |
| PN-24DTA-C | 04DU9-1KN | N/A | 6.0P |
| PN-24DTA-C | 04DU9-1SN | N/A | 6.0P |
| PN-24DTA-C | 04DU9-1ZN | N/A | 6.0P |
| PN-DTA (PRT) | 05DU9-BN | N/A | 6.0P |
| PN-DTA (PRT) | 04DU9-BN | N/A | 6.0P |
| PN-DTA (PRT) | 04DU9-1KN | N/A | 6.0P |
| PN-DTA (PRT) | 04DU9-1SN | N/A | 6.0P |
| PN-DTA (PRT) | 04DU9-1ZN | N/A | 6.0P |
| PN-DTA (CCT) | 04DU9-BN | N/A | 6.0P |
| PN-DTA (CCT) | 04DU9-DN | N/A | 6.0P |
| PN-DTA (CCT) | 04DU9-1KN | N/A | 6.0P |
| PN-DTA (CCT) | 04DU9-1SN | N/A | 6.0P |
| PN-DTA (CCT) | 04DU9-1ZN | N/A | 6.0P |
| PN-4BRT | 02IS5 | N/A | 6.0Y |

## Service Requirements

In the event of equipment malfunction, all repairs will be performed by NEC or an authorized distributor. It is the responsibility of users requiring service to report the need for service to NEC or to one of their authorized distributors.

If trouble is experienced with this equipment, please contact NEC Unified Solutions, Inc., at 800-TEAM NEC (800-832-6632) for repair and/or warranty information. If the trouble is causing harm to the telephone network, the telephone company may request that you remove the equipment from the network until the problem is resolved.

If the equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice is not practical, the telephone company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The telephone company may make changes in its facilities, equipment, operations, or procedures that affect the operation of the equipment. If this happens, the telephone company will provide advance notice so that you can make necessary modifications in order to maintain uninterrupted service.
NO REPAIRS CAN BE DONE BY THE CUSTOMER.

# Direct-Inward Dialing (DID) Calls

Allowing this equipment to be operated in such a manner as to not provide for proper answer supervision is a violation of Part 68 of the FCC's rules.

PROPER ANSWER SUPERVISION IS WHEN:

a.) This equipment returns answer supervision to the PSTN when DID calls are:

- 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

b.) This equipment returns answer supervision on all DID calls forwarded to the PSTN.

- Permissible exceptions are:
- A call is unanswered
- A busy tone is received
- A reorder tone is received

## EQUAL ACCESS REQUIREMENTS

This equipment is capable of providing users access to interstate providers of operator services through the use of access codes. Modification of this equipment by call aggregators to block access dialing codes is a violation of the Telephone Operator Consumers Act of 1990.

**Caution:** *The use of a monitoring, recording or listening devices to eavesdrop, monitor or record telephone conversations or other sound activities, whether or not contemporaneous with its 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 the telephone conversation, such as using a beep tone or other notification methods or require the consent of all parties to the telephone conversation, prior to monitoring or recording a telephone conversation. Some of these laws incorporate strict penalties.*

# Regulatory Information On Analog Telephones

NEC single-line telephones comply with Part 68 of FCC Rules. On the bottom of the equipment is a label that states, among other information, the FCC registration number and ringer equivalence number (REN) for the equipment. If requested, this information should be provided to the telephone company.

The equipment uses the following USOC jacks: RJ11C.

The equipment should be used only behind a PBX or KTS. The REN is used to determine the number of devices that may be connected to the telephone line. Excessive RENs on the telephone line may result in the devices not ringing in response to an incoming call. In most, but not all, areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to the line as determined by the total RENs, contact the telephone company to determine the maximum REN for the calling area.

## Hearing Aid Compatibility

The Dterm terminals provided for the *UNIVERGE* NEAX 2000 IPS are hearing aid compatible. FCC rules prohibit the use of non-hearing aid compatible telephones.

NEC-type single-line telephone sets used in conjunction with the *UNIVERGE* NEAX 2000 IPS are hearing aid compatible. If other than NEC-type single-line telephone sets are to be used with this system, ensure that these are hearing aid compatible.

## Industry Canada CS-03

Certification number: 140 5976 A

Load Number of the equipment: 1.0

**NOTICE**: The Industry Canada label identifies certified equipment. The certification means that the equipment meets certain telecommunications network protective operational and safety requirements. The department does not guarantee the equipment will operate to the user's satisfaction.

Before installing the equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above conditions might not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or installations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request that the user disconnect the equipment.

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

**Caution:** *Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.*

**NOTICE**: The Load Number assigned to each terminal device denotes the percentage of the total load to be connected to a telephone loop which is used by the device, to prevent overloading. The termination on a loop may consist of any combination of devices subject only to the requirement that the total of the load numbers of all the devices does not exceed 100.

# Safety Certifications

This equipment has been listed by Underwriters Laboratories and found to comply with all the applicable requirements of the standard for telephone equipment U.L. 1459. This equipment complies with Canadian Standards Association's standard C 22.2 No. 225.

## Safety Considerations

When using telephone equipment, basic safety precautions should always be followed to reduce the risk of fire, electric shock, and injury. Precautions include the following:

- Never install telephone wiring during a lightning storm.
- Never install a telephone jack in a wet location, unless the jack is specifically designed for wet locations.
- Never touch an uninsulated telephone wire or terminal, unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.

***Note:*** *More detailed precautions are included in this manual.*

## Safety Instructions

1. Never install telephone wiring during a lightning storm.
2. Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.
3. Never touch un-insulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
4. Use caution when installing or modifying telephone lines.
5. Read and understand all instructions.
6. Follow all warnings and instructions marked on the product.
7. Unplug this product from the wall outlet before cleaning. Do not use liquid cleaners or aerosol cleaners. Use a damp cloth for cleaning.
8. Do not use this product near water, for example, under water pipes near a bathtub, sink, or laundry tub, in a wet basement, or near a swimming pool.
9. Do not place this product on an unstable cart, stand, or table. The product may fall, causing serious damage to the product.
10. Slots and openings in the cabinet and the back or bottom are provided for ventilation, to protect it from overheating, these openings must not be blocked or covered. The openings should never be blocked by placing the product on a bed, sofa, rug, or other similar surface. This product should never be placed near or over a radiator or heat register. This product should not be placed in a built-in installation unless proper ventilation is provided.
11. This product should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power source available, consult with your local power company.
12. This product is normally connected with a three-wire grounding type plug, a plug having a third (grounding) pin. This plug will only fit into a grounding type power outlet. This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace your obsolete outlet. Do not defeat the safety purpose of the grounding type plug.
13. Do not allow anything to rest on the power cord. Do not locate this product where the cord will be abused by persons walking on it.
14. Do not overload wall outlets and extension cords as this can result in the risk of fire or electric shock.
15. Never push objects of any kind into this product through cabinet slots as they may touch dangerous voltage points or short out parts that could result in a risk of fire or electric shock. Never spill liquid of any kind on the product.
16. To reduce the risk of electric shock, do not disassemble this product, but take it to a qualified serviceman when some service or repair work is required. Opening or removing covers may expose you to dangerous voltages or other risks. Incorrect reassembly can cause electric shock when the appliance is subsequently used.

17. Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:
    a.) When the power supply cord or plug is damaged or frayed.
    b.) If liquid has been spilled into the product.
    c.) If the product has been exposed to rain or water.
    d.) If the product does not operate normally by following the operating instructions. Adjust only those controls that are covered by the operating instructions, because improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the product to normal operation.
    e.) If the product has been dropped or the cabinet has been damaged.
    f.) If the product exhibits a distinct change in performance.
18. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electric shock from lightning.
19. Do not use the telephone to report a gas leak in the vicinity of the leak.
20. When installing the LC card for providing the analog telephone interface that is connected to off-premise site, use PN-4LLCB card.
21. Do not install the PN-4ODTA card for providing the analog trunk interface that is connected to off-premise site.
22. Warning for US and Canada only

**WARNING**

**This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:**
**(1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.**
**This Class A digital apparatus complies with Canadian ICES-003.**
**Cet appareil numérique de la classe A est conforme a la norme NMB-003 du Canada.**

# Chapter 1 Introduction

This manual provides an overview of the *UNIVERGE* NEAX®2000 IPS (Internet Protocol Server) stored program control digital electronic PBX. An introduction to the technical characteristics is included, along with a description of available system applications.

## System Information – *UNIVERGE* NEAX®2000 IPS

**The *UNIVERGE* NEAX® 2000 IPS** (Internet Protocol Server) is a full-featured IP based communications system providing a rich feature set with pure Voice over IP (VoIP) communications (peer to peer connections), across corporate Local and Wide Area Networks (LAN and WAN). The *UNIVERGE* NEAX 2000 IPS DtermIP telephones are designed to provide a converged infrastructure at the desktop, with a 100 Base T Ethernet connection to the LAN and built-in hub for a PC connection to the telephone itself. The system can provide peer-to-peer connections between DtermIP telephones with voice compression, offering existing Dterm Series i telephone features. On the WAN side, the system can provide peer-to-peer connections over IP networks with the voice compression, on a CCIS basis (CCIS over IP) or Remote PIM (Remote PIM over IP).



***UNIVERGE* NEAX 2000 IPS**

The *UNIVERGE* NEAX 2000 IPS can provide legacy station/trunk interfaces to support the existing Time Division Multiplexing (TDM) based infrastructure, such as analog telephones, analog networks, and digital networks (T1/E1, ISDN etc.). At maximum configuration, the system can provide 1020 ports for IP and legacy devices, and 256 ports for Application cards. Communications between legacy stations/trunks and DtermIP telephones/IP networks are made via IP PAD, which converts packet-based voice data to TDM-based voice data, and vice versa. Both peer-to-peer connections and TDM-based connections are controlled the Main Processor (MP) card. The MP card incorporates a built-in Device Registration Server (DRS) and a single interface point of IP connection to IP telephone, MATWorX, and OAI/ACD servers.

*UNIVERGE* NEAX 2000 IPS users have access to hundreds of service features that are used in building unique telephony applications that enhance productivity, reduce operating costs and improve communications efficiently. The innovative modular hardware and software design allows efficient, effective growth within each module from its minimum to its maximum configuration. The *UNIVERGE* NEAX 2000 IPS software design is as advanced as its hardware. It ensures the system will support evolving applications and have the reliability needed to compete in today's world and into tomorrow's. The software is designed with modularity in mind. Together, these modular building blocks allow customers to initially buy what they need and add capacity and capabilities as the business demands, resulting in a greater degree of cost control for new installations and for upgrades to features, capacities and the software versions.

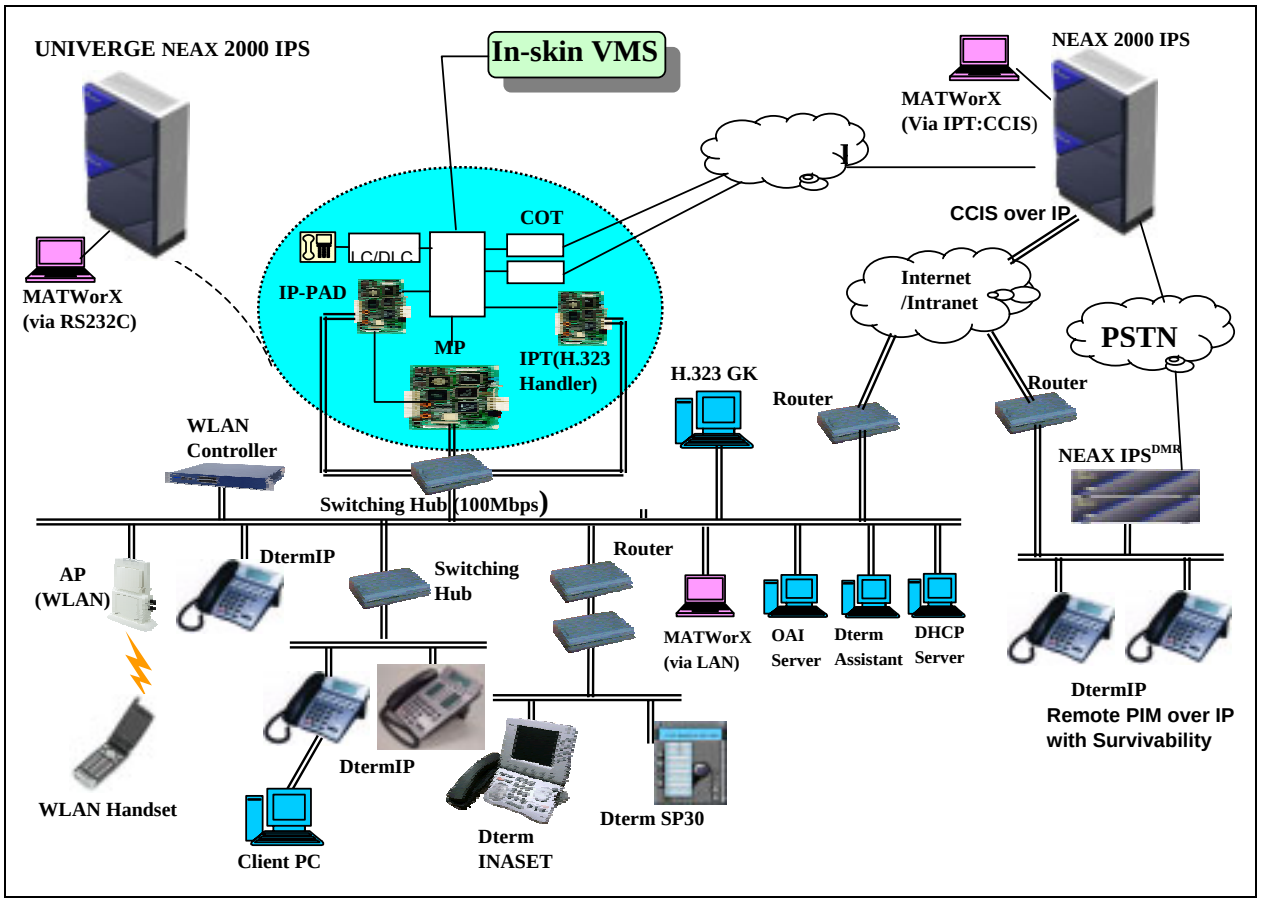



**Figure 1-1 System Outline of an IPS**

# Hardware Architecture

### Hybrid System of IP (peer-to-peer connection) and TDM Switching

The *UNIVERGE* NEAX 2000 IPS supports both pure IP switching (peer-to-peer connections) and Time Division Switching (TDM). The pure IP switching is provided for communications between DtermIPs and for CCIS/Remote PIM connections with another *UNIVERGE* NEAX 2000 IPS/ *UNIVERGE* NEAX IPS$^{DML}$/ 2400 IPX. On the other hand, the TDM switching is provided for communications between legacy stations/trunks. Connections between DtermIP/CCIS or Remote PIM over IP and legacy stations/trunks are made via IP PADs, which converts packet-based voice data to TDM-based voice data, and vice versa.

### Powerful, One-board Main Processor (MP) with Integrated Functionality

The *UNIVERGE* NEAX 2000 IPS Main Processor (MP) is the heart of pure IP connections and TDM-based connections. The MP employs a high-speed CPU, which is equivalent with Pentium. With this processing power and System On Chip (SOC) technology, the MP integrates Device Registration Server (DRS), and AP01 (OAI) functions. Also, by means of today's advanced LSI technology, the MP card size is minimized and On-board Ethernet Interface card is mounted on the MP without using an additional slot space in the PIM. This interface card is linked with LAN for call control processing of DtermIP and inter-work with MATWorX and OAI server.

The MP provides LAN control function, System-based Device Registration Server (DRS), Built-in FP, Built-in OAI, Built-in SMDR, Built-in CCH-IPT, 33 MHz PCI BUS, Memory (Basic/Expansion), TDSW (1024 CH × 1024 CH), 16-line CFT, PB Sender, Clock, PLO two ports (Receiver Mode/Source Mode), two RS-232C Ports, two-line DAT (Recording duration: a maximum of 128 seconds), DK, 4-line PB Receiver, Modem for remote maintenance (33.6 kbps), internal Music-on-Hold Tone, BUS Interface. BUS Interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. One card is required per system.

### Reduced Hardware with IP based Architecture

The DtermIPs connected to the LAN do not require DLC cards because they can be interfaced directly with the LAN and connected with peer-to-peer basis. When the DtermIP is connected to a station/trunk that is using TSW, the speech path between LAN and TSW is made via IP PAD under the call processing control of the MP. The DtermIP can be expanded simply adding the terminal itself and IP PAD if traffic volume is increased. With this system architecture, the hardware such as DLC, PIM, Power Supply etc. is reduced and easy moves, adds, and changes can be realized.

| Standard TDM Hardware | Peer to Peer IP Hardware |
|---|---|
| ▪ Line & Trunk Cards<br>▪ Application Processors<br>▪ Firmware Processors | ▪ SPN-8IPLA IP PAD<br>▪ PZ-M606-A<br>▪ PN-24IPLA IP PAD*<br>** The PN-24IPLA is a daughter board for the 8IPLA when up to 32 IP PADs for desired.* |

### Enhanced Built-in Firmware Processor (FP) on MP

The Firmware Processor card (FP) provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, and monitors I/O Bus and PCM BUS. When the system consists of three PIMs or more, one each of this card is mounted respectively in PIM 2, PIM 4, and PIM 6.

### Extended Application Processor (AP) Port Capacity

The *UNIVERGE* NEAX 2000 IPS provides maximum 256 AP ports and it is independent of the 1020 ports for the Line/Trunk (LT), therefore, more AP cards such as T1/E1 digital link cards can be used in the system.

### Universal Slot

One PIM provides 12 card slots for Line/Trunk (LT). Also, these card slots can be used for Application Processor (AP) cards without complicated limitation. This makes easy quotation and installation, and more AP cards can be mounted in one PIM.

### Unified Circuit Card Size

All circuit cards for the *UNIVERGE* NEAX 2000 IPS are designed in one size (PN-type), and installed in the PIM. This maximizes the efficiency of slot utilization of the PIM.

### High Density Line/Trunk Cards

The major line/trunk cards used in the *UNIVERGE* NEAX 2000 IPS are provided with 8 circuits per card. This allows the physical system size to be compact.

### DC/DC Power Supply for –48V

The PIM houses optional DC/DC Power Supply for the cards which require –48V power such as the CSI card used for interface of Zone Transceiver of wireless system. Since this power supply is mounted in the space under the AC/DC power, no additional Power Module/card slots are required.

### Built-in DRS (Device Registration Server) on MP

The *UNIVERGE* NEAX 2000 IPS incorporates DRS (Device Registration Server) on the MP. DRS provide Log-in/Log-out management of DtermIP including Registration and Authentication. Also, the built-in DRS can be inter-worked with DHCP server to provide easy administration on IP address.

### Office Data Backup Enhancement

The office data of the *UNIVERGE* NEAX 2000 IPS is stored in Flash ROM; therefore the backup period is extended compared with previous IVS series which were using RAM with battery.

### Various Installation Methods

To meet the specific needs of the customer's environment, the *UNIVERGE* NEAX 2000 IPS provides the following installation methods:

- Floor Standing Installation
- Wall-mounting Installation
- IEC standard 19 inch Rack-mounting Installation

### Station to Station Connection

For DtermIP to DtermIP connection (Peer to Peer connection), the voice data is transmitted and received directly between DtermIPs on the LAN. For Dterm Legacy terminal connection, the IP-PAD card and VCT card are required to transmit and receive the voice data. These cards are used to control and convert the voice data. The MP card in either of the connections above manages the control signals.

### CCIS Connection

DtermIP to DtermIP connection (Peer to Peer connection) via CCIS is available only when the destination office is *UNIVERGE* NEAX 2000 IPS or *UNIVERGE* NEAX 2400 IPX. The system provides only Point to Multipoint connection.

### Maintenance

MATWorX IPS is used as the maintenance program for the *UNIVERGE* NEAX 2000 IPS. Direct connection (RS-232C), Modem connection and LAN (TCP/IP) connections are available to connect to the MAT (Maintenance Administration Terminal).

### Dual MP System

The system complies with dual control system on Main Processor.

> **Note:** *Since the system employs Cold Standby processing in MP changeover, the calls in progress are terminated as a result of the MP changeover. Also, during the MP changeover, the call originating/receiving and service feature access are not effective. (It takes about 30 to 60 seconds to complete the MP changeover.)*

### Remote PIM over IP with Survivability

The *UNIVERGE* NEAX 2000 IPS can have a PIM installed at a remote site through an IP network. At the main site, the *UNIVERGE* NEAX 2000 IPS/IPS$^{DML}$ is installed and *UNIVERGE* NEAX 2000 IPS /IPS$^{DMR}$ are installed at the remote site. The main site controls call processing and service feature access for station users located at both the main and remote sites. When the Remote PIM cannot be connected with main site due to the IP network and/or main PBX failure, the Remote PIM initializes the system and re-starts operation by its own Main Processor (survival mode). In the survival mode, almost all service features are provided to the station users accommodated in Remote PIM. When the IP network/main PBX recovers, the Remote PIM can be restored to normal mode with a system initialization by manual operation or automatically (Selectable by system data setting).

- IPS Remote PIM with CP24-D MP
- IPS$^{DMR}$ with CP31-D MP

# Software Architecture

*UNIVERGE* NEAX IPS systems offer Software Keys (TDM and AP) and Peer-to-Peer IP Seat Licenses, which are designed with modularity in mind. This approach allows customers a greater degree of cost-control for new installations and for upgrades to features, capacities and software series.

Installing the *UNIVERGE* NEAX IPS systems requires activating Basic System software, optional features key software, and seat licenses. The software is loaded into MP by using MATWorX IPS. Once the software is loaded, that software cannot be used to OTHER system (Copy protected). A maximum of 32 Key CD can be loaded per system. If the system is Dual MP System, the software is required to load on each MP in order to register the software data into each MP. (Dual MPs are configured with the same serial number)

**Generic Program (Basic System Software)**

| Description | Remarks |
|---|---|
| 64 Port System Software | Basic Business/Hotel/Motel Features for:<br>64 LT Ports, 5 T1/DCH, 48 ISDN-BRI Trunks.<br><br>NEC Customer Software License Agreement Required |

**Optional Software**

| Description | Remarks |
|---|---|
| **IPS Software Key (CD) (Parent Capacity Option)** | |
| IPS Software Key (CD) | Compact Disk that holds selected Key files from below |
| **Capacity Option (used w/CD)** | |
| LT 64 PORT | 64 Port Line/Trunk Key (incremental)<br>Expands LT Ports from 64 to 1020 Ports in increments of 64. Software and provides LT Port Licenses in 64 port increments from 64 to1020 ports. Stand alone system maximum 512 LT ports, Remote PIM Network maximum 1020 LT ports. |
| CCIS Link (1)<br>CCIS Link (4)<br>CCIS Link (8) | Adds support for one CCIS Link<br>Adds support for four CCIS Link<br>Adds support for eight CCIS Link |
| IPT Card (1)<br>IPT Card (4)<br>IPT Card (8) | Adds support for one IP trunk card<br>Adds support for four IP trunk card<br>Adds support for eight IP trunk card |
| Event Based CCIS (ECCIS) Key | Adds Event Based CCIS capability |
| Wireless Software Key | Adds Wireless with support of 128 ZT's and 256 PS's |
| Wireless Software Key 8 PS | Adds 8 Wireless ports each key above the initial 256 |
| T1/PRI Software Key | Expands capacity of T1 to 10 and PRI to 32 spans |
| IP Remote PIM 1 Site License | Adds IP Remote Capability and is required for each Remote Site |
| 8 IP Seat License Key | DtermIP terminals are controlled by the CPU and do not use digital line cards, Instead they require DtermIP seat licenses. The licenses are available in 8 seat increments and are cumulative. For example, if you have 8 existing DtermIP Seats and need a total of 16, add another 8-seat license for a total of 16 seats. |
| SP-30-4 Seat License | Each SP-30 4 Seat License can support up to four simultaneous sessions via the SP-30 Soft-Phone. The SP30 Soft-Phone also requires IP 8 Seat License. To support eight simultaneous soft-phone sessions would require eight Soft-Phone Licenses and 8 IP Seat Licenses. |
| SP-30-4 ACD Seat License | Each SP-30 4 ACD Seat License can support up to four simultaneous sessions via the SP-30 ACD Soft-Phone and can support either Business or ACD mode. The SP30/SP30 ACD Soft-Phone also requires IP Seat License. To support eight simultaneous soft-phone sessions would require eight Soft-Phone Licenses plus 8 IP Seat Licenses. |

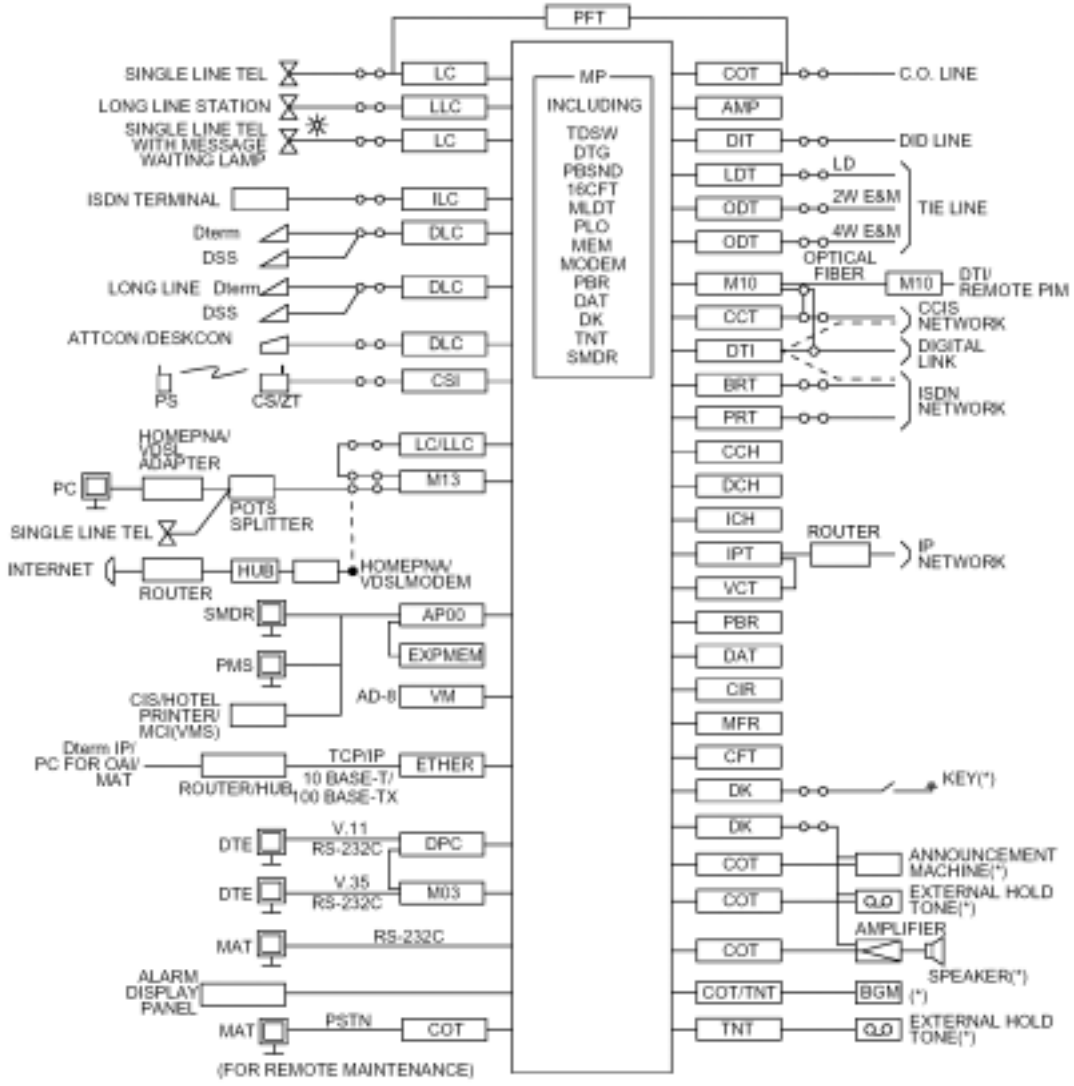
## Technical Terms

| SYMBOL | DESCRIPTION | SYMBOL | DESCRIPTION |
|---|---|---|---|
| AP00 | SMDR/Hotel Application Card | LC | Line Circuit Card (for Single Line Telephone) |
| AUC | Analog Universal Circuit Card (Long Line Circuit, DID Trunk) | MAT | Maintenance Administration Terminal |
| BGM | External Music Source for Dterm Back Ground Music Service | MDF | Main Distribution Frame |
| BRT | Basic Rate Interface Trunk Card | MEM | Main Memory |
| CCH | Common Channel Handler Card | MFR | MF Receiver/ MFC Receiver/ Sender Card |
| CFT | 32 Circuit Party Conference Trunk | MLDT | Melody Trunk |
| CIR | CALLER ID Receiver Trunk Card | MODEM | Modem |
| COT | C.O. Trunk Card | MP | Main Processor Card |
| CSI | CS/ZT Interface Card | PFT | Power Failure Transfer |
| ZT | Zone Transceiver | PMS | Property Management System |
| DAT | Digital Announcement Trunk Card | OAI | Open Application Interface |
| DCH | D-channel Handler Card | ODT | OD Trunk Card (2/4 wire E&M) |
| DID/DIT | DID Trunk Card | PBR | PB Receiver Card |
| DK | External Relay/Key Interface Card | PBSND | PB Sender |
| DLC | Digital Line Circuit Card (for Dterm, DESKCON) | PLO | Phase Locked Oscillator |
| DSS | DSS Console | PS | Personal Station |
| DTI | Digital Trunk Interface Card | PRT | ISDN Primary Rate Interface Trunk Card |
| DTG | Digital Tone Generator | SMDR | Station Message Detail Recording |
| ETHER | Ethernet Control Card | TDSW | Time Division Switch |
| EXPMEM | Memory Expansion Card | TNT | Tone/Music Source Interface Card |
| ICH | ISDN-channel Handler Card | VCT | CODEC Card |
| ILC | ISDN Line Circuit Card | VM | Voice Mail Card |
| IPT | IP Trunk Card | KEY | External Key |

# Trunking Diagram

This figure shows a typical trunking diagram of the *UNIVERGE* NEAX 2000 IPS system.

**NOTE:** *The equipment marked with (*) is provided by the customer.*



# *UNIVERGE* NEAX® 2000 IPS$^{DML}$

The *UNIVERGE* NEAX IPS$^{DML}$ is equipped with all the features and functions of the *UNIVERGE* NEAX 2000 IPS, with a smaller space requirement. It is a full-featured PBX that supports advanced networking, pure peer-to-peer IP telephony connectivity and traditional TDM switching capabilities. Designed primarily for pure converged IP networks, the *UNIVERGE* NEAX IPS$^{DML}$ can also accommodate a mixed (i.e., TDM and IP) converged IP network or standalone solution.

The ***UNIVERGE* NEAX IPS$^{DML}$** (Internet Protocol Server Distributed Model) supports up to 952 peer-to-peer IP stations and 56 TDM ports in a single modular chassis. Up to eight chassis can be stacked providing maximum capacity of 440 TDM ports while still supporting peer-to-peer IP stations. It uses the same SPN-CP31E and the Main processor. The same line/trunk cards, application processor cards and software of the *UNIVERGE* NEAX 2000 IPS and comes equipped for 19" rack mounting. It offers superior port density; each chassis only occupies two Rack Units (2RU).

## Characteristics of the *UNIVERGE* NEAX IPS$^{DML}$

**Compact and small size MODULAR CHASSIS**
One MODULAR CHASSIS provides 8 card slots /56 LT ports and up to 8 MODULAR CHASSIS can be used per system. (8 virtual LT ports are available per MODULAR CHASSIS in addition to 56 LT ports.)

**MP (Main Processor)**
PN-CP31-E for *UNIVERGE* NEAX IPS$^{DML}$

**Ring Generator Unit (RGU)**
Ring Generator Unit (RGU) for the *UNIVERGE* NEAX IPS$^{DML}$ is provided with PZ-PW139 card. The PZ-PW139 card provides ringing supply to analog SLT.

**Power Failure Transfer (PFT)**
Power Failure Transfer (PFT) for the *UNIVERGE* NEAX IPS$^{DML}$ is provided with PZ-4PFTA card. The PZ-8PFTB for the *UNIVERGE* NEAX 2000 IPS is not available for the IPS$^{DML}$.

**Installation Methods**
The *UNIVERGE* NEAX IPS$^{DML}$ can be installed on the desktop or into the 19-inch rack. Wall Mount Installation is not available.

# *UNIVERGE* NEAX® 2000 IPS$^{DMR}$

**The *UNIVERGE* NEAX IPS$^{DMR}$** (Internet Protocol Server Distributed Model Remote) is a *UNIVERGE* NEAX IPS$^{DML}$ that has been optimized for Remote PIM over IP applications. The *UNIVERGE* NEAX IPS$^{DMR}$ uses the SPN-CP31-D as the Main Processor. This system targets users who have up to 30 relatively small offices that accommodate 10-30 extensions at the Remote Site.

The MP card at Remote Site has the same system data as that at Main Site, because Remote Site automatically gets the data from Main Site at the time of setup. In normal operation, Main Site automatically copies the system data to Remote Site through the network once a day.

Because the CP31-D is a cost down CPU, the following options that are built-in on the CP24 are not available with the CP31:

- No built-in DAT.
- Only one RS Port.
- No built-in DK (external/relay key).
- No MN Alarm Indication

## System Outline

- The MP card at Main Site controls system processing, and Remote Site follows the Main Site.
- Remote Site can accommodate most terminals and trunks such as Dterm, Single-Line telephone, PS, DtermIP, COT, ISDN, etc. The Attendant Console, Dterm Attendant position, and Add-on Module are **not** supported at the Remote Site.
- Local Switch (TDSW) at Remote Site controls connections within the Remote Site if possible.
- In the case of connections between Main-Remote and Remote-Remote, the voice path is connected via Peer-to-Peer or IP-PAD.
- If the communications between Main-Remote are interrupted, the Remote Site survives by itself after the system reset.

## Advantages

- The system regards the terminals accommodated in both Main Site and Remote Site as the extensions in the same office. Therefore, the service transparency is superior to CCIS.
- Remote PIM over IP has no limitation of distance between Main and Remote.
- Remote Site has a switching function at local. This provides the effective configuration of C.O. line. In addition, the Remote Site can accommodate AP cards. This is an advantage to accommodate ISDN lines especially.
- The Remote Site survives by itself even if the link between Main and Remote is disconnected. Therefore, the impact to users at Remote Site will be smaller if the link between Main and Remote is disconnected.
- This feature can reduce the bandwidth used on the WAN that is connected to CO lines at Remote Site, rather than DtermIP at remote location or the Media Converter (MC) accommodation.

## Remote PIM over IP

Remote PIM over IP targets users who have 1-30 relatively small offices that accommodate 10-30 extensions at the Remote Site. When *UNIVERGE* NEAX IPS$^{DMR}$ and *UNIVERGE* NEAX 2000 IPS PIM are installed at remote site, and connected to a *UNIVERGE* NEAX 2000 IPS or *UNIVERGE* NEAX IPS$^{DML}$ at main site over IP network, the Main Site system controls and maintains the remote DMR and PIM operation as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

- IPS$^{DMR}$: IPS Distributed Model Remote (with CP31-D)
- IPS PIM: IPS Distributed Model (with CP24-D)

The *UNIVERGE* NEAX IPS$^{DMR}$ is designed primarily for distributed IP networking but also supports traditional analog and digital trunks for connection to the Public Switched Telephone Network (PSTN). The *UNIVERGE* NEAX IPS$^{DMR}$ supports up to 128 peer-to-peer IP stations and 56 TDM ports in a single modular chassis. Up to two chassis can be stacked providing maximum capacity of 112 TDM ports while still supporting as many as 128 peer-to-peer IP stations.

> **Note:** *The MP card at Remote Site has the same system data as the CPU at the Host Site; the Host Site automatically downloads system data to the Remote Site at the time of setup. In normal operation, Main Site automatically downloads a copy the system data to Remote Site through the network once a day.*

Because the CP31-D is designed as a Remote PIM CPU, the following options that are built-in on the CP24 are not available with the CP31:

- No built-in DAT
- Only one RS Port
- No built-in DK (external/relay key)
- No MN Alarm Indication

### Network Conditions and Payload

| Item | Requirement | Remarks |
|---|---|---|
| Protocol | TCP/IP transparent | |
| Maximum Delay Time | 120ms(one way)/240ms(return)<br>150ms(one way)/300ms(return) | Support the quality class A, B of IP Telephone |

**Bandwidth Requirement**

| Established Voice Calls | | With G7.23.1 (5.3k/6.3k) Compression | With G729a ( 8k) Compression | Without Compression (G.711) |
|---|---|---|---|---|
| 6 | Control | 4.1 Kbps | 4.1 Kbps | 4.1 Kbps |
| | Voice | 31.8/37.8 Kbps | 48 Kbps | 432 Kbps |
| 8 | Control | 4.3 Kbps | 4.3 Kbps | 4.3 Kbps |
| | Voice | 42.4/50.4 Kbps | 64 Kbps | 576 Kbps |
| 12 | Control | 4.3 Kbps | 4.3 Kbps | 4.3 Kbps |
| | Voice | 63.6/75.6 Kbps | 96 Kbps | 864 Kbps |
| 16 | Control | 4.5 Kbps | 4.5 Kbps | 4.5 Kbps |
| | Voice | 84.8/100.8 Kbps | 128 Kbps | 1152 Kbps |
| 24 | Control | 4.5 Kbps | 4.5 Kbps | 4.5 Kbps |
| | Voice | 127.2/151.2 Kbps | 192 Kbps | 1728 Kbps |
| 32 | Control | 4.9 Kbps | 4.9 Kbps | 4.9 Kbps |
| | Voice | 169.6/201.6 Kbps | 256 Kbps | 2304 Kbps |
| 48 | Control | 4.9 Kbps | 4.9 Kbps | 4.9 Kbps |
| | Voice | 254.4/302.4 Kbps | 384 Kbps | 3456 Kbps |
| 64 | Control | 5.8 Kbps | 5.8 Kbps | 5.8 Kbps |
| | Voice | 339.2/403.2 Kbps | 512 Kbps | 4608 Kbps |
| 72 | Control | 5.8 Kbps | 5.8 Kbps | 5.8Kbps |
| | Voice | 381.6/453.6 Kbps | 576 Kbps | 5184 Kbps |
| 96 | Control | 6.7 Kbps | 6.7 Kbps | 6.7 Kbps |
| | Voice | 508.8/604.8 Kbps | 768 Kbps | 6912 Kbps |

**Note:** *This information is an estimation based on an established call. Slightly Higher Control values will occur at time of call origination and termination.*

**Base values**

- Originating from a station: 9.6 Kbps/Call (estimated)
- Terminating to a station: 5.76 Kbps /Call (estimated)
- Originating to C.O: 11.5 Kbps/Call (estimated)
- Terminating from C.O: 5.76 Kbps/Call (estimated)
- Keep Alive to Remote Site: 0.032Kbps (estimated)
- Other control packets for Remote Site: 4Kbps (estimated)
- G.723.1 voice: 5.3Kbps (one-way)
- G.729a voice: 8Kbps (one-way)
- G.711 voice: 64Kbps (one-way)

The above base values are primarily used for call setup with the exception of keep alive; 0.032Kbps with no voice traffic. Connections between IP PAD are half duplex, established call utilization is G.711 voice: 64Kbps, G.723.1 voice: 5.3/6.3Kbps, or G729a voice: 8Kbps. Peer-to-Peer IP station calls are full duplex, compression can be specified by location numbers in system data. Peer-to Peer IP station calls even though full duplex will utilize one-way for Bi-directional networks such as T1. Peer-to Peer IP station calls over Asymmetrical networks such as ADSL may realize higher bandwidth utilization, compression can be specified by location numbers in system data.

### Advantages

The system regards the terminals accommodated in both Host Site and Remote Site as the extensions in the same office. Feature transparency is superior to CCIS.

Remote PIM over IP can accommodate AP cards such as ISDN PRI and T1.

This feature can reduce the bandwidth used on the WAN that is connected to CO lines at Remote Site, rather than Dterm IP at remote locations.

Since all Remote PIM over IP sites are treated as extensions in the same office, software and applications only have to be implemented in the host site. This provides centralized use of application for example distributing ACD agents in the DMR locations. CCIS requires each location to have separate software and applications.

CCIS over IP can be combined with Remote PIM over IP to accommodate larger network configurations. Up to 255 host sites can be connected via CCIS, each host site can have up to 30 Remote PIM over IP locations.

### Service Conditions

1. Host site can be *UNIVERGE* NEAX 2000 IPS, *UNIVERGE* NEAX IPS$^{DML}$, or *UNIVERGE* NEAX 2000 Retrofit system. Remote PIM over IP is available in any combination of the following CPUs.
   Main Site: CP24-D, CP27-B, CP31E, CP26-B, CP28-B
   Remote Site: CP31-D, CP24-D, CP26-D

2. Software and Key CD for the whole system must be loaded at the Host Site. No software or key's can be loaded into the Remote Site.

3. All system data changes for the whole system must be performed in the Host Site. No system data changes can be done in the Remote Site.

4. The CPU card at Remote Site has the same system data as the CPU at Main Site; the Host Site automatically downloads its system data to the Remote Site at the time of setup. In normal operation, Host Site automatically copies the system data to Remote Site through the network once a day.

5. Remote Site automatically operates by itself (survival mode) when Keep Alive signal (sent every 30 sec) between the Host Site and Remote Sits is interrupted. When Keep Alive is interrupted the Remote Site is reset to change the operation from normal mode to survival mode.

6. Remote Site in survival mode checks at 30 seconds intervals if the communications to Main Site are possible. When Keep Alive is detected, the Remote Site automatically is reset to change the operation from survival mode to normal mode.

7. When unstable conditions occur in the network, the Remote Site can be manually set to survivable mode (override automatic) until stability in the network is established. This prevents the Remote Site from resetting normal mode to survivable mode etc.

## Required Hardware and Software

### Host Site

| Equipment Name | Remarks |
|---|---|
| PZ-M606-A | On board Ethernet Interface card |
| SPN-8IPLA IP PAD | 8 Port PAD with built-in compression |
| PZ-24IPLA | 24 Port PAD Expansion, mounts on SPN-8IPLA |
| R-PIM 1 Site License | 1 required for each Remote site |

**Note:** *Registration of Host CPU and software required*

### DMR Site

| Equipment Name | Remarks |
|---|---|
| PZ-M606-A | On board Ethernet Interface card |
| SPN-8IPLA IP PAD | 8 Port PAD with built-in compression |
| PZ-24IPLA | 24 Port PAD Expansion, mounts on SPN-8IPLA |

**Note:** *Registration "not" required*

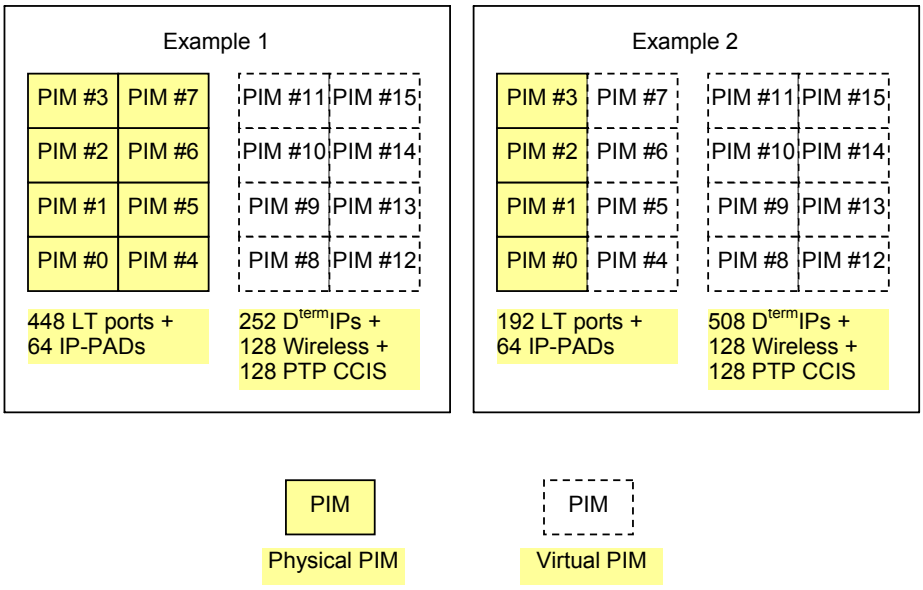
# Chapter 2 System Configuration

## Module Configuration

The *UNIVERGE* NEAX® 2000 IPS consists of single or multiple Port Interface Modules (PIM) depending on the system configuration, and there are two types of PIMs; "Physical" PIM and "Virtual" PIM. The Physical PIM is "hardware" PIM which is used to accommodate an MP, FPs, IP PADs, legacy LT cards, AP cards, and power supply units. One Physical PIM provides up to 64 LT ports and up to 8 Physical PIMs can be accommodated in a Stand Alone system. The Virtual PIM is a "software" PIM and provides up to 64 ports per PIM for use by system programming as DtermIP telephones, Wireless PS stations or Peer to Peer (PTP) CCIS trunks. The system consists of up to 16 PIMs, by the combination of Physical PIMs and Virtual PIMs, thus providing 1020 ports. When the use of Virtual PIMs exceeds 8 then the number Physical PIMs is reduced by one for each additional Virtual PIM required.

The illustration below shows examples of 1020-port configuration by the combination of TDM LT ports, DtermIP telephones, Wireless PS stations and Peer to Peer (PTP) CCIS trunks.

**Figure 2-1 System Configuration with DtermIP (1020-Port Configuration)**



The figure 2-2 shows another example of 1020-port configuration by combination of legacy LT ports and DtermIP telephones.

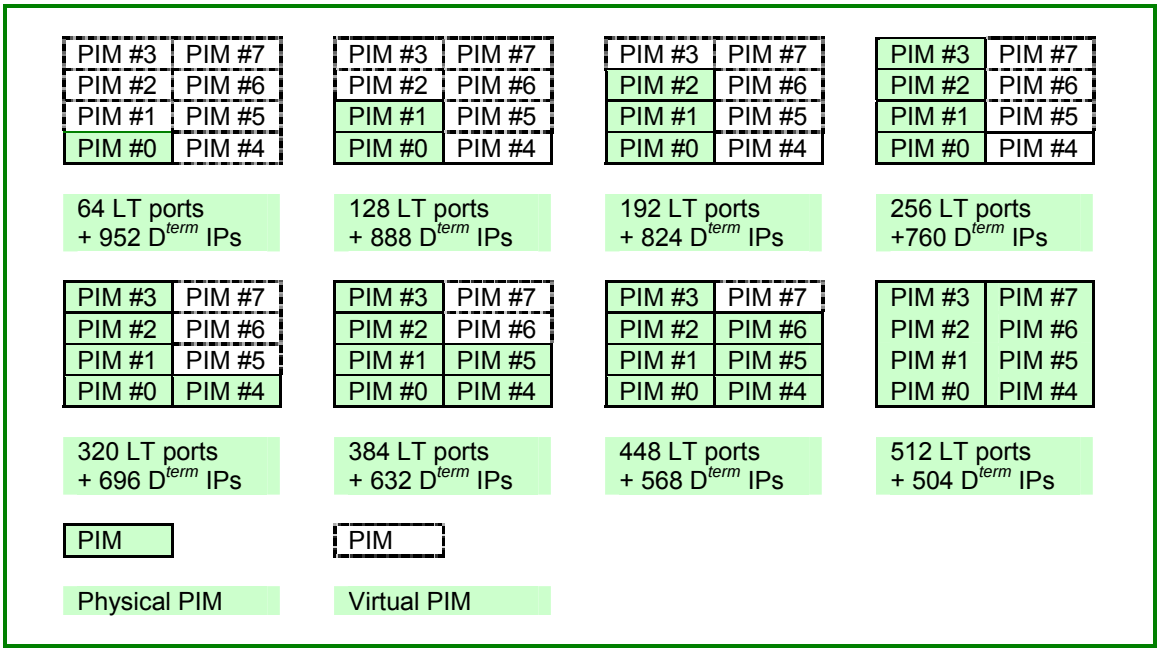



**Figure 2-2 System Configuration with DtermIP (1020-Port Configuration)**

# Installation Methods

The *UNIVERGE* NEAX 2000 IPS provides three installation methods as follows:

- Floor Standing Installation
- Wall Mounting Installation
- 19-inch Rack Mounting Installation

## Floor Standing Installation

In Floor Standing Installation, the *UNIVERGE* NEAX 2000 IPS is comprised of up to 8 Port Interface Modules (PIMs).

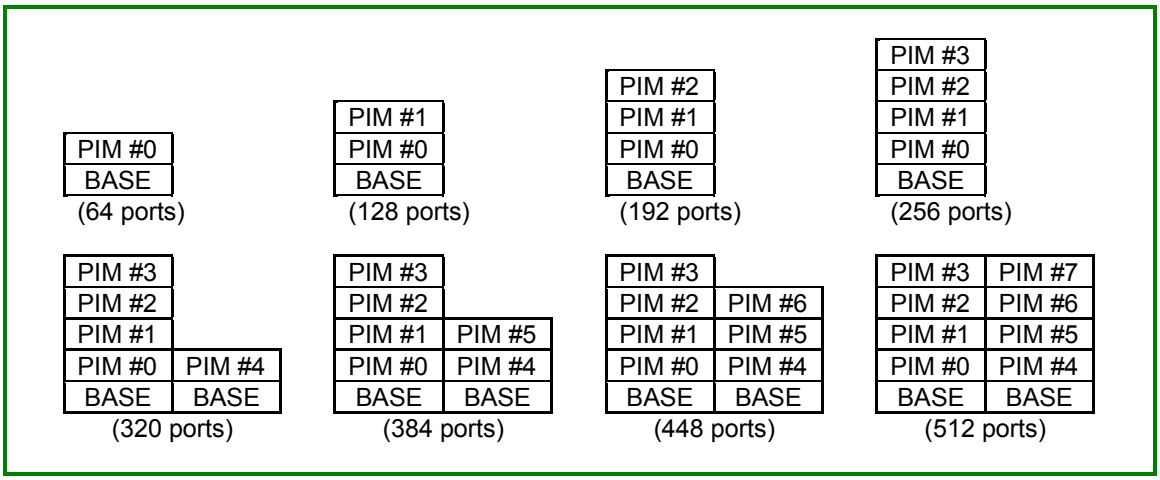



**Figure 2-2 System Configuration in Floor-standing Installation**

## Wall-mounting Installation

The *UNIVERGE* NEAX 2000 IPS can be wall-mounted with single or multiple PIM configurations (maximum of eight PIMs).

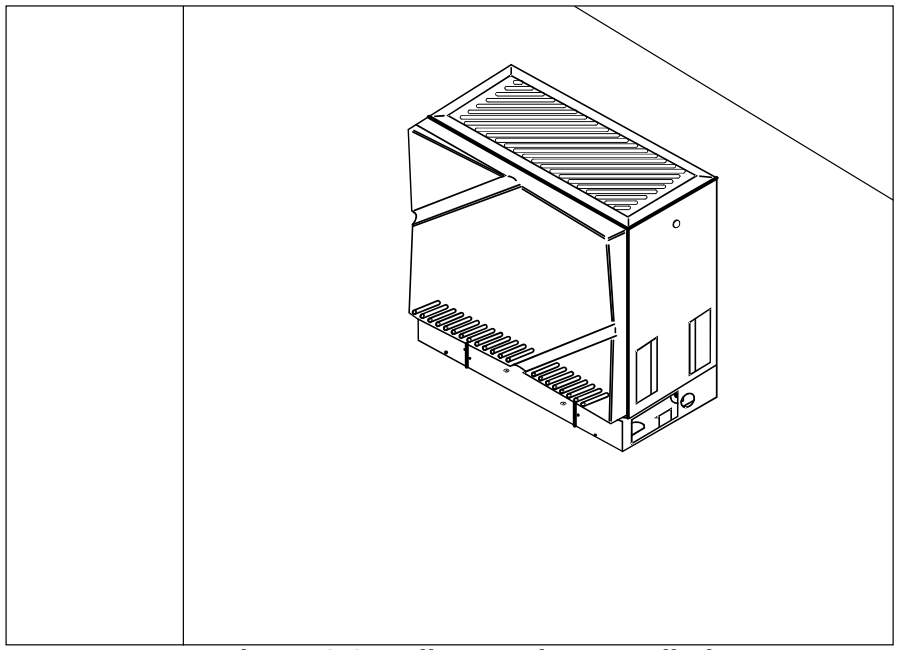


**Figure 2-3 Wall-mounting Installation**

## 19 inch Rack-mounting Installation

The *UNIVERGE* NEAX 2000 IPS can be mounted in the IEC-standard 19 inch rack up to four PIMs. (IEC: International Electro-technical Commission)

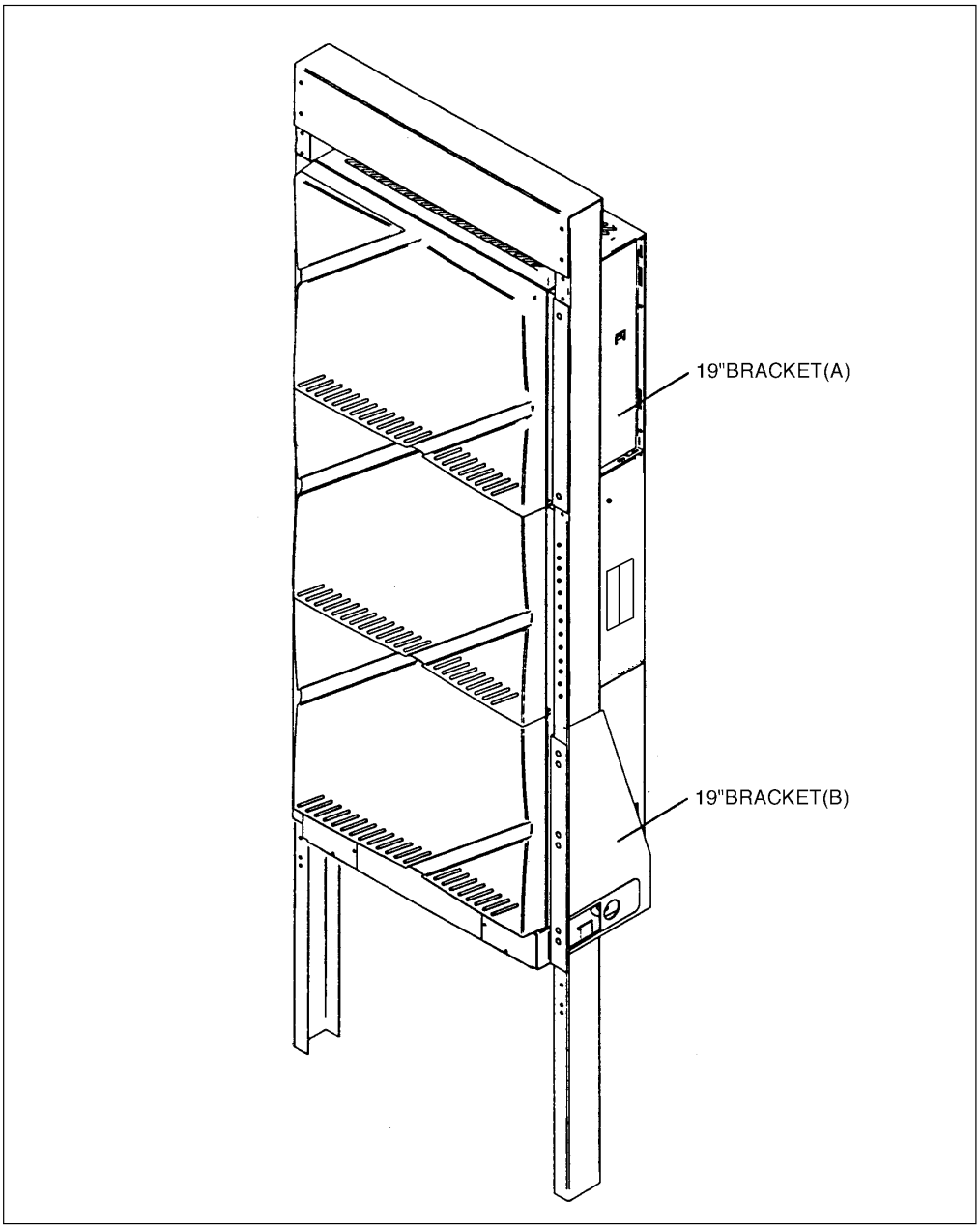



**Figure 2-4 19-inch Rack-mounting Installation**

# Modules and Installation Hardware

The *UNIVERGE* NEAX 2000 IPS is comprised of up to 8 Port Interface Modules (PIMs).

## Modules

### (1) Port Interface Module (PIM)

A PIM provides 13 card slots for common control, Line/Trunk (LT), and Application Processor (AP) cards. It also houses an AC/DC Power Supply, DC/DC Power Supply (for -48V), and batteries for protection from short-term (about 30 min.) power interruption. Four champ connectors for Line/Trunk (LTC 0 to 3) are located at the lower front side of the PIM. A PIM provides a maximum of 12 card slots for Line/Trunk (LT) and Application Processor (AP) cards. At maximum configuration, the system is comprised of 8 PIMs.

There are two types of PIM (PIMMJ and PIMMK) depending on the system type as follows.

| Type of PIM | Single MP System | Dual MP System |
|---|---|---|
| PIM MJ | Used for PIM 0-7 | Used for PIM 1-7 |
| PIM MK | Not used | Used for PIM 0 |

| | |
|---|---|
| PIM MJ (PIM3) | PIM MJ (PIM7) |
| PIM MJ (PIM2) | PIM MJ (PIM6) |
| PIM MJ (PIM1) | PIM MJ (PIM5) |
| **PIM MJ (PIM0)** | PIM MJ (PIM4) |
| **(Single MP System)** | |

| | |
|---|---|
| PIM MJ (PIM3) | PIM MJ (PIM7) |
| PIM MJ (PIM2) | PIM MJ (PIM6) |
| PIM MJ (PIM1) | PIM MJ (PIM5) |
| **PIM MK (PIM0)** | PIM MJ (PIM4) |
| **(Dual MP System)** | |

**Unit Configuration**

### (2) Battery Module (BATTMH)

The BATTMH is an optional module for installing optional long-term (about 3 hours) backup batteries. The BATTMH is designed to accommodate batteries covering up to a 4-PIM system (2 BATTMHs support maximum system configuration). The BATTMH is available for Floor Standing Installation. (When the system is Wall-mounting/19 inch Rack-mounting configuration, the BATTMH cannot be installed with the PIM.)

**Modules**

| Abbrev | Description | Remarks |
|---|---|---|
| PIMMJ | SN1729 PIMMJ | Single MP System: PIM 0 - PIM 7<br>Dual MP System: PIM 1 – PIM 7 |
| PIMMK | SN1730 PIMMK | Single MP System: Not used<br>Dual MP System: PIM 0 |
| BATTMH | SN1731 BATTMH | 1 per STACK, Max.2 per system |

## Installation Hardware

### Base/Top Assembly

The Base/Top Assembly includes a Base Unit and a Top Cover for the PIM. One Base/Top Assembly is required for each PIM stack. The Base Unit also serves as the AC power distribution panel for up to a four PIM configuration.

### Hanger Assembly

The Hanger Assembly is used for Wall-mounting Installation. One set of Hanger Assembly is required for each PIM.

### 19 inch Bracket

The 19-inch Bracket is a set of hardware used for 19-inch Rack-mounting Installation. The 19-INCH RACK BRACKET (A) is installed on both sides of the PIM. One set of 19 inch Bracket (A) is required for each PIM. The 19-INCH RACK BRACKET (B) is installed at the BASE of stack. One 19-INCH BRACKET (B) is required for each stack.

If the system is 2 PIM or more configurations with 19-INCH BRACKET (B), one set of 19-INCH BRACKET (A) is also required for the topmost PIM.

### Optional Brackets

The Mounting Bracket is used for Floor Standing Installation. Without Mounting Bracket, 1.1G shockproof is provided for 1 to 3-module stack and 0.5G shockproof is provided for 4 or more module stack. To enhance the shockproof capability to 1.1G, one set of Mounting Bracket is required for each 4 or more module stack and attached to the topmost PIM.

The I/F Bracket is used for Floor Standing Installation to joint the neighboring topmost PIM in 6 PIM or more configurations. One set of I/F Bracket is required for multiple stacks.

The Base Tray Assembly is used for Floor Standing Installation for stationary equipment (UL complied). One set of Base Tray Assembly is required for each stack.

**Installation Hardware**

| Abbrev | Description | Quantity |
|---|---|---|
| Top Cover | TOP COVER ASSEM | 1/STACK<br>(BASE ASSEM is local supply) |
| Base/Top ASSEM | SN1545 BASERE | 1/STACK |
| Hanger Assem | HANGER ASSEM (UL) | 1/PIM (Wall-mounting Installation) |
| 19 inch Bracket | 19 INCH RACK BRACKET (A) | 1/PIM<br>(19 inch Rack-mounting Installation) |
| | 19 INCH RACK BRACKET (B) | 1/STACK<br>(19 inch Rack-mounting Installation) |
| Mounting Bracket | MOUNTING BRACKET | OPTION (1/STACK) |
| I/F Bracket | I/F BRACKET ASSEM | OPTION (1/SYSTEM) |
| Base Tray | BASE TRAY ASSEM | OPTION (1/STACK) |

## *UNIVERGE* NEAX 2000 IPS SYSTEM POWER SUPPLY

### AC/DC Power Supply

The AC/DC Power Card is mounted in the left side of each PIM. The AC/DC Power card provides power to all circuit cards, which reside in the PIM. AC power requirements are as follows:
Input Voltage: 90 to 132 Vrms or 180 to 264 Vrms (selectable by switch) 50/60 Hz

### DC/DC Power Unit

The DC/DC Power Unit is mounted under the AC/DC Power Card and generates -48 V power for the circuit cards that need such power.

### Battery Backup

#### Internal Short-term option

For customers requiring battery backup, short-term and/or long-term options are available. Two 3.4AH batteries are required per PIM, and installed inside of each PIM. Backup time is approx. 30 minutes when PHS (Wireless PS) is not accommodated and approx. 10 minutes when PHS (Wireless PS) is accommodated in the system.

#### External Long-term option

Two 24AH batteries are required per each 2 PIMs, and installed inside of Battery Module in a stack basis. Backup time is approx. 3 hours when PHS (Wireless PS) is not accommodated and approx. 2 hours when PHS (Wireless PS) is accommodated in the system. The batteries are varied depending on the requested backup time. The battery shall be locally provided.

## Circuit Cards

The circuit cards used for *UNIVERGE* NEAX 2000 IPS is divided into the following three types. According to these card types, the mounting locations of card and port allocation of the Time Division Switch are varied.

- Common Control Cards
  - Main Processor (MP)
  - Firmware Processor (FP)
  - Ethernet
  - Power

- Line/Trunk (LT) Cards
  - IP PAD, Line Circuit (LC), Central Office Trunk (COT), Tie Line Trunk (LDT/ODT), etc.

- Application Processor (AP) Cards
  - SMDR/PMS/CIS/Hotel Printer Interface (AP00)
  - T1/E1 Digital Trunk Interface (DTI)

# IPS System Conditions

The *UNIVERGE* NEAX 2000 IPS is an IP communication system that integrates voice terminals through Peer-to-Peer connection to the IP network. The system is a hybrid system to accommodate both IP multiline terminals (DtermIP) and the Legacy PBX's terminals (Legacy terminal). Line/Trunk cards and Application Processor cards can be mounted in the system to provide the Legacy PBX features that use the Time Division Switch (TDSW).

### Station-To-Station Connection

Station-to-Station connection is available on the LAN. For DtermIP-to-DtermIP connection (Peer-to-Peer connection), the voice data is transmitted and received directly between DtermIPs on the LAN. For DtermIP-to-Legacy terminal connection, the IP-PAD card is required to transmit and receive the voice data. This card is used to control and convert the voice data. The MP card manages control signals in both types of connections.

### Public Network/TIE Network Connection

The system can be connected with a Public Network or Tie Line Network. When the DtermIP communicates with the DtermIP/Legacy terminal in the destination office via Public Network or Tie Line Network, the IP-PAD card and the trunk card are required to transmit and receive the voice data.

### CCIS Connection

The system can be connected with the IP network by No. 7 Common Channel Inter-office Signaling (CCIS) via the Virtual IPT, when the destination office is 2000 IPS or 2400 IPX. For DtermIP-to-DtermIP connection via CCIS (Peer-to-Peer connection), the voice data is transmitted and received directly between DtermIPs via the IP network (CCIS via IP). For DtermIP-to-Legacy terminal connection via CCIS, the IP-PAD card is required to transmit and receive the voice data. This card is used to control and convert the voice data. The MP card has a built-in Virtual IPT and the Virtual IPT manages control signals in both types of connections.

### H.323 Connection

The system can be connected with the IP network by ITU-T recommendation H.323 protocol. The system can be connected to the terminal and network equipment according to H.323 protocol. For DtermIP-to-DtermIP connection via the IP network with H.323 protocol, the IPT card and IP-PAD card are required to transmit and receive the control signal and voice data. For voice compression, the 4VCT card is required. For Legacy terminal connection via the IP network with H.323 protocol, the IPT card is required. For voice compression, the 4VCT card is required.

### Overall System Conditions

- To connect the MP (PN-CP24-D, PN-CP27-B, and PN-CP31-D) card to the LAN, ETHER (PZ-M606-A) card is required on the MP card. PZ-M697 (ETHER) card is mounted on the PN-CP31-E (MP) card.
- One Virtual FP/AP card provides 64 ports to connect the Line/Trunk cards.
- The DTMF sender signal width of Dterm/DtermIP is 112-128 ms.

### Dterm IP

- For the DtermIP, an AC-DC adapter or inline power patch panel is required.
- The hold tone for DtermIP is only “Minuet”. The hold tone set by CM48 Y=3 are not effective for DtermIP.

### Peer-to-Peer Connection

- For the communication between DtermIPs, the voice data is transmitted and received directly, without converting voice packets into PCM and voice compression in the system.

### DRS=Device Registration Server

- The System-based DRS executes DtermIP registration.
- The Network-based DRS is not available for the DtermIP registration.

### Public Network/TIE Line Network Connection

- For the DtermIP communication between offices, the IP-PAD card is required.
- Peer-to-Peer connection is not available in this connection.

### H.323 Connection

- When connecting to the IP network with H.323 protocol, the IPT card and 4VCT card are required.
- When connecting DtermIP to the IP network with H.323 protocol, the IP-PAD card is required.
- Peer-to-Peer connection between offices is not available.
- Connection via the Intranet is only supported.
- For Voice over IP (H.323), a H.323 Gatekeeper is required for converting between IP address and station number.
- The IP trunk provides only Point-to-Point connection.

### CCIS Connection

- Peer-to-Peer connection between DtermIPs via CCIS is available only when the destination office is 2000 IPS or 2400 IPX.
- The Virtual IPT can be connected to a maximum of 127 trunks.
- The Virtual IPT provides only Point-to-Multipoint connection.
- When a call over Peer-to-Peer connection via CCIS is put on hold and then answered at the same station, Elapsed Time Display returns to 0:00:00.
- When the destination office uses the physical IPT card, for example, when connecting to the former PBX system, the IPT card and 4VCT card are required in both offices.
- Conditions for Link Down Notice for CCIS connection are shown below.
  - Link Down Notice is available only for Dterm and DtermIP accommodated in the 2000 IPS and IPS$^{DML}$/IPS$^{DMR}$. This is not available for a single line telephone and Attendant Console.
  - For message display, Dterm/DtermIP with 24-digit or more LCD is recommended. 16-digit LCD may not display all messages properly.
  - Notification message can be displayed regardless of idle or busy state of Dterm/DtermIP, writing the message over the present display. After six seconds, the display returns to the time display automatically.
  - The system detects a Link Down on the condition that TCP connection between offices is interrupted. The Link Down is notified to the Dterm/DtermIP at 15-20 seconds later from the system detects the Link Down.
  - Link Down Notice is available only for the CCIS connection via Virtual IPT. CCIS connection with CCT/DTI card or LDT/ODT card is not available.
  - When the link between offices connected by CCIS via Virtual IPT is interrupted, the lamp of Dterm/DtermIP button becomes the state as shown below. Then press the button, the LCD of the Dterm/DtermIP displays the following.

| COLOR AND STATE OF BUTTON | | STATE AND OPERATION | LCD DISPLAY |
|---|---|---|---|
| Red/Flashing (Momentarily) | 0.125 seconds ON-<br>0.125 seconds OFF | Link Down occurrence | - |
| Red/Flashing (Slowly) | 0.5 seconds ON-<br>0.5 seconds OFF | Press the button after Link Down occurrence | Link Down to CCIS |
| OFF | - | Link restoration | - |
| OFF | - | Press the button after Link restoration | Normal Condition: CCIS |

  - When the link between offices recovers, the flashing lamp of the button goes out.

### CCIS and H.323 Connection

- The service features requiring continuous voice transmission such as Background Music should not be used because the traffic may reduce overall performance of the LAN.
- In the voice communication via the Internet, the connection and communication delay may occur and the voice quality may deteriorate.
- The Virtual IPT and IPT card does not support Dynamic Host Configuration Protocol (DHCP) service.

### IP-PAD

- The PN-8IPLA IP-PAD is required for the following connections.
  - DtermIP-to-Legacy terminal connection
  - DtermIP-to-Public Network/Tie Line Network connection
  - External hold tone connection
  - Conference Trunk (CFT) connection
  - Digital Announcement Trunk (DAT) connection

- The PZ-24IPLA (24DSP) card is mounted on the PN-8IPLA (IP-PAD) card when more than 8 IP PADs (16/24/32) are required.

- FAX Communication with IP PAD cards.
  - PN-8IPLA IP-PAD-C supports G.711 and G.726 Pass-Through FAX communication
  - When providing the G.711 pass-through FAX communication with the PN-32IPLA-A card, following condition is required.

    PN-32IPLA IP PAD-E is mounted.

    OR

    PN-32IPLA IP PAD-D and PN-16VCTAA IP PAD-B are mounted.

    (When mounting two 16VCT cards, both should be PN-16VCTAA IP PAD-B.)
  - When providing the G.726 pass-through FAX communication with the PN-32IPLA-A card, following condition is required.

    PN-32IPLA IP PAD-E and the PN-16VCTAA IP PAD-B are mounted.

    (When mounting two 16VCT cards, both should be PN-16VCTAA IP PAD-B.)
  - 8IPLA does not support T.30 FAX. (T.30 FAX requires the 32IPLA-A)
  - When providing the T.30 FAX communication the PN-32IPLA-A IP PAD-D/E card and the PN-16VCTAA IP PAD-B are required.

# *UNIVERGE* NEAX® IPS$^{DML}$ System Configuration

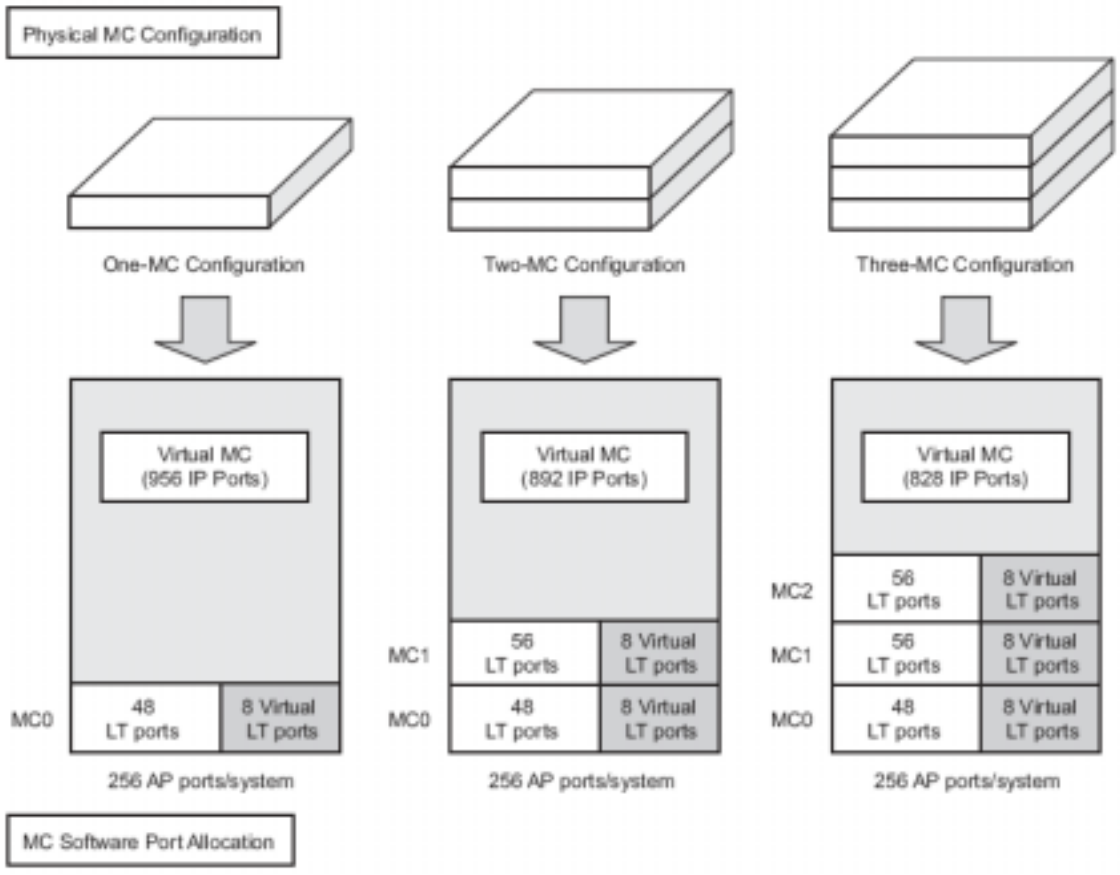
The ***UNIVERGE*** **NEAX IPS$^{DML}$** consists of from one to eight MODULAR CHASSIS depending on the system configuration. There are 2 types of MODULAR CHASSIS; "Physical Modular Chassis" and "Virtual Modular Chassis ".

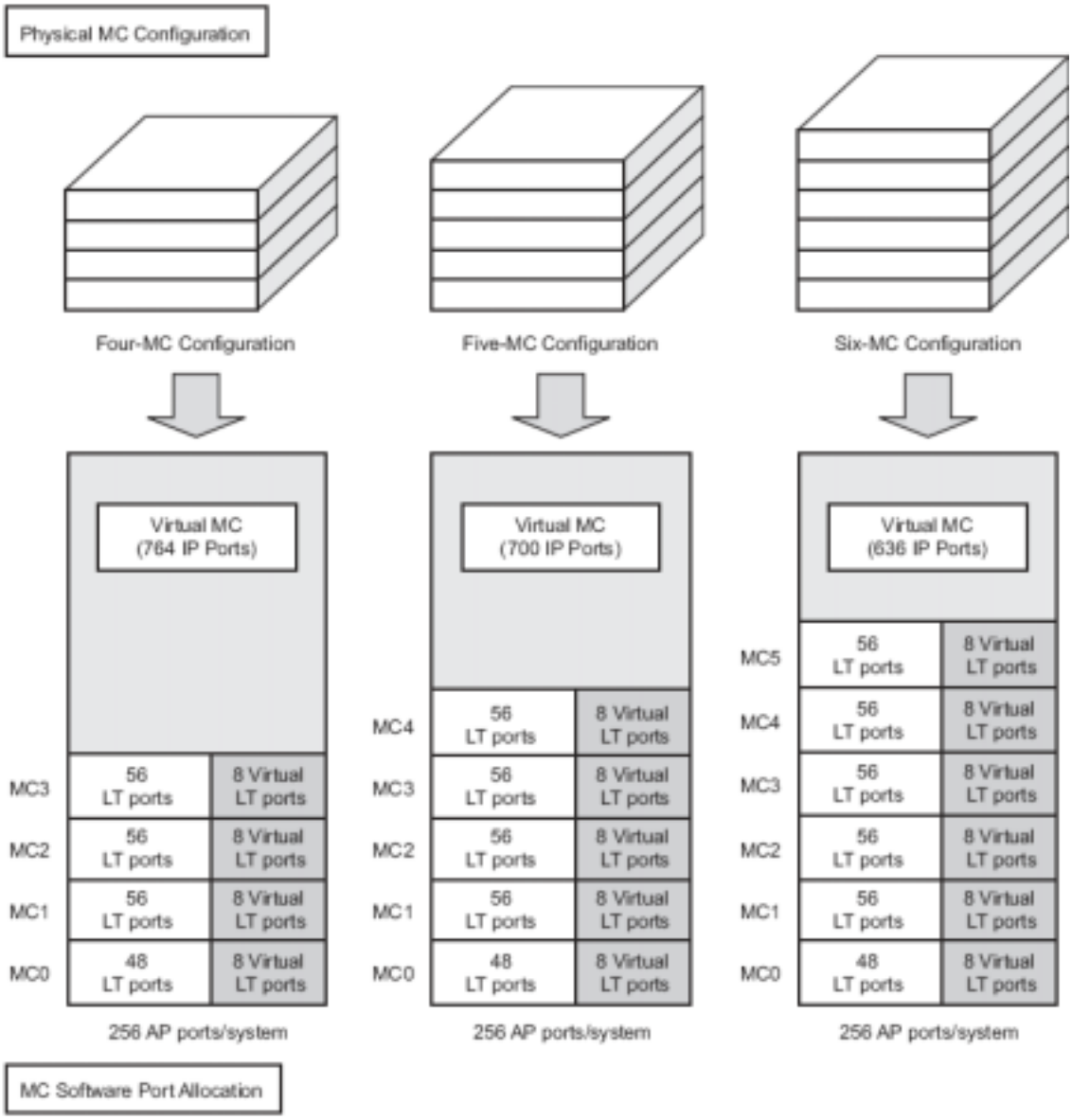
The Physical Modular Chassis is “hardware Modular Chassis” and is used to accommodate an MP, FPs, IP PADs, legacy LT/AP cards, and power supply units. The Physical Modular Chassis provides 56 LT ports in hardware slots and provides 8 ports in software port allocation (56 LT ports and 8 virtual ports). One Modular Chassis provides 8 card slots including one card slot for Main Processor (MP) and other 7 slots for Line Trunk (LT)/Application Processor (AP) cards; 56 LT ports and 8 virtual LT ports; AC, LTC, BUS cable connectors and power switch which are located at the rear side of Modular Chassis. The Physical Modular Chassis can be installed on the desktop or into the 19-inch rack only.

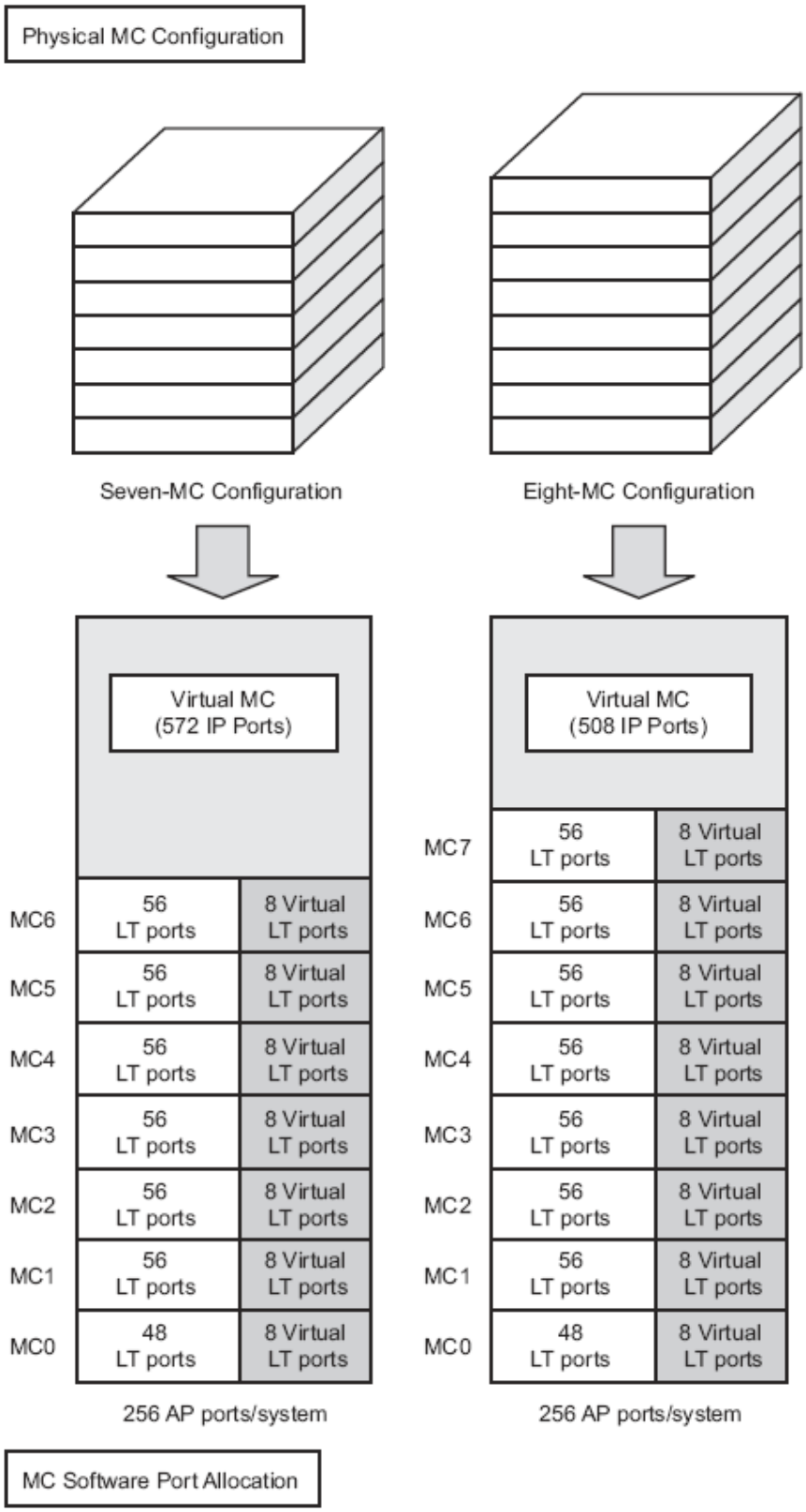
The Virtual Modular Chassis is a “software Modular Chassis” with a port capacity of 64 ports. The Virtual MC is a “software” MC and is used to accommodate IP telephones by system data programming. The port capacity of the Virtual MC is different depends on the number of Physical MC.

The port allocation varies with the version of system.

**Software Port Allocation for Each Physical MC Configuration**




Physical MC Configuration
Four-MC Configuration
Five-MC Configuration
Six-MC Configuration
Virtual MC
(764 IP Ports)
Virtual MC
(700 IP Ports)
Virtual MC
(636 IP Ports)
MC5
MC4
MC3
MC2
MC1
MC0
56
LT ports
8 Virtual
LT ports
48
LT ports
256 AP ports/system
MC Software Port Allocation


Physical MC Configuration
Seven-MC Configuration
Eight-MC Configuration
Virtual MC
(572 IP Ports)
Virtual MC
(508 IP Ports)
MC7
MC6
MC5
MC4
MC3
MC2
MC1
MC0
56 LT ports
8 Virtual LT ports
48 LT ports
256 AP ports/system
256 AP ports/system
MC Software Port Allocation

# *UNIVERGE* NEAX® IPS$^{DMR}$ System Configuration

The ***UNIVERGE*** **NEAX IPS$^{DMR}$** consists of from one to two MODULAR CHASSIS depending on the system configuration. There are 2 types of MODULAR CHASSIS; "Physical Modular Chassis" and "Virtual Modular Chassis ".

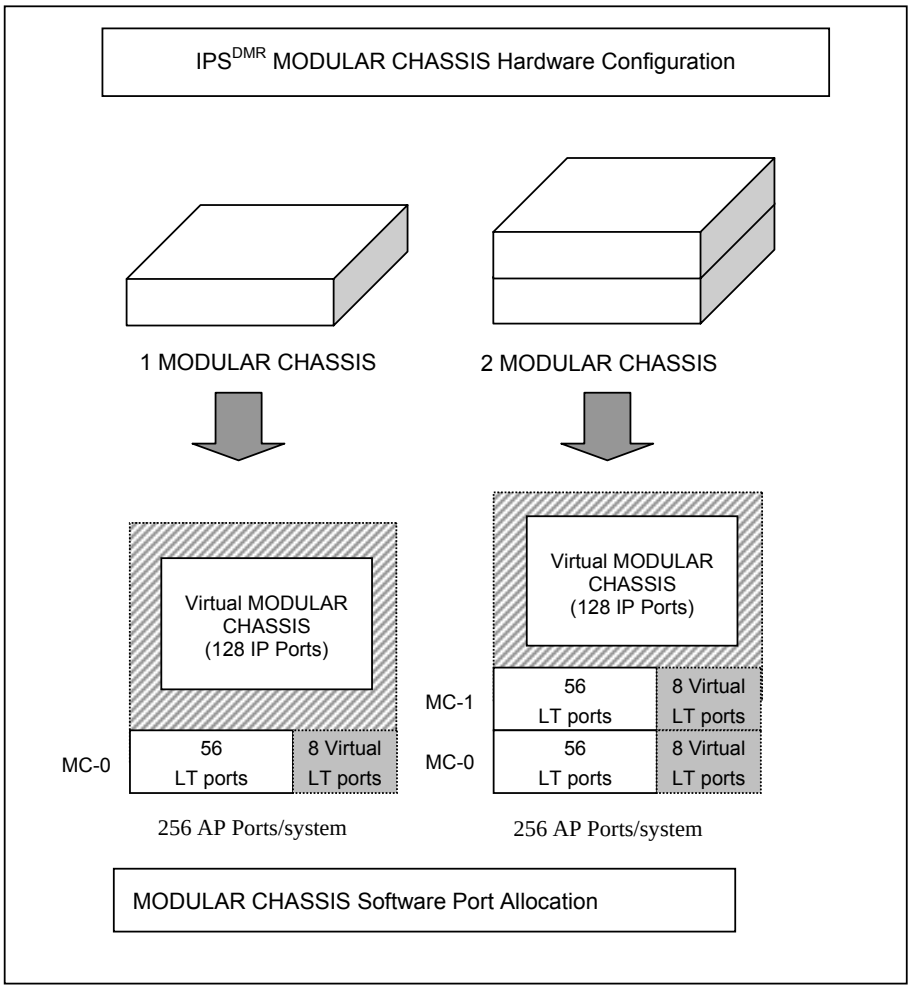
The Physical Modular Chassis is “hardware Modular Chassis” and is used to accommodate an MP, FPs, IP PADs, legacy LT/AP cards, and power supply units. The Physical Modular Chassis provides 56 LT ports in hardware slots and provides 8 ports in software port allocation (56 LT ports and 8 virtual ports). One Modular Chassis provides 8 card slots including one card slot for Main Processor (MP) and other 7 slots for Line Trunk (LT)/Application Processor (AP) cards; 56 LT ports and 8 virtual LT ports; AC, LTC, BUS cable connectors and power switch which are located at the rear side of Modular Chassis. The Physical Modular Chassis can be installed on the desktop or into the 19-inch rack only.

The Virtual Modular Chassis is a “software Modular Chassis” with a port capacity of 64 ports. A maximum of two Virtual Modular Chassis can be assigned per IPS$^{DMR}$ for a total of 128 ports used to accommodate IP stations by system data programming.

**The *UNIVERGE* NEAX IPS$^{DMR}$** locates the maximum of 64 FP/AP cards per system, at multiple Remote Sites. The number of FP/AP cards accommodated at one Remote Site should be a maximum of eight including the MP built-in FP. If more than eight FP/APs are assigned, the system does not operate normally. The maximum number of Remote Sites is 30. The MP card at Remote Site has the same system data as that at Main Site, because Remote Site automatically gets the data from Main Site at the time of setup. In normal operation, Main Site automatically copies the system data to Remote Site through the network once a day. Because the CP31-D is a cost down CPU, the following options that are built-in on the CP24 are not available with the CP31:

- No built-in DAT.
- Only one RS Port.
- No built-in DK (external/relay key).
- No MN Alarm Indication

The following illustration shows Modular Chassis hardware configurations, software port allocation, face layout and rear view of Modular Chassis for The *UNIVERGE* NEAX IPS$^{DMR}$.



# Remote PIM over IP System Outline

When IPS$^{DMR}$ and 2000 IPS PIM are installed at Remote Site, and connected to a Main Site over IP network, the Main Site system controls and maintains the operation of Remote Sites as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

IPS$^{DMR}$: IPS Distributed Model Remote (with PN-CP31-D)

IPS PIM: IPS (with PN-CP24-D)

### Outline of System Operation

- The MP card of Main Site controls call processing, and Remote Site follows the Main Site in normal operation mode.
- Remote Site can accommodate analog Single Line telephones, Dterm, PS, DtermIP, and LT/AP cards.
- Local Switch (TDSW) at Remote Site connects an outside party when the Remote Site is directly connected to the PSTN/GSTN.
- In the case of connections between Main Site and Remote Site, or Remote Site and Remote Site, the voice path is connected via Peer-to-Peer or IP-PAD.
- If the communications between Main Site and Remote Site are interrupted, the Remote Site starts a survival mode operation after the system reset.

### Outline of Survival Mode Operation

- Remote Site system watches a Keep Alive signal sent from Main Site regularly.
- If a line failure occurs (Keep Alive signal is not received), Remote Site resets the own system and starts survival mode operation as a stand-alone system to control the call processing within the Remote Site.
- During survival mode operation, Remote Site system checks regularly whether the communications with Main Site is possible or not. When the Remote Site regards that the communications are possible, the Remote Site will change over to the normal mode to communicate with the Main Site automatically or manually.

### Outline of Survival Mode Operation

- This feature displays the link state between Main Site and Remote Site on the designated Dterm or DtermIP at both sites, and allows users at both sites to notice the link failure.

# Remote PIM over IP System Conditions

## General Conditions

- The same version of software must be installed in the MP cards of Main and Remote Sites.
- The way of loading and conditions of Key CD are following.
  - The same number of Key CD for remote site license as the number of remote sites is required for Remote PIM over IP system. The required Key CD for the whole system must be loaded to the Main Site.
  - Total number of terminals that can be accommodated in the Remote Site depends on the number of port and number of license allocated from total Key CD data of Main Site. The total Key CD data is loaded to Main Site and divided between Main Site and Remote Sites as following examples.

    Example 1: When 256-port Key CD is loaded to the Main Site and 64 ports are used in the Remote Site. Remote Site: 64 ports are allocated from the Main Site. Main Site: 192 ports are available
    Example 2: When 128-IP license Key CD is loaded to the Main Site. The license is allocated in order of connecting IP telephone regardless of Main Site or Remote Site.

  - After Key CD data is loaded to the Main Site, the service feature, number of port, and number of license to be used by each Remote Site are sent to the flash memory of each Remote Site's MP card automatically. When Remote Sites operate with survival mode, this data is used.
  - Data to be used by each Remote Site is stored in the flash memory every 10 minutes. When a Remote Site starts operating with survival mode within 10 minutes from starting up, the system does not operate normally because the data is not stored in flash memory. The system should be observed for more than 10 minutes after starting up.
  - The number of legacy terminal that depends on the number of PIM in TDSW system depends on the number of port in Remote PIM over IP system. For Key CD data, one PIM license for TDSW system is converted to 64 ports for Remote PIM over IP system.
- The number of accommodated terminals/trunks in Main Site and Remote Site should be a maximum of 1020 ports in the whole system.
- The TCP/IP network is required between Main Site and Remote Site. The closed and bandwidth guaranteed network is preferable, such as IP-VPN (Layer 3 VPN) or wide area Ethernet service (Layer 2 VPN). The following table shows the permissible delay time in the network.

| Recommended/ Maximum Value | Permissible delay time | |
|---|---|---|
| | One-way | Return-way |
| Recommended | 100 ms. | 200 ms. |
| Maximum Value | 120 ms. | 240 ms. |

If the network is short of the requirement, it may cause the delay operation of system, the delay and deterioration of voice packets, disconnection of calls, and frequent changeover to survival mode at Remote Site.

- This feature is available in the Retrofit System (used as Main Site). The system that using the following MP cards can be mixed used in Remote PIM over IP system in any combination.

  Main Site: PN-CP24-D, PN-CP27-B, PN-CP31E, PN-CP26-B, PN-CP28-B

  Remote Site: PN-CP31-D, PN-CP24-D, PN-CP26-B

  ***NOTE:*** *PN-CP27-B/PN-CP28-B cannot be used as Backup CPU system in Remote Site.*

- This feature is not compatible with Fusion service of 2400 IPX.

## Conditions for System Configuration

- The number of Remote Sites is a maximum of 30.
- The total number of FP/AP at Main Site and all Remote Sites should not exceed 64 including the MP built-in FP, Virtual FP, and Virtual IPT.
- The number of FP/AP accommodated at one Remote Site should be a maximum of eight including the MP built-in FP.
- Remote Site can accommodate the following FP/AP/LT cards.
  - FP: One MP built-in FP and two Virtual FP
  - AP: BRT, 24DTI, 24PRT (ISDN-PRI)
  - LT: 8IPLA, DLC, LC, COT, ODT

  ***NOTE:*** *Attendant Console, Desk Console, Add-On Module, are not mountable at Remote Site.*
- Remote Site cannot accommodate the following FP/AP/LT cards.
  - FP: FP card (CP15) is not mountable at Remote Site.
  - AP: AP00 (SMDR), AP00 (DBM), ICH, CCH, DCH (Q-SIG/Q931a), 4RSTB (MFC/T1-ANI/E911), CFTC (32-party conference), IPT
  - LT: 8RSTG (PBR), ILC (ISDN Terminal), 4RSTF (Caller ID station), PN-8LCAD (LLC), PN-4LLCB (LLC), PN-8PFTB (PFT), 4VCT, DK00, DAT

  ***NOTE:*** *Four-line built-in PBR on the MP card is available at Remote Site.*
- MP built-in FP must be set to each Remote Site.
- The port for FP card/MP built-in FP must be assigned with every 8 port. If the assignment value cannot be divided by 8, the value that the remainder is omitted is assigned to the FP card/MP built-in FP automatically.
- The available IP-PAD card in the Remote Site is PN-8IPLA.
- For the system capacity of ISDN system or CCIS system, refer to the system capacity of each system manual. Set the whole Remote PIM over IP system within the system capacity and be sure to mount the handler card in the Site where the interface card is installed.
- The number of ports accommodated at one Remote Site should be a maximum of 256. This is the sum of maximum 128 physical ports (one built-in FP), maximum 128 IP ports (two Virtual FP), and AP ports such as ISDN, etc.

### Conditions for Survival Mode at Remote Site

- Remote Site starts survival mode operation in the following cases.
  - When the communications (Keep Alive signal) in every 30 seconds between Main Site and Remote Site are interrupted for the predetermined time set by CM0B Y=31-60>50 on normal mode operation.
  - When Remote Site cannot be connected to Main Site or is not allowed connecting to Main Site after the system reset of the Remote Site.
- Remote Site is reset automatically to change the operation from normal mode to survival mode when it detects an interruption of the communications from/to Main Site.
- When Remote Site starts the survival mode operation, the fault information "Initialize by CAT or MAT" is registered to the MP card of Remote Site. In addition, "Communication error occurrence between Main Site and Remote Site" is registered to the MP card of Main Site at 20 seconds later from the predetermined time set by CM0B Y=31-60>50.
- Remote Site on survival mode checks at every 30 seconds if the communications to Main Site are possible. When the Remote Site regards that the communications are possible, "Communication error restoration between Main Site and Remote Site" is registered to the MP card of Main Site at 20 seconds later from the predetermined time set by CM0B Y=31-60>51.
- When Remote Site on survival mode regards that the communications to Main Site are possible, manual changeover (system reset) is required. Automatic changeover (re-connection to Main Site) is also selectable in system data programming. At the automatic changeover, Remote Site system is initialized and the calls on going are disconnected due to the reset of terminals.

### Service Conditions

- Set the unique location number to each location group for proper setting of the IP-PAD channel selection and CODEC for voice compression.
- The system clock at Remote Site synchronizes with the system clock at Main Site. If the communications to Main Site are interrupted, Remote Site does not synchronize and operates at the hardware clock on the MP card of Remote Site.
- Remote Site cannot accommodate the Virtual IPT and the IPT card (H.323).
- The service features requiring continuous voice transmission such as Background Music and Internal Zone Paging should not be used at Remote Site considering the traffic on the network.
- Multiple line service should not be used among different sites considering the traffic on the network.
- Automated Attendant should not be used at Remote Site considering the traffic.
- Unavailable services at Remote Site are as follows.
  - Attendant Console, Desk Console, Add-On Module, and ISDN Terminal are not mountable.
  - Caller ID - Station is not available.
  - CCIS is not available.
  - All services using MP built-in DK or DK card are not available.
  - MP built-in DAT and Digital Announcement Trunk (DAT) card are not available.
  - RS Port No.1 is not available with CP31-D.

- Set the unique trunk route to each Site.
- Different numbering plan for every site must not be assigned.
- Since Main Site controls OAI, OAI terminals cannot be used at Remote Site during survival mode operation.
- Each Remote Site must provide a hold tone to the station at the Remote Site.
  - External hold tone (using Jack on the MP card): Tone source is required for each site.
  - Internal hold tone: Available
  - Hold tone using DAT: Not available
- The maximum station number which can be used in SMDR or PMS Interface is as follows.
  - MP built-in SMDR/PMS on IP: 1020 stations
  - SMDR/PMS with AP00 on RS-232C: 504 stations
- You can connect MAT to Remote Site via RS-232C or LAN.
- SNMP is not available at Remote Site.
- The IP Remote Site does not have DHCP server and client function.
- The MJ/MN alarm indications are not available at Remote Site. If a fault occurs at Remote Site, the fault is notified to the Main Site and MJ/MN alarm is indicated at the Main Site.

### Conditions for Link Down Notice

- Link Down Notice is available only for Dterm and DtermIP accommodated in Main Site and Remote Site. This is not available for a single line telephone and Attendant Console.
- For message display, Dterm/DtermIP with 24-digit or more LCD is recommended. 16-digit LCD may not display all messages properly.
- Notification message can be displayed regardless of idle or busy state of Dterm/DtermIP, writing the message over the present display. After six seconds, the display returns to the time display automatically.
- The system detects a Link Down on the condition that UDP connection between Main Site and Remote Site is interrupted. The Link Down is notified to the Dterm/DtermIP at 20-50 seconds (time programmable).
- When the link between Main Site and Remote Site is interrupted, the lamp of Dterm/DtermIP button becomes the state as shown below. Then press the button, the LCD of the Dterm/DtermIP displays the following.

| COLOR AND STATE OF BUTTON | | STATE AND OPERATION | LCD DISPLAY |
|---|---|---|---|
| Red/Flashing (Momentarily) | 0.125 seconds ON-<br>0.125 seconds OFF | Link Down occurrence | - |
| Red/Flashing (Slowly) | 0.5 seconds ON-<br>0.5 seconds OFF | Press the button after Link Down occurrence | Link Down to Site xx<br>(xx: Site No.) |
| Green/Flashing (Intermittently) | 0.125 seconds ON-<br>0.125 seconds OFF-<br>0.125 seconds ON-<br>0.625 seconds OFF | Remote Site is survival mode after Link restoration | - |
| Green/Flashing (Intermittently) | 0.125 seconds ON-<br>0.125 seconds OFF-<br>0.125 seconds ON-<br>0.625 seconds OFF | Press the button with Remote Site is survival mode after Link restoration | Link Down to Site xx<br>(xx: Site No.) |
| OFF | - | Remote PIM is normal mode after Link restoration | - |
| OFF | - | Press the button with Remote Site is normal mode after Link restoration | Normal Condition:<br>R-PIM |

- Link restoration is notified to the Dterm/DtermIP at 20-50 seconds. If the time is not set by programming, the Link restoration is notified to the Dterm/DtermIP at 110-150 seconds later from the system detects the Link Ready. After the link is ready, the lamp of Dterm/DtermIP button keeps flashing during the Remote Site operates on survival mode. Since the color of lamp and the indication interval changes, an administrator at the Remote Office can changeover the system operation from survival mode to normal remote mode according to this indication. When the link between Main Site and Remote Site recovers and the Remote Site starts subordinate operation to the Main Site (normal mode), the flashing lamp of the button goes out.

# Chapter 3 Terminals

A variety of terminal equipment may be connected to the *UNIVERGE* NEAX 2000 IPS. The following equipment may be installed with the system.

- SN716 DESK CON (Attendant Console)
- Dterm Series i Analog Terminals
  - Single-Line Analog
  - Hospitality Single-Line Analog
- Dterm Series i (IP) Terminals
  - 4-Line display DtermIP
  - 8-Line display DtermIP
  - 16-Line display DtermIP
  - 16LD display DtermIP
  - 32-Line display DtermIP
- Dterm Series i (TDM) Digital Terminals
  - 2-Line Digital
  - 4-Line display Digital
  - 8-Line non-display Digital
  - 8-Line display Digital
  - 16-Line display Digital
  - 16LD display Digital
  - 32-Line display Digital
  - 60 Console Add-On Module/DSS/BLF
  - 16LD Desi-Less ADM
- Dterm Series E (TDM) Digital Terminals
  - 2-Line Digital
  - 8-Line non-display Digital
  - 8-Line display Digital
  - 16-Line display Digital
  - 32-Line display Digital
  - 60 Console Add-On Module/DSS/BLF
- Dterm Cordless Terminals
  - Dterm Cordless II
  - Dterm Cordless Lite II
  - Dterm Headset Cordless II
  - Plantronics CT Cordless
- PS - Wireless Handset
- Dterm SP30 (IP Soft Phone)
- INASET

# SN716 Desk Con

The SN716 Desk Console has an ergonomic design and provides full access to all PBX Console features. It connects to the *UNIVERGE* NEAX 2000 IPS using the same circuit cards as the Dterm Series i/E terminals. The SN716 Desk Console operates on a switched-loop basis with a maximum of 6 Attendant loops terminating at each console on the associated Interface card. The Attendant uses these loops for answering, originating, holding, extending, and reentering calls. When Attendant loop release is used, the number of loops is effectively increased to a maximum of 12 for each console. The SN716 Desk Console also provides flexible key assignments to meet the operator's needs. Key assignments are semi-fixed by system default data, but may be changed via programming.

The *UNIVERGE* NEAX 2000 IPS supports a maximum of eight SN716 DESK Consoles.



### SN716 DESKCON Features

- Character LCD (4x40 characters)
- LCD designation strips
- Software-controlled LCD loop key
- Full access to PBX features
- Headset connectivity
- Recorder connectivity

**Dimensions:** 10 inches (25.4 cm wide x 9 inches (22.9 cm) deep x 4 inches (10.2 cm) high

### Line Conditions of the SN716 DESKCON

The cable length between the DLC card and terminal varies depending on the type of terminal. The table shows the line conditions of the Attendant Console.

**Line Conditions of the SN716 DESKCON**

| Interface Cards Type | Power Options | Cable Length* (Cable 0.5/24 AWG) |
|---|---|---|
| PN-8DLCL/8DLCP | PN-PW00 | 1000 ft. (304 m) |
| | AC Adapter | |
| PN-4DLCM/4DLCQ | PN-PW00 | 1000 ft. (304 m) |
| | AC Adapter | 4000 ft. (1200 m) |

*Cable length is based on the diameter of the cable and the terminal impedance.

### SN716 DESKCON Exclusive Features

While the DESKSON has full access to PBX features the SN716 DESKCON has the following exclusive features.

| DESKCON Exclusive Features | |
|---|---|
| Attendant Assisted Calling | Call Waiting Display |
| Attendant Camp-on<br>(Full and Semi-automatic) | Common Route Indial |
| Attendant Called/Calling Name Display | Dialed Number Identification Service (DNIS) |
| Attendant Called/Calling Number | Incoming Call Identification |
| Attendant Call Selection | Individual Trunk Access |
| Attendant Console Lockout-Password | Multi-Function Key |
| Attendant Do Not Disturb Setup and Cancel | Multiple Console Operation |
| Attendant Interposition Calling/Transfer | Pushbutton Calling - Attendant Only |
| Attendant Lamp Check | Serial Call |
| Attendant Listed Directory Number | Time Display |
| Attendant Loop Release | Trunk Group Busy Display |
| Attendant Programming | Unsupervised Trunk-to-Trunk Transfer By Attendant |
| Attendant Training Jacks | Attendant Delay Announcement |
| Audible Indication Control | Attendant Lockout |
| Call Processing Indication | Attendant Overflow |
| Call Queuing | Attendant Override |
| Call Splitting | |

**Note:** *For Detail of each feature refer to Chapter 9 Feature Description.*

## Functions and use of Keys and Lamp Indications

| LOCATION NUMBER | KEY OR LAMP DESIGNATION | FULL NAME | BASIC /OPTION | KEY /LAMP | FUNCTION |
|---|---|---|---|---|---|
| 1 | L1-L6 | Loop | Basic | Key | The attendant answers the call associated to the particular loops. Loop keys are usually used to reenter to held calls, answer automatic recalls. |
| 2 | L1A-L6A | Loop Lamp A | Basic | Lamp (one per loop) | Steady green lamp indicates attendant connected to the loop, or called station has answered. Flashing green lamp indicates call waiting to be answered. Steady red lamp indicates call party busy. Flashing red lamp indicates call held at the console. |
| 3 | L1B-L6B | Loop Lamp B | Basic | Lamp (one per loop) | Flashing red lamp indicates automatic recall has been activated. |
| 4 | Push-button Dial | Push-button Dial | Basic | Key | Allows the attendant to:<br>• Process incoming calls<br>• Originate calls<br>• Activate various service features |
| 5 | SRC | Source | Basic | Key & Lamp | Allows the attendant to speak with the calling party. The associated lamp lights when the attendant is connected. The source trunk/station number will be shown in the number display field. |
| 6 | DEST | Destination | Basic | Key & Lamp | Allows the attendant to speak with the called party. The associated lamp lights when the attendant is connected. The destination station/trunk number will be shown in the number display field. |
| 7 | Talk | Talk | Basic | Key | Allows the attendant to join in a three-way conference with the calling and called parties. When connection is established, both SRC and DEST lamps will light. |
| 8 | Cancel | Cancel | Basic | Key | Allows the attendant to:<br>• Disconnect the calling (source) or called party (destination) from the loop.<br>• Disconnect an outgoing trunk or tone seized by the attendant.<br>• Disconnect the station recalling attendant for transfer assistance. |
| 9 | Hold | Hold | Basic | Key | Allows the attendant to hold a call at the console and/or to activate it to serial call state. |
| 10 | Release | Release | Basic | Key | Allows the attendant to release from an established connection freeing the console for processing of new calls. |
| 11 | Answer | Answer | Basic | Key & Lamp | Allows the attendant to answer incoming calls in the order in which they arrive at the console. |
| 12 | Start | Start | Basic | Key & Lamp | Allows the attendant to extend an outgoing call to a station. Completion of outgoing pulse will be recognized. |

## Functions and use of Keys and Lamp Indications (Cont.)

| | | | | | |
|---|---|---|---|---|---|
| 13 | Incoming Call Identification: These eight non-locking keys with associated lamps provide attendant access to specific types of incoming calls. A flashing lamp indicates a call waiting to be answered. A steady lamp indicates a call answered. The standard arrangement of these keys is shown on the face layout. | | | | |
| | LDN | Listed Directory Number | Basic | Key & Lamp | Incoming central office trunk call. |
| | TIE | Tie Line | Basic | Key & Lamp | Allows the attendant to answer incoming Tie Line calls when the distant station dials access digit to the attendant. |
| | Busy | Call Forwarding-Busy Line | Basic | Key & Lamp | Allows the attendant to answer incoming calls to specified station when the station is busy. |
| | ATND | Attendant | Basic | Key & Lamp | Incoming station call. |
| | NANS | Call Forwarding-Don't Answer | Basic | Key & Lamp | Allows the attendant to answer incoming calls to specified station when the station does not answer within the predetermined time. |
| | Recall | Recall | Basic | Key & Lamp | Incoming station call for attendant assistance in transferring an established outside call to another station. |
| | Option | Optional | Option | Key & Lamp | Additional incoming special service calls, such as FX. |
| | Option | Optional | Option | Key & Lamp | Additional incoming special service calls, such as FX. |
| 14 | PAGE | Page | Basic | Key & Lamp | Allows the attendant to connect with pager. (overhead paging) |
| 15 | REC | Record | Basic | Key & Lamp | Allows the attendant to connect with recorder. |
| 16 | EMG | Emergency | Basic | Key & Lamp | Allows the attendant to answer incoming calls from the station where the station leaves the receiver off. |
| 17 | BV | Busy Verification | Basic | Key & Lamp | Allows the attendant to enter into station-to-station connection. |
| 18 | TRKSL | Trunk Selection | Basic | Key & Lamp | Allows the attendant to individually select a desired trunk. |
| 19 | Call Park | Call Park | Basic | Key & Lamp | Allows the attendant to establish incoming call to Call Park. **Note:** *The attendant can connect this call once again by dialing a specific number and individual number of the console from an ordinary extension telephone.* |
| 20 | SC | Serial call Set | Basic | Key & Lamp | Allows the attendant to be automatically recalled when the station user replaces the handset, by depressing the key after extending a central office incoming call to the station user. |
| 21 | SVC | Supervisory Call Set | Basic | Key & Lamp | Allows the attendant to supervise a call by depressing the key after extending a central office incoming call to the station user. |
| 22 | Option | Optional | Option | Key & Lamp | Allows the attendant access to various optional features provided as required. Each key has an associated lamp. |
| 23 | Mute | Mute | Basic | Key & Lamp | Allows the attendant to cut off the voice transmission. |
| 24 | Alarm | Alarm | Basic | Key | Steady lamp indicates trouble conditions in the PBX. |
| 25 | Position Available | Position Available | Basic | Lamp | When the attendant position (console) is available to process calls, the lamp lights. |

## Functions and use of Keys and Lamp Indications (Cont.)

| | | | | | |
|---|---|---|---|---|---|
| 26 | Position Busy | Position Busy | Basic | Key & Lamp | When the key is pressed, the lamp will light, and the console will become not available.<br>**Note:** *Press the button if operators leave their seats.* |
| 27 | Night | Night | Basic | Key & Lamp | Allows the console to place in the night answer mode and lights the associated lamp. Releasing the key restores the console to normal operation. |
| 28 | Up Down | Up and Down | Basic | Key | Allows the attendant to adjust volume of the receiver, ringer and contrast of the LCD. |
| 29 | LCD | Liquid Crystal Display | Basic | Display | The following information will be displayed:<br>1st line:<br>The kind of party that connects to the attendant, the number of the waiting calls, the date and time.<br>2nd line:<br>Tenant number, station class of service and station number belonging to the destination (called) party.<br>3rd Line:<br>Tenant number, station class of service and station number belonging to the source (calling) party.<br>4th Line:<br>Optional indication, such as trunk busy. |
| 30 | Multi-Function Key | Multi-Function Key | Basic | Display | Keys while idle:<br>Mode—allows access to DAY/NIGHT mode and LKOUT (Console Lock out mode)<br>Prog—allows access for programming DISA, System Speed Dial, Date & Time and Tone Ringer<br>Keys while answering or originating:<br>SPB—Out Pulse Short<br>LPB—Out Pulse Long<br>SHF—Flash Over Trunk<br>Keys while calling a busy station:<br>B.V—Busy Verification<br>Keys while calling a DND station:<br>DDOVR—DND override<br>Keys while accessing Hotel feature:<br>RC—Room Cut off<br>MW—Message Waiting<br>DD—Do not Disturb<br>WU—Wake up call<br>RESET—Reset |

# Dterm Series i Analog Terminals

The Dterm Series i is available in two distinct analog models.



**Single-Line**



**Hospitality Single-Line**



**Low End Single-Line**

### Descriptions of the Analog Terminals

| | |
|---|---|
| DTR-1 (WH) TEL | Fully modular with Redial key, Flash key, Message Waiting lamp, Data Jack and Ring/Handset Receive Volume. |
| DTR-1 (BK) TEL | |
| DTR-1HM (WH) TEL | Fully modular with Redial key, Flash key, Message Waiting Lamp, Data Jack, eight programmable Feature/Speed Dial keys and Ring/Handset Receive Volume. |
| DTR-1HM (BK) TEL | |
| DTH-1-1 (BK) TEL | Fully modular with Built-in Wall Mount Unit, Flash Key, Mute Key, Handset Cord, Message Waiting, Redial Key, three different Ring Tones, Volume Control |

### Line Conditions of the Analog Terminals

| Terminal Type | Card Type | Cable Length* (Cable 0.5/24 AWG) |
|---|---|---|
| DTR-1<br>DTR-1HM<br>DTH-1-1 | PN-8LCAK<br>(max. 600 ohms loop resistance) | Approximately:<br>1.43km (.88 miles) |
| | PN-AUCA<br>(max. 2500 ohms loop resistance) | Approximately<br>12.29km (7.63 miles) |
| | PN-4LLCB<br>(max. 2500 ohms loop resistance) | Approximately<br>12.29km (7.63 miles) |
| | PN-8LCAD<br>(max. 600 ohms loop resistance) | Approximately:<br>1.43km (.88 miles) |

*Cable length is based on the diameter of the cable and the terminal impedance.

### Specification for Analog Terminals

| Item | Description |
|---|---|
| Size | 224mm x 165mm +/5mm (Length x Width) |
| Color | Black or White |
| Dial Pad | 12-Key Dial Pad: 4 Rows and 3 Columns;<br>Metropolitan Dial Pad with Alphabet, * and # buttons;<br>Button 5 has a Raised Dot |
| Type of Dial | DTMF and Dial Pulse |
| Function Buttons | Hook flash, Redial Key on DTR-1-1; Hookflash, Program, Redial, Monitor, and Hold Keys on DTR-1HM-1 |
| Message Waiting Lamp | Neon Lamp with Window Design -Glow Through Filter Raised from Surface with MW and Incoming Ring Indication |
| Operating Voltage | Activation Voltage 88V to 108V, Deactivation Voltage 53V or Less |
| Speed Dials | (DTR-1HM Only) 8 Buttons, Maximum 21 Digits |
| Hookflash Timer | 630+/10ms (Fixed) |
| Redial Key | Maximum of 31 digits |
| Ring Vol. Control | 3 Levels (Soft, Medium, Loud) Programmable |
| Ring Tone Pitch Control | 3 Levels (Slow, Medium, Fast, Off) Programmable |
| Handset Receiver & Speaker Volume | 6 Levels (Volume Key) |
| Handset | Hearing Aid Compatible, Dynamic Type Element |
| Handset Cord | 12 feet |
| Directory Card | Large Convenient Directory Card |
| Data Jack | Dedicated Jack; Used for Connection to Modem, Speakerphone, etc., located on back of telephone |
| Wall-Mount Unit | Built-in |
| Electrostatic Discharge | Can Withstand +/- 20kv Discharge |
| Approvals | c-UL (Ul 60950 3rd Edition) FCC part 15, 68,<br>IC (Industry Canada) |

## Dterm Series i (IP) Terminals

The DtermIP gives you the freedom to tailor your platform and telephony applications for your business even as the business grows and your needs expand. With its advanced digital circuitry, the DtermIP consists of several distinct models to meet users' diverse telephone terminal needs.

DtermIP terminals are designed to provide ergonomic form and user-friendly functions. DtermIP terminals offer an adjustable LCD display unit with menu-driven soft key operation, allowing users to program terminals at the desktop. The LCD panels are equipped with three lines of display, each with 245 characters. Standard features include headset jacks, wall mounts and adjustable base units.

DtermIPs has four soft keys located just under the display of each terminal. These menu-driven soft keys allow user's convenient access too many telephony features. According to the status of the multi-line terminal, functions of the soft keys are displayed in the third line on the LCD. If the status of the terminal changes, the soft keys displayed will change automatically.

Dedicated function keys provide easy one-touch access to the most common telephone operations. These keys include: feature, recall conference, redial, hold, transfer, answer, speaker, microphone, directory and message.

DtermIP terminals are ideal choices for users that are connected through a managed IP network. DtermIP terminals are class B devices and comply with U.S. FCC regulations for office and residential use as well as the Canadian Inference-causing Equipment Regulations.



**8-Button Display**



**16-Button Display**



**32-Button Display**



**16-Button LD**



**4-Button Display**

## DtermIP Terminal Features

**DtermIP 4D**

- 4 Line/Feature Access/ Programmable Feature Access Keys
- 9 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

**DtermIP 8D**

- 8 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

**DtermIP 16D**

- 16 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

**DtermIP 16LD**

- 16 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

**DtermIP 32D**

- 32 Line/Feature Access/ Programmable Feature Access Keys
- 11 Dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, Message
- 4 Local Soft Keys Controls (detail functions are dependent on PBX)

## DtermIP Terminal Features

| Display Features | |
|---|---|
| Liquid Crystal Display (LCD) | All Models: 3 lines by 24 Characters |
| Brightness Control LCD Contrast | All Models: Yes |
| Adjustable LCD Display | Dterm 4, 8, 16, 16LD, 32 models only |
| Backlit Display Compatible | Dterm 16D (BL) model only |
| **Receiver Volume Control** | |
| Handset | All Models: Yes |
| Full Duplex Speaker Phone | All Models: Yes |
| Ring Volume Control | All Models: Yes |
| **Miscellaneous** | |
| On-Line Firmware Upgradeable | All Models: Yes |
| DHCP | All Models: Client Support |
| Call Message Indicator | All Models: Yes |
| Headset | Dterm 8, 16, 16LD, 32 models only |
| Supported Adapters | AD(A)-2R (Local Recording)<br>PS(A)-R (Local Line Survivable) |
| Adjustable Base | All Models: Yes*<br>**For Dterm 4D - Terminal height can be adjusted with removal/addition of the base unit.* |
| Built-in Wall Mount | All Models: Yes |
| Built-in Headset Jack | Dterm 8, 16, 16LD, 32 models only |
| Housing Color: Black | All Models: Yes |
| Housing Color: White | Dterm 8, 16, 16LD, 32 models only |

## DtermIP Terminal Specifications

| Network Parameters | |
|---|---|
| Internet Layer | All Models: IPv4 |
| IP Protocol | All Models: NEC Peer-to-Peer (only) |
| Jitter Buffer | All Models: Max 300msec (10msec steps) |
| Payload Interval | All Models: 10ms ~ 40ms (10ms steps) |
| IP Addressing | All Models: DHCP or Static Assignable |
| QoS | All Models: 801.p, ToS and Diff-Service |
| **Power Support Options** | |
| External | All Models: AC: 24V, Current: 750mA |
| Operating Temperature | All Models: 0 - 40 deg C (32 - 103 deg F) |
| Spare Pair Power | All Models: Yes |
| In-Line Power | All Models: Yes |
| External Power via AC adapter | All Models: Yes (Optional adapter) |
| Quality of Service | All Models: Yes<br>Layer 2: 802.1p/Q; Layer 3: IP Precedence, Diff-Services |

| **Equipment Specification Size** (W x D x H) | |
|---|---|
| **DtermIP 4D** | Tilt up: 7.57" x 8.69" x 5.57" |
| | Tilt down: 7.57" x 8.69" x 3.80" (without stand unit) |
| | Weight: 1.98lbs |
| **DtermIP 8D** | Tilt up: 9.09" x 8.54" x 5.28" |
| | Tilt down: 9.09" x 8.54" x 4.17" |
| | Weight: 2.51lbs |
| **DtermIP 16D** | Tilt up: 9.09" x 8.54" x 5.28" |
| | Tilt down: 9.09" x 8.54" x 4.17" |
| | Weight: 2.51lbs |
| **DtermIP 16LD** | Tilt up: 9.09" x 9.88" x 5.28" |
| | Tilt down: 9.09" x 9.88" x 4.17" |
| | Weight: 2.91lbs |
| **DtermIP 32D** | Tilt up: 9.09" x 9.57" x 5.28" |
| | Tilt down: 9.09" x 9.57" x 4.17" |
| | Weight: 2.84lbs |

## DtermIP Terminal Specifications

**DtermIP 4D**

- Voltage: 48V
- Current: 90ma
- Power Consumption: 4.32W
- Audio Algorithm: G.711, G.729A
- Protocol Support: 802.3af (CDP and NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45

**DtermIP 8D**

- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

**DtermIP 16D**

- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

**DtermIP 32D**

- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

**DtermIP 16LD**

- Voltage: 48V
- Current: 92ma
- Power Consumption: 6.4W
- Audio Algorithm: G.711, G.729A, G.723.1
- Protocol Support: 802.3af and CDP (NDP with ILPA integration)
- 10/100 base T (IEEE 802.3), RJ 45 multi-port Switch

# Dterm Series i (TDM) Digital Terminals

The Dterm Series i Terminals are designed to provide ergonomic form and user-friendly functions. With advanced digital circuitry, the Dterm Series i consists of distinct models to meet users' diverse telephone terminal needs.

Dterm Series i Terminals offer adjustable display and non-display units with menu-driven soft key operation, allowing users to program terminals at the desktop. Standard features include headset jacks, wall mount units and adjustable base units. The display units are equipped with large LCD panels with three lines of display, each with 24 characters. A 16-button backlit display version is available for installations in dimly lighted areas such as restaurants, night clubs, and residential applications. Easy to see in either dark or bright applications, the backlight feature may expand installation opportunities and markets.

The Dterm Series i Display Terminals have four soft keys located just under the display of each Terminal. These menu-driven soft keys allow users' convenient access too many features. The state of the terminal will determine what soft key is available to the user. According to the status of the Multiline Terminal, functions of the soft keys are displayed in the third line on the LCD. If the status of the Multiline Terminal changes, the soft keys displayed will change automatically.

Dedicated function keys provide easy one-touch access to the most common telephone operations. These keys include: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Microphone, Directory, and Message.

All Dterm Series i telephones are an ideal choice for both businesses and remote users in residential home offices. All Dterm Series i telephones are Class B devices and comply with U.S. FCC regulations for office and residential use, and with requirements of the Canadian Interference-Causing Equipment Regulations.



**2-Button Non-Display**



**8-Button Non-Display**



**8-Button Display**



**16-Button Display**



**32-Button Display**



**60-Line DSS/BLF Console**



**4-Button Display**



**16-Button LD**



**16LD-Desi-Less ADM**

## Standard features of the Digital models

- Large Message Waiting LED
- 24 Character, 3 Line LCD on display equipped models
- Tilt LCD Unit
- Adjustable Base
- Built-in Wall Mount Unit
- Built-in Headset Jack Connector **Note 1**
- 14 Programmable Ring Tones
- Speed Dial/DSS Buttons
- Programmable Line Keys with 2 Color LED
- Backlit Display on 16-Button Model
- Four Local Soft key Controls (detail functions are dependent on PBX, only provided on terminals with display)
- Eleven dedicated Function Keys: Feature, Recall, Conference, Redial, Hold, Transfer, Answer, Speaker, Mic*, Directory*, and Message*. (*Functionality dependant upon system software.)
- Built-in Half Duplex Handsfree Unit
- Snap-in Options Available: **Note 1**
  - AP(R): Analog TEL connection with Ringing Signal Generation
  - AP(A): Analog TEL connection without Ringing Signal Generation or Disconnect Signal
  - AD(A): Tape-recorder connection
  - CT(A): CTI Adapter, RS-232-C (9-pin) interface
  - IP-R: VoIP Adapter

**Note 1:** *Not available with DTR-2DT and DTR-4D terminals.*

## Series i Multi-line Terminal Descriptions

| | |
|---|---|
| DTR-2DT (BK) TEL | 2 LINE TERMINAL - available in black (BK) or white (WH).<br>(does not support optional adapters)<br>Fully modular with 2 Flexible, 2-color LED Line keys, nine Function Keys, built-in Speakerphone, electronic volume and tone controls. |
| DTR-2DT (WH) TEL | |
| DTR-4D (BK) TEL | 4 LINE TERMINAL – available in black (BK) only.<br>(does not support optional adapters)<br>Fully modular with 4 Flexible, 2-color LED Line keys, nine Function Keys, built-in Speakerphone, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls. |
| DTR-8 (BK) TEL | 8 LINE TERMINAL – available in black (BK) or white (WH).<br>Fully modular with 8 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, four softkeys, Large LED, Electronic Volume and Tone Controls and tilt stand. |
| DTR-8 (WH)TEL | |
| DTR-8D (BK) TEL | 8 LINE DISPLAY TERMINAL - available in black (BK) or white (WH).<br>Fully modular with 8 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls, and tilt stand. |
| DTR-8D (WH) TEL | |
| DTR-16D (BK) TEL | 16 LINE DISPLAY TERMINAL - available in black (BK) or white (WH).<br>Fully modular with 16 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls and tilt stand. |
| DTR-16D (WH) TEL | |
| DTR-32D (BK) TEL | 32 LINE DISPLAY TERMINAL - available in black (BK) or white (WH).<br>Fully modular with 32 Flexible, 2-color LED Line keys, eleven Function Keys, built-in Speakerphone, headset jack, wall mount unit, 24-character by 3-line display, four softkeys, Large LED, Electronic Volume and Tone Controls and tilt stand. |
| DTR-32D (WH) TEL | |
| DCR-60 (BK) CONSOLE | ATTENDANT ADD-ON CONSOLE - Requires an AC-R ADP (included).<br>Fully modular with 48 programmable, 2-color LED keys (for station trunk appearances), 12 Function keys with red LED, and tilt stand. Terminal available in: black (BK) or white (WH). |
| DCR-60 (WH) CONSOLE | |
| 16 LINE Back-Lit | 16 LINE Back-Lit DISPLAY TERMINAL<br>Fully modular with 16 flexible, 2-color LED line keys, eight function keys, built-in speakerphone, headset jack, wall mount unit, back- lit 24-character by 3-line display, four softkeys, large LED, electronic volume and tone controls and tilt stand. Terminal available in: black (BK). |
| 16LD-ADM | Dterm Series i Add-on Module extend the capabilities of the 16LD terminals with additional buttons and an LCD display. This expansion module adds 16 buttons, increasing the total number of buttons with one module. The large LCD display of the 16LD-ADM allows quick and easy identification of associated buttons. The 16 buttons on each of the expansion modules can be programmed as speed dial keys or a directory number. |

### Specification for Series i Digital Terminals

| Item | Description |
|---|---|
| Curl Cord Length | 12ft |
| Weight (no handset) | 510g (min.) |
| LCD Display | 24 digit x 3 line (alphanumeric and some characters) no back light |
| Ringing Sound Level | Max. 80dBSPL (in output limit condition)<br>Max. 86dBSPL (in normal condition) |
| Built in Hands Free | Half duplex |
| Items Provided with Instrument | Line cord, Directory card |
| Handset Cradle | K type compatible |
| LCD angle | 14~42.5 deg. (on the desk, no housing tilt)<br>25~53.5 deg. (on the desk, housing tilt up)<br>-4.4 deg. (wall mounting) |
| Housing Face Angle | 14~25 deg. (on the desk)<br>-4.4 deg. (wall mounting) |
| Recommended Headset | NEC Headsets |
| Other | HAC |

### Dterm Series i Digital Terminal Options

| Item | Description |
|---|---|
| WM-R Unit | Series i Multi-line Terminals with an AP(R)-R, AP (A)-R, CT (A)-R, and/or an IP-R Unit can be wall mounted using the WM-R Unit. |
| AD(A)-R Unit | Provides Series i Multi-line Terminals ability to interface with recording device. |
| AP( R)-R Unit | Provides Series i Multi-line Terminals ability to interface with analog device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. Provides ringing to analog device connected. Requires an AC-R ADP. |
| AP(A)-R Unit | Provides Series i Multi-line Terminals ability to interface with analog device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. No ringing is provided. |
| CT(A)-R Unit | Connects a Series i Multi-line Terminal to a PC providing a complete turnkey package with graphical telephone user interface and call logging. Shipped with Multi-line Phone Kits software. Supports Serial interface. |
| IP-R Unit | A compact plug-and-play device that installs into the base of a Dterm Series i display terminal. Integrated two-port 10/100baseT Ethernet pass through hub that permits using one port to connect the network interface card (NIC) from the PC to the IP network. The other is plugged directly into a LAN or an IP network device such as a router, DSL modem or cable modem. Requires an AC-R ADP. |
| AC-R Unit | AC adapter for DSS/BLF Console, AP(A), AP(R), or IP-R Unit |

## Dterm Series i Line Conditions

| Cable Length Note 1 | | | Standard | with AC Adapter Note 4 |
|---|---|---|---|---|
| Dterm Series i | Dterm 8 / 8D | 8DLC | 300m (984ft) | **Note 2** |
| | | 4DLC | 300m (984ft) | 1200m (3937ft) |
| | | 2DLC | 850m (2789ft) | 1200m (3937ft) |
| | Dterm 16/16D | 8DLC | 200m (656ft) | **Note 2** |
| | | 4DLC | 200m (656ft) | 1200m (3937ft) |
| | | 2DLC | 850m (2789ft) | 1200m (3937ft) |
| | Dterm 32/32D | 8DLC | 200m (656ft) | **Note 2** |
| | | 4DLC | 200m (656ft) | 1200m (3937ft) |
| | | 2DLC | 850m (2789ft) | 1200m (3937ft) |
| | DSS/BLF Console **Note 3** | 8DLC | - | 300m (984ft) |
| | | 4DLC | - | 1200m (3937ft) |
| | | 2DLC | - | 1200m (3937ft) |

***Note 1:*** *Cable length is based on the following conditions.*

- *Diameter of the cable is 0.5 mm.*
- *The Protection arrester is not inserted between the terminal and PBX.*

***Note 2:*** *When using 8DLC card, it is not available for long line function, even if it is equipped with AC Adapter.*

***Note 3:*** *The DSS/BLF Console requires local AC/DC supply.*

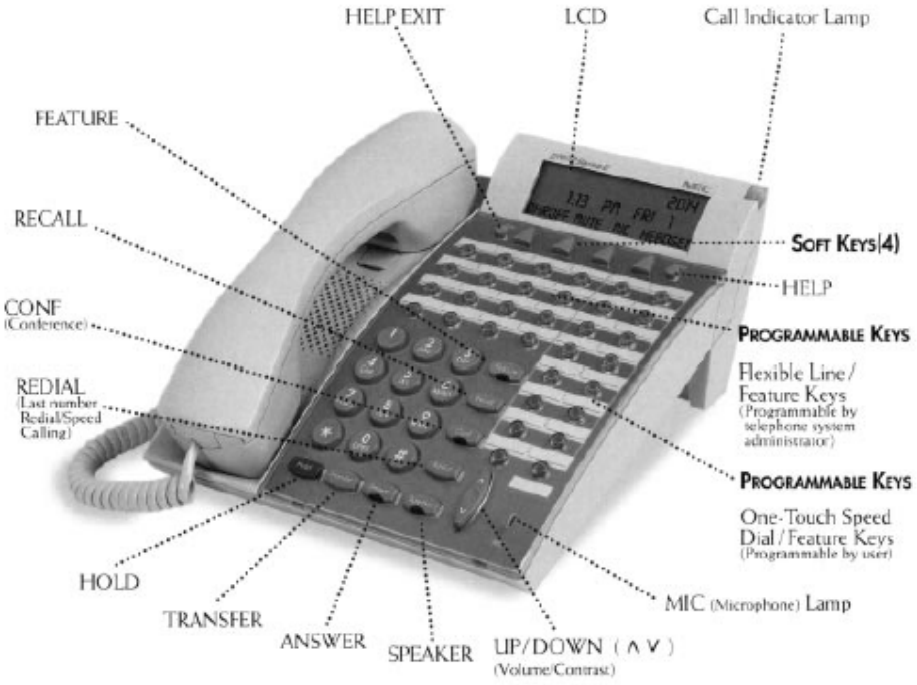
***Note 4:*** *DTR-2DT and DTR-4D terminals do not support long line adapters.*

# Dterm Series E (TDM) Digital Terminals

The Dterm Series E terminals are designed to provide ergonomic form and user-friendly functions. With advanced digital circuitry, the Dterm Series E consists of distinct models to meet users' diverse telephone terminal needs.

Dterm Series E terminals offer adjustable display and non-display units with menu-driven soft key operation, allowing users to program terminals at the desktop. The display units are equipped with large LCD panels with three lines of display, each with 24 characters. Each terminal offers an optional full duplex speaker phone operation for two-way conversation. Standard features include headset jacks, wall mount units and adjustable base units.

The Dterm Series E Display Terminals have four soft keys located just under the display of each Terminal. These menu-driven soft keys allow user's convenient access too many features. The state of the terminal will determine what soft key is available to the user. According to the status of the Multiline Terminal, functions of the soft keys are displayed in the third line on the LCD. If the status of the Multiline Terminal changes, the soft keys displayed will change automatically.



## Dterm Series E Terminals



**8-Line Non-Display**



**8-Line Display**



**16-Line Display**



**32-Line Display**



**60-Line DSS/BLF/Add-On Module**

## Dterm Series E Terminals

**8-Line Non-Display**

- 8 programmable line/feature keys with two-color LED indication
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Message waiting LED
- This terminal is available in Black (BK) or White (WH).

**8-Line Display**

- 8 Programmable line/feature keys with two-color LED
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Messaging Waiting LED
- 24-character, 3-line, adjustable Liquid Crystal Display (LCD)
- Four Softkeys
- This terminal is available in Black (BK) or White (WH).

**16-Line Display**

- 16 Programmable line/feature keys with two-color LED
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Messaging Waiting LED
- 24-character, 3-line, adjustable Liquid Crystal Display (LCD)
- Four Softkeys
- This terminal is available in Black (BK) or White (WH).

**32-Line Display**

- 16 Programmable line/feature keys with a two-color LED
- 16 SPD/DSS keys with two-color LED
- 8 dedicated Function Keys
- Built-in speakerphone
- ADA compatibility
- Large Messaging Waiting LED
- 24-character, 3-line, adjustable Liquid Crystal Display (LCD)
- Four Softkeys
- This terminal is available in Black (BK) or White (WH).

**60-Line DSS/BLF/Add-On Module**

- 60 programmable line keys (each with a two-color LED)
- This terminal is available in Black (BK) or White (WH).

## DTERM SERIES E FEATURES

- Four Local Soft key Controls (detail functions are dependent on PBX)
- Large Message Waiting LED
- 24 Character, 3 Line LCD
- Tilt LCD Unit
- Adjustable Legs
- Built-in Wall Mount Unit
- Built-in Headset Jack Connector
- Speed Dial/DSS Buttons
- Programmable Line Keys with 2 Color LED
- Eight dedicated function keys (Feature, Recall, Conf, Redial, Hold, Transfer, Answer & Speaker)
- Built-in Half Duplex Handsfree Unit
- Optional: Full Duplex HFU (consist of HFU-U Unit, External Microphone Unit and AC adapter)
- Snap-in Options Available:
  - ADA-U (Ancillary Device Adapter)
  - APR-U (Analog Port Adapter with Ringing)
  - HFU-U (External Handsfree Unit)
  - ACA-U (AC Adapter Unit)
  - WMU-U (External Wall mount Unit)
  - WMU-W (External Wall mount Unit for DTP-2 Terminals)

## Dterm Series E Digital Terminal Options

| Item | Description |
|---|---|
| ADA-U UNIT | Ancillary Device Adapter;<br>Used for External Recording Devices |
| APR-U UNIT | Analog Port Adapter with Ringing, used to connect external analog devices such as PC Modem, analog Cordless etc. Requires ACA-U Unit for local power. |
| APA-U UNIT | Analog Port Adapter without Ringing, Used for external speakerphones.<br>Requires ACA-U Unit for local power when the connected device is over<br>50 ft. from the APA-U Unit. |
| HFU-U UNIT | Full duplex Speakerphone with external microphone. |
| CTA-U | CTI Adapter Unit |
| ACA-U UNIT | AC Adapter provides local power for APR-U Unit, APA-U Unit, CTA-U Unit, HFU-U Unit, IPW-U Unit, DESKCON, and Dterm long line. |
| WMU-U UNIT | Wall Mount Unit, Required when using terminal adapters attached to phone. If not using terminal adapters built-in wall mount in phone is used. |
| IPW-2U (ELC) | The IPW-2U (ELC) adapter is a full duplex switch, which can be installed on any of the Dterm Series E display phones. With this adapter, you can upgrade the 8, 16, 32-button Dterm Series E terminals to VoIP when using the PN-32IPLA (ELC) IP station card. |
| IPW-2U (P-P) | The IPW-2U (P-P) adapter is a full duplex switch, which can be installed on any of the Dterm Series E display phones. With this adapter, you can upgrade the 8, 16, 32-button Dterm Series E terminals to VoIP when using Peer to Peer connection. |

## Dterm Series E Line Conditions

| **Loop Resistance Note 1** | | | | |
|---|---|---|---|---|
| Analog Telephone Set | Standard | | 600 ohms | |
| | Option | | 2,500 ohms | |
| Loop Start Trunk | Exchange Line | | 1,700 ohms | |
| | Tie or DID Line | | 2,500 ohms | |
| **Cable Length Note 2** | | | **Standard** | **with AC Adapter** |
| Dterm Series E | Dterm 8 / 8D | 8DLC | 300m (984ft) | **Note 3** |
| | | 4DLC | 300m (984ft) | 1200m (3937ft) |
| | | 2DLC | 850m (2789ft) | 1200m (3937ft) |
| | Dterm 16/16D | 8DLC | 200m (656ft) | **Note 3** |
| | | 4DLC | 200m (656ft) | 1200m (3937ft) |
| | | 2DLC | 850m (2789ft) | 1200m (3937ft) |
| | Dterm 32/32D | 8DLC | 200m (656ft) | **Note 3** |
| | | 4DLC | 200m (656ft) | 1200m (3937ft) |
| | | 2DLC | 850m (2789ft) | 1200m (3937ft) |
| | Dterm 8D-1IP | _ | Max. 328 ft. between Ether/IP-PAD and Router/ Switching Hub | |
| | Dterm 16D-1IP | _ | Max. 328 ft. between Ether/IP-PAD and Router/ Switching Hub | |
| | Dterm 32D-1IP | _ | Max. 328 ft. between Ether/IP-PAD and Router/ Switching Hub | |
| | DSS/BLF Console **Note 4** | 8DLC | - | 300m (984ft) |
| | | 4DLC | - | 1200m (3937ft) |
| | | 2DLC | - | 1200m (3937ft) |
| Operator Position | Attendant Terminal | | Same as Dterm Series E | |
| | SN716 Desk Console | 8DLC | 300m (984ft) | **Note 3** |
| | | 4DLC | 350m (1148ft) | 1200m (3937ft) |
| | | 2DLC | 350m (1148ft) | 1200m (3937ft) |

**Note 1:** *Loop resistance includes an internal resistance of telephone set or distant exchange.*

**Note 2:** *Cable length is based on the following conditions.*
- *Diameter of the cable is 0.5 mm.*
- *The Protection arrester is not inserted between the terminal and PBX.*

**Note 3:** *When using 8DLC card, it is not available for long line function, even if it is equipped with AC Adapter.*

***Note 4:*** *The DSS/BLF Console requires local AC/DC supply.*

# Dterm Cordless Terminals

With the NEC Cordless product line, the distance of your handset cord does not govern your work area. The mobility, convenience and reliability of these cost-effective telephones empower employees to answer calls regardless of their location. Potential customers do not waste time playing telephone tag, and employees are not glued to their desk waiting for that all-important call. Business is not lost because of callers being able to reach a live person. The NEC Cordless application is the ideal solution for those businesses that require mobility but do not want to invest in a full-blown wireless solution.



**Dterm Cordless II**



**Dterm Cordless Lite II**



**Dterm Headset Cordless II**



**Plantronics CT Cordless**

## DTERM CORDLESS II

Dterm designed for employees who need digital multiline desktop functionality in a cordless handset. The 2x16 line display provides valuable calling information. The Dterm Cordless II is designed to eliminate noise and provides a range of up to 350 feet.

### Dterm Cordless II Telephone Features

- Advanced Digital Technology eliminates noise and provides an extended range.
- Direct interface to digital interface port on NEC telephone systems.
- Programmable Function Keys let you utilize features of your telephone system with the Dterm Cordless Multiline Terminal.
- 2x16 Digit LCD displays messages, including caller name and number.*
- Conference calling saves time and improves office communication.
- Compatibility enhances the Electra Elite IPK, *UNIVERGE* NEAX® 2000 and *UNIVERGE* NEAX 2400 telephone systems.
- Message LED indicates messages in voice mail.
- Headset jack for hands-free operation.
- Hold mode for handling multiple calls.
- Transfer function for switching lines.
- Color- black

Note: Different displays - The Dterm Cordless II Multiline Terminal handset operates with the NEC Electra Elite IPK, *UNIVERGE* NEAX 2000 and *UNIVERGE* NEAX 2400 systems supporting the Dterm Series III/E/i interface. Connection with each system will result in system-specific displays.

**Dterm Cordless II Multiline Terminal will display caller name or number if Caller ID is a feature of the host system.*

### Dterm Cordless II Specifications

The Dterm Cordless II complies with FCC and IC parts 15 and 68.

General:

- Frequency Control: Phase Lock Loop
- Modulation: Spread Spectrum
- Operating Temp: 0° to 50° C (32° to 122° F)
- Output Power: 63 mW
- Occupied Bandwidth: 1,600 kHz
- Data Transmission Speed: 688 bps

Base Unit:

- Receive/Transmit Freq.: 902 MHz - 928 MHz
- Power Requirements: 10 Vdc from supplied AC adapter
- Size: 4-1/4 in. W x 7-1/2 in. D x 2-1/4 in. H
- Weight: approximately 15.4 oz.

Handset:

- Receive/Transmit Freq.: 902 MHz - 928 MHz
- Power Requirements: Nickel-Cadmium Battery Pack

Battery Charger:

- Power Requirements: 9V DC from supplied AC adapter
- Size: 2-1/5 in. W x 1-2/3 in. D x 8-2/3 in. H (with antenna)

## DTERM CORDLESS LITE II

Dterm designed for employees who need digital multiline desktop functionality in a cordless handset. The 2x16 line display provides valuable calling information. The Dterm Cordless Lite is designed to eliminate noise and provides a range of up to 150 feet. The Dterm Cordless Lite is ideal when the range of the Dterm Cordless II is not required.

### Dterm Cordless Lite II Features

- 900 MHz FM with ADPCM
- 2-line, 16-digit LCD Display
- Dterm Cordless Lite II Headset Jack
- Channel Selection Control
- Ringer Volume Control
- Handset Volume Control
- Single Key Access to: Conference, Hold, Transfer, and Mute features
- Four Programmable Keys: F1-F4
- Separate Charging Stand with Spare Battery Charging Capability
- Auto Standby
- AutoTalk
- Silent Alarm
- Out of Range Protection
- Low Battery Protection System
- Key Pad Lock Feature
- Wall Mountable Separate Base Unit
- Wall Mountable Separate Charging Unit
- Easy Installation
- Compact Handset Design
- Use with an NEC Digital Multiline Terminal or as a stand-alone device

### Dterm Cordless Lite II Telephone Specifications

General:

- Frequency Control: Phase Lock Loop
- Modulation: 900 MHz FM with ADPMC (digital)
- Operating Temperature: 0° to +50° C (+32° to +122° F)
- Bandwidth: 50 kHz
- Data Transmission Speed: 688 bps

Base Unit:

- Receive/Transmit Frequency: 902 MHz - 928 MHz
- Power Requirements: 10 Vdc from supplied AC adapter
- Size: 4-1/4 in. W x 7-5/8 in. D x 2-1/4 in. H
- Weight: Approximately 13.7 oz.

Handset:

- Receive/Transmit Frequency: 902 MHz - 928 MHz
- Power Requirements: NiMH Battery
- Size: 2 in. W x 1-1/4 in. D x 5-1/2 in. H (with antenna)
- Weight: Approximately 5.2 oz. with battery
- Battery: 700 mAh, 3.6V
- Talk Mode: 6 hours (typical)
- Standby Mode: 5 days (typical)

Battery Charger:

- Power Requirements: 9V DC from supplied AC adapter
- Size: 1-3/8 in. W x 1-1/2 in. D x 2-1/4 in. H

## DTERM HEADSET CORDLESS II

Dterm Headset Cordless II allows mobility and sound quality to handsfree conversations on your Dterm Series i TDM phone with an APR-R Adapter. This NEC cordless, along with an APR-R device, eliminates the need for a handset mechanical lifter. The Dterm Headset Cordless II and Series I with an APR-R allow you to conveniently and automatically answer calls with the touch of a button. Because it goes on and off hook electronically, you don't have to be at your desk to answer or hang up the phone.

- No Mechanical Lifter Needed
- Roam up to 300 feet in a typical office environment.
- Talk for up to eight hours without recharging.
- Fully recharge the headset in less than three hours; 80% recharged in 1.5 hours.
- Answer and end calls while away from your desk with the touch of a button.
- Noise-canceling microphone.
- Quickly mute the microphone with the touch of a button.

## Plantronics CT CORDLESS

Package includes Base Unit, Handset (with a standard battery 6 hours of Talk Time or 3½ days of Standby), Belt Clip, M175 Headset, User Guide and AC Adapter. This compact unit is an Analog single line device that operates in the 2.4 GHz Digital Spread Spectrum frequency range. This unit is compatible with and Analog single line port, Analog Port Adapter or Analog telephone line from Telco.

### Specifications

- 2.4 GHz Cordless Headset Phone
- Range of Up To 150 Feet
- Audible Low Battery Indicator
- Single Line Operation
- Ultra-Compact Remote Unit with Belt Clip
- Variable Range Volume Control
- 10 Number Speed Dial
- Page / Find Feature
- Redial / Flash
- Mute with Audible Reminder
- Talk / Charge / Power Indicator Lights
- Built In Headset Stand

# Wireless Dterm PS Handset

### DTERM PS (Personal Station)

The Dterm PS offers portability, clear digital signal quality and has the same features of a desktop phone. Weighing less than 4 oz. with 6 hours of continuous talk time and 300 hours on standby, the Dterm PS phone has been uniquely designed for people on the run. The Dterm PS provides the freedom and convenience of a mobile phone with the advantages and features of a desktop handset.

- Continuous coverage – walk freely around your workplace while on a call.
- No static or fading – audio quality indistinguishable from a desktop phone.
- Small lightweight handset with long battery life.
- Low power operation for compatibility in any environment.
- Up to 512 PS Handsets
- 216 PS Simultaneous Connections



## DTERM PS (Personal Station)



**PS Handset**

**Dterm PS Features**

- 2-Line Operation with Number Sharing
- Voice Mail Message Waiting
- Calling Party Name and Number display
- Support for Wireless Roaming
- Call Hold & Transfer
- Directory Dial (100 entries with name)
- Speed Dial (20 entries with name)
- Calling Party Number Redial & Last Number Redial (5 entries each)
- Headset Operation with Automatic Answer
- Vibration Alert
- Support for Modem Data
- 4 line, 18 Digit LCD Display (with scrolling)
- User-defined soft keys

And the provisioning of the Dterm PS data port facility to allow software updates protects your investment.

### Accessories for the Dterm PS

- Standard Battery Charger Base Unit

- Lithium-ion Battery (provides up to 4 hours of continuous talk-time and 300 hours on standby)

- Leather Cases
  - Leather Case with Swivel Clip - Includes a Swivel Belt Clip with Quick Release
  - Leather Case with Fixed Clip Includes a fixed Belt Clip

  Both cases offer an Integrated D-Ring for use with the optional adjustable necklace**.** The adjustable necklace (from 23" to 42") provides security from dropping the handset while, for example, carrying it in a lab coat pocket.

- Dterm PS Headset
  - Head band type
  - Single ear cover
  - 47" cord

  The Dterm PS Headset is a classic banded headset with a microphone extension. The Dterm PS Headset is typically used in cases of prolonged active use (for example, customer service agents).

- Dterm PS Ear Piece
  - Lower profile Ear Bud receiver
  - In-line Microphone
  - 47" cord

  The Dterm PS Ear Piece is an ear-bud type with an in-line microphone. The Ear Piece is for occasional use or where concealment is desired (for example, hotel security).

- Dterm PS Modem Cable
  - Connects to the modem jack of your laptop or palmtop
  - Supports speeds up to 19.2Kbps
  - Autodial from PC Application (Application must support dialing without waiting for dial tone)

- Dterm PS Assistant Software - The Dterm PS Assistant software and interface cable will allow you to enter the directory and speed-dial entries on your PC and upload to your Dterm PS.

  You can now:
  - Enter, Synchronize and backup your Directory and Speed Dial entries
  - Copy and paste entries from your Desktop Organizer and upload to your Dterm PS.
  - "Clone" all or some of your Dterm PS handset Directory and Speed Dial entries, so everyone in your organization has access to key personnel and department phone numbers.

  There are two versions available: Dterm PS Personal Assistant and the Dterm PS Group Assistant. The Personal Assistant supports only one Dterm PS handset while the Group Assistant is designed to manage and maintain Directory and Speed Dial entries for an entire workgroup. The PS Interface Cable connects to the PC serial port and to the Dterm PS interface port, on the bottom of the handset.

## Dterm SoftPhone (SP30)

The Dterm SP30 allows customers to capitalize on the advantages of a converged voice and data network whether they're in the office or on the road. The Dterm SP30 combines traditional business communication needs with the data applications your customers require.

The Dterm SP30 optimally delivers high quality voice via a USB-connected headset. With a simple drag and drop, the Dterm SP30 allows telephone dialing from other telephone directory applications such as Microsoft Outlook®, HTML pages and Word® documents, etc. In addition, the Dterm SP30 provides an interface to Microsoft's Telephony Application Programming Interface (TAPI) via NEC OpenWorX integration, allowing TAPI-enabled applications, such as Outlook and ACT, to make and receive calls. The Dterm SP30 has the ability to use a wireless handset (PS) for all voice connections instead of the USB handset. The Dterm SP30 can be displayed in 1 of 4 different colors (black, red, gold and neon blue) in order to reflect the "personality" of the day. The Dterm SP30 also allows for 3 different modes of operation.

**Maximized mode:** Access to full line of softphone features such as application sharing, member lists, conference mode, chatting capabilities, Internet access and many others are just one click away.

**Compact Mode:** L-shaped user interface, operating in a small footprint on the PC screen. Compact view allows the softphone to remain active while another application window such as a Word document; database file or email is the primary focus on the PC. With the compact view, the most popular features of the converged softphone are just a click away.

**Task Mode:** The softphone can be minimized and shown as a task within a Microsoft Operating System. While operating in this mode, the softphone will output an audio notification to the user upon receiving an incoming call. It will be up to the user to utilize the hot key in order to activate the Dterm SP30 application and answer the call.

**ACD Mode:** The Dterm SP30 ACD client can support two operation modes, ACD and Business. The Business mode is the default mode. When the ACD mode is configured a new set of function keys becomes available to the Dterm SP30 ACD user. A popup menu is also available when in the ACD terminal mode. While in ACD mode the Instant Messaging feature functions to the MY Line of the terminal, not the ACD Line.

**The SP30 features three modes of operation:**

**Maximized mode:**
Access to full line of Softphone features.




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**Compact mode:**
L-shaped screen interface offers small footprint.



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**Task mode:**
Minimized icon in system tray.



## Key Features and Benefits

The Dterm SP30 offers a full complement of station and converged features for an important segment of mobility and PC-based applications. Even though a majority of the hardware telephone features function on the softphone, the main focus for the softphone is its ability to deliver PC capabilities to the telephone.

**Displays call records in Quick Reference List via the Call Log Window:** All calls are recorded in a Call Log. Telephone number, date of the call and identification as to whether or not the call was received and answered are all logged. You can find at a glance when and with whom you talked. The Dterm SP30 also offers a call memo function whereby you can record the key points of the call so that you can visually identify the individual records.

**Links with PC applications:** Now you can collaborate on a white board application (Word® or Excel® document or any other application) that is operating on your PC and the PC at a distant site. Simply place a call to the far-end Dterm SP30 user and activate the application you will be collaborating on. You are now able to have a more productive conversation.

**Internet Link:** The Dterm SP30 can be assigned with a common database link for fast access to a particular site. This link could be an Internet link that needs to be accessed when receiving or placing a call (i.e., you receive a call from a customer and need to look up information that is contained in a database). Previously, you needed to locate and launch the application before loading the file. Now with the Dterm SP30, simply select the Internet link and the database file is opened, bringing it all together in one user interface.

**Real Time Communication:** In addition to providing a voice and data collaboration link, you can also chat with a remote Dterm SP30. This is ideal for the real quick conversation you need with a distant Dterm SP30 user. Maybe you're on a call and need to get a quick response from a co-worker. There's no need to put the customer on hold and call the co-worker for an answer. Instead, send a chat message and get your answer in real time. The Dterm SP30 will store all your chat messages in a log so that you can use them for future reference. Additional features of the Dterm SP30: forwarding control selection for different call modes, videoconferencing capabilities, automatic downloading of telephony features to the display and diagnostic capacities for audio problem notification.

**Presence/Status:** The presence/status functionality allows the user to confirm a buddy's presence/status with a visual indication (ICON) and text message on the Dterm SP30 GUI. The presence/status information is provided by the OpenWorX servers (LSI) package.

**8/6 Party Conference Control:** When the trunk conferencing card is configured within the voice switch and the voice trunk channel is configured in the program utility for the Dterm SP30, a user can dynamically setup and save future dated collaboration conferences. All the conferences can be activated immediately upon configuration or saved for a future date upon activation. When the conference is activated, the Dterm SP30, with the help of the Voice sever, places calls to the configured participants based on the number which are user predefined. Conferences do not require users to be Dterm SP30 users for voice only conference calls. Only Dterm SP30 users will utilize additional features like application sharing and messaging during call.

**PHS/PCS Collaboration:** PHS/PCS collaboration links the wireless handset capabilities to the Dterm SP30 GUI. In this mode, all the voice connections will be directed across the PS infrastructure instead of through Dterm SP30 USB handset. The Dterm SP30 gui will give the indication that it has received a call such as caller ID, ringing information and notification as to the state of the line. The only thing that changes is voice path direction being directed to the PS handset. The Dterm SP30 will provide data collaboration interface. The benefits for this functionality are as follows:

1. User has a more reliable voice connection with the voice server. The PS provides a constant voice connection over that of the IP voice connection.
2. Dterm SP30 user now has a handset which can be ported from place to place without the need for programming call forward functionality of the phone.
3. Now the user can be reached by one number, no matter if call forwarding is set or the user is located next to the Dterm SP30.
4. User now has a physical device which they can use for dialing and answering calls instead of working with mouse's and keyboards.
5. User is no longer locked to their PC for receiving and answer calls. The user is as portable as the in-building wireless network.

**Voice Recording:** A user will be able record the voice connection and save the wave file on a network storage place of their choice. It is recommended all voice recording be stored on a local hard drive and not a network drive. With the activation of the voice recording of the Dterm SP30, no extra recording equipment other than the Dterm SP30 phase 2 application with a USB headset is required. For those specific regions which require the notification to the remote party that a voice connection is being recorded, the Dterm SP30 provides for the setting of an automatic beep tone in the configuration menus. Beep tone can also be conditioned to send a tone notification at user selectable intervals.

**One Key Operation:** The space bar key on the keyboard for the Dterm SP30 PC can be configured in such a way to allow fast access to placing calls, receiving calls and termination of calls. You are no longer required to utilize a combination of the keyboard and mouse to access these telephone features.

**Call Log Export:** The Dterm SP30 provides for the capability of saving the call log file in a ".cvs" file format in addition to the administrator/user selecting a network storage place for the file. When opening the file with application render capable of displaying "*.cvs" files, columns breaking out caller direction, date and time, caller ID, duration of the call, notification of whether the call was recorded and memo information are displayed.

**Voice Quality Alarm:** The Dterm SP30 is able to provide notification to the user when it is perceived to be transmitting and/or receiving voice problems. When the Dterm SP30 application detects a voice problem an alarm is lit on the GUI and information about the audio problem is stored in the maintenance log file.

**2 Line Display:** When the Dterm SP30 is set for a compact mode, the display which is shown is only capable of displaying one line of information. When configuring the Dterm SP30 to a 2 line display, the display is able to switch (1 second intervals) between the normal 2 lines of display.

**Automatic Idle Return:** This function allows for automatic disconnection during ROT or BT tone being heard due to disconnecting by remote party. Administrator/user can choose to disable or enable this function within the user configuration. The default for this configuration is disabled.

**PB tone sending:** During a call, a PB tone of the dial number which is set on the member button can be sent by selecting the [send PB tone] with a right-mouse click of speed dial number.

**Pause into Dialing string:** This feature allows the user to insert a pause into the dialed number during the following conditions:

- Drag/drop into the LCD or telephone ICON.
- Copy/Paste
- Placing/transferring a call with "Call to:" tag
- Placing/transferring a call and sending the PB tone with using the member button.

**Last Number Redial:** A Dterm SP30 user can do a quick access and call to the most recent call placed by the SP30. During the idle stage a user can redial the telephone numbers previously dialed by selecting the [Redial] in pop up menu which appears by right-clicking on a "Call" button.

**Redial via Call Log:** The Dterm SP30 is capable of displaying the last 32 calling/called telephone numbers. This is done by depressing the Up/Down arrow key (Keyboard) when the LCD field is highlighted.

- First number displayed on LCD screen will be the most recent called/received telephone number.
- Pressing the arrow up will cycle from the most recent to the latest calls/received telephone number.
- Pressing the arrow down button will cycle from the last to most recent calls received/placed by the user.

**One Touch Button in Compact Mode:** A button called the "One Touch" button has been added to the compact panel. The "One Touch" button allows access quick access to your speed dial list (Buddy List) that is conditioned on the main GUI under the Member buttons. By clicking the "One Touch" button, you can place a call or transfer a call to any of the member's numbers that has pre-registered on the main GUI.

**Notification of Call Received with No Answer:** The Dterm SP30 Call Log button which is located on the main and compact panel will flash in blue when a call was receive by the Dterm SP30 but not answered. Before the integration of the blinking call Log button the only way a user was noted that a call was received was if the remote party left a voice mail. In those case where there was no voicemail message was left, the user had no way of identifying whether calls were received or not. Users can confirm visual that a call was received and immediately reply to the caller when they become available.

**Application Collaboration:** The NEC Dterm SP30 allows users to share ideas, information and programs in a variety of ways while either in a point-to-point connection or 6/8 party conference mode.

- **Videoconferencing -** The Dterm SP30 audio and videoconferencing feature lets you communicate with anyone on the NEC Network.
  - Share ideas, information, and applications using video and audio
  - Send and receive real-time images using Windows-compatible equipment
  - Allows for broadcasting of the live video to other Dterm SP30 users which might not have video transmission capabilities.
  - Use of a video camera to instantly view items, such as hardware devices, road conditions or even personnel, which are displayed in front of the camera

- **Whiteboard -** The whiteboard lets you collaborate in real time with other Dterm SP30 users via graphic design. With the whiteboard, you can review, create and update graphic information.
  - Manipulate contents by clinking, dragging and dropping information on the white board with a mouse/keyboard.
  - Copy, cut and paste information from any Windows-based application into the whiteboard.
  - Use different-colored pointers to easily differentiate participant's comments.
  - Save the whiteboard contents either at the local side or distant end location
  - Load saved whiteboard pages, enabling you to prepare information before a conference, then drag and drop it into the whiteboard during an audio meeting

- **Chat -** The chat functionality lets you conduct real-time conversations via text with as many Dterm SP30 users as you like. With chat, you can:
  - Type text messages to communicate with other co-workers during a conference
  - All messages are sent in a whisper mode so that they are only received by one party
  - All messages sent and received are saved automatically in the chat log
  - Automatic pop up notification when a chat message is received
  - ICON notification within the chat log identifying different states of the messages

- **File Transfer -** File transfer lets you send one or more files to distant Dterm SP30 users. With file transfer, you can:
  - Send a file to other Dterm SP30 users
  - Accept or reject transferred files

- **Application Sharing -** Dterm SP30 gives you better control over how shared programs are displayed on your desktop and give the person sharing the program control over who uses it.
  - View shared programs in a frame, which makes it easy to distinguish between shared and local applications on your desktop
  - Minimize the shared program frame and do other work if you don not need to work in the current conference program.
  - Easily switch between shared programs using the shared programs taskbar.
  - Approve conference participants' requests to work in the program you introduce.
  - Allow or prevent others from working in a program using the sharing dialog menu.

# INASET™

INASET terminals are members of the Dterm IP family. INASET terminals have a Web browser with a large color display and a built-in multi-port Ethernet switch for connectivity to the user's local PC. INASET terminals bring a wealth of information to the desktop, including short text display messages and Web pages specifically tailored for the small screen format

The INASET's basic load includes a graphical telephony application that provides an abundance of telephony information and desktop control that is easy to use with its menu-based interface. Information and controls accessible via the softkeys and feature buttons include:

- **Line status** showing a visual icon display for the status of all assigned phone lines and DSS lines.
- **Caller information** showing a visual text display for things such as time, date and call status information.
- **Telephony Directory** for storing, searching and dialing different profiles which you can categorize and store in one of three different groups: corporate, personal and group.
- **Web access** providing browsing capabilities to display HTML web-based information located on the Internet or Intranet. Also includes support for Java applets.
- **Virtual keys** providing access to features, functions and recent keys activated on the terminal. The user can program display and functions for how they see fit with limited or no administrative support necessary.

By converging a company's voice and data networks, there is only one network to manage. The INASET includes a built-in switch, so you can use a single Ethernet switch port for the computer (data) and the INASET. Because it's an IP based telephone, it can be installed anywhere on a corporate IP network. The INASET is Dynamic Host Configuration Protocol (DHCP) compatible and doesn't require the need to be co-located with the *UNIVERGE* NEAX PBX equipment. INASET supports G.711, G.729a and G.723.1 audio compression for low-bandwidth requirements.



## INASET Applications

The INASET is specifically suited for the enterprise environment, including the following end users: managers, purchasing agents, consultants and call agents. But this is just the beginning. This advanced IP business terminal can be programmed to do much more than a standard business phone. For example, the customizable user interface (that's developed with the SDK) can replace single line phones that are typically installed in cafeterias, break rooms, lobbies and manufacturing floors. The INASET's programmability is also ideal for traders, stockbrokers, real estate brokers, executives, PR professionals and any other occupation in which professionals use Internet-related information to interact with the public.

- **Personalization at the desktop:** Individual customization can bring flexibility to the desktop. Software developers can enhance traditional features such as Answer, Redial, Conference, Recall and Help menus.
- **Centralized services:** Information or features that need to be accessed by numerous individuals within the organization can be centralized on servers. Directory service functions, Intranet web site information, customer records, CTI applications and other work group applications are examples of centralized services that could improve portability and use throughout the enterprise network.

### INASET terminals

| Description | Remarks |
|---|---|
| ITR-240G-1 (BK) | The INASET 240G has a 3" LCD with a gray scale 240 x 160 pixel display and 16 programmable keys.<br>Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol).<br>Available in black only. |
| ITR-320C-1 (BK) | The INASET 320C has a 5.1" color display and 32 programmable keys.<br>Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol).<br>Available in black only. |
| ITR-320G-1 (BK) | The INASET 320G has a 5.1" LCD with gray scale display and 32 programmable keys. Supports Power over Ethernet (POE) (802.3af and Cisco Discovery Protocol).<br>Available in black only |

### INASET Accessories

| Description | Remarks |
|---|---|
| WM-RL UNIT | Wall mount for the ITR-LC-1 |
| INASET Headset Cord 12’ | 12 Foot Black Handset Cord |
| INASET Headset Cord 25’ | 25 Foot Black Handset Cord |
| AC-R | AC adapter for the INASET |
| ILPA-R | 802.3af power Dongle for INASET Original |
| ADA-2R | Audio recording adapter for IP terminals |
| MIC-R (BK/WH) UNIT | External Microphone for INASET 240, 320s |
| PSA-R | Analog survivable adapter |

### INASET Basic Network Capability

- Dual port 10/100 Mbps Ethernet switch
- Internal voice packet prioritization
- VLAN support (802.1q & 802.1p frame tagging)
- Configurable ToS bits (DiffServ and IP Precedence support)
- Powering
- Inline power (unused pairs) or
- Direct powering with external wall-plug adapter

### PBX System Requirements

- *UNIVERGE* NEAX 2000 IPS R6.1or higher
- IPPAD Card
- 8 Seat Licenses: Supports up to 8 INASET terminals

# Chapter 4 Equipment Description

## Modules

| Abbrev | Name Code | Remarks |
|---|---|---|
| SN1729 PIMMJ | PIM | Port Interface Module (PIM)<br>Maximum 64 physical ports per PIM.<br>Houses two batteries for protection from short power interruption (for 30 minutes).<br>At maximum configuration, the system consists of eight PIMs, and provides a total of 512 physical ports (64 ports × 8). |
| SN1730 PIMMK | PIM | Port Interface Module (PIM) for Backup CPU System - Maximum 64 physical ports per PIM. Houses two batteries for protection from short power interruption (for 30 minutes).<br>One PIM is required per Backup CPU System. |
| COVER PARTS ASSEM | COVER PARTS ASSEM | Cover Parts Assembly<br>One cover parts assembly is required for each PIM. |
| SN1545 BASERE | BASE/TOP ASSEM | Base/Top Cover Assembly<br>One base and top cover assembly is required for each stack. |
| TOP COVER ASSEM | | |
| SN1731 BATTMH | BATTMH | Battery Module for housing PIM or CS (ZT) backup batteries<br>Houses two pairs of batteries for protection from long power interruption (for 3 hours). |
| SN-1737 IPSMJ | MC | Modular Chassis (MC) with Ringer. Each MC supports 56 physical ports (7 LT/AP card slots). |

# Installation Hardware

| Equipment Name | Function |
|---|---|
| HANGER ASSEM (UL) | Wall Hanger Assembly<br>One HANGER ASSEM is required per PIM for Wall Mounting installation. |
| 19 INCH RACK BRACKET (A) | 19-inch Rack Mounting Bracket Type A<br>One bracket is required for one PIM configuration.<br>One bracket is required for the top PIM of multiple module configuration. |
| 19 INCH RACK BRACKET (B) | 19-inch Rack Mounting Bracket Type B<br>One bracket is required for the bottom module of multiple module configuration. |
| MOUNTING BRACKET | Safety Mounting Bracket<br>Used as an overhead hanger for Floor Standing Installation. Wire, chain or eyebolts are to be locally provided, to secure the bracket. To be installed on the top PIM in four or more modules of stack. It provides 1.1G shockproof construction. |
| I/F BRACKET ASSEM | Inter Frame Bracket Assembly<br>Used to joint the frames in two-frame configuration, for Floor Standing Installation. |
| BASE TRAY ASSEM | Base Tray Assembly<br>One BASE TRAY is required per one frame for Floor Standing Installation of Stationary Equipment. |
| RACK MOUNT KIT (U) | For DM (E) Modular Chassis - Two 19” rack-mounting ears, quantity two 1” ears. One set is included with System Package and each Expansion Module. |

# System Power

| Abbrev | Name Code | Remarks |
|---|---|---|
| PZ-PW121 | AC/DC PWR | Main Power Supply Card: One card is pre-installed per PIM.<br>Input: AC120 V/240 V (50 Hz/60 Hz)<br>Output: –27 V (4.4 A), +5 V (7.2 A), CR (38 mA),<br>+90 V (80 mA) |
| PZ-PW122 | DC/DC PWR | Power Supply Card for Cell Station (Zone Transceiver):<br>Input: –24 V DC<br>Output: –48 V DC (1.7 A)<br>One card per PIM.<br>A maximum of 16 CS (ZTs) backed up by one card. |
| PN-PW00 | EXTPWR | Power Supply Card for DESKCON:<br>• Provides –48 V DC power.<br>• A maximum of four cards per frame (4PIMs).<br>A maximum of three cards per PIM.<br>• Occupies two physical slots width per card. |
| PW131 | PZ-PW131 | For DM (E) Modular Chassis - Main Power Supply without Generator and Message Waiting. One is included in each PIMMJ. |
| PW139 | PZ-PW139 | For DM (E) Modular Chassis - Ring Generator Unit (RGU) mounts on top of the PZ-PW131. One is included in each PIMMJ.<br>Ring Voltage: 85/75Vac; Ring Current: 0.05A |

# Common Control Cards

| Abbrev | Name Code | Remarks |
|---|---|---|
| MP | SPN-CP24D | Main Processor Card<br>Provides LAN control function, system-based Device Registration Server (DRS), built-in FP, Virtual FP, built-in OAI, built-in SMDR on RS-232C, built-in SMDR on IP, built-in PMS on IP, Virtual IPT, Virtual CSH, 33 MHz PCI BUS, Memory (SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), two RS-232C ports, 2-line DAT (Recording duration: Maximum 128 seconds), DK, 4-line PB receiver, Modem for remote maintenance (33.6 Kbps), Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS.<br>Supports MP Program Download (FTP download for stand alone mode only).<br>One card is required per system. |
| MP | SPN-CP27B | Main Processor Card for Backup CPU system<br>Provides LAN control function, system-based Device Registration Server (DRS), Virtual FP, built-in OAI, built-in SMDR on RS-232C, built-in SMDR on IP, built-in PMS on IP, Virtual IPT, Virtual CSH, 33 MHz PCI BUS, Memory<br>(SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), two RS-232C ports, 2-line DAT (Recording duration: Maximum 128 seconds), DK, 4-line PB receiver, Modem for remote maintenance (33.6 Kbps), Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS.<br>Two cards are required per Backup CPU system. One card for active MP [MP0] and another card for stand by MP [MP1]. |
| MP | PN-CP31-E | Main Processor Card (DML)<br>Provides LAN control function, System-based Device Registration Server (DRS), built-in FP, Virtual FP, built-in OAI, built-in SMDR on RS-232C, built-in SMDR on IP, built-in PMS on IP, Virtual IPT, Virtual CSH, 33 MHz PCI BUS, Memory (SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH×1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), two RS-232C ports, 2-line DAT (Recording duration: Maximum 120 seconds), DK, 4-line PB receiver, Modem for remote maintenance (33.6 Kbps), internal Music-on-Hold tone, BUS interface.<br>BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS, MP Program Download (FTP). One card is required per system. |

# Common Control Cards (Cont.)

| Abbrev | Name Code | Remarks |
|---|---|---|
| MP | PN-CP31-D | Main Processor Card (DMR)<br>Provides LAN control function, System-based Device Registration Server (DRS), built-in FP, 33 MHz PCI BUS, Memory (SDRAM 32 MB, Flash ROM 9 MB), TDSW (1024CH × 1024CH), 16-line CFT, PB sender, Clock, 2-line PLO (receiver mode/source mode), one RS-232C port, 4-line PB receiver, internal Music-on-Hold tone, BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing, cable delay timing, monitors I/O Bus and PCM BUS.<br>Supports MP Program Download (FTP download for stand alone mode only).<br>One card is required per system. |
| ETHER | PZ-M606-A | Ethernet Control Card<br>Mounted on MP card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol. Provides Auto Negotiation function.<br>• Set to ON/OFF by the office data setting<br>[For Series 3400 software or later]<br>10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card. |
| FP | PN-CP15 | Firmware Processor Card<br>Provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. When the system consists of three PIMs or more, one each of this card is mounted respectively in PIM2, PIM4, and PIM6. |
| ETHER | PZ-M697<br>(for DML) | Ethernet Control and inter-module BUS interface Card Mounted on PN-CP31-E card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol.<br>Provides Auto Negotiation function.<br>– Set to ON/OFF by the office data setting.<br>10BASE-T/100BASE-TX twisted pair cable is connected directly to this card. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS.<br>Occupies two physical slots width per card on PN-CP31-E card. |
| FP | PN-CP33<br>(for DML) | Firmware Processor Card<br>Provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. When the system consists of three MCs or more, one each of this card is mounted respectively in the MP slot of MC2, MC4, and MC6. |

# Line/Trunk (LT) Cards

The following table shows a summary of the Line/Trunk (LT) cards for *UNIVERGE* NEAX 2000 IPS. The LT cards may be installed in slot 00 to 11 of PIM 0-7, with the following conditions.

- Maximum 64 ports per PIM

| Abbrev | Name Code | Ports | Remarks |
|---|---|---|---|
| LC | PN-8LCAK | 8 | 8-line Analog Line Circuit Card for Single Line Telephones<br>Loop resistance : Maximum 600 Ω<br>Provides Message Waiting Lamp control, momentary open, Line Test functions for each circuit. |
| | PN-8LCAD | 8 | 8-line Analog Long Line Circuit Card for Single Line Telephones; Provides Caller ID Display on analog station.<br>Loop resistance: Maximum 600 Ω<br>Provides Message Waiting Lamp control, momentary open functions for each circuit. |
| | PN-4LLCB | 4 | 4-line Analog Long Line Circuit Card for Single Line Telephones; Provides Caller ID Display on analog station.<br>Loop resistance for PB/DP type:<br>PB : Maximum 1200 Ω<br>DP (20 PPS) : Maximum 1700 Ω<br>DP (10 PPS) : Maximum 2500 Ω<br>Including the internal resistance of the distant office equipment Provides Message Waiting Lamp control, momentary open/reverse functions for each circuit.<br>PZ-PW122 is required. |
| | PN-AUCA | 2 | 2-Line Long-Line card provided with the Power Failure Transfer (PFT) function.<br>Used as 2-Line LLC with PFT or 2-Line DID<br>Loop Resistance: MAX.2500ohms (LLC)<br>Two Reverse Circuits Included<br>On-Board Power (-48VDC) included |
| DLC | PN-8DLCP | 8 | 8-line Digital Line Circuit Card for Dterm Series *i*/E/III, DSS Console, ATTCON, DESKCON<br>[–27 V version, 2-wire type] |
| | PN-8DLCL | 8 | 8-line Digital Line Circuit Card for Dterm Series *i*/E/III, Dterm Series IPK/Elite/Electra Pro, DSS Console, ATTCON, DESKCON<br>[–27 V version, 2-wire type] |
| | PN-4DLCT | 4 | 4 Circuit Digital Line Card (-48V), used for Dterm Series E and Series i Terminals. PZ-PW122 Power Supply Required to provide Long Line function. |

# Line/Trunk (LT) Cards (Cont.)

| Abbrev | Name Code | Ports | Remarks |
|---|---|---|---|
| ILC | SPN-2ILCC | 8 | 2-line ISDN Line Circuit Card<br>Provides a physical interface to ISDN Terminals.<br>Occupies 8 time slots per one card. |
| COT | PN-8COTS | 8 | 8-line Central Office Trunk Card<br>(Loop Start/Ground Start Trunk)<br>Provides loop detection, sending/detecting ground on Tip/Ring wire. |
| | PN-8COTQ | 8 | 8-line Central Office Trunk Card<br>(Loop Start Trunk)<br>Provides loop detection, receiving/sending the Caller ID (CLASS SM) signal. μ-law only. |
| DID | PN-4DIDA | 4 | 4-line Direct Inward Dialing Trunk Card<br>Provides loop detection, sending reverse signal and PB to DP signal conversion.<br>Equipped with –48 V DC-DC on-board power supply. |
| ODT | PN-4ODTA | 4 | 4-Line Out Band Dialing Trunk Card<br>Used as either a 2-wire E&M trunk or a 4-wire E&M trunk.<br>Maximum 4 cards per PIM.<br>When it accommodates in the LT08 to LT11 slots of the PIM, connect speech line circuits 2 and 3 with the CN1 connector on the front side of the card. |
| IP PAD | PN-8IPLA | 8 | 8-channel IP-PAD Card<br>Provides LAN Interface, Packet Assembly/Disassembly to accommodate Legacy Line/Trunk interface. And provides voice compression DSP control functions such as voice compression control, DTMF relay and FAX relay.<br>• Voice compression protocols: G.711 (64 kbps), G.723.1 (5.3 Kbps/6.3 Kbps), G.729a (8 Kbps)<br>• FAX protocol: Pass-through (G.711, G.726)<br>Provides Auto Negotiation function.<br>• Set to ON/OFF by the switch setting<br>When mounting 24DSP (PZ-24IPLA) card, this card can provides up to 32-channel of IP-PAD (When using G.723.1, provides up to 24-channel of IP-PAD).<br>Two cards can be accommodated per built-in FP/FP card, maximum 8 per system.<br>10BASE-T/100BASE-TX twisted pair cable is connected directly to this card. |
| | PZ-24IPLA | 24 | 24-channel DSP Card for 8-channel IP-PAD Card<br>Provides Packet Assembly/Disassembly to accommodate Legacy Line/Trunk Interface. And Provides voice compression DSP control functions such as voice compression control, DTMF relay same as IP-PAD (PN-8IPLA) card.<br>Used to expand the IP-PAD channel up to 32-channels. |

# Line/Trunk (LT) Cards (Cont.)

| Abbrev | Name Code | Ports | Remarks |
|---|---|---|---|
| 8RST | PN-8RSTG | 8 | 8-line PB Receiver Card<br>Used for a PB station line, DID or tie line. |
| 4RST | PN-4RSTF-A | 4 | 4-line Sender Card for Caller ID Display on analog Single Line Telephones PN-4LLCB is required. |
| DAT | PN-4DATC | 8 | 4-line Digital Announcement Trunk Card<br>Recording duration: Maximum 120 seconds<br>Occupies 8 time slots per one card. |
| DK00 | PN-DK00 | 0 | 8-circuit External Relay Control/External Key Scan Card. Provides the above-mentioned control functions on a per circuit basis. |
| TNT | PN-TNTA | 4 | 2-line Tone/Music Source Interface Card<br>Used for BGM or Music on Hold.<br>Provides two jacks for an external tone/music source. |
| VCT | SPN-4VCTI-B | 4 | 4-channel CODEC Card for IP Trunk Card<br>Provides voice compression DSP control functions such as voice compression control, DTMF relay and FAX relay.<br>• Voice compression protocols: G.711 (64 Kbps), G.723.1 (5.3 Kbps/6.3 Kbps), G.729a (8 Kbps)<br>• FAX protocol: T.30<br>Used together with IP Trunk (PN-IPTB) card.<br>Four cards can be accommodated per IP Trunk (PN-IPTB) card, maximum 32 per system. |
| | SPN-4VCTI-B (H.323) | 4 | 4-channel voice translator CODEC card for H.323 includes BUS cable. Voice compression: G729A /G723 /G711. Up to qty 3 are used with one SPN-IPTB-B (H323) to provide 12 IP Trunk Channels. Max 3 VCT's per each IPT, max 32 VCT's per system. VCT cards connect together and to IPT via front connected bus cable. One cable is included with each VCT card. |
| CSI | SPN-4CSIA | 16 | 4-line Zone Transceiver Interface Card<br>Used to interface with the ZT, based on U-interface.<br>Maximum four ZTs can be connected per CSI card.<br>Occupies 16 time slots per one card. |
| Optical Modem | PN-M10 | 0 | Optical Fiber Interface Card<br>Provides optical fiber interface for T1/E1 Digital Trunk Interface (1.5 M/2 Mbps)<br>Line length : 10 km (6.2 miles) or less<br>Line coding : CMI |
| PFT | PZ-8PFTB | 0 | 8-line Power Failure Transfer Card<br>To be mounted in PFT slot of PIM.<br>One card per PIM. |
| | PZ-4PFTA | 0 | 4 Circuit Power Fail Transfer (PFT).<br>Mounts in each IPS DM (E) IPSMJ (includes mounting hardware and POWER CA-PFT.) |

# Application Processor (AP) Cards

The AP cards may be installed in slot 00 to 11 of PIM 0-7, with the following conditions:

- Maximum 24 cards per system
- Maximum 256 ports per system

| Abbrev | Name Code | Ports | Remarks |
|---|---|---|---|
| DTI | SPN-DTA (PRT)-A | 25 | 1.5M Digital Trunk Interface<br>Used for either T1 or PRI. Used for T1 Interface - 24 channel T1 (Max 10 T1, using 24 APTS, Max of 240 DTI Ports); Used for ISDN PRI Trunk Interface with D-Channel Handler for 24 channel ISDN Primary Rate (23B+D) included (uses 25 APTS). Lower/Upper AP Hwy.<br>RoHS compliant |
| | SPN-30DTC-UA | 30 | 2M Digital Trunk Interface<br>Uses 31 Application Time Slots. Max of 120 DTI Ports w/this Interface. AP Number 4-15 & 20-31 Lower AP Hwy Applications; 2M AMI (30ch) T1, CCIS (30B+C) requires SPN-SC00, CCH-D (AP) for each CCIS Link. |
| PRT | SPN-DTA (PRT)-A | 25 | 1.5M Digital Trunk Interface<br>Used for either T1 or PRI. Used for T1 Interface - 24 channel T1 (Max 10 T1, using 24 APTS, Max of 240 DTI Ports); Used for ISDN PRI Trunk Interface with D-Channel Handler for 24 channel ISDN Primary Rate (23B+D) included (uses 25 APTS). Lower/Upper AP Hwy. RoHS compliant |
| CCT | SPN-DTA (CCT)-B | 25 | Can be used as either 24 channel CCIS span or Common Channel Handler for CCIS used with T1. Card is a 24 channel CCIS Interface with C-Channel Handler included. 1.5M Digital Trunk Interface using 25 Application Time Slots. Max 8 CCIS Links. Lower/Upper AP Hwy. Used for CCIS with Point to Point T1. RoHS compliant |
| BRT | SPN-4BRTA-H | 8 | 4 circuit Basic Rate (2B+D) Interface Trunk Card Accommodates 4 two-channel PCM digital lines. Both Point-to-Point and Point-to-Multi Point are supported. Lower/Upper AP Hwy. RoHS compliant. |
| SIG | SPN-DTA (QSIG)-A (AP) | 25 | QSIG combo card with Protocol Handler and 24DTAC. Supports the following Supplementary Services (SS): SS-CLIP, SS-CLOP, SS-CLIR, SS-CONP, SS-CNIR and SS-CONR. Implementation of ID-SS (ETS 300 173) and Name ID-SS (ETS 300 238) is based on basic specifications of ETS 300 172 by ETSI at the Q reference point between PBX's |
| CFT | SPN-CFTC | 32 | 32 Party Digital Conference<br>(Group Call/Meet Me)<br>• Group Call Automatic<br>• Group Call Broad Cast<br>• Group Call Two Way Calling |

# Application Processor (AP) Cards (Cont.)

| Abbrev | Name Code | Ports | Remarks |
|---|---|---|---|
| CCH | SPN-DTA (CCT)-B | 1 | Common Channel Handler for CCIS<br>Can be used as either Common Channel Handler for CCIS used with T1 or 24 channel CCIS span. Card used as Common Channel Handler for CCIS is used with T1 and is 1 per Destination Site, Max 8 CCIS Links. Used for CCIS with Point to Point T1. Lower/Upper AP Hwy.<br>RoHS compliant |
| ICH | SPN-SC03B 8ICH | 4 | D-Channel Handler (8ICH) with Flash ROM Uses 4 Application Time Slots, Required for BRI Station Side (One SC03 for every four PN- 2ILCA cards) AP Number 4-12, Lower AP Hwy |
| CSH | SPN-SC03B 8CSH-C | 4 | Wireless ZT Handler (CSH) with Flash ROM Supports both S-Interface and U-Interface. Supports Short Text Message and Dukane Nurse Call system. Uses 4 Application Time Slots. This card is required to expand the number of PS from 256 to 512. |
| AP00 | SPN-AP00B MRC-C | 2 | Provides four RS-232C ports and is used for Call Accounting (SMDR) both 2400 and 1400 formats, Property Management System (PMS), Hotel-Printer and Message Center Interface (MCI), CCIS Centralized SMDR/MCI; PZ-M537 expands Buffer (1600 - 27000). RS-232C: Synchronous One Port, Asynchronous Three Ports. AP Number 4-15 & 20-31 Lower AP Hwy |
| | SPN-AP00B MRC-H | 2 | Application Processor card<br>One AP00 card per system. Functions Include: SMDR 2400 Format, SMDR 2400 Extended Format, Station to Station SMDR, MCI, DND Group, Executed Task Print Out for MP-PMS and H/M Printer for<br>MP-PMS. (Does not support 1400 SMDR format) RoHS compliant |
| EX MEM | PZ-M537 | 0 | Expansion Memory for AP00-B |

**Note 1:** *The Expansion Memory is the on-board sub-memory card for the AP00-B. It provides additional memory capacity.*

| System Capacity (SMDR call record) | w/o EXPMEM | with EXPMEM |
|---|---|---|
| SPN-AP00B MRC-C(AP) | 1600 | 27000 |
| SPN-AP00B MRC-H(AP) | 2620 | 23580 |

# Application Processor (AP) Cards (Cont.)

| Abbrev | Name Code | Ports | Remarks |
|---|---|---|---|
| 4RST | SPN-4RSTBA-B | 4 | 4 Circuit MF Sender for T1 ANI<br>Supports T1 Feature Group D (number only).<br>Uses 4 Application Time slots, AP Number 4-15 & 20-31 Lower AP Hwy. Max: 4-Cards per system including PN-8RSTA/PN-4RSTF/PN-4RSTG and SPN- -4RSTB-911(AP). |
| | SPN-4RSTBA-911 | 4 | 4 Circuit MF Receiver for Enhanced 911 Requires CAMA Trunks. Uses 4 Application Time Slots, AP Number 4-15 & 20-31 Lower AP Hwy. Max: 4 - Cards per system including PN-8RSTA/PN-4RSTF/PN-4RSTG and SPN-4RSTB-B (AP). |
| Caller ID Class | SPN-4RSTC-A | 4 | 4 Circuit FSK Receiver for Analog Trunk CALLER ID Max: 4-Cards including RSTB and RSTB-911AP Number 4-12, Lower AP Hwy, Used with PN-8COTS |
| IPT | SPN-IPTB-A | 0 | 16 Channel IP Trunk Card.<br>(G.729a. Payload 40ms), VoIP P-P and PMP. Max.1/PIM, Max 8 cards per system, Max 127 channels per system. Has built-in CCH supporting up to 255 nodes. Has one RJ45 connectors for 10/100 Ethernet/Fast Ethernet to router. Up to quantity 4 SPN-4VCT are connected to one SPN-IPT. |
| | SPN-IPTB-B(AP) (H.323) | 0 | VoIP H.323 IP Trunk card. Supports up to 12 channels. (G.729a. PAYLOAD 40ms). Also requires CCIS and IPT key per IPTB and NEC GK1000 software. Has one RJ45 connectors for 10/100 Ethernet/Fast Ethernet to router. Up to quantity 3 of SPN-4VCT are connected to one SPN-IPT. |
| Wireless Roaming | SPN-AP00B DBM-C | 2 | Database Memory Card for WCS IP Roaming. - Required for Wireless Roaming and OAI applications using free Location Facility (FLF)", PBX receives a copy of the database from OAI server by FLF function. The PBX stores this database on the SPNAP00B DBM-B card, where it can refer to the data and convert ID codes to station numbers. Expands number of Wireless PS handsets to 512. |
| | SPN-SC01 DCH-Q | 1 | DCH Card for Wireless Roaming (Q.931.a)<br>1 Card per system.<br>Installed With PN-24DTA-A / 30DTC.<br>AP Number 4-15 & 20-31 Lower AP Hwy |

# Chapter 5 Feature Description

## Business/Hotel Motel Features

### Account Code
This feature, when used with Station Message Detail Recording (SMDR), allows station users and Attendants to enter a cost accounting or client billing code (up to 16 digits) into the system.

### Add-On Module
This feature allows the Add-On Module to be combined with a Multiline Terminal when there are insufficient line or trunk keys provided at the Multiline Terminal. When the Add-On Module unit keys are programmed as line/trunk keys, the additional 25 lines/trunks and the existing lines/trunks set for the Multiline Terminal can be accessed directly (maximum of 49 lines/trunks). The station speed dialing function can be assigned for all keys on the Add-On Module unit. Also, one of the last 3 keys can be used as a Day/ Night change key.

### Alarm Indications
Faults are indicated by the Major/Minor (MJ/MN) lamps located on the AC/DC Power Supply and, optionally, an external alarm display unit. Station Application not applicable.

### Alphanumeric Display
The LCD on Multiline Terminals is used to provide alphanumeric information including clock/calendar and call processing information. Station Application All Multiline Terminals with an LCD display.

### Analog Port Adapter
This feature allows an Analog Port Adapter unit combined with a legacy Multiline Terminal to connect to an analog terminal such as an analog telephone, Modem, and PC with built-in Modem. There are two communication modes for the terminal connected via the Analog Port Adapter as shown below:

1. Single Port Mode
   A Multiline Terminal and an analog terminal share the same port. In this mode, the Multiline Terminal and the analog terminal cannot be used simultaneously.
2. Dual Port Mode
   A Multiline Terminal and an analog terminal use different ports. In this mode, the Multiline Terminal and the analog terminal can be used simultaneously.

### Announcement Service
This feature allows station users to record messages on Digital Announcement Trunk (DAT) cards. When a station user dials the feature access code for this feature, the user receives the corresponding message from the system. Also Announcement Service can be used to provide a voice message in the following cases;

- An incoming C.O. line/Tie line call has been transferred and encounters a busy or no answer condition
- An incoming DID line/Tie line call has been terminated to a station and encounters a busy or no answer condition
- Internal Recorded Message in place of Music on Hold
- Night Announcement

### Answer Key
An Answer Key is provided on all Multiline Terminals. The Answer Key can be used to answer incoming calls on outside lines, and primary or secondary extensions. When the Answer Key is used to answer an incoming call with a call in progress, the first party is placed on hold and the second party is connected. If the Answer Key is depressed while in a three-party call, the user can alternate between each party and a Broker's Call is established.

### Attendant Assisted Calling
This feature allows a station user to ask an Attendant for assistance in originating a call. Three methods are available: non-delay, delay, and passing dial tone.

### Attendant Camp-on
This feature permits the Attendant to hold an incoming call in a special mode when the desired station for the transfer is busy. The Attendant sends a Camp-On tone to the busy station. When that station becomes idle, it is automatically alerted and connected to the waiting party.

### Attendant Console
### SN716 DESKCON
The Attendant Console (SN716 DESKCON) operates on a switched-loop basis with a maximum of 6 Attendant loops terminating at each console on the associated Interface card. The Attendant uses these loops for answering, originating, holding, extending, and reentering calls. When Attendant loop release is used, the number of loops is effectively increased to a maximum of 12 for each console.

#### Attendant Called/Calling Name Display
This feature provides a display of the calling/called party's name on the Attendant Console LCD for Attendant Called/Calling Name Display. On attendant-to-station calls, the LCD display the name assigned to the primary extension of the station. On attendant-to-trunk calls, the LCD displays the name assigned to the trunk route of the trunk.

#### Attendant Called/Calling Number
This feature provides a display of the station number and station name on the Attendant Console during an Attendant-to-station connection. During an Attendant-to-trunk connection, the same display shows the trunk route designation and a trunk identification code (4 digits).

#### Attendant Call Selection
This feature allows assignment of keys on the Attendant Console to particular types of trunk routes (such as WATS or FX) and particular types of service calls (such as Attendant recalls, intercept calls, etc.). LEDs indicate the type of incoming call and pressing the associated key allows the Attendant to answer the calls in any order.

#### Attendant Console Lockout - Password
This feature allows the Attendant Console to be set into a lockout mode. This disables the console from originating or receiving calls and setting or resetting service features. To return the Console to its manual operating condition a password is required.

#### Attendant Do Not Disturb Setup And Cancel
The Attendant has the ability to enter and remove individual stations from Do Not Disturb (DND). Additionally, the Attendant can set one preassigned group of stations into, or out of, Do Not Disturb.

### Attendant Interposition Calling/Transfer
This feature allows any Attendant to directly converse with another Attendant and also allows Attendants to transfer calls from their console to another Attendant's console in systems where Multiple Console Operation has been provided.

### Attendant Lamp Check
This function is used to check the status of keys, lamps, and LCDs mounted on the Attendant Console to verify that various operations of the Attendant Console are functioning normally. The check is done by a preset procedure.

### Attendant Listed Directory Number
This feature provides a display of the Listed Directory Number on the Attendant Console when the operator has answered a Listed Directory Number call.

### Attendant Loop Release
This feature allows an Attendant Console loop to become available for a second call as soon as the Attendant has directed the first call to a station, even if that station does not answer.

### Attendant Programming
This function is allowed only for the Attendant Console and is used to execute DISA code set up, speed dial programming, and system clock set up operations.

### Attendant Training Jacks
The SN716 DeskCon provides two headset/handset jacks on the comnsole, for trining operations. Normal call handling procedures apply. When jacks are used for training, both handsets can be used for listening and talking.

### Audible Indication Control
This feature allows the Attendant to adjust the volume of audible indications received at the Attendant Console.

### Call Processing Indication
This feature provides visual indications of all calls being processed or awaiting processing at the Attendant Console.

### Call Queuing
This feature provides the Attendant the ability to handle a series of exchange network calls in the order of their arrival, (first in, first out) thereby eliminating unnecessary delays.

### Call Splitting
This feature allows the Attendant to confer privately with one party on an Attendant handled connection without the other party overhearing.

### Call Waiting Display
This feature provides a visual indication to the Attendant when one or more calls are waiting to be answered.

### Common Route Indial
This feature allows assignment of incoming DID calls to different Attendant Call Selection keys based on the last 4 digits dialed into the system. Up to eight individual Listed Directory Numbers can be assigned in system programming. When an incoming call to any of these trunks is received, an Attendant Call Selection key will flash and the LCD display will indicate the Listed Directory Number associated with that trunk route.

### Dialed Number Identification Service (DNIS)
This feature provides a display of the company name on the Attendant Console when the Attendant has answered a Listed Directory Number or a Tie Line call.

### Incoming Call Identification
Incoming calls are identified by various means. Refer to Attendant Called/Calling Number, Attendant Call Selection, Attendant Source Key, Attendant Listed Directory Number and Common Route Indial Features and Specifications.

### Individual Trunk Access
The Attendant Console is provided with the ability to access each individual trunk by dialing an associated identification code. This allows detection of faulty trunks during regular testing or after complaints. The Customer Administration Terminal (CAT) or Maintenance Administration Terminal (MAT) has the capability to then busy out the trunk until repair is made.

### Multi-Function Key
This feature allows the top row of keys on the Attendant Console to perform and display multiple functions in accordance with the status of call processing.

### Multiple Console Operation
This feature allows more than one Attendant Console to operate within the same system.

### Pushbutton Calling - Attendant Only
This feature permits an operator to place all calls over Dual-Tone, Multi-Frequency (DTMF) lines from the pushbutton keypad on the Attendant Console.

### Serial Call
This feature is activated by the Attendant when an incoming calling party wishes to speak with more than one internal party. When the internal station subsequently disconnects from the Central Office line call, the Central Office party automatically rings back to the same Attendant.

### Time Display
This feature provides a digital time display on the Attendant Console LCD. Time is constantly displayed on the Attendant Console LCD. The clock display of the Attendant Console is synchronized with the clock in the sysytem.

### Trunk Group Busy Display
A visual indication is supplied to the Attendant when all trunks in a particular trunk group are busy.

### Unsupervised Trunk-to-Trunk Transfer By Attendant
This feature allows an Attendant to transfer an incoming or outgoing call on one trunk to an outgoing trunk and exit the connection before the called party answers.

### Attendant Delay Announcement
This feature provides an announcement, via a Digital Announcement Trunk Card, to external calls that are not answered by the attendant within a predetermined time.

### Attendant Lockout
This feature denies an Attendant the ability to reenter an established trunk or station connection without being recalled by that station after the call is put in consultation hold.

### Attendant Overflow
When an incoming call, which has terminated from a trunk to the Attendant Console, remains unanswered after a predetermined time period, this feature provides a change to Night Service for that particular trunk, or an overflow to an outside trunk.

### Attendant Override
This feature permits an Attendant to enter a busy connection (station or trunk) using the Attendant Console. When this feature is activated, a warning tone is sent to the connected parties after which, they are connected with the Attendant in a three-way bridge.

### Authorization Code
An Authorization Code is a numerical code which will temporarily change a station's Class of Service to a Class of Service assigned to that Authorization Code. This new Class of Service allows access to trunks, dialing patterns, and/or features that would otherwise be restricted.

### Automated Attendant
This feature allows the system to answer incoming trunk calls. The system will supply a message and/or dial tone to the caller. The caller can then dial the desired extension number and be directed to that station.

### Automatic Call Distribution (ACD)
The Automatic Call Distribution (ACD) feature permits incoming calls to terminate to a prearranged group of stations. Calls are distributed in the order of arrival to idle terminals within the group, based on which terminal has been idle the longest period of time. Stations may log on/log off from the ACD group. Supervisor stations may monitor conversations of agents.

#### Busy In/Busy Out - ACD
This feature allows an agent in an ACD group to log their station onto or off of the group. This allows the system to control whether a call directed to the pilot number of the ACD group goes to that station or not. This prevents incoming calls from being directed to stations at which no agent is available.

#### Call Waiting Indication - ACD
This feature provides a visual indication when an incoming call to an ACD group is placed in queue, due to an "all agents busy" condition. On external relay controlled indicator or an LED on a Multiline Terminal can be used to provide Call Waiting Indication.

#### Delay Announcement - ACD
This feature allows the system to provide a recorded announcement to an incoming caller placed in queue to an ACD group. A single announcement, or two separate announcements, can be provided.

### Hunt Past No Answer - ACD
This feature allows calls targeted at an ACD group to hunt past an agent's station, after a no answer condition, if the agent forgets to log off of the group and the agent is unable (or not available) to answer the call.

### Immediate Overflow - ACD
This feature allows a call directed to an ACD group to immediately overflow to another ACD group, upon encountering an "all agents busy" condition.

### Priority Queuing - ACD
This feature allows the system to prioritize incoming calls by trunk route and on a per station basis, when the call enters an ACD queue. When a call is considered as a priority, it is placed at the beginning of the queue.

### Queue Size Control - ACD
On incoming DID/Tie line calls, the system can be assigned a threshold which limits the number of calls in queue. When the queue size threshold is exceeded, incoming callers are connected to busy tone.

### Silent Monitor - ACD
This feature provides the ACD group supervisor with the ability to monitor a call to an ACD agent. The silent monitor function gives no indication (as an option) to either the agent or the calling party.

## Automatic Call Distribution (ACD) with Management Information System (MIS)
The Automatic Call Distribution (ACD) with (MIS) provides a management information system to be used in conjunction with the built-in ACD features of the system. The MIS incorporates a supervisor's terminal for real-time monitoring of agent activity, amber and red alarms, and hard-copy summary reports.

## Automatic Camp-on
An incoming Direct Inward Termination (DIT) call which has been terminated to a busy station can be Camped-On automatically. When the busy station becomes idle, the station is automatically called and connected to the camped on incoming trunk call.

## Automatic Change to Daylight Saving Time
This feature allows the 2000 IPS system clock to automatically change from standard time to daylight saving time, and vice versa. Schedule to change to/from daylight saving time is programmed by system data programming.

## Automatic Number Identification (ANI)
This feature receives the calling subscriber's number automatically sent from T1 network using MF signaling and displays the calling number on the LCD of a Multiline Terminal and an Attendant Console.

## Automatic Recall
This feature works as a timed reminder. When a call remains on Hold, Camp-On or ringing unanswered for a fixed interval after being transferred, the station that initiated the hold, transfer, or Camp-On is automatically alerted.

### Automatic Wake-up
This feature allows the system to be programmed to automatically call guest rooms or administration stations at specified times. Upon answering, the guest is connected to a recorded announcement or music source. A printout of unanswered or blocked Automatic Wake-Up attempts for each guest room is provided using the Hotel/Motel printer.

### Background Music
Background Music can be provided on a dial-up basis over legacy Multiline Terminal speakers. Incoming voice announcements, ringing and recalls override Background Music. Up to 10 music programs can be offered.

### Back Up CPU
*UNIVERGE* NEAX 2000 IPS provides a dual CPU system with two MP cards. When Emergency Notification from hardware is detected, the changeover from an active MP card to a standby MP card will occur. If the active MP card becomes out of order for any reason, the standby MP card starts up automatically. The standby MP card employs a Cold Standby System that will restart initialization by the changeover from the active MP card.

### Bandwidth Control
This feature allows assigning an available bandwidth threshold for VoIP Traffic within a location and between locations, and to restrict Outgoing/Incoming Calls when the VoIP Traffic exceeds the threshold. The location is a group of VoIP devices [IP Enabled Dterm, IP PAD, or Peer-to-Peer IP Trunks (built-in IP Trunks)], that have the same VoIP communication parameter (such as CODEC Selection List and ToS field) values assigned. When the VoIP Traffic over CCIS exceeds the threshold, the call can be routed to Legacy Trunks (TDM Network). When exceeding the threshold, the system can store fault information and provide external alarm indication.

### Boss / Secretary Calling
A secretary with a Multiline Terminal can use an appearance of the boss' extension to screen calls for that extension, and announce and/or transfer calls to that extension. Additionally, the secretary can call the boss during a busy condition and send a Message Waiting Indication to the boss' station.

### Broker's Call
This feature allows a Multiline Terminal or Single Line Telephone user to alternate between two parties, talking to one party while the other party remains on Hold on the same line. The Multiline Terminal user uses the TRF or ANS key to alternate between the two parties. The Single Line Telephone user uses the Hold feature to alternate between the two parties.

### Call Back
This feature allows a calling party to set an automatic Call Back when a busy or no answer condition is encountered. When the busy station becomes idle, the station that set the Call Back will be called. In case of Call Back no answer, the Call Back to the setting station is initiated immediately after the called station goes on hook after making a call or accessing a feature.

## Call Forwarding

Call Forwarding allows calls directed to a station to be routed to another station, an Attendant, an outside number or voice mail equipment. The types of Call Forwarding provided are:

- Call Forwarding - All Calls
- Call Forwarding - Busy Line
- Call Forwarding - No Answer
- Call Forwarding - Destination
- Multiple Call Forwarding - All Calls
- Multiple Call Forwarding - Busy Line
- Multiple Call Forwarding - No Answer
- Split Call Forwarding - All Calls
- Split Call Forwarding - Busy Line
- Split Call Forwarding - No Answer
- Attendant Call Forwarding Setup and Cancel
- Call Forwarding - Override
- Group Diversion

### Attendant Call Forwarding Set-up and Cancel

All of the various types of Call Forwarding can be set up or canceled from both Attendant Consoles.

### Call Forwarding - All Calls

This feature allows all calls directed to a particular extension to be rerouted to an alternate destination, regardless of the busy or idle status of the extension. Call Forwarding - All Calls can be set by an Attendant Console, the individual station user, a Multiline Terminal with a secondary appearance of the station's extension, or from another station (which can program itself to be the destination of the rerouting).

### Call Forwarding - Busy Line

This feature permits a call to a busy extension to be routed to a pre-designated station, Attendant Console, or voice mail equipment. Call Forwarding - Busy Line can be set or canceled by an Attendant Console, the individual station user, or a Multiline Terminal with a secondary appearance of the station's extension.

### Call Forwarding - No Answer

This feature reroutes calls to extensions which do not answer. These calls can be rerouted to another station, an Attendant Console or voice mail equipment. Call Forwarding - No Answer can be set by the individual station user, an Attendant Console, or by a Multiline Terminal with a secondary appearance of the station's extension.

### Call Forwarding - Destination

This feature allows a station (A) user to set Call Forwarding - All Calls from another station (B) within the system, to the user's station (A).

### Multiple Call Forwarding - All Calls

When a forwarded call is rerouted to a station that has also set a Call Forward, the call can be forwarded to another station. A call can be forwarded up to a maximum of five times, as specified in system programming.

### Multiple Call Forwarding - Busy Line
This feature permits a call to a busy station to be forwarded multiple times to a designated idle station.

### Multiple Call Forwarding - No Answer
This feature permits a call to an unanswered station, the ability to be forwarded multiple times to a designated station that does not have Call Forwarding - No Answer set or to the Attendant Console.

### Split Call Forwarding - All Calls
This feature allows all internal and external calls to a busy extension to be rerouted to different destinations individually, regardless of the busy or idle status of the extension. According to the type of incoming call (Station, C. O. Line, Tie Line, or a call terminated from internal office or via CCIS); Call Forwarding or Split Call Forwarding can be selected.

### Split Call Forwarding - Busy Line
This feature allows internal and external calls to a busy extension to be rerouted to separate destinations. Destinations may be an internal station, Attendant Console, or voice mail. And according to the type of a caller (Station/C.O. Line/Tie Line) or a call terminated from internal office or via CCIS, Call Forwarding or Split Call Forwarding can be selected.

### Split Call Forwarding - No Answer
This feature allows internal and external calls to a busy extension to be rerouted to separate destinations. Destinations may be an internal station, Attendant Console, or voice mail. And according to the type of a caller (Station/C.O. Line/Tie Line) or a call terminated from internal office or via CCIS, Call Forwarding or Split Call Forwarding can be selected.

### Call Forwarding – Logout (DtermIP)
This feature allows a call terminated to an IP Station in Logout Status to be forwarded to a designated Station, Outside Number, Attendant Console or Digital Announcement Trunk (DAT). This feature is also
applicable to the IP Stations that the LAN Cable is pulled out of or the power is off.

### Call Forwarding - Override
This feature allows the call forward destination station to call the station which set call forward. The call forward setting will be ignored.

### Group Diversion
This feature allows all calls terminated to an extension that are not answered within a predetermined time to be forwarded to a pre-designated station.

## Call Park
This feature enables a station user or attendant to place a call into pre-designated Call Park locations. The station user or attendant is then free to process other calls. This feature is available system wide and for individual tenants.

### Call Park - System
When a call is parked by Call Park-System, the call can be retrieved from Call Park by any station in the system.

### Call Park - Tenant
When a call is parked by Call Park - Tenant, the call can be retrieved from Call Park-Tenant by any station within the tenant from which the call was originally parked.

### Call Pickup

This feature enables a station user to answer any call directed to another station, to a station within the user's own Call Pickup Group, or to a station within a different Call Pickup Group. Three Call Pickup methods are available: Call Pickup - Direct, Call Pickup - Group, and Call Pickup - Designated Group.

#### Call Pickup - Direct

This method permits a station user to pickup a call to any other station in the system by dialing a specific Call Pickup feature access code and the number of the called extension.

#### Call Pickup - Group

This method permits a station user to answer any calls directed to other extensions in their preset pickup group by dialing a Call Pickup - Group feature access code.

#### Call Pickup - Designated Group

This method permits a station user to answer an incoming call directed to another group by dialing the Call Pick-up - Designated Group feature access code and any station within the group to which the ringing station belongs.

### Call Redirect

Without answering incoming calls or held calls that terminate to the line keys of a Multiline Terminal, the calls can be transferred to a pre-programmed station or Voice Mail System. Two transferring destination number can be designated per tenant, in system data programming. This feature can be used together with the Caller ID Display feature.

### Call Transfer

This feature permits a station user to transfer a call to another station in the system directly, or with assistance from the attendant.

#### Call Transfer - All Calls

This feature permits a station user to transfer incoming or outgoing calls to another station within the system without attendant assistance.

#### Call Transfer - Attendant

This feature permits a station user, while connected to an internal or outside call, to signal the Attendant and have the Attendant transfer the call to another station within the system or to an outside connection.

### Caller ID Class

This feature receives the calling subscriber’s name and number sent from a public network using a MODEM signal and displays the name or number on an LCD of a Multiline Terminal and Attendant Console.

### Caller ID Display

Without answering incoming calls or held calls which terminate to the line keys of a Multiline Terminal, the calling party's information can be confirmed by the indications on the LCD. When a station is in conversation and the CID display key is pressed, the following information will be displayed on the lower line of the LCD.

### Caller ID Station
This feature enables a user to connect analog telephones with Caller ID display function, and provides the calling party's number and name on the display without answering incoming calls.

### Camp-on
This feature provides selected stations or outside calls with Camp-On capability to a busy internal station. Two Camp-On methods are provided. The call waiting method allows a station or an outside party to camp itself on to a busy station. The transfer method allows a transferred outside call to be camped-on to a busy station.

### Centrex Compatibility
A combination of features allows full integration of the *UNIVERGE* NEAX 2000 IPS with Centrex service.

### Check In / Check Out
When this feature is activated, the following operations occur:

- Check In
- Room Cutoff is cleared.
- Check Out
- Room Status printout is supplied.
- Do Not Disturb is reset.
- Room Cutoff is set.
- Message Waiting is reset.
- Automatic Wake Up is cleared.

### CID Call Back
When an Incoming Call is terminated from a Station and Trunk with Caller ID (Calling Number Information), and the Called Station does not answer, the Calling Number is registered to the system memory and MW Lamp is lit. After the Calling Number is registered, by operating from a Station, Confirmation, Delete, or Call Back is available. When the Called Station answers, the Calling Number is also registered to the Last Number Redial Memory. The Station can search and call back to that number by the operation of the Last Number Redial/Stack Dial feature.

### CID Call Routing
This feature allows designating a call terminating system based on the Calling Party Number received from the network.

### Class of Service
This feature permits all stations to be assigned a Class of Service in accordance with the degree of system use desired. The Class of Service is used to assign restrictions for trunk access and feature access.

### Code Restriction
This feature allows the *UNIVERGE* NEAX 2000 IPS to be programmed to restrict outgoing calls according to specific area and/or Central Office codes. This restriction is controlled on the basis of a three-digit area code or six-digit area and office code numbering plan.

### Conference (Three/Four Party)
This feature provides a station user the ability to add-on another party (trunk or station) to a call already in progress. Single Line Telephone users can add up to one additional party and Multiline Terminal users can add up to two additional parties.

### Conference (Eight Party)
This feature permits a station user or Attendant (conference leader) to establish a Conference among as many as eight parties (including the Conference leader).

### Conference (32 Party)
This feature permits a Station User (PS, Multiline Terminal, Single Line Telephone), Attendant, or a Trunk Party to establish a Conference among as many as 32 Parties (including the Conference Leader). Two Conference methods are available: Group Call and Meet-Me Conference.

#### Group Call
This feature enables a Station User (PS, Multi-line Terminal, Single Line Telephone) within the system or a Trunk Party to establish a Conference among as many as 32 Parties. It also enables a Station User to page a maximum of 31 Parties simultaneously, excluding the Conference Leader. Three Group Call methods are available: Group Call - Automatic Conference, Group Call - Broadcasting, and Group Call - 2-Way Calling.

#### Group Call - Automatic Conference
This feature enables a Station User to establish a Conference among as many as 32 Parties. From a station or Attendant, a maximum of 31 Stations/Trunks can be paged simultaneously except the Conference Leader. The Paged Stations/Trunks are assigned to the simultaneous Paging Groups as participants by the System Data beforehand.

#### Group Call - Broadcasting
This feature enables a Station User to page a maximum of 31 Parties simultaneously except the Group Call Leader. After Paged Parties answer, the leader can speak to the Paged Parties (the Paged Parties only hear the leader's voice). The Paged Stations/Trunks are assigned to the simultaneous Paging Groups as participants by the System Data beforehand.

#### Group Call - 2-Way Calling
This feature enables a Station User to page a maximum of 31 Parties simultaneously except the Group Call Leader. After one of the Paged Parties answers, paging becomes 2-Way Calling between the leader and the first answered party and automatically stops paging other parties. The Paged Stations/Trunks are assigned to the simultaneous Paging Groups as participants by the System Data beforehand.

#### Meet-Me Conference
This feature enables Station Users (PS, Multi-line Terminal, Single Line Telephone) within the system, Attendants, or Trunk Parties to join a Conference as many as 32 Parties by dialing a specific Access Code. The Conference participants are automatically connected to the Conference Trunk an alert tone signals new arrivals. Conference participants may call in at preset time or may be directed to do so by a Conference coordinator once all the participants have joined the conference leader can lock out the conference..

### Consecutive Speed Dialing
For Speed Dialing, all digits are registered as a Speed Dialing Code. In the case of Consecutive Speed Dialing, the common portion of the number is registered as a speed calling code, and the remaining digits of each number are dialed by each individual calling station or by using a Station Speed Dial key on a Multiline Terminal.

### Consultation Hold
This feature permits a station user to hold any incoming or outgoing CO call, tie line call, or any intra-office call while originating a call to another station user within the system.

### Customer Administration Terminal (CAT)
In addition to the Maintenance Administration Terminal (MAT), programming of the *UNIVERGE* NEAX 2000 IPS can be done from selected Multiline Terminals with LCD. The designated Multiline Terminals can be placed in program mode, and system data can then be changed. To prevent unauthorized changes, password levels are assigned, providing authorization for access to certain areas of programming and denying access to others.

### Data Line Security
This feature allows line circuits that are used for data transmission to be protected from interruptions such as Attendant Camp-On, Executive Override, and Attendant Override.

### Delayed Hotline
When a station user goes off-hook and waits for a certain period of time without dialing operation, the station user is automatically routed to a specified Hotline station or an attendant. If the station user dials a number before the preprogrammed time, the station user can make a call as usual.

### Delayed Ringing
This feature enables trunks and station lines to ring immediately at the terminating station, but also, after a programmable period of time has elapsed, to ring at secondary Multiline Terminals with that trunk or line appearance.

### Diagnostics
To assist maintenance personnel, the *UNIVERGE* NEAX 2000 IPS provides diagnostic capabilities such as fault code generation, device status information and alarm information recording which can be accessed from the Maintenance Administration Terminal (MAT) or Customer Administration Terminal (CAT).

### Dial by Name
This feature allows a Multiline Terminal user to search for a desired number by name. The number and name are registered in the system and they are shown on Multiline Terminal LCD. The Multiline Terminal user can search for the desired number by name using up or down soft keys. When the Multiline Terminal user finds the desired number, the call can be originated by pressing the Line/Trunk key or going off hook.

### Dial Conversion
The system can be assigned to use rotary Dial Pulse (DP) or Dual Tone Multi-frequency (DTMF) trunks and stations. This feature provides for the repeating of digits dialed by the station user onto the C.O. trunks.

### Direct Data Entry
This feature allows a maid or other hotel personnel to enter numeric data to the Property Management System (PMS), using the guest room station for entry through dial operation. The same numerical data can be output to a Hotel/Motel Printer by system data programming.

### Direct Digital Interface
This service feature provides the capability to connect trunks from the *UNIVERGE* NEAX 2000 IPS directly to T1 carrier links using either a private or public network.

### Direct Inward Dialing (DID)

This feature provides for incoming calls from the exchange network (except FX or WATS) to reach any station within the system without attendant assistance.

#### DID Call Waiting

This feature allows an incoming call on a DID trunk or a tie line to automatically be Camped-On to the destination station if the destination station is busy.

#### DID Digit Conversion

This feature allows the *UNIVERGE* NEAX 2000 IPS to convert the digits received from the serving C.O. to valid station numbers when the C.O. numbering plan differs from the desired station numbering plan.

#### DID Name Display

This feature allows Name Assignment for a DID Number received from a Public Network, and displays the name on an LCD of a Multi-line Terminal or Attendant Console.

### Direct Inward System Access (DISA)

This feature allows an outside caller to access the system using an exchange network connection without Attendant or station assistance. The outside user may originate calls over any or all of the system's facilities such as WATS, FX, Tie Line or CCSA. The outside user can also directly call stations and access miscellaneous trunks for such features as dictation access.

#### Call Forwarding set by DISA

This feature allows an outside caller to set Call Forwarding – All Calls Direct Inward System Access (DISA) code.

### Direct Inward Termination (DIT)

This feature automatically routes incoming network exchange calls directly to a pre-selected station without Attendant assistance. The call can then be processed by the called party. Three-party Conference, Call Transfer, etc., are handled in the same manner as any normal trunk call.

### Direct Outward Dialing (DOD)

This feature permits any station user the ability to gain access to the exchange network by dialing an access code and receiving new dial tone. The user may then proceed to dial the desired exchange network number.

### Direct Station Selection/Busy Lamp Field (DSS/BLF) Console

This feature allows a Direct Station Selection/Busy Lamp Field (DSS/BLF) Console to be associated with a legacy Multiline Terminal. When the buttons on the DSS/BLF Console unit are programmed for Direct Station Selection (DSS) buttons, up to 60 stations can be directly accessed in addition to those already appearing on the Multiline Terminal. Busy status for each station is indicated by a red LED associated with each button. In addition, the DSS console can provide the following functions:

- Message Waiting - Set/Cancel/Status Display
- Do Not Disturb - Set/Cancel/Status Display
- Automatic Wake Up No Answer - Status Display/Cancel
- Agent Busy Out - UCD - Status Display
- Line Lockout - Status Display
- Room Cutoff - Set/Cancel/Status

### Busy Out Status Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Busy Out Status Console. This feature is activated by use of a Function Mode key on a DSS/BLF Console. Busy Out Status for each station is indicated by a red LED associated with each button.

### Do Not Disturb Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Do Not Disturb (DND) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. DND set status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel the DND status of other stations using the DND Console.

### Message Waiting Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Message Waiting (MW) Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Message Waiting status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/reset MW status using the MW Console.

### Room Cutoff Console

This feature allows a DSS/BLF Console unit associated with a Multiline Terminal to be used as a Room Cutoff Console. This feature is activated by the use of a Function Mode key on a DSS/BLF Console. The Room Cutoff status for each station is indicated by a green LED associated with each button. In addition, the Multiline Terminal user can set/cancel Room Cutoff to another station using the Room Cutoff Console.

### Wake Up No Answer Console

This feature allows an EDW-48-2A unit associated with a Multiline Terminal to be used as a Wake Up No Answer (WU) Console. This feature is activated by a function mode key on a DSS/BLF Console. The No Answer status for each station is indicated by a flashing green LED associated with each button.

## Distinctive Ringing

This feature provides Distinctive Ringing patterns to the station so that the station user can distinguish between internal and external incoming calls. This feature also enables the LED associated with the line key of the Multiline Terminal to flash in two colors according to the kind of incoming call.

## Do Not Disturb

This feature restricts incoming calls to a station and can be set by an individual station or from the Attendant Console. Placing a station in Do Not Disturb (DND) does not prevent a station from originating a voice or data call or from receiving a data call. This feature also allows a station to ensure privacy from telephone interruptions while on an outgoing call. Additionally, the Attendant Console can place a group of stations in the Do Not Disturb condition.

### Do Not Disturb – Group

This feature allows the system to schedule to set/cancel Do Not Disturb for a group of stations at appointed time. The system has up to four patterns of timetable, and each timetable has time to set/cancel Do Not Disturb in a day (the time is programmable by the system). The timetable can be assigned for day of the week. The different timetable can also be assigned for specific dates of the year.

### Do Not Disturb - Hotel/Motel

This feature allows the Attendant Console(s), Hotel/Motel Front Desk Instrument(s), guest stations or Property Management System (PMS) terminal(s) to place individual stations into Do Not Disturb. Calls can be placed from stations set in DND.

### Do Not Disturb-System

This feature simultaneously restricts incoming calls to a pre-assigned group of stations by operation from the Hotel/ Motel Front Desk Instrument(s). Attendant Console(s) and Hotel/Motel Front Desk Instruments can use the DND OVR key to override this Do Not Disturb setting.

## Dterm Assistant

Dterm Assistant is Web-based software which resides on the server and provides end users with the ability to maintain Dterm Multi-line Terminals and the IPS telephony features (such as Speed Dialing) from a Web-enabled PC. The Dterm Assistant operates in a client-server environment and can manage multiple IPS systems over a Local Area Network (LAN)/Wide Area Network (WAN).

## DtermIP

DtermIP is an IP-based Multiline Terminal which provides a built-in capability of peer-to-peer IP communications. The *UNIVERGE* NEAX 2000 IPS system provides the DtermIP with same IP communications capabilities of an IP Enabled Dterm (The IP Enabled Dterm is a Dterm Multiline Terminal with an add-on IP adapter unit).

## Elapsed Call Timer

This feature provides a display of the elapsed time while a Multiline Terminal with LCD is connected to any trunk.

## Enhanced 911

This feature allows the PBX to transmit a caller's emergency service identification information to an Enhanced 911 Emergency system. The 911 notification is also provided to the EMG key of a designated Attendant Console/Dterm.

## Executive Calling

This feature allows a station to be assigned a VIP class. This provides special ringing to a called station when that station is idle, and automatic sending of three tone bursts to a called station when that station is busy, provided the call was originated from a station assigned as VIP class.

## Executive Override

This feature allows selected users to override a busy condition on a called station. A warning tone is transmitted to both stations in the busy call before the busy condition is overridden, and a three-party Conference is then established.

## External Paging with Meet-Me

This feature allows a station user or attendant dial-access to local voice paging equipment and connects both parties automatically after the paged party has answered the page by dialing an access code.

### Fax Arrival Indicator
When a call from a C.O. line (Direct-Inward-Termination, Direct-Inward-Dialing, Automated Attendant), station or tie line has terminated to a facsimile machine, a related lamp on a pre-designated Multiline Terminal is caused to light, indicating reception of a facsimile call.

### FAX over IP
This feature allows the system to transmit facsimile communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN). Since PBX regards facsimile equipment as one of ordinary telephones, IP Packet Assembler/Disassembler (IPPAD) and Voice Compression Trunk (VCT) are required for facsimile uses over IP network same as legacy stations. The facsimile transmission procedure (T.30 or G.711/G.726 pass-through) is supported with IP-PAD/VCT.

### Feature Activation from Secondary Extension
This feature allows the Multiline Terminal user to access an appearance of another extension and program certain features from that extension.

### Flexible Line Key Assignment
Multiline Terminals can have any desired line-key assignment. This feature permits assignments to be tailored to each individual's needs. (The terminal's primary extension line appearance is the only line key that cannot be reassigned.)

### Flexible Numbering Plan
The *UNIVERGE* NEAX 2000 IPS has a Flexible Numbering Plan. All access codes and station numbers and can be assigned in system programming. Refer also to the Single Digit Dialing Features and Specifications, which further increases the flexibility of the system.

### Flexible Ringing Assignment
This feature allows lines on Multiline Terminals to be individually programmed to ring or not ring.

### Forced Account Code
This feature forces the user to enter an Account Code (up to 8 or 10 digits) for all outgoing calls. The Account Code must be dialed before dialing the outgoing number. Calls are processed only when the dialed Account Codes are valid.

### Group Call by Pilot Number Dialing
This feature allows a Station User (Multi-line Terminal/Single Line Telephone/PS) or a Trunk Party to page a group of Stations simultaneously by dialing a Pilot Number. The maximum of 32 Stations can be assigned to a Paging Group, and the Paging Group is associated with the Pilot Number. After one of the Paged Stations answers, the paging becomes a 2-Way calling between the Calling Party and the first answered Station and automatically stops paging other Stations.

### Group Listening
When a Multiline Terminal user makes a call using the handset, pressing the SPKR key will allow others to listen through the built-in speaker of the Multiline Terminal. The user may continue talking on the handset at the same time.

### Hands-free Answerback
This feature allows the station user to answer a voice call without lifting the handset.

### Hands-free Dialing and Monitoring
This feature allows the station user to dial or monitor a call without lifting the handset.

### Hold
This feature permits a user to Hold a call in progress. After Hold has been set, the station user can make or answer new calls.

#### Automatic Hold
This feature allows a Multiline Terminal user to automatically hold a call in progress when other line/trunk key is pressed.

#### Call Hold
This feature permits a user to Hold a call in progress by sending a hookflash and dialing the Call Hold feature access code, or by pressing the Call Hold key. This line can then be used for originating another call or returning to a previously held call.

#### Dual Hold
This feature permits a station user who is placed on Hold by another station to place that station on Hold also.

#### Exclusive Hold
This feature allows a Multiline Terminal user to place a call on Hold and to exclude all other station users from retrieving the held call.

#### Non-exclusive Hold
This feature allows a Multiline Terminal user to place a call on Hold that may be retrieved by any station that has an appearance of the held line.

### Hotel/Motel Attendant Console
The Attendant Console can be programmed to function as a Hotel/Motel Attendant Console. In addition to the business features and functions of the Attendant, the Hotel/Motel Attendant Console can set Room Cutoff (individual and group), Automatic Wake Up, Message Waiting, and Do Not Disturb (individual and group).

### Hotel/Motel Front Desk Instrument
A Multiline Terminal with LCD can be programmed to function as a Hotel/Motel (H/M) Front Desk Instrument. This can be used to set and cancel standard H/M features such as Message Waiting, Do Not Disturb, Automatic Wake Up, and Room Cutoff.

### Hotline - Inside/Outside
This feature causes the terminal to place a call to another station or to an outside party automatically when the user selects the Hotline extension.

### House Phone
This feature allows selected stations to reach the Attendant simply by going off-hook.

### Individual Attendant Access
This feature permits a user to call a specific Attendant by dialing an Attendant call code.

### Intercept Announcement

This feature provides the automatic interception of Direct Inward Dialing (DID) and Tie Line calls which cannot be completed due to unassigned station or level. The caller hears a recorded Intercept Announcement that informs the caller that an inoperative number was reached, and may supply the number for information.

### Intercom

Three types of Intercoms are available: Manual Intercom, Automatic Intercom, and Dial Intercom. Each type of Intercom provides access to a small group of Multiline Terminals with simplified calling methods.

#### Manual Intercom

The Manual Intercom groups have up to six Multiline Terminals sharing a common signal path. Users can call other members of the Manual Intercom group by pressing a Manual Intercom key; each press sends a tone burst over the speakers of all the terminals in the group. When another user answers the call, a speech path is activated.

#### Automatic Intercom

Automatic Intercom provides a path for Voice Announcement Calls with Handsfree Answerback between two Multiline Terminals using a line key. Private conversations can be held by using the Multiline Terminal handsets. The Busy/Idle status of the associated Multiline Terminal is displayed on the Automatic Intercom line key LED.

#### Dial Intercom

Dial Intercom comprises up to 10 Multiline Terminals which can call each other using a dedicated Dial Intercom line key with abbreviated dialing. Dial Intercom calls can be Voice Announce with Handsfree Answerback or ringing calls.

### Internal Tone/Voice Signaling

Multiline Terminals can signal incoming internal calls by Voice Announcement or by ringing according to the programmed mode (Voice first or Ring first) of the called terminal. The caller can dial the digit 1 to change from Voice Announcement to Ring Tone or vice versa. The Multiline Terminal assigned this feature can program the following two modes:

- Voice Mode: allows an incoming call to terminate with Voice Announcement.
- Tone Mode: allows an incoming call to terminate with ringing.

### Internal Zone Paging with Meet-Me

This feature allows the Attendant Console and system users to page over the built-in speakers of the Multiline Terminals within the assigned zone or all zones.

### IP Enabled Dterm

This feature provides a Dterm Series E/Series i terminal, if equipped with an IP adapter unit with a capability to provide a converged infrastructure at the desktop, with a 10Base-T/100Base-TX Ethernet connection to corporate Local Area Networks (LAN). The IP Enabled Dterm can communicate with other IP Enabled Dterm and CCIS network (IP based) on a peer-to-peer connection basis. And, the IP Enabled Dterm also communicates with legacy stations and trunks (TDM based) via IPPAD (IP Packet Assembler/Disassembler). The IP Enabled Dterm provides users with all features currently available in Dterm Series E/Series i terminals.

### Last Number Redial
This feature allows users to redial the last station-to-station or outside number they dialed using a feature access key or a feature access code. This is useful when the called station is busy or does not answer.

### Least Cost Routing - 3/6 Digit
This service feature allows the *UNIVERGE* NEAX 2000 IPS to be programmed to route outgoing calls over the most economical facility (WATS, FX, DDD). Based on the individual area code and office code dialed (6-digit analysis), the system examines the programmed tables and uses the trunk in the order specified.

### Line Lockout
This feature automatically releases a station from the common equipment if the station remains off-hook for longer than a programmed interval before dialing. Howler tone may be programmed to be sent to the station in Line Lockout.

### Line Pre-selection
This feature provides the station user with two ways to select an idle, held, recalling, or ringing line before going off-hook.

### Maid Status
This feature allows the Hotel/Motel (H/M) Front Desk Instrument, Property Management System (PMS) terminal, or guest room station (using special access code) to register the condition of each guest room.

### Maintenance Administration Terminal (MAT)
The Maintenance Administration Terminal (MAT) is a personal computer that provides an interface to the PBX via the system's CPU card. The MAT PC must have the MATWorX program properly installed to communicate with the PBX. MATWorX is required for system software registration and activation. MATWorX is a Graphical User Interface (GUI) program that provides an efficient method to manipulate the PBX database. This program contains extensive help files, Usage Wizards and Tool Tips, with hyperlinks imbedded in the text. The hyperlinks provide quick access to the appropriate Add-In modules. Add-In modules provide a user friendly intuitive method to customize the PBX database.

### Message Center Interface (MCI)
This feature provides an interface with a customer supplied Voice Mail System (VMS) that can send Message Waiting lamp control data to the system. The Message Center Interface (MCI) can provide the following operations:

1. When terminating the call to the VMS, the system sends call connection status information to the VMS through the MCI.
2. The VMS sends the Message Waiting Lamp on data to the MCI.
3. The system, upon receiving this control data from the MCI, illuminates the Message Waiting lamp of the corresponding station.
4. The VMS, upon receiving retrieved message information, will send the Message Waiting lamp control data requesting the system to extinguish the Message Waiting lamp of the corresponding station.

## Message Registration

This feature provides output from the system to a call accounting system using an RS-232C connector. This allows the Hotel/Motel clerk to retrieve the information needed to charge for local and toll calls.

## Message Reminder

This feature allows a user or Attendant to turn on the message waiting (MW) lamp of a Single Line Telephone, or the Message Reminder (MSG) LED of a Multiline Terminal (if assigned).

## Message Waiting

### Message Waiting – Single Lamp

This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, administrative station, Voice Mail System (VMS) or Property Management System (PMS) terminal to light a lamp (on an uninterrupted or interrupted basis) on a Single Line Telephone or Multiline Terminal to indicate a message is waiting.

### Message Waiting – Multiple Lamp

This feature allows the Attendant Console, Hotel/Motel (H/M) Front Desk Instrument, administrative station, Voice Mail Systems (VMS) or Property Management System (PMS) terminal to light multiple line keys on a Multiline Terminal, to indicate a message is waiting. This allows multiple individuals who share the same Multiline Terminal to receive their own Message Waiting indication.

### Voice Message Waiting

In addition to the lamp indication control, this feature also provides the Voice Message Waiting service that an originating station user can set the Message Waiting with a recorded message by using the Digital Announcement Trunk (DAT) card.

- Voice Message Waiting – System

  An originating station user can choose the recorded message to be set by dialing the message number associated. The messages are recorded by the predetermined station.

- Voice Message Waiting - Individual

  When setting Message Waiting, an originating station user announces the message to be recorded after dialing the station number.

## Miscellaneous Trunk Access

This feature allows the connection of various types of external facilities. In addition to Loop and Ground Start Trunks, the following can also be interfaced with the *UNIVERGE* NEAX 2000 IPS: CCSA Lines Code Calling Equipment, Dictation Equipment, Foreign Exchange (FX) Lines, Radio Paging Equipment, and Wide Area Telephone Service (WATS) lines. Refer to separate features, Direct Inward Dialing (DID), and Tie Line Access for more applications of Miscellaneous Trunk Access.

### CCSA Access
This feature allows connection to or from a CCSA (Common Control Switching Arrangement) network. A CCSA network is a special, privately-leased network constructed for one customer's exclusive use that replaces or augments the public switched network services. The outgoing connections using CCSA lines are accomplished in the same manner as a normal outgoing call. Incoming calls are received from the CCSA network as a series of digits from the network instead of a ringing signal, and the connection is established in the same manner as a Direct Inward Dial (DID) or Tie Line connection. For Tie Line applications, the customers can construct a network with their own numbering plan. In a CCSA application, the numbers are issued by the C.O. following the CCSA network numbering plan.

### Code Calling Equipment Access
Code Calling Equipment consists of external paging units and external dialers requiring dial access from the *UNIVERGE* NEAX 2000 IPS.

### Dictation Equipment Access
This feature permits dial access to customer provided Dictation Equipment, and in some instances allows them to maintain telephone dial control of normal dictation system features.

### Foreign Exchange (FX) Access
An FX line is a line that is extended and terminated at a distant Central Office. With this feature, outgoing calls over the FX line become a local call at the distant C.O.

### Radio Paging Equipment Access
This feature provides station users dial access to Radio Paging Equipment, and to selectively tone - or voice/ tone-alert individuals carrying pocket paging devices. The paged party (when on premises) can be connected to the paging party by going to the nearest station and dialing an answer back code.

### Wide Area Telephone Service (WATS) Access
This feature allows any station user direct dial access to outgoing WATS lines.

## Mobility Access
2000 IPS supports Mobility Access (MA) feature including enquiry calls and activation/deactivation by a user. This feature allows a user to make/receive telephone calls from a remote location such as a cell phone via the PBX, as if the user is making and receiving calls from his/her desk phone.

## Modem over IP
This feature allows the system to transmit modem communications over IP network, via Local Area Networks (LAN) and corporate Wide Area Network (WAN).

## MP Program Download
NEAX 2000 IPS provides Online MP Program Download via IP network. The IPS downloads the MP upgrade program from FTP server using MATWorX IPS. Immediate or scheduled changeover to the upgrade program is available. It is also possible to change back to the previous program that was in use before the changeover (changeback).

### Multiple Language Display
A language displayed in the LCD of a Multiline Terminal and Attendant Console can be selected on a per-station basis or on a per-operator console basis. The available languages are the same as used on a system-wide basis.

### Multiline Terminal Attendant Position
A Multiline Terminal with LCD can be programmed to function similar to an Attendant position. This Attendant position has limited access to Attendant related features and functions and can be substituted where an Attendant is required but an Attendant Console is not necessary. When a DSS/BLF console unit is associated with this Attendant Multiline Terminal enhanced operation is available.

### Music on Hold
This feature plays music when a line is placed on hold. Music is provided by a circuit board memory chip, IP adpter, or a local music source, such as a CD player or a radio.

### Night Service
This feature provides a variety of methods for handling incoming calls when the system is in night mode. These include:

- Attendant Night Transfer
- Call Rerouting
- Day/Night Mode Change by Attendant Console
- Day/Night Mode Change by Station Dialing
- Night Connection-Fixed
- Night Connection-Flexible
- Trunk Answer Any Station

#### Attendant Night Transfer
When the Attendant Console is in Night Service, any operator directed calls (dial 0 calls) are automatically routed to a preprogrammed station. Priority Calls and Off-Hook Alarms which terminate to an Attendant are also routed by this feature.

#### Call Rerouting
This feature provides flexible reroute capabilities for a variety of calls when the system is in night mode.

#### Day / Night Mode Change by Attendant Console
This feature provides activation of DAY/NIGHT Mode Change by depressing a predetermined key from the Attendant Console.

#### Day / Night Mode Change by Station Dialing
This feature allows selected stations to activate a change from day mode to night mode by dialing a special code.

#### Day/Night Mode Change by System Clock
This feature provides automatic activation of Day/Night Mode Change by using System Clock.

#### Night Connection - Fixed
This feature allows incoming calls normally terminated to the Attendant to reroute to a predetermined station when the system is placed in Night Service.

#### Night Connection - Flexible
This feature provides incoming calls normally terminated to the fixed night station to be Call Forwarded to another station.

#### Trunk Answer Any Station (TAS)
This feature allows any station, other than one with incoming restrictions, to answer incoming calls when the system is in the night mode. When this feature is activated, incoming exchange network calls will activate a common alert signal at the customer premises. By dialing a specified code, any station may answer the call and then extend it to any other station by means of the Call Transfer feature.

#### Overflow for TAS Queue
If the TAS Call is not answered by predetermined time, the call will be forwarded to predetermined Station/Attendant Console/Announcement Service.

#### Queue Limit for TAS
When a DID Call is converted to TAS and the number of used Lines reaches queue limit, this feature provides the system to restrict the next call terminating.

### No CID Call Routing
This feature allows designing a call terminating system based on the reason for absence of calling party number received from the network.

### Off-hook Alarm
This feature allows a station user to call the Attendant, or a pre-designated station, by simply staying off-hook for a preprogrammed period of time. The calling number is automatically displayed at the Attendant Console, or the pre-designated station if equipped with an LCD.

### Off-Premises Extensions
This feature allows the connection of a single line telephone in an off-premises location. The connection to the Off-Premises Extension can be through direct copper or through the local telephone company.

### Open Application Interface (OAI)
Provides a computer-to-PBX interface, allowing a computer to control the function of the *UNIVERGE* NEAX 2000 IPS. The *UNIVERGE* NEAX 2000 IPS can be customized to accommodate most customer applications. Application software can be provided by NECAM, an outside software house, or a customer.

### Pad Lock
This feature temporarily restricts telephones from making unauthorized calls by dialing special access code when station users are away from their seats.

### Periodic Time Indication Tone
This feature provides a periodic tone to the station user who has made an outgoing call. This feature can be allowed or denied for each station.

### Pooled Line Access
A line key can be assigned to access Pooled Lines. Each line key will allow incoming, outgoing, or both-way access to a trunk route.

### Power Failure Transfer
This feature provides for specified trunks to be automatically connected to designated Single Line Telephones in the event of AC power loss. It is normally used when the system is not equipped with reserve power.

### Priority Call
This feature allows the Attendant to answer a call before other calls, at the Attendant's discretion.

### Privacy
This feature restricts Multiline Terminal users from depressing a busy line button and entering a conversation unless permitted by the Multiline Terminal user currently on that line button or if the line button is assigned for Direct Privacy Release.

#### Direct Privacy Release
This feature allows a station user with a secondary appearance of another extension in the system to access that extension when it is being used by someone else. This feature allows for a simplified method for establishing a conference. In addition, this feature can be used to emulate PC dialing, where a single line extension connected to a PC can appear on a Multiline Terminal and be accessed by the Multiline Terminal user after the PC is completed dialing.

#### Manual Privacy Release
This feature allows a Multiline Terminal user to enter a conversation on a busy line button if the Multiline Terminal user already in the conversation allows them by releasing Privacy.

### Private Lines
Only a C.O. trunk assigned to that specific station is seized when a station user originates an outgoing C.O. call or when an incoming C.O. call is terminated at the station designated by Direct-In-Termination. In this manner, stations and C.O. trunks are to be associated on a 1-to-1 basis.

### Property Management System Interface
The *UNIVERGE* NEAX 2000 IPS provides a data interface to a locally provided Property Management System (PMS). This enables communication between the *UNIVERGE* NEAX 2000 IPS and the PMS in order to provide computer control of Hotel/ Motel features.

### Proprietary Multiline Terminal
There are five Multiline Terminals that can be used with the system.

- Dterm Series i
- Dterm Series E
- IPK/Electra Elite

#### Automatic Idle Return
This feature returns a station to the idle state after 3 seconds of reorder tone is received due to the distant end disconnecting.

### Called Station Status Display

This feature provides a display on the status of a called station on the LCD of the calling Multiline Terminal.

### Calling Name and Number Display

This feature provides a display on the LCD of the Multiline Terminal receiving a call, indicating the station number or trunk number of the incoming call.

### Dynamic Dial Pad

This feature allows to make an outgoing call at first hand by pressing a ten key of Multiline Terminal, without pressing a Speaker key or going off-hook.

### Group Fearture Key

This feature provides the multiple line keys of Multiline Terminal with DSS/BLF function. A station user can monitor the status of group member stations such as idle, busy, hold and ringing by lamp indication of the multiple line keys. If a line is idle, a station user can make a call to the member station by pressing the associated line key (instead of pressing DSS key). If a line is ringing or is on hold, a station user can pick up the incoming call or the call on hold by using My Line, and the ringing or held line is made idle.

### Handsfree Unit

The built-in Handsfree Unit enables full Handsfree operation for both internal and external calls (No optional Handsfree Unit is required).

### I-Hold / I-Use Indication

Multiline Terminals provide indication of which line keys have been placed on Hold, or are in use by that Multiline Terminal. The LED associated with the line key will give the appropriate indication.

### Microphone Control

All Multiline Terminals are equipped with a Microphone Control button with an associated LED.

### Multiple Line Operation

This feature allows for the appearance of multiple lines on the Flexible Line Keys and feature keys of all Multiline Terminals.

### Mute Key

This feature allows the distant extension user, of a station user that presses a mute key during conversation, not to hear the station user's voice though the station user can hear the distant extension user's voice. By pressing the mute key again, the mute status returns to original conversation.

### My Line Number Display

This feature enables a Multiline Terminal to display My Line (Primary Extension) number on the LCD during the Multiline Terminal in idle state.

### Preset Dialing
This feature allows a Multiline Terminal user to prepare and verify a number in the display on the LCD before dialing. When a wrong number is entered, the user can correct the number before originating the call.

### Prime Line Pickup
This feature allows a Multiline Terminal user to go off hook and originate a call from the line assigned as the Prime Line without depressing the associated line key.

### Recall Key
Each Multiline Terminal is equipped with a Recall Key that is used to generate a hookflash to access features provided by the outside exchange, or to abandon a call while retaining the line for origination of another call.

### Relay Control Function Key
This feature provides a Multiline Terminal with the ability to activate/deactivate relays (on a PN-DK00) to control external devices.

### Ring Frequency Control
The ring frequency of the Multiline Terminal can be controlled on a station basis in system programming (14 frequencies are available) or by use of a function key on the Multiline Terminal.

### Ringing Line Pickup
This feature provides the ability to answer any call ringing into a Multiline Terminal by just lifting the handset.

### Soft Keys
According to the status of the Multiline Terminal, function keys (Soft Keys) are displayed in the third line on the LCD. If the status of Multiline Terminal changes, the Soft Keys will change automatically. Also if the Help key is pressed, explanation of indicated Soft Keys are shown on the LCD.

### Volume Control
Multiline Terminals are equipped with common Volume Control keys for:

- Built-in Speaker / Handset Receiver Volume.
- Ring Volume.
- C.O. Transmission Level.
- LCD contrast.
- Ring Tone Frequency

The Volume Control keys are located on the lower front side of Multiline Terminals (UP and DOWN).

## Remote Hold
This feature allows a Multiline Terminal user to hold it on the line button of transferred terminal, by pressing the Hold key.

### Remote PIM over IP
When IPS$^{DMR}$ and 2000 IPS PIM are installed at remote site, and connected to a 2000 IPS or IPS$^{DM}$ at main site over IP network, the Main Site system controls and maintains the IP Remote PIM operation as one single system. If a communication failure occurs between the Main Site and Remote Site, the Remote Site automatically changes over to a survival mode and operates as a stand-alone system.

### Reserve Power
This feature provides backup power from a 24V battery source in the event of a commercial power failure.

### Resident System Program
This feature provides the installers a simple procedure to have the system generate system data according to the system hardware configuration, thereby providing immediate operation and shorter programming time. When activated, the system scans hardware configuration (such as line/trunk card slot location) and assigns system data (such as extension numbers, trunk numbers, etc.) according to a predetermined generic program assignment.

### Return Message Schedule Display
This feature permits any station user to register his Return Schedule from his phone when he leaves his desk or the premises, and have the Return Schedule displayed on a calling Multiline Terminal with a Liquid Crystal Display (LCD) during his absence.

### Room Cutoff
This feature allows the following types of terminals to temporarily restrict guest room telephones from making unauthorized calls when guests are away from their rooms. This feature allows the same restriction when the rooms are in Check Out status:

- Attendant Console
- Hotel/Motel (H/M) Front Desk Instrument
- Property Management System (PMS) terminal
- Guest room telephones using a special access code

There are two types of Room Cutoff conditions depending on the type of calls restricted.

- External Call Restriction: All outgoing calls from guest room stations are restricted in the Room Cutoff status. (Only internal calls are available.)
- Toll Call Restriction: All toll calls from guest room stations are restricted during Room Cutoff status. (Internal and local calls are available.)

### Room Status
This feature provides the Hotel/Motel (H/M) Front Desk Instrument with a visual display of the guest's room status. A supplementary print out (individual and summary) can be provided.

### Route Advance
This feature automatically routes outgoing calls over alternate facilities when the first choice trunk group is busy. Users select the first choice route by dialing the corresponding access code, and the equipment then advances through alternate trunk groups only if the first choice is busy.

### Save and Repeat
This feature allows a Multiline Terminal to save a specific dialed number and then redial that number at a later time.

### Security Alarm
This feature provides an indication on the Attendant Console when a contact closure occurs.

### Semi-Automatic Attendant Camp-On
This feature permits the Attendant to hold an incoming call in a special mode when the desired station for the transfer is busy. The Attendant sends a Camp-On tone to the busy station. When that station becomes idle, the Attendant is recalled automatically. After the Attendant answers the recall, the station is called automatically. When the station answers the Attendant call and the attendant releases, the station is automatically connected to the waiting party.

### Set Relocation
This feature enables two stations to be moved from one location to another without reprogramming station data at MAT.

### Single Digit Dialing
This feature provides the station user the ability to dial single digit codes to access certain features while still allowing the same digit dialed to be used as the first digit of guest room station numbers.

### Single Digit Feature Access Code
This feature allows stations to access certain other system features by the direct dialing of a Single Digit Access Code, while receiving Busy Tone or Ringback Tone.

### SNMP
Simple Network Management Protocol (SNMP) is a standard protocol for TCP/IP network management, which enables a network management application software to query a management agent (network device such as router, PC host, and hub) using a supported MIB (Management Information Base). The MIB is a database of network performance information that is stored on the network devices. The *UNIVERGE* NEAX 2000 IPS can support the SNMP standard MIB (MIB-II, defined in IETF RFC 1213) and private MIB and TRAP. This feature also enables the network management system (SNMP manager) to manage the 2000 IPS via Network Address Translation (NAT).

### Software Line Appearance (Virtual Extensions)
This feature permits assignment of circuits which do not physically exist, to be used as secondary extensions on Multiline Terminals. There are 1020 software lines (minus the number of Dterms and DtermIP) that can be assigned to line keys and used as desired.

### Stack Dial
This feature enables a Multiline Terminal or an Attendant Console to remember the numbers dialed in the preceding five calls, including the last number dialed. The stack dial numbers are sequentially displayed on the LCD display, thus allowing the station user to make an outgoing call by selecting the desired dialed number from the display.

### Station Hunting
Three Station Hunting arrangements are available. Station Hunting - Circular processes the call no matter which station in the hunt group is called. Station Hunting - Terminal initiates a hunt only when the pilot number of a hunt group is called. Station Hunting - Secretarial is initiated when a busy secretarial station in a Station Hunting - Circular group or Station Hunting - Terminal group is reached.

### Station Hunting - Circular
When a busy station in a hunt group is called, this feature permits the call to be processed automatically through the hunt group in a preprogrammed order from that station's position within the hunt group.

### Station Hunting - Terminal
When a pilot number is dialed and that number is busy, sequential Station Hunting will be initiated. However, if a number other than the pilot number is dialed and that number is busy, busy tone will be provided rather than initiate Station Hunting.

### Station Hunting - Secretarial
This feature allows assignments to be given to members of Terminal and Circular Hunting groups to reroute calls (when their hunting group is all busy) to a back-up hunting group.

## Station Message Detail Recording (SMDR)
This feature provides a call record for outgoing station-to-trunk calls and incoming trunk-to-station calls (including Data Call). This facilitates cost control by identifying trunk use and misuse by individual stations. Station Message Detail Recording (SMDR) enables call billing to customers and clients, and provides a means for checking local telephone bills.

### Station to Station SMDR
Based on the existing feature of SMDR output, this feature enables the output of an SMDR record for station-to-station calls (two-station connection). This output is based on the NEAX 2400 SMDR Format and can be interfaced from the MP via RS232C, MP via TCP/IP or AP00B MRC-F via RS232C. This feature can be enabled for SLT, Dterm, Dterm IP, PS, or Softphone.

## Station Service Status Display
This feature provides a maintenance person to know a service feature setting status for a specific station at a glance. By entering single command, the maintenance person can see a list of service features/station status regarding to the specific station.

## Station Speed Dialing
This feature allows a station user to dial frequently called numbers by dialing an access code and an abbreviated code, or by depressing a feature key or line key assigned for Station Speed Dialing capability.

## Step Call
This feature allows the Attendant or station user, after calling a busy station, to call an idle station by simply dialing an additional digit. This feature will operate only if the number of the idle station is identical to that of the busy station in all respects, except the last digit.

## Supervisory Control of Peripheral Equipment
When various types of peripheral equipment (such as facsimiles, dictation equipment, Voice Mail, etc.) are connected to the line circuits of the *UNIVERGE* NEAX 2000 IPS, this feature allows the loop of the line circuit concerned to open for a programmable interval, and send a release signal to the peripheral equipment when the calling party disconnects.

## System Clock Setup by Station Dialing
This feature enables a Station User to set up the System Clock, from Single Line Telephone, Multi-line Terminal, and PS.

### System Speed Dialing
This feature provides all users the ability to dial frequently called numbers using an abbreviated call code.

### Tenant Service
This feature provides for more than one organization (tenant) to share the same *UNIVERGE* NEAX 2000 IPS system. Through system programming, each organization may be restricted to its own Central Office trunks, Attendant Consoles and extension group. In addition, incoming calls are directed to the specific tenant.

### Terminal Login via NAT
The *UNIVERGE* NEAX 2000 IPS supports network configuration via a router with NAT. NAT (Network Address Translation) is a technology that translates an internal local IP address to a globally unique IP address before sending packets to the outside network. NAT is configured on the router at the border of the inside network and the outside network. NAT is used for efficient use of global IP addresses and for security (shield internal addresses from the public Internet).

- Remote connections of IP Enabled Dterm via NAT
  - Provide communications between IP Enabled Dterm located under the same NAT.
  - Provide communications between IP Enabled Dterm located under different NAT.

### Tie Lines
This feature allows any station user dial access or direct access to an E&M Tie Line.

### Tie Line Tandem Switching
This feature allows trunk-to-trunk connections through the *UNIVERGE* NEAX 2000 IPS without the need for any Attendant assistance or control. The major use of this feature is in association with a dial tandem tie line network to allow tie line connections and incoming tie line calls automatic access to, and completion of, local Central Office calls.

### Timed Forced Release
This feature allows the system to restrict a long-time call to reduce phone rates for trunk calls. For an outgoing trunk call, an incoming trunk call and a tandem call, when the preset time elapsed, the call is released forcibly.

### Timed Queue
When a user originates an outgoing trunk call and the called party is busy or does not answer, the caller can set the Timed Queue feature. When this feature is set, the trunk seizure is repeated and the number is redialed after a predetermined time interval.

### Timed Reminder
This feature allows the system to be programmed to automatically call stations at specified times. Upon answering, the station is connected to a recorded announcement or music source.

### Trunk - Direct Appearances
This feature allows Multiline Terminal users the ability to access a CO line or E&M Tie Line without dialing an access code. For this feature, trunks must be assigned to the line keys on the Multiline Terminal. Incoming calls on CO lines can be answered on the appropriate trunk line appearance.

### Trunk Queuing - Outgoing
This allows a station user, upon encountering a busy signal on a trunk, to dial a feature access code and enter a first-in, first-out queue. As soon as an outgoing trunk becomes available, stations in the queue will be called back on a first-in, first-out basis.

### Trunk-to-Trunk Connection
This feature provides any station user with the ability to conference together two outside trunk calls and abandon the connection without dropping the Trunk-to-Trunk Connection.

### Uniform Call Distribution (UCD)
The Uniform Call Distribution (UCD) feature permits incoming calls to terminate to a prearranged group of stations. Calls are distributed in the order of arrival to idle terminals within the group, based on which terminal has been idle the longest period of time. Stations may log on/log off from the UCD group. Supervisor stations may monitor conversations of agents.

#### Busy In/Busy Out-UCD
This feature allows an agent in a UCD group to log their station onto or off of the group. This allows the system to control whether a call directed to the pilot number of the UCD group goes to that station or not. This prevents incoming calls from being directed to stations at which no agent is available.

#### Call Waiting Indication-UCD
This feature provides a visual indication when an incoming call to a UCD group is placed in queue, due to an “all agents busy” condition. An external relay controlled indicator or an LED on a Multiline Terminal can be used to provide Call Waiting Indication.

#### Delay Announcement-UCD
This feature allows the system to provide a recorded announcement to an incoming caller placed in queue to a UCD group. A single announcement, or two separate announcements, can be provided.

#### Hunt Past No Answer-UCD
This feature allows calls targeted at a UCD group to hunt past an agents station, after a no answer condition, if the agent forgets to log off of the group and the agent is unable (or not available) to answer the call.

#### Immediate Overflow-UCD
This feature allows a call directed to a UCD group to immediately overflow to another UCD group, upon encountering an “all agents busy” condition.

#### Priority Queuing-UCD
This feature allows the system to prioritize incoming calls by trunk route and on a per station basis, when the call enters a UCD queue. When a call is a considered as priority it is placed at the beginning of the queue.

#### Queue Size Control-UCD
On incoming DID/Tie line calls, the system can be assigned a threshold that limits the number of calls in queue. When the queue size threshold is exceeded, incoming callers are connected to busy tone.

### Silent Monitor-UCD

This feature provides the UCD group supervisor with the ability to monitor a call to a UCD agent. The silent monitor function gives no indication (as an option) to either the agent or the calling party.

## Uniform Numbering Plan (UNP) -Voice and Data

In the numbering plan for a network to be configured through the use of Tie Lines, a Uniform Numbering Plan (UNP) is employed. When UNP is employed, a station user from any PBX within the network can call a desired party by using a uniform dialing method based on the UNP.

## Variable Timing Parameters

This feature gives the IPS the versatility to change timing duration using the Maintenance Administration Terminal (MAT) or the Customer Administration Terminal (CAT). All timing parameters are set initially in the Resident System Program. These timing parameters can be changed according to the customer's requirements.

## Voice Guide

This feature provides a station user with an announcement that informs:

1. The result of the operation when the station user set or canceled the service feature, instead of service set tone.
2. Which service has been set to the station; such as, Call Forwarding - All Calls, Do Not Disturb or Message Waiting, when the station goes off-hook, instead of special dial tone.

## Voice Mail Integration

This feature is used to interface the *UNIVERGE* NEAX 2000 IPS with a locally provided stand-alone type Voice Mail System (VMS). The VMS, connected to the *UNIVERGE* NEAX 2000 IPS single line circuit (LC), is controlled by sending/receiving DTMF signals using this LC.

The VMS's voice mail feature can be used by accessing this VMS directly from an extension. If a station sets its call forwarding destination to the VMS, calls to this station are connected to the VMS, and the messages can be registered according to the VMS instruction. In addition, the Message Waiting lamp of the station can be turned on automatically by the VMS.

## Voice Mail Private Password

Voice Mail Password can be prevented from displaying in LCD of Multiline Terminals when connected to the Voice Mail System.

## Voice Mail Transfer

This feature has two functions that provide streamlined transfer access to voice mail.

1. One touch access to VMS: When an Attendant transfers an external call to a station, and if the station is busy or unanswered, the Attendant can transfer the call to a VMS by dialing "9" or by pressing a function key provided for this feature.
2. Transferring Camp-On call to VMS: When an Attendant sets Camp-On to a busy destination station for an external call, and if the destination station does not answer by predetermined time, the call can be automatically transferred to a VMS.

## Voice over IP - (H.323)

Voice over IP allows the system to transmit voice conversations over a corporate Intranet using ITUT H.323 protocol.

## Voice over IP Log Collection

This feature provides Voice over IP (VoIP) log collection functions for analyzing VoIP related problems caused by IP device, usage method, network environment and other effects. A maintenance person will be able to understand its cause and figure out how to solve the problem. The following functions are supported.

### Call Log Collection

It enables the collected call log to be checked when the user reports a malfunction. Call log is output to the RS port of MP card. It includes the speech time, number of sent/received packets and number of received packets lost.

### Fault Log Collection

It enables a real time check when a malfunction exists at registration or checking of accumulated fault information by a maintenance person. Fault log is output to the RS port of MP card. It includes reset cause and reset time, number of sent/received control packets, number of resent control packets, number of control packets that fails to resend, number of abnormal control packets received and number of times control sequence becomes abnormal.

## Whisper Page

This feature allows a secretary to interrupt the boss in a private way. By pressing a feature key or dialing an Access Code, the secretary station can voice override the conversation between the boss and another party (station or trunk). When the conversation is interrupted, the boss can hear the secretary but the other party is unaware of the Voice Override.

# CCIS Features

### Attendant Camp-On With Tone Indication - CCIS
This feature permits the Attendant; when the desired station at another switching office is busy, to hold an in-coming call in a special waiting mode. A distinctive Camp-On tone is sent to the busy station when the Attendant sets Camp-On. When that station becomes idle, it is automatically rung and connected to the waiting trunk party.

### Attendant Controlled Conference - CCIS
This feature permits an Attendant (2400 IPX) to establish a conference, through CCIS, with up to eight parties of stations and/or trunks (inside and outside parties).

### Automatic Recall - CCIS
This service feature works as a time reminder. When an Attendant-handled call through CCIS remains on hold, camped-on, or ringing unanswered for a fixed interval, the Attendant is automatically alerted.

### Brokerage - Hot Line - CCIS
This feature provides a ringdown connection between two stations, each using a Multiline Terminal, in different offices in the CCIS network.

### Busy Lamp Field (BLF) - CCIS
This feature provides a busy status indication of the predetermined stations within the CCIS network. The visual indication is provided with a red LED associated with each DSS button on the DSS/BLF Console and Multiline Terminal. Pressing the DSS button allows a direct access to the preprogrammed station within the CCIS net-work.

### Busy Verification - CCIS
This feature permits an Attendant, via the Attendant Console on the 2400 IPX or the 2000 IPS, to interrupt a busy station's call at another switching office connected through CCIS.

### Call Back - CCIS
This feature provides inter-office Call Back. A station who has dialed a busy station at another office can set Call Back - CCIS by dialing a feature access code. When this feature has been set, the setting station will ring as soon as the busy station becomes available.

### Call Forwarding - All Calls - CCIS
This feature permits all calls destined for a particular station to be routed to another station or to an Attendant Console, in another office in the CCIS network, regardless of the status (busy or idle) of the called station. The activation and cancellation of this feature may be accomplished by either the station user or an Attendant.

### Call Forwarding - Busy Line – CCIS
This feature permits a call to a busy station to be immediately forwarded to a pre-designated station or to an Attendant Console in another office in the CCIS network.

### Call Forwarding – Don't Answer - CCIS
This feature permits a call to an unanswered station to be forwarded to a pre-designated station or to an Attendant Console in another office, when the called station does not answer after a predetermined time period.

### Call Forwarding - Intercept – CCIS
This feature allows calls to an inoperative number, through a CCIS trunk, to be intercepted and automatically routed to a recorded announcement informing the caller that an inoperative number was dialed and giving the Listed Directory Number for information.

### Call Forwarding - Override - CCIS
This feature allows a target station user (Station A) to call a station (Station B) which has Call Forwarding – All Calls - CCIS set.

### Call Processing Indication - CCIS
This feature provides visual indications of all CCIS calls being processed or waiting processing at the Attendant Console.

### Call Transfer - All Calls - CCIS
This feature allows a station user to transfer incoming or outgoing Central Office, intra-office and inter-office calls to another station in the CCIS network, without Attendant assistance.

### Call Transfer - Attendant – CCIS
This feature permits a station user, while connected to a CCIS network call, to transfer a call to an Attendant Console via the CCIS network.

### Called Station Status Display - CCIS
This service feature provides, on the LCD display of the calling Multiline Terminal, a display of the called station status of the remote office within the CCIS network.

### Calling Name Display - CCIS
This feature permits the station name of a calling or called party at another switching office, through the CCIS network, to be displayed either on a Multiline Terminal or an Attendant Console.

### Calling Number Display - CCIS
This feature permits the number of a calling or called party at another switching office to be displayed either on a Multiline Terminal or an Attendant Console.

### CCIS Networking via IP
This feature provides CCIS networks with Voice over IP (VoIP) or Peer-to-Peer IP capabilities to provide a converged infrastructure over corporate Wide Area Networks (WAN). The IP Enabled Dterm can communicate with other IP Enabled Dterm over the CCIS network (IP based) on a peer-to-peer connection basis. And, the legacy terminals (TDM-based terminals) can communicate with other legacy terminals or IP Enabled Dterm terminals over the CCIS network, via IP-PAD (IP Packet Assembler/Disassembler). Voice compression of G.729a (8Kbps) and G.723.1 (5.3Kbps/6.3Kbps) is available for those connections. The CCIS Networking via Peer-to- Peer IP provides users with all TDM-based CCIS functionality, such as feature transparency, centralized management, and centralized facilities. There are two types of connections available for CCIS Networking via IP:

- **CCIS Networking via IP (Peer-to-Peer Connections Basis)** When the distant systems are 2000 IPS, the systems are connected on a peer-to-peer basis. The CCIS call control signals are transmitted between the built-in IP trunks (CCIS Handler; CCH) on the MP card, over the Local Area Networks and Wide Area Networks (LAN and WAN). For connections between IP Enabled Dterm terminals, voice signals are also transmitted over the LAN and WAN. For connections between legacy terminals, voice signals are transmitted via IP-PADs. This connection is also available when the distant systems are 2400 IPX supporting peer-to-peer connections.

- **CCIS Networking via IP (Non Peer-to-Peer Connections Basis)** When the distant systems are 2000 IVS2, the systems are connected with IP trunks [including Voice Compression Trunks (VCT)], via Time Division Switch (TDSW). Voice signals of IP Enabled Dterm terminals are transmitted via IP-PADs, while those of legacy terminals are directly connected to the IP trunks. Call control signals between the systems are also transmitted over the IP trunks. Voice compression of G.729a (8 kbps) and G.723.1 (5.3 kbps / 6.3 kbps) can be provided by the IP trunks with VCT cards. This connection is also applicable when the distant systems are 2400 IPX not supporting peer-to-peer connections.

### Centralized Billing - CCIS
This feature is used to collect billing information from each office within the network and to direct it to the associated center office. Billing information is then forwarded to the central billing centers via RS232C interfaces.

### Centralized Day/Night Mode Change - CCIS
This feature switches the Day/Night mode of a remote office, linked to the main office (2400 IPX) via CCIS, in accordance with the Day/Night mode switching on the Attendant Console at the main office.

### Centralized E911 – CCIS
This feature allows the system to transmit a calling party number to the 911 Emergency systems over CCIS tandem connection.

### Consultation Hold - All Calls - CCIS
This feature permits a station user, within the CCIS network, to hold any incoming or outgoing public network or Tie Line call while originating a call to another station within the CCIS network.

### Data Line Security - CCIS
This service feature allows the lines which are used for data transmission through CCIS to be protected from interruptions such as Attendant Camp - On, Busy Verification - CCIS, Executive Right of Way, and Attendant Override.

### Deluxe Traveling Class Mark - CCIS
This feature provides outgoing call restrictions within the CCIS network. The following three types of restrictions are allowed:
- Deluxe Traveling Class Mark Restriction
- Route Restriction
- Numbering Restriction

### Dial Access to Attendant – CCIS
This feature allows a station user to call an Attendant Console by dialing an operator call code through the CCIS network.

### Digital Display - Station - CCIS
This service feature provides a display of the station number on the Attendant Console, during an Attendant-to-station connection, within the CCIS network.

### Digital Display - Trunk - CCIS
This service feature provides the Attendant with a visual indication of incoming and outgoing trunk calls during an attendant-to-trunk connection within the CCIS network. Trunk Group number and trunk identification code are displayed.

### Direct-In Termination - CCIS
This feature automatically routes incoming exchange calls through CCIS to a pre-assigned station in the network, without Attendant assistance.

### Distinctive Ringing – CCIS
This feature provides distinctive station ringing patterns for terminated calls, through the CCIS network, so that a station user can distinguish between incoming internal and external calls.

### Do Not Disturb - CCIS
This feature allows a station user to establish Do Not Disturb (DND) status on a temporary basis, during which time access to the station from CCIS calls will be denied.

### Dual Hold – CCIS
This feature allows two connected Multiline Terminals to be placed on hold simultaneously over the CCIS link. This enables the held parties to answer or originate a call from a secondary extension or the idle primary extension.

### Elapsed Time Display – CCIS
This feature provides an LCD which shows the duration of time that a Multiline Terminal is connected to any trunk through the CCIS network.

### Flexible Numbering of Stations – CCIS
This feature allows voice and data station numbers to be assigned to any instrument in the CCIS network, based solely upon numbering plan limitations.

### Hands-Free Answerback - CCIS
This feature allows a Multiline Terminal station user to respond to a voice call, through the CCIS network, without lifting the handset.

### Hot Line – CCIS
This feature allows two stations, at different nodes in the CCIS network, to be mutually associated on an automatic ringdown basis through the CCIS network.

### House Phone – CCIS
This feature allows selected stations to call an Attendant Console, through the CCIS network, simply by going off hook.

### Incoming Call Identification - CCIS
This feature allows an Attendant to visually identify the type of service and/or trunk group which is arriving or waiting to be answered at the Attendant Console through the CCIS network.

### Individual Attendant Access – CCIS
This feature permits a station user to call a specific Attendant Console, in the CCIS network, using an individual Attendant Identification Number.

### LDN Night Connection – CCIS
This feature routes Listed Directory Number (LDN) calls to a pre-selected station, in the CCIS network, when the Night mode has been entered.

### Link Alarm Display - CCIS
This feature provides an indication on external equipment when the CCIS link is connected/disconnected, when the system is initialized, or when the CCH is in make-busy.

### Link Reconnect - CCIS
This feature provides the system connected to CCIS network with the capability to release the redundant CCIS link connection and re-connect the link within the system for efficient usage of the CCIS links.

### Message Waiting Lamp Setting - Attendant – CCIS
This feature allows an Attendant, in the 2400 IPX, to set or cancel a Message Waiting lamp indication, through the CCIS network, on a station in 2000 IPS.

### Message Waiting Lamp Setting - Station – CCIS
This feature allows a station user, in the 2400 IPX, to set or cancel a Message Waiting lamp indication, through the CCIS network, to a station in 2000 IPS with this feature.

### Miscellaneous Trunk Access - CCIS
This feature provides access to all types of external and customer-provided equipment/facilities, such as Tie Line and exchange network, along with Dictation, Paging Access - CCIS and Code Calling through the CCIS network.

### Miscellaneous Trunk Restriction - CCIS
This feature denies certain stations and dial-repeating tie trunks access to specific trunk groups, such as Tie Line, exchange network, Dictation or Paging Access - CCIS through the CCIS network.

### Multiple Call Forwarding - All Calls - CCIS
This feature allows the last hop of a Multiple Call Forwarding - All Calls sequence to be forwarded over a CCIS network to a station in another office.

### Multiple Call Forwarding - Busy Line – CCIS
This feature allows the last hop of a Multiple Call Forwarding - Busy Line sequence to be forwarded over a CCIS network to a station in another office.

### Multiple Call Forwarding – Don’t Answer- CCIS
This feature allows the last hop of a Multiple Call Forwarding - Don’t Answer sequence to be forwarded over a CCIS network to a station in another office.

### Multiple Console Operation – CCIS
This feature provides console operation where Attendant Consoles are installed in more than one node in the CCIS network.

### Network Station Number - CCIS (FCCS)
When 2000 IPS is connected to a 2400 IPX via CCIS link, Network Station Number can be moved to other office within the network by a simple command operation from the Centralized MAT in the 2400 IPX.

### Night Connection - Fixed – CCIS
This feature routes calls normally directed to the Attendant Console to a pre-selected station in another office, through the CCIS network, when the Night mode has been entered.

### Night Connection - Flexible – CCIS
This feature provides an inter-office night connection service, via the CCIS network, when the calling station and the night station belong to different offices.

### Outgoing Trunk Queuing - CCIS
This feature allows a CCIS network station, upon encountering an all trunk busy signal, to dial a specified access code and enter a first-in, first-out queue. As soon as a CCIS trunk becomes available, stations in the queue will be called back on a first-come, first-served basis.

### Paging Access - CCIS
This feature provides dial access to paging equipment from an Attendant Console or a station, through the CCIS network.

### Restriction from Outgoing Calls - CCIS
This feature automatically restricts users of pre-selected stations from placing outgoing calls and/or certain miscellaneous trunk calls through CCIS, without Attendant assistance.

### Service Display - CCIS
This feature generates LCD displays on the Multiline Terminal corresponding to the various features as they are initiated.

### Single-Digit Station Calling - CCIS
This feature allows the assignment of Single-Digit Station numbers.

### Station-Controlled Conference - CCIS
This feature allows any station of the 2400 IPX to establish a conference among a maximum of eight parties of stations and/or trunks (inside and outside parties) of the 2000 IPS, through CCIS.

### Station-to-Station Calling - CCIS
This feature permits any station user to dial another station directly, through CCIS, without Attendant assistance.

### Station-to-Station Calling - Operator Assistance – CCIS
This feature allows a station user to call another station in the CCIS network, with the assistance of an Attendant Console operator.

### Toll Restriction - 3/6 Digits - CCIS
This feature allows the system to be programmed to restrict outgoing calls, through CCIS, according to specific Area and/or Office Codes. This restriction is determined on the basis of a three-digit Area Code or six-digit Area and Office Code numbering plan.

### Trunk Answer from Any Station - CCIS
This feature allows any station, not restricted from incoming calls, to answer incoming calls when the network is in Night mode.

### Trunk-to-Trunk Restriction - CCIS
This feature allows Trunk-to-Trunk tandem restriction by caller's information sent from each office (e.g., caller is a trunk) through the CCIS network.

### Uniform Numbering Plan – CCIS
In a CCIS network, a Uniform Numbering Plan enables a station user to call any other station in the network. Two alternative numbering plans are provided. In the first plan, the station user dials any digit station number from three to eight. The location of the office is identified by the first one-, two-, or three-digit of the station number. In the second plan, the station user dials a one-, two- or three-digit office code and any digit station number from two to eight.

### Voice Call - CCIS
This feature provides a voice path, through the CCIS network, between a Multiline Terminal in one office and a Multiline Terminal in another office. This path is established from the calling party to the called party's built-in speaker. If the called party's **MIC** lamp is on, the called party can have a conversation in hands-free.

### Voice Mail Integration – CCIS

This feature allows any station user in the CCIS network to utilize the Voice Mail System (VMS) with the Message Center Interface (MCI).

### Voice Mail Live Record over CCIS

Previously in a CCIS network of multiple IPS systems all sharing a centralized voice mail only the site where the voice mail was attached could use the Live Record feature. With R12.1 on the IPS it is now possible to use Live Record (automatic or manual recording) from all IPS systems with R12.1 installed. When a tandem connection is required to reach voice mail only the originating and destination IPS offices need to have R12.1 software installed.

### Voice Mail Private Password - CCIS

Voice Mail Password can be prevented from displaying in LCD of Multiline Terminals when connected to the voice mail system via CCIS.

# ISDN Features

### Call-By-Call Service Selection
Services can be selected on a call-by-call basis to all channels of a single PRI interface according to applications. Services may be used on any available channel, unlike Trunk Provisioning Service, in which services are assigned to specific channels.

### Called Party Recognition Service [Direct-In Termination (DIT)]
This feature provides an incoming Direct-In Termination (DIT) call via an ISDN trunk to be connected to a predetermined station. This application can be used for a station or modem.

### CLI Transparency
This feature provides a Calling Line Identity (CLI) received from ISDN or CCIS trunk to transparently send outgoing ISDN trunk, in case of tandem connection (ISDN-to-ISDN or CCIS-to-ISDN).

### CPN to Network - Present
This feature allows the ISDN network to be informed of the Calling Party Number and Name (CPN) when a call originates from a terminal connected to the System.

### CPN to Terminating User - Display
This feature provides a visual display of the originating station's number and sub-address information on a Multiline Terminal for incoming ISDN calls. This provides the terminal user with a quick and accurate way to identify the originating station's number (Calling Number).

### DID Addressing
This feature allows incoming ISDN-PRI calls to terminate to stations, attendant console, automated attendant, etc., based on the called party number. Direct Inward Dial trunks will be terminated to programmed destinations without attendant assistance.

### DID Addressing and DOD Addressing
This feature allows the system to use DID and DOD on the same B-Channels. Trunk Provisioning Service election is not required. (B-Channels can be used for DID and DOD without separating the trunk routes.)

### Event-Based CCIS
This feature allows a PBX customer who does not have a tie line (or when a customer cannot use the tie line due to busy or fault of the line), to use the various CCIS feature by using the ISDN line as a CCIS virtual tie line, on the IMX to *UNIVERGE* NEAX 2000 IPS connection or the *UNIVERGE* NEAX 2000 IPS to IPS connection.

### ISDN Terminal
This feature provides the system with an ISDN Terminal or Terminal Adapter (TA). ISDN Terminal to ISDN Terminal, ISDN Terminal to ISDN Trunk, ISDN Trunk to ISDN Terminal, ISDN Terminal to Single Line Telephone, ISDN Terminal to Multiline Terminal, and ISDN Terminal to PS connections are available.

### MEGACOM® Access/WATS
AT&T's MEGACOM® (WATS) network, as well as WATS from other carriers, can be used.

### MEGACOM® 800 Service/800 WATS Ultra WATS
AT&T's MEGACOM® 800 (Inward WATS) network, as well as 800 WATS provided by other carriers, can be used.

### MULTIQUEST® /900 Service

AT&T's MultiQuest® service can be used. (It is a "900"-type service.) Also, 900 service provided by other carriers can be used.

### Sub-address – Present

This feature allows a Primary Rate Interface ISDN trunk to transfer the Called Party Sub-address information to a destination ISDN station when the call is originated by the system. Dialing the called party station number and sub-address is required.

### Trunk Provisioning Service Selection

Each channel of a PRI interface can be dedicated to a particular service. Services are designated to specific channels; once designated, a channel can be used only for that service.

### Q-SIG Circuit Switched Basic Call – ETSI Version

Q-SIG conforms to ETS 300 172. This feature enables NEC's PBX to connect to NEC's PBX or other manufacturer by using Layer 3 protocol for the signaling for the support of circuit mode bearer services at the Q reference point.

The following services can be provided in conjunction with this feature:

a. Common station/voice service

Call Forwarding - All Calls **Note 1**

Call Forwarding - Busy Line **Note 1**

Call Forwarding - No Answer **Note 1**

Caller ID Display **Note 1**

Direct-In Termination **Note 1**

Call Transfer - All Calls **Note 1**

Hot Line **Note 1**

Do Not Disturb **Note 1**

Calling Number Display

b. Attendant Console service

System Speed Dialing **Note 2**

Call Queuing

c. Network features

Trunk-to-Trunk Connection

Brokerage - Hot Line

Least Cost Routing - 3/6 Digit

Tie Lines

**Note 1:** *Service not based on the specification of Q-SIG supplementary service.*

**Note 2:** *Service provided with some limitations.*

# Wireless Features

### Analog PBX Interface
This feature allows a secretary to interrupt the boss in a private way. By pressing a feature key or dialing an Access Code, the secretary

### Announcement - PS No Answer / Announcement - PS Out of Zone
This feature allows calls to a PS which cannot be paged in a predetermined period of time to be routed to the announcement notifying the calling party that the PS cannot answer. Announcements can be divided between Announcement-PS No Answer and Announcement-PS Out of Zone depending on the PS condition.

### Call Forwarding-Not Available
When a PS is power off or out of zone, a call directed to the PS is forwarded to a VMS, and a voice mail message can be recorded to the VMS and checked from the PS. Also the VMS can page the PS automatically after the voice mail message is recorded.

### Call Forwarding-PS Out of Zone
This feature allows calls directed to a PS which is power off or out of zone to be routed to an another station, an Attendant, an outside number or voice mail equipment. The call forwarding destination can be set/cancelled from the PS station or MAT/CAT.

### CCIS Interface
This feature allows the WCS to be integrated with NEC PBXs with CCIS interface.

### Calling Name Display - PS
Without answering incoming calls or hold calls terminates to the PS, the calling party's name can be confirmed by the indications on the LCD.

### Calling Number Display - PS
This feature provides a display on the LCD of a PS receiving a call, indicating the station number.

### DTMF Signal Sender
This feature allows a PS user to send the DTMF signal (tone) to the called party (terminal, voice mail system, etc.) while engaged in communication.

### Handover
When the signal transmission quality becomes inferior, a PS re-originates a call automatically and seizes another radio channel, making the WCS handover the call to another zone transceiver to maintain the speech quality.

### Individual PS Calling
This feature allows the calling party to page the individual PS.

### Last Number Redial-PS
This feature enables a PS to store the numbers dialed in the previous five calls including the last number dialed. The stack dial numbers are sequentially displayed on the LCD, allowing the station user to make an outgoing call by selecting the desired dialed number from the display.

### Multi-Line Operation-PS
PS equipped with two line keys, L1 key and L2 key, and different station numbers can be assigned to each of two lines. The number assigned to L1 key of the PS is called My Line and the other number assigned to L2 key is called Sub Line. My Line and Sub Line of a PS can be assigned to appear on the Flexible Line Keys of a Dterm and the Dterm can share the PS lines.

### Multi-Site Roaming
PS user can originate or receive a call in any place of a network provided by the plural PBXs which are interfaced by JT-Q931-a.

### *UNIVERGE* NEAX 2000 IPS Wired for Wireless
This feature allows the *UNIVERGE* NEAX 2000 IPS to have the Wireless PBX feature. *UNIVERGE* NEAX 2000 IPS provides several integrated features adding on the adjunctive configuration.

### Number Sharing
This feature allows the Dterm user to have a PS as Sub Station and to get service with one telephone number. In case that one user has both Dterm and PS, with this feature used, the user is not required to have separate two telephone numbers.
- When user is at his desk, a call is terminated to Dterm.
- When user leaves his desk with PS, a call is automatically terminated to PS.

In this feature, the Dterm and PS are referred as Main Station and Sub Station, respectively.

### Out of Zone Indication
When a PS user moves out of the service area and the electric field strength becomes weak, this feature notifies it to the user with the warning tone and the LCD display.

### Overlap Dialing
This feature allows a PS user to receive dial tone and dial the desired number to originate a call.

### Preset Dialing
This feature allows a PS user to confirm the number to be dialed before originating a call.

### PS Authorization
This feature is to confirm the identity of a PS to avoid an unauthorized PS from accessing the system.

### PS Location Indication
This feature is when PS calls a Multiline Terminals/Attendant Console or vice versa, this feature allows indicating the location name of the ZT that is connected to the PS on the LCD of the Multiline Terminal.

### PS Location Registration
This feature allows the WCS to supervise the location of each PS, upon receiving the location registration request, to allow call termination.

### PS Location Search
This feature allows a Multiline Terminal user to search and display a location name of a specific PS number on the LCD of the Multiline Terminal. The location name is a ZT name where the PS is located in the ZT's radio zone.

### Q.931a Roaming over IP Trunk
PS user can originate or receive a call in any place of a network provided by multiple PBXs that communicate by JT-Q931a protocol over IP Trunks (IPT+VCT) or Peer to Peer Trunks.

### Radio Channel Changeover
This feature is to supervise and changeover the channel automatically to avoid the interference and to maintain the speech quality.

### Short Message Notification (OAI)
This feature enables a Short Text Message (STM), once arrived at a mail box of the STM Server (external equipment), to be automatically distributed to the addressee PS via Open Application Interface (OAI). This feature also provides "TM Full" notification on the display (LCD) of the address PS that is busy. If the PS is out of zone, the feature makes retransmission of the STM after the PS returns to the zone.

### Soft Phone Wireless Handset
This feature allows a Soft Phone (Dterm SP30) user to use a PS terminal as the handset of the Soft Phone. The user can select a handset (Soft Phone Handset or PS) by pressing the Handset Switch key on the Soft Phone.

### Speech Encryption
This feature protects a call from being tapped.

### Speed Dial-PS
This feature allows a PS user to dial the certain frequently called numbers using two-digit abbreviated call codes.

### Station Hunting - Not Available
This feature allows a call placed to a PS station, which is out of zone or power off, to be forwarded to an idle station in a hunt group. Two Station Hunting arrangements are available.

- Station Hunting - Circular processes the call regardless of which station in the hunt group is called.
- Station Hunting - Terminal initiates a hunt only when the pilot number of a hunt group is called.

### Voice Mail Indication
When a message is mailed in the PS, an indication of the envelop icon is displayed in the LCD of the PS.

### WLAN Handset Registration/Service
This feature allows the 2000 IPS to accommodate the WLAN Handset (MH210/MH220/MH250) as a station. A terminal is connected to the 2000 IPS through Access Point and SIP Handler and operates under the control of the virtual CS on the SIP Handler. Station-to-Station calls between WLAN stations and calls between WLAN stations and existing stations/trunks are available.

# Chapter 6 System Overview

## Operating Environment

| Operating Condition | Temperature Relative Humidity | |
|---|---|---|
| Normal Operations | 5°C - 30°C<br>41°F - 86°F | 15% - 65%<br>Non-Condensing |
| Short Periods** | -0°C - 40°C<br>32°F - 104°F | 15% - 90%<br>Non-Condensing |
| During Storage and Transit | -18°C - 50°C<br>0°F - 122°F | 8% - 90%<br>Non-Condensing |
| **Not to exceed 72 consecutive hours or 15 days in a year. | | |

## Grounding Requirements

The system grounding must have a specific ground resistance and AC noise level, and is to be connected to a predetermined terminal in the PBX. Standard grounding requirements are as shown below:

- Communication grounding : Less than 10 ohm
- Protective ground for PIM : Less than 10 ohm

***Note:*** *The AC ripple on these various grounds should be less than 0.5Vp-p.*

**CAUTION**

Grounding circuit continuity is vital for safe operation of telecommunication equipment. Never operate this equipment with the grounding conductor

The following specific requirements apply to ground wiring.

An equipment grounding conductor that is at least as large as the ungrounded branch-supply conductors is to be installed as part of the circuit that supplies the PBX. Bare, covered, or insulated grounding conductors are acceptable. Individually covered or insulated equipment grounding conductors shall have a continuous outer finish that is either green, or green with one or more yellow stripes. The equipment grounding connector is to be connected to ground at the service equipment.

The attachment-plug receptacles in the vicinity of the PBX are all to be of a grounding type, and the equipment grounding conductors serving these receptacles are to be connected to earth ground at the service equipment.

# AC Power Requirements

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| AC Input Voltage | 90 to 132Vac or 180 to 264Vac; 47 to 64Hz |
| AC Input Current | 3.5A(at 100V), 2.0A(at 200V) |

# Installation

The following items are required for correct operation.

a.) Adequate space accommodation

b.) Adequate ventilation

c.) Commercial AC power

### Main Equipment

The installation of the *UNIVERGE* NEAX 2000 IPS is comprised of up to 8 Port Interface Modules (PIMs). A PIM provides 13 card slot for Common Control, Line/Trunk (LT), and Application Porcessors (AP) cards. It also houses an AC/DC Power Supply, DC/DC Power Supply (for –48V), and batteries for protection from short-term (about 30 minutes) power interruption.

Cabling inside the unit, between the switching equipment and the MDF, can all be done by plug-and-jack connections, while printed circuit cards can easily be plugged into the edge connectors. On all installations, a special provision for plug-and-jack connections completely eliminates possible errors in wiring. This allows the installation to be done easily and smoothly.

### Mounting Circuit Cards

(1) Before mounting the circuit cards, confirm the following items.

- Wrist Strap is connected to Frame Ground.
- Switch settings of circuit cards are already completed.
- The “SW1” switches of all PZ-PW121 cards are turned off.

(2) Mount circuit cards into their mounting positions according to the “Bay Face Layout” and “Port Assignment Table” given in the Office Data Programming Manual.

### Various installation Methods

To meet the specific needs of the customer's environment, *UNIVERGE* NEAX 2000 IPS provides the following installation methods:

- Floor Standing Installation
- Wall-mounting Installation
- IEC standard 19 inch Rack-mounting Installation

# System Administration

In this system, the Customer Administration Terminal (CAT) or Maintenance Administration Terminal (MAT) is used for programming the system data. The CAT is a digital multi-function telephone (Dterm) which is equipped with function keys, a dial pad and LCD and interfaces with the system via the MP card. The Maintenance Administration Terminal (MAT) is a personal computer that provides an interface to the PBX via the system CPU card. The MAT PC must have the MATWorX™ program properly installed to communicate with the PBX. MATWorX is required for system software registration and activation.

## Password Entry

In a system with password service, a maintenance person is required to enter an authorization level number (Password Level) and appropriate password prior to engaging in programming the system data with the MAT/CAT. A maximum of eight (8) Password Levels can be set up. The number of commands that the maintenance person can access is determined by the Password Level.

## Resident System Program

This resident system program generates system data automatically according to the system hardware configuration, thereby providing immediate operation and shorter programming time. When activated, the system scans hardware configuration (such as line/trunk card location) and assigns the system data (such as station numbers, trunk numbers, etc.) according to a predetermined generic program assignment.

### Service Conditions

1. This service is applicable for equipment installed in PIM0 through PIM3.
2. Data for any vacant slot is not assigned.
3. Virtual stations are not assigned.
4. A line/trunk card (PN-AUCA/PN-DK00/ /PN-4DAT/PN-4RSTF/PN-4VCTI) is not assigned, even if mounted.
5. An application card is not assigned, even if mounted.
6. No tenant assignment is provided. (Tenant 01 is assigned)

# Maintenance

## System Diagnostics

When a fault occurs in the system, an audible and visual indication will be given at the following units:

- External alarm indicating unit
- Fault messages reported at MAT*WorX* for remote reporting
- Alarm lamps in front of each package mounted in the frame

## Self Diagnostic/System Messages

The *UNIVERGE* NEAX 2000 IPS provides a sophisticated array of self-diagnostic routines that are continually and automatically being performed. Various system messages are printed when a fault occurs in a central processor system, switch network processors. Many other miscellaneous system messages and change of key status messages are also printed.

## Customer Administration Terminal (CAT)

The Customer Administration Terminal (CAT) is a digital multi-function telephone (Dterm) which is equipped with function keys, a dial pad and LCD and interfaces with the system via the MP card. Programming of the system can be done from selected Multiline Terminals with LCD. The designated Multiline Terminals can be placed in program mode, and system data can then be changed. To prevent unauthorized changes, password levels are assigned, providing authorization for access to certain areas of programming and denying access to others.

### Service Conditions

1. Programming from a Customer Administration Terminal can only be accomplished when the system is online.
2. All Multiline Terminals with LCD scanned during initialization will be Customer Administration Terminals.
3. The commands CM00 (Office Data All Clear) and CM01 (Office Data Partial Clear) cannot be accessed from the CAT. The CAT cannot delete itself from the system program.
4. Only two Customer Administration Terminals can be in program mode at the same time.
5. The data that can be changed from the CAT can be limited by the Password level assigned. There are eight levels of Passwords that can be assigned in system programming. The relation between Password level and access to available commands is also assigned in system programming.
6. A password can consist of a maximum of any eight digits with the following limitation: The password cannot be CCCCCCCC or FFFFFFFF.
7. Caution should be exercised when assigning Passwords to command authorization levels. If a password is forgotten, access to system programming will be limited and a system initialization with subsequent programming may be required.
8. When the Customer Administration Terminal is offline for programming, it cannot access normal terminal functions.

## Maintenance Administration Terminal (MAT)

The Maintenance Administration Terminal (MAT) is a personal computer (PC) that is used for programming and maintenance of the *UNIVERGE* NEAX 2000 IPS system. The MAT can provide a Maintenance Printout, Peg Count information and fault message output. Additionally, the MAT can be used to Remove and Restore to service any station in the system and can read or save system data from disks. The MAT can assign the Key Data for the Attendant Console. The MAT requires an IBM or compatible PC running Microsoft Windows 98, NT 4.0, 2000 or XP and MAT*WorX*.

MATWorX is a Graphical User Interface (GUI) program that provides an efficient method for manipulating the PBX database. This program contains extensive help files, Usage Wizards and Tool Tips, with hyperlinks imbedded in the text. The hyperlinks provide quick access to the appropriate Add-In modules. Add-In modules provide a user-friendly, intuitive method for customizing the PBX database.

MATWorX add-ins makes it easy for you to add or remove PBX features at any time. Add-ins are modular components that let you program specific features such as Caller ID, Station Assignments, Day/Night Modes, and Line Key Assignments.

Because add-ins are modular, you can add, remove, and upgrade them individually from within MATWorX. Add-ins let you modify your PBX's features without having to upgrade the MATWorX application itself. MATWorX also gives you a convenient way to launch other commonly used applications, such as Microsoft Word or Excel, directly from its Toolbar.

## Connecting to a PBX

There are three ways to connect your PC to an NEC PBX:

- Use a modem to establish a dial-up connection.
- Use a serial cable to establish a direct connection.
- Use TCP/IP over your Local Area Network (LAN), Requires DeviceServerWorX (DSW)

The method you use depends on how you installed and configured the device to which you want to connect. A serial cable direct connection offers better performance than a modem connection, but requires that the PC and device be within 50 feet of each other. A TCP/IP connection offers excellent performance and flexibility but requires a network connection to both your PC and the device.

## PBX Configuration Wizard

The PBX Configuration Wizard is a custom tool in MAT*WorX* that enables you to establish the proper communication settings between your computer and the *UNIVERGE* NEAX 2000 IPS. The Wizard asks you simple questions and then uses the information to automatically configure the connection for the PC and the PBX.

### Service Conditions

1. Connection through modems is available, providing remote maintenance capabilities.
2. MAT*WorX* can be connected to the system either directly or remotely. Direct connection is through the RS connector on the MP card. Remote connection is available via either an internal modem on the MP card or an external modem for high speed. Remote connection via the internal modem is through the COT card. Connection between the modem and the COT is accomplished through internal switching of the TDSW. Remote connection via an external modem is through the MP card.
3. The following functions can be performed from MAT*WorX*:
   - System, station, and trunk data entry, change, and copy.
   - Loading, saving, and verification of system data to and from a disk.
   - ROM check readout of generic program.
   - Display of fault/fault cleared messages.
   - On-site or remote access to the system.
   - Printout of system data (when printer is connected to PC).
   - Display and setting of system clock/calendar.
   - Numbering Plan
   - Least Cost Routing (LCR)
   - System initialize
   - UCD/Station Hunting/Call Pickup – Group
4. The PC used with MAT*WorX* must have an RS-232C port, and cannot be located more than 50 feet (15m) from the system when connected on premises.
5. When stations or trunks are expanded, moved, or changed, office data for a Multiline Terminal key/station/ trunk can be copied and multiple assignments of related office data is possible.

## MACH Script Editor

This is a powerful timesaving tool that enables you to create a list of *UNIVERGE* NEAX 2000 IPS commands that perform tasks in the PBX. This list is referred to as a script, which can be saved and run at anytime. You can also use the MACH Script Editor to perform many other operations.

## Remote Maintenance

This feature allows station and trunk changes or reassignments to be performed without a site visit by service personnel, and can be used to retrieve fault codes prior to visiting a site. One Remote Maintenance center can service an unlimited amount of systems, thus reducing the amount of personnel to maintain each site, travel costs and customer billing for each site.

### Service Conditions

1. The following additional equipment is required for this feature:
   - A modem at the maintenance center and one at each remote site. (When the internal modem of the Main Processor (MP) is used, no modem at each remote site is required)
   - A cable for connection between the MP and the on-site modem. (When the internal modem of the MP is used, the above cable is not required)

2. The internal modem of the MP is compatible with the following specifications:

| | |
|---|---|
| ITU-T V.22 1200 bps | ITU-T V.32 4800/9600 bps |
| ITU-T V.22 bis 2400 bps | ITU-T V.34 19.2 k/33.6 kbps |
| Bell 212A 1200 bps | |

3. Any one of the following connections are also required for access to the modem:

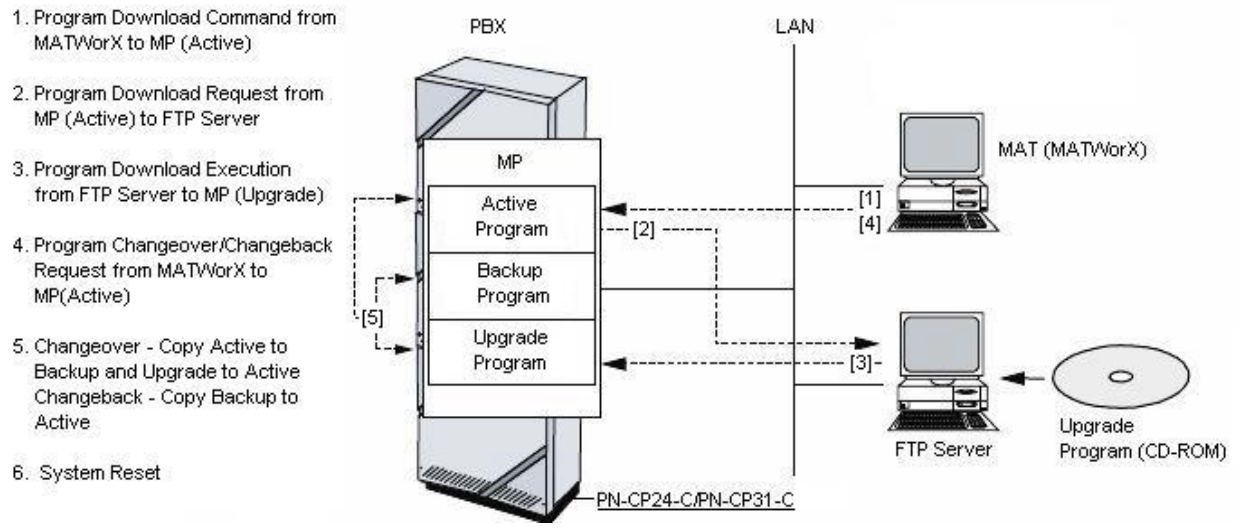
| | |
|---|---|
| A dedicated line | Attendant controlled transfer |
| Direct Inward System Access (DISA) | Direct Inward Termination |

4. The following operations can be executed from the Remote Maintenance location:
   - Retrieval of fault data
   - Retrieval of Peg Count information
   - Deletion or addition of system data (line, trunk, etc.) using a preprogrammed security password
   - Data assignment by device number (stations, trunks, and Attendant Console)
   - Copying of station data from one station to other stations (when adding sequential stations in groups)
   - Release / Reconnection of backup batteries
   - Display of station line status

# MP Program Download

The UNIVERGE NEAX 2000 IPS provides Online MP Program Download via IP network using the CP24D, CP31E, and CP31D. This feature allows an MP upgrade program to be downloaded to the UNIVERGE NEAX 2000 IPS MP card with the PBX on-line and during the download process all features and functions are available. The MP Program Download feature is available to a stand alone system or a Remote system in a Remote PIM network.

The IPS downloads the MP upgrade program from FTP server using MATWorX. MATWorX and the FTP server can be installed and running on the same PC. A single FTP server or multiple FTP servers can be used. Immediate or scheduled changeover to the upgrade program is available. It is also possible to change back to the previous program that was in use before the changeover (changeback).




**MP Download Process**

## Service Conditions

### General Service Conditions

- The MP program download can be executed to the PN-CP24-D/PN-CP31-D (MP card) when a PZ-M606-A is mounted. PZ-M697 (ETHER) card is mounted on the PN-CP31-E (MP) card.
- For the Retrofit and Backup CPU systems remote download of MP program is not available.
- The call processing is stopped while the system is initialized by the program changeover execution.
- Remote download of AP program is not available.
- While the PBX is off-line, you cannot execute the program downloading, or program changeover/changeback/program version matching (immediate). However, it is possible to read various information and set the schedule (date/time) for program changeover/changeback/program version matching even while the PBX is off-line.

### Program Download Service Conditions

- Do not reset the PBX and the MATWorX during MP program downloading.
- Do not pull out the LAN cable during MP program downloading.
- FTP (File Transfer Protocol) is used as the protocol for the remote download of MP program.
- Only the file specified in “File Name” of CHECKSUM.TXT can be downloaded to the PBX.
- If you do not set the directory name to save the program files and CHECKSUM.TXT by MATWorX, they are saved to the FTP directory (default).
- If you do not set the password and user ID of the FTP server by MATWorX, login to the FTP server with anonymous.
- If required FTP server information is not set when executing the MP program download, “DATA NOT SET” is displayed.
- For the TCP port, the specified port is used for file transfer (control), and the port in front of it is used for the file transfer. For example, when the TCP port is set to 3000, Port No. 3000 is used for the file transfer (control), and Port No. 2999 is used for the file transfer (Port 21-File transfer (control), Port 20-File transfer: default).
- If you read the issue number of the standby side (outdated side) and change the settings of FTP information during MP program downloading, “WAIT, BUSY NOW” is displayed.
- When using a single FTP server to update multiple sites in remote PIM network schedule the file transfer at least 7 minutes or more apart so that bandwidth and the number of connections to the FTP server are not exceeded.

**Program Changeover Service Conditions**

- The system is initialized automatically after executing the program changeover any data that is not backed up will lost. Be sure to execute the office data backup before executing the program changeover.
- If an error is detected during system initialization after executing the program changeover, the PBX starts automatically with the previous program.
- If the program download is executed while the program version matching is being executed, the program version matching is interrupted, and program download is executed.
- If the program changeover (immediate) is executed while the program version matching is being executed, the program is changed after the program version matching is completed.
- The program changeover cannot be executed if the program downloading or program version matching is interrupted or fails. If the program changeover (immediate) is set, the setting is invalid.
- The program changeover for which the schedule (date/time) has been set is executed only when the PBX is on-line.
- When immediate program changeover is executed after a schedule (date/time) for program changeover execution is set, the schedule is invalid.

### Required Hardware

| Equipment | Comments |
|---|---|
| CP24D (IPS/IPS Remote PIM)<br>CP31E (IPS$^{DML}$)<br>CP31D (IPS$^{DMR}$) | Main and Remote sites |
| Upgrade Program | |
| MATWorX IPS | Version 13.0.0 |
| PC/Work station for FTP<br>**Note** | Windows 2000 Server/Windows 2000 Professional Windows XP Professional (Internet Information Server (IIS) supplied with Windows 2000/XPProfessional can be used) |

**Note:** *MATWorX and FTP server program can be used on the same PC.*

# MA4000 Management System

MA4000 Management System is a web based, powerful voice server management configuration suite. MA4000 offers centralized management for the *UNIVERGE* NEAX 2000 IPS Voice Server as well as simple and powerful tools for managing moves, adds and changes. MA4000 also offers flexibility and security and allows the end user to manage their voice server in the same ways they manage their networks.

- Web Browser Operation
- Email Alarm Notification
- Alarm Client Notification
- LDAP Auto Provisioning Service
- LDAP Authentication
- Windows Authentication
- IT Friendly Interface
- Flexible Access Levels (Roles)
- Application Program Interface (API & SDK)
- Security
  - HTTPS supported
  - Audit Trail Logging
- Customizable Portals
- Command Line Interface
- Integration to OpenWorX
- Centralized Authentication Service (NEC CAS)
- System Health Monitoring
- Range programming that is schedulable

The MA4000 IPS Assistant gives the power of station management directly to the end-user. Limited by access rights, an administrator can allow his users to manage their own phones. Supported features include Button Programming, Call Forwarding and Speed Dial Programming.

System Requirement

The following are the specifications for the Web Client machines. These machines only access the MA4000 via a web browser and have no actual software installed on them.

| WEB Client | Minimum | Recommended |
|---|---|---|
| Processor: | Intel® Pentium® 450 MHz | Intel Pentium 700 MHz or higher |
| RAM: | 64 MB | 128 MB or more |
| Hard Drive Space: | 10 MB | 100 MB or more |
| Video: | 800 x 600 SVGA Monitor | 1024 x 768 SVGA Monitor |
| Web Browser: | Any HTTP 1.1 compliant | Internet Explorer 5.5 or greater |
| Input Device: | Mouse & 101 Key Keyboard | Mouse & 101 Key Keyboard |
| Operating System: | Windows® 98 SE | Windows 2000/XP with latest Service Pack |

The following requirements are for a single user system running the MA4000 Management System software.

| WEB Server | Minimum | Recommended |
|---|---|---|
| Processor: | Intel® Pentium® 450 MHz | Intel Pentium 2 GHz or higher |
| RAM: | 1 GB (1024 MB) | 1 GB (1024 MB) |
| Hard Drive Space: | 500 MB | 500 MB or more |
| Video: | 800 x 600 SVGA Monitor | 1024 x 768 SVGA Monitor |
| Web Browser: | Any HTTP 1.1 compliant | Internet Explorer 5.5 or greater |
| Drives: | CD ROM | CD ROM |
| Input Device: | Mouse & 101 Key Keyboard | Mouse & 101 Key Keyboard |
| Ethernet Port: | 10/100 MB | 10/100 MB |
| USB Port: | At least one unused USB Port | At least one unused USB Port |
| Web Server Used: | Microsoft Internet Information Server version 5.0 or higher | Microsoft Internet Information Server version 5.0 or higher |
| Operating System: | Windows® 2000/XP Professional with latest Service Pack | Windows 2000/2003 Server with latest Service Pack |
| Database: | Microsoft® SQL MSDE with the latest Service Pack | Microsoft SQL 2000 Personal or Standard depending on the OS with the latest Service Pack |

The following requirements are for a single user system running the MA4000 Management System software.

| WEB Server | Minimum | Recommended |
|---|---|---|
| Processor: | Intel® Pentium® 450 MHz | Intel Pentium 2 GHz or higher |
| RAM: | 512 MB | 512 MB |
| Hard Drive Space: | 500 MB | 500 MB or more |
| Video: | 800 x 600 SVGA Monitor | 1024 x 768 SVGA Monitor |
| Web Browser: | Any HTTP 1.1 compliant | Internet Explorer 5.5 or greater |
| Drives: | CD ROM | CD ROM |
| Input Device: | Mouse & 101 Key Keyboard | Mouse & 101 Key Keyboard |
| Ethernet Port: | 10/100 MB | 10/100 MB |
| USB Port: | At least one unused USB Port | At least one unused USB Port |
| Web Server Used: | Microsoft Internet Information Server version 5.0 or higher | Microsoft Internet Information Server version 5.0 or higher |
| Operating System: | Windows® 2000/XP Professional with latest Service Pack | Windows 2000/2003 Server with latest Service Pack |
| Database: | Microsoft® SQL MSDE with the latest Service Pack | Microsoft SQL 2000 Personal or Standard depending on the OS with the latest Service Pack |

## Equipment List

| DESCRIPTION | NOTES |
|---|---|
| MA4000 Management Software | Includes all MA4000 Programs; System Manager and a USB Key |
| MA4000 SQL Processor Lic (1) | Includes MS SQL 2000 Standard & Personal DB (1 CD) Includes one processor license for unlimited client browser access to the MA4000 Manager & Assistant system |
| MA4000 CUST Provided DB Option | Enables MA4000 to work with a Customer Provided MS SQL 2000 Database. Customer is responsible for providing the database and for all licensing |
| MA4000 MSDE DB Option | Enables MA4000 to work with MSDE Database. Simultaneous client-browser access is limited to five |
| MA4000 SQL 2000 Generic PKG | Provides Microsoft SQL 2000 database CD with one folder containing Personal Edition and one folder containing Standard Edition |
| MA4000 IPS Manager Option | 1 required when managing UNIVERGE NEAX 2000 IPS Voice Servers |
| MA4000 IPS Ext Lic (1) | 1 License is required for each IPS extension to be managed |
| MA4000 IPS Assistant Lic. (1) | 1 License is required for each IPS extension using the Desktop interface |
| MA4000 LDAP APS Option | Enables the MA4000 LDAP Auto Provisioning Service (LAPS) |

## Optional Server and Hardware

| DESCRIPTION | NOTES |
|---|---|
| 120Rd-1 w/ Win2K3 Server **Note** | Includes 2.8 GHz Processor, 1GB RAM, 1-36GB Hard Drive, Windows 2003 Server Operating System |
| Logitech PS/2 Wheelmouse OEM Ivory | Logitech Mouse |
| Intel 104 Keyboard PS/2 | Intel Keyboard |
| NEC Acusync AS500 Monitor | NEC 15” Monitor |
| 2.8GHz/512KB CPU KIT (120Rd-1) | Additional CPU for 1U Servers |
| 512MB LP SDRAM (2x256MB) KIT120Rd-1 | Additional 512MB RAM for 1U Servers. 1U Servers support up to 1.5GB of RAM |
| HDD 36G 15K RPM HOTPLUG KIT | Additional Hard Drive for 1U Servers. 1U Servers support up to 3 Hard Drives. |
| 1-CHANNEL U320 RAID w/64M CACHE (120Lg) | RAID Controller for RAID 1 (Two Hard Drives), RAID 5 (Three Hard Drives) |

**Note:** *Server above does not include Keyboard, Mouse or Monitor. These items must be added separately*

# *UNIVERGE* NEAX 2000 IPS Documentation List

NEC offers a full complement of documents for the *UNIVERGE* NEAX 2000 IPS product line. Technical documentation is available on Compact Disk (CD ROM) or on the WEB through NTAC On-Line. This section lists all documents included on the Compact Disk (CD ROM).

| ***UNIVERGE* NEAX 2000 IPS Technical Manuals** | |
|---|---|
| Call CenterWorX Business System Manual | Maintenance Manual |
| Call CenterWorX MIS Admin Manual | OAI System Manual |
| Call CenterWorX MIS Installation Manual | Office Data Programming |
| Call CenterWorX MIS Reports Manual | Q-SIG System Manual |
| CCIS System Manual | Q-SIG System Manual (PRT) |
| Command Manual | Retrofit System Guide |
| DML Hardware Installation Guide (440 Port) | Request for Proposal |
| DRS Installation and Configuration Guide | Software Upgrade Guide |
| Feature Programming Manual | SNMP Implementation |
| General Description | System Manual |
| INASET Installation Guide | Upgrade Guide |
| Installation Procedure Manual | WCS System Manual (PCS) |
| IPS/DML Configuration Guide | Wireless System Manual |
| ISDN System Manual | WLAN System Manual |
| MatWorX Installation Guide | |

| ***UNIVERGE* NEAX 2000 IPS Features and Specification** | |
|---|---|
| Business Hotel Features and Specifications | ISDN/Q-SIG Features and Specifications |
| CCIS Features and Specifications | SMDR/MCI/PMS Interface Specification |
| CCWX ACD Features and Specifications | WCS Features and Specifications |

| ***UNIVERGE* NEAX 2000 IPS User Manuals** | |
|---|---|
| Dterm Assistant IPS User Guide | ITR-4D-3 Dterm IP Installation Guide |
| Dterm Series E Agent Console User Guide | INASET Installation Guide |
| Dterm Series E Supervisor Console User Guide | INASET for the NEAX 2000 IPS User Guide |
| Dterm Series E Telephone User Guide | MatWorX IPS User Guide |
| Dterm Series I / Dterm IP User Guide | Power Patch Panel User Guide |
| Dterm Series I and Dterm IP Installation Guide | SN716 Desk Console User Guide |

# Chapter 7 Specifications

## Processors

The *UNIVERGE* NEAX 2000 IPS, IPS$^{DML}$, and IPS$^{DMR}$ are distributed multiprocessor systems. Their control system consists of a Main Processor (MP), Firmware Processors (FP), and Application Processors (AP). Both the FP and APs execute their predetermined functions under the control of the MP.

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| Control System | Stored Program Control |
| Processor Type | 32-Bit (ElanSC520) |
| Program Storage | Flash ROM |
| Office Data Storage | Flash ROM |
| Processor Architecture | Central |
| Program Updates | Floppy Disk |
| Time Division Matrix | PCM Time Division (1,024 x 1.024; Non-Blocking) |

### Memory

| CARD NAME | PROCESSOR TYPE | MEMORY CAPACITY | |
|---|---|---|---|
| | | ROM | RAM |
| MP (PN-CP24-D) | 32-Bit (ElanSC520) | 16MB (Flash ROM) | 64MB (SDRAM) |
| MP (PN-CP27-B) | 32-Bit (ElanSC520) | 8MB (Flash ROM) | 32MB (SDRAM) |
| MP (PN-CP31-E) | 32-Bit (ElanSC520) | 16MB (Flash ROM) | 64MB (SDRAM) |
| MP (PN-CP31-D) | 32-Bit (ElanSC520) | 16MB (Flash ROM) | 64MB (SDRAM) |
| FP (PN-CP15) | 16-bit (25MHz) | === | 768KB |
| FP (PN-CP33) | 16-bit (25MHz) | === | 768KB |
| AP (PN-AP00-B) | 16-bit (8MHz) | 512KB (Flash ROM) | 512KB |

## Common Control Cards

| Name Code | Remarks |
|---|---|
| PN-CP24-D | Main Processor Card for *UNIVERGE* NEAX 2000 IPS.<br>One card is required per system. |
| PN-CP27-B | Main Processor Card for *UNIVERGE* NEAX 2000 IPS Dual MP System.<br>One card is required per system. |
| PN-CP31-E | Main Processor Card for *UNIVERGE* NEAX 2000 IPS$^{DML.}$<br>One card is required per system. |
| PN-CP31-D | Main Processor Card for *UNIVERGE* NEAX 2000 IPS$^{DMR.}$<br>One card is required for each Remote Site. |
| PZ-M606A<br>(for IPS and DMR) | Ethernet Control Card:<br>• Mounted on MP card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol.<br>• 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card. |
| PZ-M697<br>(for DML) | Ethernet Control and inter-module BUS interface Card:<br>• Mounted on PN-CP31-E card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol.<br>• 10BASE-T/100BASE-TX twisted pair cable is connected directly to this card.<br>• BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS.<br>• Occupies two physical slots width per card on PN-CP31-E card. |
| PN-CP15<br>(for IPS) | Firmware Processor Card:<br>Provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. When the system consists of three PIMs or more, one each of this card is mounted respectively in PIM2, PIM4, and PIM6. |
| PN-CP33<br>(for DML) | Firmware Processor Card<br>Provides Line/Trunk interface, Memory (RAM 768 KB), and inter-module BUS interface. BUS interface functions as a driver/receiver of various signals, adjusts gate delay timing and cable delay timing, monitors I/O Bus and PCM BUS. When the system consists of three MCs or more, one each of this card is mounted respectively in the MP slot of MC2, MC4, and MC6. |

Major specifications and functionality of the *UNIVERGE* NEAX IPS MPs are shown below:

| Item | | PN-CP24-D<br>PN-CP27-B |
|---|---|---|
| Central Processing | | ElanSC520 |
| System Memory | | CP24D: Flash ROM (16MB), SDRAM (64MB)<br>CP27B: Flash ROM (8MB), SDRAM (32MB) |
| Network Switching | | 1,024 · 1,024 Time Division Switch **(Note 1)** |
| 3-Way Conference | | 16 sets of 3-way conference circuitry |
| DTMF Signal Sender | | 32 circuits (digit 0 to 9, *, and # are generated) |
| Music-on-Hold | | 10 types are available **(Note 2)** |
| Mini Jack | | 1 for External Music Source for MOH **(Note 2)** |
| Audible Tone Generator (DTG) | | Available |
| Phase Lock Oscillator (PLO) | | 2 ports (Source/Receiver) |
| Built-in SMDR | | Available |
| Built-in MCI | | Available |
| Built0in PMS | | Available |
| Built-in FP0 | | Available |
| BS00 Function | | Available |
| DTMF Receiver | | 4 circuits |
| AP01 Function | | Available |
| Built-in DRS | | Available |
| Virtual IPT / Virtual CSH | | Available |
| System data copy from active MP to standby MP | | CP27B (Dual MP) only |
| MP Program Download (FTP) | | Available with CP24D only **(Note 3)** |
| MAT Interface | Direct Connection | 1 port |
| | Remote Connection w/Built-in MODEM | 1 port **(Note 4)** |
| | LAN Connection | Available |
| External Alarm Indication | | MJ and MN |
| DAT | | 2circuits (120 seconds per circuit) |
| DK00 | | 2 circuits (relay drive x1, external key scan x1) |
| Application Key Program | | In EPROM |

**Note 1:** *Refer to the Figure 4-2 on next page for the port allocation of the Time Division Switch.*

**Note 2:** *In case of Dterm IP, the synthesized melody is provided built in to IP adapter.*

**Note 3:** *MP Program Download (FTP) is for Stand Alone Systems Only (CP24D.)*

**Note 4:** *No. 1 Port includes a built-in modem for a remote connection of the MAT.*

Major specifications and functionality of the *UNIVERGE* NEAX IPS$^{DML}$ MP are shown below:

| Item | | PN-CP31-E |
|---|---|---|
| Central Processing | | ElanSC520 |
| System Memory | | CP31E: Flash ROM (16MB), SDRAM (64MB) |
| Network Switching | | 1,024 · 1,024 Time Division Switch **(Note 1)** |
| 3-Way Conference | | 16 sets of 3-way conference circuitry |
| DTMF Signal Sender | | 32 circuits (digit 0 to 9, *, and # are generated) |
| Music-on-Hold | | 10 types are available **(Note 2)** |
| Mini Jack | | 1 for External Music Source for MOH **(Note 2)** |
| Audible Tone Generator (DTG) | | Available |
| Phase Lock Oscillator (PLO) | | 2 ports (Source/Receiver) |
| Built-in SMDR | | Available |
| Built-in MCI | | Available |
| Built0in PMS | | Available |
| Built-in FP0 | | Available |
| BS00 Function | | Available |
| DTMF Receiver | | 4 circuits |
| AP01 Function | | Available |
| Built-in DRS | | Available |
| Virtual IPT / Virtual CSH | | Available |
| MP Program Download (FTP) | | Available |
| MAT Interface | Direct Connection | 1 port |
| | Remote Connection w/Built-in MODEM | 1 port |
| | LAN Connection | Available |
| External Alarm Indication | | MJ and MN |
| DAT | | 2circuits (120 seconds per circuit) |
| DK00 | | 2 circuits (relay drive x1, external key scan x1) |
| Application Key Program | | In EPROM |

**Note 1:** *Refer to the Figure 4-2 on next page for the port allocation of the Time Division Switch.*

**Note 2:** *In case of Dterm IP, the synthesized melody is provided built in to IP adapter.*

Major specifications and functionality of the *UNIVERGE* NEAX IPS$^{DMR}$ MP are shown below:

| Item | PN-CP31-D |
|---|---|
| Central Processing | ElanSC520 |
| System Memory | CP31D: Flash ROM (16MB), SDRAM (64MB) |
| Network Switching | Remote PIM |
| 3-Way Conference | 16 sets of 3-way conference circuitry |
| DTMF Signal Sender | 32 circuits (digit 0 to 9, *, and # are generated) |
| Mini Jack | 1 for External Music Source for MOH |
| Audible Tone Generator (DTG) | Available |
| Phase Lock Oscillator (PLO) | 2 ports (Source/Receiver) |
| Built-in FP0 | Available |
| DTMF Receiver | 4 circuits |
| Built-in DRS | Available |
| MP Program Download (FTP) | Available |

### Firmware Processor (FP)

Firmware Processors (FP) are required when more than two PIMs/Modular Chassis (MC) are used. The FP provides supervision and status analysis of line/trunk ports, which reside in the MC or PIM. The FP provides the bus interface for I/O Bus, PCM Bus, and Alarm Bus in a multiple-PIM configuration. The major specifications of the FP are shown below:

- Central Processor Unit: 16-bit (25 MHz)
- Memory: Program Area (384 kb), Work Area (384 kb)
- BS01 Function

| Name Code | Remarks |
|---|---|
| PN-CP15 | Firmware Processor Card used with the *UNIVERGE* NEAX 2000 IPS. |
| PN-CP33 | Firmware Processor Card used with the *UNIVERGE* NEAX 2000 IPS$^{DML}$. |

# Power

## AC Power Requirements

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| AC Input Voltage | 90 to 132Vac or 180 to 264Vac; 47 to 64Hz |
| AC Input Current | 3.5A(at 100V), 2.0A(at 200V) |

## AC Power Consumption / Thermal Output (Maximum)

| DESCRIPTION | AC Power Consumption (KVA) | | Thermal Output (BTU) | |
|---|---|---|---|---|
| | 100V | 200V | 100V | 200V |
| 1-PIM/MC | 0.35 | 0.40 | 1,195 | 1,365 |
| 2-PIM/MC | 0.70 | 0.80 | 2,389 | 2,730 |
| 3-PIM/MC | 1.05 | 1.20 | 3,584 | 4,096 |
| 4-PIM/MC | 1.40 | 1.60 | 4,778 | 5,461 |
| 5-PIM/MC | 1.75 | 2.00 | 5,973 | 6,826 |
| 6-PIM/MC | 2.10 | 2.40 | 7,167 | 8,191 |
| 7-PIM/MC | 2.45 | 2.80 | 8,362 | 9,556 |
| 8-PIM/MC | 2.80 | 3.20 | 9,556 | 10,922 |

## Battery Requirements

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| Max. Battery Capacity | 260AH per 4 PIM (65AH (12V) x 8) |
| DC Input Voltage for Battery | -24V |
| Built-in Battery Requirements | 3.4AH (12V) x 2 (approx. 30min. backup) |
| Physical Size of Built-in Battery (one 12V battery) | 133(W) x 60(H) x 67(D) mm |

## Operating Environment

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| Ambient Temperature | 0 to 40□ |
| Relative Humidity | Max. 90% (non-condensing) |

## Electrical Characteristics (Central Office Trunk)

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| Insulation Resistance | 15 mega-ohms or more at 100Vdc |
| DC Resistance | On-hook conditions: 30 mega-ohms<br>Off-hook conditions: 1,700 ohms |
| Impedance | On-hook conditions: 20 kilo-ohms (300 to 3,400Hz)<br>8 kilo-ohms (at 24Hz)<br>Looped conditions: 600 ohms |
| Leak Current | 0 mA at on-hook conditions |

### Transmission Characteristics (For TDM Circuits)

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| PCM Coding System | A-law/U-law |
| Insertion Loss | 0.15 dB at 1KHz |
| Return Loss | 20 dB or more (300 to 3,400Hz) against 600 ohms |
| Longitudinal Balance | 59 dB or more (300 to 3,400Hz) |
| Attenuation/Frequency Distortion | -0.2 dB to +0.7 dB (300 to 3,400Hz) |
| Group Delay Distortion | 0 to 0.3msec. (500 to 2,800 Hz) |
| Total Distortion | 25 dB (Input signal:-45 dBm0)/40 dB (input signal:0 dBm0) |
| Idle Channel Noise | -67 dBmop or less (psophometric noise)<br>-50 dBm0 or less (single frequency noise) |
| Impulsive Noise | 0 counts at -35 dBm |
| Cross Talk Attenuation | 90 dB or more |
| Inter-modulation Products | -40 dB or more |
| Spurious In-Band Signals | -49 dBm0 or less |
| Signal Attenuation | Attenuation rate: 12 dB per octave or more at 3.4 kHz above<br>Attenuation level: -40 dBm or less at 3.4 kHz and above -70 dBm or less at 50 kHz and above |

# System Capacity

### IPS System Capacity (Single MP System)

| Item | | System Capacity Note 1 | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| | | **1PIM** | **2PIM** | **3PIM** | **4PIM** | **5PIM** | **6PIM** | **7PIM** | **8PIM** |
| LT Card | (No. of Ports) | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| | (No. of Cards) | 12 | 24 | 36 | 48 | 60 | 72 | 84 | 96 |
| AP Card | (No. of Ports) | Max. 256 ports per system | | | | | | | |
| | (No. of Cards) | 12 | 24 | | | | | | |
| Total Number of Lines (Analog Single Line Tel. + Dterm) | | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| IP PAD (No. of Channel) **Note 2** | | 64 | | 128 | | 192 | | 256 | |
| Analog Single Line Telephone (Lines) | Standard | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| | Long **Note 3** | 48 | 96 | 144 | 192 | 240 | 288 | 336 | 384 |
| Dterm (Lines) | Standard | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| | Long | 24 | 48 | 72 | 96 | 120 | 144 | 168 | 192 |
| DtermIP/ INASET/ SP30 (Peer to Peer Connection) | | 952 | 888 | 824 | 760 | 696 | 632 | 568 | 504 |
| Dterm PS | | 512 | | | | | | | |
| Zone Transceiver (ZT) | | 16 | 32 | 48 | 64 | 80 | 96 | 112 | 128 |
| ISDN Terminal (2ILCC) | P-P Connection | 24 | 32 | 32 | 32 | 32 | 32 | 32 | 32 |
| | P-MP Connection | 48 | 64 | 64 | 64 | 64 | 64 | 64 | 64 |
| Central Office Trunk (Lines) | Loop Start | 64 | 128 | 192 | 256 | 256 | 256 | 256 | 256 |
| | DID w/4DIT | 48 | 96 | 144 | 192 | 240 | 256 | 256 | 256 |
| Tie Line Trunk (Lines) | 2W or 4W E&M | 48 | 96 | 144 | 192 | 240 | 256 | 256 | 256 |
| | Type 1 w/PW122 | 12 | 24 | 36 | 48 | 60 | 72 | 84 | 96 |
| CCIS Trunk (Peer to Peer Connection) | | Max. 127 | | | | | | | |
| DTI/CCIS Digital Link **Note 4** | 1.5M | DTI: 10, CCIS: 8 | | | | | | | |
| | 2M | 8 | | | | | | | |
| ISDN | 1.5M (PRT) | 8 | | | | | | | |
| | 4BRT (card) | 6 | 12 | 18 | 24 | 24 | 24 | 24 | 24 |
| IP Trunk | | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
| PFT Connections | 8PFT | 8 | 16 | 24 | 32 | 40 | 48 | 56 | 64 |
| 3-Party Conference | | Max. 16 conference groups per system | | | | | | | |
| 8-Party Conference Group | | Max. 8 conference group per system | | | | | | | |
| 32-Party Conference Group | | Max. 8 conference group per system | | | | | | | |
| DTMF Sender | | Max. 32 circuits per system | | | | | | | |
| DTMF Receiver | | 16 | | DTMF Receiver | | | | | |
| SN716 Desk Console | | 8 | | | | | | | |
| Attendant Terminal (Dterm ATT Position) | | Max. 8 sets per system | | | | | | | |

**Note 1:** *"PIM" in this table means "Physical PIM."*
**Note 2:** *Number of IP PAD card is maximum 2 per FP.*
**Note 3:** *Message Waiting Lamp is not available when the 4LLC is mounted in slot 08 to 11.*
**Note 4:** *The total number of trunk line and DTI channel shall be 256 or less. (Each trunk line and DTI channel are required to assign the "Trunk Number" by system data programming and maximum number of system parameter for "Trunk Number" is 256.)*

## IPS System Capacity (Single MP System cont'd.)

| Item | System Capacity | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| | 1PIM | 2PIM | 3PIM | 4PIM | 5PIM | 6PIM | 7PIM | 8PIM |
| SMDR Interface | Max. 2 interface ports (RS-232C) / Max.1 interface port (IP) per system | | | | | | | |
| PMS Interface | Max. 1 interface port (RS-232C) / Max. 1 interface port (IP) -per system | | | | | | | |
| ACD / MIS or OAI Interface **Note 5** | Max. 1 interface port per system | | | | | | | |
| Remote PIM over IP (Number of Sites) | Up to 30 (depending on network configurations) | | | | | | | |
| DID Dial Conversion | 1000 | | | | | | | |
| Call Forwarding-Outside Set | 496 | | | | | | | |
| Authorization Code / Forced Account Code / Remote Access to System (DISA) Code | 3000 | | | | | | | |
| Message Reminder Set | 1024 | | | | | | | |
| Name Display / Guest Name Display | 512 | | | | | | | |
| Speed Calling-Station (Station Speed Dial) Set | 10000 | | | | | | | |
| MP built-in SMDR Call Record | 1024 | | | | | | | |

**Note 5:** *It is possible to connect 4 applications at the same time.*

## IPS System Capacity (Dual MP System)

| Item | | System Capacity Note 1 | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| | | 1PIM | 2PIM | 3PIM | 4PIM | 5PIM | 6PIM | 7PIM | 8PIM |
| LT Card | (No. of Ports) | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| | (No. of Cards) | 11 | 23 | 35 | 47 | 59 | 71 | 83 | 95 |
| AP Card | (No. of Ports) | Max. 256 ports per system | | | | | | | |
| | (No. of Cards) | 11 | 23 | 24 | | | | | |
| Total Number of Lines (Analog Single Line Tel. + Dterm) | | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| IP PAD (No. of Channel) **Note 2** | | 64 | | 128 | | 192 | | 256 | |
| Analog Single Line Telephone (Lines) | Standard | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| | Long **Note 3** | 44 | 92 | 140 | 188 | 236 | 284 | 332 | 380 |
| Dterm (Lines) | Standard | 64 | 128 | 192 | 256 | 320 | 384 | 448 | 512 |
| | Long | 22 | 46 | 70 | 94 | 118 | 142 | 166 | 190 |
| DtermIP/ INASET / SP30 (Peer to Peer Connection) | | 952 | 888 | 824 | 760 | 696 | 632 | 568 | 504 |
| Dterm PS | | 512 | | | | | | | |
| Zone Transceiver (ZT) | | 16 | 32 | 48 | 64 | 80 | 96 | 112 | 128 |
| ISDN Terminal (2ILCC) | P-P Connection | 24 | 32 | 32 | 32 | 32 | 32 | 32 | 32 |
| | P-MP Connection | 48 | 64 | 64 | 64 | 64 | 64 | 64 | 64 |
| Central Office Trunk (Lines) | Loop Start | 64 | 128 | 192 | 256 | 256 | 256 | 256 | 256 |
| | DID w/4DIT | 44 | 92 | 140 | 188 | 236 | 256 | 256 | 256 |
| Tie Line Trunk (Lines) | 2W or 4W E&M | 48 | 96 | 144 | 192 | 240 | 256 | 256 | 256 |
| | Type 1 w/PW122 | 12 | 24 | 36 | 48 | 60 | 72 | 84 | 96 |
| CCIS Trunk (Peer to Peer Connection) | | Max. 127 | | | | | | | |
| DTI/CCIS Digital Link **Note 4** | 1.5M | DTI: 10, CCIS: 8 | | | | | | | |
| | 2M | 8 | | | | | | | |
| ISDN | 1.5 (PRT) | 8 | | | | | | | |
| | 4BRT (card) | 6 | 12 | 18 | 24 | 24 | 24 | 24 | 24 |
| IP Trunk | | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
| PFT Connections | 8PFT | 8 | 16 | 24 | 32 | 40 | 48 | 56 | 64 |
| 3-Party Conference | | Max. 16 conference groups per system | | | | | | | |
| 8-Party Conference Group | | Max. 8 conference group per system | | | | | | | |
| 32-Party Conference Group | | Max. 8 conference group per system | | | | | | | |
| DTMF Sender | | Max. 32 circuits per system | | | | | | | |
| DTMF Receiver | | 16 | | 32 | | | | | |
| SN716 Desk Console | | 8 | | | | | | | |
| Attendant Terminal (Dterm ATT Position) | | Max. 8 sets per system | | | | | | | |

**Note 1:** *"PIM" in this table means "Physical PIM."*
**Note 2:** *Number of IP PAD card is maximum 2 per FP.*
**Note 3:** *Message Waiting Lamp is not available when the 4LLC is mounted in slot 08 to 11.*
**Note 4:** *The total number of trunk line and DTI channel shall be 256 or less. (Each trunk line and DTI channel are required to assign the "Trunk Number" by system data programming and maximum number of system parameter for "Trunk Number" is 256.)*

## IPS System Capacity (Dual MP System, Cont'd)

| Item | System Capacity | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| | 1PIM | 2PIM | 3PIM | 4PIM | 5PIM | 6PIM | 7PIM | 8PIM |
| SMDR Interface | Max. 2 interface ports per system (RS-232C)<br>Max.1 port per system (IP) | | | | | | | |
| PMS Interface | Max. 1 interface port (RS-232C)<br>/ Max. 1 interface port (IP) -per system | | | | | | | |
| ACD / MIS or OAI Interface **Note 5** | Max. 1 interface port per system | | | | | | | |
| Remote PIM over IP (Number of Sites) | Up to 30 (depending on network configurations) | | | | | | | |
| DID Dial Conversion | 1000 | | | | | | | |
| Call Forwarding-Outside Set | 496 | | | | | | | |
| Authorization Code / Forced Account Code / Remote Access to System (DISA) Code | 3000 | | | | | | | |
| Message Reminder Set | 1024 | | | | | | | |
| Name Display / Guest Name Display | 512 | | | | | | | |
| Speed Calling-Station (Station Speed Dial) Set | 10000 | | | | | | | |
| MP built-in SMDR Call Record | 1024 | | | | | | | |

**Note 5:** *It is possible to connect 4 applications at the same time.*

## IPS$^{DML}$ System Capacity

| Item | | System Capacity Note 1 | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| | | 1 MC | 2 MC | 3 MC | 4 MC | 5 MC | 6 MC | 7 MC | 8 MC |
| Physical Module Chassis | | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
| Physical MC Port for Line/Trunk | | 48 | 104 | 160 | 216 | 272 | 328 | 384 | 440 |
| Built in FP (on MP) | | 1 | | | | | | | |
| FP card (PN-CP33) | | 0 | | 1 | | 2 | | 3 | |
| IP PAD (No. of Channel) | **Note 2** | 32 | 64 | 128 | | 192 | | 256 | |
| Analog Single Line Telephone (Lines) | Standard | 48 | 104 | 160 | 216 | 272 | 328 | 384 | 440 |
| | Long | 12 | 24 | 36 | 48 | 60 | 72 | 84 | 96 |
| Dterm (Lines) | Standard | 48 | 104 | 160 | 216 | 272 | 328 | 384 | 440 |
| | Long | 8 | 16 | 24 | 32 | 40 | 48 | 56 | 64 |
| DSS/BLF Console | | 8 | | 16 | | 24 | | 32 | |
| DtermIP/ INASET / SP30 (Peer to Peer Connection) | | 956 | 892 | 828 | 764 | 700 | 636 | 572 | 508 |
| Dterm PS | | 512 | | | | | | | |
| Zone Transceiver (ZT) | | 16 | 32 | 48 | 64 | 80 | 96 | 112 | 128 |
| ISDN Terminal (2ILCC) | P-P Connection | 12 | 26 | 32 | | | | | |
| | P-MP Connection | 24 | 52 | 64 | | | | | |
| Central Office Trunk (Lines) | Loop Start | 48 | 104 | 160 | 216 | 256 | | | |
| | DID w/4DIT | 24 | 52 | 80 | 108 | 136 | 164 | 192 | 220 |
| Tie Line Trunk (Lines) | 2W or 4W E&M | 24 | 52 | 80 | 108 | 136 | 164 | 192 | 220 |
| | Type 1 w/PW122 | 4 | 8 | 12 | 16 | 20 | 24 | 28 | 32 |
| CCIS Trunk (Peer to Peer Connection) | | Max. 127 | | | | | | | |
| DTI/CCIS Digital Link **Note 3** | 1.5M | DTI: 10, CCIS: 8 | | | | | | | |
| ISDN | 1.5 (PRT) | 8 Main, 32 Total | | | | | | | |
| | 4BRT (card) | 5 | 10 | 15 | 20 | 24 | | | |
| IP Trunk (IPT card or H.323 card) | | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
| PFT Connections | 4PFTA | 4 | 8 | 12 | 16 | 20 | 24 | 28 | 32 |
| 3-Party Conference | | Max. 16 conference groups per system | | | | | | | |
| 8-Party Conference Group | | Max. 8 conference group per system | | | | | | | |
| 32-Party Conference Group | | Max. 8 conference group per system | | | | | | | |
| DTMF Sender | | Max. 32 circuits per system | | | | | | | |
| DTMF Receiver | | 16 | | 32 | | | | | |
| SN716 Desk Console | | 8 | | | | | | | |
| Attendant Terminal (Dterm ATT Position) | | Max. 8 sets per system | | | | | | | |

**Note 1:** *"MC" in this table means "Physical Module Chassis."*

**Note 2:** *Number of IP PAD card is maximum 2 per FP.*

**Note 3:** *The total number of trunk line and DTI channel shall be 256 or less. (Each trunk line and DTI channel are required to assign the "Trunk Number" by system data programming and maximum number of system parameter for "Trunk Number" is 256.)*

## IPS$^{DML}$ System Capacity (Cont'd.)

| Item | System Capacity | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| | 1 MC | 2 MC | 3 MC | 4 MC | 5 MC | 6 MC | 7 MC | 8 MC |
| SMDR Interface | Max. 2 interface ports per system (RS-232C)<br>Max.1 port per system (IP) | | | | | | | |
| PMS Interface | Max. 1 interface port (RS-232C)<br>/ Max. 1 interface port (IP) -per system | | | | | | | |
| ACD / MIS or OAI Interface **Note 4** | Max. 1 interface port per system | | | | | | | |
| Remote PIM over IP (Number of Sites) | Up to 30 (depending on network configurations) | | | | | | | |
| DID Dial Conversion | 1000 | | | | | | | |
| Call Forwarding-Outside Set | 496 | | | | | | | |
| Authorization Code / Forced Account Code / Remote Access to System (DISA) Code | 3000 | | | | | | | |
| Message Reminder Set | 1024 | | | | | | | |
| Name Display / Guest Name Display | 512 | | | | | | | |
| Speed Calling-Station (Station Speed Dial) Set | 10000 | | | | | | | |
| MP built-in SMDR Call Record | 1024 | | | | | | | |

**Note 4:** *It is possible to connect 4 applications at the same time.*

# IP Remote Network Capacity

### Total System Capacity (Main plus Remote)

| Item | | Capacity |
|---|---|---|
| LT Ports | | 1020 |
| AP Ports | | 256 per site |
| Analog Single Line Tel. + Dterm (all stations) | | 980 |
| IP PAD (No. of Channel) | | 256 |
| DtermIP/INASET (PTP Connection) | | 952 |
| PS | | 512 |
| Zone Transceiver (ZT) | | 128 |
| ISDN Terminal | | 64 |
| Central Office Trunk (Lines) | | 256 |
| Tie Line Trunk (Lines) | 2W/4W E&M | 192 |
| CCIS Trunk (Peer to Peer Connection) | | 127 |
| DTI/CCIS Digital Link | 1.5M/2M | DTI: 10/CCIS: 8 Links |
| ISDN | 1.5M (PRT) | 8 Main / 32 Total |
| | 4BRT (card) | 24 |
| IP Trunk | | 8 |
| PFT Connections | | 64 |
| 3-Party Conference | | Max. 16 conference groups |
| 8-Party Conference Group | | Max. 8 conference groups |
| 32-Party Conference | | Max. 8 conference groups |
| DTMF Sender/Receiver | | Max. 32 circuits |
| Attendant Consoles | | 8 |
| Attendant Terminal (Dterm ATT Position) | | Max. 8 sets |
| SMDR Interface | | Max. 2 interface ports per system (RS-232C)<br>Max.1 port per system (IP) |
| PMS Interface | | Max. 1 interface port (RS-232C)<br>/ Max. 1 interface port (IP) -per system |
| ACD / MIS or OAI Interface | | Max. 1 interface port |
| Remote PIM over IP | | Up to 30 (depending on network) |
| DID Dial Conversion | | 1000 |
| Call Forwarding-Outside Set | | 496 |
| Authorization Code / Forced Account Code / Remote Access to System(DISA)Code | | 3000 |
| Message Reminder Set | | 1024 |
| Name Display / Guest Name Display | | 512 |
| Speed Calling-Station (Station Speed Dial) Set | | 10000 |
| MP built-in SMDR Call Record | | 1024 |

## IPS$^{DMR}$ Capacity

| Number of PHYSICAL MODULAR CHASSIS | | Capacity Per MC | |
|---|---|---|---|
| | | 1 | 2 |
| LT card **Note 1** | No. of ports | 56 | 112 |
| | No. of cards | 7 | 14 |
| AP card | No. of ports | Max. 256 ports per site | |
| | No. of cards | 7 | 14 |
| IP-PAD | No. of channel | 32 | 64 |
| Analog Single Line Telephone (Lines) | 8LC **Note 2** | 48 | 104 |
| Dterm (Lines) | Standard | 48 | 104 |
| DtermIP/ INASET (Peer to Peer Connection) **Note 3** | | 128 | |
| Central Office Trunk (Lines) | Loop Start | 48 | 104 |
| | DID w/4DIT | 28 | 56 |
| | 2W/4W E&M | 12 | 24 |
| DTI | 1.5M | 6 | 8 |
| ISDN | 1.5M(PRT) | 6 | |
| | 4BRT (card) | 5 | 10 |
| PFT Connections | | 4 | 8 |

**Note 1:** *Each Modular Chassis has 24 Virtual LT Ports that can only be used to expand the PAD channels from 8 to 32 using the 8IPLA w/24IPLA.*

**Note 2:** *Message Waiting Indication available via Stutter Dial Tone.*

**Note 3:** *Remote PIMs Support up to 2 Virtual PIMs for assignment of DtermIP/ INASET only.*

## IPS PIMMJ (As Remote PIM) Capacity

| Number of PHYSICAL PIMS | | Capacity Per PIM | |
|---|---|---|---|
| | | 1 | 2 |
| LT card | No. of ports | 64 | 128 |
| | No. of cards | 8 | 16 |
| AP card | No. of ports | Max. 256 ports per site | |
| | No. of cards | 12 | 24 |
| IP-PAD | No. of channel | 32 | 64 |
| Analog Single Line Telephone (Lines) | 8LC | 64 | 128 |
| Dterm (Lines) | Standard | 64 | 128 |
| | Long Line | 24 | 48 |
| DtermIP/ INASET (Peer to Peer Connection) **Note 1** | | 128 | |
| Central Office Trunk (Lines) | Loop Start | 64 | 128 |
| | DID w/4DIT | 48 | 96 |
| | 2W/4W E&M | 24 | 48 |
| DTI | 1.5M | 10 | |
| ISDN | 1.5M(PRT) | 6 | |
| | 4BRT (card) | 6 | 12 |
| PFT Connections | | 8 | 16 |

***Note 1:*** *Remote PIMs Support up to 2 Virtual PIMs for assignment of DtermIP/ INASET only.*

## System Capacity (Cont'd)

- **CCIS (p-p/p-mp) and Peer-to-Peer**

| Payload Size | G.729a | G.711 | G.723.1 |
|---|---|---|---|
| 20 ms | 8 Channel | 8 Channel | ------ |
| 30 ms | 16 Channel | 16 Channel | 16 Channel |
| 40 ms | 16 Channel | 16 Channel | ------ |

- **VoIP (H.323)**

| Payload Size | G.729a | G.711 | G.723.1 |
|---|---|---|---|
| 20 ms | 6 Channel | 5 Channel | --- |
| 30 ms | 8 Channel | 7 Channel | 8 Channel |
| 40 ms | 12 Channel | 10 Channel | --- |

- **Payload size for Virtual IPT**

| Payload Size | Max. Voice Channels Per IPT | | |
|---|---|---|---|
| | G.729a | G.711 | G.723.1 |
| 10 ms. | 4 | 4 | – |
| 20 ms. | 8 | 8 | – |
| 30 ms. | 16 | 16 | 16 |
| 40 ms. | 16 | 16 | – |

# IP Specifications

<table>
<tr><th colspan="3">Item</th><th>Specifications</th><th>Remarks</th></tr>
<tr><td colspan="3">Voice Encoding</td><td>G.729a<br>G.723.1 (5.3 k/6.3 k)<br>G.711</td><td>8 kbps CS-ACELP<br>MP-MLQ/ACELP<br>64 kbps PCM</td></tr>
<tr><td colspan="3">IP-PAD</td><td>32 channels per card<br>Automatically seized per call</td><td></td></tr>
<tr><td colspan="3">FAX Communication Feature</td><td>FAX Relay Method (T.30)  IP</td><td>PAD card is required.<br>G3 FAX (up to 14.4 kbps)<br>Super G3 Reciprocal: Not allowed</td></tr>
<tr><td colspan="3">DTMF Signal</td><td>H.245</td><td>H.323 IPT/IP-PAD/DtermIP</td></tr>
<tr><td colspan="3" rowspan="4">Inter-office/Intra-office Signaling</td><td>H.245</td><td>DtermIP to DtermIP connection<br>DtermIP to IP-PAD connection</td></tr>
<tr><td>PROTIMS over IP</td><td>DtermIP to UNIVERGE NEAX 2000 IPS connection</td></tr>
<tr><td>CCIS over IP</td><td>Point to Multipoint connection</td></tr>
<tr><td>H.323</td><td>H.323 IPT card and IP-PAD card are required</td></tr>
<tr><td colspan="3">Jitter Control</td><td>Dynamic Jitter Buffer</td><td></td></tr>
<tr><td colspan="3">QoS<br>(Quality of Service)</td><td>• TOS, IP Precedence<br>• DiffServ</td><td></td></tr>
<tr><td colspan="3">LAN Interface</td><td>10BASE-T/100BASE-TX</td><td>Auto Negotiation is available.<br>100BASE-TX is recommended.</td></tr>
<tr><td colspan="3">Echo Canceller<br>(IP-PAD)</td><td>G.168</td><td></td></tr>
<tr><td rowspan="2">Payload Size</td><td colspan="2">DtermIP/CCIS Virtual IPT</td><td>10 ms-40 ms<br>(G.723.1: 30 ms unit)</td><td>Max. voice channels per card<br>10 ms: 12 ch<br>20 ms: 20 ch<br>30 ms: 30 ch<br>40 ms: 32 ch</td></tr>
<tr><td colspan="2">H.323 IPT</td><td>20 ms.-40 ms. (10 ms. increments)<br>(G.723.1: 30 ms. fixed)</td><td>Maximum voice channels per card<br>G.729a G.711 G.723.1<br>20 ms.: 6ch 5ch -<br>30 ms.: 8ch 7ch 8ch<br>40 ms.: 12ch 10ch -</td></tr>
<tr><td colspan="2" rowspan="2">PAD Control</td><td>0 dB to +16 dB (+2 dB unit)<br>0 dB to –16 dB (–2 dB unit)</td><td>Setting is available per Location No.</td></tr>
<tr><td>0 dB to –16 dB</td><td>For connection via the IPT card</td></tr>
</table>

# *UNIVERGE* NEAX IPS$^{DML}$/IPS$^{DMR}$ System Specifications

| Item | Specifications |
|---|---|
| DML<br>System Capacity | LT ports: Max. 56 ports / MODULAR CHASSIS,<br>- (Max. 64 ports including 8 virtual LT ports/Modular Chassis)<br>- Max. 440 ports / system (IPS$^{DML}$)<br>- AP ports: Max. 256 ports / system<br>- IP ports: Max. 952 ports (IPS$^{DML}$)<br>- Card slots: 8 slots / MODULAR CHASSIS (including 1 slot for MP card) |
| DMR<br>System Capacity | LT ports: Max. 56 ports / MODULAR CHASSIS,<br>- (Max. 64 ports including 8 virtual LT ports/Modular Chassis)<br>- Max. 112 ports / system (IPS$^{DMR}$)<br>- AP ports: Max. 256 ports / system<br>- IP ports: Max. 128 ports (IPS$^{DMR}$)<br>- Card slots: 8 slots / MODULAR CHASSIS (including 1 slot for MP card) |
| Circuit Card Mounted in Modular Chassis | All LT/AP cards of the *UNIVERGE* NEAX 2000 IPS can be used for the IPS$^{DML}$ with the exception of 4LLCB and 4ODTAcards. |
| Power | AC100V – 240V (automatically adjusted) |
| Installation Method | Desk top-setting, 19” rack-mounting |
| Conditions | Temperature: 5°C – 40°C (when the system is operating)<br>Humidity: 20% - 80% (when the system is operating) |
| Cooling | Cooling by FAN |
| Safety Standard | Complied with UL60950, CSA22.2 No. 950, EN60950, AS3260 |
| EMC | Complied with VCCI Class A, FCC Part 15 Class A, EN55022 Class A, AS/NZS 3548 Class A |

# Line Conditions

| Description | | Specifications |
|---|---|---|
| **Loop Resistance (including Telephone Set)** | | |
| Analog Standard Line<br>Analog Long Line | | Max. 600 ohms<br>Max. 2,500 ohms (DP 10pps),<br>Max. 1,700 ohms (DP 20pps)<br>Max. 1,200 ohms (DTMF) |
| **Loop Resistance (including Opposite End Resistance)** | | |
| Central Office Trunk | | Max. 1,700 ohms |
| Tie Line Trunk (Loop Dial) | | Max. 2,500 ohms |
| Tie Line Trunk (E&M) | | Max. 900 ohms (only E-wire condition) |
| **Cable Length** **Note 1** | | |
| SN716 Desk Console | | |
| | 8DLC/4DLC/2DLC Card | Max. 350 meters<br>(Max. 300 meters for 8DLC card) |
| | 4DLC/2DLC Card with AC Adapter | Max. 1,200 meters |
| Dterm Series i/E **Note 2** | | |
| | 8DLC/4DLC Card | Max. 200 meters<br>(Max. 300 meters for Dterm 8 and Dterm 8D) |
| | 2DLC Card | Max. 850 meters |
| | 4DLC/2DLC Card with AC Adapter | Max. 1,200 meters |
| DSS/BLF Console | | |
| | 4DLC/2DLC Card with AC Adapter | Max. 1,200 meters |
| Zone Transceiver | | |
| | ZTII-U (for U-Interface) | Max. 1,210 meters (2-wire), Max. 2,<br>100 meters (4-wire) @-48V<br>Max. 970 meters (2-wire), Max.<br>1,700 meters (4-wire) @-45V |
| | ZTII-U with AC Adapter | Max. 3,950 meters |

**Note 1:** *Cable length is based on cable with 0.5mm diameter and without lightning arresters*
**Note 2:** *DTR-2DT and DTR-4D terminals do not support long line adapters.*

# Zone Transceiver Line Conditions

**Zone Transceiver Line Conditions**

| | U-Interface Line Conditions | |
|---|---|---|
| **PBX Power Supply at –48V (Standard)** | **2 Wire** | **4 Wire** |
| 0.4φ | 762m (2500ft) | 1310m (4300ft) |
| 0.5φ | 1219m (4000ft) | 2103m (6900ft) |
| 0.65φ | 1676m (5500ft) | 2895m (9500ft) |
| 0.9φ | 2438m (8000ft) | 3958m (13000ft) |
| **PBX Power Supply at –45V (Note)** | **2 Wire** | **4 Wire** |
| 0.4φ | 609m (2000ft) | 1066m (3500ft) |
| 0.5φ | 975m (3200ft) | 1706m (5600ft) |
| 0.65φ | 1432m (4700ft) | 2499m (8200ft) |
| 0.9φ | 2131m (7000ft) | 3718m (12200ft) |
| **PBX Power Supply at –43V (Note)** | **2 Wire** | **4 Wire** |
| 0.4φ | 548m (1800ft) | 944m (3100ft) |
| 0.5φ | 883m (2900ft) | 1524m (5000ft) |
| 0.65φ | 1340m (4400ft) | 2316m (7600ft) |
| 0.9φ | 1948m (6400ft) | 3352m (11000ft) |
| **Local Power Supply (with AC Adapter)** | | |
| 0.4φ | 3350m (11000ft) | |
| 0.5φ | 3958m (13000ft) | |
| 0.65φ | 3958m (13000ft) | |
| 0.9φ | 3958m (13000ft) | |

**Note:** *Zone Transceiver is provided –48V power in normal condition. The table at –45V and at –43V show ZT Line Conditions in the event of lower voltage status.*

# Traffic Capacity

Traffic Capacity

| Number of PIMs | 1PIM | 2PIM | 3PIM | 4PIM | 5PIM | 6PIM | 7PIM | 8PIM |
|---|---|---|---|---|---|---|---|---|
| Traffic Capacity | Max. 2500 BHCA | | Max. 5000 BHCA **Note** | | Max. 7500 BHCA **Note** | | Max. 8000 BHCA **Note** | |

**Note:** *The traffic load of each FP shall be 2500 BHCA or less.*

# DRS (Device Registration Server)

DRS (Device Registration Server)-System Based

| Features of Built-in DRS | | Description | Remarks |
|---|---|---|---|
| Max Registration Terminal | | MP | 952 with CP24D/CP27B/CP31E<br>128 with CP31D |
| Log-in | Login without password | Not Available | Use blank as a password |
| | Authentication by DRS-Network Based | Not Available | |
| | Authentication by DRS-System Based | Available | |
| | Authentication by MAC Address | Available | |
| | Confirmation when overriding | Available | |
| Log-out | Dialing Log-out feature access code | Available | |
| | Function key | Available | |
| | Soft key | Available | |
| DHCP | Inter-working with DHCP server | Available | |

# Chapter 8 System Performance

## Transmission Characteristics

| Transmission Characteristics | |
|---|---|
| Cross Talk Attenuation | More than 70 dB at 1000 Hz |
| Idle Circuit Noise | Less than -65 dBm |
| Insertion Loss<br>(relative to 1KHz-10 dBm) | ▪ Station-to-Station - Typical 6 dB<br>▪ Station-to-Trunk - Typical 0.5 dB<br>▪ Trunk-to-Trunk - Typical 0.5 dB at 0 dB PAD control |
| Longitudinal Balance Trunk Side | Better than 58 dB |
| PCM Characteristics | ▪ Line Rate - 1.544 Mbps<br>▪ μ-Law<br>▪ Meets North America TI-04 Standards |
| Return Loss | More than 15 dB (300 ~ 3,400 Hz) |
| Line Impedance | Station: 600 W<br>Trunk: 600 or 900 W |
| Leakage Resistance | More than 20,000 W |

# Rotary Dial Pulse and DTMF Signaling

## (1) Rotary Dial Signal

| Description | Specifications | |
|---|---|---|
| | Receiving | Sending |
| Dial Speed | 9 to 22 pps | 10 pps +/- 0.8pps<br>20 pps +/- 0.8 pps |
| Break Ratio | 55 to 77 % | 67 +/- 3% or 62 +/- 3% |
| Inter-Digit Pause | Min. 256 msec. | 300 to 1,000 msec.(10 pps) |
| Switch-Hook Flash Detection | 384 to 2,300 msec. | Not applicable |

## (2) DTMF Signaling

| Description | Specifications | | | | |
|---|---|---|---|---|---|
| | Receiving | Sending | | | |
| Signal Code | High Frequency Group | | 1,209Hz | 1,336Hz | 1,477Hz |
| | Low Frequency Group | 697Hz | 1 | 2 | 3 |
| | | 770Hz | 4 | 5 | 6 |
| | | 852Hz | 7 | 8 | 9 |
| | | 941Hz | * | 0 | # |
| Frequency Deviation | +/- 1.8 % | +/- 0.8 % | | | |
| Signal Duration | Min. 40 msec. | 64 or 128 milli-sec. | | | |
| Inter-Digit Pause | Min. 40 msec. | 32 to 240 msec. | | | |
| Signal Level | -46 to -5 dBm | -10 dBm (low group)<br>- 8 dBm (high group) | | | |
| Unwanted Frequency Components | Not Applicable | 40 dB below the power of signal frequency | | | |

## Multi-frequency Compelled (MFC) – R2 SIGNAL

### (1) MFC Frequency Value

| Frequencies | Forward Signals(Hz) | Backward Signals(Hz) |
|---|---|---|
| F0 | 1,380 | 1,140 |
| F1 | 1,500 | 1,020 |
| F2 | 1,620 | 900 |
| F3 | 1,740 | 780 |
| F4 | 1,860 | 660 |
| F5 | 1,980 | 540 |

### (2) MFC Combinations

| Combination Number | Frequencies |
|---|---|
| 1 | F0 + F1 |
| 2 | F0 + F2 |
| 3 | F1 + F2 |
| 4 | F0 + F3 |
| 5 | F1 + F3 |
| 6 | F2 + F3 |
| 7 | F0 + F4 |
| 8 | F1 + F4 |
| 9 | F2 + F4 |
| 10 | F3 + F4 |
| 11 | F0 + F5 |
| 12 | F1 + F5 |
| 13 | F2 + F5 |
| 14 | F3 + F5 |
| 15 | F4 + F5 |

### (3) Sender/Receiver Specifications

| Description | Specifications |
|---|---|
| **Sender** | |
| Sender Transmitted Level | -8 dbm to -11.5 dBm |
| Frequency Variation | +/- 2 Hz |
| **Receiver** | |
| Sensitivity Range | -35 dBm to 0 dBm |
| Frequency Variation | +/- 12 dBm |

# Audible Tones and Ringing Signal

### Audible Tones

**Specifications of Audible Tone**

| TONE | FREQUENCY | INTERRUPTION |
|---|---|---|
| Dial Tone (DT) | 350 Hz mixed with 440 Hz | Continuous |
| Special Dial Tone (SDT) | 350 Hz mixed with 440 Hz | 0.125 sec. ON, 0.125 sec. OFF |
| Busy Tone (BT) | 480 Hz mixed with 620 Hz | 0.5 sec. ON, 0.5 sec. OFF |
| Reorder Tone (ROT) | 480 Hz mixed with 620 Hz | 2.5 sec. ON, 0.25 sec. OFF |
| Howler Tone (HWT) | 2,400 Hz interrupted by 16 Hz | Continuous |
| Service Set Tone (SST) | 350 Hz mixed with 440 Hz | Continuous |
| Ring Back Tone (RBT) | 440 Hz mixed with 480 Hz | 1 sec. ON, 3 sec. OFF |
| Hold Tone (HDT) | 480 Hz mixed with 620 Hz | 0.25 sec. ON, 0.25 sec. OFF<br>0.25 sec. ON, 1.25 sec. OFF |
| Second Dial Tone | 440 Hz mixed with 480 Hz | 0.25 sec. ON, 0.25 sec. OFF<br>0.25 sec. ON, 1.25 sec. OFF |
| Call Waiting Ring Back Tone | 440 Hz mixed with 480 Hz | 1 sec. ON, 1 sec. OFF |

### Ringing Signal

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| Frequency | 20 Hz (Nominal) |
| Voltage | 75 Vrms (Nominal) |
| Interruption: | 2 sec. ON, 4 sec. OFF (for external call)<br>1 sec. ON, 2 sec. OFF (for internal call) |

**Note:** *The 2000 IPS has the capability to detect the above type of signal from Central Office and to transmit the above type of signal to PBX stations.*

# Built-In Modem on MP Card

| DESCRIPTION | SPECIFICATIONS |
|---|---|
| MODEM | 33.6 kbps |

# Dimensions and Weight

### *UNIVERGE* NEAX 2000 IPS

| Main Equipment | | |
|---|---|---|
| | **Dimensions (W x D x H: mm)** | **Weight (kg)** |
| PIM<br>(Fully card-mounted) | Approx. 430 x 223 x 353<br>(16.9" x 8.8" x 13.9") | Approx. 11.5<br>(25.35 lbs) |
| BASE | Approx. 430 x 205.2 x 61.6<br>(16.9" x 8.08" x 2.43") | Approx. 3.0<br>(6.61 lbs) |
| BASE TRAY (for UL) | Approx. 435 x 224.6 x 66.2<br>(17.1" x 8.84 x 2.6") | Approx. 1.7<br>(3.75 lbs) |

### *UNIVERGE* NEAX IPS$^{DML}$/IPS$^{DMR}$

| Main Equipment | Dimensions<br>(W x D x H; mm) | Weight (kg) |
|---|---|---|
| *UNIVERGE* NEAX IPS$^{DML}$<br>*UNIVERGE* NEAX IPS$^{DMR}$ | 430(W) x 365(D) x 88(H) mm | Approximately 7Kg /<br>Modular Chassis<br>(when all slots are occupied) |

### Heat Dissipation

| PIM NO. | MAX. AC POWER CONSUMPTION (W/h)* | MAX BTU (BTU/h) |
|---|---|---|
| 1 | 360 | 1226 |
| 2 | 720 | 2452 |
| 3 | 1080 | 3678 |
| 4 | 1440 | 4904 |
| 5 | 1800 | 6130 |
| 6 | 2160 | 7356 |
| 7 | 2520 | 8582 |
| 8 | 2880 | 9805 |