Subject: SOPHO 2000 IPS and IPS DM
Release: up to 12.2
Date: August 2006

The SOPHO 2000 IPS provides the flexibility of the existing TDM-technology, with over 400 features, while providing a bridge to the IP-technology of the future. The SOPHO 2000 IPS was the first IP Telephony system that supports 100% peer-to-peer connectivity and offers all the system reliability and features associated with traditional TDM PBXs.

The capacity of a single 2000 IPS Peripheral Interface Module (PIM) is 64 TDM ports and 448 IP extension ports. With a maximum of 8 modules a complete system can be built of up to 512 TDM ports and a capacity of 1020 virtual (IP + TDM) stations .

A smaller version, the 2000 IPS DM, is positioned more for the IT environment: it is housed in a thin PC like chassis for 19" rack mounting and each chassis only occupies two Rack Units (2RU). It supports up to 40 TDM ports (in a single module) but has capacity to the full 448 IP extension ports. Up to three chassis can be stacked providing maximum capacity of 120 legacy TDM ports while still supporting as many as 320 peer-to-peer IP stations. It uses the same CPU, line/trunk cards, application processor cards and software of the SOPHO 2000 IPS.

The remote PIM concept is combining the benefits of distributed networking with full feature transparency, without giving up reliability. Especially for this concept also the DM-remote (DMR) is introduced. With the Remote PIM the IP station capacity is expanded from 448 -> 956.

Trunk Features: Analogue trunk, ISDN and SIP Trunk Interface for Germany (Toplink)
Tie lines: QSIG and CCIS, a proprietary tie-line protocol
Mobility Access (Basic + Enquiry Calls)
In-skin Voice Mail / Unified Messaging, NAT/NAPT support
SMDR over IP, PMS over IP, Fax over IP (FoIP), Modem over IP (MoIP)
CSTA interface: OAI providing call monitoring and control for external applications

Benefits:

☑ **Advanced Technology (complete IP system on one card)**
☑ **Supports TDM & IP Switching**
☑ **Peer-to-Peer IP Telephony Connectivity**

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The *SOPHO 2000 IPS* General Description Guide has been developed to provide information on the *SOPHO 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:

NEC Philips Unified Solutions

SOPHO 2000 IPS Product Management

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NL1200 JD, Hilversum, The Netherlands

http://www.nec-philips.com

CONTENTS:

Regulatory Information

This preamble provides the necessary compliance and type approval statements.

Approval aspects

In general no specific approval activities are expected for countries within the European Community. Once available the formal Declaration Of Conformity for the new products can be downloaded via the website of the Approvals & Legislation group of Customer Services. For countries outside the European Community requests for type approval support can be made via the Approvals & Legislation group of Customer Services.

FCC

In compliance with the USA's Federal Communications Commission (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.

EU directives ROHS and WEEE

The equipment is in compliance with the requirements of EU directives on:

- the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), 2002/95/EC
- waste electrical and electronic equipment (WEEE), 2002/96/EC

In order to meet ROHS regulations with relase 12.2 several boards will be phased out - see Chapter 4 for an overview.
Another consequence is that the H.323 trunk interface will be abandoned in favor of SIP trunking.

Service Requirements

In the event of equipment malfunction, all repairs will be performed by NEC Philips Unified Solutions or an authorized distributor. It is the responsibility of users requiring service to report the need for service to NEC Philips Unified Solutions or to one of their authorized distributors.
If trouble is experienced with this equipment, please contact NEC Philips Unified Solutions, at +31 35 689 1111 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.
USA only: 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.

1 Introduction

1.1 Release Overview

SOPHO 2000 IPS release 6

This first release provided the flexibility of the existing technology, with over 400 features, while providing a bridge to the technology of the future.

The SOPHO 2000 IPS was the first IP Telephony system that supports 100% peer-to-peer connectivity and offers all the system reliability and features associated with traditional TDM PBXs.

The SOPHO 2000 IPS delivers the same rich feature set as any traditional PBX, but with the added functionality of supporting peer-to-peer IP switching alongside TDM switching. This protects users' legacy investments while allowing them to move to true IP switching at their own pace.

The capacity of a single 2000 IPS module was 64 TDM ports and 448 IP extension ports. With a maximum of 8 modules a complete system could be built of up to 512 TDM ports.

A smaller version, the 2000 IPS DM, iss positioned more for the IT environment: it is housed in a thin PC like chassis for 19" rack mounting and each chassis only occupies two Rack Units (2RU). It supports up to 40 TDM ports (in a single module) but has capacity to the full 448 IP extension ports. Up to three chassis can be stacked providing maximum capacity of 120 legacy TDM ports while still supporting as many as 320 peer-to-peer IP stations. It uses the same CPU, line/trunk cards, application processor cards and software of the SOPHO 2000 IPS.

SOPHO 2000 IPS release 8

This release with the 3300 Series R8 software brought lots of new things (hardware and software) and support for new external applications and devices like "Softphones" and terminals. For information on that, refer to the appropriate pages. With this release a number of new features are added to the system, to name a few:

- The remote PIM concept was introduced. This concept is combining the benefits of distributed networking with full feature transparency. Without giving up reliability. Especially for this concept also the DM-remote (DMR) is introduced.
- A smaller granularity IP-PAD (8 ports), enabled us to be more price competitive for the smaller hybrid or IP configurations. This card has build-in voice compression
- Enlargement of networking capacity
- Enhancements on QSIG

SOPHO 2000 IPS release 9

The SOPHO 2000 IPS expanded its capabilities once again with the release of 3400 Series R9 software. 3400 series R9 software was released with feature enhancements, expanded capacity of terminals and virtual stations. The terminals included in the over all expansion are Analog, Digital and Digital IP. The total Virtual station capacity was expanded from 768 to 1020, resulting in a port expansion of IP stations from 448 to 956.
R9 enhancement provides the analog trunk transmission plan for Europe and South Africa. The 8 Circuit Analogue Trunk Card provides 8 Loop Start trunks with Loop detection using the PN-8COTU card.
Other additional features are SMDR over IP, PMS over IP, Fax over IP (FoIP), Modem over IP (MoIP), LAN interface speed setting, more analog trunk transmission plans, QSIG and ISDN improvements, MATWorX enhancements and OAI enhancments.

SOPHO 2000 IPS release 10
The 3500 series R10 software was released with new features, such as:
- Remote Software Download (MP),
- Short Message Service (SMS) Transparency,
- Analog station CLI-FSK,
- In-skin Hub.

In addition there were enhancements for User Mobility in a Remote PIM Network, ISDN AOC-E, 32-Party Conference, SMDR, and DND.

SOPHO 2000 IPS release 11
The 3600 series R11 software was released with new features, such as:
- SIP Trunk Interface for Germany (Toplink),
- Dterm Preset Dialing (Off-line Number Preparation) and
- Automatic Change of Daylight Saving Time.

New hardware was also released, such as:
- a new 4-circuit digital long line circuit card and
- a new 4-channel MFC receiver/sender card.

In addition there were enhancements for OAI, SMDR for Internal Calls, Multi-language Display per terminal, ISDN – COLP/COLR for Spain, Single Digit Feature Access Code, etc.

SOPHO 2000 IPS release 12.1
With the release of 3700 series R12.1 software is being released with new features such as:
- Mobility Access (Basic + Enquiry Calls),
- NAT support and
- Voicemail Live Recording over CCIS.

In addition, there are enhancements on OAI and Dterm attendant position for centralized operator configuration.

SOPHO 2000 IPS release 12.2
The SOPHO 2000 IPS continues to provide new and enhanced features with the release of 3700 series R12.2 software, such as Remote MP software download/upgrade (for DMR in Remote PIM system), Delayed Hotline and Automatic Login to Home Station Number. In addition, there are enhancements on Mobility Access, OAI (to improve call handling by PC-based attendant), SIP trunks, Dial-by-Name, etc. New boards are also introduced for RoHS compliance.

1.2 Future

Some features are not supported with the first issue of Release 12.2, e.g.
- ISDN-CCBS (Completion of Calls to Busy Subscriber)
- CLI of outside caller to remote MA user
- Tone changes for South Africa need to be tested first
- CLI presentation with Cal Forwarding to Outside
- Hotel/Motel Printer Enhancements
- Pronto visual aid for desk console needs to be tested first
- New version of 4-circuit MFC Receiver/Sender Card for Brazil (SPN-4RSTBA-C(AP))
- IPS DM(E) Basic System Package (IPS-Lite) with PIMMI

This is indicated where this is applicable as supported in Release 12.3. Note however these features will become available on short term. Once available and supported this document will be updated.

1.3 System information Sopho 2000 IPS

The SOPHO 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 system can consist of 1 to 8 Peripheral Interface Modules (PIM) providing a capacity range of 50 to almost 1000 stations.
The SOPHO 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 the same features as for digital (TDM) based Dterm Series telephones. 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).

The SOPHO 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 telephone s/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 with 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.
SOPHO 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 SOPHO 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.

The SOPHO 2000 IPS is the revolutionary companion to the SOPHO iS3000 series. As such, it has all the selling features of the SOPHO iS3000 within an IP transport environment. Since the SOPHO 2000 IPS supports traditional circuit-switched telephony (Time Division Multiplexed) on both the trunk and line sides as well as peer-to-peer IP telephony on both the trunk and line sides, this simultaneous compatibility allows existing users of SOPHO telephony to retain their existing TDM equipment (thus protecting their original investment) as they begin to make the migration to pure IP telephony. All versions of the SOPHO iS3000 can be connected to the SOPHO 2000 IPS easily.

The SOPHO 2000 IPS can function as a standalone device providing telephony connectivity for a mid-sized-to-small office. But it can likewise be networked with other

SOPHO 2000 IPS/ IPS DM nodes giving all users the look and feel of "one network." Up to 255 SOPHO 2000 IPS nodes can be networked together.
Note: DPNSS is not supported

1.3.1 The IP nature of SOPHO 2000 IPS

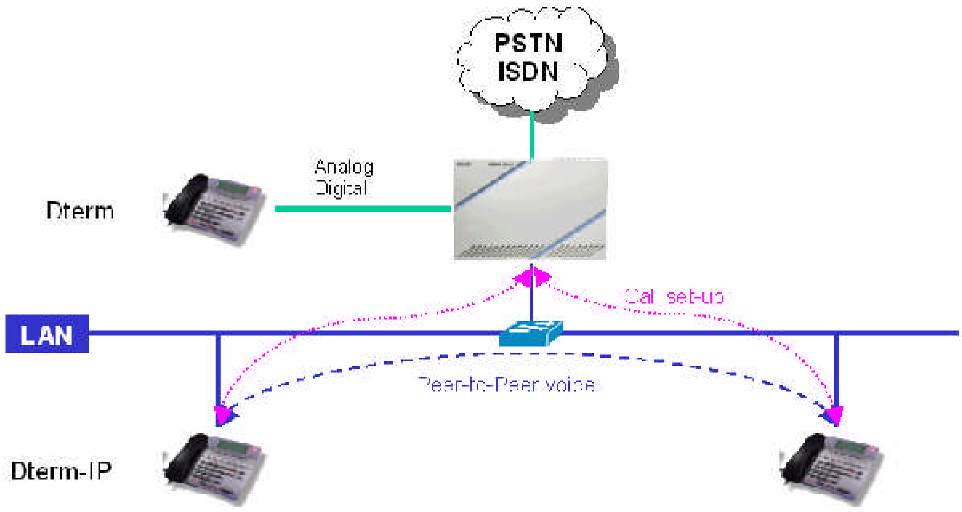
The SOPHO 2000 IPS has more than 400 telephony features to fit most of the customer demands, an overview of these features is given in Appendix D. For a detailed description of each of these features, refer to the "Business, Hotel Features & Specifications" manual.

Because of it's importance and that fact that it is quite new compared to the iS3000, Peer-to-peer IP telephony (and VLAN) is described in the following chapters.

In addition to the anticipated saving from consolidated transport, peer-to-peer IP telephony promises long term benefits resulting from reduced equipment requirements. Traditional TDM trunk cards and line cards will no longer be required in pure peer-to-peer IP telephony systems because a switch as such is no longer required. The network itself becomes the switch. The fact that the SOPHO 2000 IPS can function in and support a "hybrid" network with traditional digital/analogue switching, IP/TDM/IP switching and pure peer-to-peer IP switching means that users can continue to utilize their existing equipment while they begin to phase in Next Generation Telephony which will lay the foundation for the networks of the future.

1.3.1.1 Peer-to-Peer IP Telephony Connectivity

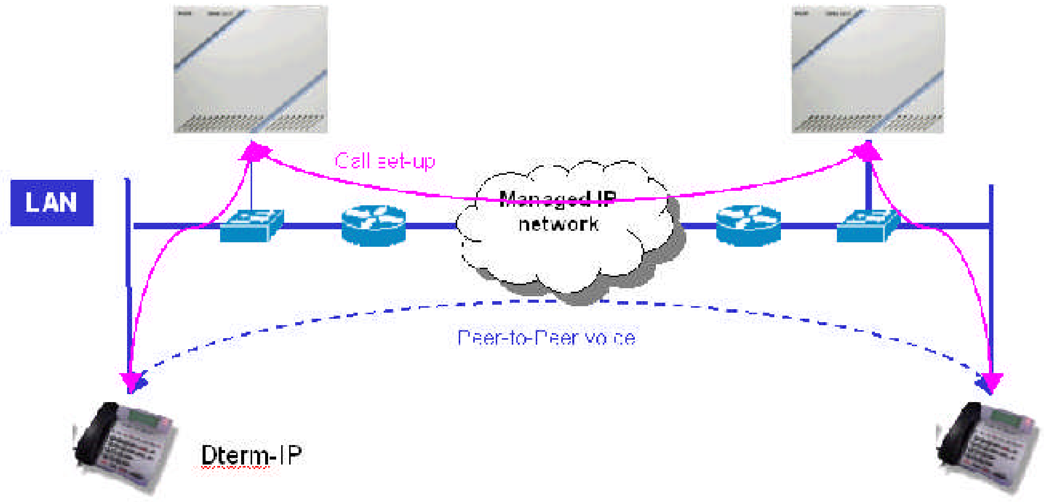
This powerful capability lies at the core of the SOPHO 2000 IPS and its significance should not be underestimated. Most IP-enabled Telephony systems have allowed users to take advantage of the cost savings associated with consolidated transport, but they have done so by adding a card(s) to the switch which acts as a gateway/protocol converter or they provide an adjunct gateway that attaches to the switch and performs the same functions. However, while both of these solutions succeed in providing the conversions required for converged transport, they nonetheless still require that connectivity be established and maintained in the TDM/PBX switching device itself.
The SOPHO 2000 IPS supports a new kind of connectivity (peer-to-peer) which does not require that calls be routed through a traditional Time Division Multiplexed (TDM) switching engine.



The SOPHO 2000 IPS assists in the call setup process and provides the full range of traditional telephony features. Once the call is established, the “voice”-data is “carried” only by the network from station to station. It does not occupy resources in the SOPHO 2000 IPS, which only “re-enters” when requested to provide a function, such as conferencing with a third party.

1.3.1.2 Peer-to-Peer IP Connectivity in the LAN / WAN

The previous drawing above depicts intra-nodal peer-to-peer connectivity where the participating stations are elements on the same LAN. However, the SOPHO 2000 IPS also supports inter-nodal Peer-to-Peer connectivity, enabled by its support of CCIS, Common Channel Inter-office Signaling. Thus, the eventual, long-term equipment reduction and its associated cost savings span the entire network.
See the next drawing for Node-to-Node Peer-to-Peer IP Connectivity.



1.3.2 Voice Mail (MyMail@Net 510i)

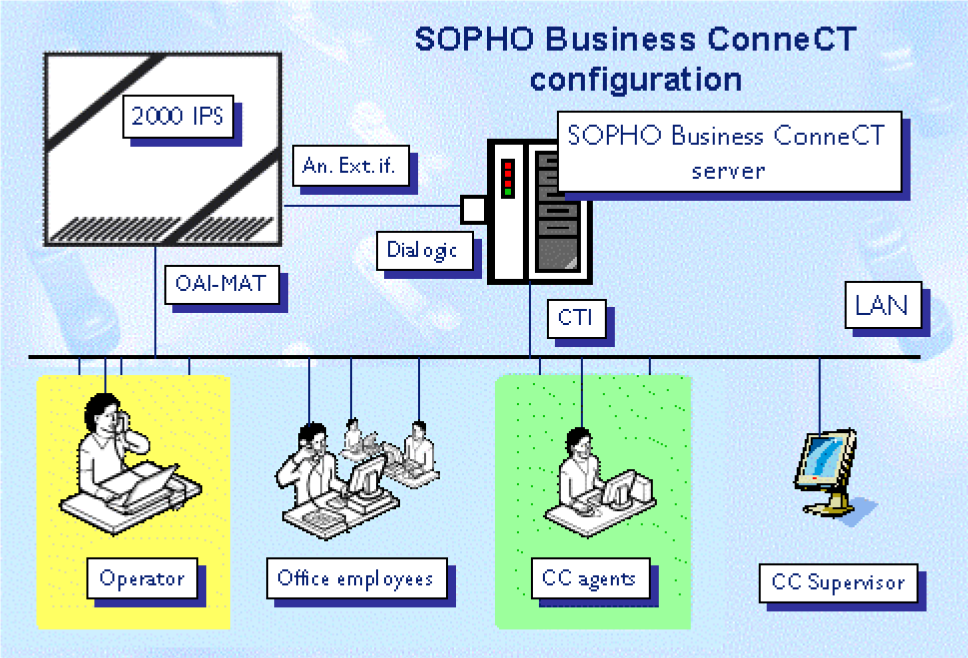
MyMail@Net 510i is an in-switch version of the MyMail@Net 510 specifically designed for the SOPHO 2000IPS IP-PBX. It operates on the Linux operating system and comes with a wide range of optional packages: ViewMail modules, integration with most E-mail systems, in- and outbound fax functionality and Hospitality integration with 16 multi-lingual prompt sets and 17 guest languages.
With the optional ViewMail modules it is possible to conveniently manage voice mail directly from the users desktop PC, to control live telephone traffic and messages, both voice and email.
MyMail@Net 510i is available in 2 to 16 voice port and 1-4 fax port configurations with solid-state storage capacity of 2 Gbyte. Each system also includes five free seats of the ViewMail applications.

1.3.3 Wireless (IP-DECT)

IP DECT offers the mobility to move around the building while enjoying uninterrupted access to telephony, along with other features, such as voicemail and text messages.
IP DECT is the marriage between DECT and VoIP, where the Radios (RFP in DECT terminology and Access Points in VoIP terminology) communicate with the system via the LAN infrastructure.
The Philips SOPHO 2000 IPS provides a unique set of features with DECT terminals. A choice of business phones is available, enabling individual user needs to be met.

1.3.4 SOPHO Business ConneCT

SOPHO Business ConneCT is a server based application that uses the OAI application interface of the 2000 IPS to provide the desktop users several options, grouped as options for the employee, operators and, ACD agents.
Below diagram shows the simple configuration of Business ConneCT in the customer environment:



Interfaces to PBX and users are all IP based except for the Voice Mail and Auto-attendant/IVR module which require a Dialogic board. This keeps the hardware costs as low as possible. Running on a standard server PC with Windows operating system.

The Employee features provide easy and complete directory access, call handling, presence management, group display and call state information, all included in a single window.

The operator features provides directory information, busy lamp field and easy call queuing & transferring, and is customisable to the users preference. Also for employees that temporarily have to fill in the operator position, the intuitive interface provides a very effective solution.

The Agent feautures provide basic call routing for the typical small helpdesk environment to enhanced functionality like skill based routing, email routing, Auto-attendant/IVR, caller identification and complete supervisor functions.

1.4 Hardware Architecture

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

The SOPHO 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 SOPHO 2000 IPS/ SOPHO 2000 IPS DM/ SOPHO 2000 IPS DML/ iS3000. 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.

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

The SOPHO 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), AP01 (OAI) functions, which are provided by an additional card in the previous IVS series. 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.

The SOPHO 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.

1.4.2.1 Firmware Processor (FP)

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.
The first FP to service PIM 0 and 1 is incorporated in the MP.

1.4.3 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:
- Line & Trunk Cards
- Application Processors
- Firmware Processors

Peer to Peer IP Hardware:
- SPN-8IPLA IP PAD
- PZ-M606-A
- PN-24IPLA IP PAD (The PN-24IPLA is a daughter board for the 8IPLA for up to 32 IP PADs)

1.4.4 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.

1.4.5 Unified Circuit Card Size

All circuit cards for the SOPHO 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

1.4.6 High Density Line/Trunk Cards

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

1.4.7 Extended Application Processor (AP) Port Capacity

The SOPHO 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.

1.4.8 Office Data Backup Enhancement

The office data of the SOPHO 2000 IPS is stored in Flash ROM; therefore the backup period is extended compared to systems using RAM with battery.

1.4.9 Various Installation Methods

To meet the specific needs of the customer's environment, the SOPHO 2000 IPS provides the following installation methods:
- Floor Standing Installation
- Wall-mounting Installation
- IEC standard 19 inch Rack-mounting Installation

1.4.10 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 IPPAD 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.

1.4.11 CCIS Connection

DtermIP to DtermIP connection (Peer to Peer connection) via CCIS is available only when the destination office is SOPHO 2000 IPS or SV7000.
The DtermIP connection via CCIS is Peer to Peer to the ISG (IP interface card) in the iS3000.
The system provides only Point to Multipoint connection.

1.4.12 Maintenance

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

1.4.13 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.)

1.4.14 Remote PIM over IP with Survivability

The SOPHO 2000 IPS can have a PIM installed at a remote site through an IP network. At the main site, the SOPHO 2000 IPS/IPS DM is installed and SOPHO 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 MP
- IPS DMR with CP31 MP

1.5 Software Architecture

SOPHO IPS systems use Software Keys (TDM and AP) and Peer-to-Peer IP Seat Licenses to define a modular system. This approach allows customers a greater degree of cost-control for new installations and for upgrades of features, capacities and software series.
Installing the SOPHO IPS systems requires activating Basic System software, and optional features and seat licenses from the software key (on a Floppy Disk). 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 software keys 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)

1.5.1 Generic Program (Basic System Software)

64 Port System Software provides Basic Business/Hotel/Motel Features for:
- 64 LT Ports,
- 5 T1's /E1's,
- 5 ISDN-PRI DCH's,
- 48 ISDN-BRI Trunks.

NEC Customer Software License Agreement Required

1.5.2 Optional Software

The Key Keeper (FD) Floppy Disk holds the selected Key files from the options below:

Description	**Remarks**
LT 64 PORT	64 Port Line/Trunk Key (incremental) Expands LT Ports from 64 to 1020 Ports in increments of 64. Stand alone system maximum 512 LT ports, Remote PIM Network maximum 1020 LT ports.
CCIS Link (1) CCIS Link (4) CCIS Link (8)	Adds support for one CCIS Link Adds support for four CCIS Link Adds support for eight CCIS Link
IPT Card (1) IPT Card (4) IPT Card (8)	Adds support for one IP trunk card Adds support for four IP trunk card Adds support for eight IP trunk card
Event Based CCIS(ECCIS) Key	Adds Event Based CCIS capability
T1/E1 6 to 10	Expands T1/E1 capacity between 144 to 240 channels

ISDN DCH 5 to 8	Expands capacity between 5 DCH Cards and 8 DCH cards
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 SeatLicenses.

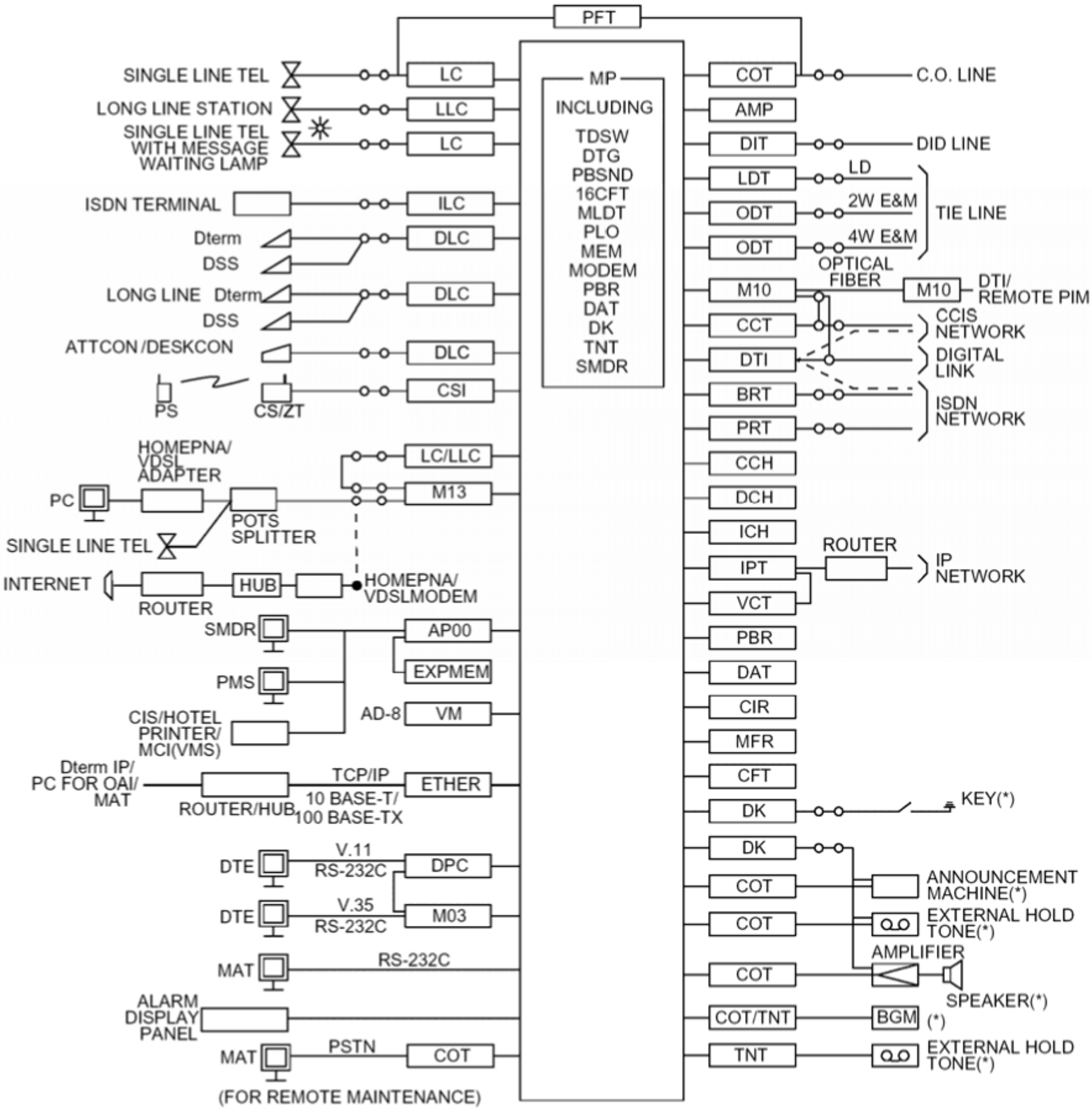
1.6 Technical Terms

SYMBOL	DESCRIPTION	SYMBOL	DESCRIPTION
AP00	SMDR/Hotel Application Card	M10	Optical Interface Card
AUC	Analog Universal Circuit Card (Long Line Circuit, DID Trunk)	MAT	Maintenance Administration Terminal
BGM	External Music Source for Dterm	MDF	Main Distribution Frame
BRT	Basic Rate Interface Trunk	MEM	Main Memory
CCH	Common Channel Handler Card	MFR	MF Receiver/ MFC Receiver/Sender Card
CFT	6/10 Party Conference Trunk	MLDT	Melody Trunk
CIR	CALLER ID Receiver Trunk	MODEM	Modem
COT	C.O. Trunk Card	MP	Main Processor Card
CSI	CS/ZT Interface Card	PFT	Power Failure Transfer
CS/ZT	Zone Transceiver (For North America/ Latin	PMS	Property Management System
DAT	Digital Announcement Trunk	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	PBSND	PB Sender
DLC	Digital Line Circuit Card (for Dterm, ATTCON, 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	16CFT	16 Circuit Four Party Conference Trunk
LC	Line Circuit Card (for Single Line Telephone)	KEY	External Key
AP00	SMDR/Hotel Application Card	M10	Optical Interface Card

1.7 Trunking Diagram

This figure shows a typical trunking diagram of the SOPHO 2000 IPS system (some items are not available in EMEA):

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



1.8 SOPHO IPS DM

The SOPHO Internet Protocol Server Distributed Model (SOPHO IPS DM) is designed as a complement to the SOPHO 2000 IPS, and optimized for 19” rack deployment. The IPS DM is more tailored for the IT environment and has more emphasis on deploying IP telephony than TDM telephony, although the latter is also available.

The SOPHO IPS DM supports peer-to-peer IP telephony connectivity in the LAN and the WAN, while at the same time enabling the enterprise to maximize flexibility in deploying IP. Designed primarily for pure IP networks, the SOPHO IPS DM can also accommodate a mixed (i.e., TDM and IP) converged IP network or standalone solution.

The SOPHO IPS DM supports all the same features and functionality of the SOPHO 2000 IPS, providing voice communication services and features superior to many of today’s IP based telephone systems. TDM switching provides for communication between legacy stations and traditional analog and digital trunks or tie-lines. Furthermore, the SOPHO IPS DM. It is designed primarily for IP networking but also supports traditional analogue trunks and digital trunks for connection to the PSTN. Pure IP switching provides communications between Dterm IP phones and also provides CCIS network connections with up to 255 SOPHO 2000 IPS and other SOPHO IPS DM systems. TDM switching provides for communication between legacy stations and trunks.

The SOPHO IPS DM has two other variants with an adapted processor board (CP31) for smaller capacities, designated as the SOPHO IPS DML (Light) and IPS DMR (Remote).

1.8.1 SOPHO IPS DM and IPS DML (Light)

The SOPHO IPS DM (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 two chassis can be stacked providing maximum capacity of 112 legacy TDM ports while still supporting as many as 888 peer-to-peer IP stations or more depending on the amount of TDM stations used. It uses the same CPU, line/trunk cards, application processor cards and software of the SOPHO 2000 IPS and comes equipped for 19” rack mounting. It offers superior port density; each chassis only occupies two Rack Units (2RU).

The SOPHO IPS DML (Internet Protocol Server Distributed Model Light) is an SOPHO IPS DM that has been optimized for a Small Office Stand Alone IP Solution with from 10 to 80 IP telephones. The SOPHO IPS DML uses the SPN-CP31 as the Main Processor. This system targets users with up to 112 TDM station, 128 IP stations and can be used as an end point in a peer-to-peer CCIS network.
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

1.8.2 Characteristics of the SOPHO IPS DM and IPS DML

Compact and Small Size Modular Chassis

One Modular Chassis provides 7 card slots for 56 LT ports and up to 2 Modular Chassis can be used per system. (8 virtual LT ports are available per Modular Chassis in addition to 56 LT ports.)

2 types of MP (Main Processor)

MP can be selected from the following options by customer requirements.
- PN-CP24 for SOPHO IPS DM , the same MP as the SOPHO 2000 IPS.
- PN-CP31 for SOPHO IPS DML

Power Failure Transfer (PFT)

Power Failure Transfer (PFT) for the IPS DM/IPS DML is provided with PZ-4PFTA card. The PZ-8PFTB for the SOPHO 2000 IPS is not available for the IPS DM/IPS DML.

Installation Methods

Wall Mount Installation is not available. The SOPHO IPS DM /IPS DML can be installed on the desktop or into the 19-inch rack.

1.8.3 SOPHO IPS DMR (R8)

In the feature release R.8 the DM can be turned into an IPS DMR (Distributed Module Remote) by putting a dedicated (simpler) MP in stead of the normal 2000 IPS Main Processor. As such the DMR can act as a remote PIM (over IP) for a 2000 IPS Main site.

The SOPHO IPS DMR (Internet Protocol Server Distributed Model Remote) is a SOPHO IPS DM that has been optimized for Remote PIM over IP applications. The SOPHO IPS DMR uses the SPN-CP31 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 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 SOPHO 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 SOPHO 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.

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.

1.9 List of main benefits

- Capitalize on Converged WAN Infrastructures – By transporting voice signals across the Wide Area Network as IP packets, companies can integrate their voice traffic with their data traffic thereby paying for service and maintenance of one network rather than two.
- Capitalize on Converged LAN Infrastructures – Similarly, the SOPHO 2000 IPS allows users to establish voice calls across the 10/100 Ethernet Local Area Network utilising the existing plant cabling and allowing single cable termination to the desktop. Again, common administration can also produce savings.
- The distributed concept of the 2000IPS provides
  - Survivability with full functionality
  - Up to 15 sites as one single system
- TDM / IP with full functionality
- Excellent HW Reliability
- Integration with iS3000
  - Share same network protocol
  - Share similar applications
- Complete hotel/motel package
  - In-switch hotel functionality
  - Certified with most of PMS vendors
- Affordable and secure wireless IP with IP DECT
- SIP trunk is available
- Integrated Application Suite (BCT) on a single server provides:
  - Operator
  - Employee portal, also allows free seating
  - CC agent
- Proven technology: 100.000 sold worldwide
- In-skin Voice Mail/Unified Messaging option
- Secure O.S. (proprietary) dedicated for call processing
- Demonstrated in one of the first projects with full Microsoft LCS integration

2 System configurations

2.1 Module configurations

The SOPHO 2000 IPS uses a modular Port Interface Module (PIM) design, reduced energy and space requirements, flexible universal port architecture, Flash ROM technology and high reliability.

The SOPHO 2000 IPS consists of a single or multiple Port Interface Modules (PIM), depending on the system configuration; 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. Up to 8 Physical PIMs can be accommodated in a Stand Alone system. Port Interface Modules can accommodate four, eight and 32-port line/trunk (L/T) interface cards and the Application Processor (AP) cards. Because of a highly integrated Central Processor Unit the SOPHO 2000 IPS also eliminates the need for some separate application processor cards by combining their functions onto its CPU.
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.out of 1 to 8 Physical PIM.
The PIM has special slots for the CPU and FP cards and 12 universal slots for Line Terminating circuits (LT) and Application Processor circuits (AP); although the slots can accommodate both LT and AP cards.

The first PIM (identified as PIM0) contains the CPU and a 1024x1024 TDM switching matrix, TDSW, with 1024 x 64kbps channels in and 1024 x 64kbps channels out (= non-blocking).
512 ports of the TDM switch are reserved for

- Line Terminating circuits (LT), for:
  - analogue terminals, with 4 or 8 port analogue station cards
  - proprietary digital terminals (Dterm digital), with 2 or 8 port digital station cards
  - ISDN video and data terminals, 2 circuit ISDN station card; note <u>not</u> suited for S_0 voice terminals
  - analogue trunk lines
- and for IP-PAD Application Processor circuits (AP):
  - to address proprietary IP terminals (Dterm IP), accessed via IP Packet Assembler/Disassembler circuits (IP-PAD) with 8 or 32 channels
  - The IP-PAD can be expanded with voice compression circuits (VCT) for IP voice over WAN – rarely used in a LAN, 16 channels on a card. (for more details on the IP-PAD card, refer to section 2.8)

256 ports of the TDM switch are reserved for:

- Application Processor circuits (AP), for:

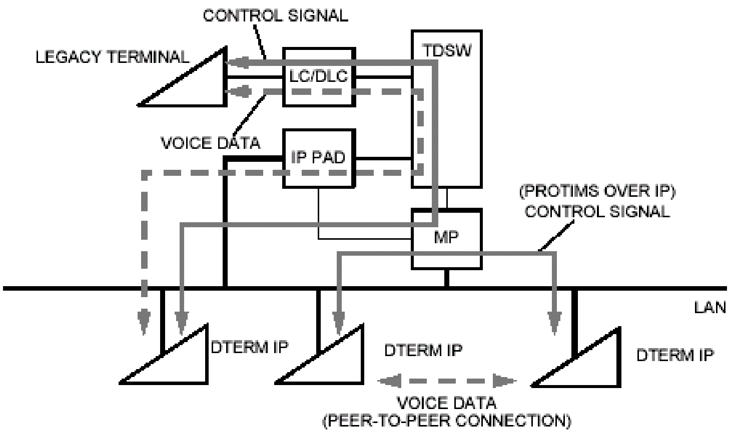
- o ISDN PRI card
- o ISDN BRI card, with 4 circuits
- o multi-unit IP networking, with proprietary CCIS signaling protocol, either via analogue lines, ISDN, E1 or IP (IP-PAD)
- o Q.SIG networking
- o H.323 IP trunking

Distributed Processing is the basis of the architecture, with the following components:

- CPU Main Processor (MP), consisting of Control Cards and System Software, providing Main Processor function, Ethernet interface, guarding the IP Licenses
- Firmware Processor (FP), controlling Line/Trunk Circuit Cards, Analogue Station/Trunk, Digital Station, RST, IP PAD, DAT, VCT, Fiber Interface, External Relays, and station-side ISDN
- AP Application Processors are cards running a protocol or application, e.g. Digital Trunks, Data/Protocol Handler Cards, IPT and 32 Party Conference

As an option, a dual CPU can be installed in a (special) first PIM. The second Main Processor (MP) is in “Cold” standby. In the event of MP failure, the standby MP will “wake up” and take over – the transition time depends on the installed configuration (average time 30~120 seconds).

A separate FP Firmware Processor card is required for every other two PIMs. For the first two PIMs, however, the CPU also contains the Firmware Processor (FP) function for the first two PIMs. The FP card is mounted respectively in PIM 2, PIM 4, and PIM 6. Firmware processors (FP) are responsible for the supervision and control of 128 Line/Trunk Ports (64 ports/physical PIM). 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.

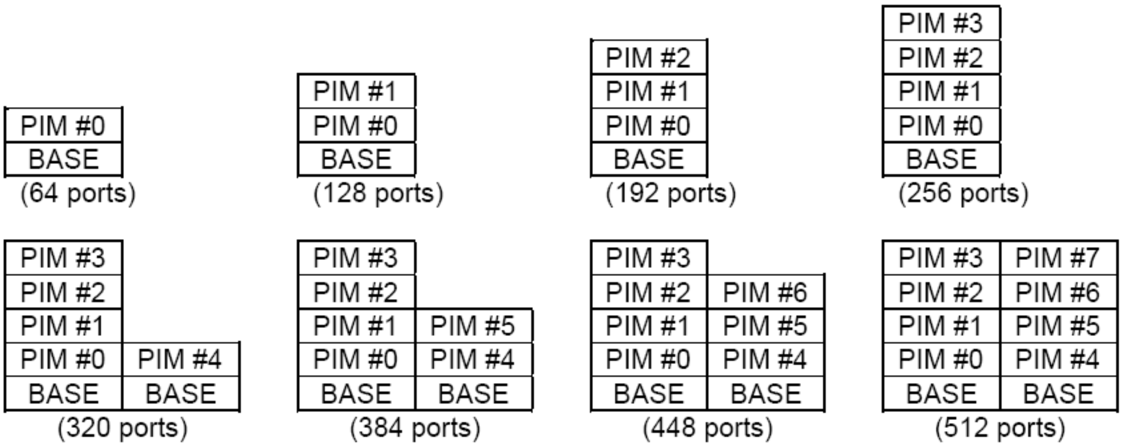


2.2 Installation Methods

The SOPHO 2000 IPS provides three installation methods as follows:

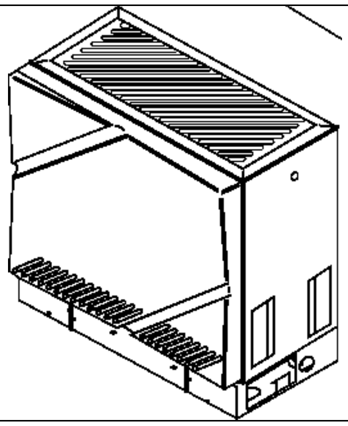
- Floor Standing Installation
- Wall Mounting Installation
- 19-inch Rack Mounting Installation

2.2.1 Floor Standing Installation



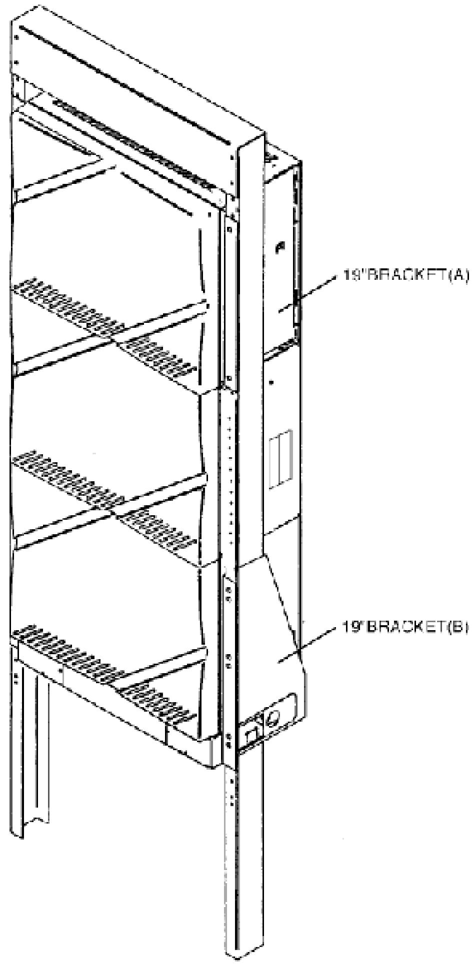
In Floor Standing Installation, the SOPHO 2000 IPS is comprised of up to 8 Port Interface Modules (PIMs).

2.2.2 Wall-mounting Installation



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

2.2.3 19 inch Rack-mounting Installation



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

2.3 Modules and Installation Hardware

2.3.1 Port Interface Module (PIM)

The SOPHO 2000 IPS is comprised of up to 8 Port Interface Modules (PIMs).
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 depending on the system type as follows:

- PIMMJ is normally used in every PIM, but
- PIMMK is only used in PIM0 in a dual system instead of PIMMJ

2.3.2 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 one stack of 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.

2.3.3 Installation Hardware

Base/Top Assembly (1 per stack)	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 (1 per PIM)	The Hanger Assembly is used for Wall-mounting Installation. One set of Hanger Assembly is required for each PIM.
19 inch Bracket (A: 1 per PIM) (B: 1 per stack)	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.
Mounting Bracket (Optional 1 per stack)	This is used for Floor Standing Installation. Without Mounting Bracket, 1.1 G 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 1.1 G, one set of Mounting Bracket is required for each 4 or more module stack and attached to the topmost PIM.
I/F Bracket (Optional 1 per system)	This 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.
Base Tray Assembly (Optional 1 per stack)	This is used for Floor Standing Installation for stationary equipment (UL complied). One set of Base Tray Assembly is required for each stack.

2.3.4 SOPHO 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) 5 0/60 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.
Battery Backup 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.

2.4 Circuit Cards

The circuit cards used for SOPHO 2000 IPS are 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)

2.5 IPS System Conditions

The SOPHO 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).
Overall System Conditions:

- To connect the MP (PN-CP24, PN-CP27, and PN-CP31) card to the LAN, ETHER (c) card is required on the 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.

2.6 IPS Terminal interfaces

2.6.1 ANALOGUE STATION INTERFACE

The SOPHO 2000 IPS supports analogue stations with an 8 port (max. 600 ohms loop resistance) card or a long-line 4 port (max. 2500 ohms loop resistance) card.
Modem communication speed may be limited at around 33.6Kbps -it depends on customer environment.

2.6.2 DIGITAL STATION INTERFACE

The TDM terminals are connected with a 2 or 8 port interface card and both require 1-cable pair. Reference the charts below for cable distances.

Dterm Series Line Card	Cable Distance w/o Power Adapter	Cable Distance w/Power Adapter
PN-8DLC	200 m	300 m
PN-2DLC	850 m	1200 m

The PN-8DLC card is also used to connect the DSS Console and the attendant console DESKCON (SN716).

2.6.2.1 Long line DLC card (4DLC) (R11)

The PN-4DLCT is a new digital long 4-line circuit card for Digital Dterm phones and DSS/BLF Consoles, which allows the PIM card slots to be used more efficiently.
The maximum cable length is 850 meters at 0.5mm-diameter cable. One card provides 4 Dterm/DSS Console interface circuits. PZ-PW122 (-48VDC power card) is required. The PN-4DLCT replaces PN-2DLCN.
<u>Required Hardware:</u>
PN-4DLCT
PZ-PW122
<u>No IPS software requirements</u>

2.6.3 ISDN Station Card

The 2 S_0 Circuit ISDN Station Card provides Basic Rate ISDN support for 4 Video & Data Terminals which do comply to the standard. Note that standard S_0 voice terminals are not supported. This is (currently) also not planned for the roadmap

2.6.4 IP STATION INTERFACE

The interface is achieved with a 32 Channel IP PAD (SPN-32IPLAA IP PAD-C), which provides Packet Assembly/Disassembly to accommodate Legacy Line/Trunk/Feature interface for the IP terminals via the TDM switch. It runs on G.711, no voice compression. A maximum of 8 per system is possible, 2 per FP.
It can be combined with 16 Channel CODEC Cards (SPN-16VCTAA IP PAD-A) to provide voice compression (G.723.1, G.729a) and 14.4.kbps Fax. One or two VCTs can be mounted on each IPLA.

The IP Signaling used by the Dterm uses a proprietary protocol (PROTIMS) and does not support H.323 with Gatekeeper.
Standard H.323 terminals can be connected to the 2000 IPS via an external gatekeeper and utilizing the H.323 IP trunk card (SPN-IPTB-B H323).

QoS settings use TOS, IP Precedence and Diffserv.
Echo Canceller complies with G.168.
VLAN complies with IEEE 802.1Q.

DiffServ (Differentiated Services) type of QoS supports:

- Peer-to-Peer CCIS
- Point-to-Multipoint IP Trunk -CCIS
- H.323 Trunking
- Dterm IP to Dterm IP
- Dterm IP to IPPAD
- Dterm IP/IPPAD to Peer to Peer CCIS

2.6.4.1 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.

2.6.4.2 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 IPPAD 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.
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.

2.6.4.3 DRS (= Device Registration Server)

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

2.7 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, IP-PAD cards and trunk cards are required to transmit and receive the voice data.

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

2.7.1 CCIS Connection

The system can be connected in an IP network by No. 7 Common Channel Inter-office Signaling (CCIS) via the Virtual IPT to other 2000 IPS systems, and to 2400 IPX, iS3000 or SV7000.

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.

- Peer-to-Peer connection between DtermIPs via CCIS is available only when the destination office is 2000 IPS, 2400 IPX or SV7000.
- 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 DM/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- 0.125 seconds OFF	Link Down occurrence	-
Red/Flashing (Slowly)	0.5 seconds ON- 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.

2.7.2 H.323 Connection (obsolete)

The system could be connected to the terminal and network equipment according to ITU-T recommendation H.323 protocol. This has been abandoned in favor of SIP.

2.7.3 Q.SIG (QSIG) Protocol Support to 3rd party PBX's

This feature allows the SOPHO 2000 IPS to provide basic connection service when interfacing to a QSIG network.

A 30 B-channel 2Mps (E1) digital interface and a QSIG D-channel handler is required for each physical interface.
Currently supported functionality is:

- Basic Call (ECMA 143)
- Generic Functional Procedures (ECMA 165)
- Identification Services (ECMA 148)
- Name Identification (ECMA 164)

Between iS3000 and 2000 IPS the QSIG link has been tested and verified. Be aware that between 2000 IPS and 3rd party PBX's, one carefully should check compliance.
Currently a test at the TI test lab is scheduled to certify / approve QSIG.
This is a first step in the integration between iS3000 and 2000 IPS. With the development of CCIS on the iS3000 the integration will go much further.

2.7.4 SIP Trunk with Toplink, Germany, Basic calls (R11)

In R11 software, 2000 IPS supports SIP-based connections with a German Provider, Toplink, for cost saving of customer's telephone expenses.

The IPS supports:
- DDO calls to toplink network
- DDI calls from toplink network
- Calling line identifications – Presentations (CLIP)
- Calling party number display on Dterm
- Session Timer - Send/receive keep alive signals
- Receive fragment packets (max. 3000bytes)
- Fault message registration
    - IPS reboot (SIP trunk, MP)
    - Link disconnect
    - Session timer - timeout
- Alternating routing to PSTN
    - When SIP trunk or SIP network is failed, 2000 IPS can reroute the outgoing call to PSTN.
- Add/delete E.164 “+” sign
    - Add “+” sign in front of telephone number (outgoing call)
    - Delete “+” sign in front of telephone number (incoming call)

2.7.4.1 SIP Trunk Enhancements (R12.2)

NAT Support

The 2000 IPS SIP trunk supports network configuration via a router with NAT/NAPT function (NAT: Number Address Translator, NAPT: Number Address Port Translator). NAT support provides efficient use of global IP addresses and provides better security that shields internal addresses from the public Internet.

Typically, the 2000 IPS with SIP trunk is on one side of a router with NAT/firewall function and the SIP server is on the other side. The SIP-trunk has a private IP-address while the SIP server and the outside of the router have a global IP-address. In the SIP messages sent and received by 2000 IPS, inside the payload of IP-packets, the source IP address of the 2000 IPS SIP trunk is indicated. This source address in the SIP message is processed by the SIP server, therefore it should be a global address that refers to the SIP trunk. If the source IP address in the SIP message is the private address, correct SIP communication cannot be established.

The NAT function of the router converts a source IP address in the IP header from private address to global address, and vice versa, but cannot convert the source IP address in the message body, so it can not change the SIP messages. In R12.2, the IPS can change the source IP address in the SIP message from the private IP-address to the global IP-address.

Note: When the NAT function is used, it is necessary to do an inter-working test with providers and routers beforehand.

RTP Monitoring/Statistics (this is a maintenance function)

The SIP trunk can monitor if RTP packets are correctly received when the SIP trunk call is being established. If no RTP packets are received for 10 seconds, the SIP trunk connection can be released and the fault message is stored in the system.
This feature prevents unnecessary channel seizure of the SIP trunk.

The feature also provides functions for analyzing problems regarding SIP trunk calls. It can collect call logs on SIP trunk calls. These statistics can be retrieved by MATWorX.

Outband DTMF

The 2000 IPS SIP trunk supports out-of-band DTMF based on RFC2833, which provides more reliable DTMF digits transmission over a SIP network.
This feature provides DTMF digits transmission between the IPS SIP Trunk and an external SIP terminal using RTP packets (out-of-band signals), instead of using regular audio packets.

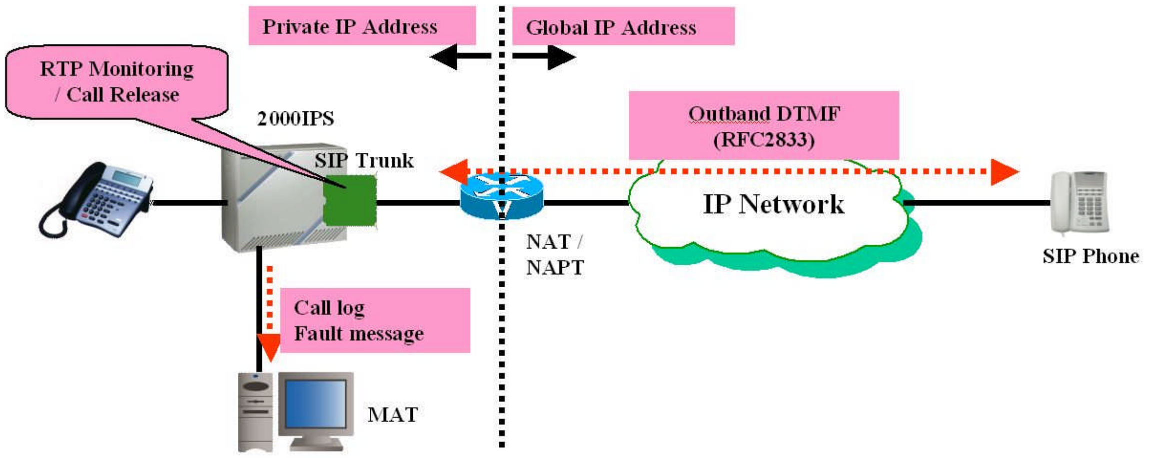
- When the outband DTMF packets are received from the external SIP terminal, the IPS detects the digits information from the received RTP packets and send the DTMF tones to the IPS station.
- When sending the outband DTMF packets to the external SIP terminal, the IPS converts the DTMF tones received from the station to the RTP packets based on RFC2833, instead of regular audio packets.

Tone Disabler

This feature is designed for improved FAX communications over SIP network.
By detecting phase inversion of FAX signal tone (V.25 2100Hz tone), the system changes a setting of an echo canceller and NLP (Non Linear Processor) for the SIP trunk connection.

- When the phase inversion is detected, the echo canceller and NLP set OFF for the FAX communications.
- When the phase inversion is not detected, the echo canceller and NLP remains the same as voice communications.

This feature can be effective on a SIP trunk card basis by system data programming.



2.8 The IP-PAD and VCT

To support the IP telephony capabilities a Packet Assemble/Disassemble (PAD) gateway card is available. This card is used to convert the speech path of the IP network to the legacy interfaces in the system (TDM based interfaces), or visa versa. Besides the conversion function on 32IPLAA IP-PAD C card the system can provide compression protocols via the adjacent 16VCTAA IP-PAD A card, a maximum of 2 VCT cards can be applied per IP-PAD**.**

So, the IP-PAD is required for the following connections/statuses:
- Connections between IP Enabled Dterm telephones and legacy stations/trunks
- Connections between IP Enabled Dterm telephones and IP trunks (H.323)
- Connections between legacy stations/trunks and IP trunks (H.323)
- Connections for CCIS networking via IP from/to legacy stations/trunks
- While IP Enabled Dterm telephones are on hold (Consultation Hold, Call
- Transfer, Music-on-Hold)
- When any override service is activated (Executive Override)
- 3-/4-Party conference including IP Enabled Dterm telephones

The following table provides an overview of the bandwidth occupation in relation with the different compression rates.

PAYLOAD	BANDWITH
G.711 /40ms (64k)	72 Kbps
G.711/10ms (64k)	96 Kbps
G.723.1/30ms (5.3K)	16 Kbps
G.723.1/30ms (6.3k)	17 Kbps
G.729a/10ms (8k)	40 Kbps
G.729a/40ms (8k)	16 Kbps

2.8.1 8 ports IP-PAD (including compression)

With release 8 a smaller granularity IP-PAD is introduced. This card has 8-channels with on-board compression. Apart from that, it can be expanded by an optional 24-channel daughter board.
This is done for two reasons:
- Better fit for smaller configs/small amount of IP phones. So also more choice of boards
- Possibility to easily upgrade later

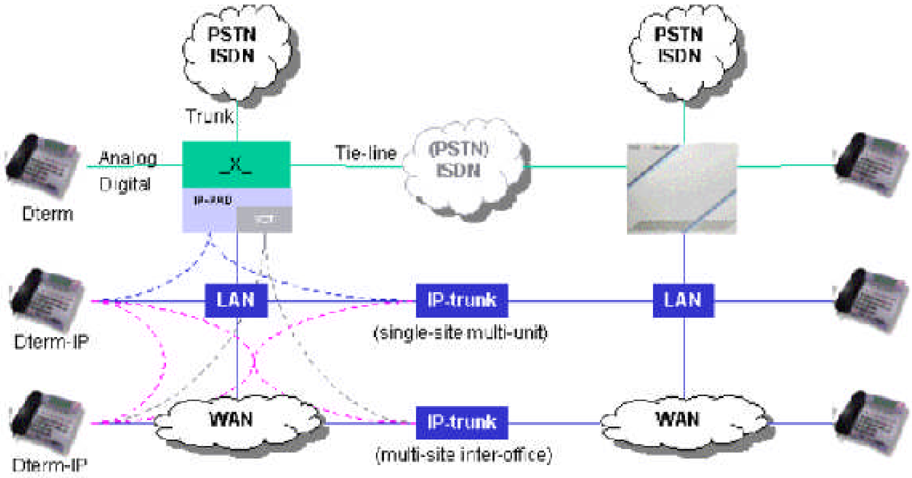
The following table shows the specifications of the 8 channel board (SPN-8IPLA) and the 24 channel daughter board (SPZ-24IPLA):

Function	Description	Remarks
Network interface	Ethernet- IEEE 802.3 (10BASE-T)- IEEE 802.3u (100BASE-TX)- Auto Negotiation	100Mbps recommended
Number of channels	8	
Number of	Can be expanded by adding a Sub card on the Main card.	

expanded channels	+24ch (+16ch when using G723.1)	
Voice Codec	G711 (64Kbps), G729a (8Kbps), G723.1 (5.3Kbps)	G723.1 (5.3Kbps only)
Voice Payload	10-40ms (10ms increment), 30ms fixed with G723.1	
Jitter Buffer	10-300ms	
DTMF Relay	Available	
FAX Relay	Available with (G.711), regarded as voice call	Pass-Through & G.711
Modem Relay	Not supported	Under Study
VLAN	Tag VLAN (IEEE802.1p)	
QoS	IP Precedence, Diffserv	
PAD	-14dB --- +14dB	
EC	G.168 (Max.64ms), with NLP (Non Linear Processor) function	

Further information about configuring, compatibility and compression can be found in Appendix C.

2.8.2 When are IP-PAD and VCT required



Notes: *Conferences, including 3 and 4 party also require the IP-PAD, even when MoH is provided internally by the DtermIP*

In table form:

To / From	Dterm Ana/Digi	TDM trunk/tie	Dterm-IP LAN	IP trunk LAN	Dterm-IP WAN	IP trunk WAN
Dterm Ana/Digi	-	-	IP-PAD	IP-PAD	IP-PAD VCT	IP-PAD VCT
TDM trunk/tie	-	-	IP-PAD	IP-PAD	IP-PAD VCT	IP-PAD VCT
Dterm-IP LAN	IP-PAD	IP-PAD	-	-	-	-
IP trunk LAN	IP-PAD	IP-PAD	-	-	-	-
Dterm-IP WAN	IP-PAD VCT	IP-PAD VCT				
IP trunk WAN	IP-PAD VCT	IP-PAD VCT				

2.9 Fax and modem traffic

2.9.1 T.30 Fax over IP (obsolete)

T.30 ITU-T standard FAX handshake protocol is supported to help facilitate FAX over a much broader range of customer premise equipment. The facsimile transmission procedure (T.30) is supported via T.30 FAX relay on the VCT package; PN-32IPLA-C card and PN-16VCTA-A card must be used; Note since these cards are not RoHS compliant and will be phased out, this feature will no longer be supported!

Facsimiles can be sent between SOPHO 2000 IPS and SOPHO IPS DM systems.
If a Super G3 facsimile is used, the transmission speed will be equivalent to G3.

Minimum network requirements such as available bandwidth, QoS, jitter buffer value larger than network assumed value, and clock synchronization play an important factor for successful G.711 FAX transmissions.
In some applications bandwidth is extremely limited making network problems virtually impossible to avoid. For these applications T.30 FAX is recommended.

Required bandwidth for FAX Connection:

Connection Conditions	Required Bandwidth (One-way)
G711, Payload=40ms	150 Kbps (estimated)
T.30, G729a, Payload=40ms, Communication speed=14.4Kbps (No IP Header compression in Router)	23.6 Kbps (FAX Payload=78byte)
T.30, G729a, Payload=40ms, Communication speed=14.4Kbps (with IP Header compression in Router)	16.6 Kbps (FAX Payload=78byte)

2.9.2 FAX Communication with IP PAD cards (Release 9):

Fax over IP (FoIP) is now available in G.711 non-compressed or G.726 compression mode.

- 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.

2.9.3 Modem over IP, Pass-through mode between 8IPLA IP-PADs (R9)

With R9 software and SPN-8IPLA PAD-B the system detects the modem connection and uses pass-through mode for better through put and success rate. Connection speeds range from 9.6 kbps to 24 kbps; those speeds vary depending on the CODEC used (G.711 or G.726) and network configuration.
Required Hardware: SPN-8IPLA IP PAD-B

2.10 IP networking

2.10.1 MP, IP-PAD LAN interface speed setting enhancement to 100Mbps/full duplex mode(fixed) (R9)

R9 software adds fixed 100 Mbps Full Duplex to the CPU/M606, which is adjusted via office data programming.
Control for 100 Mbps Full Duplex or auto-negotiate 10/100 Half Duplex on the 32 IPLA and 8 IPLA IP PAD is done via switch settings on each card.
Required Hardware:
- SPN-8IPLA IP PAD-A/B with SC-3330 Firmware
- SPN-32IPLAA IP PAD-C/D/E with SC-3353 Firmware

2.10.2 VLAN Tagging

The SOPHO 2000 IPS supports VLAN-based on IEEE 802.1Q (Tag VLAN) to logically separate “voice”-data from the other traffic on the LAN/WAN. VLAN lessens the possibility of packet collision and prevents voice quality degradation from lack of available bandwidth.

2.10.3 NAT support (R12.1)

This feature provides efficient use of global IP addresses and better security that shields internal addresses from the public Internet).
NAT (Network Address Translation) is a technology that translates the internal local IP addresses to globally unique IP addresses 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). In R12.1 software, the 2000 IPS support network configuration via a router with NAT.

Remote IP Dterm connection via NAT
- Provide communications between Dterm IP located under the same NAT.
- Provide communications between Dterm IP located under different NAT.

Remote PIM over IP via NAT

- Provides communications between Dterm IP within the same remote site (Not provide communications between Dterm IP via NAT).

Required Software and Hardware

- SPN-8IPLA IP PAD-C
- Dterm IP (ITR-xx-3P TEL) with firmware rev.12.82 or later
- Dterm SP30 Ver.7 or later

2.10.4 In-Skin Hub (R10, obsolete)

Note: this card is not RoHS compliant and will be phased out.
With the In-kin Hub no external hub is required to accommodate LAN cables from the IPS and IP-related devices. Also, since the In-skin Hub is powered by the PBX, the operating power for the Dterm IP phones connected to the hub can be backed up even if commercial power failure occurs.
The 8-port In-skin Switching Hub can be used as a LAN cable terminator to accommodate LAN cables from the system (MP/M606-A, IP-PAD) and IP-base peripheral devices (MAT, SMDR, OAI application processor, etc.). The In-skin Hub also provides a Power over Ethernet (PoE) capability based on 802.3af or NEC proprietary protocol. This PoE function can be used for a backup power feeding to Dterm IP phones connected to the In-skin Hub.
Required Software and Hardware:

- SPN-8ETIA ETHER SW
- PZ-PW122 (IPS) or PN-PW03 (DM/DMR) for PoE

2.11 Miscellaneous Application Processor cards

2.11.1 RS232C ports

The 4 Circuit Multifunction Card (PN-AP00B MRC-C) provides 4 RS232C ports for SMDR, PMS, MCI and Hotel Printer. It is required in the Main System for Centralized Billing and Voice Mail.
With the Memory Expansion Card for AP00B (PZ-M537), the call record storage for SMDR can be expanded from 1,600 to 27,000 records.

2.11.2 OAI with FLF

The SPN-AP00B DBM-C(AP) for PBC, provides features for OAI applications using FLF (?); requires release 9.

2.12 In-Skin Voice Mail / Unified Messaging: MyMail@Net 510i

MyMail@Net 510i combines voice mail, automated attendant and audiotext into a completely integrated business solution that will help you communicate more effectively with the people who matter most to your business-your customers and co-workers.

MyMail@Net 510i is an in-switch version of the MyMail@Net 510 specifically designed for the SOPHO 2000IPS IP-PBX.
Feature packages that extend the functional capabilities of MyMail@Net 510i include ViewMail modules, integration with most E-mail systems, in- and outbound fax functionality and Hospitality integration with 16 multi-lingual prompt sets and 17 guest languages.
ViewMail gives the user control over his live telephone traffic and messages, both voice, and email, on his desktop PC.

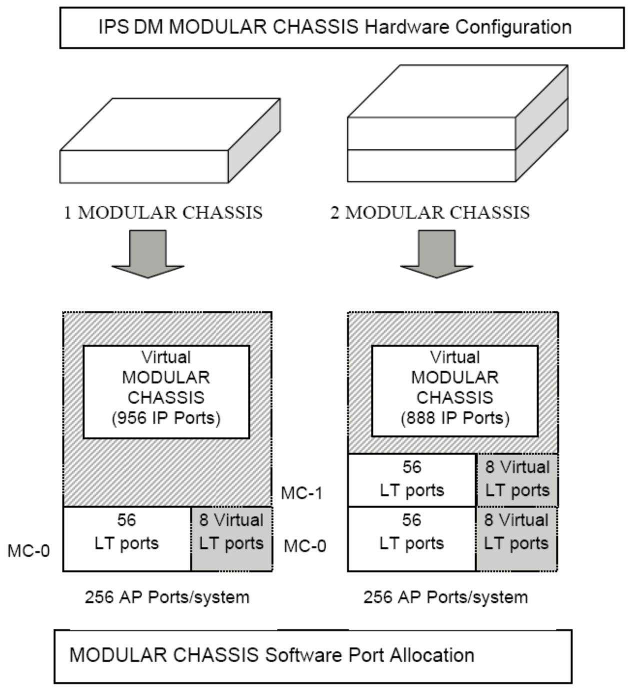
2.13 Patch Panel (24-Port)

The 24-Port Patch Panel is specifically designed for use on the SOPHO 2000 IPS and IPS DM. One Patch Panel provides connectivity for 24 legacy stations or trunks (three 8 port card slots). 1 RU rack mount provides a clean and convenient method for connecting up to 24 RJ-11 or RJ-45 connectors. The panel provides one 50-pin male champ connector and convenient breakout of the alarm signal leads pin 25 and 50 to the PBX.

2.14 SOPHO IPS DM/IPS DML/IPS DMR System Configuration

2.14.1 SOPHO IPS DM Modular Chassis (MC)

The SOPHO IPS DM consists of from one to two Modular Chassis depending on the system configuration. The Modular Chassis provides 56 LT ports in hardware slots and provides 64 ports in software port allocation (56 LT ports and 8 virtual ports). There are 2 types of Modular Chassis; "Physical Modular Chassis" and "Virtual Modular Chassis". The Physical Modular Chassis is a “hardware Modular Chassis” and is used to accommodate an MP, IP PADs, legacy LT/AP cards, and power supply units. The Virtual Modular Chassis is “software Modular Chassis” and is used to accommodate IP stations by system data programming. The port capacity of the Virtual Modular Chassis is varied depending on the number of Physical Modular Chassis. The Modular Chassis can be installed on the desktop or into the 19-inch rack only.
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 following illustration shows Modular Chassis hardware configurations, software port allocation, face layout and rear view of Modular Chassis for IPS DM.



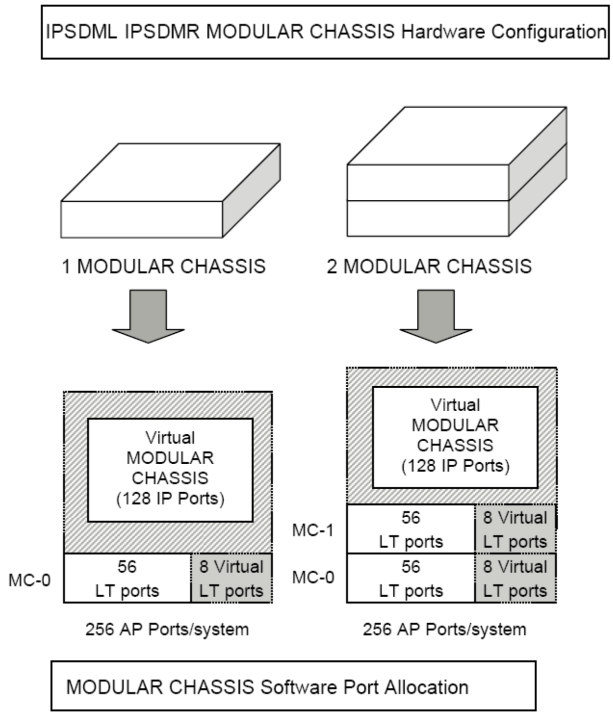
2.14.2 SOPHO IPS DML/ IPS DMR Modular Chassis (MC)

The SOPHO 2000 IPS DML/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, IP PADs, legacy LT/AP cards, and power supply units. The Physical Modular Chassis provides 56 LT ports in hardware slots and provides 64 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 IPSD ML/IPS DMR for a total of 128 ports used to accommodate IP stations by system data programming.

The SOPHO IPS DML is an SOPHO 2000 IPS DM that has been optimized for Small Office Stand Alone IP Solution with from 10 to 80 IP telephones. The SOPHO IPS DML uses the SPN-CP31 as the Main Processor. This system targets users with up to 112 TDM station, 128 IP stations and can be used as an end point in a peer-to-peer CCIS network. 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 SOPHO 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 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 SOPHO IPS DML/IPS DMR.



2.15 Remote PIM over IP (Release 8)

With the remote PIM concept, a normal 2000 IPS PIM, an IPS DM or a IPS DMR can be put at a remote site (connected via IP to a main system) an be a fully integral part of that main system. The Main Site system controls and maintains the remote DM(R) 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. In this way a really distributed survivable remote system can be build. When using a 2000 IPS and an IPS DM as the remote PIM one can also look at it as a true networked multi node configuration.

- IPS DM: IPS Distributed Model (with CP24). Comparable with normal 2000 IPS with a more limited TDM capacity and a smaller housing (2 rack units)
- IPS DMR: IPS Distributed Model Remote (with CP31). This is an IPS DM with a special scaled down main processor, e.g. no on-board modem.

See next table Remote PIM over IP CPU Application, to find out which CPU can be used in which kind of situations.

		Remote Site				
	CPU type		CP24		CP27 dual CPU	CP31
Main site		PIM type	IPS PIMMG	DM PIMMD	IPS PIMMH	DMR PIMD
	CP24	IPS PIMMG	yes	yes	note 1	yes
		DM PIMMD	yes	yes	note 1	yes

	CP27 dual CPU	IPS PIMMH	yes	yes	note 1	yes

Note 1: Can work only in single CPU mode (to be confirmed by NEC)

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.

2.15.1 Outline of Operation

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.

On-line Moves, Adds and Changes

When LC/DLC/COT cards are added or deleted (Moves Adds & Changes- MAC) in Remote Site through on-line programming these cards are in real time being assigned to the system, even if the system data copy is not activated to the Remote site.

2.15.2 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 FD are following.
  - The same number of Key FD for remote site license as the number of remote sites is required for Remote PIM over IP system. The required Key FD 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 FD data of Main Site. The total Key FD data is loaded to Main Site and divided between Main Site and Remote Sites as following examples.

    Example 1: When 256-port Key FD 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 FD 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 FD 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 FD 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, PN-CP27, PN-CP26, PN-CP28
  Remote Site: PN-CP31, PN-CP24, PN-CP26
  NOTE: PN-CP27-A/PN-CP28-A cannot be used as Backup CPU system in Remote Site.
- This feature is not compatible with Fusion service of SV7000 (and 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 (ISDN/Q-SIG/Q931a), 4RSTB (MFC/T1-ANI/E911), 4RSTC/D (Caller ID trunk), CFTC (32-party conference), IPT
  - LT: 8RSTG (PBR), ILC (ISDN Terminal), 4RSTF (Caller ID station), CFTA/B (6/10-party conference), PN-8LCAD (LLC), PN-4LLCB (LLC), PN-8PFTB (PFT), 4VCT, DK00, DAT

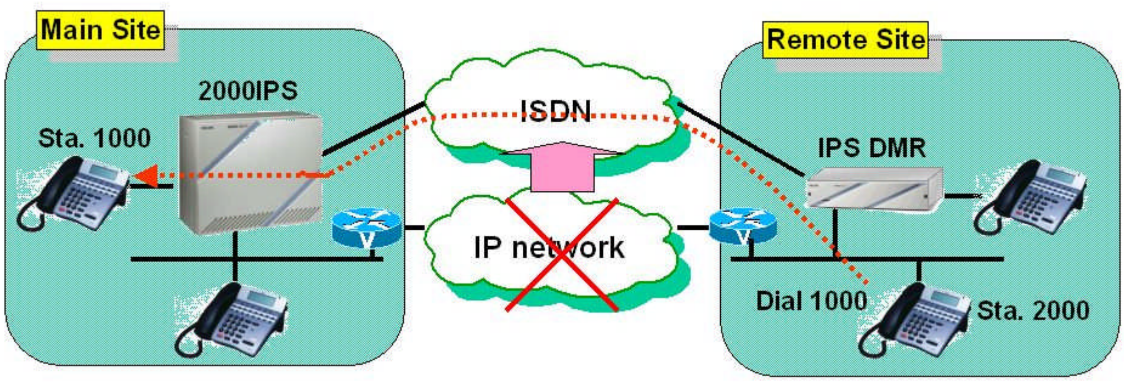
  NOTE: *Four-line built-in PBR on the MP card is available at Remote Site.*
- MP built-in FP must be set to each Remote Site.
- The port for FP card/MP built-in FP must be assigned with every 8 port. If the assignment value cannot be divided by 8, the value that the remainder is omitted is assigned to 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=3 1-60>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=3 1-60>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.

Backup Call Routing to ISDN in Remote PIM Survival Mode (R12.2)

Calls within a Remote PIM over IP network are made by dialing a station number. If IP network fails, the Remote Site system works as a survival mode. In this case, prior to R12.2, the station user have to dial an outside telephone number associated with the called station number to call to the station in the Remote PIM network. In R12.2 software, if IP network fails, calls to Main/Remote Sites can be automatically routed via ISDN network. When the calling station dials the called station number, the system translates that number to the outside telephone number and makes the outgoing call via ISDN.



Service Conditions

- Host site can be SOPHO 2000 IPS, SOPHO IPS DM
- Software and Key FD for the whole system must be loaded at the Host Site. No software or key's can be loaded into the Remote Site.
- 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.
- 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.
- 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.
- 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.
- 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.
- Set the unique location number to each location group for proper setting of the IPPAD 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.
- 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.
- The backup call routing mechanism, to translate numbers, is available also for calls from main office to a remote site, when survival mode is active.
- The Calling Number presented to the called party is the ISDN DDI number instead of station number (e.g. 00356891234 instead of 1234). Calling Name information is not presented.
- SMDR call records for calls made by Remote PIM extensions while in survival mode, are lost and will not be included in Management@Net Accounting application.

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	L C D DI S PLAY
Red/Flashing (Momentarily)	0.125 seconds ON- 0.125 seconds OFF	Link Down occurrence	-
Red/Flashing (Slowly)	0.5 seconds ON- 0.5 seconds OFF	Press the button after Link Down occurrence	Link Down to Site xx (xx: Site No.)
Green/Flashing (Intermittently)	0.125 seconds ON- 0.125 seconds OFF- 0.125 seconds ON- 0.625 seconds OFF	Remote Site is survival mode after Link restoration	-

Green/Flashing (Intermittently)	0.125 seconds ON- 0.125 seconds OFF- 0.125 seconds ON- 0.625 seconds OFF	Press the button with Remote Site is survival mode after Link restoration	Link Down to Site xx (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: 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.

2.15.3 Bandwidth Requirement

Established Voice Calls		With G7.23.1 (5.3k/6.3k) Compression	With G729a (8k) Compression	Without Compression
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.71 1 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.71 1 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.

2.15.4 Required Hardware and Software

The following hardware is required (apart from the basic system and terminals) to accommodate for the remote PIM over IP configurations:

Host site:
- IP PAD card (with optional compression)
- 64 PORT SYS SOFTWARE
- LT PORT KEYS (qty must equal Host site plus all Remote sites)
- 8 SEAT IP LICENSE (qty must equal Host site plus all Remote sites)
- R-PIM 1 site license (1 required for each Remote site)

Note: Registration of Host CPU and software required.

Remote Site (a PIM, DM or DMR configuration):
- 2000 IPS (with CP24), a IPS DM (with CP24) or a IPS DMR (with CP31)
- appropriate IP PAD card (with optional compression)

Note: Registration "not" required

2.16 Language and Country version support

For the European market the tone plans and languages of the different countries are implemented and made selectable.

- Country tone plans (set with Digital Tone Generator program):
  - Germany
  - Italy
  - Netherlands
  - Austria
  - Belgium
  - Spain
  - Sweden
  - UK
  - Denmark
  - Greece
  - Swiss
- Languages for Dterm / Deskcon (English is default)
  - German
  - French
  - Dutch
  - Italian
  - Spanish
  - Portuguese
  - Swedish
  - Danish

3 Terminals

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

Analog Terminals

- BaseLine
- Dterm Single-Line Analog
- Dterm Hospitality Single-Line Analog

Dterm Series Digital Terminals	TDM	IP
• 2-Line Digital	Dterm	
• 4-Line display Digital	Dterm	DtermIP
• 8-Line non-display Digital	Dterm	
• 8-Line display Digital	Dterm	DtermIP
• 16-Line display Digital	Dterm	DtermIP
• 32-Line display Digital	Dterm	
• 60 Console Add-On Module/DSS/BLF	Dterm	
• 16 LD display Digital	Dterm	DtermIP
o 16 LD Desi-Less ADM (not in EEC)	Dterm	
• IP Soft Phone		Dterm SP30
• IP display terminal with XML (not in EEC)		Inaset

DECT (wireless) terminals

• Advanced business set	C944
• Standard business set	Zenia
• Low-end business set	C244
• Industrial Handset	I600

Attendant consoles

- Deskcon
  - Pronto, visual aid equipment on Desk Console
- SV60E
- BusinessConneCT Operator

3.1 Analogue Terminals

3.1.1 BaseLine:

Features

Ringer tone selection	Pause insertion	Indirect memories: 10
Ringer volume control	Earth recall	Memory back-up
Receiver Vol. Control	Time-break recall	Line Power
Ringer Led	Last Number Redial	Post dialling DTMF
Message Waiting Led	Save key (for temporarily storing)	Wall mounting
Mute	Direct memories: 3	
Tone dialing		

The set is supplied with a multilingual user guide and a line cord with an RJ11 plug on both sides. The set is line powered and no back-up battery is required.

The Recall function is adjustable via switches under the memo card. There is a choice of Earth Recall and Timebreak Recall. Timebreak Recall can be set to 100mS, 300mS and 600 mS. Default at delivery it is 100mS on the Europe version and the SA version, 600mS on the International one. The Ringer LED / Message Waiting LED can also be selected via a switch under the memo card.
The Europe version of the set is based on TBR38.

Note: The BaseLine Pro with display is not advised, because the display cannot be used for CLI. with the 2000 IPS.

3.1.2 Analogue Dterm:

DTR-1	DTR-1HM
	
SINGLE LINE TELEPHONE Requires one analogue port. Fully modular with Redial key, Flash key, Message Waiting lamp, Data Jack and Ring/Handset Receive Volume.	SINGLE LINE HOTEL/MOTEL TELEPHONE requires one analogue port. Fully modular with Redial key, 'Flash' key, Message Waiting Lamp, Data Jack, eight programmable Feature/Speed Dial keys and Ring/Handset Receive Volume[1]

Line Conditions for Analog Dterm Terminals

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

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

Specification for Analog Dterm Terminals

Item	Description
Size	165 x 226 x 100 mm^3 (Width x Length x Height)
Dial Pad	12-Key Dial Pad: 4 Rows and 3 Columns; Metropolitan Dial Pad with Alphabet, * and # buttons; Button 5 has a Raised Dot
Type of Dial	DTMF and Dial Pulse

[1] Teledex hotel sets can be sourced directy by the NSO via Teledex, refer to section 4.19.1.3

Item	Description
Function Buttons	Hook flash, Redial Key on DTR-1 -1; Hookflash, Program, Redial, Monitor, and Hold Keys on DTR-1HM-1
Color	– Black/silver
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-1 HM 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
Environmental Conditions	Temperature: • rated: 0° C to +40° C • transport: -10° C to +50° C Humidity: • rated: 10 to 90% RH • transport: 5 to 95% RH
EMC and safety	All SOPHO Dterm Analogue models carry a CE mark.
Approvals	c-UL (Ul 60950 3rd Edition) FCC part 15, 68, IC (Industry Canada)

3.2 Dterm Digital Terminals

The Dterm Series Digital 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 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.

DTR-2

2-Button Non-Display

2 LINE TERMINAL
(does not support optional adapters)
Fully modular with 2 Flexible, 2-color LED Line keys, nine Function Keys, built-in Speakerphone, electronic volume and tone controls.

DTR-8

8-Button Non-Display

8 LINE TERMINAL
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-4D

4-Button Display

4 LINE TERMINAL – available in black (BK) only (does not support optional adapters).

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-8D

8-Button Display

8 LINE DISPLAY TERMINAL

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-16D

16-Button Display

16 LINE DISPLAY TERMINAL

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-32D

32-Button Display

32 LINE DISPLAY TERMINAL
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.

DCR-60 CONSOLE

60-Line DSS/BLF Console

ATTENDANT ADD-ON CONSOLE - Requires an AC-R ADP (included).
Fully modular with 48 programmable, 2-color LED keys (for station trunk appearances), 12 Function keys with red LED, and tilt stand.

DTR-16LD

16-Button LD

16 LINE Back-Lit DISPLAY TERMINAL
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.

16LD-ADM

16LD-Desi-Less ADM

Add-on Module (USA only)

Extends 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.

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 1 6-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.*

Important note: Although the keys on the 8D/16D/32D look the same, they are not the same. The 8D en 16D do not have the possibility for speed dialing, only for programming line keys / function keys. On the 32D terminal, speed dial buttons can be programmed (max 16 speed dials). Then there are two options:

- keys 1-16 are used as function keys and 17-32 are speed dials
- key 1-24 are used as function keys and 25-32 are speed dials

Specification for Dterm Series 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) 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) 25~53.5 deg. (on the desk, housing tilt up) -4.4 deg. (wall mounting)
Housing Face Angle	14~25 deg. (on the desk) -4.4 deg. (wall mounting)
Recommended Headset	NEC Headsets
Other	HAC

Dterm Series Digital Terminal Options

Adapter Code	Description
CT(A)-R	Connects the Dterm Digital terminals (except the DTR-2DT) 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.
AD(A)-R	Provides the Dterm Digital terminals (except the DTR-2DT) with the ability to interface with a recording device.
AP(R)-R	Provides Dterm Digital terminals (except the DTR-2DT) ability to interface with analogue device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. Provides ringing to analogue device connected.
AP(A)-R	Provides Dterm Digital terminals (except the DTR-2DT) ability to interface with analogue device such as a cordless telephone, facsimile machine, external speakerphone, Automatic Dialer or modem. No ringing is provided.
IP-R	A compact plug-and-play device that installs into the base of a Dterm Digital terminals (except the DTR-2DT and the DTR-8)).An integrated two-port 10/100baseT Ethernet pass through hub that permits using one port toconnect the network interface card (NIC) from the PC to the IP network. The other is pluggeddirectly into a LAN or an IP network device such as a router, DSL modem or cable modem.Requires an AC-R ADP.
WM-R	Dterm Digital terminals (except the DTR-2DT) 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.

Dterm Series i Line Conditions

Cable Length	**Note 1**		**Standard**	**with AC Adapter Note 4**
Dterm Digital Series	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.

3.3 Dterm IP Terminals

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.

Convergence Features

- Two 10/100 full duplex Ethernet ports- One which connects the Dterm IP to the local Ethernet network, the other provides connectivity for a local workstation.
- Three types of powering options
    - Local AC adapter
    - Cisco Inline Power (Cisco discovery protocol CDP) for those infrastructures with an installed base of Cisco gear
    - Spare pair power across the Ethernet network.
- Transportable QoS, which follows the user no matter where they log in.
- Multiple Voice Coding support, which automatically negotiates to a common setting.
- G.711 providing an international standard for encoding/ decoding telephony signals on a 64 Kbps non compressed channel.
- It also supports the compression algorithms G.729A (8Kbps) and G.723.1 (5.3/6.3 Kbps).

Automatic Login to Home Station Number (R12.2)
Prior to R12.1, an IP terminal with MAC Address Authentication Mode can be temporarily used as a terminal with Password Authentication Mode (Login/Logout Mode). In this case, once the IP terminal with MAC Authentication Mode is logged out, a user have to login by manual operations. In R12.2 software, new operation mode is added: Fixed Connection Mode. In this mode, an IP terminal works as follows.

- An IP terminal is normally used like MAC Authentication Mode (no need to login / logout operation).
- If necessary, the terminal can be temporarily logged out and can be used as someone's own terminal by login with his/her station number & password. After he/she logged out the terminal, the terminal is automatically logged in to the home station number (e.g. conference room telephone).

Maximum 256 IP terminals can be assigned for the Fixed Connection Mode telephone. The Dterm SP30 (Soft Phone) will require future ver.11 or later to support this feature.

Dterm IP Program Download

This feature provides the method to download the latest firmware program of IP Enabled Dterm from the FTP/TFTP server automatically via system programming.
Program download can be activated by:

- Appointed Time
- Time of Login
- Designated Terminals

Physical Features

- Tiltable LCD Unit
- Adjustable Legs
- Built-in Wall Mount Unit

Note:
The difference between the Dterm IP and Dterm TDM is that the IP terminal has the functionality to login/logout and the IP terminal has it's own Music on Hold (MoH) source.
Dterm IP terminals connect to the LAN using a 100 BASE-TX cable.

The range of Dterm IP phones now consists of 5 different –3P models that all support the 802.3af standard for Power over Ethernet without the need for an adapter. The default is the -3P model however there is still a limited stock of –2P models available in the Spares list.

ITR-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)
4-Button Display	

ITR-8D

8-Button Display

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)

ITR-16D

16-Button Display

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)

Note: The Dterm Digital 16D(BL) version offers all the functionality of the 16D but is equipped with backlit display for special use in restaurants, bars, clubs and other badly lit areas. It does not require additional power.

ITR 32D

32-Button Display

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)

ITR 16LD

16-Button LD

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

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 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

3.4 Dterm SoftPhone SP30

The Dterm SoftPhone SP30 receives and places high quality voice calls over a VoIP network on the user's PC via a USB-connected headset. Call control is with a graphical user interface (GUI) that uses simple drag and drop features. The 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 OpenWorX integration, allowing TAPI-enabled applications, such as Outlook and ACT, to make and receive calls.
Most of the features and functions which exist on IP Dterm and/or IP enable Dterm can be instantly transported to the SoftPhone upon activation (log in).

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.

Note: *The Dterm SP30 as ACD client can support two operation modes, ACD and Business. The Business mode is the default mode and the ACD mode is not used in EMEA.*

Key Features

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 SP30, the main focus for the SP30 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 SP30 user and activate the application you will be collaborating on. You are now able to have a more productive conversation.
- Internet Link: The 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 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 SP30. This is ideal for the real quick conversation you need with a distant 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 SP30 will store all your chat messages in a log so that you can use them for future reference. Additional features of the 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 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 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 SP30 users for voice only conference calls. Only SP30 users will utilize additional features like application sharing and messaging during call.
- 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 SP30, no extra recording equipment other than the 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 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 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 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 SP30 is able to provide notification to the user when it is perceived to be transmitting and/or receiving voice problems. When the 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 that is shown is only capable of displaying one line of information. When configuring the 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 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 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 SP30 Call Log button which is located on the main and compact panel will flash in blue when a call was receive by the 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 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
- o **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
- o 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.

•

Dterm SP30 System Requirements

- Dterm SP30 Software CD
- SoftPhone seat Licenses: 1 license per 4 clients
- IP Seat License: 1 license per 8 seats

Operational Requirements

- PC: IBM-PC/AT Compatible
- OS: WindowsXP/2000 (required DirectX ver8.1 or higher)
- Memory: 256MB or more
- CPU: Pentium III 800MHz or more
- HDD: 50MB free space or more
- Microsoft XML: Version 3.0 or Higher (required for presence feature)
- Microsoft Netmeeting: Version 3.01 (required for application collaboration and video conferencing)
- Audio devices: Recommended USB handset or headset (to be ordered separately)

3.5 INASET™ (USA only)

Note: the INASET is not RoHS compliant and is not available in EEC countries.
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. The INASET's basic program load includes a graphical telephony application that provides telephony information and desktop control, including short text display messages and Web pages specifically tailored for the small screen format, that are 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.

The INASET includes a built-in switch, so you can use a single Ethernet switch port for the computer (data) and the INASET. The INASET is Dynamic Host Configuration Protocol (DHCP) compatible and supports G.71 1, G.729a and G.723.1 audio compression for low-bandwidth requirements.

3.6 Wireless terminals

The SOPHO 2000 IPS provides a unique feature with DECT terminals. The radio stations are connected via the LAN. The protocol that is used to carry feature information to and from the 2000 IPS is based on the PROTIMS protocol, which is also used for DTERM IP phones.

DECT handsets	C944 Advanced business set	Zenia Standard business set	C244 Low-end business set	I600 Industrial Handset
Specifications				
Weight	140 gr.	110 gr	140 gr.	110 gr.
Display size	5x16	4x14	3x16	4x16
Stand-by/Talk time	200/20hr.	240/16 hr.	200/20 hr.	100/7 hr.
Directory (ext. numbers)	100	100	50	100
Call log (extension nbrs)	30	-	20	30
Text messaging	+	-	-	+
Loudspeaker	+	+	+ / -	+
Clock and Alarm	+	-	-	-
Subscriptions	10	5	5	10
Vibrating silent ring	+	+	-	+
Dust- and waterproof	-	-	-	+
Man-down and SOS alarm	-	-	-	+

3.7 ATTENDANT CONSOLE

3.7.1 Dterm Digital 32D

The Dterm Digital 32D is especially suitable for operator functions. Along with 16 programmable keys (each with a two-colour LED) and 16 quickselection keys, it offers hands-free functionality, a headset socket and incomingcall/ message-waiting indicator lights. The terminal also has an adjustable liquid crystal display (LCD) screen with three lines of 24 characters per line. The Dterm Digital 32D can be programmed to manage 24 lines and eight quickselection keys, allowing the most common telephone operations to be accessed through a single press of a key.

Functions available to users include: recall, conference, redial, transfer, reply, hands-free and park, among others. The practical headset connector and the hands-free function provide multiple usage options and, during a call, the reception volume can be increased by three levels and reduced by two from the standard setting (six volume levels in total). The terminal also offers four ringer volume settings (off, low, medium and high) and three ringer tones (slow, medium and fast).

The additional module with 60 programmable line keys/function keys is particularly suitable to expand a switchboard operator console, available for the SOPHO Dterm Digital only.

Note: *Beside the Dterm 32D, also the 8D and 16D can be used when less functionality is required. If IP connectivity is required, the SOPHO DtermIP, 16D and 32D will fit.*

3.7.2 DeskCon

The Desk Console (SN753 DESKCON) operates on a switched-loop basis with a maximum of six Attendant Loops terminating at each Console on the associated Interface Card. The Attendant uses these loops for answering, originating, holding, extending, and re-entering calls. When Attendant Loop Release is used, the number of loops is effectively increased to a maximum of 12 for each Console.

The Desk Console is designed ergonomically, with user line keys on the left, the keyboard in the centre and frequently used function keys on the right. With this arrangement, the switchboard operator can answer or switch calls, access the keyboard and perform call-management functions in a simple, efficient and intuitive way. The rotating LCD screen, with four lines of 40 characters, displays the type of call and the extension number, as well as queued calls and other important information, all in real time. Switchboard operators can change both ringer and audio volume to compensate for background noise or to suit personal preferences.
It connects to the SOPHO IPS using the same circuit cards as the Dterm Series terminals. The IPS can support up to 8 SN753 Desk Consoles maximum.

Highlights

- LCD with 4 lines, 40 characters per line
- Software-controlled LCD loop key
- Full access to PBX features
- Headset connectivity
- Recorder connectivity

Call handling features

- The display unit shows: date and time, type of call connected to the operator, extension number, call timer, number of calls waiting and much more.
- The user line keys blink to indicate an incoming call and the light indicators show the status of the call: call in progress, call waiting, redial, prolonged waiting, waiting for reply or automatic redial.
- Source and destination keys allow the operator to speak to either the caller or the person called.
- A conversation key allows the operator to join the caller and the called party in a three-way conversation.
- Special function keys include: an emergency key for answering an incoming call from an extension with the handset disconnected for a long period of time, a line-selection key to engage a selected line, a mute key to exclude the voice of the switchboard

operator (e.g. for conferring in private) and a start key for initiating an outside call requested by an extension.
- Connection via handset or headset, depending on the operator's choice.

Dimensions: 10" wide x 9" deep x 4" high.

The chart below shows the **max cable length** that can be achieved for connecting the DeskCon to the system depending on the card it connects to.

	With AC adapter
PN-2DLCN	300 m
PN-8DLCP	1200 m

3.7.3 Interface for visual aid equipment on Desk Console

The Desk Console provides a serial interface for connection to visual aid equipment for visually impaired and blind operators, to be tested in practice for proper operation.

As user interface the Pronto can be provided, manufactured by Baum (www.baum.de). The Desk Console display information is provided to the operator via the Pronto. Call handling is done via the Desk Console keyboard. The Pronto can provide audio information to the operator via loudspeaker or mixed with the Desk Console handset signal.

The Pronto has the following key characteristics:
- Braille-elements
- Audio information (English, Italian, Spanish, German, French or Dutch)
- Text-editor, through ActivSync convertible to MS-word documents
- MP-3 Player integrated
- Data transfer from reading machine POET, ScannaR or Ovation.

The implementation requires a serial interconnection cable and optionally an audio-mixer that combines the Deskcon handset signal with the Pronto audio information.

3.7.4 SuperVisor 60E

The SV60E is an advanced Windows based workstation operator console, with multitasking capabilities to enable your telephone operator to combine call handling with other office applications.

The software architecture of the SV60E is based on industry standards. It runs on Windows NT, Windows 2000 and Windows XP.

The SV60E software contains three major parts:
- Call handling software;
- Directory handling software;
- Large database for storage and retrieval of directory information or messages.

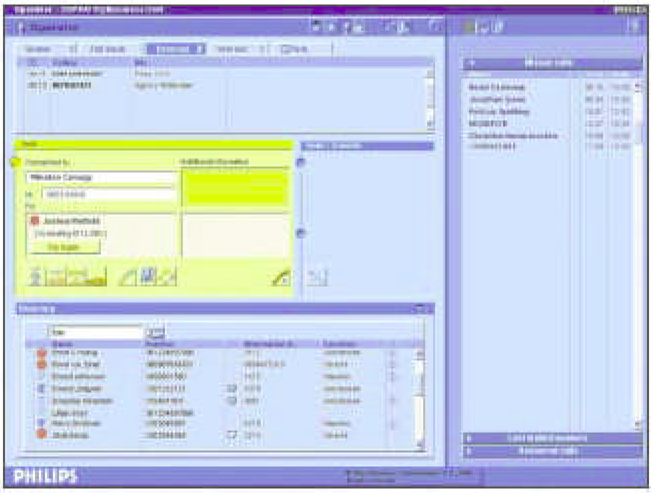
To make the SV60 software able to communicate with the SOPHO 2000 IPS the OpenWorX BAS module is required.

Networking aspects
The SV60E can share the name database with other applications such as:
- One or more SV60E consoles
- Management@Net
- OpenWorX for name display on a telephone set or for name browsing on the Feature Phones
- Property Management Systems

Interconnections are made with standard LAN hard- and software in the following configurations:
- Client-server (MSDE or SQL based)
    - o Note: In case more than three applications are used and name changes must be available to every console at once, a separate database server is advised. This is also recommended with more than 5000 names in the directory. The server is also required in combination with MyOffice@Net.
- Data Distribution Services

3.7.5 BusinessConeCT Operator

The Operator application of BusinessConneCT is a multimedia workstation for the telephone operator. It has an easy-to-use, easy-to-learn graphical user interface, easy-to-understand icons, an extensive and flexible name directory, and various messaging facilities. Its display information is designed to suit operator tasks. Calls are stored in different queues, showing the operator where a call has originated at a glance.



Other information available in the operator screen:
- external or internal call, rerouted
- presence status
- time caller has been waiting before the call was answered

Call handling features

- Full or pop-up screens provide a clear overview of different areas: queues, call handling, directory and availability lists. • Calls are handled by means of short cut keys. Drag & drop or point & click options allow operators to work quickly and efficiently.
- Separate source and destination areas allow operators to speak to the caller and/or the person called.
- Clear queue information, including the number of calls waiting, is displayed in different call categories: external incoming trunk and tie-line calls, internal operator assistance calls, fall back calls (failed DID calls that were not answered, for example) and parked calls.
- Operators have the option to retrieve a call from the top of a queue (first in, first out) or select a specific call from the queue list.
- Directory browsing in the company, external, personal or web directory enables numbers to be looked up quickly
- Free seating allows operators to log in at any workplace, using a handset (Dterm) and/or headset.
- (Note that a blind operator solution will also become available for the Business ConneCT Operator)

BusinessConneCT uses an OAI connection to the 2000 IPS.

Server platform

Processor	3 Ghz (2.4 Ghz minimal)
Memory	1 GB RAM (512 MB minimal)
Database engine	MSDE 2000, Microsoft SQL server 2000 Standard edition
Operating system	Windows 2000 Server, Windows Server 2003 Standard Edition, Windows Server 2003 Web Edition (MSDE only)

Client platform

Processor	1 Ghz
Memory	256 MB RAM
Operating	Windows XP system professional, Windows 2000 professional/server, Windows 2003 server
Web browser	IE 6.0 or higher

4 Equipment description (Logistics)

This section defines the List of Deliverables for the SOPHO 2000 IPS and SOPHO IPS DM and DMR. The DMR was not introduced with **the 2000 IPS call processing software R6.2**, but is part of the **R8 introduction time schedule.**

4.1 RoHS compliancy

Because of European legislation on Restriction of Hazardous Substances (RoHS), some system cards and components needed to be redesigned. Existing components that are not RoHS compliant will be phased out gradually. The components that are not RoHS compliant will not be manufactured anymore.

The following table shows an overview of the components that are not RoHS compliant and that will be replaced by a new version:

<table>
<tr><th>Existing board (to be phased out)</th><th>New board</th></tr>
<tr><td>9600 511 02788 SPN-CP24C MP(PBC)</td><td>9600 511 04895 SPN-CP24D MP(PBC)</td></tr>
<tr><td>9600 511 51435 SPN-CP27A MP(PBC)</td><td>9600 511 04896 SPN-CP27B MP(PBC)</td></tr>
<tr><td>9600 511 02791 SPN-CP31C MP(PBC)</td><td>9600 511 04897 SPN-CP31D MP(PBC)</td></tr>
<tr><td>9600 511 04204 SPN-30PRTA-D(AP)</td><td>9600 511 04836 SPN-DTA(PRT)-A(AP)</td></tr>
<tr><td>9600 511 51284 SPN-30DTCC-A(AP)</td><td>9600 51x xxxxx SPN-DTA with firmware</td></tr>
<tr><td>9600 511 51286 SPN-30CCTA-A(AP)</td><td>9600 51x xxxxx SPN-DTA with firmware</td></tr>
<tr><td>9600 511 51294 SPN-30PRTA-QSIG(AP)</td><td>9600 51x xxxxx SPN-DTA with firmware</td></tr>
<tr><td>9600 511 51229 SPN-AP00B MRC-C(AP)
9600 511 51486 PZ-M537</td><td rowspan="2">9600 511 04899 SPN-AP00D MRC-A(AP)</td></tr>
<tr><td>9600 511 03588 SPN-AP00B MRC-F(AP)
9600 511 51486 PZ-M537</td></tr>
<tr><td>9600 511 50125 SPN-CFTC (AP)</td><td>9600 511 04838 SPN-CFTC-A (AP)</td></tr>
</table>

Notes

- For the 2Mbit/s QSIG, Channel Associated Signalling and CCIS interface, the SPN-DTA(PRT)-A(AP) card will be provided in replacement. This card must be upgraded in the field with the proper firmware.
- The new AP00D card is both a new version of AP00B MRC-F card with same functionality and an alternative for AP00B MRC-C i.e. old AP00 card with AP00 PROG-D1 firmware.
  This alternative however does not support integrated low-end hotel functionality, i.e. a full functional front desk instrument and billing via hotel printer, PMS over RS232.
  For more details about the differences, see chapter 4.
- The new CFTC-A(AP) card is both a new version of CFTC(AP) card with same functionality and an alternative for CFTB card.
  For more details about the differences, see chapter 4.

The following table shows an overview of the components that are not RoHS compliant and that will be phased out, for some of these an alternative exists:

Existing board (to be phased out)	**Alternative board**
9600 511 51245 PN-2ODTB *(E&M trunk)*	9600 511 51256 SPN-4ODTA
9600 511 50024 PN-2DLCN *(digital long line)*	9600 511 03311 PN-4DLCT
9600 510 02513 SPN-32IPLAA IPPAD-E 9600 511 53136 SPN-16VCTAA IP PAD-B	9600 511 02540 SPN-8IPLA IPPAD-C 9600 511 51254 PZ-24IPLA
9600 511 50120 PN-CFTB *(6/10 conference)*	9600 511 04838 SPN-CFTC-A(AP)
9600 511 50137 PN-4LCAA *(LC with RGU)*	9600 511 04540 IPS DM(E) PIMMI(PBC)
9600 511 50228 PN-M10 *(fiber optic modem)*	No alternative board - no market
9600 511 02498 SPN-8ETIA ETHER SW *(in-skin hub)*	No alternative board - no market
9600 511 51300 ALM DSPP *(External alarm box)*	No alternative product - no market
9600 511 50132 SPN-IPTB-A(AP) (CCIS) 9600 511 50133 SPN-4VCTI-A (CCIS)	No alternative boards - not introduced
9600 511 50135 SPN-IPTB-B(AP) (H.323) 9600 511 50136 SPN-4VCTI-B (H.323)	No alternative boards - superseded by SIP

Notes:
1. Due to the phase out of 32IPAA+VCT IPPAD, T.30 fax support will no longer be available. However the 8IPLA and 8+24 IPLA support fax pass-through for G.711 and G.726.
2. Alternative for H.323 trunk is SIP-trunk. The market demand for H.323 trunk has practically disappeared.
3. The CCIS trunk card that will be phased out was intended for IP networking of 2000 IPS with IPK. This feature will not be supported anymore. There is no impact on the CCIS-over-IP networking capability between 2000 IPS nodes, SV7000 and iS3000 (via the IP-PAD card).

4.2 Basic System Packages

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051104207	**IPS DM SYS PKG-A**	upto 11	-	X	-	**IPS DM Basic System Package (IPS Lite)** **- Component Change: SPN-4BRTA-F(AP)**
960051104632	**IPS DM(E) SYS PKG (PBC)** contains: - IPS DM(E) PIMMI (PBC) - RACK MOUNT KIT (J) - SPN-CP31D MP (PBC) - PZ-M606-A - SPN-8IPLA IP PAD-C - SPN-4BRTA-G (AP) - PN-8LCAA - PN-4DATC	12.3	-	X	-	**IPS DM(E) Basic System Package (IPS-Lite)** · Available with R12 or later software · RoHS-compliant Replaces IPS DM SYS PKG-A (PBC)

4.3 Modules

The Basic Module of the SOPHO 2000 IPS is the PIMMG (PBC). The system capacity is defined by building configurations from 1 to maximum 8 PIMMG (with **the 2000 IPS SW 3200 Series R6.2**). In case a dual processor configuration is required PIM 0 (First PIM) should be a PIMMH; the PIMMH requires two SPN-CP27 processors.

The Basic Module of the SOPHO IPS DM is the PIMMD (PBC). The system capacity is defined by building configurations from 1 to 3 PIMMD.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150006	**ICS VS PIMMG(PBC):** SN1663 PIMMG COVER PARTS ASSEM(PBC) PZ-PW126 AC CORD-B-A PWR CNT CA-D SIDE PANEL(EU) LABEL BASE B		X	-	-	**PIMMG** **Port Interface Module (PIM) for single CPU** • Maximum 64 physical ports per PIM. • 13 Card Slots (12 LT/AP Slots + 1 MP/FP Slot) • Houses two batteries for protection from short power interruption (for 30 minutes). • At maximum configuration, the system consists of eight PIMs
960051153000	**ICS VS PIMMH (PBC)** SN1664 PIMMH COVER PARTS ASSEM(PBC) PZ-PW126 AC CORD-B-A PWR CNT CA-E SIDE PANEL(EU) LABEL BASE B		X	-	-	**PIMMH** Port Interface Module (PIM) for Back-Up CPU: (PIM0 only) • Maximum 64 physical ports per PIM. • 13 Card Slots (11 LT/AP slots + 2 MP slots) • Houses two batteries for protection from short power interruption (for 30 minutes). • At maximum configuration, the system consists of one PIMMH and seven PIMMG's
960051151121	**ICS VS BASE (PBC)** SN1545 BASE NP-254031 NAME PLATE (CE) TOP COVER ASSEM PWR CNT CA-E LABEL BASE B PL-C.P.I.MSX3X8X3GF (2 ea.)		X	-	-	**Base/Top Cover Assembly:** • One base and top cover assembly is required for each stack. • CE Marking label and EMC Class A Warning label are put on the BASE.
960051151013	**48-TW-0.7 CONN CA**		X	-	-	**BUS cable (connects PIMMG/PIMMH to PIMMG)** • Required for multiple PIM Configurations • Used to connect PIM 0 to PIM 1, PIM 0 to PIM 2, PIM 2 to PIM 3, PIM 0 to PIM 4, PIM 4 to PIM 5,, PIM 4 to PIM 6 AND PIM 6 to PIM 7.. Max. 7 cables per system.
960051151348	**COVER PARTS ASSEM (PBC)**		X	-	-	**Triangle Corner for Front Cover** (INCLUDED WITH PIMMG and PIMMH)
960051151129	**ICS VS BATTMG (PBC)** SN1671 BATTMG COVER PARTS ASSEM (PBC) LABEL BASE B		X	-	-	**External Battery Module** (PIM for mounting external batteries) • Max 4 batteries, 4 x (166mm long x 175mm wide x 125mm high) • Requires one BATT CA EXT per module.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051153001	**IPS DM PIMMD (PBC)** SN1651 IPSMD PZ-PW131 109P0624H7D09 FAN FRONT COVER ASSEM (PBC) LABEL BASE B	upto 11	-	X	X	**DM PIMMD** Port Interface Module (PIM) w/o Ringer • Maximum 40 Physical Ports per PIM • 6 Card Slots (5 LT/AP Slots + 1 MP/FP Slot) • CE, EMC Class A Warning and Manufacturing Label. • DM system at maximum configuration consists of three PIMs and provides a total of 120 physical ports (40 ports x 2). • DMR system at maximum configuration consists of two PIMs
960051104540	**IPS DM (E) PIMMI (PBC)**	12.3	-	X	X	**IPS DM PIM** · Max. 7 LT card slots per PIM · Max. 56 LT ports per PIM · AC-DC Power Unit (PZ-PW131) and Ring Generator Unit (PZ-PW139) is included · AC Power Cord is not included. Replaces IPS DM PIMMD (PBC)
960051104653	**PZ-PW139**	12.3	-	X	X	**Ring Generator Unit (RGU) for IPS DM** · RoHS-compliant

4.4 Installation hardware

The following brackets are defined for the SOPHO 2000 IPS. The use depends on selection of type of installation: wall mounted, floor standing or 19-inch rack mounting.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151369	**19 INCH RACK BRACKET (A)**		X	-	-	**19" Bracket for Rack Mounting of PIM** One per PIM 2, 3, & 4 in first tack and one per PIM 6, 7, & 8 in second stack. Used with 151371.
960051151371	**19 INCH RACK BRACKET (B)**		X	-	-	**19" Bracket for Rack Mounting first PIM in Module Stack.** One used for PIM 1 in first Module Stack and one used for PIM 5 in second Stack.
960051151372	**MOUNTING BRACKET**		X	-	-	**Bracket for Floor Standing Installation** Prevents Stack from Falling Down 1 per Stack (Attach's to Top most PIM)
960051151373	**HANGER ASSEM**		X	-	-	**Wall Mounting Bracket 1 per PIM**
960051151374	**ICS VS I/F BRACKET ASSEM**		X	-	-	**Multiple Stack Bracket Used to join neighboring PIM in second stack**
960051151375	**BASE TRAY ASSEM**		X	-	-	**Base tray for floor standing installation.**

Brackets for connecting the PIMMD modules and for installation into 19 inch rack:

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151380	**RACK MOUNT KIT(J)**		**-**	**X**	**X**	**2 RU (19 inch) Rack Mounting Bracket Kit.** • Qty two Brackets per kit. • One Kit is required for each PIMMD when Rack Mount
960051151381	**JOINT BRACKET KIT(J)**		**-**	**X**	**X**	**Joint Bracket Kit for connecting PIM's in multiple PIM configurations.** • Qty two Brackets per kit. • One Kit is required for two PIM configuration, two Kits required for three PIM configuration.

4.5 Power Unit

The following power units can be supplied within the system, the PW122 is required in case Long line LC cards are used.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151497	**PZ-PW126**		**X**	**-**	**-**	**MAIN POWER AC/DC PWR** • INPUT: AC240 V (60 Hz) • OUTPUT: –27 V (4.4 A), +5 V (7.2 A), CR (38 mA), +90 V (80mA) (INCLUDED WITH PIMMG and PIMMH)
960051151487	**PZ-PW122**		**X**	**-**	**-**	**POWER SUPPLY CARD FOR –48V.** • INPUT: DC-24V OUTPUT: DC-48V (1.7A) • Used for Long Lines with 151220 PN-4LLCB • Mounts in LT/AP card slot. • Max 1 PZ-PW122 card per PIMMG/PIMMH.

4.6 Common Control Cards

The following Common control cards are available for the SOPHO 2000 IPS.

The Main Control Processor (MP) is the central control processor of the system. The ether control daughter board is a plug-on board for the MP that provides peer-to-peer IP connection to the Ethernet and provides the interface for the OAI and MAT. The 2000 IPS with **SW 3200 Series R6.2** supports both a single and a dual processor concept.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151428	**SPN-CP24B MP(PBC)**	**upto 9**	**X**	**X**	**-**	**MAIN PROCESSOR: One card required per system.** • CPU: AMD Elan SC520 Pentium Equivalent • Memory: (FROM=8MB, SDRAM=32MB) • Built-In Functions; ¾ Pty CFT =16, DTG, PB Senders=32, RS232C = 2 PORTS at 38.4kbps, DK=1, SYSTEM CLOCK, Hold Tone Selectable Internal Melody or External Source (1 Jack/input), 4PBR, 2DAT Circuits, FP0 function. PZ-M537 Memory Expansion, DRS function, AP01 & CC01 w/M606. Major and Minor Alarm indications • PLO: PORT (MASTER/SLAVE) • MODEM: 33.6 kbps Internal • Registration: CPU and software required
960051102788	**SPN-CP24C MP (PBC)**	**10 upto 11**	**X**	**X**	**-**	**Main Processor for IPS** • Provides expanded memory for Remote Software Download in R10 software (CP24B: FROM=8MB, SDRAM=32MB, CP24C: FROM=16MB, SDRAM=64MB) • Provides same built-in functions as SPN-CP24B MP
960051104895	**SPN-CP24D MP (PBC)**	**12.1**	**X**	**X**	**-**	**Main Processor for IPS** • Same functionality as SPN-CP24C MP (PBC) except number of music source(8) Note1 • R11 or earlier software cannot be worked with this card. • RoHS-compliant Replaces SPN-CP24C MP(PBC)
960051151435	**SPN-CP27 A MP (PBC)**		**X**	**-**	**-**	**MAIN PROCESSOR for Dual MP used in PIMMH.** Two cards required per system. • CPU: AMD Flan SC520 Pentium Equivalent • Memory: (FROM=8MB, SDRAM=32MB), Key ROM. • Built-In Functions; ¾ Pty CFT =16, DTG, PB Senders=32, RS232C = 2 PORTS at 38.4kbps, DK=1, SYSTEM CLOCK, Hold Tone Selectable Internal Melody or External Source (1 Jack/input), 4PBR, 2DAT Circuits, FP0 function. PZ-M537 Memory Expansion, DRS function, AP01 & CC01 w/M606, FP0. • PLO: PORT (MASTER/SLAVE) • MODEM: 33.6 kbps Internal • Registration: CPU and software required
960051104896	**SPN-CP27B MP(PBC)**	**12.1**	**X**	**-**	**-**	**Main Processor for IPS Backup MP system** • Same functionality as SPN-CP27A MP (PBC) except number of music source(8) Note1 • R11 or ealier software cannot be worked with this card. • RoHS-compliantReplaces SPN-CP27A MP(PBC)
960051151492	**PZ-M606-A**		**X**	**X**	**X**	**Ethernet Control Daughter card** • Optionally mounts on CP24, CP27 and CP31. MP cards to accommodate Ethernet and transmit/receive TCP/IP protocol. Required for OAI, Peer-Peer IP and TCP/IP MAT. • 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card via RJ45 connector.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151429	**SPN-CP31A MP(PBC)**	upto 9	-	-	X	**Remote Processor for Remote PIM Over IP** One Card required in each Remote PIM. • CPU: AMD Elan SC520 Pentium Equivalent • Memory: (FROM=8MB, SDRAM=32MB) • Built-In Functions; ¾ Pty CFT =16, DTG, PB Senders=32, RS232C = 1 PORT (D-SUB), SYSTEM CLOCK, Hold Tone Selectable Internal Melody or External Source (1 Jack/input), 4PBR, FP0 function, DRS function, AP01 & CC01 w/M606. Major Alarm indication • MODEM: None • Registration: None
960051102791	**SPN-CP31C MP (PBC)**	10 upto 11	-	X	X	**Main Processor for DM/DMR** . Provides expanded memory for Remote Software Download in R10 software (DM only) (CP31A was FROM=8MB, SDRAM=32MB. CP31C is: FROM=16MB, SDRAM=64MB) Note that when the CP31C is used for the DMR, the Remote Software Download is not supported. - Provides same built-in functions as SPN-CP31A MP
960051104897	**SPN-CP31D MP (PBC)**	12.1	-	X	X	**Main Processor for IPS DM/DMR** · Same functionality as SPN-CP31C MP (PBC), except number of music source(8) Note1 · R11 or ealier software cannot be worked with this card. · RoHS-compliant Replaces SPN-CP31C MP(PBC)

NOTE 1:

- Due to RoHS compliance, a melody IC for Music-on-Hold on MP board is changed to a new one.
- Due to musical copyright reason, some music contents have been removed from the melody IC
- The following table shows the comparison of the music contents between current and new MP board

	Existing MP (will be phased out)	**New MP**
	PN-CP24-C PN-CP27-A PN-CP31-C	PN-CP24-D PN-CP27-B PN-CP31-D
1	Menuet [default setting]	Menuet [default setting]
2	Nocturne	Nocturne
3	For Elise	For Elise
4	The Maiden’s Prayer	The Maiden’s Prayer
5	When the saints go marching in	When the saints go marching in
6	Amaryllis	Amaryllis
7	Spring (from Four Seasons)	Spring (from Four Seasons)
8	Ich bin ein Musikante	Ich bin ein Musikante
9	If you love me	Not Available
10	Let it be	Not Available

11	It's a small world	Not Available

4.7 Firmware Processor (FP) Card

Firmware processors (FP) are required when more than 2 PIMs are used. For a 1 or 2 PIM configuration a Firmware processor is not required as this is standard available functionality on the SPN-CP24B and SPN-CP27A.

The firmware processor provides supervision and status analysis of Line/Trunk (LT) ports that reside in the PIM. The FP card is installed in slot 12 of PIM 2, 4 and 6.
For the IPS DMR the maximum configuration consists of 2 PIMs.
The FP card is installed in slot 12 of PIM 2 of a DM

The following firmware processor is available for the SOPHO 2000 IPS/DM

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
9600 51151243	SPN-CP32 FP (PBC)		X	X	-	**FIRMWARE PROCESSOR** • Required when 3 or more PIM's are mounted • DM = Max 1 card per system, mounts in third PIM (PIM 2). • IPS= Max 3 cards per system, One each mounts in third PIM (PIM 2), fifth PIM (PIM 4) and seventh PIM (PIM 6).

4.8 Line and Trunk (LT) Cards

Line / Trunk (LT) cards are listed in the following sections, LT cards resides directly under the control of the FP of the associated PIM. The following conditions apply (2000 IPS):

- Max. 512 ports per system
- Max. 64 ports per PIM
- Max.128 ports per FP

The number of allowed LT ports is controlled by licenses.

4.8.1 Analogue station

Cards for analogue lines:

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150114	PN-8LCAA		X	-	-	**8L ANALOGUE LINE CIRCUIT** • Loop Resistance: Max. 600ohms • Message Waiting Control: 8 Circuits • Momentary Open: 8 Circuits
960051002589	PN-8LCAD	10	X	-	-	**8 Circuit Analog Station card** · Provides CLI display (FSK) on analog station, together with PN-4RSTH · CLI: 8 circuits · Ring Trip: 8 circuits · Loop Resistance: Max. 600 ohms · Line Voltage: -27Vdc · Available in R10 or later software
960051103309	PN-8LCAF					8-line Analog Station Card with Message Waiting IndicationI (MWI) - Polarity Reversal - Loop resistance: Max. 600 ohms - MWI - Polarity Reversal: 8 circuits - Available with R11 or later software - RoHS compliant
960051151220	PN-4LLCB	11	X	X	X	**Four-circuit Analogue Long Line Card for Single Line Telephones:** Loop resistance for PB/DP: • PB : Maximum 1200 ohms • DP (20 PPS) : Maximum 1700 ohms • DP (10 PPS) : Maximum 2500 ohms Including the internal resistance of the distant office equipment. • Provides Message Waiting Lamp control, momentary open/ reverse functions for each circuit. • IPS= PZ-PW122 is required. Max 376 stations/ 94 cards per system • DM & DMR= PN-PW03 is required. DM max 48 stations/12 cards per system. DMR max 32 stations/8 cards per system. DM & DMR require one PN-4LCAA to be mounted per PIM to provide Ring Generator to backplane. •Loop Resistance: MAX. 2500 ohms (LLC)
960051150137	PN-4LCAA		-	X	X	**Four Circuit Analogue Station Card with Ring Generator.**

The following circuit cards are dedicated for usage in the IPS DM only.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150137	**PN-4LCAA** See Note		-	X	X	**Four Circuit Analogue Station Card with Ring Generator**
960051151252	PN-PW03		-	X	X	**-48V DC/DC Power Supply Card** Used with PN-4LLCB for long line. Maximum one PN-PW03 per PIM
960051151493	PZ-4PFTA		-	X	X	**Power Fail Transfer Circuit**

Note: the first analogue card in an IPS DM should be the 4LCAA as the IPS DM is not equipped with a ringing generator.

4.8.2 Digital Station

The following cards are applied for Digital lines. To these cards the Dterm series I terminals, the DeskConsole and the DSS Console can be connected.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150223	PN-8DLCP		X	X	X	**8 DIGITAL LINE CIRCUIT** • Used for D^{term} Digital, DSS Console & DESK CON (SN-753)
960051103311	PN-4DLCT (from release 11)	11	X	-	-	**4-line Digital Long Line Circuit Card (-48V line power feeding)** - Used for Dterm Series i, DSS/BLF Console and Desk Console - Requires PZ-PW122 for -48V line power feeding - RoHS compliant
960051150224	PN-2DLCN	upto 10	X	-	-	**2-line Digital Long Line Circuit Card** • Used for Dterm Digital, DSS Console & DESKCON (SN-753) • Provides Line Test function. • Equipped with –48 V DC-DC on-board power supply.

4.8.3 ISDN Station

The following cards provide the interface for the ISDN Basic rate (S0) station interface and D-channel handler. The S0-bus interface (2ILCA) is equipped for video & data purposes only.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150208	SPN-2ILCA		X	X	-	**2 Circuit ISDN Station Card (S/T 4 wire Interface)** • ETSI ISDN –1 Basic Rate Video & Data Terminals Requires SPN-SC03B 8ICH(AP) for every four SPN-2ILCA cards.
960051151233	SPN-SC03B 8ICH(AP)		X	X	-	**ISDN station -channel Handler Card:** • Provides the D-Channel signaling interface and controls a maximum of four ILC cards (Layer 2 and 3).

4.8.4 Central Office Trunk

The following cards provide the analogue trunk (SS-type signaling) and tie line (E&M type signaling) interface. The 8COT card can also be used in combination with paging equipment (use PN-DK00 for the relay part). In combination with the PFT card the 8COT card can offer power failure switch over from internal analogue lines to external analogue lines.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151244	**PN-8COTU** See note		**X**	**X**	**X**	**8-line Central Office Trunk Card (Loop Start Trunk):** • Provides loop detection.
960051151245	**PN-2ODTB**	**upto 11**	**X**	**X**	**-**	**2-line Out Band Dialing Trunk Card.*** • Used as either a 2-wire E&M trunk or a 4-wire E&M trunk. • Equipped with –48 V DC-DC on-board power supply. • Both No. 0 and No. 1 circuits must be set to the same purpose
96005115125	**SPN-4ODTA**	**8**	**X**	**?**	**?**	**4 circuit analog 2W/4W Tie Line card.** PZ-PW122 (-48V Power Supply) is required in case of E&M Type-I signaling. Maximum of four SPN-4ODTA per PIM, due to the power capacity of PZ-PW122. This card is available from R8 (not available for R6.2). PN-2ODTB is needed for R6.2.

Note:
For the analogue trunk interfaces the PN-8COTU card is available however with following constraints:

- The PN-8COTU supports TBR21, the investigations for TBR38 are still not finished. Already is known that for Italy, Spain, South Africa, and Brazil the current PN-8COTU requires adaptations. Implementations for these NSOs are scheduled for SW R9.
- As some country specific transmission plans do not match 100% with the PN-8COTU supported transmission plan, voice and Modem/Fax signals to and from the PSTN over the analogue trunk can be deformed.
- For voice calls over the analogue trunk the audio/voice quality will be affected, echo can occur or the user will hear a deformed audio/voice signal.
- For Modem/Fax calls over the analogue trunk the mismatch causes data failures resulting in low(er) modem/fax transfer speed rates.

As the PN-8COTU supports the SS/PD signaling, the Analogue Trunk will not be able to detect a forward release. Therefore Direct Dial-In (DDI) is not supported on the PN-8COTU.

4.8.5 Register Sender

The following card supplies the system with additional DTMF receiver registers.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function

960051151289	**PN-8RSTG**		**X**	**X**	**-**	**8 CIRCUIT PB RECEIVER DTMF Register** used with analogue single lines.
960051102787	**PN-4RSTH**	**10**	**X**	**X**	**-**	**4 Circuit Frequency Shift Key (FSK) Sender Card for analog station CLI-FSK (Sirio terminal by Telecom Italy)** - Provides analog station CLI-FSK used with PN-4LLCB - Max. 4 cards per system
960051102632	**SPN-4RSTBA-A (AP)**	**upto 11**	**X**	**X**	**-**	**4-circuit MFC Receiver/Sender Card for Brazil** - Product change from 960051151276 SPN-4RSTB-A(AP) due to manufacturing discontinuity - Provides the same functionality as 960051151276 SPN-4RSTB-A(AP)
960051104448	**SPN-4RSTBA-C(AP)**	**12.3**	**X**	**X**	**-**	**4-circuit MFC Receiver/Sender Card for Brazil** · Add MFC-ANI function for Venezuela Replaces SPN-4RSTBA-A(AP)

4.8.6 Conference

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150120	**PN-CFTB**	**upto 11**	**X**	**X**	**-**	**Enhanced Conference Trunk (less loss)** • 6-Pty CFT (1-Conference Group Per Card) or • 10-Pty CFT (1-Conference Group Per 2-Cards) • MAX 4 Cards per system Can be used for Group Call Features.
960051104838	**SPN-CFTC-A(AP)**	**12.2**	**X**	**X**	**-**	**32-Party Conference Trunk card** · Provides Group call, Meet-Me Conference and Station/Attendant Controled Conference · Some operation differences between SPN-CFTC-A (AP) and PN-CFTB. See Note2. · RoHS-compliant Replaces SPN-CFTC(AP) and PN-CFTB

NOTE 2:

- Due to RoHS compliance, PN-CFTB and PN-CFTC are consolidated into PN-CFTC-A.
- The following tables show a feature comparison between PN-CFTB vs. PN-CFTC-A:

Feature Capacity

	CFTB (will be phased out)	**CFTC**
Card type	LT card	AP card
Number of conference parties	6 parties (1 card) 10 parties (2 cards)	32 parties
Number of ports per card	8 LT ports (1 card) 16 LT ports (2 cards)	32 AP ports
Number of cards per system	4 cards	8 cards (max.256 ports)
Number of conference group per system	6-party x 4 10-party x 2 6-party x 2 + 10-party x 1	8-party x 4 16-party x 2 16-party x 1 + 8-party x 2

		32-party x 1

Feature Comparison

	CFTB (will be phased out)	**CFTC**
Station Controlled Conference	X	X (Note)
Attendant Controlled Conference	X	X (Note)
Meet-Me Conference	---	X
Group Calling – Automatic Conference	X	X
Group Calling – Broadcasting	---	X

Note: Operating procedure is different between CFTB and CFTC-A (see next section).

Operating Procedure Comparison

- Station/Attendant Controlled Conference

CFTB (will be phased out)	CFTC
1. The conference leader goes off hook, or ATT presses a LOOP key. 2. Dial the access code A for CFTB and receive special dial tone. 3. Dial the number of the 1st party to be added to the conference. 4. Press the Transfer key or SHF after the party answers and then dial the access code B to add them into the conference. 5. Sequentially call all desired parties and connect them to CFTB by dialing the access code B.	1. The conference leader goes off hook, or ATT presses a LOOP key. 2. Dial the number of the 1st party to be added to the conference. 3. Press the Transfer key or SHF after the party answers and then dial the access code for CFTC. 4. The conference leader goes on hook, or ATT presses a RELEASE key. The 1st party is connected to the conference. 5. Repeat procedure 1. to 4. above for all desired parties and connect them to CFTC.

- Station Controlled Conference (Dterm line key conference)

CFTB (will be phased out)	**CFTC**
The 6/10 party conference key and 5 or 9 line keys for participants must be assigned to Dterm of the conference leader. 1. Press the line key. 2. Dial the number of the 1st party to be added to the conference. 3. After the party answers, press the Hold key. 4. Sequentially call all desired parties using other line keys, and placing each on hold after the party answers. 5. After holding the last party, press the Conference feature key. The last party is connected to CFTB. 6. Sequentially press the holding line keys to connect the participants to CFTB.	Line keys for the same number as participants must be assigned to Dterm of the conference leader. 1. Press the line key. 2. Dial the number of the 1st party to be added to the conference. 3. After the party answers, press the Hold key. 4. Sequentially call all desired parties using other line keys, and placing each on hold after the party answers. 5. After holding the last party, repeat below procedure to connect the participants to CFTC: a) Press the holding line key b) SHF c) Dial access code for CFTC d) Go on hook to connect the participant to CFTC.

4.8.7 Digital Announcement

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150115	PN-4DATC		X	X	-	**4 Circuit Digital Announcement Trunk** (120 Seconds Per Channel)

4.8.8 Relay

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151203	PN-DK00		X	X	-	**8-circuit External Relay Control/External Key Scan Card** • Provides the above-mentioned control functions on a per circuit basis.

4.8.9 IP-PAD

The following cards provide a Packet Assemble/Disassemble (PAD) gateway from and to the IP network from the legacy interfaces in the system (TDM based interfaces). Besides the conversion function on 32IPLAA IP-PAD C card the system can provide compression protocols via the adjacent 16VCTAA IP-PAD A card, a maximum of 2 VCT cards can be applied per IP-PAD**.**

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151257	SPN-32IPLAA IP PAD-C	upto 8	X	X	X	**32-channel IP-PAD Card** • Provides Packet Assembly/Disassembly to accommodate Legacy Line/Trunk interface. • 16VCT is not required when G.711 (64K) Voice Compression is used. • Used with a maximum of two 16VCT cards when G.723.1, G.729a Compression is required • Two cards can be accommodated per built-in FP/FP card, a maximum of eight per system. • 100 BASE-TX twisted pair cable is connected directly to this card.
960051002513	SPN-32IPLAA IP PAD-E	9 upto 11	X	X	X	**32-channel IP-PAD Card** Provides a choice to set "100Mbps/full duplex mode(fixed)", in addition to "Auto Negotiation". Provides FAX over IP with: - G.711 pass through (without 16VCT card) - G.726 pass through (with SPN-16VCTAA IP PAD-B) New Firmware

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151236	SPN-16VCTAA IP PAD-A	upto 8	X	X	X	**16-channel CODEC Card for IP-PAD:** • Voice compression protocols: • G.723.1, G.729a, G.711, • FAX (14.4 Kbps), DTMF signals. • Used with SPN-32IPLAA IP PAD-C card. • Two cards can be accommodated per SPN-32IPLAA IP PAD-B card, maximum 16 per system.
960051153136	SPN-16VCTAA IP PAD-B	9 upto 11	X	X	X	**16-channel CODEC Card for IP-PAD**
960051151253	SPN-8IPLA IP PAD- A	upto 8	X	?	?	**8 Channel IP PAD and VCT card.** Supports G.711, G.723.1, G.729a codec T.30/T.38 FAX is not supported.
960051002604	SPN-8IPLA IP PAD-B	upto 9	X	?	?	**8 Channel IP PAD and VCT card.** Provides a choice to set "100Mbps/full duplex mode(fixed)", in addition to "Auto Negotiation". New Firmware: version must be rev. 2.00 or later
960051102540	SPN-8IPLA IP PAD-C	10	X	?	?	**8 Channel IP PAD and VCT card.** Add log collection function for VoIP calls in R10 software.
960051151254	PZ-24IPLA		X	?	?	**24 Channel IP PAD and VCT daughter board** Mounts on SPN-8IPLA to provide up to 32 PAD/VCT Channels. Supports G.711, G.723.1, G.729a codec In case of G.723.1, max. 16ch per card. T.30/T.38 FAX is not supported.

4.8.10 Optical Interface

This card provides an optical interface to E1/T1 trunk interfaces in the system, for DTI/CCIS/remote PIM configurations.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150228	**PN-M10** See note	upto 11	X	X	-	**Optical Fiber Interface Card:** • Provides optical fiber interface for E1 Digital Trunk Interface (2 Mbps) • Line length: 10 km (6.2 miles) or less. • Line coding: CMI

Note: was not introduced and now obsolete.

4.9 Application Processor (AP) Cards

The Application Processor (AP) Cards reside directly under the control of the MP. The following AP cards can be used in the system. The following conditions apply (2000 IPS), independent from the number of LT ports used :

- Max. 24 cards per system
- Max. 256 ports per system

The number of allowed LT ports is controlled by licenses.

4.9.1 Digital Trunk and coax cable connection

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151489	**PZ-M542**		**X**	**X**	**X**	**Coaxial Cable Connection Card:** • Used to connect a coaxial cable for the Digital Trunk Interface. • Max three Digital Trunk Interface cards can be connected to each PZ-M542. • Max two cards can be connected to LTC connector of each PIM.

4.9.2 ISDN

ISDN is supported by the 2000 IPS / IPS DM on a Primary rate card (30B+D): SPN-30PRTA and on a Basic rate card (2B+D): SPN-4BRTA

The following features are supported in SW R6.2:

- Basic call
  Basic call (BC)
  Overlap receiving
  Enhanced release protocol
- Number identification
  Calling Line Identification Presentation (CLIP)
  Calling Line Identification Restriction (CLIR)
- Advice of Charge (for Germany, Italy and Netherlands)
  Advice of charge during Call (AOC-D)
  Advice of charge at the end of the Call (AOC-E)

ISDN features in SW R8.

- Basic call
  B-channel negotiation (only exclusive)
- Number identification
  Connected Line Identification Presentation (COLP)
  Connected Line Identification Restriction (COLR)
- ISDN Addressing

ISDN functions in SW R9.

- ISDN Incomplete (for DDI-fail)

The following ISDN trunk interfaces are available: Basic and Primary rate trunk interface and D-channel handler card for the 2ILCA.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151285	SPN-30PRTA- A(AP)	upto 8	X	X	X	**ISDN Primary Rate (30B+D) Interface Card** • Provides a built-in D-Channel Handler (DCH).
960051151437	SPN-30PRTA-B(AP)	upto 9	X	X	X	**ISDN Primary Rate (30B+D) Interface Card** Provides ISDN Overlap Sending, Addressing, COLP/COLR and B-Channel Negotiation. Release 8: replaces SPN-30PRTA-A (AP), new firmware
960051102940	SPN-30PRTA-C (AP)	upto 10	X	X	X	**ISDN Primary Rate (30B+D) Interface Card** Add ISDN AOC-E for Greece, Luxemberg, Portugal, Spain and Sweden in R10 software. (in addition to Austria, Belgium, Denmark, Germany, Italy, Netherlands and Switzerland) New Firmware
960051104204	SPN-30PRTA-D (AP)	11	X	X	X	**ISDN Primary Rate Interface Trunk Card (30B+D)** Provides ISDN COLP/COLR for Telefonica (Spain) with R11 or later software (In addition to ETSI Standards) New Firmware
960051104836	SPN-DTA(PRT)-A(AP)	12.2	X	X	X	**ISDN Primary Rate Interface Trunk (30B+D) / E1 trunk card** · Provides the same functionality as SPN-30PRTA-D, plus ISDN-CCBS supplementary service (ISDN-CCBS is available with R12.3) · RoHS-compliant Replaces SPN-30PRTA-D(AP)
960051151293	SPN-4BRTA-C(AP)	upto 8	X	X	X	**4-line Basic Rate (2B+D) Interface Trunk Card** • Accommodates four two-channel PCM digital lines. • Supports point-point and point-multi-point
960051151438	SPN-4BRTA-D (AP)	upto 9	X	X	X	**4-line Basic Rate (2B+D) Interface Trunk Card** Provides ISDN Overlap Sending, Addressing, COLP/COLR and B-Channel Negotiation. Release 8: Replaces SPN-4BRTA-C (AP), new firmware
960051102939	SPN-4BRTA-E (AP)	upto 10	X	X	X	**4-line Basic Rate (2B+D) Interface Trunk Card** Add ISDN AOC-E for Greece, Luxemberg, Portugal, Spain and Sweden in R10 software. (in addition to Austria, Belgium, Denmark, Germany, Italy, Netherlands and Switzerland)
960051104204	SPN-4BRTA-F (AP)	upto 11	X	X	X	**4-Circuit ISDN Basic Rate Interface Trunk Card (2B+D)** Provides ISDN COLP/COLR for Telefonica (Spain) with R11 or later software (In addition to ETSI Standards) New Firmware
960051105101	SPN-4BRTA-G(AP)	12.2	X	X	X	**4-Circuit ISDN Basic Rate Interface Trunk card (2B+D)** · Provides the same functionality as SPN-4BRT-F, plus ISDN-CCBS supplementary service (ISDN-CCBS is available with R12.3) · RoHS-compliant Replaces SPN-4BRTA-F(AP)

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151233	**SPN-SC03B 8ICH(AP)**		X	X	-	**ISDN station -channel Handler Card:** • Provides the D-Channel signaling interface and controls a maximum of four ILC cards (Layer 2 and 3).

4.9.3 CCIS

The following cards provide the CCIS interface via point-to-point 2Mbit interface or Common channel protocol handler. A CCIS interface requires a CCIS card software key (license).

Use of CCIS over IP does not require additional hardware, in this case only the CCIS Card and IPT Card software key (license) are required.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151286	**SPN-30CCTA-A(AP)**	upto 11	X	X	-	**CCIS (2 Mbps) Trunk Card:** • Provides a built-in Common Channel Handler (CCH) for digital 2M Point-to-Point CCIS.
960051151282	**SPN-SC00 CCH-D(AP)**		X	X	-	**Common Channel Handler Card:** • Transmits/receives signals on the common signaling channel of CCIS. • Used with SPN-30PRTA-A(AP) to provide Event – Based CCIS via ISDN PRI. • Used with SPN-4BRTA-C(AP) to provide Event – Based CCIS via ISDN BRI.

Note: the SPN-30CCTA-A(AP) card will be replaced by the SPN-DTA with special firmware.

4.9.4 Q-SIG

QSIG is supported by the 2000 IPS on a Primary rate card (30B+D): SPN-30PRTA
The following features & supplementary services are supported in SW R6.2:

- Basic call (BC)
- Generic Functional Procedures (GF)
- Number identification
  - Calling Line Identification Presentation (CLIP)
  - Calling Line Identification Restriction (CLIR)
  - Connected Line Identification Presentation (COLP)
  - Connected Line Identification Restriction (COLR)
- Name identification
  - Calling / Connected Name Identification Presentation (CNIP)
  - Calling / Connected Name Identification Restriction (CNIR)

Support of future QSIG features are part of the roadmap and are described in section Future.

The following card provides the Q-SIG interface via a point-to-point 2Mbit interface. A Q-SIG interface requires a CCIS card software key (license).

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151294	**SPN-30PRTA-QSIG (AP)** PN-30PRTA SC-3176 IPS PRTA PROG-A1 SP-1045 SHS PAD PROG-A	**upto 11**	**X**	**X**	**-**	**ISDN Primary Rate (30B+D) Interface Card:** • Provides a built-in D-Channel Handler (DCH). • Provides a built-in QSIG Protocol Handler

Note: the SPN-30PRTA-QSIG (AP) card will be replaced by the SPN-DTA with special firmware.

4.9.5 H.323 IP Trunk (obsolete)

The following card is required for H.323 IP trunk interface. The card supports voice compression protocols via the associated 4VCT card.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150135	**SPN-IPTB-B H323 (AP)**	**upto 11**	**X**	**X**	**-**	**IP Trunk Card:** • Accommodates H.323 in the IP network and transmits/receives compressed voice or signals over IP network. • Maximum eight SPN-IPTB-B H323 cards per system. • One SPN-IPTB-B H323 card is used with max three PN-4VCTI-B H323 cards (12 channels). • 10 BASE-T/100 BASE-TX twisted pair cable is connected directly to this card.

The following card is required for H.323 trunk interface. The card supports voice compression protocols. The card is used adjacent to the IPTB card (IP trunk interface).

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150136	**SPN-4VCTI-B W/CA H323**	**upto 11**	**X**	**X**	**-**	**4-channel CODEC Card for IP Trunk.** VoIP (H.323) Voice compression protocols: • G.723.1, G.729a, G.711, • Used with SPN-IPTB-B H323(AP) card. • Up to three cards can be accommodated per PN-IPTB-B card, maximum 24 VCT cards per system.

Note these cards are not RoHS compliant and will be phased out.

4.9.6 SIP Trunk

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051102541	SPN-8SIP TRK-C (AP)	upto 11	X	X	-	**8-channel SIP Trunk Card (for TopLink, Germany)** - Can be expanded to 32 channels with 960051151254 PZ-24IPLA sub card (mounted on PN-8IPTA). - Max. 2 SIP trunk card per system - Max. 64 channels per system (Two SPN-8SIP TRK with PZ-24IPLA). - Codec: G.711, G.729a - FAX: G.711 pass-thorugh - DTMF: G.711 pass-through - Available with R11 or later software
960051105122	SPN-SIP TRK-E(AP)	12.2	X	X	-	**8-channel SIP Trunk Card** · Add NAT support, RTP Monitoring/Statistics , Tone Disabler and Outband DTMF(RFC2833) with R12.2 software · RoHS-compliant Replaces SPN-8SIP TRK-C(AP)

4.9.7 Call Accounting

The following cards are applied for adding Call accounting options (SMDR, Centralized SMDR) in the system (a Built-in SMDR interface is supplied by the MP). The AP00 and M537 extend the number of buffered SMDR-records. Besides this function the AP00 card also provides interface for PMS, CIS printer, Hotel Printer and MCI.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151229	SPN-AP00B MRC-C (AP)	upto 7	X	X	-	**Application Processor Card:** • Provides four RS-232C ports, and is used for SMDR, Hotel Printer, CIS, PMS, MCI, CS report functions. • One card per system
960051151258	SPN-AP00B MRC-E (AP)	8 upto 10	X	X	-	**Required when using R8 Feature Enhancement to DND (Automatic Set/Rest at appointed time)** (PMS and Hotel Printer are not supported)
960051103588	SPN-AP00B MRC-F (AP)	upto 11	X	X	-	**Application Processor card** - One AP00 card per system - Used for SMDR ,CCIS-centralized SMDR and MCI (Does not support PMS and Hotel Printer) - Provides SMDR for Station-to-Station Calls with R11 or later software (Does not support CCIS-Centralized SMDR) - Provides Executed Task Printout for MP-PMS (IP-PMS) with R11 or later software

960051151486	**PZ-M537**	**upto 11**	**X**	**X**	**-**	**Memory Expansion Card for AP00 Card:** • The system capacity is expanded when PZ-M537 is mounted on PN-AP00 (AP00) card. • AP00 supports up to 1,600 Call Records Received without PZ-M53 and up to 27,000 Call Records Received with PZ-M53.
960051104899	**SPN-AP00D MRC-A(AP)**	**12.2**	**X**	**X**	**-**	**Application Processor card** · Same functionality as SPN-AP00B MRC-F (AP) + PZ-M537, plus - AOC-E unit information output on SMDR interface - Immediate printout of call detailed information on Hotel/Motel Printer - Maid Status change result on Hotel/Motel Printer · Some feature differences between SPN-AP00B MRC-C(AP) and SPN-AP00D MRC-A(AP). See Note3. · RoHS-compliant Replaces SPN-AP00B MRC-C and SPN-AP00B MRC-F(AP).

NOTE 3
Due to RoHS compliance, the PN-AP00-B card is changed to the PN-AP00-D card.
The following table shows a feature comparison between PN-AP00-B vs. PN-AP00-D:

Hotel Feature Comparison:

Feature	AP00-B (with SC-3005 AP00 PROG-D1) (will be phased out)	AP00-D (with SC-3561 IPS MRCA PROG-C1)	PMS-IP + AP00-D (with SC-3561 IPS MRCA PROG-C1)	PMS-IP without AP00-D	Remarks
PMS interface	RS-232C	NO	IP	IP	
ISDN AOC-E on PMS interface	NO	NO	YES (via IP)	YES (via IP)	
ISDN AOC-E Unit on SMDR interface	NO	YES	YES (See Remarks)	YES (See Remarks)	SMDR information can be delivered either via PMS interface or via SMDR interface
H/M Front Desk Instrument (Dterm)					
Message Waiting – set/reset	YES	YES	YES	YES	
Room Cutoff – set/reset	YES	YES	YES	YES	
Don't Disturb – set/reset	YES	YES	YES	YES	
Automatic Wakeup – set/reset	YES	YES	YES	YES	
Check-in/Check-out – set/reset	YES	NO	NO (See Remarks)	NO (See Remarks)	Check-in/Check-out from PMS only
Maid Status change	YES	NO (See Remarks)	YES	YES	Maid Status change from guest station only
Room Status display	YES	NO	NO (See Remarks)	NO (See Remarks)	Room status can be provided by external PMS
Display totaled call charge per station	YES	NO	NO (See Remarks)	NO (See Remarks)	Call charge information can be provided by external PMS
H/M Printer connect to the AP00					
Printout for H/M feature set/reset result	YES	NO	YES (See Remarks)	NO	Refer to chapter 5 with feature descriptions
Immediate printout for call details record	YES	YES	YES	NO	

Feature	AP00-B (with SC-3005 AP00 PROG-D1) (will be phased out)	AP00-D (with SC-3561 IPS MRCA PROG-C1)	PMS-IP + AP00-D (with SC-3561 IPS MRCA PROG-C1)	PMS-IP without AP00-D	Remarks
Printout for totaled call charge per station (key operation from H/M Front Desk Instrument)	YES	NO (See Remarks)	NO (See Remarks)	NO (See Remarks)	Call charge information can be provided by external SMDR or PMS
Printout for detailed call charge per station (key operation from H/M Front Desk Instrument)	YES	NO (See Remarks)	NO (See Remarks)	NO (See Remarks)	Call charge information can be provided by external SMDR or PMS
Printout for Room Status (key operation from H/M Front Desk Instrument)	YES	NO	NO (See Remarks)	NO (See Remarks)	Room status information can be provided by external PMS

Note: Phase out of AP00-B with AP00 PROG implies that low-end hotel solution (with all functionality inside 2000 IPS and not PMS connected) will no longer be available. There was limited market demand for this functionality. Because the new AP00 does not offer all the system's Hotel Functionality on the Front Desk instrument and the billing information via the hotel printer, a low end external PMS solution is currently being investigated as an alternative.

4.9.8 Conference

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150125	**SPN-CFTC (AP)**	**till 11**	**X**	**X**	**-**	**32-Party Conference Trunk Card.**
960051104838	**SPN-CFTC-A(AP)**	**12.2**	**X**	**X**	**-**	**32-Party Conference Trunk card** · Provides Group call, Meet-Me Conference and Station/Attendant Controled Conference · Some operation differences between SPN-CFTC-A (AP) and PN-CFTB. See Note2. · RoHS-compliant Replaces SPN-CFTC(AP) and PN-CFTB

4.9.9 OAI

The following cards are applied for OAI applications using FLF:

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051002498	**SPN-AP00B DBM-C(AP)**	**9**				For PBC, this provides the same features as SPN-AP00B DBM-B(AP) for OAI applications using FLF. The reason for the change is to coincide with other NEC markets (such as US, APAC, LATAM) for future maintenance purpose. New Firmware

4.10 The 8-port In-skin Switching Hub (obsolete)

Note: this card in not RoHS compliant and will be phased out.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051102498	SPN-8ETIA ETHER SW	10 till 11	X	?	-	**8-port In-skin Switching Hub** - Layer 2 switch functions - Eight 10BASE-T/100BASE-TX ports, Auto-negotiation - Auto MDI/MDI-X - Max. 2,048 MAC address table - PoE: 802.3af or NEC protocol (signal-pair feeding) for bakup DtermIP power Number of DtermIP connections per hub: Max. 7 (IPS), Max. 4 (DM/DMR) Requires PZ-PW122 (IPS) or PN-PW03 (DM/DMR) for PoE - Max. 1 hub per PIM - Occupies 2 physical card slots for mounting the Hub (Occupies 1 physical card slots when the hub mounts in Slot 00 (IPS PIM) or Slot 02/05 (DM PIM))

4.11 Power Fail Transfer

These cards provide power fail transfer options from internal analogue lines to external analogue trunk lines during power fail situations. Typical cards combined with these options are 8LC, 4LC and 8COT cards.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150119	PZ-8PFTB		X	-	-	**8-line Power Failure Transfer Card:** • Mounted in PFT slot of PIM. • One card per PIM.
960051151493	PZ-4PFTA		-	X	X	**4-line Power Fail Transfer Circuit**

4.12 Cables

The following cables are available for bus, inter-PIM, power and battery connections.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function

960051151013	**48-TW-0.7 CONN CA**		**X**	**-**	**-**	**AC Power Cord;** • Connects to AC240 V (60 Hz) via power distribution board. • Wires FG, Neutral and Line to terminals in the Base Unit.
960051151061	**AC CORD-B- A**		**X**	**-**	**-**	**AC Power Cord - connects to AC240 V (60 Hz) via power distribution board** Wires FG, Neutral and Line to terminals in the Base Unit.
960051151060	**AC CORD-D-EU**		**X**	**-**	**-**	**AC Power Cord - connects to AC240 V (60 Hz) via (CEE Euro standard) connector.** Wires FG, Neutral and Line to terminals in the Base Unit.
960051151065	**AC CORD-D-UK**		**X**	**-**	**-**	**AC Power Cord - connects to AC240 V (60 Hz) via UK standard connector.** Wires FG, Neutral and Line to terminals in the Base Unit.
960051151043	**AC CORD-E-E**		**-**	**X**	**X**	**AC cord for Europe** • Connects PW-131 to commercial power.
960051151044	**AC CORD-E-UK**		**-**	**X**	**X**	**AC cord for United kingdom** • Connects PW-131 to commercial power.
960051151003	**BATT CA EXT**		**X**	**-**	**-**	Connects between PZ-PW126 and external batteries.
960051151002	**BATT CA INT**		**X**	**-**	**-**	Connects between PZ-PW126 and internal batteries mounted in each PIM.
960051151039	**BATT CA-P5**		**-**	**X**	**X**	**Battery Cable for External Battery Back-Up** • Connects PW-131 or PZ-4PFT to External Batteries • One BATT Cable is required for each PIM.
960051151038	**BUS-0.4 CA-PA**	**till 11**	**-**	**X**	**X**	**Bus Cable for extension of bus in multiple PIM configurations.** • One BUS Cable is required for two PIM configuration, two BUS Cables are required for three PIM configuration.
960051104627	**BUS-0.4 CA-X**	**12.3**	**-**	**X**	**X**	**Bus cable between new DM PIM (0.4m)** Replaces 960051151038 BUS-0.4 CA-PA · RoHS-compliant
960051104628	**BUS-0.4 CA-CHG**	**12.3**	**-**	**X**	**X**	**Bus cable between old DM PIM(960051153001) and new DM PIM (960051104540) (0.4m)** · RoHS-compliant
960051151029	**MAT CA-T**		**X**	**X**	**-**	**MAT CABLE FOR LOCAL MAT 6.6ft. (2.0m)** •PC SIDE: 9PIN D-SUB
960051151008	**PWR CA-A**		**X**	**-**	**-**	**DC Power Cable - 1.6 feet, 0.5 meters long** • Connects PZ-PW126 to PZ-PW126 • Used to connect PIM 0 to PIM 1, PIM 2 to PIM 3, PIM 4 to PIM 5, PIM 6 to PIM 7. Max 4 cables per system.
960051151031	**PWR CNT CA-D**		**X**	**-**	**-**	**Power Control Cable;** • Used to connect PZ-PW126 (in PIM1- PIM7) and external batteries. (INCLUDED WITH PIMMG)
960051151032	**PWR CNT CA-E**		**X**	**-**	**-**	**Power Control Cable;** • Connects between PIM 0 PZ-PW126 and Base Unit (INCLUDED WITH PIMMH and BASE)
960051151004	**RS NORM-4S CA-A**		**X**	**-**	**-**	**AC Power Cord;** • Connects to AC240 V (60 Hz) via (CEE Euro standard) connector. • Wires FG, Neutral and Line to terminals in the Base Unit.

960051151007	**RS PRT-15S CA-A**		**X**	**X**	**-**	**RS-232C CABLE, 49.2 ft. (15.0m);** • Connects SPN-AP00 to Printer
960051151021	**RS RVS-4S CA-C**		**X**	**X**	**-**	**RS-232C CABLE, 13.1 ft. (4.0m)**

4.13 Software and Licenses

The following sections give an overview of the Software and license options (software keys) of the system. The basic Software package and software keys are supplied on a diskette. Extended software keys (capacity options) are supplied on a separate key keeper (diskette). In addition to this PBC supplies the 2000 IPS SW and firmware and MatWorX SW on a separate CD-ROM. On the initial start up of the system should be loaded from the System software floppy disk and key keeper to be made operational, license activation is done via the registration wizard in the MatWorX SW.

4.13.1 Licensing

On the SOPHO 2000 IPS system, SOPHO Dterm IP telephones are controlled by the CPU and do not use digital line cards! Instead they require Dterm IP seat licenses.
These licenses are available in increments of 8 seats and are cumulative, making it easy to add seats as needed. For example, if you have 8 existing Dterm IP seats and need a total of 16, you can simply add an-other 8-seat license to reach the total of 16 seats.

The standard CPU software permits use – without license - of:

- 1 E1 card
- 48 LT Ports

This can be expanded– as an option for TDM but required for IP systems – with ‘64 Port System Software’ permitting the use of:

- 5 E1 Cards
- 5 ISDN PRI Cards
- 64 LT Ports
- 96 ISDN BRI Trunk channels

Further expansion is possible (some of which are required for an IP system) with licenses controlled by a ‘Key Keeper’, providing the following options:

- 8 to 448 IP Seat (required for IP system)
- LT Ports expansion from 64 to 128 or
- LT Ports expansion from 64 to 256 or
- LT Ports expansion from 64 to 512
- LT Ports expansion from 128 to 256
- LT Ports expansion from 128 to 512
- LT Ports expansion from 256 to 512
- 4 Soft-Phone
- CCIS Link (1) (at least required for IP system)
- CCIS Link (4)
- CCIS Link (8)
- IPT Card (1) (at least required for IP system)
- IPT Card (4)
- IPT Card (8)
- Event Based CCIS
- E1 expansion from 6 to 10 Cards

- ISDN PRI expansion from 5 to 8 Cards
- Remote PIM (required for IP system)

4.13.2 System Software

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960052150585	**64 Port Sys Soft – 3200 Series R6.2 (FD)**	**6**	**X**	**X**	**-**	**3200 Software Floppy Disk** • Business/Hotel/Motel Features; • Supports up to 64 LT Ports, 5 E1's, 5 ISDN-PRI, 96 ISDN-BRI Trunks.
960052150586	**64 Port Sys Soft - 3300 Series R8 (FD)**	**8**	**X**	**X**	**-**	Required to provide new R8 Features
960052002605	**64 Port Sys Soft - 3400 Series R9 (FD)**	**9**	**X**	**X**	**-**	Required to provide new R9 features.
960052102941	**64 Port Sys Soft - 3500 Series R10 (FD)**	**10**	**X**	**X**	**-**	Required to provide new R10 features.
960052104206	**64 Port Sys Soft - 3600 Series R11 (FD)**	**11**	**X**	**X**	**-**	- Required to provide new R11 features.
960052104517	**64 Port Sys Soft - 3700 Series R12.1 (FD)**	**12.1**	**X**	**X**	**-**	- Required to provide new R12.1 features.
960052105197	**64 Port Sys Soft - 3700 Series R12.2 (FD)**	**12.2**	**X**	**X**	**-**	**R12.2 System Software** (SC-3551 IPS BSC PROG-M1)
960004151000	**2000 IPS SW CD-ROM**		**X**	**X**	**X**	**SW CD-ROM for the 2000 IPS Software**

4.13.3 Capacity option

The basic system can be enhanced with the following capacity options.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960052150600	**LT PORTS 64 to 128**		**X**	**X**	**-**	**Expands LT Ports from 64 to 128 Ports**
960052150601	**LT PORTS 64 to 256**		**X**	**-**	**-**	**Expands LT Ports from 64to 256 Ports**
960052150602	**LT PORTS 64 to 512**		**X**	**-**	**-**	**Expands LT Ports from 64 to 512 Ports**
960052150629	**LT-64 Port**	**8**	**X**	**-**	**-**	**Required to achieve 1020 ports in a Remote PIM network.** Ports above 512 are required in 64 port increments
960052150609	**CCIS 1 CARD**		**X**	**X**	**-**	**Adds support for one CCIS Link**
960052150610	**CCIS 4 CARDS**		**X**	**X**	**-**	**Adds support for four CCIS Links**
960052150611	**CCIS 8 CARDS**		**X**	**X**	**-**	**Adds support for eight CCIS Links**
960052150618	**IPT 1 CARD**		**X**	**X**	**-**	**Adds support for one IP Trunk Card**

960052150620	**IPT 4 CARDS**		**X**	**X**	**-**	**Adds support for up to four IP Trunk Cards**
960052150628	**IPT 8 CARDS**		**X**	**X**	**-**	**Adds support for up to eight IP Trunk Card. Available with 2100 Series Software or higher.**
960052150624	**ECCIS**		**X**	**X**	**-**	**Adds Event Based CCIS capability.**
960052150641	**8 SEAT LICENSE**		**X**	**X**	**-**	**Adds support for up to eight IP stations.** Required for Peer-to-Peer IP Dterm's, INASET and Soft-Phone (SP30).
960052150496	**SOFT-PHONE 4 SEAT LICENSE**	**till 6.2**	**X**	**X**	**-**	**Adds support for up to four SP30's.** Used with 150641 8 SEAT LICENSE.
960052150497	**SP30 - 4 Seat License**		**X**	**X**	**-**	**Required for SP30 software CD.** 8 Seat IP License is also required
960052150680	**R-PIM 1 Site License**	**8**	**X**	**X**	**-**	**Adds support for Remote PIM over IP.** One license required for each Remote PIM. Each License is loaded in Main system.

For Add-on to existing Licenses:

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960052150603	**LT PORTS 128 to 256**		**X**	**-**	**-**	**Expands LT Ports from 128 to 256 Ports**
960052150604	**LT PORTS 128 to 512**		**X**	**-**	**-**	**Expands LT Ports from 128 to 512 Ports**
960052150605	**LT PORTS 256 to 512**		**X**	**-**	**-**	**Expands LT Ports from 256 to 512 Ports**
960052150612	**CCIS 1 to 4 CARDS**		**X**	**X**	**-**	**Expands CCIS Links from 1 to 4-Links**
960052150613	**CCIS 1 to 8 CARDS**		**X**	**X**	**-**	**Expands CCIS Links from 1 to 8-Links**
960052150614	**CCIS 4 to 8 CARDS**		**X**	**X**	**-**	**Expands CCIS Links from 4 to 8-Links**
960052150622	**IPT 1 to 4 CARDS**		**X**	**X**	**-**	**Adds support for IP Trunk from 1 to 4 cards**
960052150625	**IPT 1 to 8 CARDS**		**X**	**X**	**-**	**Adds support for IP Trunk from 1 to 8 cards**
960052150627	**IPT 4 to 8 CARDS**		**X**	**X**	**-**	**Adds support for IP Trunk from 4 to 8 cards.**
960052150606	**T1/E1 6 to 10 Cards**		**X**	**X**	**-**	**Expands T1/E1 Capacity between 144 to 240 Channels.**
960052150608	**ISDN DCH 5 to 8 Cards**		**X**	**X**	**-**	**Expands ISDN PRI Capacity between 5 DCH Cards and 8 DCH Cards.**

4.14 Alarm Display (obsolete)

The Alarm display unit provides an optional external alarm display unit to indicate Power On/OFF, Major Alarm and Minor Alarm status.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function

960051151300	ALARM DSPP	upto 11	X	X	X	**External Alarm Display Unit.** Requires locally supplied cable to connect between Alarm DSPP and MDF. Provides Power-On, Major - Alarm and Minor-Alarm indications.

Note: discontinued.

4.15 Patch Panel

The 24 Port patch panel is meant for installation in a 19" rack, in use with the SOPHO 2000 IPS or IPS DM(R). The standard RJ21 cabling and gender converter can be used to connect swiftly to the PIM. Each patch panel has a one-one relation with the Champ (RJ21) connector in the PIM.
The 24 port patch panel is very attractive because of the right connections (pin 25 and 50 of PIM 0 for minor and major alarm).

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051150230	24 PORT PATCH PANEL (PBC)		X	X	X	**Patch Panel for Digital and Analogue Station / Trunk ports.** One Patch Panel for every three-card slot in PIM. • Pre-wired private labeled 24 Port Patch Panel with RJ45 jacks on the front and 50 Position male champ connector on the back. • 19" Rack Mount Dimensions: 482.6mm x 43.9mm (1RU)

4.16 Spare parts IPS DM

Following spare parts have been defined for the SOPHO IPS DM:

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151306	109P0624H7D09 FAN		-	X	X	FAN UNIT (One included in each PIMMD)
960051151494	PZ-PW131		-	X	X	AC/DC Power (One included in each PIMMD)

4.17 MAT software

The MatWorX SW provides the Operational Maintenance interface with the system, either via the RS232C interface on the MP or via the IP-interface on the M606.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960050221410	**MATWorX IPS (PBC)**	**till 8**	**X**	**X**	**X**	**SW CD-ROM MatWorX 6.3.2**
960050221410	**MATWorX IPS (PBC)**	**9**	**X**	**X**	**X**	Required to provide new R9 MATWorX features New Software
960050221410	**MATWorX IPS**	**10**	**X**	**X**	**X**	Required to provide new R10 MATWorX features New Software (Replaces SA-1620 IPS MAT PROG-PJ11)
960050221410	**MATWorX IPS**	**11**	**X**	**X**	**X**	- Required to provide new R11 MATWorX features New Software (Replaces SA-1700 IPS MAT PROG-PK11)
960050221410	**MATWorX-IPS (PBC)**	**12.1**	**X**	**X**	**X**	MATWorX Ver.11.5.0+
960050221410	**MATWorX-IPS (PBC)**	**12.2**	**X**	**X**	**X**	MATWorX Software (SA-1744 IPS MAT PROG-PL11, Ver.11.6.0)

4.18 Desktop

Following sections give an overview of the Desktop terminals (analogue, digital, IP and Operator Desk Console).

4.18.1 Analogue Terminals

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960053780022	**DTR-1-1P (BK) TEL**		**X**	**X**	**X**	**Single Line Analogue Telephone** • Fully modular with Redial key, Flash Key, Message Waiting Lamp, Data Jack and Ring/handset receive volume adjustment.
960053780027	**DTR-1HM-1P (BK) TEL**		**X**	**X**	**X**	**Single Line Analogue Telephone** • Fully modular with Redial key, Flash Key, Message Waiting Lamp, Data Jack, eight programmable Feature/Speed Dial Keys and Ring/handset receive volume adjustment.

4.18.2 Digital terminals

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
9600 53780033	**DTR-2DT-1P (BK) TEL**		**X**	**X**	**X**	**2 Line Non-Display Digital Terminal:** **• 2 Flexible Line Keys with 2-Color LED** **• Eight Function keys** **• Built-in Speaker Phone** **• Electronic Volume and Tone Control** **• Does not support optional adapters**
960053780083	**DTR-8-1P (BK) TEL**		**X**	**X**	**X**	**8 Line Non-Display Digital Terminal:** **• 8 Flexible Line Keys with 2-Color LED** **• Eight Function keys** **• Built-in Speaker Phone** **• Headset Jack** **• Electronic Volume and Tone Control** **• Wall Mount Unit**
960053780085	**DTR-8D-1P (BK) TEL**		**X**	**X**	**X**	**8 Line with Display Digital Terminal** • 24 character by 3 Line Display • Four Soft-Keys • 8 Flexible Line Keys with 2-Color LED • Eight Function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit
960053780087	**DTR-16D-1P (BK) TEL**		**X**	**X**	**X**	**16 Line with Display Digital Terminal;** • 24 character by 3 Line Display • Four Soft-Keys • 16 Flexible Line Keys with 2-Color LED • Eleven Function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit
960053780043	**DTR-16D(BL)-1P (BK) TEL**		**X**	**X**	**X**	**16 Line Terminal with Back Lit Display** The Dterm Digital 16D(BL) offers all the functionality of the 16D but is equipped with back-lit display for special use in restaurants, bars, clubs and other badly lit areas. It does not require additional power.
960053780089	**DTR-32D-1P (BK) TEL**		**X**	**X**	**X**	**32 Button with Display Digital Terminal;** • 24 character by 3 Line Display • Four Soft-Keys • 24 Flexible Line Keys with 2-Color LED • Eleven Function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit
960053780051	**DTR-16LD-1P(BK)**		**X**	**X**	**X**	**16 Line with Display Digital Terminal;** • 24 character by 3 Line Display • Four Soft-Keys • 16 Flexible Line Keys with 2-Color LED and extra LCD • Eleven Function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960053780095	**DCR-60-1P (BK) Console**		**X**	**X**	**X**	**60 Button DSS/BLF or Add-On Module;** • Requires 8w AC Adapter

4.18.2.1 Digital Adapters

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960053780111	**AD(A)-RP Unit**		**X**	**X**	**X**	**Used for Tape-Recorder connection**
960053780112	**AP(R)-RP Unit**		**X**	**X**	**X**	**Analogue Telephone Adapter with ringing** • Requires 8w AC Adapter
960053780113	**AP(A)-RP Unit**		**X**	**X**	**X**	**Analogue Telephone Adapter without ringing or disconnect supervision.**
960053780114	**CT(A)-RP Unit**		**X**	**X**	**X**	**Computer Telephony Adapter (RS.232C,9 pin)**
960053780115	**IP-RP Unit**		**X**	**X**	**X**	**VoIP Adapter** • Requires 8w AC Adapter or POE
960053780116	**WM-RP Unit**		**X**	**X**	**X**	**Wall Mount Unit**

4.18.3 IP terminals

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960053780019	**ITR-4D-3P (BK) TEL**		**X**	**X**	**X**	**4 Button with Display IP Terminal** • 24 character by 3 Line Display • Four Soft-Keys • 4 Flexible Line Keys with 2-Color LED • 13 fixed function keys • Built-in Speaker Phone • Electronic Volume and Tone Control • Wall Mount Unit • Requires 8w AC Adapter or POE
960053780091	**ITR-8D-2P (BK) TEL**	**EoL**				**8 Button with Display IP Terminal**
960053780023	**ITR-8D-3P (BK) TEL**		**X**	**X**	**X**	**8 Button with Display IP Terminal** • Ethernet switch port • 24 character by 3 Line Display • Four Soft-Keys • 8 Flexible Line Keys with 2-Color LED • 15 Fixed function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit • Requires 8w AC Adapter or POE

960053780093	**ITR-16D-2P (BK) TEL**	EoL				**16 Button with Display IP Terminal**
960053780028	**ITR-16D-3P (BK) TEL**		**X**	**X**	**X**	**16 Button with Display IP Terminal** • Ethernet switch port • 24 character by 3 Line Display • Four Soft-Keys • 16 Flexible Line Keys with 2-Color LED • 15 Fixed function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit • Requires 8w AC Adapter or POE
960053780045	**ITR-32D-3P (BK) TEL**		**X**	**X**	**X**	**32 Button with Display IP Terminal** • Ethernet switch port • 24 character by 3 Line Display • Four Soft-Keys • 32 Flexible Line Keys with 2-Color LED • 15 Fixed function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit • Requires 8w AC Adapter or POE
960053780090	**ITR-16LD-3P (BK) TEL**		**X**	**X**	**X**	**16 Button with Display IP Terminal** • Ethernet switch port • 24 character by 3 Line Display • Four Soft-Keys • 16 Flexible Line Keys with 2-Color LED and extra LCD • 15 Fixed function keys • Built-in Speaker Phone • Headset Jack • Electronic Volume and Tone Control • Wall Mount Unit • Requires 8w AC Adapter or POE

4.18.4 Dterm Spare Parts

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960053780502	**HANDSET W/O CORD-P (BK)**		**X**	**X**	**X**	**Black Handset without cord**
960053770502	**LINE CORD 2.1 METER-P (BK)**		**X**	**X**	**X**	**Black 2.1 meter (7 feet)** Line Cord connects from base of terminal to wall jack.
960053780505	**HANDSET HANGER KIT-P (BK) (25)**		**X**	**X**	**X**	**Small plastic piece attached to base, under handset.** Reversible for handset Desk/Wall use. Packaged in quantities of 25.
960053780530	**HANDSET CORD 3.6 METER-P (BK)**		**X**	**X**	**X**	**Black Handset cord 3.6 meters (12 feet)**
960053780535	**HANDSET CORD 7.6 METER-P (BK)**		**X**	**X**	**X**	**Black Handset cord 7.6 meters (25 feet)**

4.18.5 Dterm Desi Labels

Desi labels are the labels used behind the transparent cover of the Dterm Phones, for e.g. type of Dterm station a separate label is available, it is possible to customize these labels using the DESI SW in the MATWorX SW, or via separate tool.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960053780492	DESI Label DCR-60-1P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DCR-60-1P
960053960053	DESI Label DTR-16D-1P, ITR-16D-2P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DTR-16D-1P, ITR-16D-2P
960053960054	DESI Label DTR-8-1P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DTR-8-1P
960053780495	DESI Label DTR-8D-1P, ITR-8D-2P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DTR-8D-1P, ITR-8D-2P
960053780492	DESI Label DCR-60-1P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DCR-60-1P
960053780496	DESI Label DTR-1-1P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DTR-1-1P
960053780497	DESI Label DTR-1HM-1P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DTR-1HM-1P
960053780498	DESI Label DTR-2DT-1P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DTR-2DT-1P
960053780499	DESI Label DTR-32D-1P (MS) (PKG 25)		X	X	X	Desi Label (Metallic Silver) for the DTR-32D-1P

4.18.6 Desk Console

The operator desk console for the system (connects to a DLC card).

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960051151449	**SN753 DESK CON -A** SN716 DESK CON F-A 6P6C(IV) LINE CORD INTENDED USE		X	X	-	**SN753 DESK CONSOLE;** • Maximum of 8 Consoles per system • Requires 15w AC Adapter • Intended Use document is included.
960051200285	SN536 DCHST A-A		X	X	-	Handset & Cradle for SN-753 DESKCON

4.18.7 AC/DC Adapters

Local AC/DC adapters For Dterm IP, DeskCon and DSS console.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960012058000	AC/DC Adapter 24V/8W INT		X	X	X	AC/DC Adapter 24V/8W for the ITR-8D-2P, ITR-16D-2P and DCR-60-1P Console (INT version)
960012059000	AC/DC Adapter 24V/8W UK		X	X	X	AC/DC Adapter 24V/8W for the ITR-8D-2P, ITR-16D-2P and DCR-60-1P Console (UK market)
960012060000	AC/DC Adapter 24V/15W INT		X	X	X	AC/DC Adapter 24V/15W for the SN753 DESK CON -A
960012061000	UK-Plug for AC/DC Adapter		X	X	X	AC Plug for AC/DCAdapter 24V/15W (UK market)

4.18.8 Headset

GN Netcom Headset for Dterm, Dterm IP and DeskCon.

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
960012053000	Headset		X	X	X	Headset for use with Dterm, Dterm IP and SN753 DeskCon

4.19 Installation materials, local MDF and Battery options

Installation methods

The 2000 IPS can be installed in three ways:

- Floor standing installation
- Wall mounting installation
- 19-inch Rack mounting installation

A Floor standing installation typically consists of 1 base unit and 1 to 4 PIMs build together to one stack, therefore the maximum configuration (8 PIMs) is build of two stacks of 4 PIMs. When an external battery option is required the BATTM will be installed on top of the base and the first PIM on the BATTM.

Each type of installation requires different installation materials (e.g. brackets and assembly materials), for floor standing, wall mounting and 19 inch rack installation these materials will automatically be generated by Prophix when the type of installation is selected.

Main Distribution Frame

The SOPHO 2000 IPS can be wired in the following ways:

- Via MDF

- Via Patch panel

MDF options:
The following MDF options are available (selection in Prophix):

MDF Option	Cables options	Connection block options
No MDF	• No cables • cables 2 meter • cables 4 meter • cables 8 meter	
MDF Adjacent (=is3030 MDF)		connecting blocks yes/no
MDF at distance	• cables 4 meter • cables 8 meter	connecting blocks yes/no

The existing iS3030 cabinet based MDF is made available for the 2000 IPS. The 2000 IPS system needs to be connected via the backpanel with 4 Champ (RJ21) connectors. Ethernet connections are placed directly on the various cards (IP trunk, IP pad and ethernet card). Prewired cables can be selected with Sofycom blocks . This implies that 1 cable is connected with 3 x 10 da Sofycom blocks.

Patchpanel
The following options result in a patch panel solution. The patch panel solution saves installation time as this option provides a completely prepared cabling solution, without additional installation effort. Available materials for a completely pre-wired solution:

- Patch panel 24 RJ45 -RJ21
- RJ21 RJ21 cable 2 meter
- RJ21 RJ21 cable 4 meter
- RJ21 RJ21 cable 8 meter
- Gender convertor

Batteries
The following battery back up solutions can be provided:

- No battery backup
- Battery backup Internal short term
- Battery backup external long term

For the 2000 IPS as much as possible existing componenents, products and suppliers already in use have been selected by PBC. The (existing) hawker batteries are in principle the preferred batteries. The 2000 IPS requires in total a 24 V external battery.

Battery back-up internal short term
Within each PIM there is the possibility to place two 12 volts 3.2 AH batteries providing a back-up time of 30 minutes. The batteries for this solution are placed inside the PIM.For connecting battries to the system and interconnecting the two batteries the internal battery cable BATT CA INT is required.

Battery back-up external long term

Within a separate external battery back-up cabinet the following batteries can be placed:
Two 24AH Batteries per 2-PIMs provides 3-Hours

Two 38AH Batteries per 2-PIMs provides 4-Hours
Two 65AH Batteries per 4-PIMs provides 4-Hours
Two 65AH Batteries per 2-PIMs provides 8-Hours

The external battery cabinet is only possible in case a floorstanding stack installation is configured. External battery backup will result in the selection of either the BATTMG module or the existing iS3000 battery cabinet NB.

Documentation

Details concerning the installation of the 2000 IPS or IPS DM can be found in following documents:

- SOPHO 2000 IPS - Installation procedures manual
- SOPHO 2000 IPS - DM Hardware Installation Guide
- SOPHO 2000 IPS - Battery cabinet & MDF installation guide

Materials

12NC	Description	Release	2000 IPS	IPS DM	IPS DMR	Function
3522 059 22420	WIRING TOOL MDF SOFYCOM		X	X	X	iS3000 MDF equipment
9562 155 64000	BATTERY CABINET NB		X	X	X	Battery / power equipment
9562 157 48100	FLOORSTAND ML -1		X	X	X	iS3000 MDF equipment
9562 157 54100	MDF iS3030		X	X	X	iS3000 MDF equipment
9562 157 58100	LOGO MDF IS3030		X	X	X	iS3000 MDF equipment
9562 160 00200	DISCONN.BLOCK 10X2		X	X	X	iS3000 MDF equipment
9562 160 02100	CONNECT BLOCK SOFY 10X2		X	X	X	iS3000 MDF equipment
9562 160 03100	CARTRIDGE OVERV. ARRESTOR		X	X	X	iS3000 MDF equipment
9562 160 04100	OVERVOLTAGE ARREST. SOFYC		X	X	X	iS3000 MDF equipment
9562 160 05100	EARTH BLOCK + CORD 10X2		X	X	X	iS3000 MDF equipment
9562 160 06100	Sofycom disconnecting plug		X	X	X	iS3000 MDF equipment
9562 160 08100	4WIRE TESTCONNECTOR SOFY		X	X	X	iS3000 MDF equipment
9562 161 72000	BATTERY SET 25AH		X	X	X	Battery / power equipment
9562 162 67000	CABLE SET NB DOUBLE AUTON		X	X	X	Battery / power equipment
9600 016 21000	Mains Cable 2.5m Int		X	X	X	Battery / power equipment
9600 016 41000	Cable RJ21-3xSofy 2mtr		X	X	X	25 pair MDF cable from PIM to MDF (Sofycom) 2mtr

9600 016 42000	**Cable RJ21-RJ21 4mtr**		**X**	**X**	**X**	**25 pair MDF cable from PIM to MDF (Sofycom) 4mtr**
9600 016 43000	**Cable RJ21-RJ21 8mtr**		**X**	**X**	**X**	**25 pair MDF cable from PIM to MDF (Sofycom) 8mtr**
9600 016 44000	**RJ21-RJ21 4M 90 Deg**		**X**	**X**	**X**	**25 pair MDF cable from PIM to MDF 4m (both ends RJ21)**
9600 016 45000	**UTP CAT 5e cable 3m**		**X**	**X**	**X**	**UTP Cat 5 Cable for connection to Ethernet (Dterm IP)**
9600 016 46000	**Gender Changer Centronics**		**X**	**X**	**X**	**Gender Changer for MDF Cable or to 24 port patch panel**
9600 120 48000	**BATTERY SET 24V 24AH**		**X**	**X**	**X**	**Battery / power equipment**
9600 120 49000	**BATTERY 12V 3.2AH**		**X**	**X**	**X**	**Battery / power equipment**
9600 120 50000	**Key Keeper**		**X**	**X**	**X**	**Diskette for software keys (licences)**

4.19.1 Supply conditions and localized versions

The SOPHO 2000 IPS and IPS DM platform contain localized information such as:

- Supported tone plans
- Supported languages (e.g. on Dterm)
- Localized User guides
- Country specific articles

In addition to this in different countries other conditions apply with respect to

- 8COTU Analogue trunk
- supported ISDN functions.
- Supported QSIG features
- Restrictions with respect to supported articles

4.19.1.1 Localized User guides

The following localized User guides are made available per market:

12NC	Document	NSO
9600 075 01000	SOPHO Dterm U.Guide Anlg	UK and ISO, INT version
9600 075 02000	SOPHO Desk Cons. U.Guide	UK and ISO, INT version
9600 075 11000	SOPHO DeskConsole UG-DE	BE, DE, AT, CH
9600 075 12000	SOPHO DeskConsole UG-IT	CH, IT
9600 075 13000	SOPHO DeskConsole UG-FR	BE, CH
9600 075 14000	SOPHO DeskConsole UG-ES	ES
9600 075 15000	SOPHO DeskConsole UG-NL	BE, NL
9600 075 16000	SOPHO DeskConsole UG-SE	SE
9600 075 17000	SOPHO DeskConsole UG-DK	DK
9600 075 00100	SOPHO Dterm U.Guide Dgtl	INT Version
9600 075 03100	SOPHO Dterm UG-DE	BE, DE, AT, CH
9600 075 04100	SOPHO Dterm UG-IT	CH, IT
9600 075 05100	SOPHO Dterm UG-FR	BE, CH
9600 075 06100	SOPHO Dterm UG-ES	ES
9600 075 07100	SOPHO Dterm UG-NL	BE, NL

9600 075 08100	SOPHO Dterm UG-SE	SE
9600 075 09100	SOPHO Dterm UG-DK	DK
9600 075 73000	SOPHO Dterm UG-PT	PT

4.19.1.2 Country version support

The List of deliverables (LOD) for 2000 IPS / IPS DM SW R6.2 is defined based upon the introduction in the European markets. On some country specific articles the definition of the LOD will differ, with SW R6.2 this is defined for the European markets and the UK market. The country specific items are automatically generated when using Prophix.

Power cord for 2000 IPS and IPS DM:

Article	Europe	UK
AC power cord IPS DM	960051151043 - AC CORD-E-E	960051151044 - AC CORD-E-UK
AC power cord 2000 IPS	960051151060 - AC CORD-D-EU	960051151065 - AC CORD-D-UK

AC/DC Adapter for Dterm IP, DSS and DeskCon.

Article	Europe	UK
AC/DC Adapter Dterm IP, DSS	9600 120 58000 - AC/DC Adapter 24V/8W INT	9600 120 59000 - AC/DC Adapter 24V/8W UK
AC/DC Adapter DeskCon	9600 120 60000 AC/DC Adapter 24V/15W	9600 120 60000 - AC/DC Adapter 24V/15WINT + 9600 120 61000 -UK-Plug for AC/DC Adapter

4.19.1.3 Support of articles in local market

For the following articles local market conditions may be different:

- Power over Ethernet: PowerDsine products in the 6xxx Series provide Dterm proprietary detection. As alternative also the Dterm in-line power supports Cisco Discovery Protocol (CDP).
- The DECT server solution for 2000 IPS is not offered for the markets: IT and ZA.
- Hotel sets can be sourced directly via Teledex, see www.teledex.com. The Pearl and the Pearl S models will be certified, because they have a European look and feel. You should specify with your order form that the sets will be connected to a NEC PBX to make sure that the correct versions of the sets are delivered (e.g. that will support Message waiting indication on the 2000 IPS).

5 FEATURE DESCRIPTION

Index to features:

5.1 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.

Analog Station CLI-FSK (R10)
This feature provides calling party number before answering incoming calls. It allows the system to connect analog telephones with Caller ID display function as PBX stations, and provides the calling party number on the display when terminating incoming calls. The calling party number information is transmitted to the called telephone using Frequency Shift Key (FSK) signals based on ETSI EN300 659, off-hook data transmission during ringing. When the calling party number does not exist, the Reason for Absence of the CLI is displayed.
<u>Required Software and Hardware:</u>

- PN-4RSTH
- PN-4LLCB
- PZ-PW122

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 (SN753 DESKCON)
The Attendant Console (SN7 16 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 CLI information with DDI fail (R11)
Any operator (DeskCon, SV60 or Business ConneCT operator) can see the identity of the Calling Number (CLI) in case any of the following DDI fail actions occur: -no answer, busy, unassigned, DND. This should also be supported by the CCIS interface
Calling and Called Party Number Display
Called Party Number Display in case of Failed DID calls to Operator

In R11 software, called party number can be displayed on an operator console (Dterm-Attendant Position, Desk Console), when failed DID calls (busy, no answer, unassigned number or Do Not Disturb) are routed to the operator. This feature is applicable for a standalone and a CCIS network system.

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 Semi-automatic Camp-on (R9)

This feature provides a convenient way for the Attendant (SN753) to preannounce caller information to the destination station. When the destination station is busy, the Attendant places the incoming caller in Semi-Automatic Camp-On. When the destination station becomes idle the Attendant is notified and can announce the call and automatically release it to the destination station. Prior to R9 the IPS only supported Automatic Camp-On to a busy station. The differences between Automatic and Semi-automatic Attendant Camp-on is a procedure to recall the camped-on station:

Automatic Attendant Camp-on

1. When the camped-on station becomes idle state(by hang-up or switch hook flash), the ""camped-on station"" is called.
2. The camped-on station answers the call, and can talk to the calling party.

Semi-automatic Attendant Camp-on

1. When the camped-on station becomes idle state, the "Attendant" is called.
2. The Attendant answers the call.
3. The camped-on station is called. The Attendant receives ringback tone.
4. The camped-on station answers the call. The camped-on station can talk to the Attendant.
5. The Attendant press the Release key. The camped-on station can talk to the calling party.

Notes

1. *The Automatic or Semi-automatic Camp-on can be assigned on a system basis.*

2. *Attendant Loop Release feature cannot be used when this feature is used.*
3. *Semi-automatic Camp-on over CCIS is not supported.*
4. *OAI cannot be used where the system uses the Semi-automatic Camp-on.*

Attendant Support of DND override(except for CCIS)

Any operator type (Dterm, DeskCon, SV60 and Business ConneCT) is able to override a DND situation. This is not (yet) supported over CCIS.

Attendant Training Jacks

The SN753 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 (MIS)

The Automatic Call Distribution (ACD) with Management Information System (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 of Daylight Saving Time (R11)

This feature allows the 2000 IPS system clock to automatically change from standard time to daylight saving time (summer time), and vice versa by pre-assigned command data.

Schedule to change to/from daylight saving time is programmed by MATWorX. This change is executed at 02:00am standard time. Change to/from daylight saving time or not can be programmed on a location number basis.

The daylight saving time is applicable to below features:

- Clock display on Dterm/Desk Console
- Time stamp on SMDR and PMS output
- Hotel Printer Service for IP-PMS
- Time stamp on Message Reminder
- Automatic Wake-Up
- Day/Night Mode Change – Automatic
- System Clock Change from Station
- LCR – Time of Day Routing

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 Reconnect to Held Party (R11)
In R11 software, a station in below case can automatically reconnect to the held party:
- Station-A is connected to station-B/outside party-B.
- Station-A does consultation hold station-B/outside party-B, then calls outside party-C.
- When outside party-C hangs up before station-A hangs up, station-A automatically reconnects to the held party (outside party-B).

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
SOPHO 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:

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

All of the various types of Call Forwarding can be set up or cancelled 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).

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.

CLI to ISDN on Call Forwarding - outside (R12.3)

When an incoming trunk call is forwarded by Call Forwarding-Outside, CLI of the external calling party or of the forwarding station can be presented to ISDN by system data programming.

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.

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.

Ring tone instead of BT if CFD to busy external party (R11)

Call forwarding to an external number. If the external number is busy, busy tone is heard

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 display enhancements (R8)
Previously, incoming trunk Caller ID (CID) Name or Number was displayed for 6 seconds after the call is answered. Incoming station calls, the number displayed for the duration of the call, but the name was only displayed for 6 seconds after the call was answered. R8 enhancement allows the incoming trunk CID Name or Number to be displayed for the duration of the call and incoming station calls display both name and number for the duration of the call (CM08>537 programming option, default is 6 seconds).
Previously, outgoing trunk calls displayed the dialed number for 6 seconds after the call is answered. Outgoing station calls, the number displayed for the duration of the call, but the name was only displayed for 6 seconds after the call is answered. R8 enhancement allows the out going trunk call dialed number to be displayed for the duration of the call and out going station call to display both name and number for the duration of the call. (CM08>538 programming option, default is 6 seconds).

Calling Party Number Display – Dterm (R11)
1) Prior to R11, when the system uses TAS (Trunk Answer from any station) as a trunk call terminating method, calling party number was displayed at the time when the incoming call is answered. In R11 software, the calling party number can be displayed before answering the call. Max. 8 Dterms can be assigned per tenant for the answering positions, by system data programming.

2) R11 software provides an option to choose "blinking or not" of the calling party number display, when receiving an incoming call to a Dterm.

Caller ID - Station (ETSI-FSK), CLI on Analogue station (R10)
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.
A new card (PN-8LCAD) is made available, for 8 analogue ports with provision of CLI information. CLI information is provided via "FSK", not via "DTMF".

Previously, CLI on analogue lines could only be provided on 4 port long line card (PN-4LLCB) in combination with PZ-PW122 power supply. The power supply is not needed for PN-8LCAD.
Caller ID stations cannot be accommodated in remote PIM.
Required Hardware: PN-8LCAD and PN-4RSTH (Sender).

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 SOPHO 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 SOPHO 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 (Six/Ten Party)
This feature permits a station user or Attendant (conference leader) to establish a Conference among as many as six or ten 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.

Conference card: alert tone on new participant (R10)

This enhancement provides alert tone to all conference participants when a new call joins the existing conference. This allows the conference participants to be aware when a new participant arrives. The Conference can be locked/unlocked to allow/disallow additional participants. This can prevent contents of the telephone conference from being overheard. Existing systems with 32-Party conference cards can take advantage by upgrading to R10. No new hardware is required: SPN-CFTC (AP)

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 SOPHO 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 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.

Delayed Hotline (R12.2)

When a station user goes off-hook and waits for a pre-programmed time without dialling, the station user is automatically routed to a predetermined station or an operator (Desk Console). If the station user dials a number before the pre-programmed time, the station user can make a call as usual.

Max. 100 stations can be assigned for the delayed hotline stations (including normal hotline stations).

Analogue single line telephones, Dterm and Dterm IP can be configured as delayed hotline stations.

Lobby telephones in hotels or hospitals are typical applications of the delayed hotline feature. If a visitor knows a station number, he/she can make a call by dialling that station number. If the visitor does not know the station number, he/she waits for a short time period, and is then automatically routed to an operator or the reception.

Desk Console: Lockout operation (R9)

This enhancement allows the Desk Console to be set into a lockout mode (by the softkey operation). This disables the console from making or receiving calls and setting or canceling service features. To return the Console to its manual operating condition, a password is required.

Notes

1. The lengh of the password can be up to 8 digits (combination of 0 to 9, * and #).
2. The password is assigned by the MAT or CAT.
3. The lockout status is cleared when the system power failure or MP/FP reset is occurred.
4. Day/Night mode change by pressing the Night key on the Desk Console is restricted during the lockout mode.

<u>Required Software and Hardware:</u>

- SN753 DESKCON

Diagnostics

To assist maintenance personnel, the SOPHO 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.

Erase Directory Entry (R12.2)

Prior to R12.2, a Dterm user can add and modify his directory data from the telephone set. In R12.2 software, the Dterm user can erase the directory data from the telephone set. This is effective for station-based directory data only (system-based directory data cannot be added, modified and erased from the telephone set).

Character Deletion (R12.2)

Prior to R12.2, when a Dterm user enters wrong character or number during directory search or directory data entry, the user cannot modify the input characters or numbers (need back from the start). In R12.2 software, the Dterm user can delete the input characters or numbers from the last input and re-enter the correct characters or numbers.

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 (PM S), 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 SOPHO 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 Routing in case of no CLI (R11)
This feature allows a malicious call without CLI to be handled by an operator, a predetermined answering position or announcement
In R11 software, when an ISDN DID call with no CLI (and no reason of absence of CLI) is received, the DID call can be routed to one of the below options per tenant basis:

- A station based on the received DID number
- LED color on the Dterm line key and ring tone pattern of the Dterm can be different between incoming call with CLI and without CLI, by system data programming
- A predetermined station
- Desk Console
- Digital Announcement Trunk (DAT)

Required Hardware:

- SPN-4BRTA-F (AP)
- SPN-30PRTA-D (AP)

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 SOPHO 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 (R8)

Previously Do Not Disturb was only available to be set by the individual Dterm users. The R8 enhancement allows the system to schedule set/cancel Do Not Disturb feature for a group of stations at pre-programmed times. The system has up to four timetables; each timetable has time to set/cancel Do Not Disturb for seven-day period (Sunday-Saturday) and 24 hours a day in 5-minute increments. Different timetables can be assigned for specific dates of the year.

- DND status can be indicated by led associated with "DND" function key programmed on a Dterm or DSS/BLF Console.
- Pressing the associated DND function key or the DSS/BLF key to set/cancel the DND feature will have no effect if the station is set to DND Group.
- The DND status set by DND Group can be temporarily changed on a individual station basis by dialing the DND feature set/cancel code or by the Attendant Console

Required Hardware
- SPN-AP00B MRC-E(AP)

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.

Do Not Disturb on Sub- or Virtual line (R10)
In addition to Prime Line, DND set/reset is available for Sub-line appearances and Virtual lines. When DND is set, it is indicated by the Sub-line and Virtual line lamp key.

Do Not Disturb Override by Operator (R12.1)
This enhancement provides increased service level of the call handling in the centralized operator configuration.
Prior to R11, an operator (Dterm attendant position/Desk Console) can override only an internal station that sets the Do Not Disturb feature. In R12.1 software, this feature is applicable over the CCIS network. The operator can override a DND station in the distant office (connected via CCIS). When the DND station is idle, the DND station will be rung. When the DND station is busy, the operator can override the busy connection and 3-party conference is established.

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 SOPHO 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).

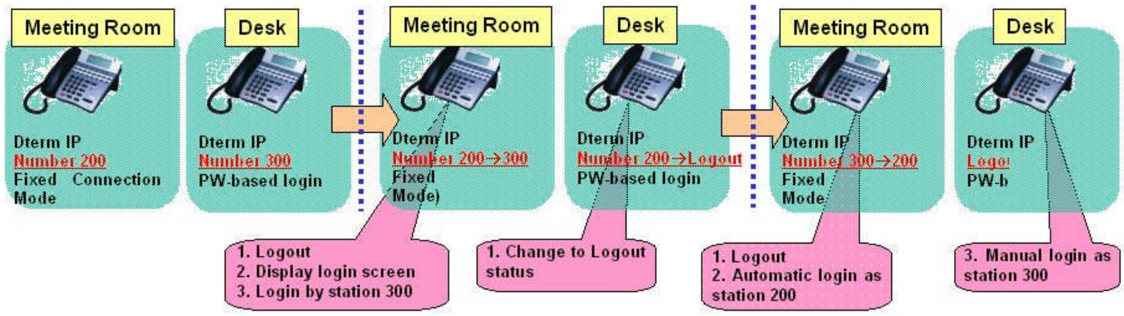
DtermIP Automatic Login to Home Station Number (R12.2)
Prior to R12.1, an IP terminal with MAC Address Authentication Mode can be temporarily used as a terminal with Password Authentication Mode (Login/Logout Mode). In this case, once the IP terminal with MAC Authentication Mode is logged out, a user have to login by manual operations. In R12.2 software, new operation mode is added: Fixed Connection Mode. In this mode, an IP terminal works as follows.

- An IP terminal is normally used like MAC Authentication Mode (no need to login / logout operation).

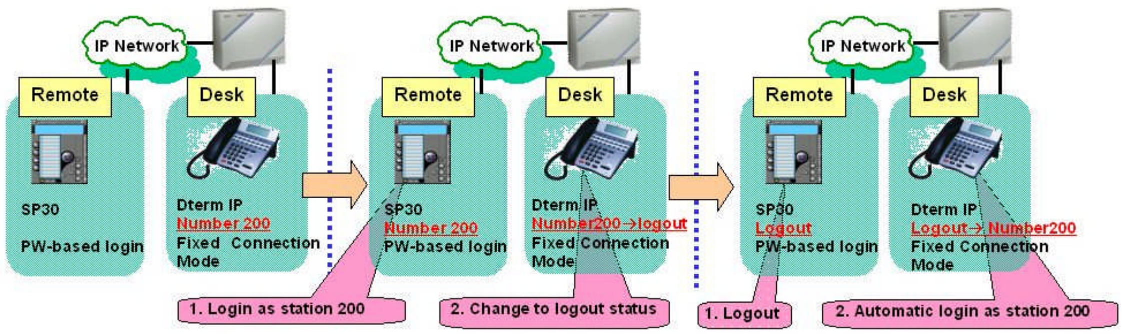
- If necessary, the terminal can be temporarily logged out and can be used as someone's own terminal by login with his/her station number & password. After he/she logged out the terminal, the terminal is automatically logged in to the home station number (e.g. conference room telephone).

Maximum 256 IP terminals can be assigned for the Fixed Connection Mode telephone.
Note: Dterm SP30 (Soft Phone) with Ver.10.2.0 or later.

Case-1: A Meeting room telephone is assigned as Fixed Connection Mode. Someone wants to use the meeting room telephone temporarily as his/her own telephone. He/she login with his/her station number. After logout, the meeting room telephone can be automatically logged in to its home station number.



Case-2: Person A has his desk phone (station 200) with Fixed Mode Connection Mode. He moves outside an office and he login with his station number (200) from his soft phone. At this time, his desk phone becomes logout status. After he logout his soft phone, his desk phone is automatically logged in to station number (200).



Dterm: My Line number display during idle state (R9)

This enhancement provides a Dterm phone with LCD display to display its My Line number during idle state. This provides a convenient and quick way to visually identify the station number assigned to the digital terminal.
Note: This feature is available on a basis of station class of service.

Dterm Preset Dialing, Off-line number preparation (R11)

This feature allows a Dterm user to prepare and verify a number in the display before dialing. When a wrong number is entered, the user can correct the number (delete the last-entered digit or return to idle state without dialing). When the user is sure that the number is correct, lift the handset or press the speaker key (or idle line key) to dial the number.

E&M TIE LINES
The SOPHO 2000 IPS supports both 2 and 4-wire, Type I and Type V, E&M Tine Line connection to other SOPHO IPS systems. An ODT (Outbound Dialing Trunk) card is required for 2 or 4-wire E&M Tie Line interface. When using a 2-wire application, one ODT card supports two Tie Lines. When using a 4-wire application, one ODT card supports one Tie Line. The SOPHO 2000 IPS supports a maximum of 256 trunks.

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.

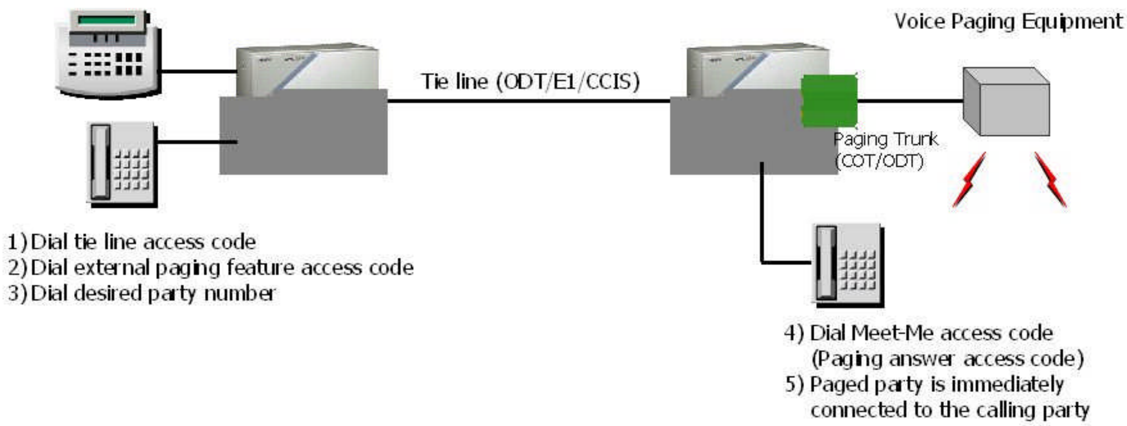
Executed Task Printout for MP-PMS (R11)
Enable AP00 to print out executed records for Wake Up call, Do Not Disturb etc. by coordination with PMS LAN interface

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.

External Paging with Meet-Me over Tie Line (R12.2)
Prior to R12.2, the External Paging with Meet-Me feature is available in a standalone configuration. Now, in R12.2 software, this feature is also available over tie line network. A station or an operator can dial-access to voice paging equipment located in another PBX connected via tie line trunk and connects both parties automatically after the page party has answered the page by dialing an access code.
Available tie line trunk type is 4ODT (2-wire/4-wire E&M trunk), E1 trunk and CCIS trunk (These trunks must support release signal sending).
Available paging trunk (external voice paging equipment interface) type is 4ODT and 8COT.



Required Hardware:

- Tie line trunk: 4ODT, E1 trunk or CCIS
- Paging trunk: 8COT with DK00, or 4ODT

Extended SMDR - CCIS, IPS to IPS, IPS to SV7000 (R9)
Extended SMDR allows both Office Number and PSTN Calling Number to be sent to Main/Center location with Centralized SMDR-CCIS. This is available for IPS to IPS and IPS to SV7000, when an SV7000 is in the network it must be the main/center location for Centralized SMDR.
Required Software and Hardware:
- SPN-AP00B MRC-C/E(AP) with SC-3168 Firmware

Flexible Line Key Assignment, One Touch key (R8)
Previously Account Code Access and Account Code could be programmed into a single One Touch-Key, the telephone number had to be programmed into a second One Touch-Key providing a two key operation. R8 enhancement allows for Account Code Access + Account Code + Trunk Access Code/LCR Access Code + Telephone Number (not to exceed a total of 26 digits) to be programmed into a single One Touch-Key.

Group Feature Key (R10)
This feature allows a quick convenient way to screen calls without tying up the boss's line. It provides the ability for a secretary to have the boss's line appear on a multifunction key. When the boss's line rings and the secretary pushes the key the call jumps to the secretary's prime line. This frees up the boss's line for additional calls. The secretary can transfer the call to the boss by pushing the multifunction key, which acts like DSS key.

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.71 1/G.726 pass-through) is supported with IPPAD/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 SOPHO 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.

Flexible Ringing Assignment by Day/Night Mode (R12.2)
Prior to R12.2, a ringing-on/off can be assigned per Dterm line/trunk key. In R12.2 software, in addition to this, the ringing-on/off per Dterm line/trunk key can be assigned by Day/Night mode status. This enhancement is applicable for Dterm, Dterm IP and Soft phone SP30.
Example: When a Dterm station 2000 has station 2001, 2002 and 2003 as sub-line, the following ringing-on/off pattern can be configured:

Line Key Nr.	Station Number	Day Mode	Night Mode
1	2000 (My Line)	Ringing-on	Ringing-on
2	2001	Ringing-on	Ringing-off
3	2002	Ringing-off	Ringing-on
4	2003	Ringing-off	Ringing-off

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 – Automatic Conference (6/10-Party)

Automatic Conference
This feature allows a Multiline Terminal user or single line telephone user within the system to establish a conference among as many as six or ten parties. From a Multiline Terminal /Single Line Telephone, a maximum of 9 stations can be paged simultaneously plus the originator. The stations are assigned to the simultaneous paging groups as participants by the system data beforehand.

2 Way Calling

This feature allows a Multiline Terminal/Single Line Telephone to page a maximum of fifteen parties simultaneously including the originator. After one of paged parties answers, the paging becomes the 2 Way Calling between the originator and the first answered party, automatically stops paging other parties. The stations are assigned to the simultaneous paging groups as participants by the system data beforehand.

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.

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 SOPHO 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.

Message Waiting Indication by Polarity Reversal (R11)

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 SOPHO 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 SOPHO 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 (USA)
This feature allows any station user direct dial access to outgoing WATS lines.

Mobility Access (from R21.1)
This feature provides a user the possibility to make/receive telephone calls from a remote location via the PBX, as if the user works from his/her desk phone.

From R12.1 software, 2000 IPS supports basic Mobility Access (MA) feature including enquiry calls and activation/deactivation by user, with enhancements in R12.2 for Call Forwarding and CLI (this feature is equivalent of SOPHO Mobility Access (SMA) in iS3000).
Required license: Mobility Access License (8 seats)

Activation/deactivation of MA feature from MA station (R21.1)
The MA feature can be activated/deactivated from a MA station telephone. The MA station dials access code for MA activation/deactivation or press the MA activation/deactivation feature button on Dterm, dials password + MA station number. During the MA feature is activated, the lamp on the MA feature button is lit.

Activation/deactivation of MA feature from remote telephone (R21.1)
The MA feature can be activated/deactivated from a remote telephone. The remote user dials DDI number for SMA activation/deactivation, dials password + MA station number.

Call from MA user - DTMF based (R21.1)
Remote MA user can access to internal PBX station and external party (via PBX), like DISA feature. The MA user dials DDI number for MA access, receive dial tone, then dials desired internal station number or external party number using DTMF. CLI display on the internal station shows the MA station number. CLI display on the external party shows the CLI number of the MA station.

Calls towards MA user (R21.1)
When an internal station or external party makes a call to a MA station, the call is forwarded to a remote MA user. CLI display on the MA user telephone shows the CLI of the MA station number.

Call Forwarding – All Calls set/cancel from a remote MA user (R12.2)
A remote MA-user can activate/deactivate the “Call Forwarding – All Calls” feature for his/her MA station. In this case, the call towards the MA station is forwarded to the Call Forwarding – All Calls destination, instead of forwarded to the remote MA user.

Cal Forwarding – Busy when a remote MA user is busy or out of radio zone (R12.2)
When a remote MA user is busy or out of radio zone, a call towards the MA user can be forwarded to a Call Forwarding – Busy destination, if it is set for the MA station.

Enquiry calls - triggered by second call, without GSM arrangement (R21.1)
A remote MA user (GSM phone) can activate an enquiry call (triggered by second call) during the conversation of a MA call:

1. During the conversation of a MA call, the MA user press CALL button.
2. The internal station user is placed on consultation hold by GSM network.
3. The MA user dials a DDI number for MA access.
4. The PBX checks the CLI of the incoming call and recognizes this is the second call from the same MA user
5. The PBX disconnects the second call and places the first MA call on consultation hold.
6. The MA user dials desired station/external party number for enquiry.

7. The MA user can alternate between first call and second call by doing the 1) to 6) above.

<u>Send CLI of outside caller to remote MA user (R12.3)</u>
When an outside caller dials in to an MA station and that call is forwarded to a remote MA user, the remote MA user can see the CLI of the outside caller

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
Sopho 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).

MP Program Download for Remote Site (R12.2)
MP program downloading via FTP server is extended for Remote Site system (DMR) in the Remote PIM over IP network.

Multi-language support on Dterm (per terminal) (R11)
Prior to R11, language displayed in the Dterm and Desk Console was selected on a system-wide basis. Multinational institutes or multilingual countries (e.g. Belgium and Switzerland) require Multi-Language Display. With R11, the language can be selected on a per-station basis (on a per-operator console basis in case of Desk Console).

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.

Night Service: Failed DDI Calls to a Night Station (R11)

Prior to R11, a night station can answer failed DID calls (destination busy, unanswered or not existing number) when the Desk Console is switched to night mode, but not failed DID calls (destination Do Not Disturb). In R11 software, the night station can answer failed DID calls with destination setting Do Not Disturb. This feature is available only in a standalone configuration.

Prior to R12.1, when a failed DID call (destination busy, no answer, etc.) is forwarded to a Dterm attendant position in the night mode, the call is routed to the attendant position and continues ringing at the attendant position. In R12.1 software, the failed DID call can be routed to a night station, instead of ringing the attendant position.

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 SOPHO 2000 IPS. See separate chapter for OAI features.

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

Hotel/Motel Printer Service for IP-PMS (R11, R12.3)

This printout information can be used as an evidence of Hotel/Motel service set/cancel activities. It is also useful for troubleshooting regarding the PMS interface.
In case of IP-PMS, below information regarding Hotel/Motel service can be automatically output to a Hotel/Motel Printer connected to AP00 card:

- Automatic Waku-Up Set/Cancel
  When Automatic Wake-Up feature is set/canceled from a guest station or an administrative terminal (Dterm Front Desk Terminal, DSS Console, Desk Console or PMS terminal), date & time, guest station number and wake-up time are printed out (Setting/canceling station number or terminal type is also printed out when the feature is set or canceled from the administrative terminal)
- Automatic Wake-Up Result
  When Automatic Wake-Up feature is executed, the result of the wake-up feature is printed out (Start Ringing, Answered, No Answer, Busy or Incomplete).
- Do Not Disturb Set/Cancel
  When Do Not Disturb feature is set or canceled from a guest station or an administrative terminal, date & time and guest station number are printed out (Setting/canceling station number or terminal type is also printed out when the feature is set or canceled from the administrative terminal)
- Message Waiting Set/Cancel
  When Message Waiting feature is set or canceled from an administrative terminal, date & time, guest station number, and setting/canceling station number (or terminal type) are printed out.
- Room Cut-Off Set/Cancel

When Room Cut-Off feature is set or canceled from an administrative terminal, date & time, guest station number, and setting/canceling station number (or terminal type) are printed out.

- Check-In/Check-Out Set/Cancel
  When Check-In/Check-Out is set or canceled from a PMS terminal, date & time and guest station number are printed out.
- PMS Interface Established/Failed
  When PMS interface is established or failed, date & time of these events is printed out.

In R12.3, below additional information will be output to the Hotel/Motel Printer.

- Maid Status change results
- Immediate printout for call detailed records
  This printout information can be used as an evidence of Hotel/Motel service set/cancel activities. It is also useful for troubleshooting regarding the PMS interface.

Required Software and Hardware:

- SPN-AP00D MRC-A (AP)
- RS PRT-15S CA-A
- RS-232C Printer (locally provided)

Proprietary Multiline Terminal

The Multiline Terminals that can be used with the system:

- Dterm Series i
- Dterm Series E

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.

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).

Q.SIG

This feature allows the SOPHO 2000 IPS to provide basic connection service when interfacing to a QSIG network. A 30 B-channel 2Mps (E1) digital interface with a Q.SIG D-channel handler (SPN30PRTA-QSIG) card is required for each physical interface.

QSIG Overlap Sending/Receiving (R9)

R9 enhancement provides an option to use overlap sending/receiving for QSIG, based on ETS 300 172. Max 32 digits of the called number can be sent/received with SETUP + INFO message(s); in case of tandem connection, max. 24 digits of the called number. Tandem connection is supported for a call from/to ISDN, CCIS, QSIG, E1/T1 or analog E&M trunk.

Note: In the overlap sending, T304 timer is fixed as 15s (though ETSI defines 20s or more).

Required Hardware: SPN-30PRTA-QSIG(AP) or SPN-24PRTA-QSIG(AP).

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 MP Software Download (R10)

This feature provides a remote MP software download function via an IP network when the system is on-line mode. By the command from the MATWorX, the MP software stored in the external FTP server is downloaded to the MP. The downloaded software stored in the MP is changed over to the existing working MP software immediately after the download is completed or based on the schedule programmed by the MATWorX. If the changeover fails, fallback to the old software can be executed.

This feature provides an easy way for software updates from a remote maintenance center to systems dispersed geographically.

Required Software and Hardware:

- SPN-CP24C MP (PBC)
- SPN-CP31C MP (PBC); Note: When the SPN-CP31C is used for remote site MP in the Remote PIM network, the remote software download feature is not supported.
- MATWorX IPS for R10

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.

Prior to R11, single-digit feature access codes (postdialing digit) were fixed by the system (e.g. 2=call back, 3=Executive Override). With R11 software, these codes are flexible by system data programming. 0 to 9, * and # can be used for these codes.

Short Message Service (SMS) Transparency (R10)
This feature enables the use of the Short Message Service provided by Telecom Italy using the analog SMS terminals. The feature allows analog terminals with the SMS function to make/receive a call to/from a Short Message Service Center (SM-SC) and send/receive a short text message. The SMS is a service provided by Telecom Italy that allows text messages to be sent and received. The text message communications are controlled by the SMS terminal and SM-SC and transmitted using Frequency Shift Key (FSK) signals over the speech path. When the SMS call is received, the Calling Line Identity (CLI) of the SM-SC will be displayed on the SMS terminal.
<u>Required Hardware:</u>
- PN-4RSTH
- PN-4LLCB
- PZ-PW122

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 SOPHO 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 (Last number redial)
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.

R11 software provides below enhancements regarding the Stack Dial feature.

Tone Suppression

Feature Dial Tone after pressing the Redial key on Dterm can be suppressed by system data programming. The tone suppression is effective from 2nd or later press of the Redial key (The tone is provided In case of 1st press of the Redial key from an on-hook state). This will provide an option for users who do not like to hear the tone during searching the redial number.

Extend Time-out Timer

Timeout timer after pressing the Redial Key is extended from 6 to 16 seconds. This will provide the user with more time to check the displayed number.

Automatic Idle Return after Timeout

After the timeout timer after pressing the Redial key has expired (16 sec), the Dterm can be automatically returned to an idle state by system data programming (Prior to R11, Re-Order Tone continues sending until the Dterm user replaces the handset or presses the Speaker key to return to the idle state).

No NUT in Redial List (Stack Dial)

On time-out the set shall go back to idle and place a hint text on the display to replace the handset or go on-hook with the LSP key.

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.

SMDR Call Release Reason (R10)

Output of Call Release Reason and Call Received Time can be useful to determine recourse requirements. Enabling SMDR output of abandoned incoming call improves services because it becomes possible to judge whether resources are excessive or deficient.

Call Release Reason (Abandoned Call)

Prior to R10, an SMDR can output a call release reason of:
- Normal: an incoming call is released after called station answers the call.
- Diverted/Transferred: an incoming call is released after the call is diverted or transferred and is answered by the forwarding/transferring destination.

In R10 enhancement, a release reason of “Call Abandoned” can also be output to SMDR. When an incoming trunk call is released before the called station answers, the abandoned call information can be output to the SMDR call record.

Call Received Time

In R10 enhancement, in case of an incoming trunk to station call, a Call Received Time can be output to the SMDR call record, in addition to the Call Answered Time. This enhancement allows an external SMDR application can provide a Ring Time before answering the incoming call.

Required Hardware

- SPN-AP00B MRC-E with SC-3168 PROG-A1 firmware rev.3.0 or later

SMDR for Station-to-Station Calls (R11)

R11 software supports SMDR for station-to-station calls, in addition to station-to-trunk calls, trunk-to-station and trunk-to-trunk calls. This enhancement can be provided by station class of service. The SMDR call record can be output via MP-IP port, MP-RS232C port or AP00-RS232C port (PMS-IP port and CCIS Centralized Billing are not supported). The output format can be 2400 standard format or 2400 extended format. This feature can be enabled for SLT, Dterm, Dterm IP, PS or Softphone.

Required Hardware:

- PZ-M606-A (in case of MP-IP port)
- SPN-AP00B MRC-F (AP) (in case of AP00-RS232C port)

SMDR for ISDN-COLP/COLR for Telefonica, Spain (R11)

Prior to R11, 2000 IPS supports the ISDN COLP/COLR supplementary service based on the ETSI standards. In R11 software, the IPS also supports the ISDN COLP/COLR based on the Spanish national standards used by Telefonica.
These are now identified with the following methods:
1. ISDN-COLP/COLR (ETSI EN 300 097-1) standard ISDN Connected number information element (IE) is part of codeset 0.
2. Spanish ISDN-COLP/COLR ISDN Connected number information element (IE) is handled if it is part of codeset 5.

COLP: Connected Line Identification – Presentation
COLR: Connected Line Identification – Restriction

Required Hardware:
- SPN-4BRTA-F (AP)
- SPN-30PRTA-D (AP)

SMDR/PMS interface on IP (R9)

SMDR on IP: SMDR in R9 now has the ability to transmit over TCP/IP via the built-in AP00 function on the CPU (on IP port of the MP card, M606A). Only one connection is allowed using port 60010. Once the SMDR over IP function is enabled the associated RS port on the CPU is disabled and cannot be used for any other function. Also, call records are sent to the external SMDR device, requested by the SMDR device. Max. 1024 call records are stored in the MP.
- In a Remote PIM configuration, the SMDR device must be connected to the Main site.
- In case of CCIS Centralized Billing, the SMDR on IP can be used at a center office, but not at a local office. In this case, the center office sends the call records from the local office to the SMDR device when received. (Not wait for receiving the request from the SMDR device)

PMS on IP: PMS in R9 now has the ability to transmit over TCP/IP via the built-in AP00 function on the CPU (via IP port of the MP card, M606A). Only one connection is allowed using port 60050. When using PMS over IP an AP00 cannot be used in the system. The IP interface is compatible with model 90/120 message specifications.
- Hotel/Motel printer cannot be used when the PMS on IP is used.
- Check-in/Check-out cannot be set/cancel from Dterm-Hotel/Motel Front Desk Terminal.
- In a Remote PIM configuration, the PMS must be connected to the Main site.

Required Software and Hardware:
- PZ-M606-A

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.

Station Speed Dialing with Account Code (R11)

Prior to R11, One-Touch Keys on Dterm can store station numbers or trunk access code + outside telephone numbers. In R11 software, Account Code, Forced Account Code or Authorization Code can also be programmed on the One-Touch Keys.
E.g.:

- “Account Code access code” + “Account Code” + “Trunk Access Code” + “Outside Telephone Number”
- “Forced Account Code access code” + “Forced Account Code” + “Trunk Access Code” + “Outside Telephone Number”
- “Authorization Code access code” + “Authorization Code” + “Trunk Access Code” + “Outside Telephone Number”

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 SOPHO 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.

System Speed Dialing (R8)
Previously a maximum of 300 System Speed dialing buffers for 3-digit abbreviated code (000 - 999) were available. R8 enhancement allows for 10.000 system speed dialing buffers using 4-digit abbreviated codes (0000-9999). System speed dialing with 3-digit abbreviated code is also available as before. Since system speed dialing with 4 digit abbreviated code uses station speed dialing memory, the available number of station speed dialing memory blocks is reduced concurrently for Dterm One touch key/station speed dialing.

System Speed Dialing: Programmable Length of Abbreviated Code (R11)
Prior to R11, an abbreviated code of System Speed Dialing was 4-digit number (fixed). With R11 software, the abbreviated code can be programmed in 1- to 8-digit number. 0 to 9 can be used for the abbreviated code digit.

Tenant Service
This feature provides for more than one organization (tenant) to share the same SOPHO 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 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 SOPHO 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 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.

User Mobility (IP Remote PIM Network)

This feature allows users visiting other locations (main or remotes) in the IPS Remote PIM network to log-in to an IP Dterm using their home station number and password. Mobile/visiting employees retain their home station in both normal and survivable mode.

DID number, features/applications can be retained by the employee at any location in the network. This allows the mobile/visiting employees to retain their productivity.

User Mobility supports two scenarios.
1. User moves with their IP terminal from one site to another (the MAC address of the terminal is not changed).
2. User moves only with the station number rather than the IP terminal (the MAC address of the terminal is changed). In this case, they are allowed to log in (override) on a terminal without logging out on the terminal you used before moving.

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 SOPHO 2000 IPS with a locally provided stand-alone type Voice Mail System (VMS). The VMS, connected to the SOPHO 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.

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.

5.2 CCIS features

Networking - a powerful feature of the SOPHO 2000 IPS that sets it above competitive systems of comparable size. This concept is based on the common Channel Inter-Office Signaling (CCIS) protocol.
This protocol derives its functionality from the CCITT SS#7 recommendations. The CCIS signaling link, common to all voice and data channels, exchanges information relating to addressing (dialed digits, calling/called number), supervisory functions (call setup & termination), network accounting & management (centralized billing and fault reporting).

The key capabilities and benefits are:

- 95% Feature Transparency
- Centralized Services
- Facilities Optimization
- Equipment Interchangeability
- LAN/WAN connectivity

With CCIS networking provides:

- Centralized Services
    - Centralized Voice Mail
    - Centralized SMDR/Call Accounting
    - Centralized Attendant
    - Centralized Emergency call facilities
    - Busy Lamp Field for Predetermined Stations

  Initial benefits include the reduction of time spent training users that are transferred between locations and reduction of transport costs based on geographical location of remote site.
- Facilities Optimization
    - Centralized Public Switched Network Services: Analogue, E1, ISDN PRI/BRI, Calling Party Identification
    - Equipment Interchangeability

  Initial benefits include: the ability to move equipment between locations; reduce time spent training end-users transferred between offices; and reduce costs associated with specialized telephone services.
- Station Number Sending
  Provides individual identification of each user regardless of station location on the network.
- Calling Party Information
    - Caller ID for Incoming Transferred Call over CCIS
    - 16 Digit CPN/ANI/Caller ID over CCIS (via LCR, Least Cost Routing)
    - Caller ID Analogue Station over CCIS

  Users will benefit from receiving Calling Party information regardless of trunks/station location on the network and when calls are transferred from another system on the network.
- Busy Lamp Field for Predetermined Stations
  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 Multi-line Terminal. Pressing the DSS button allows a direct access to the programmed station within the CCIS Network. Managers and supervisors at remote sites without a local assistant

can receive administrative support from the another site.

- Multiple Call Forwarding All/Busy/No Answer
  This feature allows the last hop of a Multiple Call Forwarding sequence to be forwarded over a CCIS Network to a station in another office. When Multiple Call Forwarding over CCIS link is allowed in system programming, the number of Call Forwarding over CCIS link is up to 7 combined with Multiple Call Forwarding-All Calls/Busy Line/Don't Answer. When Multiple Call Forwarding over CCIS link is not allowed in system programming, Call Forwarding over CCIS link is available only once.
  Provides additional flexibility for telephone coverage to customers that require creative solutions that support their business needs.
- and many user facilities, such as: Call back, Call forward DND (especially useful for workers in virtual offices without local administrative support), Dual hold, Hot line, Voice call (intercom), Hands-free answerback.
  Alternate Routing: provides alternate routing from IP trunks to other PSTN trunks (E1/ISDN) when the data network or intranet is not available.
  This is a major advantage over IP solution vendors using a server platform that cannot easily (if at all) provide alternate routing to PSTN in an all trunks busy condition.
- Bandwidth Control 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. 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.

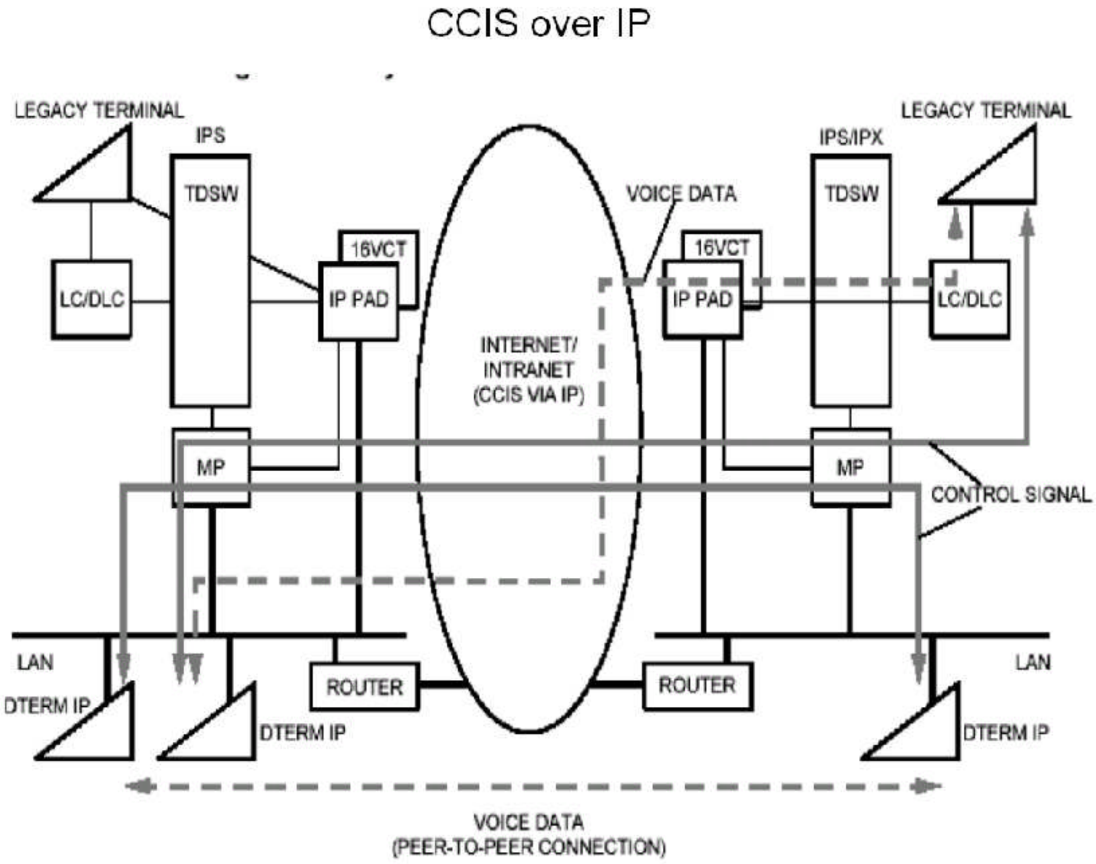
CCIS networking provides Transparent Voice Networking Over Wide Area Transport, i.e. not just within a LAN, but also via:

- o TCP-IP via Frame Relay/ATM/E1
- o E1/ISDN PRI/ISDN BRI
- o E&M TIE Lines/64K Leased Lines

Typically used for Campus Environments & Contiguous Properties providing direct connection via Copper Cable or Fiber Optic Cable.
Interoperability with all networking transport mediums reduces the cost of connecting remote offices/buildings based on geographical location of the remote site.

See the table in the appendix showing the channel cards and CCIS channel handler cards used for each type of network connection.

CCIS over IP supports connectivity between legacy terminals, IP terminals, and Legacy–to-IP terminals. The MP Card manages Control Signals in both types of connections.

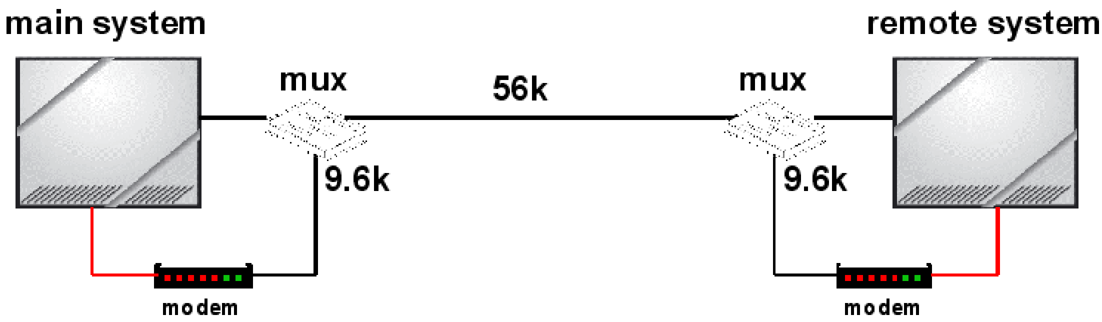


For Peer-to-Peer connection, the Voice Data is transmitted and received directly between Dterm IP via Intranet.
For Dterm IP-to-Legacy Terminal connection, the IP PAD card and VCT card are required to control and convert the Voice Data.

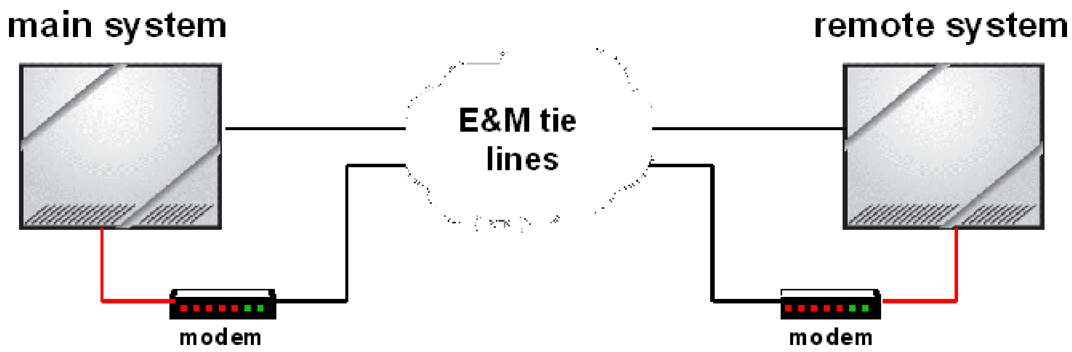
Customer's who are not ready for pure IP, can still utilize IP connections between sites by installing the IP Trunk Card (IPT) and take advantages of the benefits afforded by eliminating the support and maintenance of a separate voice infrastructure. Off course, peer-to-peer calls are not possible over such a network but only within each LAN domain.

5.2.1 CCIS over E&M LINES

Example with leased line:

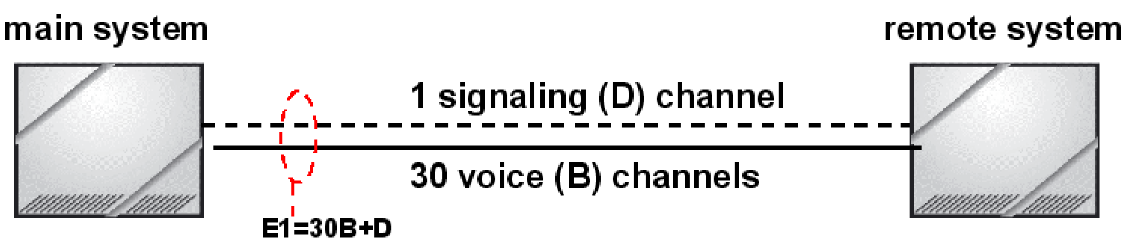


Example with E&M tie-line network:



PN-2ODTB - E&M Tie Line cards and SPN-SC00 CCH-D channel handler required per site connected. When building a point-to-point network, the maximum number of connections to remote locations is 8 per system

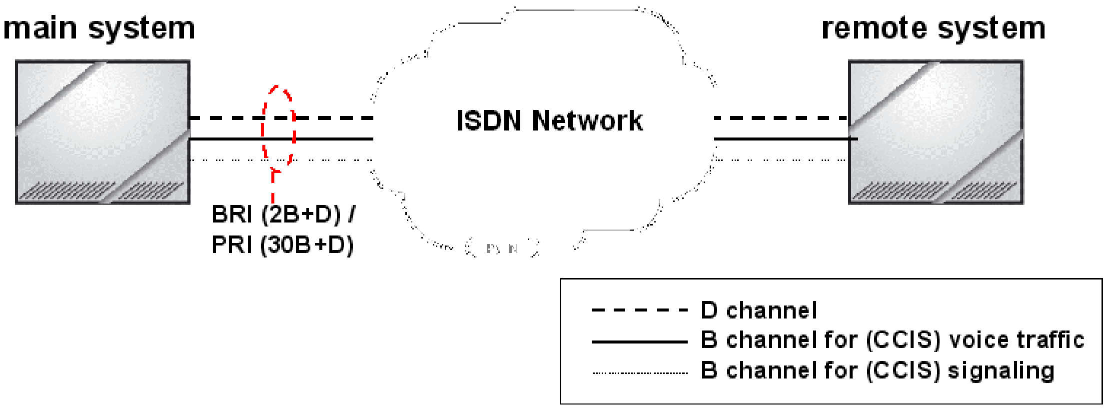
5.2.2 CCIS over Digital Trunk (E1) Interface



Uses the CCIS E1 card (SPN-30CCTA-A) with built-in CCIS Channel Handler:

When building a point-to-point E1 network, the maximum number of connections to remote locations is 8 per system.
See the specific manuals for the details.

5.2.3 CCIS over ISDN (event driven)



ISDN PRI (30PRTA) or BRI (4BRTA) cards and a CCIS channel handler (SPN-SC00 CCH-D) card are required per site connected. In the PRI card 29 voice channels remain, in the BRI card 3.
When building a point-to-point network, the maximum number of connections to remote locations is 8 per system.

5.2.4 CCIS over IP

This requires the same cards as for IP terminals, i.e. IP-PAD and VCT.

5.2.5 Overview of 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 – No 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.
Enhancements in R12.2:

- When an incoming trunk call is forwarded by Call Forwarding-No Answer over CCIS and the call forwarding destination is busy, that call can be returned to the forwarding station by system data programming.
- When an incoming trunk call is forwarded by Call Forwarding-No Answer over CCIS to an operator in the distant office, that call is appeared on the NANS key of the Desk Console.

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 peerto-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.

Rerouting at originating office - CCIS (R11)
In case of CCIS network, when trunks at the destination office are fully busy or failed, the destination office sends message to the origination, and the origination office carries out re-routing process

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 (R12.1)
This feature provides live recording of emergency calls, malicious calls, etc. within the CCIS network by centralized voicemail system.
Prior to R11, the 2000 IPS with voicemail system provides live recording feature in a standalone configuration only. In R12.1 software, this feature can be expanded within the CCIS network. The Dterm station (TDM Dterm, IP Dterm, and Dterm SP30) can record the conversation with another station/trunk within the CCIS network by the centralized voicemail.
When a tandem connection is required to reach voice mail only the originating and destination IPS offices need to have R12.1 software installed.

<u>Required Hardware</u>
- Active Voice voicemail system (???) supporting extended AAINFO message

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.

5.3 ISDN FEATURES

5.3.1 ISDN interfaces

Two standard ISDN interface cards for the 2000 IPS are available:

- The Digital Trunk Interface card with D-channel Handler provides a single 30B+D circuit for ISDN PRI.
- The Basic Rate Interface Trunk Card supports 4 two-channel PCM digital lines (2B+D) for ISDN BRI.

Required Hardware and Software

The following hardware is required for the support of the above functionality:

R8:
- SPN-30PRTA-B (AP)
- SPN-4BRTA-D (AP)

R10:
- SPN-30PRTA-C (AP)
- SPN-4BRTA-E (AP)

5.3.2 Overview of ISDN features

On these interfaces the following ISDN services will be supported (according to the ETSI specs):	From release
▪ Basic call	6.2
▪ Calling Line Identification Presentation (CLIP)	6.2
▪ Calling Line Identification Restriction (CLIR)	6.2
▪ Advice of charge end of call (AOC-E) for NL, DE, IT; AT, CH, DK; GR, LUX, PT, ES, SE	8; 9; 10
▪ AOC-E Output to PMS or SMDR Interface	12.2
▪ Overlap Receiving	6.2
▪ Overlap Sending	8
▪ Addressing (support of E.164 numbering plan as Type of Number / Numbering Plan Identifier combination)	8
▪ Connected Line Identification Presentation (COLP)	8
▪ Connected Line Identification Restrictions (COLR)	8
▪ B-channel negotiation	8
▪ Completion of Calls to Busy Subscriber (CCBS)	12.3

Addressing

This supplementary service supports E.164 numbering plan as TON (Type of Number) / NPI (Numbering Plan Identifier) combination in Calling and Called Party Number IE (Information Element) for outgoing calls. It adds trunk access -, national access - and international access code to the calling party numbers in the callers list

Advice of Charge at the End of the call (AOC-E) has become available for the following country specific implementations: Austria, Belgium, Denmark, Germany, Greece, Italy, Luxemburg, Netherlands, Portugal, Spain, Sweden and Switzerland. The other European countries must set to "Germany" because this country uses the standard implementation of the AOC service. The AOC information received will be displayed on the calling Dterm display phone, and be output to the SMDR port. (See table above for country release.)

AOC-E Output to PMS Interface (R12.2)
In R12.2 software, either ISDN AOC-E unit or AOC-E charge information can be output to the PMS interface (both information cannot be output at the same time). The output of either AOC-E unit or AOC-E charge information can be selected by system data programming.

AOC-E Output to SMDR Interface (R12.2)
Prior to R12.2, ISDN AOC-E "charge" information can be displayed on Dterm and output to an SMDR interface. In addition, in R12.2 software, ISDN AOC-E "unit" information can be output to the SMDR interface (both AOC-E unit and charge information can be output at the same time).

Required Hardware for AOC-E output or PMS or SMDR:

- ISDN BRI Trunk: SPN-4BRTA-E (AP) or later
- ISDN PRI Trunk: SPN-30PRTA-C (AP) or later / SPN-DTA (PRT)-A
- PMS/SMDR-IP: PZ-M606-A
- SMDR-RS232C (MP): No additional hardware is required
- SMDR-RS232C (AP00): SPN-AP00D MRC-A (AP)

B-channel negotiation allows an ISDN route to negotiate for an alternative B-channel in the case of call collision. B-channel negotiation is can be set per ISDN route by a route option.

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.

Calling Party Number (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.

Calling Party Number (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).

Completion of Calls to Busy Subscriber (CCBS), (R12.3)
In R12.3 software, the 2000 IPS will support ISDN – CCBS (Completion of Calls to Busy Subscriber) supplementary service based on the ETSI specifications (EN 300 359-1). This service feature is a call back feature over ISDN network. The feature allows a

calling party to set an automatic call back when a called party is busy. When the busy station becomes an idle, the station that sets the call back will be called. A maximum of 32 stations can access this feature simultaneously. Analog single line telephone, Dterm, Dterm IP station can be the calling and called party of the ISDN – CCBS feature.

Note: When the ISDN – CCBS feature is used, the ISDN network provider should support this function. Sometimes the ISDN provider requests additional payment to allow this function.

Connected Line Presentation (COLP)/Connected Line presentation Restriction (COLR)
The COLP supplementary service provides the IPS to send identification of the connected party in the IPS, to the calling party. Reference specifications: EN 300 094, ETS 300 096, EN 300 097

The COLR supplementary service provides the connected party in the IPS to prevent presentation of the connected party's ISDN number (and subaddress), to the calling party. Reference specifications: ETS 300 095, ETS 300 096, EN 300 098

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 SOPHO 2000 IPS connection or the SOPHO 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.

ISDN-to-ISDN Tandem Connection: BT on Busy External Party (R11)
Prior to R11, in the above case, calling party will hear ring back tone (RBT) first, and then later this becomes busy tone (BT). It appears if someone is rejecting your call. In R11 software, this problem is solved.
In case someone has set a call forwarding to an external number, if that person is called and the external number is busy, the calling party will now hear busy tone (BT).

Notification to caller when outgoing call seizes 2nd choice trunk for ISDN PRI/BRI (R9)
In case of an outgoing tie line call where 2nd choice trunk is ISDN trunks, when all tie line trunks are busy or in failure and the 2nd choice trunk (=ISDN trunk) is seized, the

caller will receive a special tone to notify the 2nd choice trunk is seized. This will provide the caller an option to stop making the expensive call. The notification tone will continue about 240ms to 320ms (fixed value).

- This feature is effective when the 2nd choice trunk route is ISDN PRI or BRI trunks.
- The notification tone is sent to the caller in case of making an normal ISDN call, when this feature is assigned.

Overlap sending

This service allows the IPS to send called party number digits to the network in successive messages

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.

5.4 OAI FEATURES

R12.1 OAI has been enhanced for PC-based attendant (SV60E) in CCIS. With these enhancements a PC-based attendant can provide improved call handling, also for CCIS calls.
In R12.2 software, the IPS provides several OAI enhancements to improve call handling by PC-based attendant.
Business ConneCT 2.2 and SV60E 3.4 will support the Camp-on-busy capability

Break-in over CCIS (R11)

Prior to R11, Break-in via OAI is available for a busy connection with an internal station only. In R12.1 software, break-in via OAI is available for a busy connection with a station in the CCIS network using SCF22. After the break-in via OAI, 3-party conference is established. Maximum number of simultaneous break-in connections per system is 16 including station monitor.

Camp-on Busy for operator over CCIS (R12.2)

Prior to R12.2, an OAI application can set camp-on to a busy station in the same PBX only (standalone configuration only). In R12.2 software, the OAI application can set camp-on to a busy station over a CCIS network, using SCF 19.

- When the called station has set Call Forwarding-All Calls and the forwarding destination is busy, the forwarding destination is camped-on.
- When the called station has set Call Forwarding-Busy Line and the forwarding destination is busy, the originally called station is camped-on.
- When a camp-on busy is set over CCIS network, an OAI application can receive status notifications for the station which the camp-on busy is set.

Reminder: only one caller can be camped-on at the same time.

Connected party status notification over CCIS/QSIG/ISDN (R11)

Prior to R11, when a station monitored by SMFR makes an outgoing trunk call (CCIS/QSIG/ISDN), the system provides only the outgoing trunk information in the system to OAI. In R12.1 software, the system can provide the connected party status information over CCIS/QSIG/ISDN to OAI using SMFN 12 and SMFR 139. Below information is provided via OAI:

- Connected party number
- Connected party status for CCIS/QSIG call (ring, answer, disconnect, outgoing call failed, connected party change, override set, camp-on set)
- Connected party status for ISDN call (ring, answer, disconnect)

Split Call Forwarding – All Calls/Busy/No Answer (R12.2)
In R12.2 software, an OAI application can set/cancel the Split Call Forwarding – All Calls/Busy/No Answer feature using the OAI facility SSFR 8, 9 and 10. This provides different handling of internal and external calls for Call Forwarding.

Status notification for camp-on busy (R12.1)
In R12.1 software, below status notification for camp-on busy feature can be provided via OAI.

- When camp-on busy is set, the status of the camp-on busy destination can be notified via OAI (SMFN 1-8).
- When the camp-on busy call is answered,
    - The status of the busy station can be notified via OAI (SMFN 2-8).
    - The status of the held party can be notified via OAI (SMFN 6-1).
- When the broker's call (shuttle) is activated by the busy station, the status of the party held during the broker's call can be notified via OAI (SMFN 6-1).

This enhancement is applicable for both camp-on methods: call waiting method and call transfer method.

System Speed Dialing (R12.1)
In R12.1 software, System Speed Dialing via OAI is available using the OAI facility SCF1 and SCF7.

5.4.1 Overview of OAI features

Back tone can be sent for monitored number (R9)
R9 enhancement allows OAI-AP to send below monitoring requests simultaneously by SMFR FN=124, instead of using SMFR FN=125 and SMFR FN=127. The monitored object is station or trunk.

- Make Call, Receive Call, Answer Call, Release Call and Hold Call
- 2nd Party Monitor: This allows the OAI-AP to send ringback tone to the calling party automatically when the call is terminated to the called party in the 2nd Party Monitoring.

Break-in over CCIS (R12.1)
Prior to R11, Break-in via OAI is available for a busy connection with an internal station only. In R12.1 software, break-in via OAI is available for a busy connection with a station in the CCIS network using SCF22. After the break-in via OAI, 3-party conference is established. Maximum number of simultaneous break-in connections per system is 16 including station monitor.

Camp-on-Busy recall can be queued and answered by OAI (R11)
For recall of OAI SCF10: Camp-on-Busy recall can be queued and answered by OAI application (e.g. PC-Attendant)

Called number, of rerouted DID call via CCIS, over OAI (R11)
In R11 software, 2000 IPS can generate a called party number of failed DID call (busy, no answer) to OAI application by SMFN. Called party can be an internal station or CCIS party.

Camp-on Busy for operator over CCIS (R12.2)
Prior to R12.2, an OAI application can set camp-on to a busy station in the same PBX only (standalone configuration only). With this feature PC-based attendants, like Business ConneCT 2.2 and SV60E 3.3, can provide improved call handling for CCIS calls.
In R12.2 software, the OAI application can set camp-on to a busy station over a CCIS network, using SCF 19.

- When the called station has set Call Forwarding-All Calls and the forwarding destination is busy, the forwarding destination is camped-on.
- When the called station has set Call Forwarding-Busy Line and the forwarding destination is busy, the originally called station is camped-on.
- When a camp-on busy is set over CCIS network, an OAI application can receive status notifications about the station for which the camp-on busy is set.

Reminder: only one caller can be camped-on at the same time.

Connected party status notification over CCIS/QSIG/ISDN (R12.1)
Prior to R11, when a station monitored by SMFR makes an outgoing trunk call (CCIS/QSIG/ISDN), the system provides only the outgoing trunk information in the system to OAI. In R12.1 software, the system can provide the connected party status information over CCIS/QSIG/ISDN to OAI using SMFN 12 and SMFR 139. Below information is provided via OAI:

- Connected party number
- Connected party status for CCIS/QSIG call (ring, answer, disconnect, outgoing call failed, connected party change, override set, camp-on set)
- Connected party status for ISDN call (ring, answer, disconnect)

Enhancements for PC-based attendant (SV60E) in CCIS (R12.1)
With these enhancements a PC-based attendant can provide improved call handling, also for CCIS calls.

Exclusive hold activation and notification by OAI (R9)
R9 enhancement allows an OAI-AP to receive exclusive hold notification and activate/deactivate exclusive hold by SCF 11 and 12.

- Exclusive notification: STS=2 (exclusive hold) is added to SMFN FN6 (hold notification).
- Activate/deactiveate exclusive hold: SCF FN=11(answer call) and FN=12(hold call) supports exclusive hold.

Forward notification (R9)
R9 enhancement provides forward notifications to OAI-AP when answering a call by Call Forwarding - All/Busy (SMFN=2 STS=5/6).

Note: When the incoming call by call forwarding is answered by Dterm Sub-line, the IPS sends SMFN FN=2 STS=1.

Forward notification across CCIS (R9)
R9 enhancement allows the IPS to send a distinctive notification for Call Forwarding-All/Busy/No Answer (SMFN FN=1 STS=4/5/6, SMFN FN=2, STS=5/6/7) to the OAI-AP, in such case. Also, calling party number and station number that sets the call forwarding are added to 2ndPartyAdd2 info of SMFN FN=1 STS=4/5/6 and SMFN FN2 STS=5/6/7.
Note: When the incoming CCIS call by call forwarding is answered by Dterm Sub-line, the IPS sends SMFN FN=2 STS=1.

Incoming/Answer notification enhancements (R9)
R9 enhancements provide an option to use below parameters in the Incoming/Answer notifications(SMFN1-0/2-0/2-1) when executing SCF6 and SCF8).

1. When an incoming call by SCF6 or SCF8 is received, 3rd Party Line1 of SMFN1-0 can be set to "a station/trunk connected to 2nd Party".
2. When an incoming call by SCF6 is answered, 3rd Party Line2 of SMFN2-0/1 can be set to "a station/trunk connected to 2nd Party".

OAI events in case of Call Back (Automatic Ring Back) calls (R11)
In R11 software, 2000 IPS can generate the PBX status information regarding call back-recall to OAI applications by SMFN (INCOMING, ANSWER, RELEASE).

OAI Fault registration (R11)
In R11 software, 2000 IPS can store fault message regarding OAI.

- Association between IPS and OAI application is established
- Association between IPS and OAI application is disconnected
- IPS detects fault status by health check

The fault message can be retrieved by MATWorX.

Retrieval of terminal type possible (R9)
R9 enhancement allows OAI-AP to retrieve line status with terminal type from the IPS(SMFR FN=1 MRID=4). The IPS sends back the below response to the OAI-AP. The monitoring object is station only.

- Terminal type: analog SLT, Dterm and wirelss PS(PHS/PCS)
- Line status notified: Idle, Busy, Lockout and Make BusyNote: For wiress PS, this feature is effective for L1(My line) only.

Split Call Forwarding – All Calls/Busy/No Answer (R12.2)
(Difference in handling internal and external calls for Call Forwarding)
In R12.2 software, an OAI application can set/cancel the Split Call Forwarding – All Calls/Busy/No Answer feature using the OAI facility SSFR 8, 9 and 10.

Status notification for camp-on busy (R12.1)
In R12.1 software, below status notification for camp-on busy feature can be provided via OAI.

- When camp-on busy is set, the status of the camp-on busy destination can be notified via OAI (SMFN 1-8).
- When the camp-on busy call is answered,

  - o The status of the busy station can be notified via OAI (SMFN 2-8).
  - o The status of the held party can be notified via OAI (SMFN 6-1).
- When the broker's call (shuttle) is activated by the busy station, the status of the party held during the broker's call can be notified via OAI (SMFN 6-1).

This enhancement is applicable for both camp-on methods: call waiting method and call transfer method.

System Speed Dialing (R12.1)
In R12.1 software, System Speed Dialing via OAI is available using the OAI facility SCF1 and SCF7.

5.5 Certification Program 2000 IPS

The Certification Program developed by the International Expert Training Centre (IETC) ensures that NSOs and partners can identify experts who know how to implement the product range to the best of their advantage.

The Certification Program is currently defined for the SOPHO 2000 IPS platform. In the future other products will be included.

What sort of Certifications exist?

CE-Certification: At the end of three weeks of training on the SOPHO 2000 IPS, the engineer has gained a lot of knowledge. In the months thereafter he or she will gain more knowledge and skills about the product by experience and self-study. When the engineer feels confident about this, he or she can apply for the CE Certification examination. A **Customer Engineer** who passes this exam is considered skilled enough to install and configure basic installations and to do regular maintenance and troubleshooting.

SE-Certification: After passing the Master Tech training, the **Support Engineer** should gain several months of on the job experience before applying for the SE-engineer certificate. This level of Certification confirms that the engineer is experienced enough to investigate and, if possible, solve complicated problems.

EUT-Certification: End User Trainers must be able to give product training to end-users. This Certification examination checks whether the candidates have enough product knowledge. The exam consists of a training course that the end-user trainer must deliver in front of fellow End User Trainers.

SU-Certification: Sales Support Engineers need to have sufficient knowledge about functionality, characteristics and boundaries. This examination will be more theoretical than practical.

TR-Certification: Trainers who will be giving training on a certain subject to people in their own organization, and to partners, need to ensure they have a high level of knowledge about the product. Trainers should also be able to pass on this knowledge in a didactical and acceptable way. This Certification will guarantee that training given by local trainers has the same level of quality as training given by IETC trainers.

TC-certificate: A local **Training Centre** must have a good training environment and enough equipment to enable TR-Certified trainers to deliver an optimal training session.

More information about this subject can be found in the Certification Program of the Customer Services Training Centre.

The information for the Customer Services Center training program can be found as follows (present on the NSONET pages Product Support, Training Centre): Certification program

6 Installation, Programming and Maintenance

6.1 Operating Environment

Operating Condition	Temperature Relative Humidity	
Normal Operations	5_{o}C - 30^{o}C 41^{o}F - 86^{o}F	15% - 65% Non-Condensing
Short Periods**	-0^{o}C - 40^{o}C 32^{o}F - 104^{o}F	15% - 90% Non-Condensing
During Storage and Transit	-18^{o}C - 50^{o}C 0_{o}F - 122^{o}F	8% - 90% Non-Condensing
**Not to exceed 72 consecutive hours or 15 days in a year.		

6.2 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 0.5 Vp-p.

CAUTION
Grounding circuit continuity is vital for safe operation of telecommunication equipment.
Never operate this equipment with the grounding conductor disconnected!

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.

6.3 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

6.4 Installation

The following items are required for correct operation.

1. Adequate space accommodation
2. Adequate ventilation
3. Commercial AC power

Main Equipment

The installation of the SOPHO 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-andjack 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, SOPHO 2000 IPS provides the following installation methods:

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

See also NSONET Product Support Pages, 2000 IPS. For details concerning installation refer to the installation manuals.

The 2000 IPS Systems delivered to the customer are prepared for installation in the following way:

- The CPU has been loaded with System SW and software keys (licences).
- The System has been loaded with standard scripting, based upon the configuration defined in Prophix.
- On start up therefore all boards will be “in service” and related functionality is prepared.
- Customisation to be completed by the NSO service engineer. (e.g. internal and external number plan, specific functionality requested by customer)

6.4.1 Language and Country version support

For the European market the tone plans and languages of the different countries are implemented and made selectable.
Language and country version support should be selected by entering the proper commands on the 2000 IPS / IPS DM.
In the system the following parameters need be set for the proper language and country version support:

- Nation code
- A/μ law designation
- Digital Tone Generator program selection, selects country tone plan:
  - Germany
  - Italy
  - Netherlands
  - Austria
  - Belgium
  - Spain
  - Sweden
  - UK
  - Denmark
  - Greece
  - Swiss
- Language selection for Dterm / Deskcon (English is default)
  - German
  - French
  - Dutch
  - Italian
  - Spanish
  - Portuguese
  - Swedish
  - Danish

For selection of the tones on the Dterm for the countries in Europe a new DTG program has been generated. The following two DTG files contain tones for each countries:

- SP3758 D1 issue 1.07: Netherlands, Germany, Italy, Austria, Belgium, Spain, Sweden and UK.
- SP3744 E1 issue 0.04: Denmark, Greece, Switzerland.

6.5 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 MAT WorXTM 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-CFTA/PN-CFTB/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)

6.5.1 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.

6.5.2 Maintenance Administration Terminal (MAT)

The Maintenance Administration Terminal (MAT) is a Personal Computer (PC) used for programming and maintenance of the system by the service engineer.
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.

The MAT can be connected to the system either directly or remotely. There are three ways to connect your PC to an SOPHO 2000 IPS:

- 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.

6.5.3 Management@Net

Management@Net is enhanced with a portal for the support of the 2000 IPS, allowing the customer a single point of access to all his/her management applications independent of the platform.
For the 2000 IPS the following applications will be accessible:

- Accounting Package based on MTS TABS web solution
- OpenWorX Manager
- DECT Manager iS3000 and DECT Manager 2000 IPS

See for more details the separate product description of Management@Net 3.0.

6.5.4 MATWORX

MATWorX is a Graphical User Interface (GUI) program that provides an efficient method for maintaining 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.

The MATWorX IPS version 7.0.0 offers many enhancements to make the configuration of Remote PIMs over the IP network simpler. Enhancements related to the Remote PIM over IP capability were one of the main focuses of this MATWorX IPS release. The primary focus of these enhancements were in the following areas:

- Display PBX Main Site and Remote Site information
- When connecting to Remote Sites provide a warning message
- Display accommodation states of extensions and trunks with site information
- Office data conversion for Remote PIM over IP system FP/AP number expansion (expand from00-31 to 00-63)
- When connecting to a Remote Site that operates in normal mode the MATWorX functions have been limited to avoid mistakes that could be made by the technician

6.5.4.1 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 SOPHO 2000 IPS. The Wizard asks you simple questions and then uses the information to automatically configure the connection for the PC and the PBX.

6.5.4.2 MACH Script Editor

This is a powerful timesaving tool that enables you to create a list of SOPHO 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.

<u>Service conditions:</u>

1. MATWorX can be used with any standard Windows PC. The PC used with MATWorX must have an RS-232C port, and cannot be located more than 15m from the system when connected on premises.
2. MATWorX can be connected to the system either directly or remotely. Direct connection is through the RS connector on the MP card. The MAT CA-T cable connects MATWorX to the RS connector. 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 (Central Office Trunk) 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 the MAT:
   - 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 MATWorX 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 assignment of related office data is possible.

6.5.4.3 MATWorX enhancements

Release 9:

- IP PAD Setting Add-In (New); new add-in to maintain IP-PAD information and adjust the quality of communication among IP devices.
- Graphical Configuration Report Add-In (Enhancement); Adds 8IPLA and 24IPLA to the existing graphical configuration report (GCR)
- Data Setting Add-In (Enhancement): Adds data setting add-in for 8IPLA/24IPLA.
- Ease of Operation (Improvement):
    - It is now possible to change the font size and window size of the MOC add-in and the Mach Script Editor.
    - The following have been added to MATWorX Scheduler. MATWorX Scheduler activates applications at specified times:
        - AP Program Download
        - LEN List-Up
        - Fault Information Display
        - Mach Script Editor
        - Office Data Save, Load and Verify

Required Software:

- MATWorX IPS Version 8

Release 10:

- New Remote Software Download add-in enables to activate the software download to MP and to program a mode for change over to the new software: immediately or scheduled.
- New Station Speed Dial Data add-in enables multiple station speed dial data to be viewed and edited on the same screen. The data can be imported from/exported to csv file.
- New Digit Conversion add-in enables all numbers to be viewed and edited on the same screen, spread sheet layout with cells and rows.
- Enhancement of the existing LCR add-in: all programmed area codes, maximum digits and next pattern can be viewed and edited on the same screen, spread sheet layout with cells and rows.

Release 11:

- Add-in enhancement for quick station data entry
  - Quick data entry of station number, trunk number and trunk route number. (Consecutive number entry at 1-click operation, etc.)
  - Easy setting of station data
    - Trunk Restriction Class (CM1201)
    - Service Restriction Class (CM1202, CM1207)
    - Tenant Number (CM1204) Dterm – Prime Line (CM93)
    - Station Name (CM77)
    - Dterm – Key Assignment (CM90)
- Support Windows 2003

  From R11 MATWorX, MATWorX can be run on Windows 2003 operating system. On the Other hand, R11 MATWorX does not support older OS, which can be used for R10 or earlier MATWorX.

Windows	95	98	98SE	Me	NT4	2000	2003	XP
R10 MATWorX	X	X	X	X	X	X	N/A	X
R11 MATWorX	N/A	N/A	N/A	X	N/A	X	X	X

- Support Wireless LAN terminal and SIP handler setting by Add-in

  This add-in provides GUI-based data setting related to in-skin SIP server and WLAN handset.
- Load licenses from HDD or USB Memory (Registration Wizard)

  Prior to R11 MATWorX (Registration Wizard), software licenses are loaded from a floppy disk Drive only. In R11 MATWorX (Registration Wizard), software licenses can be loaded from a hard disk drive or USB memory.
- Sending Ping command

  In R11 software, 2000 IPS can send a Ping command to an external IP device (e.g. OAI server) to know the device exists and the IP network is working. MATWorX sends a request for sending Ping command to the 2000 IPS, with a desired IP address. The 2000 IPS sends ICMP echo to the IP device with that IP address, and receives ICMP echo reply from the IP device. The 2000 IPS sends the result of Ping command to the MATWorX.
- Other Enhancements
  - Add shortcut icon of the MATWorX on the Desktop screen when the MATWorX is installed
  - “Welcome” dialog box is appeared when starting up the MATWorX to guide a MATWorX user to next actions.
  - Provide a new menu bar and toolbar configuration to search appropriate commands more easily (For existing MATWorX user, existing menu bar and toolbar configuration can be also selected by option menu).
  - Provide an option to choose a background color of dialog box, etc.

Release 12.1: MATWorX IPS Ver. 11.5.0

This enhancement provides a complete GUI-based data programming solution for the 2000 IPS system.

- Most NEC logo and texts removed

  In order to adapt the customer requirements, most of the NEC logo and texts appeared in the MATWorX screen are removed.
- Command script interface

Command script interface allows an external application to communicate with the 2000 IPS system via MATWorX based on the PBX script data. The external application is System Programming Assistant (SPA) which is developed by PBC. This application complements the MATWorX to provide GUI-based system data programming.

Release 12.2: MATWorX IPS Ver. 11.6.0
The list of MATWorX Add-Ins, which is enhanced to support the R12.2 related features:

- Service Restriction Add-in
- Station Assignment Add-in
- Station Data Copy / Move / Swap Add-in
- Digit Conversion Add-in
- Numbering Plan Add-in
- Traffic Measurement Add-in
- Office Data Save / Load / Verify Add-in
- Trunk Route Add-in
- Setting System Accommodation Data Add-in
- MOC Mode Add-in

MATWorX PC Requirements:

	Minimum Requirements	Recommended Configuration
CPU	Pentium II 350MHz	Pentium IV 1GHz
RAM	128MB	256MB
Hard Disk	230MB or more	
Monitor	SVGA (800 x 600) High Color (16 bit)	SVGA (1024 x 768) True Color (32 bit) 15-inch or larger monitor
Peripherals	CD-ROM drive, Keyboard, MouseRS-232C or LAN interface	
Operating System	Microsoft Windows Me, Windows 2000, Windows XP, Windows 2003 (Latest Service Pack (SP) should be applied)	

6.5.4.4 Traffic management

The SOPHO 2000 IPS provides traffic management reports to be used for overall analysis of system performance. MATWorX is used to request and display the type of report, sample measurement time period, and time increments of reports. There are over twenty types of reports.

6.5.4.5 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.

<u>Service Conditions</u>

1. The following additional equipment is required for this feature:

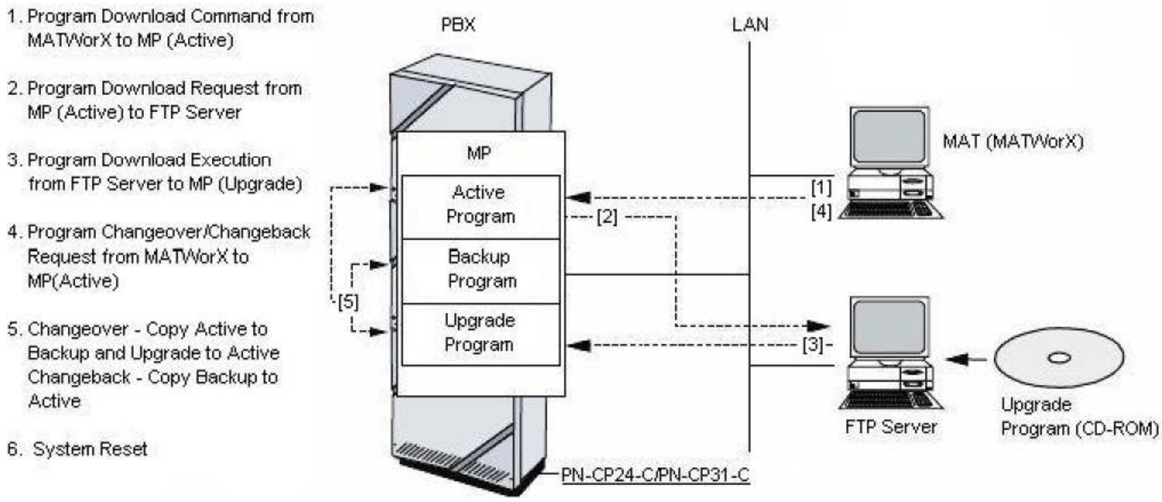
- A modem at the maintenance center and one at each remote site. (When the internal
- modem of the Main Processor (MP) is used, no modem at each remote site is required)
- A cable for connection between the MP and the on-site modem. (When the internal
- modem of the MP is used, the above cable is not required)

2. The internal modem of the MP is compatible with the following specifications:
   - ITU-T V.22 1200 bps
   - ITU-T V.22 bis 2400bps
   - ITU-T V.32, 4800/9600 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 controlledtransfer
   - 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

6.5.4.6 MP Program Download

The SOPHO 2000 IPS provides Online MP Program Download via IP network using the CP24 (IPS/PSDM/ IPS Remote PIM) and CP31 (IPS DML/ IPS DMR). This feature allows an MP upgrade program to be downloaded to the SOPHO 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

<u>Service Conditions</u>

General Service Conditions

- The MP program download can be executed to the PN-CP24/PN-CP31 (MP card) when a PZ-M606-A is mounted.
- 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.
- When the PN-32IPLA is mounted in the system, the MP program download (FTP) is not available. When the PN-32IPLA-A or PN-8IPLA is mounted, it is available in the system.

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.

<u>Required Hardware/Software:</u>

Main and Remote sites	CP24 (IPS/IPS DM /IPS Remote PIM) CP31 (IPS DML /IPS DMR)
Upgrade Program	Version 10.0.0
MATWorX IPS	
PC/Work station for FTP	Windows 2000 Server Windows 2000 Professional Windows XP Professional

Notes: MATWorX and FTP server pro gram can be used on the same PC.
Internet Information Server (IIS), supplied withWindows 2000/XP Professional can be used.

6.5.4.7 MP Program Download (FTP) for Remote Site (R12.2)

With R12.2, MP program downloading via FTP server is extended for Remote Site system (DMR) in the Remote PIM over IP network. This feature provides an easy and economical way of software updates for systems dispersed geographically from a remote maintenance center.

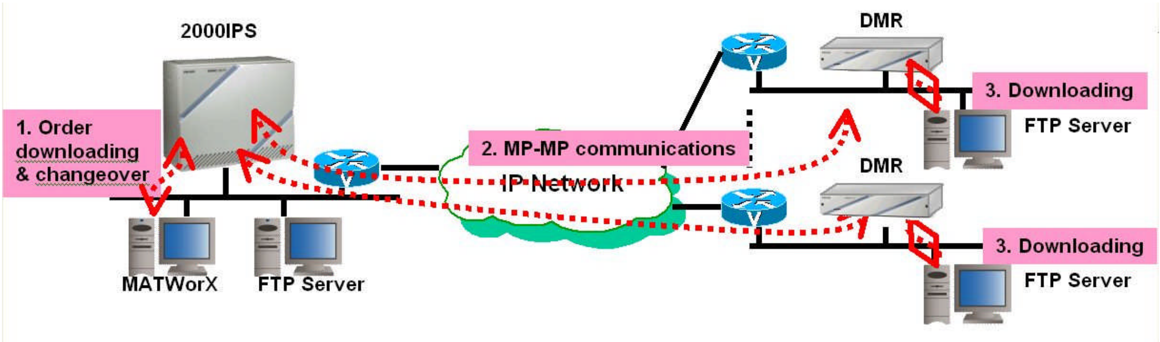
By command entry from the MATWorX in the main site, the MP software stored in the external FTP server is downloaded to the MP of the remote site PIM. The downloaded software stored in the MP is changed over to the existing working MP software. By system data programming, below operation patterns are available.

- Scheduled downloading & scheduled changeover
- Immediate downloading & immediate changeover
- Immediate downloading & scheduled changeover

If the changeover fails, fallback to the old software can be executed in below patterns.

- Automatic fall back (in case that system reset occurs 5 times in 3 minutes)
- Manual fall back (by command data entry from MATWorX)
- Scheduled fall back

The FTP server can be located in the main site or in each remote site.



6.5.5 Diagnostics

6.5.5.1 Log Data Collection for VoIP Calls (R10)

This feature allows speeding up the isolation of an error as it provides functions for analyzing VoIP related problems. Collects call logs and fault logs on VoIP calls. This enables the collected call log to be checked when the user reports a malfunction. Enable real time check when a malfunction exists at registration or accumulated checking of error information by a maintenance person. Call logs and Fault logs can be referenced from MATWorX and MAT/CAT mode.

Required Software and Hardware:

- SPN-8IPLA IP PAD-C or firmware upgrade required.

6.5.5.2 On-line MP-FP Command Output via RS-232 (R12.2)

Prior to R12.2, MP-FP traces can be retrieved during off-line mode. In R12.2, the MP-FP traces can be retrieved during on-line mode (SENCE switch=4), by entering commands (CMF6, Y=2). The traces data can be retrieved by MATWorX or can be output via RS-232C (Modem built into MP).

Requires MATWorX IPS Ver.11.6.0 or later version.

6.5.5.3 Self Diagnostic/System Messages

The SOPHO 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.

6.5.5.4 Station Service Status Display (R12.2)

By entering single command (CME4, YY=00, 01), a service engineer can see a list of service features/station status regarding to a specific station. This information can be displayed on a MATWorX or Dterm CAT.

6.5.5.5 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

6.5.6 System Programming Assistant

The System Programming Assistant (SPA) is an extension to MATWorX to improve the install time of a PBX with a customer.

The software runs on current laptops on Windows 2000 and higher with at least 256MB of memory, and can run connected to a 2000 IPS as well as disconnected from the 200 IPS. The software runs on a standalone PC without any connection to a specific server.

SPA will run as an extension to MATWorX. A toolbar button will be made available in MATWorX to start SPA. It will interact with the MACH Script Editor by running scripts and processing of the results. MATWorX will connect via the PAM (PBX Access Module) interface to the 2000 IPS.

Features:

Load Configuration	A service engineer can load a previously saved configuration. This configuration could also be imported from a (‘Hoorn’) script or downloaded from the 2000 IPS.
Save Configuration	A service engineer can save the configuration in the user interface to a file. The file will include any changes made to the configuration. The configuration could be imported from the ‘ISC’ script or uploaded to the 2000 IPS.
Import (ISC) Script	A service engineer can import a script that was generated by the ISC. The 2000 IPS commands will be read from the file, interpreted and saved in a 2000 IPS configuration. The 2000 IPS configuration can then be saved via the Save Configuration use case.
Generate/Export Script	A service engineer can generate scripts for exporting the changes made to the configuration or to export the whole configuration. This script is then saved in a file and can be uploaded later with the MACH Script Editor to the 2000 IPS. It is also possible to upload directly to a 2000 IPS via

	the Upload to 2000 IPS use case.
Download from 2000 IPS	A service engineer can choose to get the configuration from the 2000 IPS as an alternative to import the '(Hoorn)' script. All configuration items that are shown in the user interface and the all configuration items available in the '(Hoorn)' scripts will be read from the 2000 IPS. It is possible to download the complete configuration of the 2000 IPS as well as a specific (specified) part of the configuration.
Upload to 2000 IPS	A service engineer can upload the changes or the whole configuration to the 2000 IPS. This use case is an extension to the Generate/Export Script use case. Not only is the script generated but it is also uploaded to the 2000 IPS via MATWorX.

6.5.7 MA4000 Management system

MA4000 Management System is a web based, powerful voice server management configuration suite. MA4000 offers centralized management for the SOPHO 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

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:	10/1 00 MB	10/1 00 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 SOPHO 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)

6.5.7.1 MA4000 IPS Assistant

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.

6.6 Documentation

NEC Philips Unified Solutions offers a full complement of documents for the Sopho 2000 IPS product line. Technical documentation is available on Compact Disk (CD ROM) or on the WEB through NSOnet. This section lists all documents included on the Compact Disk (CD ROM):

Description	12NC	Intended reader
SOPHO 2000 IPS - Request for Proposal Reference Guide	3522 009 11241	Installation/maintenance technicians, Sales representatives
SOPHO 2000 IPS - General description	3522 009 11251	Installation/maintenance technicians, Sales representatives
SOPHO 2000 IPS - Configuration guide	3522 009 11261	Installation/maintenance technicians
SOPHO 2000 IPS - System manual	3522 009 11271	Installation/maintenance technicians
SOPHO 2000 IPS - Command manual	3522 009 11281	Installation/maintenance technicians
SOPHO 2000 IPS - Installation procedures manual	3522 009 11291	Installation/maintenance technicians
SOPHO 2000 IPS - Business/Hotel Feature spec.manual	3522 009 11301	Installation/maintenance technicians, Sales representatives
SOPHO 2000 IPS - Business/Hotel Feature progr.manual	3522 009 11311	Installation/maintenance technicians, Sales representatives
SOPHO 2000 IPS - CCIS Feature & specification manual	3522 009 11321	Installation/maintenance technicians, Sales representatives
SOPHO 2000 IPS - CCIS System manual	3522 009 11331	Installation/maintenance technicians
SOPHO 2000 IPS - ISDN System manual	3522 009 11341	Installation/maintenance technicians
SOPHO 2000 IPS - ISDN Feature & specification manual	3522 009 11351	Installation/maintenance technicians
SOPHO 2000 IPS - QSIG System manual	3522 009 11361	Installation/maintenance technicians
SOPHO 2000 IPS - OAI system manual	3522 009 11371	Installation/maintenance technicians
SOPHO 2000 IPS - Office data programming manual	3522 009 11381	Installation/maintenance technicians
SOPHO 2000 IPS - Maintenance manual	3522 009 11391	Installation/maintenance technicians
SOPHO 2000 IPS - Data Interface System manual	3522 009 11401	Installation/maintenance technicians
SOPHO 2000 IPS - DM Hardware Installation Guide	3522 009 11431	Installation/maintenance technicians

MATWorX installation guide	3522 009 11531	Installation/maintenance technicians
MATWorx user guide	9600 075 26001	End user
SOPHO Dterm U.Guide Anlg	9600 075 01001	End user
SOPHO Desk Cons. U.Guide	9600 075 02001	End user
SOPHO DeskConsole UG-DE	9600 075 11001	End user
SOPHO DeskConsole UG-IT	9600 075 12001	End user
SOPHO DeskConsole UG-FR	9600 075 13001	End user
SOPHO DeskConsole UG-ES	9600 075 14001	End user
SOPHO DeskConsole UG-NL	9600 075 15001	End user
SOPHO DeskConsole UG-SE	9600 075 16001	End user
SOPHO DeskConsole UG-DK	9600 075 17001	End user
SOPHO Dterm UG-INT	9600 075 00001	End user
SOPHO Dterm UG-DE	9600 075 03001	End user
SOPHO Dterm UG-IT	9600 075 04001	End user
SOPHO Dterm UG-FR	9600 075 05001	End user
SOPHO Dterm UG-ES	9600 075 06001	End user
SOPHO Dterm UG-NL	9600 075 07001	End user
SOPHO Dterm UG-SE	9600 075 08001	End user
SOPHO Dterm UG-DK	9600 075 09001	End user

All user guides can standard be ordered via Prophix. All other documentation can be ordered through the regular channels.

All service documentation is provided in English and contains the information provided by NEC for this product range, the documentation covers both APAC, USA and EMEA information. Where applicable NEC has noted that certain functionally is applicable to EMEA (marked PBC in older versions).

In all other cases readers are advised to take notice of the following:
Manuals provided by NEC contain information for the whole lifeline of the 2000 IPS, this includes downwards compatible system / products/ functions such as the 2000 IVS and IVS2 system. As well as functionality and products not offered for the EMEA market, such as:

- Wireless System (WCS): instead IP-DECT will be offered by PBC,
- Dterm PS II (cordless handset): instead Dect handsets of PBC port folio are used.
- Dterm Cordless Terminals: instead PBC handset portfolio is used
- all terminal ranges that proceed the Dterm Series I, such as the Dterm series E
- Dterm SP10 (Softphone), a newer version of the softphone, SP30, has been introduced for EMEA
- Dterm Extenders
- Business Attendant Console (BAS), instead SV60E will be offered with integration on the OpenWorX platform and database.
- External ACD-MIS from NEC
- QueWorx / CallcentreWorX: instead CC250- will be offered.
- NEAXMail solution (e.g. AD8): instead MyMail@Net 510 or 560 will be offered.

Note: AIMWorX and OpenWorX are available

All functionality offered on the 2000 IPS platform with applications is described in the product descriptions of the associated subject.

6.7 Training

6.7.1 End-user training

The information for the Customer Services Center end user training program can be found on the NSONET (NSONET pages: Product Support, Training Center):

Course overview - Overview of courses and calendar
End user training - Overview of end user training program

6.7.2 Technical training

The following engineer courses are available at the Customer Services Training Center in Hilversum:

1. For the 2000 IPS platform (Customer Engineer (CE) and System Engineer (SE) level):
   - CE – SOPHO 2000 IPS Basic (5 days)
   - CE – SOPHO 2000 IPS Routing (5 days)
   - CE – SOPHO 2000 IPS VoIP (5 days)
   - SE – SOPHO 2000 IPS MasterTech 1 (5 days)

2. For the 2000 IPS applications and IP-DECT:
   - CE – SOPHO 2000 IPS Applications (3 days)
   - CE – iSMobile (IP-DECT) (3 days)

3. ICT knowledge required for 2000 IPS:
   - CE – SOPHO Advanced ICT (5 days)
   - CE – SOPHO ICT Network Security (2 days)
   - CE – Voice over IP & Quality of Service (3 days)

The information for the Customer Services Center training program can be found on the NSONET (NSONET pages: Product Support, Training Center):

Training guide - Overview of all courses, content and pre-requisites for courses
Certification program - With the 2000 IPS platform the Customer Services Training Center introduces a certification program for engineers
Course overview - Overview of courses and calendar
End user training - Overview of end user training program

6.8 Warranty and repair

6.8.1 Warranty

Warranty on SOPHO 2000 IPS and IPS DM items is valid for 12 months.
Each SOPHO 2000 IPS / IPS DM is delivered with the following warranty:

- 2000 IPS / DM hardware:
  12 Months warranty, 3 years Maintenance Obligation

- Terminals Dterm (Analogue, Digital, IP) and DeskCon 753:
  12 Months warranty, option for 3 Years maintenance
- 2000 IPS SW and licenses:
  6 Months warranty on current and previous Release, 3 years Maintenance Obligation
- Local regulations and customer specific requirements may result in different conditions.

6.8.2 Spare parts

Overview of spare parts
For the spare parts on the SOPHO IPS DM refer to: section Spare Parts IPS DM
For the spare parts on the SOPHO Dterm terminals refer to: section Dterm Spare parts.

6.8.3 Repairs

Repairable and consumable items
For the SOPHO 2000 IPS / IPS DM a number of 12 NC's have been defined as repairables, repair of these materials is provided via the regular repair channel.

Consumable parts are not replaced. In case of failure the required part should be ordered at the Commercial Order desk, during the whole lifecycle from ERL to EOS date.

A current overview of the repairable /consumable items is given in Appendix 2. As this list is continuously growing with new 12 NC's introduced with the 2000 IPS platform, please refer for up to date information concerning repairables and consumables on NSOnet.

Repair procedure
Repairable parts can be sent for repair according to the applicable service conditions. The repair service is based upon replacement policy, the original defect part is not returned to the customer. The procedure is executed as Exchange in Advance (EiA).

Software / license replacement in case of MP failure
On replacement of a defective MP, also the required licenses should be made available, as the license keys are connected to the MP of the system.
The replacement procedure for MP repairs:

- The engineer executes emergency procedure on-site which allows system to run on licenses for 5 days
- The engineer send in the malfunction MP for repair
- Registration of the data concerning the defect MP
    - Serial number of the MP
    - System ID number
- Retrieve Active licenses
- ISL returns a correct working MP
- Delivery of new Software and licenses is executed in parallel.

6.8.4 Preventative Maintenance

Cooling Fan recommendation:
We recommend that the cooling Fan that is mounted in the right side of the PIMMD should be replaced every four years in order to ensure proper functioning of the cooling system. Via the Fault Reporting Scheduling Feature, the pre-determined replacement can be notified in advance.

With respect to replacement of the fan in the SOPHO IPS DM refer to the IPS DM installation manual for further instructions.

6.8.5 MTBF

Mean Time Between Failure and Mean Time to Repair figures can be found in the manual: "2000 IPS Request for Proposal Reference Guide".

6.9 Service information

6.9.1 Standard procedures

The procedures concerning maintenance and support are described in the PQS Customer guide. Please refer to this document for more details. Exceptions to the procedures in this guide or detailed information concerning maintenance and support for the 2000 IPS platform are supplied via the NSONET pages of Customer Services in Hilversum.

Please check the Product Support pages on NSONet for the most up-to-date service information about the 2000 IPS.

6.9.2 Technical Support

Technical Support on the SOPHO 2000 IPS and IPS DM is defined according to the PQS Customer guide.

Detailed on Technical Support from Customer Services can be found on the Product Support pages of the NSONET (Product Support, Support Area, 2000 IPS).

6.9.3 Maintenance

The entry point for Problem Reports is the PQS-Helpdesk.

The 3th Line Maintenance on the SOPHO 2000 IPS and IPS DM is provided by Customer Services Hilversum. Please note that the 4th line maintenance for the 2000 IPS / IPS DM is done by NEC, through Customer Services. The Maintenance and Support contract is agreed with the supplier.

For details on maintenance, remote maintenance, service requirements, scripting and scheduling of releases please refer to the Product Support pages of the NSONET (Product Support, Support Area, 2000 IPS).

7 Specifications

7.1 Processors

The SOPHO 2000 IPS, IPS DM, 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.

The SOPHO 2000 IPS is outfitted with a new SPN-CP24 CPU. Equipped with an AMD processor, this CPU offers greater flexibility by combining some of the functions that would otherwise require additional hardware. The new CPU and including it call processing software provide the latest standard features and TDM & IP switching in the SOPHO 2000 IPS.
The new PZ-M606, an optional daughter board, may be added to the CPU to provide OAI & TCP/IP connections and also enables peer-to-peer connectivity for SOPHO Dterm IP phones stations and peer-to-peer IP CCIS.

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	
		Flash ROM	RAM
MP (PN-CP24)	32-Bit (ElanSC520)	16MB	64MB (SDRAM)
MP (PN-CP27)	32-Bit (ElanSC520)	8MB	32MB (SDRAM)
MP (PN-CP31)	32-Bit (ElanSC520)	16MB	64MB (SDRAM)
FP (PN-CP15)	16-bit (25MHz)	===	768KB
AP (PN-AP00)	16-bit (8MHz)	512KB	512KB

Main Processor (MP)

Name Code	Remarks
PN-CP24	Main Processor Card for *SOPHO 2000 IPS* and IPS DM. One card is required per system.
PN-CP27	Main Processor Card for *SOPHO 2000 IPS* Dual MP System. One card is required per system.
PN-CP31	Main Processor Card for *SOPHO 2000 IPS DML*. One card is required per system. Main Processor Card for *SOPHO 2000 IPS DMR*. One card is required for each Remote Site.

Name Code	Remarks
PZ-M606A	Ethernet Control Card: • Mounted on MP card to accommodate the Ethernet and transmit/receive a signal of TCP/IP protocol. • 10 BASE-T/100 BASE-TX twisted pair cable is connected

Major specifications and functionality of the Sopho IPS MPs are shown below:

Item		PN-CP24 PN-CP27	PN-CP31
Central Processing		ElanSC520	
System Memory		CP24/CP31: Flash ROM (16MB), SDRAM (64MB) CP27: Flash ROM (8MB), SDRAM (32MB)	
Network Switching		1,024 × 1,024 Time Division Switch	
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 1)**	
Mini Jack		1 for External Music Source for Music on Hold**(Note 1)**	
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		CP27A (Dual MP) only	
MP Program Download (FTP)		Available with CP24/CP31 **(Note 2)**	
MAT Interface	Direct Connection	1 port	1 port
	Remote Connection w/Built-in MODEM	1 port **(Note 3)**	1 port **(Note 3)**
	LAN Connection	Available	Available
External Alarm Indication		MJ and MN	MJ only
DAT		2circuits (120 seconds per circuit)	Not Available
DK00		2 circuits (relay drive x1, external key scan x1)	Not Available
Application Key Program		In EPROM	In Flash ROM

Note 1: *In case of Dterm IP, the synthesized melody is provided built in to IP adapter.*
Note 2: *MP Program Download (FTP) is for Main and Remote sites (CP24 and CP31)*
Note 3: *No. 1 Port includes a built-in modem for a remote connection of the MAT.*

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 *SOPHO 2000 IPS*.

7.2 Power

7.2.1 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)

7.2.2 AC Power Consumption / Thermal Output (Maximum

DESCRIPTION	AC Power Consumption (KVA)		Thermal Output (BTU)	
	100V	200V	100V	200V
1-PIM	0.35	0.40	1,195	1,365
2-PIM	0.70	0.80	2,389	2,730
3-PIM	1.05	1.20	3,584	4,096
4-PIM	1.40	1.60	4,778	5,461
5-PIM	1.75	2.00	5,973	6,826
6-PIM	2.10	2.40	7,167	8,191
7-PIM	2.45	2.80	8,362	9,556
8-PIM	2.80	3.20	9,556	10,922

7.2.3 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

7.2.4 Operating Environment

DESCRIPTION	SPECIFICATIONS

Ambient Temperature	0 to 40
Relative Humidity	Max. 90% (non-condensing)

7.2.5 Electrical Characteristics (Central Office Trunk

DESCRIPTION	SPECIFICATIONS
Insulation Resistance	15 mega-ohms or more at 100Vdc
DC Resistance	On-hook conditions: 30 mega-ohms Off-hook conditions: 1,700 ohms
Impedance	On-hook conditions: 20 kilo-ohms (300 to 3,400Hz) 8 kilo-ohms (at 24Hz) Looped conditions: 600 ohms
Leak Current	0 mA at on-hook conditions

7.2.6 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) -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 Attenuation level: -40 dBm or less at 3.4 kHz and above -70 dBm or less at 50 kHz and above

7.3 System Capacity

7.3.1 Release overview of System Parameters and Specifications

With release R8 of the 2000 IPS software a maximum of 64 FP/AP can be deployed in the Remote PIM network. The maximum number of Remote Sites is therefore 15. The maximum number of ports in a Remote PIM network is 1020. While in R6.2 the preliminary support for remote PIM over IP the number of ports was only 512. By having 1020 ports, the 256 analog trunks and 256 IP PAD channels are not eating up the ports from the analogue, digital and IP station ports.

The number of stations in Release 9 software has been increased to a total of 980. The combination of Analog, Digital and IP can be used to achieve the 980-station count. An IP Remote PIM network would be required to reach 980 TDM stations, because a stand-alone system only supports hardware up to 512 TDM stations. In R9 each type of station has its own limit to the number of that type of station that can be programmed.

- R9 supports up to 1020 ports per system in both standalone and remote PIM configuration
- The maximum number of Analog and Digital stations has now been increased from 512 to a total of 980 in an IP Remote PIM network. The maximum in a stand-alone system remains 512.
- IP stations have now been increased from 448 to a total of 956 for both stand-alone and Remote PIM network.

7.3.2 IPS System Capacity

Item		Capacity Per PIM 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 ***(Single MP)***	12	24	36	48	60	72	84	96
	No. of Cards ***(Dual MP)***	11	23	35	47	59	71	83	95
AP Card	No. of Ports		Max. 256 ports per system						
	No. of Cards ***(Single MP)***	12	24						
	No. of Cards ***(Dual MP)***	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)		64		128		192		256	
Analog Single Line Telephone (Lines)	Standard	64	128	192	256	320	384	448	512
	Long ***(Single MP)***	48	96	144	192	240	288	336	384
	Long ***(Dual MP)***	44	92	140	188	236	284	332	380
Dterm (Lines)	Standard	64	128	192	256	320	384	448	512
	Long ***(Single MP)***	24	48	72	96	120	144	168	192
	Long ***(Dual MP)***	22	46	70	94	118	142	166	190
DtermIP/ INASET (PTP Connection)		952	892	828	764	700	636	572	508
PS		512							
Zone Transceiver (ZT)		16	32	48	64	80	96	112	128
ISDN Station		16	32	48	64	80	96	112	128
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 E&M ***(Single MP)***	24	48	72	96	120	144	168	192
	2W E&M ***(Dual MP)***	22	46	70	94	118	142	166	190
	4W E&M ***(Single MP)***	24	48	72	96	120	144	168	192
	4W E&M ***(Dual MP)***	22	46	70	94	118	142	166	190
CCIS Trunk (Peer to Peer Connection)		Max. 127							
DTI/CCIS Digital Link	1.5M	DTI: 10, CCIS: 8							
	2M	8							
ISDN	1.5M (PRT)					8			
	2BRT (card) ***(Single MP)***	12				24			
	2BRT (card) ***(Dual MP)***	11	23	24					
	4BRT (card)	6	12	18	24				
IP Trunk		1	2	3	4	5	6	7	8
PFT Connections		8	16	24	32	40	48	56	64
3-Party Conference		Max. 16 conference groups per system							
6-/10-Party Conference	6-Party	Max. 4 conference groups per system							
	10-Party	Max. 2 conference groups per system							
32-Party Conference		Max. 8 conference group per system							
Built-in Router		Max. 8 cards per PIM							
DTMF Sender		Max. 32 circuits per system							
DTMF Receiver		16		32					

IPS System Capacity (continued)

Item	Capacity Note 1
Attendant Consoles	Max. 8 sets per system
Attendant Terminal (Dterm ATT Position)	Max. 8 sets per system
SMDR Interface	Max. 2 Interface ports (RS232) Max 1 interface port (IP) per system
PMS Interface	Max. 1 Interface port (RS232) Max 1 interface port (IP) per system
ACD / MIS or OAI Interface	Max. 1 interface port per system
Remote PIM over IP (Number of PIM at Remote Site)	Up to 15 (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

Note 1: *System Capacity is for Main site only. For Total System Capacity see IP Remote Network System Capacity.*

Note that although the specifications show that the SOPHO 2000 IPS supports only 8 CCIS links, the flexibility of IP networking allows 255 addressable nodes, greatly enhancing the networking capability of the system.

Capacity comparison for releases 6, 8 and 9
The following table compares the capacities between R6.2, R8 and R9:

Capacities (numbers)	Software release					Comments
	R6.2		R8/**R9**			
LT ports	512		1020			
DTMF receivers	32	512 LT ports	32	512/**980**	1020 LT ports	
Muber of Attendant Consoles	8		8			
Digital/Analog stations	512		512/**980**			
ISDN stations	128		128			ISDN stations main site only
IP stations	448		448/**956**			
IP pad Channels	256		256			
analog trunks	256		256			
P2P CCIS trunks						Max 127 channels main site only
AP trunks		256			256	
Number AP channels	256		256			
AP/FP cards	32		64			
Physical FP cards	4	32 AP/FP	4		64 AP/FP	Main Site only
Built-in FP on CPU	15		30			
Virtual FP for IP stations	15		30			
Remote PIM's	10		15			

Note: *The bold figures show the increased capacity with R9, but 980 is the maximum number of TDM terminals operated in an IP Remote network. In a stand-alone system the maximum number of TDM terminals is 512.*

The chart above is an example of the expanded port capacity of the IPS in R9 and is intended to help explain the addition to the Virtual extension. The total number of Virtual extensions available is calculated using the equation above. Other LT ports (CO Trunks, DAT, Registers, etc.) which are used to calculate the total number of ports used in the system does not affect the total number of virtual extension. Refer to the example below.

Station Line Size Comparison including Virtual Station Expansion:

Capacities (numbers)	Software							
	R8 Stand Alone		R8 w/Remote PIM		R9 Stand Alone		R9 w/Remote PIM	
Total Terminals	512		1020		1020		1020	
Digital	512	512	512	512	512	980	980	980
Digital IP	448		448		956		956	
Virtual	768 [1]		768 [1]		1020 [2]		1020 [2]	

Notes:

1. *To Calculate the Number of Virtual stations use this formula: 768 – Number of Digital/ IP = Virtual*
2. *To Calculate the Number of Virtual stations use this formula: 1020 – Number of Digital/ IP = Virtual*

7.3.3 IPS DM/IPS DML System 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 system	
	No. of cards	7	14
Total number of lines (Analog Single Line Telephone + Dterm)		56	112
IP-PAD	No. of channel	32	64
Analog Single Line Telephone (Lines)	8LC	56	112
	Long Line	Not Available	
Dterm (Lines)	Standard	56	112
DtermIP/INASET (PTP Connection) **Note 2**		952/128	888/128
ISDN Station		10	20
Central Office Trunk (Lines)	Loop Start	56	112
	DID w/4DIT	28	56
	2W/4W E&M	14	28
CCIS Trunk (Peer to Peer Connection)		Max. 127	
DTI/CCIS Digital Link **Note 3**	1.5M	7	DTI: 10, CCIS: 8
	2M	7	8
ISDN	1.5M (PRT)	7	8
	4BRT (card)	5	10
IP Trunk		1	2
PFT Connections		4	8
3-Party Conference		Max. 16 conference groups per system	
6-/10-Party Conference	6-Party	Max. 4 conference groups per system	
	10-Party	Max. 2 conference groups per system	
32-Party Conference		7	Max. 8 conference groups per system

Note 1: *Each Modular Chassis has 8 Virtual LT Ports that can only be used when using 32 PAD channels with the 8IPLA w/24IPLA.*

Note 2: *SOPHO 2000 IPS DML only supports a maximum of 128 DtermIP/INASET.*

Note 3: *The total number of trunk line and DTI channel shall be 256 or less. (Each trunk line and DTI channel is required to assign the "Trunk Number" by system data programming and maximum number of system parameter for "Trunk Number" is 256.)*

IPS DM/IPS DML System Capacity (Continued)

Number of PHYSICAL Modular Chassis	**Per MC**	
	1	**2**
Built-in Router	Max. 5 cards per Modular Chassis	
DTMF Sender	Max. 32 circuits per system	
DTMF Receiver	16	32
SN716 Desk Console	8	
Attendant Terminal (Dterm ATT Position)	Max. 8 per system	
SMDR Interface	Max. 2 Interface ports (RS232) Max 1 interface port (IP) per system	
PMS Interface	Max. 1 Interface port (RS232) 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 PIM for Remote Sites) **Note 5**	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 **Note 5**	1024	

Note 4: *ACD / MIS and OAI are mutually exclusive.*
Note 5: *IPS DML only supports a maximum of 128 IP stations; IPS DML does not support built-in SMDR. The IPS DML is a Stand Alone Only solution; Remote PIM's off the IPS DML are not supported.*

7.4 IP Remote Network Capacity

Total System Capacity (Main plus Remote)

Item		Capacity
LT Ports		1020
AP Ports		256
Analog Single Line Tel. + Dterm		980
IP PAD (No. of Channel)		256
DtermIP/INASET (PTP Connection)		952
PS		512
Cell Station (CS) / Zone Transceiver (ZT)		128
ISDN Station		128
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/2M (PRT)	8
	2BRT (card)	24
	4BRT (card)	24
IP Trunk		8
PFT Connections		64
3-Party Conference		Max. 16 conference groups
6-/10-Party Conference	6-Party	Max. 4 conference groups
	10-Party	Max. 2 conference groups
32-Party Conference		Max. 8 conference groups
Built-in Router		1 per Site
DTMF Sender/Receiver		Max. 32 circuits
Attendant Consoles		8
Attendant Terminal (Dterm ATT Position)		Max. 8 sets
SMDR Interface		Max. 1 interface port
PMS Interface		Max. 1 interface port
ACD / MIS or OAI Interface		Max. 1 interface port
Remote PIM over IP		Up to 15 (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

7.4.1 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 network
	No. of cards	7	14
IP-PAD	No. of channel	32	64
Analog Single Line Telephone (Lines)	8LC	56	112
Dterm (Lines)	Standard	56	112
DtermIP/ INASET (Peer to Peer Connection) **Note 2**		128	
Central Office Trunk (Lines)	Loop Start	56	112
	DID w/4DIT	28	56
	2W/4W E&M	14	28
DTI	1.5M	7	10
ISDN	1.5M(PRT)	7	8
	4BRT (card)	5	10
PFT Connections		4	8

Note 1: *Each Modular Chassis has 8 Virtual LT Ports that can only be used when using 32 PAD channels with the 8IPLA w/24IPLA.*

Note 2: *Remote PIMs Support up to 2 Virtual PIMs for assignment of DtermIP/ INA SET only.*

7.4.2 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 network
	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)	8	
	4BRT (card)	6	12
PFT Connections		8	16

Note 1: *Remote PIMs Support up to 2 Virtual PIMs for assignment of DtermIP/ INA SET only.*

7.4.3 Payload sizes

7.4.3.1 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	------

7.4.3.2 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	---

7.4.3.3 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	–

7.5 IP Specifications

Item	Specifications	Remarks
Voice Encoding	G.729a G.723.1 (5.3 k/6.3 k) G.711	8 kbps CS-ACELP MP-MLQ/ACELP 64 kbps PCM
IP-PAD	32 channels per card Automatically seized per call	
FAX Communication Feature	FAX Relay Method (T.30) IP	PAD card is required. G3 FAX (up to 14.4 kbps) Super G3 Reciprocal: Not allowed
DTMF Signal	H.245	H.323 IPT/IP-PAD/DtermIP
Inter-office/Intra-office Signaling	H.245	DtermIP to DtermIP connection DtermIP to IP-PAD connection
	PROTIMS over IP	DtermIP to *UNIVERGE* NEAX 2000 IPS connection
	CCIS over IP	Point to Multipoint connection
	H.323	H.323 IPT card and IP-PAD card are required
Jitter Control	Dynamic Jitter Buffer	
QoS (Quality of Service)	• TOS, IP Precedence • DiffServ	
LAN Interface	10BASE-T/1 00BASE-TX	Auto Negotiation is available. 100BASE-TX is recommended.
Echo Canceller (IP-PAD)	G. 168	

IP Specifications (continued)

Item		Specifications	Remarks
Payload Size	DtermIP/CCIS Virtual IPT	10 ms-40 ms (G.723.1: 30 ms unit)	Maximum voice channels per card 10 ms: 12 ch 20 ms: 20 ch 30 ms: 30 ch 40 ms: 32 ch
	H.323 IPT	20 ms.-40 ms. (10 ms. increments) (G.723.1: 30 ms. Fixed)	Maximum voice channels per card G.729a G.711 G.723.1 20 ms.: 6ch 5ch - 30 ms.: 8ch 7ch 8ch 40 ms.: 12ch 10ch -
PAD Control		0 dB to +16 dB (+2 dB unit) 0 dB to –16 dB (–2 dB unit)	Setting is available per Location No.
		0 dB to –16 dB	For connection via the IPT card

7.6 Trunk interfaces

7.6.1 E&M TIE LINES

The SOPHO 2000 IPS supports both 2 and 4-wire, Type I and Type V, E&M Tine Line connection to other SOPHO IPS systems. An ODT (Outbound Dialing Trunk) card is required for 2 or 4-wire E&M Tie Line interface. When using a 2-wire application, one ODT card supports two Tie Lines. When using a 4-wire application, one ODT card supports one Tie Line. The SOPHO 2000 IPS supports a maximum of 256 trunks.

7.6.2 Analogue Trunk interface, Transmission Plan (R9)

R9 enhancement provides the analog trunk transmission plan for Europe, based on the PBC requirements (Europe and South Africa).
The 8 Circuit Analogue Trunk Card provides 8 Loop Start trunks with Loop detection.
This R9 enhancement supports analog trunk transmission plans using (PN-8COTU card) for:

- Austria
- Belgium
- Denmark
- Germany
- Italy
- Spain
- Sweden
- Switzerland
- The Netherlands
- UK
- South Africa

Refer to below table for specifications implemented in the 2000 IPS

Country	Itfc	Zac	Zb	Li(dBr)	Lo(dBr)	Comp.(dBr)	COTCard
Austria	K2	270 Ohm+750 Ohm//150nF TBR38	270 Ohm+750 Ohm//150nF TBR38	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.5	8COTU
Belgium	K2	270 Ohm+750 Ohm//150nF TBR38	270 Ohm+750 Ohm//150nF TBR38	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.5	8COTU
Denmark	K2	270 Ohm+750 Ohm//150nF TBR38	270 Ohm+750 Ohm//150nF TBR38	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.5	8COTU
Germany	K2	270 Ohm+750 Ohm//150nF TBR38	270 Ohm+750 Ohm//150nF TBR38	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.5	8COTU
Italy	K2	600 Ohm	400 Ohm+700 Ohm//200nF	l: -6.0 s: -3.5	l: -1.0 s: -3.5	0.0	8COTU
Spain	K2	220 Ohm+820 Ohm//120nF	220 Ohm+820 Ohm//120nF	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.8	8COTU
Sweden	K2	270 Ohm+750 Ohm//150nF TBR38	270 Ohm+750 Ohm//150nF TBR38	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.5	8COTU
Switzerland	K2	270 Ohm+750 Ohm//150nF TBR38	270 Ohm+750 Ohm//150nF TBR38	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.5	8COTU
The Netherlands	K2	270 Ohm+750 Ohm//150nF TBR38	270 Ohm+750 Ohm//150nF TBR38	l: -6.0 s: -3.5	l: -1.0 s: -3.5	1.5	8COTU
UK	K2	270 Ohm+750 Ohm//150nF	270 Ohm+750 Ohm//310nF	l: -6.0 s: -3.5	l: -1.0 s: -3.5		8COTU
South Africa	K2	220 Ohm+820 Ohm//115nF	220 Ohm+820 Ohm//115nF	l: -6.0	l: -1.0	1.8	8COTU
International [1]	K2	600 Ohm	600 Ohm	l: -3.0 s: 0.0	l: +3.0 s: 0.0	0.0	8COTR
Brazil [1]	K2	900 Ohm	800 Ohm//50nF	l: -6.0 s: -4.0	l: -1.0 s: -3.0	1.8	8COTR
China [1]	K2	200 Ohm+680 Ohm//100nF	200 Ohm+680 Ohm//100nF	l: +3.5 s: +3.5	l: -3.5 s: -3.5	1.3	8COTR-A

Notes

1. Specifications for International, Brazil and China are existing implementations for the 2000 IPS

Required Hardware:

- PN-8COTU (Europe and South Africa)
- PN-8COTR (Brazil and APAC except for China)
- PN-8COTR-A (China

7.6.3 SIP Trunk with Toplink, Germany, Basic calls (R11)

In R11 software, 2000 IPS supports SIP-based connections with a German Provider, Toplink, for cost saving of customer's telephone expenses.
The IPS supports the below functions:

- DDO calls to toplink network
- DDI calls from toplink network
- Calling line identifications – Presentations (CLIP)
- Calling party number display on Dterm
- Session Timer - Send/receive keep alive signals
- Receive fragment packets (max. 3000bytes)
- Fault message registration
  - IPS reboot (SIP trunk, MP)
  - Link disconnect
  - Session timer - timeout
- Alternating routing to PSTN
  - When SIP trunk or SIP network is failed, 2000 IPS can reroute the outgoing call to PSTN.

- Add/delete E.164 “+” sign
    - Add “+” sign in front of telephone number (outgoing call)
    - Delete “+” sign in front of telephone number (incoming call)

Below table shows a summary of basic specifications of the SIP trunk interface.

Description	Specifications
Main Card	PN-8IPTA (8ch)
Sub Card	PZ-24IPLA (24ch)
The maximum channel of SIP in IPS system	(Main+Sub)<=64ch per system
Network Interface	10BASE-T/100BASE-TX, Auto Negotiation, EtherPort/Main PKG
Codec	G.711, G.729a
Payload size	20,30,40ms
Port Number	SIP: 5060, RTP: 10000-10320
Jitter Buffer size	Max. 300ms
Silence suppression	Not Supported
Echo Canceller	G.168 (64ms)
FAX	Pass-through (G.711)
DTMF	Pass-through (G.711)

Required Software and Hardware:
- CCIS license
- IPT license
- SPN-8SIP TRK-C (AP)
- PZ-24IPLA

7.7 SOPHO 2000 IPS DM, IPS DML, IPS DMR System Specifications

Item	**Specifications**
System Capacity	LT ports: Max. 56 ports / Modular Chassis, - (Max. 64 ports including 8 virtual LT ports/Modular Chassis) - Max. 112 ports / system (IPS DM/IPS DM^{L}/IPS DMR) - AP ports: Max. 256 ports / system - IP ports: Max. 952 ports (IPS DM) - Max. 128 ports / system (IPS DM^{L}/IPS DMR) - Card slots: 8 slots / Modular Chassis (including 1 slot for MP card)
Circuit Card Mounted in Modular Chassis	All LT/AP cards of the *SOPHO 2000 IPS* can be used for the IPS DM/ IPS DM^{L} with the exception of 4LLCB, 4ODTA and 4CSIA cards.
Power	AC100V – 240V (automatically adjusted)
Installation Method	Desk top-setting, 19” rack-mounting

Conditions	Temperature: 5°C – 40°C (when the system is operating) 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

7.8 Line Conditions

Description		Specifications
Loop Resistance (including Telephone Set)		
Analog Standard Line Analog Long Line		Max. 600 ohms Max. 2,500 ohms (DP 10pps), Max. 1,700 ohms (DP 20pps) 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 (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 (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

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.*

7.9 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.*

7.10DRS (Device Registration Server)

Features of Built-in DRS		Description	Remarks
Max Registration Terminal		MP	952 with CP24/CP27 128 with CP31
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	

7.11IP standards conformity.

The following table summarizes the different IP standards used with the 2000 IPS

Item	Specification
IP version	IP v.6
Voice encoding	G.729a G.723.1 (5.3kbps/6.3kbps) G.711
DTMF signal	H.245
Inter-office/intra-office signaling	H.245 Protims over IP CCIS over IP H.323
Jitter control	Dynamic Jitter Buffer
Quality of Service	TOS, IP precedence, DiffServ
LAN Interface	10BASE-T / 100BASE-T
Echo Canceller (on IP PAD)	G.168
FAX protocol	T.30; Path through (G.711,G.726)

7.12 IP PAD compatibility table

In R8 an 8-channel IP-PAD card is released with a 24-channel daughter board. The table shows the compatibility of the new IP-PAD card.

	Compression rate	Fax over IP (FoIP)	Modem over IP (MoIP)	Minimum SW Release
SPN-32IPLA IP PAD-C	G.711	T.30 or G.711(64k)	N/A	R.6.2
SPN-16VCTA IP-PAD-A (optional card for 32 IPLAA IP PAD-C)	G.723/G.729	T.30 or G.711(64k)	N/A	R.6.2
SPN-8IPLA IP PAD-A	G.711/G.723/G.729	Path-through or G.711(64k)	Path-through	R.8
PZ-24IPLA (daughter board for 8IPLA IP PAD-A)	G.711/G.723/G.729	Path-through or G.711(64k)	Path-through	R.8

7.13 Language support

The SOPHO 2000 IPS / DM supports the following languages in SW R6.2:

- English (default)
- German
- French
- Dutch
- Italian
- Spanish
- Portuguese
- Swedish
- Danish

Language support in SW R6.2 is based upon one supported language per system. Multi-lingual support in the system is a future roadmap item.
See Appendix A. Language and country version support

7.14 Security

7.14.1 Dterm IP security

When the Dterm IP is registered in the SOPHO 2000 IPS, the Login Code (DNR) and Password entered from the terminal can be encrypted. The Login Code uses a proprietary algorithm, but the Password can use a proprietary algorithm or the MD5 Algorithm. MD5 is an algorithm defined in RFC 1321 from the IETF. The Encryption Algorithm can be assigned on a system-wide basis by system programming. The encryption is available in both Login Method (with password protected) and Automatic Login Method (MAC Address Authentication).

7.14.2 Platform Security

The SOPHO 2000 IPS system can be programmed such that access to program changes is controlled by Passwords. Up to eight Password levels are available. This feature, when activated, prevents unwanted changes by unauthorized personnel.
Apart from that the system has a proprietary Operating system which is therefore not subject to viruses or hacking.

8 System performance

8.1 Transmission Characteristics

Transmission Characteristics	
Cross Talk Attenuation	More than 70 dB at 1000 Hz
Idle Circuit Noise	Less than -65 dBm
Insertion Loss (relative to 1KHz-10 dBm)	Station-to-Station - Typical 6 dB Station-to-Trunk - Typical 0.5 dB Trunk-to-Trunk - Typical 0.5 dB at 0 dB PAD control
Longitudinal Balance Trunk Side	Better than 58 dB
PCM Characteristics	EMEA: Line Rate 2 Mbps, A-Law USA: Line Rate 1.544 Mbps, µ-Law, TI-04 Standards
Return Loss	More than 15 dB (300 ~ 3,400 Hz)
Line Impedance	Station: 600 Ω Trunk: 600 or 900 Ω
Leakage Resistance	More than 20,000 Ω

8.2 Tone Plans

Within the 2000 IPS SW R6.2 the tone plans for the European markets have been defined.

8.2.1 Rotary Dial Pulse and DTMF Signaling

(1) Rotary Dial Signal

Description	Specifications	
	Receiving	Sending
Dial Speed	9 to 22 pps	10 pps +/- 0.8pps 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) - 8 dBm (high group)			

Unwanted Frequency Components	Not Applicable	40 dB below the power of signal frequency

8.2.2 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

8.2.3 Audible Tones

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 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 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

8.2.4 Tone changes for Denmark (R12.2)

The following tone specifications are implemented:

Tone Name	Frequencies	Cadence
Number Unobtainable Tone	425Hz, -3dBm	250ms-on, 250ms-off, Repeating
Park Tone	425Hz, -3dBm	250ms-on, 250ms-off, Repeating

8.2.5 Tone changes for South Africa (R12.2)

In R12.2 software, the system will provide support of South African tone plan, but this has to be tested first. The following table shows the tone specifications for South Africa:

Tone Name	Frequencies	Cadence
Internal Ring Tone	33.33Hz x 400Hz, -3dBm (suppressed carrier)	1s-on, 2s-off, Repeating
External Ring Tone	33.33Hz x 400Hz, -3dBm (suppressed carrier)	1s-on, 2s-off, Repeating
Internal Dial Tone	400Hz, -9dBm	Continuous
Enquiry Dial Tone	400Hz, -9dBm	Continuous
First External Dial Tone	33.33Hz x 400Hz, -3dBm (suppressed carrier)	Continuous
Busy Tone	400Hz, -9dBm	500ms-on, 500ms-off, Repeating
Number Unobtainable Tone	400Hz, -9dBm	2.5s-on, 500ms-off, Repeating
Congestion Tone	400Hz, -9dBm	250ms-on, 250ms-off, Repeating
Diversion Active Dial Tone	33.33Hz x 400Hz, -3dBm (suppressed carrier)	1s-on, 200ms-off, Repeating
Confirmation Tone	400Hz, -9dBm	600ms-on/200ms-off Repeating (max.32s
Call Waiting Tone Burst	400Hz, -9dBm	256ms-on, off
Camp-on Busy Tone to Caller	33.33Hz x 400Hz, -3dBm (suppressed carrier)	400ms-on, 200ms-off, Repeating
Hold Tone	33.33Hz x 400Hz, -3dBm (suppressed carrier)	400ms-on, 200ms-off, Repeating
Park Tone	400Hz, -9dBm	2.5s-on, 500ms-off, Repeating
Routing Tone	425Hz, -6dBm	7 times 75ms-on/75ms-off, followed by 1 time 75ms-on, 875ms-off, Repeating
Break-in Ticker	425Hz, -3dBm	5 times 32ms-on/480ms-off, Repeating every 4s
Add-on Ticker	425Hz, -3dBm	4 times 32ms-on/480ms-off
Howler Tone	1800Hz, -13dBm	250ms-on, 250ms-off Repeating, overlaying cadence of 32s

Note: The IPS does not support the following tones:

- Second External Dial Tone
- Delayed Hotline Dial Tone
- Paging Absent Tone
- Paging In-Progress Tone
- Bypass Tone
- Through Connect Ring Tone
- ARB Destination Ring Back Tone
- Password Tone
- Call Waiting Ticker
- Dialup Automatic Break-in Ticker

8.2.6 Ringing Signal

DESCRIPTION	SPECIFICATIONS
Frequency	20 Hz (Nominal)
Voltage	75 Vrms (Nominal)
Interruption:	2 sec. ON, 4 sec. OFF (for external call) 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.*

8.3 Built-In Modem on MP Card

DESCRIPTION	SPECIFICATIONS
MODEM	33.6 kbps

8.4 Dimensions and Weight

8.4.1 SOPHO 2000 IPS

Main Equipment		
	Dimensions (W x D x H)	**Weight**
PIM (Fully card-mounted)	Approx. 430 x 223 x 353 mm3 (16.9" x 8.8" x 13.9")	Approx. 11.5 kg (25.35 lbs)
BASE	Approx. 430 x 205.2 x 61.6 mm3 (16.9" x 8.08" x 2.43")	Approx. 3.0 kg (6.61 lbs)
BASE TRAY (for UL)	Approx. 435 x 224.6 x 66.2 mm3 (17.1" x 8.84 x 2.6")	Approx. 1.7 kg (3.75 lbs)

8.4.2 SOPHO IPS DM/IPS DML/IPS DMR

Main Equipment	Dimensions (W x D x H)	Weight
SOPHO IPS DM *SOPHO IPS* DML *SOPHO IPS* DMR	430 x 365 x 88 mm3	Approximately 7 kg / Modular Chassis (when all slots are occupied)

8.5 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

8.6 Maximum feature capacities

Application	**2000 IPS**
Speed dial	10,000
Fault Message Types	64
Station Numbering	8 Digit
Forced Account Code	3,000
Call Forward Outside	496
Message Reminder	1024
DID Conversion	1,000

Appendices

Appendix A: Repairables and Consumables

The following overview presents which items obtained from NEC are defined as repairable or consumable. Remaining items on the List of Deliverables are supplied by PBC and therefore follow the same definition as used for the iS3000 List of deliverables.

12nc	Description	Type	MTBF	Failure Rate
960051150230	24 Port Patch Panel	Consumable		
960051151002	BATT CA INT	Consumable		
960051151003	BATT CA EXT	Consumable		
960051151004	RS NORM-4S CA-A	Consumable		
960051151008	PWR CA-A	Consumable		
960051151021	RS RVS-4S CA-C	Consumable		
960051151031	PWR CNT CA-D	Consumable		
960051151032	PWR CNT CA-E	Consumable		
960051151039	BATT CA-P5	Consumable		
960051151043	AC CORD-E-E	Consumable		
960051151060	AC CORD-D-EU	Consumable		
960051151061	AC CORD-B-A	Consumable		
960051151306	109P0624H7D09 FAN	Consumable	346	
960051151348	COVER PARTS ASEM	Consumable		
960051151369	19 INCH RACK BRACKET (A)	Consumable		
960051151372	MOUNTING BRACKET	Consumable		
960051151373	HANGER ASSEM	Consumable		
960051151374	ICS VS I/F BRACKET ASSEM	Consumable		
960051151375	BASE TRAY ASSEM	Consumable		
960051151380	RACK MOUNT KIT (J)	Consumable		
960051151381	JOINT BRACKET KIT (J)	Consumable		
960051200285	SN536 DCHST A-A	Consumable		0.18%
960053780022	DTR-1-1P (BK) TEL	Consumable	84	0.10%
960053780027	DTR-1HM-1P (BK) TEL	Consumable	42	0.15%
960053780033	DTR-2DT-1P (BK) TEL	Consumable	12	0.15%
960053780111	AD(A)-RP Unit	Consumable	40	1.25%
960053780112	AP(R)-RP Unit	Consumable	17	1.25%
960053780113	AP(A)-RP Unit	Consumable	18	1.25%
960053780116	WM-RP Unit	Consumable		
960053780492	DESI DCR-60-1P(PKG 25)	Consumable		
960053780493	DESI DTR-16D-1P(PKG 25)	Consumable		
960053780494	DESI DTR-8-1P(PKG 25)	Consumable		
960053780495	DESI DTR-8D-1P(PKG 25)	Consumable		
960053780496	DESI DTR-1-1P(PKG 25)	Consumable		
960053780497	DESI DTR-1HM-1P(PKG 25)	Consumable		
960053780498	DESI DTR-2DT-1P(PKG 25)	Consumable		
960053780499	DESI DTR-32D-1P(PKG 25)	Consumable		
960053780500	HANDSET W/O CORD-P (BK)	Consumable		
960053770502	LINE CORD 2.1M-P (BK)	Consumable		
960053780505	HANDSET HANGER KIT-P(BK)(25)	Consumable		
960053780530	HANDSET CORD 3.6M-P(BK)	Consumable		
960053780535	HANDSET CORD 7.2M-P(BK)	Consumable		
960051151007	RS PRT-15S CA-A	Consumable		

960051151013	48-TW-0.7 CONN CA	Consumable		
960051151029	MAT CA-T	Consumable		
960051151038	BUS-0.4 CA-PA	Consumable		
960051151371	19 INCH RACK BRACKET (B)	Consumable		
960051151494	PZ-PW131	Consumable	9	
960051150115	PN-4DATC	Repairable	21	0.79%
960051150125	SPN-CFTC (AP)	Repairable	20	1.38%
960051150228	PN-M10	Repairable	65	0.00%
960051151229	SPN-AP00B MRC-C (AP)	Repairable	22	2.29%
960051151233	SPN-SC03B 8ICH (AP)	Repairable	17	1.87%
960051151243	SPN-CP32 FP (PBC)	Repairable	37	0.18%
960051151282	SPN-SC00 CCH-D(AP)	Repairable	13	0.52%
960051151293	SPN-4BRTA-C (AP)	Repairable	17	0.00%
960051151449	SN753 DESK CON-A	Repairable	14	2.58%
960051151486	PZ-M537	Repairable	33	0.08%
960051150119	PZ-8PFTB	Repairable	79	0.00%
960051151121	ICS VS BASE (PBC)	Repairable		0.00%
960051151203	PN-DK00	Repairable	90	0.60%
960051151487	PZ-PW122	Repairable	20	0.18%
960051151489	PZ-M542	Repairable	78	0.08%
960051151493	PZ-4PFTA	Repairable	152	
960051151497	PZ-PW126	Repairable	8	0.18%
960053780083	DTR-8-1P (BK) TEL	Repairable	14	1.50%
960053780085	DTR-8D-1P (BK) TEL	Repairable	11	1.50%
960053780087	DTR-16D-1P (BK) TEL	Repairable	10	1.50%
960053780095	DCR-60-1P (BK) Console	Repairable	12	1.00%
960053780114	CT(A)-RP Unit	Repairable	14	1.25%
960053780115	IP-RP Unit	Repairable	9	1.25%
960051150006	ICS VS PIMMG (PBC)	Repairable	61	0.05%
960051150114	PN-8LCAA	Repairable	59	0.38%
960051150120	PN-CFTB	Repairable	121	0.64%
960051150137	PN-4LCAA	Repairable	26	0.55%
960051150208	SPN-2ILCA	Repairable	20	0.37%
960051150223	PN-8DLCP	Repairable	62	0.11%
960051150224	PN-2DLCN	Repairable	133	0.11%
960051151129	ICS VS BATTMG (PBC)	Repairable		
960051151220	PN-4LLCB	Repairable	64	0.55%
960051151244	PN-8COTU	Repairable	36	0.17%
960051151245	PN-2ODTB	Repairable	39	0.07%
960051151289	PN-8RSTG	Repairable	858	0.01%
960051151300	ALARM DSPP	Repairable	12	
960051153001	IPS DM PIMMD (PBC)	Repairable	8	
960053780089	DTR-32D-1P (BK) TEL	Repairable	9	1.50%
960053780091	ITR-8D-2P (BK) TEL	Repairable	8	3.00%
960053780093	ITR-16D-2P (BK) TEL	Repairable	8	3.00%
960051150136	SPN-4VCTI-B W/CA H323	Repairable	80	1.04%
960051151284	SPN-30DTCC-A (AP)	Repairable	33	0.98%
960051151285	SPN-30PRTA-A(AP)	Repairable	28	1.37%
960051151429	SPN-CP31A MP (PBC)	Repairable	11	0.00%
960051153000	ICS VS PIMMH	Repairable	61	0.05%
960051150135	SPN-IPTB-B H323(AP)	Repairable	20	0.00%
960051151286	SPN-30CCTA-A (AP)	Repairable	28	0.00%
960051151428	SPN-CP24B MP(PBC)	Repairable	11	1.20%
960051151435	SPN-CP27A MP (PBC)	Repairable	11	1.20%

960051151492	PZ-M606-A	Repairable	90	1.57%
960051151236	SPN-16VCTAA IP PAD-A	Repairable	43	1.30%
960051151257	SPN-32IPLAA IP PAD-C	Repairable	10	4.35%
960051151294	SPN-30PRTA-QSIG (AP)	Repairable	28	0.00%

Note:

- empty field for MTBF or failure rate => status : not applicable
- NSOs are advised to base local stock on failure rates, failure rates will be updated based upon experience figures and made available by ISL on the ISL NSONET pages.
- The 12 NC that are marked “green” are defined as critical 12 NCs, this implies that these 12 NC’s are critical to the operation of the 2000 IPS or IPS DM, for these 12 NC’s NSOs are advised to keep a local stock for fast repair in case of failure when NSO has contractual obligation to respond within a certain timeframe (e.g. less than 4 hour) in which item can not be supplied from central stock.
- As the list of deliverables is subject to change, because of roadmap developments (and therefore also the list of consumables and repairables) always refer for an actual overview to the NSONET pages from ISL.

Appendix B: SOPHO 2000 IPS Networking: Channel cards, handlers

Network type		type		quantity	max	Licenses (s/w activation key)
Trunking						
Analogue	Analogue Loop start	channel card	PN-8COTU	1 per 8 trunks	[1]	none
	Analogue (E&M 2/4)	channel card	PN-2ODTB	1 per 2 trunks	[1]	none
ISDN	PRI	channel card	SPN-30PRTA-A	1 per 30 channels	[1]	ISDN DCH per card (**5** incl, **3** extra option)
	BRI	channel card	SPN-4BRTA-C	1 per 8 channels	[1]	BRI per card (48 max included)
PBX Networking						
IPS to IPS (CCIS)	Analogue (E&M 2/4)	channel card	PN-2ODTB	1 per 2 trunks [A]	[2]	
		channel handler	SPN-SC00 CCH-D **+ modem**	1 per destination	[3]	CCIS per card
	Digital (E1)	channel card	SPN-30CCTA-A	1 per 30 channels per destination [A]	[2]	E1 per card (**5** incl, **3** extra option) CCIS per card
	ISDN (Event driven)	channel card	SPN-30PRTA-A	1 per 30 channels [A]	**[9]**	ISDN DCH per **link** (**5** incl, **3** extra option)
		channel card	SPN-4BRTA-C	1 per 8 channels [A]	[2]	BRI per **link** (48 max included)
		channel handler	SPN-SC00 CCH-D	1 per destination	[4]	CCIS per card, ECCIS per system
	IP	IP access	PZ-M606-A	Needs to be present	[5]	**1 IPT and 1 CCIS**
		channel card	SPN-32IPLAA IP PAD-C	Depends on traffic [B]	[7]	
		compression	**SPN-16VCTAA IP PAD-A**	Depends on traffic [C]	[8]	
IPS to Non-IPS	iSNet (=Q.SIG)	channel card	SPN-30PRTA-QSIG	1 per 30 channels per destination	[2]	ISDN DCH per card (**5** incl, **3** extra option) **1 CCIS per destination**
	Q.SIG (ISDN PRI)	channel card	SPN-30PRTA-QSIG	1 per 30 channels per destination	[2]	ISDN DCH per card (**5** incl, **3** extra option) **1 CCIS per destination**
	IP-H323	channel card	SPN-IPTB-B H323	1 per **3** 4VCTI-B cards	[2, 6]	IPT per card, **CCIS per card**
		compression	SPN-4VCTI-B W/CA H323	1 per 4 channels [A]		
Terminal Networking						
IP	LAN	channel card	SPN-32IPLAA IP PAD-C	Depends on traffic [B]	[7]	IP seat license per IP terminal (per 8)
		compression	SPN-16VCTAA IP PAD-A	1 if FAX	[8]	
	WAN	channel card	SPN-32IPLAA IP PAD-C	Depends on traffic [B]	[7]	IP seat license per IP terminal (per 8)
		compression	SPN-16VCTAA IP PAD-A	Depends on traffic [C]	[8]	

Notes:

[A] Each CCH uses one (1) channel, i.e. each destination

[B] Refer to traffic calculation in Prophix, per 32 channels

[C] Refer to traffic calculation in Prophix, per 16 channels

[1] total max 256 trunks/channels

[2] total max 128 trunks/channels

[3] total max 8 destinations

[4] total max 4 destinations

[5] max 256 destinations

[6] total max 8 cards per system

[7] max 2 per FP in its PIM

[8] max 2 per IPLAA

[9] total 16 Channels

Appendix C: SPN-8IPLA IP PAD-A Details

Compatibility Table

IP PAD /VCT	151226 SPN-16VCTA IP PAD	151236 SPN-16VCTAA IP PAD-A	FAX over IP (FoIP)	Modem over IP (MoIP)	Minimum System Software
SPN-32IPLA IP PAD	Required G.711/G.729/G.723	Required G.711/G.729/G.723	G.711 (64K)	N/A	R4.2
SPN-32IPLA IP PAD-A G.711	Optional G.711/G.729/G.723	Optional G.711/G.729/G.723	G.711 (64K)	N/A	
SPN-32IPLAA IP PAD-B G.711	Optional G.711/G.729/G.723	Optional G.711/G.729/G.723	G.711 (64K)	N/A	R6.1
SPN-32IPLAA IP PAD-C G.711	Optional VoIP G.711/ G.729/ G.723 FoIP G.711 (64K)	Optional VoIP G711/ G.729/ G.723 FoIP T.30 or G.711 (64K)	T.30 or G.711 (64K)	N/A	R6.2
SPN-8IPLA IP PAD-A G711/G.729/G.723	Not required Has built-in VCT	Not required Has built-in VCT	Pass-Through or G.711 (64K)	Pass-Through	R8
PZ-24IPLA G711/G.729/G.723	Not required Has built-in VCT	Not required Has built-in VCT	Pass-Through or G.711 (64K)	Pass Through	R8

Channel Allocation Table

IP-PAD Channel Allocation (System Data) and FAX Support								
Voice Over IP (VoIP)	SPN-32IPLAA IP PAD-C				SPN-8IPLA IP PAD-A			
					8IPLA	PZ-24IPLA Daughter Board		
G.711	8	16	24	32	8	16	24	32
G.729a	16 (1) 16VCTAA Or (1) 16VCTA		32 (2) 16VCTAA Or (2) 16VCTA		8	16	24	32
G723.1	16 (1) 16VCTAA Or (1) 16VCTA		32 (1) 16VCTAA Or (1) 16VCTA		8	16		

SPN-8IPLA w/PZ-24IPLA supports only 16 PAD channels when G723.1 is used.
8IPLA and 24IPLA networks to 32IPLA via Voice.

FAX Over IP (FoIP)	Modem Over IP (MoIP)	SPN-32IPLAA IP PAD-C		SPN-8IPLA IP PAD-A			
				8IPLA	PZ-24IPLA Daughter Board		
T.30	N/A	16	32	N/A	N/A	N/A	N/A
	N/A	16 (1) 16VCTAA	32 (2) 16VCTAA	N/A	N/A	N/A	N/A
*Pass-Through G.711/ G.726	Pass-Through G.711/ G.726	**16 (1) 16VCTAA	**32 (2) 16VCTAA	8	16	24	32

* FAX Pass-Through is tone detect technology that sets up specific conditions end to end for FAX and Modem on 8IPLA/24IPLA networked to 8IPLA/24IPLA.

** 32IPLA must have IPAD PROG-D1 Issue 1.0 or higher firmware to support Pass-Through FAX with 8IPLA/24IPLA.

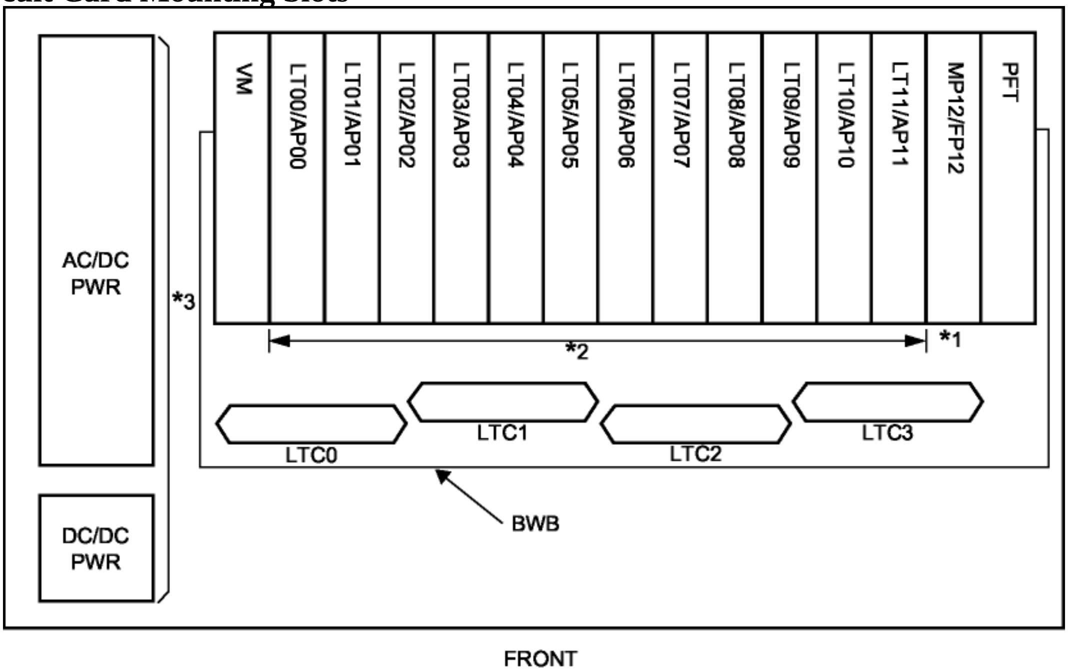
SPN-8IPLA and SPZ-24IPLA Specifications:

Function	Description	Remarks
Network Interface	Ethernet - IEEE 802.3 (10BASE-T) - IEEE 802.3u (100BASE-TX) - Auto Negotiation	100Mbps recommended
CPU	MIPS32 4KC (RISC CPU, built-in DSP)	
Number of channels	8ch	
Number of expanded channels	Can be expanded by adding a Sub card on the Main card. +24ch (+16ch when using G723.1)	
Voice Codec	G711 (64Kbps), G729a (8Kbps), G723.1 (5.3Kbps)	G723.1 (5.3Kbps only)
Voice Payload	10-40ms (10ms increment), 30ms fixed with G723.1	
Jitter Buffer	10-300ms	
DTMF Relay	Available	
FAX Relay	Available with (G.711), regarded as voice call	Pass-Through & G.711
Modem Relay	Not supported	Under Study
Peer-Peer control protocol	NEC proprietary protocol (UDP/TCP)	H245 equivalent
VLAN	Tag VLAN (IEEE802.1p)	
QoS	IP Precedence, Diffserv	
PAD	-14dB --- +14dB	
EC	G.168 (Max.64ms), with NLP (Non Linear Processor) function	

SPN-8IPLA and SPZ-24IPLA Number of simultaneous calls (Estimated)

Codec	Payload	Main Board	Main + Sub board	Remarks
G711	10ms	8	10	
	20ms	8	20	
	30ms	8	30	
	40ms	8	32	
G729a	10ms	8	10	
	20ms	8	20	
	30ms	8	30	
	40ms	8	32	
G723.1	30ms	8	16	
	60ms	8	16	

Circuit Card Mounting Slots



VM : PZ-VM00/VM00-M/VM03-M mounting slot
LT00-LT10 : Line/Trunk card mounting slots
AP00-AP10 : Application Processor card mounting slots
LT11-AP11 : Line/Trunk/Application Processor card/PN-CP27 (MP1) backup CPU mounting slots
MP12 : PN-CP24/ PN-CP27 (MP0) with backup CPU mounting slot
FP12 : PN-CP15 mounting slot
PFT : PZ-8PFTB mounting slot
AC/DC PWR : PZ-PW121/PW126 mounting slot
When using the A361 PIM-DC (PIM for –48 V DC power supply system), this slot is used for DC/DC PWR (PZ-PW135 mounting slot).
DC/DC PWR : PZ-PW122 mounting slot
When using the A361 PIM-DC (PIM for –48 V DC power supply system), this slot is not used.

Notes
(*1) PN-CP24 (MP) card is to be mounted in the MP12 slot of PIM0. PZ-M606-A (ETHER) card is to be mounted on the PN-CP24 (MP) card. PN-CP15 (FP) card is to be mounted in the FP12 slot of PIM2, 4, 6 according to the system con-figuration.
(*2) Either line/trunk cards or application processor cards can be mounted in the LT00/AP00-LT1 1/ AP11 slots of PIM0-7. For mounting condition of PN-RTA (RTA), refer to the In-Skin Router Installation Guide.
(*3) When using PIM for –48 V DC power supply system, the AC/DC PWR slot is used for the DC/DC PWR slot. And the DC/DC PWR slot is not used. The other slots can accommodate the same cards as a usual PIM.

SOPHO 2000 IPS IP PAD Mounting Conditions

PN-32IPLA and 16VCT

PN-32IPLA (IP-PAD) card is to be mounted in the LT01 and/or LT05 slots of PIM0-7.
In the following cases, other L/T card and/or AP card can be mounted in the adjoining left side slot of IP-PAD card.

IP-PAD CARD		16VCT CARD	×: ALLOWED -: NOT ALLOWED							
			OTHER LINE/TRUNK CARDS				APPLICATION PROCESSOR CARDS			
CHANNEL NO.	SLOT NO.		LT02	LT03	LT06	LT07	LT02	LT03	LT06	LT07
16 CH	LT01	NONE	×	×			×	×		
	LT05				×	×			×	×
24 CH	LT01	NONE	-	×			-	×		
	LT05				-	×			-	×
32 CH	LT01	NONE	-	-			-	-		
	LT05				-	-			-	-
16 CH	LT01	ONE CARD	-	×			-	×		
	LT05				-	×			-	×
24/32 CH	LT01	TWO CARDS	-	-			-	-		
	LT05				-	-			-	-

In the following cases, according as number of channels, do not mount any other L/T cards in the LT08-LT1 1 slots when you mount the IP-PAD card in the LT05 slot.

IP-PAD CARD		×: ALLOWED -: NOT ALLOWED							
		OTHER LINE/TRUNK CARDS				APPLICATION PROCESSOR CARDS			
CHANNEL NO.	SLOT NO.	LT08	LT09	LT10	LT11 NOTE	LT08	LT09	LT10	LT11 NOTE
16 CH	LT05	-	-	×	×	×	×	×	×
24 CH			-	-	×	×	×	×	×
32 CH		-	-	-	-	×	×	×	×

Note: *For Backup CPU system, do not mount in the LT08-LT10 slots.*

PN-8IPLA

PN-8IPLA (IP-PAD) card is to be mounted in the LT00 and/or LT04 slots of PIM0-7.
In the following cases, other L/T card and/or AP card can be mounted in the adjoining left side slot of IP-PAD card.

IP-PAD CARD		24DSP CARD	×: ALLOWED - :NOT ALLOWED											
			OTHER LINE/TRUNK CARDS						APPLICATION PROCESSOR CARDS					
CHANNEL NO.	SLOT NO.	CHANNEL NO.	LT 01	LT 02	LT 03	LT 05	LT 06	LT 07	LT 01	LT 02	LT 03	LT 05	LT 06	LT 07
8 CH	LT00	NONE	×	×	×				×	×	×			
	LT04					×	×	×				×	×	×
16 CH	LT00	8 CH	-	×	×				-	×	×			
	LT04					-	×	×				-	×	×
24 CH	LT00	16 CH	-	-	×				-	-	×			
	LT04					-	-	×				-	-	×
32 CH	LT00	24 CH	-	-	-				-	-	-			
	LT04					-	-	-				-	-	-

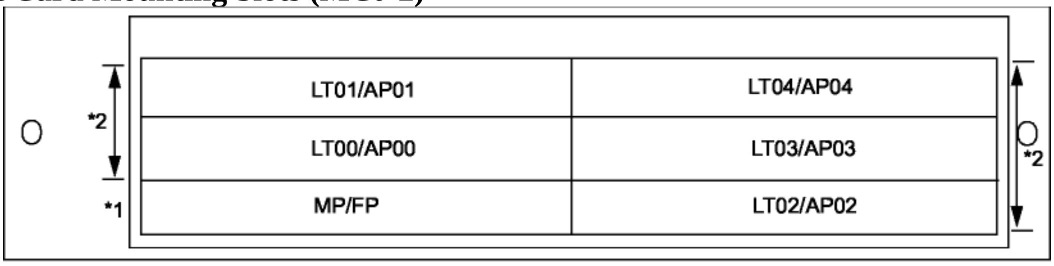
In the following cases, according as number of channels, do not mount any other L/T cards in the LT08-LT1 1 slots when you mount the IP-PAD card in the LT04 slot.

IP-PAD CARD		24DSP CARD	×: ALLOWED -: NOT ALLOWED							
			OTHER LINE/TRUNK CARDS				APPLICATION PROCESSOR CARDS			
CHANNEL NO.	SLOT NO.	CHANNEL NO.	LT08	LT09	LT10	LT11 NOTE	LT08	LT09	LT10	LT11 NOTE
8 CH	LT04	8 CH	-	×	×	×	×	×	×	×
16 CH			-	-	×	×	×	×	×	×
24 CH		16 CH	-	-	-	×	×	×	×	×
32 CH		24 CH	-	-	-	-	×	×	×	×

Note: *For Backup CPU system, do not mount in the LT08-LT10 slots.*

SOPHO IPS DM and SOPHO IPS DMR Mounting Conditions

Circuit Card Mounting Slots (MC0-2)



FRONT SIDE

LT00-LT06 : Line/Trunk card mounting slots
AP00-AP06 : Application Processor card mounting slots
MP : PN-CP24/PN-CP31 mounting slot
PFT : PZ-4PFTA mounting position
Notes (*1) PN-CP24/PN-CP31 (MP) card is mounted in the MP/FP slot of MC0. PZ-M606-A (ETHER) card is mounted on the PN-CP24/PN-CP31 (MP) card. PN-CP15/PN-CP19 (FP) card is mounted in the MP/FP slot of MC2 when the system is three-MC configuration. No card is mounted in the MP/FP slot of MC1.
(*2) Either line/trunk cards or application processor cards can be mounted in the LT00/AP00-LT04/ AP04 slots of MC0-2.
(*3) PZ-4PFTA card is mounted in the bottom of MC0-2.

PN-8IPLA

When using PN-8IPLA (IP-PAD) card without PZ-24IPLA (24 DSP) card, IP-PAD card can be mounted in LT00 slot and LT04 slot of MC0-1, and the other LT/AP card can be mounted in the vacant slot as follows.

LT02	Other LT/AP	Other LT/AP	LT06
LT01	Other LT/AP	Other LT/AP	LT05
LT00	IP-PAD	IP-PAD	LT04
		Other LT/AP	LT03

When using PN-8IPLA (IP-PAD) card with PZ-24IPLA (24 DSP) card, IP-PAD card can be also mounted in LT00 slot and LT04 slot of MC0-1. But the IP-PAD card is recommended to mount in either LT00 slot or LT04 slot to use the LT slots efficiently. And depending on the number of channels (8/16/24/32) that is assigned by office data, the other LT/AP card can be mounted in vacant slot as shown in the following mounting examples.

Using 16 channels (IP-PAD is in LT00):

LT02	Other LT/AP	Other LT/AP	LT06
LT01	AP only	Other LT/AP	LT05
LT00	IP-PAD + 24DSP	Other LT/AP	LT04
		Other LT/AP	LT03

Using 24 channels (IP-PAD is in LT00)

LT02	AP only	Other LT/AP	LT06
LT01	AP only	Other LT/AP	LT05
LT00	IP-PAD + 24DSP	Other LT/AP	LT04
		Other LT/AP	LT03

Using 32 channels (IP-PAD is in LT00)

LT02	AP only	Other LT/AP	LT06
LT01	AP only	Other LT/AP	LT05
LT00	IP-PAD + 24DSP	Other LT/AP	LT04
		AP only	LT03

Using 16 channels (IP-PAD is in LT04)

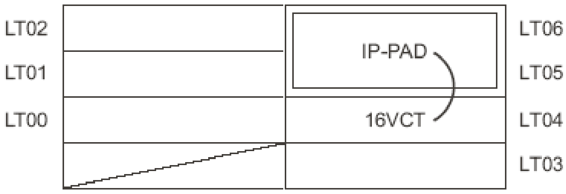
LT02	Other LT/AP	Other LT/AP	LT06
LT01	Other LT/AP	AP only	LT05
LT00	Other LT/AP	IP-PAD + 24DSP	LT04
		Other LT/AP	LT03

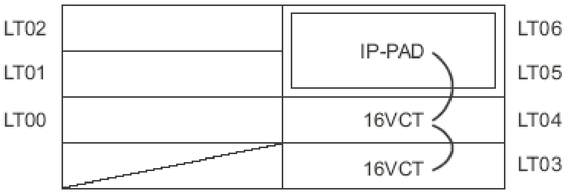
Using 24/32 channels (IP-PAD is in LT04)

LT02	Other LT/AP	AP only	LT06
LT01	Other LT/AP	AP only	LT05
LT00	Other LT/AP	IP-PAD + 24DSP	LT04
		Other LT/AP	LT03

PN-32IPLA

PN-32IPLA-A (IP-PAD) card is recommended to be mounted in the LT04 slot of MC0-2 to use the LT slots eficiently. The first 8 ports are provided by LT04 slot and remaining 24 ports are provided by Virtual LT ports. The IP-PAD card is 2-slot width physically. Therefore, any card cannot be mounted in the LT04 slot. When using PN-1 6VCTA-A (16VCT) card, they should be mounted in LT02 slot and LT03 slot to connect the front cable.





PN-4VCTI

PN-4VCTI (4VCT) card is mounted in the next to PN-IPTB (IPT) card. When mounting two 4VCT cards, IPT card and 4VCT cards are mounted in the right side slots of MC0-2. LT/AP cards can be mounted in vacant slots.