TELESİS
SOLUTIONS FOR A WIRED WORLD

X1 rev4
Switching System
TELESÍS
X1 rev4

ADVANCED DIGITAL
LARGE CAPACITY SWITCHING SYSTEM
IN BRIEF:

Thank you

... for choosing the Telesis X1 (revision 4) large capacity switching system. It is manufactured to the highest quality standards and tested vigorously to comply with requirement for its product specifications.

Telesis CZ

About Telesis CZ Ltd

Established in 1998, Telesis CZ has grown rapidly as a leading-edge provider of solutions for Central Office, Rural, Tandem, Signaling Converter, ISDN PBX, and IP Telephony applications. With 10 years experience, Telesis CZ as delivered to its customers over 0.5 million telephone lines. Our products have been chosen by public- and private-sector entities in various countries worldwide. Telesis X1 systems have been installed in more than forty countries.

Telesis CZ excellent reputation in international markets rests on excellent product design and a competitive cost/performance ratio. You may find further information about Telesis CZ and its products at http://www.telesis.cz

SYSTEM OVERVIEW:

STATE OF THE ART TECHNOLOGY

The X1 switching system complies with ITU-T Recommendations Q.541, Q.542, Q.543 concerning Digital Exchange Design Objectives. The system is designed in the beginning of 1990s, but many refinements have been incorporated since then.

The X1 switching systems have the following characteristics:

- the capacity is sufficient or easily expandable to ensure the availability of capacity needed for future new functions or services to be introduced in the exchanges,
- all equipment is mounted on printed circuit boards which can be easily plugged in,
- digital 30 channel PCM/E1 interfaces for trunks and PBX lines,
- architecture for NGN (next generation networks)
- provision of modern subscriber services according to ETSI Recommendations.
- embedded VoIP (voice over IP) technology

The X1 is capable of inter-working between all its digital and analog signaling schemes. Each variant of incoming and outgoing lines associated with a signaling system generates and receives FITEs and BITEs (forward and backward inter-working telephony events) according to on-line, real-time procedures specified for each signaling system in the ITU-T recommendations. The present interfaces and signaling schemes that are supported by the X1 exchanges are given in the following sections

SYSTEM CAPACITY

The X1 capacity depends on the demands and the place of the system in the network. Particular values are defined for each system. The X1 switching system allows, in an economic way, the realization of switching system in the capacity range of tens of thousands of subscriber lines and thousands of trunks

In order to facilitate expansions of the system in future, the X1 is built on a modular principle. It is possible to distinguish functional units that can be replaced and/or expanded.

The Telesis X1 has a modular mechanical structure. The installation of the system or expanding an already operating one by adding subscriber and/or trunk line interfaces is simple and fast.
An X1 system has a CPU module and peripheral modules. There are two types of peripheral modules: the analog and E1 peripheral interface module XPMN and the BRI interface module BPMN. These two modules differ in their backplane boards. Each peripheral interface module has its own dedicated DC-DC voltage converter and has 15 slots for interface boards.

The X1 allows thousands of H.323 endpoints to be registered into its integrated gatekeeper. These endpoints may be IP trunk routes and/or H.323 hard/soft phones with their own IP addresses. The X1 can also register to multiple gatekeepers simultaneously. Similarly, the X1 allows thousands of SIP user agents to be registered into its integrated registrar. These user agents may be IP trunk routes and/or SIP hard/soft phones with their own IP addresses. The X1 can also register to multiple registrars simultaneously. In an X1, TDM circuits, H.323 endpoints, and SIP user agents exist altogether.

Furthermore, Telesis proprietary xSIP (eXtended SIP) protocol allows value added services of the DTS digital telephone sets also to be functional over IP.

By means of expansions and adaptations it is possible to use the X1 as:

- Central Office
- Toll switch
- Tandem switch
- Transit switch
- Rural switch
- Signaling (protocol) converter
- SSP, Service Switching Point for SS7
- STP, Signal Transfer Point for SS7
- PBX
- IP Telephony System

The X1 incorporates a central processing unit that takes full advantage of the massive amount of computing power that is now available, along with the myriad of software development tools for the industrial computers. Specifically, it is based on an industrially standard processing units.

Subscriber and trunk port expansions may be done by adding boards or modules into the system.

SYSTEM RESOURCES

The Telesis X1's common control function is implemented in a centralized manner. The control and switching module, including industrial CPU, digital signal processors, memory elements, switching matrix, HDLC handlers, and all other service circuits, is duplicated for full on-line redundancy. Improved speed, security, and reliability are guaranteed by the placement of the generic programs and operating parameters in non-volatile semiconductor memory elements. The core of the X1 and the essence of its technological sophistication is the control and switching unit, which includes:

- Powerful industrial CPU
- Powerful digital signal processors
- Program and parameter memories
- Operating memories
- Switching circuits
- Tone generator/synthesizer
- HDLC circuits (for CCS and system control)
- Conference channels
- Ethernet interface
- Integrated H.323 gatekeeper and SIP registrar
- Integrated voice cipher for VoIP calls (AES-256)
- Real-Time Clock (RTC)
- Integrated DVR (Digital Voice Recorder) for
  - System messages (announcements)
  - Voice mail applications
  - Bi-directional recording of conversation
- Other service components, such as timing circuits and signaling circuits
A large number of technological factors have influenced the final design specifications of the X1. A typical example is the selection of a central processing unit which is much effected by massive increase in computing power of and large number of software development tools available industry. The X1 is based on an industrially standard processing unit. System-wide processing operations are thus carried out on this unit.

WDT circuitries on the processing unit checks the system operation. If a short system failure occurs, WDT causes the automatic restart of the software.

The estimated system recovery time for both manual and automatic restart of the system is very short (a few seconds only).

**REDUNDANCY**

To ensure the system reliability, there exists a stand-by redundant control and switching module in the Telesis X1 Switching System. The control and switching module, including industrial CPU, digital signal processors, memory elements, switching matrix, HDLC handlers, and all other service circuits, is duplicated for full on-line redundancy. The duplicated control and switching circuits and devices in detail are:

- Industrial CPU
- Digital signal processors
- Program and parameter memories
- Operating memories
- Switching circuits
- Tone generator/synthesizer
- HDLC circuits (for CCS and system control)
- Conference channels
- Ethernet interfaces
- Serial ports
- Integrated H.323 gatekeeper and SIP registrar
- Integrated xSIP registrar
- Integrated voice cipher for VoIP calls (AES-256)

- Real-Time Clock (RTC)
- Integrated DVR (Digital Voice Recorder)
- MFCR2, MFRI transceivers
- DTMF transceivers
- Caller ID transceivers
- Other timing circuits

Each control and switching module has its own DC-DC converters. Furthermore, AC-DC rectifiers are with redundancy.

**MEMORIES**

The non-volatile (NV) type memories in the Telesis X1 Switching System are used for storing:

- Boot program,
- Operational software Xymphony,
- Programmed parameters,
- Voice records (100 hours),
- CMDR call records (tens of thousands of call records), and
- System records.

NV memories are solid state integrated circuits and never loose data in case of power failures even for a long period of time.

Volatile memories (RAMs) are used for running the operational software (firmware) Xymphony within the X1.

**ETHERNET INTERFACES**

10/100 BaseT ethernet interfaces on Telesis X1 Switching Systems provide numerous IP Telephony applications and system management solutions. Some of these are:

- Web based system management
- CMDR (detailed call records) collection
• System alarm and warning messages collection
• Statistics date collection
• Voice (conversation) records collection
• Xymphony API server and client communication
• H.323 protocol support with H.225.0 version 5, H.235 version 3, H.245 version 12, H.450, AES FIPS PUB 197. Connections between the system and H.323 entities.
• SIP Session Initiation Protocol support with RFC 3261. Connection between the system and SIP entities.
• xSIP (eXtended SIP). Connection between the system and xSIP terminals

INTEGRATED AUTOMATIC VOICE RECORDER
Automatic voice recording of all calls (conversations) is critical to some operations and installations such as:
1. Public Safety and Health
2. Call Centers
3. Air, Maritime, Railway Traffic Control
The integrated DVR (digital voice recorder) within Telesis X1 Switching Systems may be used in such operations and applications for bi-directional (both calling and called party voices) recording. The combination of the integrated DVR hardware (with a storage capacity of 100 hours or more) and a PC running XPort Utility is the heart of this solution. In this solution, the integrated DVR operates as recording buffer, whereas the PC with XPort is the main archiving device.

SOFTWARE
The Telesis X1 is a Stored-Program-Controlled (SPC) system. Its leading-edge operational software, Xymphony, has been developed by Telesis specifically for this switching system. Among the many aspects of Xymphony is its advanced algorithm for creating routing tables, complete with options for Least-Cost Routing (LCR), real-time charging tables, and various subscriber/trunk classes.

With many programmable features, Xymphony makes the traffic management highly efficient and versatile. The convenience of dealing with all-in-one standard software allows the user to support as many features as desired, up to the maximum capacity: thus, an organization's capacity can grow without necessarily updating the software. Upgrading the Xymphony operating system in the field is accomplished simply by downloading a new version of it directly onto the X1's solid state disc over IP without interruption to any of the system's services.

Xymphony within the X1 System is modular and well structured and is developed by a high level language of C.

Xymphony also provides:
• easy expansion and modification when introducing new services and technologies, to guarantee high reliability,
• introduction procedures for new software parts or repairs with minimum service interruptions. Software repairs are normally introduced without any loss of answered calls. New software packages are introduced with a minimum service interruption, independent of the size of the switching system, to provide well defined procedures for collecting of information, restoring to normal operation and introduction of repairs in case of disturbances caused by some errors.

The volume and structure of database in the software package can be tailored to the size and equipment set up of the X1, provided that the system allows the assigned data stored size per function to be changed by the staff. However the database is standardized for the maximum exchange size. It is possible after the loading of the program to alter the installed amount of equipment via commands given by means of the maintenance PC.

Xymphony software provides strategies to prevent the occurrence of a software failure to escalate to a total break down and to present sufficient information for quick solving of the software fault.

The X1 system verifies the integrity of its software at regular basis.

TRAFFIC-HANDLING CAPACITY
The Telesis X1 has been tested with over 300,000 Busy Hour Call Attempts (BHCA) without any loss in the quality of service. The X1 employs a high-capacity switching matrix. The advanced design incorporates state-of-the-art Industrial CPU and Digital Signal Processors that are capable of executing hundreds of millions of instructions a second. Such highly efficient use of the system's resources, with the help of intelligent algorithms, leaves very little, if any, possibility that an overload condition will arise.
The X1 allows thousands of H.323 endpoints to be registered into its integrated gatekeeper. These endpoints may be IP trunk routes and/or H.323 hard/soft phones with their own IP addresses. Furthermore, the X1 can register to multiple gatekeepers simultaneously. For an IP to IP call, the media is direct from an H.323 endpoint to another, such a call does not use a system channel resource (i.e., PCM channel from the switching matrix). The X1 provides the address resolution and some other services according to the predefined profile of every registered endpoint. For such calls, the X1 acts as a softswitch. There can be any number of simultaneous IP to IP, H.323 calls.

The X1 allows thousands of SIP user agents to be registered into its integrated registrar. These endpoints may be IP trunk routes and/or SIP hard/soft phones with their own IP addresses. Furthermore, the X1 can register to multiple registrar simultaneously. For an IP to IP call, the media is direct from a SIP user agent to another, such a call does not use a system channel resource (i.e., PCM channel from the switching matrix). For such calls, the X1 acts as a softswitch. There can be any number of simultaneous IP to IP, SIP calls.

In the operation of the X1 system, there is a sufficient margin for overload situations. The capacity is sufficient or easily expandable to ensure the availability of capacity needed for future new functions or services to be introduced in the system.

Faulty circuits are automatically blocked by the system. Furthermore, ports may be:

- blocked both-way calls
- blocked for incoming calls only
- blocked for outgoing calls only
- manually.

**TRAFFIC MEASUREMENTS**

Traffic measurements of the X1 provide the data base from which the dimensioning, planning, operation and management of the telephone network are carried out. Information gathered from these measurements can be used for

- identifying the traffic patterns and distribution on a route and destination bases,
- determining the amount of traffic in the switching system and the network,
- monitoring the continuity and grade of service.

The following two types of measurements are available at least:

- The call records generated by the X1 contain all data (e.g. time of occurrence of signaling event, dialed digits, etc.) characterizing each individual call attempt. Very detailed traffic data can be obtained by the analysis of call records. The call records are generated continuously and are collected and stored in the maintenance PC,
- Real time traffic measurements.

The X1 is capable to perform all functions and traffic measurements described in ITU-T recommendations Q.544, E.410, E.411, E.412, E.500 and E.502. It is possible to perform simultaneously all the measurements described in ITU-T Rec. Q.544.

**Statistical Reports Format:**

**Title Field has the following information:**

- Statistics period
- # (number) of SETUP attempts generated by the system
- # of accesses carried out without referring to a Routing Criteria Table
- # of DTMF requests
- # of failed DTMFs
- # of accesses carried out by referring to the ‘Port Access’ law
- # of accesses carried out by referring to the ‘Trunk Access’ law
- # of accesses carried out by referring to the ‘Single Port Routing’ law
- # of test tone accesses carried out by referring to the ‘Test tone’ law
- # of vacant tone accesses by referring to the ‘Vacant tone’ law
- # of overflow tone accesses carried out by referring to the ‘overflow tone’ law

**Reports Field has the following information:**

- Route Number
- Total # of outgoing seizure requests from the route
- # of failed seizure requests
- Total # of successful seizure requests
CALL ROUTING CAPABILITIES

The advanced routing algorithm within the Telesis X1 system is carried out in four steps:

- First Step: Incoming A-Party Analysis,
- Second Step: B-Party (called-party) Pre-Analysis
- Third Step: B-Party (called-party) Main Analysis
- Fourth Step: Outgoing A-Party Analysis

The same routing algorithm is applied for both TDM and VoIP calls.

Incoming A-Party Analysis (optional):

Incoming A-Party (calling-party) analysis is the first (optional) step in Routing Analysis. It is applied to each originating or incoming interface. In order to facilitate this analysis, an incoming A-party analysis table is assigned to the originating user (TDM or VoIP). The incoming A-party analysis is done by analyzing the calling-party’s Number (up to 32 digits), Category, Nature of Address (or type of number), Numbering Plan, Presentation Status, Screening Status, etc. after which any or all of these parameters may be translated and then selected for a called-party pre-analysis table.

Called-Party Pre-Analysis (optional):

The Pre-Analysis is carried out prior to the main called-party analysis to truncate some called-party prefixes or reject certain addresses (destinations) altogether. Telesis has developed pre-analysis in order to simplify the main analysis, especially for when a Telesis X1 system services several operators in transit mode. Pre-analysis is applied to as many as 16-digit called-party prefixes.

Main Analysis:

The main called-party analysis is carried out on the called-party prefixes resulting from the pre-analysis. Up to 16-digit-long prefixes may be analyzed. It is possible to define the minimum and maximum digits for starting the analysis. This analysis is date- and time-dependent, i.e., for different days of the week and different times of the day, the analysis may result in different outcomes:

- Route
- Called-party number replacement (up to 32 digits)
- Call class to be used for detailed CMDR
Authorization level necessary to realize the routing
Tariff
The route that is decided on after this analysis may be:
- A local termination (i.e., subscriber)
- A trunk group from which the call is to be routed outward
- A physical address of a digital/analog subscriber/trunk port available in the switch
- A remote H.323 gatekeeper
- A remote SIP registrar
- A subscriber service
- A system service for maintenance and administrative purposes
- A system resource like integrated voice mail, message-storage hardware etc.
- A tone or a system message stored in the switch

Outgoing A-Party Analysis (optional):
Outgoing A-party (calling-party) analysis is the last (optional) step in the routing procedure. It is applied to the terminating or outgoing interface. In order to facilitate this analysis, an outgoing A-party analysis table is assigned to the called subscriber/outgoing trunk. The outgoing A-party analysis is done by analyzing the calling party's number (up to 32 digits), Category, Nature of address, Numbering Plan, Presentation Status, Screening Status, etc. and, additionally, the Called-Party's Nature of Address, after which any or all of these parameters may be translated.

SYSTEM MANAGEMENT AND BILLING
The Telesis X1 can managed over IP. System programming and updating the Xymphony operating system can easily be managed either on site or remotely, through a WEB browser connection to web server integrated into the system or XMan (eXchange Manager) connection. All system-management operations are performed without interrupting the functions of the system. The integrated web server or XMan allows:
- Subscriber administration
- Routing administration
- Trunk-circuit administration

Call-charging administration
Furthermore, the Telesis XPort utility allows:
- Detailed monitoring of signaling
- Traffic measurement including collecting detailed call-data records
- Downloading conversation records
A powerful feature of Xymphony concerns call charging. It is capable of prefix digit analysis of up to 16 digits. Destination-based tariff tables can be constructed to define:
- Unit-charge periods depending on destination
- Unit-charge periods depending on the day of the week and time of day
- Vacations and other special days for separate unit-charge periods
- Fixed-charge calls
- Charge-free calls
All unit-charge periods are specified in multiples of 100 msec (milliseconds). Hence, a unit-charge period may take a value from 100 msec to several minutes in multiples of 100 msec. Furthermore, call-charging information generated by far-end equipment can also be used.
A subscriber service (such as call forwarding, reminder) can have a separate fixed fee charge for using, activating, deactivating, or querying the service.
Call charging is processed in real time during the call. Pre-paid status within a certain amount of credit can be defined for each subscriber.
A detailed record for each call and service use may be generated by Xymphony (operating firmware of the X1) and is transmitted to Telesis XPort Software Utility (running on the maintenance or billing PC). Each record in the database holds details of a call, including:
- sequence number
- type of record
- date and time of the record generated within the Telesis system
- duration of the call in the record
- port access code (address), which is to be billed
- port access code, which made a password protected call
• outgoing port or the called port address
• digits received from the originating port
• replaced digits after routing analysis for an outgoing call
• password for a password protected call
• time of alarm indication for a wake up call
• calling party number for an incoming call
• number of the calling party
• total number of charge pulses received (real charge pulses)
• total number of locally generated charge pulses (pseudo charge pulses)
• channel information for CCS signalling ports
• reference number of the call
• switching system's name
• services used for the call like Last number redial, Dial from user pool, Dial by special key, System controlled hot line, User controlled hot line, Redial in incomplete, Redial destination busy, Alerting timeout, No information, User controlled call forwarding (BRI ISDN), User controlled unconditional call forwarding, User controlled call forwarding busy, User controlled call forwarding no reply, System controlled unconditional call forwarding, System controlled call forwarding busy, System controlled call forwarding no reply, Call transfer after answer, Call back, User controlled divert routed call, Call transfer while the destination is busy, Call transfer while the destination is ringing, Alternative routing

XPort can send call records to a printer or a text file in a user-defined line format. Its database also has a table that keeps a separate charge counter with multiple registers for accumulating data for Local, NWD, ISD calls, Subscriber Services, and Total (sum of all the other registers) for each subscriber.

The X1 has no hardware parts that require periodic maintenance. However, the call data accumulated in the X1's solid state disc should periodically be downloaded through XPort to its database to create space on the system's disc.

**SYSTEM AND USER SERVICES**

Telesis X1 switching systems provide, at least, system features, system parameters, and user services, described below. If it is desired, these may be activated or deactivated. With hundreds of programmable features and services, an X1 make traffic management highly efficient and versatile. The convenience of dealing with all-in-one system software Xymphony allows the user to support as many features as desired, up to the maximum capacity; thus, an organization's capacity can grow without necessarily updating the software. Another aspect of the system's software is an advanced algorithm for creating routing tables.

Many of the user services, mentioned below, are applicable for SIP calls with using the basic SIP supplementary services such as Invite (call hold), call forward and call transfer (with refer method). Similarly, these services are also applicable for H.323 calls with using the basic H.450 supplementary services.

For ensuring the easy and fast programming of the X1, each programmable parameter is set to a default value. The standard default values are carefully selected and optimized for a wide range of organizations' requirements. Thus, the flexibility does not bring a complexity.

| Access code |
| Account code |
| Account name |
| Advice of charge coefficient |
| Advice of charge coefficient as divider |
| AES-256 media encryption |
| Alerting message |
| Alerting transfers do not clear |
| Alternative routes |
| Alternative route causes |
| Alternative ACD system messages |
| Always try to seize first port |
|ANI (automatic number identification)   |   |
|Any IP address can register           |   |
|A-Party (calling party) analysis      |   |
|Analog trunk to trunk connects        |   |
|Answer camped call                    |   |
|Attendant positions                   |   |
|Authorization modified tone           |   |
|Authorization modified tone cadence modification |   |
|Authorization modified tone frequencies modification |   |
|Authorization modified tone level modification |   |
|Auto dialing                          |   |
|Auto dialer frequencies for detection |   |
|Automatic attendant                   |   |
|Automatic Call Distribution ACD       |   |
|B-Party (called party) analysis       |   |
|Boot Xymphony                         |   |
|Busy transfers do not clear           |   |
|Busy message                          |   |
|Busy message download                 |   |
|Busy message remove                   |   |
|Busy message upload                   |   |
|Busy tone                             |   |
|Busy tone cadence modification        |   |
|Busy tone frequencies modification    |   |
|Busy tone level modification          |   |
|Call back                             |   |
|Call center                           |   |
|Call class                            |   |
|Call forwarding busy                  |   |
|Call forwarding no reply              |   |
|Call forwarding unconditional          |   |
|Call hold                             |   |
|Call hold disabled                    |   |
|Call intrusion                        |   |
|Call intrusion authorization          |   |
|Call park                             |   |
|Call pickup                           |   |
|Call resume                           |   |
|Call routing                          |   |
|Call status                           |   |
|Call suspend                          |   |
|Call transfer                         |   |
|Called party (B-Party) analysis       |   |
|Called party numbering plan           |   |
|Caller ID                             |   |
|Calling party (A-Party) analysis      |   |
|Calling party filtering               |   |
|Calling plans                         |   |
Camp on
Camp on if no channel
Camp on tone
Camp on tone cadence modification
Camp on tone frequencies modification
Camp on tone level modification
Camp on with alerting tone
Camp on with busy tone
Charge pulse limit per call
Charge pulse on answer
Charge pulse on connection
Charge pulse period
Clear held calls
CMDR (Call Message Detailed Record)
Common pool edit
Conference
Confirmation tone
Confirmation tone cadence modification
Confirmation tone frequencies modification
Confirmation tone level modification
Copy routing table
Copy user properties
Credited extension
Credit edit
CTI (computer telephony integration)
DDI (direct dialing in)
Declare external IP address (for H.323)
Declare external IP address (for SIP)
Default max-forwards
Default Xymphony
Delayed digits as DTMF
Delayed hot line
Delayed hot line tone
Delayed hot line tone cadence modification
Delayed hot line tone frequencies modification
Delayed hot line tone level modification
Dial tone
Dial tone cadence modification
Dial tone frequencies modification
Dial tone level modification
Dial from user pool
Dial from common pool
Dial from directory
DID (direct inward dialing)
Directory
DISA (direct inward system access)
Divert routed calls
Diversion active tone
Diversion active tone cadence modification
Diversion active tone frequencies modification
Diversion active tone level modification
DNS server IP address
Do not disturb
Do not disturb activation
Do not disturb deactivation
DVR (digital voice recorder)
Enable advice of charge identifier
Enable connected number identifier
Enable pending call warning
Enable progress indicator
Forced release
Force transmit media frame length
G.168-2 echo canceler
G.711 codec
G.711 transmit buffer length
G.711 preferred receive buffer
G.711 transmit silence suppress
G.723 codec
G.723 transmit buffer length
G.723 preferred receive buffer
G.723 transmit 6.4kbps compress
G.723 transmit 6.4kbps compress
G.729 codec
G.729 transmit buffer length
G.729 transmit silence suppress
G.729 preferred receive buffer
Gateway IP address
Gateway external IP address
H.323 protocol
H.323 gatekeeper identifier
H.323 RAS multicast port UDP
H.323 unicast port UDP
H.323 call signaling port TCP
H.245 terminal type
H.450 supplementary services
Hold message
Hold message download
Hold message remove
Hold message upload
Hot line
Howler tone
Howler tone cadence modification
Howler tone frequencies modification
Howler tone level modification
Ignore checksum in signaling (for H.323)
Ignore checksum in signaling (for SIP)
Ignore declared IP addresses
Incoming call only
Inhibit call transfer warning
IP address (system)
ISDN supplementary service 3-PTY
ISDN supplementary service AOC-D/E
ISDN supplementary service CCBS
ISDN supplementary service CCNR
ISDN supplementary service CFU
ISDN supplementary service CFNR
ISDN supplementary service CLIP
ISDN supplementary service CLIR
ISDN supplementary service COLP
ISDN supplementary service COLR
ISDN supplementary service ECT
ISDN supplementary service DDI
ISDN supplementary service HOLD
ISDN supplementary service MCID
ISDN supplementary service MSN
ISDN supplementary service UUS
Last number redial
LCR (least cost routing)
Local call ring type1 tone
Local call ring type2 tone
Local call ring type3 tone
Local call ring type4 tone
Local call rings cadence modification
Local call ring tones cadence modification
Local call ring tones frequencies modification
Local call ring tones level modification
Lower algorithms disabled
Malicious call trace
Media jitter buffer
Melody composer
Message waiting
Message waiting by alerting
Message waiting callback
Multi station call
Multiple entry tone
Multiple entry tone cadence modification
Multiple entry tone frequencies modification
Multiple entry tone level modification
Multiple hold tone
Multiple hold tone cadence modification
Multiple hold tone frequencies modification
Multiple hold tone level modification
Music on hold
MWI (message waiting indicator)
Night service
No authorization check for common pool
Non discriminative ringing
No route if CMDR buffer is full
Outgoing call only
Overflow tone
Overflow tone cadence modification
Overflow tone frequencies modification
Overflow tone level modification
Override charge pulse limit
Parameter store
Park message
Password login
Password update
Permitted call class
Port route
Port status
Records not reported
Redial on alerting timeout
Redial on destination busy
Redial on incomplete dialing
Reject diverted calls
Reject diverted calls activation
Reject diverted calls deactivation
Reminder message
Reminder message download
Reminder message remove
Reminder message upload
Reminder service
Reminder service activation
Reminder service deactivation
Reminder service tone
Reminder sequence
Remote port to send messages
Registrar/Gatekeeper type
Room house keeping
Room report
Room occupancy
Room query
Routed call pickup
RTP ToS-Diffservices
Runtime status of the system
Secondary dial tone
Secondary dial tone cadence modification
Secondary dial tone frequencies modification
Secondary dial tone level modification
Security profile
<table>
<thead>
<tr>
<th>Feature</th>
<th>Feature</th>
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</thead>
<tbody>
<tr>
<td>SIP (session initiation protocol)</td>
<td>User pool update</td>
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<tr>
<td>SIP supplementary services</td>
<td>Use rport in invite message</td>
</tr>
<tr>
<td>SIP UDP signaling port</td>
<td>User service charges</td>
</tr>
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<td>Single port routing</td>
<td>Vacant tone</td>
</tr>
<tr>
<td>Speed dial memory</td>
<td>Vacant tone cadence modification</td>
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<td>Statistics period</td>
<td>Vacant tone frequencies modification</td>
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<td>Subnet mask</td>
<td>Vacant tone level modification</td>
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<td>Suspended calls cleared</td>
<td>VMail (Voice Mail)</td>
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<td>System language</td>
<td>Voice mail quota full message</td>
</tr>
<tr>
<td>System name</td>
<td>Voice mail quota full message download</td>
</tr>
<tr>
<td>System daylight saving</td>
<td>Voice mail quota full message remove</td>
</tr>
<tr>
<td>System time</td>
<td>Voice mail quota full message upload</td>
</tr>
<tr>
<td>System time zone</td>
<td>Voice mail receive</td>
</tr>
<tr>
<td>System parameter upload</td>
<td>Voice mail receive message</td>
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<tr>
<td>Tie line</td>
<td>Voice mail receive message download</td>
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<tr>
<td>Trunk call ring tone</td>
<td>Voice mail receive message remove</td>
</tr>
<tr>
<td>Trunk call ring cadence modification</td>
<td>Voice mail receive message upload</td>
</tr>
<tr>
<td>Trunk call ring tone cadence modification</td>
<td>Voice mail send</td>
</tr>
<tr>
<td>Trunk call ring tone frequencies modification</td>
<td>Voice mail send message download</td>
</tr>
<tr>
<td>Trunk call ring tone level modification</td>
<td>Voice mail send message remove</td>
</tr>
<tr>
<td>Upload Xymphony</td>
<td>Voice mail send message upload</td>
</tr>
<tr>
<td>User authorizations</td>
<td>Voice mail send message download</td>
</tr>
<tr>
<td>User parameter upload</td>
<td>Voice mail send message remove</td>
</tr>
<tr>
<td>User pool</td>
<td>Voice message on alerting</td>
</tr>
<tr>
<td></td>
<td>Voice message on setup</td>
</tr>
</tbody>
</table>
SYSTEM SYNCHRONIZATION

The synchronization algorithms in the X1 switching system provide two basic modes of synchronization - with external synchronization source and with internal clock.

In case of external synchronization source, input for reference clock signal with 2048 kHz and signal, amplitude and impedance according to ITU-T Rec. G.703, that is E1 interface.

The X1 system may be programmed such that;

• It may be the master (uses its own internal clock), or
• It may be a slave and tries to synchronize an external clock from one or more E1s and if it fails (master clock is not available), it passes to the free running mode

In case of external synchronization source: with reference to jitter, wander and slip probability, the synchronization subsystem meets ITU-T Rec. G.823 and 6.822

In case of internal clock: the synchronization subsystem includes a clock (clocks) with long- term stability better than $5 \times 10^{-5}$, acc. to ITU-T Rec. G.812, MTBF for the internal clock (with stand-by module) is 100 years.

TONES

The tones are applied to inform the subscriber about different states of the subscriber line, network situations or events in the call set-up phase, during conversation or when subscriber procedures are used to activate, deactivate, interrogate or invoke a subscriber service. The tones are applied in conformity with ITU-T Recommendation Q.35. The levels of the tones are stated in dBm0 at the exchange in which the tone is inserted. The harmonic distortion of all tones is $<1\%$. The accuracy of the timing is better than 10%.

• Cadences,
• Levels, and
• Frequencies

of the tones are programmable without interrupting the operation of the X1.

Some editable tones are:

• "Dial tone" : It is normal dial tone when the user is in idle state
• "Diversion activated" : It is tone when unconditional call diversion function is active
"Authorization modified": It is tone when allowed call class (user) is different from the allowed call class (system). That is the user modified his authorization level for password-dialing.

"Delayed hot line": It is tone when delayed hot line service is active.

"Confirmation": It is confirmation tone when a service call is accepted by the system.

"Vacant": It is tone when dialed digits are not sufficient to route the call.

"Busy": It is tone when the destination is busy.

"Overflow": It is tone when there are no system resources to route the call or to activate a service.

"Howler": It is tone when the first digit dialing time-out occurs.

"Local call ring 1": It is ring and rinback tone type-1.

"Local call ring 2": It is ring and rinback tone type-2.

"Local call ring 3": It is ring and rinback tone type-3.

"Local call ring 4": It is ring and rinback tone type-4.

"Trunk Call ring": It is ring and rinback tone for incoming trunk calls.

"Secondary dial tone": It is tone when a call is hold.

"Multiple entry": It is tone when using multiple-entry services like changing status of several room at once.

"Multiple hold": It is tone when holding multiple calls on.

"Reminder service": It is tone when reminder service is active.

"Campon": It is tone for camped calls.

ANNOUNCEMENTS

The X1 system provides up to 100 different announcements (or system messages), each of them with maximum duration of 10 minutes.

Some of these messages are used for predefined applications such as:

- Auto-attendant scenario
- Music on hold
- Wake up call
- Voice Mail (VMail) scenario

"Wav" wave files to be transferred from PC to the system should be 16 bits, 8 kHz, mono in Windows waveform audio format (wave).

- Message-00 Welcome1: First greeting message in Auto-attendant applications
- Message-01 WelcomeAlt: Alternative greeting message (perhaps in another language)
- Message-02 Busy: Busy message instead of the busy tone, caller can dial a new number, Applicable only for DC loop trunks if welcome message is played
- Message-03 BusyAlt: Alternative busy message (perhaps in another language), caller can dial a new number, Applicable only for DC loop trunks if welcomeAlt message is played
- Message-04 Alerting: Alerting message instead of an alerting (ringback) tone
- Message-05 AlertingAlt: Alternative alerting message (perhaps in another language)
- Message-06 AlertingIni: Alerting message for incoming calls through DID trunks
- Message-07 Reminder: Reminder (wake-up) call message
- Message-08 Hold: Message (or music) on hold, played repeatedly
- Message-09 VmailSend: Voice Mailbox greeting message
- Message-10 VmailSendAlt: Alternative Voice Mailbox greeting message (perhaps in another language)
- Message-11 VmailReceive: Voice Mailbox receive message (when mailbox is accessed to listen left messages)
- Message-12 VmailPending: Notification message when there are pending Voice Mails in the mailbox
- Message-13 Park: Message (or music) on park, played repeatedly
- Message-14 Welcome2: Second greeting message in Auto-attendant applications
- Message-15 Welcome3: Third greeting message in Auto-attendant applications
- Message-16 VmailQuotaFull: Notification message when personal Voice Mailbox is full (out of quota)
• Message-17 CamponToRoute: Notification message played when all ports in route are busy and the call is waiting for an outgoing available port.

• Message-18 NoAnswer: If B party do not answer, caller can dial a new number, applicable only for DC loop trunks if welcome message is played

• Message-19 NoAnswerAlt: If B party do not answer, caller can dial a new number, applicable only for DC loop trunks if welcomeAlt message is played

The remaining system messages are like ready-to-use resources. Using routing criteria tables, callers may be routed to these system messages. System administrators may decide on criteria when to play a system message. Some typical applications may be:

• dialing of a non-existing code,
• dialing of a vacant/unallocated number,
• fault or congestion in certain direction,
• restriction of use of services due to non-paid bills,
• wake-up service is activated,
• follow me or follow me no answer services is activated.
• reserved ones

FAULT LOCALIZATION
The X1 is capable to perform detection, localization and reporting of failures in the switching system, connected equipment and interfaces. Furthermore it is capable of detecting and reporting overload conditions in the system, at the interfaces and the connected equipment.

When a fault or an out-of-limit condition is detected, the X1 switching system stores all relevant fault information. When the fault or out-of-limit situation is resolved, the actions to minimize the effect are canceled.

The software design avoids propagation of faults by programs or data contamination.

REAL TIME EVENT MONITORING AND PROTOCOL ANALYSIS
Introduction:
With the Real-time Protocol Analysis feature in Telesis X1 systems, it is possible to monitor any TDM interface by decoding signaling or protocol existing on it.

Man Machine Interface:
The Telesis XPort utility is used as the MMI (Man Machine Interface) for the integrated Real-time Protocol Analyzer.

Capabilities:
The signaling (protocol) analyzer of the Telesis X1 system is capable of analyzing:

• Call Control (CC) primitives
• E1 Layer-1 (physical layer) events
• A-B-C-D bits on CAS E1
• DTMF tones
• MFR1 tones
• MFCR2 tones
• ANI (Automatic Number Identification) request
• ANI response
• Dial pulses (decadic pulses)
• Multi-frequency shuttle tones (Pulse shuttle, R1.5)
• Multi-frequency packet tones (Pulse packet 1, 2, 3a, 3b)
• Line signaling on E&M interfaces
• Basic Rate ISDN - BRI Layer 1 (physical layer)
• Basic Rate ISDN - BRI Layer 2 (data link layer)
• Basic Rate ISDN - BRI Layer 3 (network layer)
• Primary Rate ISDN - PRI (DSS1 and QSIG) Layer 1 (physical layer)
• Primary Rate ISDN - PRI (DSS1 and QSIG) Layer 2 (data link layer)
• Primary Rate ISDN - PRI (DSS1 and QSIG) Layer 3 (network layer)
CC primitives (CCPs) are the communication between Layer 3 Port Control and Call Control Engine (CCE). The control and switching hardware in a system together with its operating firmware, the Xymphony, form the CCE. The CCE provides the functionality to initiate, manage and terminate calls through the interfaces in a Telesis X1 system. In this communication, the required controls for call setup, call proceeding, and call ending occur free from the signaling type.

The decoding of CCPs is done according to the ITU-T Q.931 standard. Call modeling of all calls in Telesis X1 systems is based on ISDN standards.

A call has two sides, one is incoming (or originating) side and the other is outgoing (or termination) side. During a call control, CCPs are sent from the CCE to Layer 3 and from Layer 3 to the CCE.

The CCPs are formed of Indications, Requests and Responses. CCPs define which algorithms or operations should be executed to set up, proceed or end calls; regardless of the port's type or physical properties.

The decoded CCPs are detailed in ITU-T Q.931. Furthermore, Telesis also added some more primitives to make the monitoring and analysis complete such that:

- Call Control primitives from Layer 3 Port Control to the CCE:
  - SetupIndication
  - InfoIndication as: DialPulse, Dtmf, MF R1, MFC R2, O.BAND, From pool, COMPANION
- RingStopIndication
- AniStartIndication
- AniStopIndication
- BPartyOnHookIndication
- ReAnswerIndication
- InrIndication
- InfIndication

- Call Control primitives from the CCE to Layer 3 Port Control
  - RejectRequest
  - DisconnectRequest
  - MoreInfoRequest
  - ProceedingRequest
  - ReleaseRequest
  - SetupRequest
  - AlertingRequest
  - ConnectRequest
  - ConnectResponse
  - HoldResponse
  - HoldRejectRequest
  - NotifyRequest as: CallisDiverting, DiversionActivated, RemoteHold, RemoteRetrieval, AlertingTransfer, ActiveTransfer, ConfEstablished, ConfDisconnected, UserSuspended, UserResumed, BearerChange, 3PtyRemoved
  - RetrieveResponse
  - RetrieveRejectRequest
  - InfoRequest
  - ProgressRequest
  - SuspendResponse

- SuspendRejectRequest
- ResumeResponse
- ResumeRejectRequest
- SuspendRequest
- ResumeRequest
- ChargePulseRequest
- VoiceMessageRequest
- MaliciousCallRequest
- RingStartRequest
- RingStopRequest
- AniStartRequest
- AniStopRequest
- BPartyOnHookRequest
- ReAnswerRequest
- InrRequest
- InfRequest
- ConferenceResponse
- EncryptedMediaRequest

The decoding information on the monitor window may be displayed in various forms, such as binary, hexadecimal, and mnemonic explanations.
ANALOG SUBSCRIBERS

The analog subscriber interfaces can drive loops up to 3,000 ohm resistance (including the terminal equipment) and is capable of sending Caller ID information according to the ETSI ETS 300 659-1 standard as an FSK signal burst where the data transmission occurs during the first long silent period between two ring patterns. Line feeding voltage is 48 VDC. The subscriber interfaces have over-voltage protection, conforming to ITU-T K.20/K.21 recommendations. Additional primary protection devices to those residing in the main distribution frame may also be employed for further protection.

DIGITAL SUBSCRIBERS

Telesis digital sets connect to 4B+D ISDN interfaces (digital subscriber interfaces) in Telesis X1 systems. Each Telesis digital set is operated and powered by a single pair of wires.

ANALOG DC LOOP TRUNKS

The analog DC loop trunks can detect line DC feed, polarity reversal, and 12 or 16 kHz charge pulses and are capable of receiving Caller ID information transmitted according to the ETSI ETS 300 659-1 standard as an FSK signal burst. Over-voltage and over-current protection on analog trunks conforms to ITU-T K.20/K.21 recommendations. Additional primary protection devices to those residing in the main distribution frame may also be employed for further protection.

ANALOG E&M TRUNKS

The analog E&M trunk card has four line circuits that satisfy conditions for AT&T Type-V (or US Type-V, Type-5) connections. Each port may individually be set to either a two- or four-wire interface by appropriate configuration of the on-the-board jumpers. The Xymphony operating software supports many different line and register signaling types over the E&M lines. An additional detector on the M-wire checks the DC feed on the line for faulty conditions at far-end equipment. The analog E&M trunk card is available for 600 ohm resistive reference impedance. Each E&M port can be programmed as incoming, outgoing, both-way trunk, or unavailable.
**Signaling on analog E&M trunk lines:**
- Pulsed line with decadic-address signaling
- Continuous line with DTMF, MFC-R2, MF-R1 register signaling
- Signaling types widely employed in CIS countries:
  - ANI (Automatic Number Identification) detection and query
  - Address signaling
    - Dial Pulse
    - MFC-R1.5
    - Pulse Packet 1
    - Pulse Packet 2
    - Pulse Packet 3a
    - Pulse Packet 3b
  - Line signaling
    - CL-1B
    - OCL-1B
    - TCL-1B

**EURO ISDN BRI**
The Basic Rate ISDN types are S0 or Uk0. Point-to-point and point-to-multipoint connection schemes are possible. The line code on U interfaces is 2B1Q.

**E1 INTERFACES**
The 2.048 Mbit E1 digital interfaces can connect to 120 ohm balanced terminations. The clock of a Telesis X1 system can be programmed to synchronize with any of the E1 interfaces that may be present in the system or it may run freely. The line code is programmable as AMI or HDB3. Cyclic redundancy check (CRC4) can be enabled or disabled for each E1 interface individually. The direction of each channel of every E1 present in the system can be programmed as incoming, outgoing, both-way, or unavailable and therefore may yield fractional E1 connections.

**Signaling on 2.048 Mbit E1 (ITU-T G.703) interfaces:**
- Channel-Associated Signaling (CAS)
  - Single-bit E&M emulation
  - Two-bit ITU-T R1
  - Two-bit ITU-T R2
- Many variations of signaling types (such as CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM) widely employed in CIS countries
- Common-Channel Signaling (CCS)
  - DSS1 (Euro-ISDN in the TE direction)
  - DSS1 (Euro-ISDN in the NT direction)
  - ECMA QSIG (in the TE direction)
  - ECMA QSIG (in the NT direction)
  - ITU-T Signaling System No.7 ISUP
  - V5.2 LE and AN protocol

**SS No.7 SIGNALING**

**Standards:**
Telesis X1 systems support ISUP ETSI EN 300 356 version 3 and ITU-T recommendations.

**Applications:**
Telesis X1 systems are integrated systems that include both a voice switch and an SS7 switch. A Telesis X1 system may serve as two types of signaling points in an SS7 network:
- SSP (Service Switching Point)
- STP (Signal Transfer Point)

As an SSP, the Telesis X1 system responds to all calls bound for known signaling points by looking at its routing table to determine how to route each call and then sending a message to the destination using the ISUP (ISDN User Part) protocol.

A unique feature of the Telesis X1 system is that multiple SPs (Signaling Points) may be defined in a single system. So, several OPCs (Originating Point Codes) may exist in the same system. This results in the combining of both SSP and STP functions in a single Telesis X1 system.

As an STP, the Telesis X1 system acts as a router or gateway in an SS7 network. ISUP messages are routed based on Global Title (dialed digits) or DPC (Destination Point Code) or Calling Party Information Element parameters.

With Telesis X1’s OPC translation feature, ISUP traffic flowing in a private SS7 network may be routed to the public SS7 network.
Layers and Options:

The physical layer (MTP1, Message Transfer Part-1) of SS7 signaling in Telesis X1 systems is an E1 (ITU-T G.703) interface. 120 ohm balanced terminations are supported on any E1. The line code is HDB3 or AMI programmable. Options for the data link layer (MTP2) include programmable basic error correction methods or PCR (Preventive Cyclic Retransmission). For the network layer (MTP3), SLTM (Signaling Link Test Message) periods are set as desired, and acknowledgment of test messages (SLTA) is automatic. OPC and DPC as well as CIC (Circuit Identification Code) prefixes can also be set easily. Several parameters in ISUP messages are programmable. Some of these are:

- Calling-party number
- Calling-party category
- Calling-party nature of address
- Calling-party numbering plan
- Calling-party presentation indicator
- Calling-party screening indicator
- Called-party number
- Called-party numbering plan
- Called-party NOA-Nature of Address (fixed)
- Called-party NOA-Nature of Address (automatic by analyzing the called party number)
- Subservice
- Hop counter initial value
- Satellite indicator
- Echo control device indicator
- Load Sharing

Physical Links:

Telesis X1 systems support both terrestrial and satellite links. For terrestrial links, a basic error detection/correction function is used. Due to the delay arising in satellite links, the PCR (Preventive Cyclic Retransmission) error correction function is used.

Link and Linkset Configuration:

The link is an existing E1 interface on the Telesis X1 system. Any 64 kbps channel on an E1 (except channel 0) is programmable as the signaling path. Several links may be grouped into a linkset sharing a common signaling path. Each linkset has its own OPC and DPC. In the same Telesis X1 system, there may be several linksets, each having an individual OPC. Furthermore, selected channels of every E1 link can be programmed as unavailable for outgoing traffic, thereby yielding fractional E1 connections.

Route and Routeset Configuration:

In a Telesis X1 system, each link and/or linkset may have its own route number. It is possible to define numerous distinct routes. A given route to a particular destination and its accompanying alternate routes are grouped in a routeset. Each route in a routeset has a priority order. Routing to the next priority alternate route is possible in the event that a route becomes unavailable. Furthermore, it is possible to define which release causes stated in ISUP messages will lead to the alternate routes being used.

Message Routing:

ISUP message routing is based on dialed digits (Global Title). As an SSP, the Telesis X1 system converts dialed digits from a subscriber's line to SS7 signaling messages. As an STP, the Telesis switch may realize routing according to:

- Dialed digits
- DPC
- Calling-party number
- Category of calling party
- NOA of calling party
- Numbering plan of calling party
- Presentation status of calling party
- Screening status of calling party
ISDN SIGNALING

Standards:
Basic and primary rate ISDN interfaces are according to ETSI and ITU-T standards. QSIG ISDN D channel protocol is an ECMA (European Computer Manufacturers Association) derivative.

Types of Interfaces:
For ISDN DSS1 Basic Rate (BRI) in X1 systems, the types of interfaces supported are:

- S0 interface
- Uk0 interface

Both point-to-point and point-to-multipoint connection schemes are provided.

For ISDN Primary Rate (PRI) in X1 systems, the types of interfaces supported are:

- DSS1 Euro ISDN TE (terminal side)
- DSS1 Euro ISDN NT (network side)
- ECMA QSIG TE (terminal side)
- ECMA QSIG NT (network side)

Layers and Options:
The physical layer of primary ISDN access in Telesis X1 systems is an E1 (ITU-T G.703) interface. 120 ohm balanced terminations are supported on any E1. The line code is HDB3 or AMI programmable. CRC4 may be enabled or disabled.

Basic Rate ISDN interfaces are S0/T0 and Uk0. The connector type for S0/T0 is RJ45. Point-to-point and point-to-multipoint connection schemes are possible. With point-to-multipoint connections, both short and extended passive buses are supported. The line code on U interfaces is 2B1Q.

Several parameters in ISDN messages are programmable. Those are:

- Calling-party number
- Calling-party category
- Calling-party type of number
- Calling-party numbering plan
- Calling-party presentation indicator
- Calling-party screening indicator
- Called-party number
- Called-party numbering plan
- Called-party type of number (fixed)
- Called-party type of number (automatic by analyzing the called-party number)

ISDN Supplementary Services:
The following ISDN supplementary services are supported:

- 3PTY
- AOC-D/E
- CCBS
- CCNR
- CFU
- CFNR
- CLIP
- CLIR
- COLP
- COLR
- ECT
- DDI
- HOLD
- MCID
- MSN
- UUS
Route and Routeset Configuration:

In a Telesis X1 system, each basic and/or primary ISDN access may have its own route number. It is possible to define up to 128 distinct routes. For primary access, a given route to a particular destination and its accompanying routes are grouped in a routeset. Each route in a routeset has a priority order. Routing to the next priority alternate route is possible in the event that a route becomes unavailable. Furthermore, it is possible to define which release causes stated in Layer-3 messages will lead to the alternate routes being used.

For accessing PRI, the channel selection type is programmable as ascending, descending, or cyclic. Disabling outgoing calls from certain channels also yields fractional E1 use.

Routing:

A Telesis X1 system may realize routing calls from ISDN interfaces according to:

- Dialed digits
- Calling-party number
- Category of calling party
- TON (Type of Number) of calling party
- Numbering plan of calling party
- Presentation status of calling party
- Screening status of calling party

V5.2 PROTOCOL

V5.2 Protocol In General:

The V5.2 protocol stack is used for the connection of an Access Network (AN) to a Local Exchange (LE). It enables for various access methods, like analog telephone access and ISDN basic rate access. A V5.2 interface may use up to 16 E1 interface links. For analog access, the PSTN user port is converted into a functional part of the V5.2 protocol for signaling to the AN side. In principle, the V5.2 interface uses a concentrator type, whereby up to 16 E1 links, each carrying 30 channels (480 channels in all), may serve up to 1,920 subscribers with ¼ concentration. There is no limit in principle, to the number of V5.2 interfaces that can exist between the access network and the local exchange. A channel is allocated to a subscriber dynamically when he or she makes or receives a call.

V5.2 includes a protection protocol, allowing for redundancy in the signaling. Usually, the signaling for the 16 E1s is carried on one channel on one link (the primary link). If the signaling link fails, a secondary link is set to take over. This is similar in principle to SS7’s multiple point codes. In order to support more traffic through dynamic allocation of channels, the V5.2 protocol has several additions:

- A bearer channel connection (BCC) protocol establishes and de-establishes bearer connections required on demand, identified by the signaling information, under the control of the Local Exchange.
- A link control protocol is defined for the multi-link management to control link identification, link blocking, and link failure conditions.
- A protection protocol, operated on two separate data links for security reasons, is defined to manage the protection switching of communication channels in case of link failures.

Access Network:

The term AN (access network) refers to the network between the local exchange (or Central Office) and the subscriber. In many countries, this network is still predominantly made up of the copper-cable-based point-to-point connections. However, conventional point-to-point copper cabling has some limitations:

- It offers limited bandwidth, which is difficult to overcome.
- Inflexibility: both in time and types of service provided.
- Due to star topology (from the exchange to the subscriber), reliability is limited.
- Installation time is long.
- It is maintenance intensive due to possible cable damage and thus costly.
- Largely passive, making it difficult to manage.
- Loop length limitations (~ 10 km).
- Uneconomical in remote, isolated areas with low telephone densities.
- Prone to electromagnetic interference.

To overcome the above-mentioned issues, several vendors developed AN (access network) technologies, almost all which support the V5.2 protocol in connecting to the voice-switching Local Exchanges. Some examples of new technologies based on AN with V5.2 protocol are mentioned below:
DSLAM (Digital Subscriber Line Access Multiplexer) or Broadband Access Network Technology

A family of technologies that have begun to transform the narrow-band copper access network into a broadband network is the xDSL family of technologies. The term DSL, or digital subscriber line, refers to the modem that, when connected at either end of a normal twisted-wire pair line, converts it into a digital line capable of handling data rates well into broadband. By using higher frequencies, DSL technologies enable much higher speeds over the twisted-pair lines.

WLL (Wireless Local Loop) Technology

Bringing telephone access to subscribers in remote and isolated areas with low telephone densities by laying down a copper network may be uneconomical. WLL technology makes installation in such areas easy and more economical by utilizing radio to access subscriber premises. WLL equipment connected to Telesis system via V5.2 protocol to access rural areas via radio

DLC (Digital Loop Carrier) Technology

Optical fibers, clearly the preferred technology for transmission media, are beginning to find their place in the subscriber's loop. The tremendous advantages in terms of the information capacity of fiber, its small weight, and its size compared to copper cable are making it a very attractive technology with which to replace copper in the subscriber loop. To access to the same number of subscribers that one fiber cable can carry, we would need hundreds of twisted-wire copper cables, which makes DLC the preferred technology in many installations.

Standards:

V5.2 protocol implementation in Telesis X1 systems follows ETSI Standard EN 300 347 (Version 2). The implemented protocol covers both LE (Local Exchange) and AN (Access Network) protocols and supports analog telephone access (PSTN) and ISDN basic-rate access.

Applications:

Telesis X1 systems support:

- Both the Access Network (AN) and Local Exchange (LE) sides
  - Common control protocol
  - Protection control protocol
  - Link control protocol
  - Port control protocol
  - BCC protocol
  - ISDN messaging
  - PSTN messaging
  - Protocol conversion between the V5.2 LE protocol and any other signaling system provided by a Telesis X1 system
  - ETSI FSK Caller ID transmission from an LE to an AN
  - Charge-pulse transmission from an LE to an AN
  - Mixed use of V5.2 LE and AN protocols and other signaling systems
  - Flexible assignment of selected subscribers to a dedicated AN while allowing the other ports (subscribers and trunks) to operate as usual.

For a V5.2 interface, the mapping of the logical (C-path) to the physical C-channel is stored in timeslot 16 of the primary link as default. This default profile cannot be modified while the V5.2 interface is in operation. If the V5.2 interface consists of more than one E1 link, then the protection protocol exists for both the primary and the secondary link in parallel. Time slots 16 in the primary link and the secondary link are always physical C-channels. If protection switching occurs, the C-path is restored at timeslot 16 of the secondary link. V5.2 protocol implementation in Telesis X1 systems allows:

- assignment of 16 E1 links to one V5.2 interface (if the system capacity allows)
- assignment of Link Ids
- assignment of Interface ID
- assignment of logical C-channel ID
- setting a primary link (which carries the C-path by default)
- setting a secondary link

Frequently, service providers are hesitant to invest in a new switch in remote and isolated areas with low telephone densities. An attractive alternative is to install remote units in such areas that can act as part of an already existing switch. This solution offers ease of maintenance and operation as well as cost effectiveness. V5.2 protocol implementation in the Telesis X1 systems enable them to serve as the remote units of any exchange that conforms to ETS 300 047 V5.2 protocol standards.

Versatility:

One unique feature of Telesis X1 systems is their versatility; the V5.2 LE protocol and the V5.2 AN protocol and other signaling systems can co-exist within one system, such that:
• Some subscribers may constitute an AN of a dedicated local exchange
• Some E1 interfaces may utilize the V5.2 LE protocol so that other ANs may be connected
• Some subscribers and trunks may continue to operate in conventional mode

Networking Telesis X1 systems:

Networking Telesis X1 systems is possible by utilizing the V5.2 protocol. Note that, because switching is actively done on the LE side, the number of bearer channels between the LE and the AN should be planned carefully in accordance with the expected telephone traffic. Erlang loss formula could be used as a service quality target. To illustrate, through a 60-channel (2 x E1) capacity V5.2 interface, total erlang traffic capacity is 44.75, with a lost-call probability of 0.005 at random traffic. Consequently, for a 250-subscriber capacity X1 AN with a 60-channel V5.2 interface, erlang per subscriber is almost 0.2.

Options:
The physical layer of the V5.2 protocol in Telesis X1 systems is an E1 (ITU-T G.703) interface. 120 ohm balanced terminations are supported on any E1 link. The line code is HDB3 or AMI programmable. CRC4 error checking may be enabled or disabled. The Telesis X1 supports up to 16 AN and/or LE V5.2 interfaces, each using up to 16 x E1 interface links, provided that the E1 capacity of the X1 allows it.

R2 SIGNALING

Standards and interfaces:
Telesis X1 systems support signaling for two-bit R2 line / MFCR2 interregister according to ITU-T recommendations. Telesis X1 switching systems support signaling for MFCR2 interregister on analog Type-V E&M interfaces.

Options:
The physical interface for single-bit and two-bit signaling is an E1 (ITU-T G.703). 120 ohm balanced terminations are supported on any E1 link. The line code is HDB3 or AMI programmable. CRC4 may be enabled or disabled. Each E1 channel can be programmed as incoming, outgoing, bothway trunk, or unavailable.

Each analog E&M trunk board has four line circuits that satisfy the conditions for Type-V connections. Each port may individually be set to either a two- or four-wire interface with appropriate configuration of the on-the-board jumpers. An additional detector on the M-wire checks the DC feed on the line to detect faulty conditions in far-end equipment. The analog E&M trunk card is available for 600 ohm resistive reference impedance. Each E&M port can be programmed as incoming, outgoing, bothway trunk, or unavailable.

The following options and parameters are available on R2 signaling interfaces of Telesis X1 systems:
• Automatic calling-party category translation for calls routed into and out of the R2 signaling system
• ANI (asking and replying to calling-party number request)
• Programmable calling-party categories

Route and Routeset Configuration:
In a Telesis X1 system, each E&M interface and E1 channel may have its own route number. It is possible to define numerous distinct routes. A given route to a particular destination and its accompanying routes are grouped in a routeset. Each route in a routeset has a priority order. Routing to the next priority alternate route is possible in the event that a route becomes unavailable.

Routing:
A Telesis X1 system may realize routing according to:
• Dialed digits
• Calling-party number
• Category of calling party

R1 SIGNALING

Standards and interfaces:
Telesis X1 systems support signaling for two-bit R1 line / MFR1 interregister according to ITU-T recommendations. Telesis X1 systems support signaling for MFR1 interregister on analog Type-V E&M interfaces.
Options:
The physical interface for single-bit and two-bit signaling is E1 (ITU-T G.703). 120 ohm balanced terminations are supported on any E1. The line code is HDB3 or AMI programmable. CRC4 may be enabled or disabled. Each E1 channel can be programmed as incoming, outgoing, bothway trunk, or unavailable.

Each analog E&M trunk board has four line circuits that satisfy the conditions for Type-V connections. Each port may individually be set to either a two- or four-wire interface by appropriate configuration of the on-the-board jumpers. An additional detector on the M-wire checks the DC feed on the line to detect faulty conditions in far-end equipment. The analog E&M trunk card is available for 600 ohm resistive reference impedance. Each E&M port can be programmed as incoming, outgoing, bothway trunk, or unavailable.

Route, Routeset Configuration, Routing:
In a Telesis X1 system, each E&M interface and E1 channel may have its own route number. It is possible to define numerous distinct routes. A given route to a particular destination and its accompanying routes are grouped in a routeset. Each route in a routeset has a priority order. Routing to the next priority alternate route is possible in the event that a route becomes unavailable. The Telesis X1 system realizes routing according to the dialed digits or calling party number.

CAS SINGLE BIT E&M EMULATION
Types:
Telesis X1 systems support single-bit CAS E&M emulation with:
- Immediate-start continuous line / pulsed address signaling
- Wink-start continuous line / pulsed address signaling
- Immediate-start continuous line / DTMF address signaling
- Wink-start continuous line / DTMF address signaling
- Immediate-start continuous line / MFCR2 interregister signaling
- Wink-start continuous line / MFCR2 interregister signaling
- Immediate-start continuous line / MFR1 interregister signaling
- Wink-start continuous line / MFR1 interregister signaling
- Pulsed line / pulsed address signaling

Options:
The physical interface for CAS single-bit E&M emulation is an E1. 120 ohm balanced terminations are supported on any E1. The line code is HDB3 or AMI programmable. CRC4 may be enabled or disabled. Each E1 channel can be programmed as incoming, outgoing, bothway trunk, or unavailable. This also yields fractional E1 connections.

For single-bit pulsed or continuous line signaling, the idle states of the A,B,C, and D bits can be set as desired. The signaling bit is also programmable as any one of A,B,C, or D. The transmit and receive gains in any E1 voice channel may be increased or decreased.

Many signaling periods and timers for signal generation and detection are field programmable, such as:
- Timer to receive the first digit
- Interdigit timer
- Release timer
- Disconnect timer
- Seizure-acknowledge timer
- Seizure-pulse duration
- Seizure-acknowledge pulse duration
- EOS (End of Selection) pulse duration
- Answer-pulse duration
- Clear-pulse duration
- Charge-pulse duration
- Make/break ratio and periods for pulse dialing
- Wink-pulse duration

Both unilateral and bilateral call clearing are applicable.

Route, Routeset Configuration, Routing:
In a Telesis X1 system, each E1 channel may have its own route number. It is possible to define up to 128 distinct routes. A given route to a particular destination and its accompanying routes are grouped in a routeset. Each route in a routeset has a priority order. Routing to the next priority alternate route is possible in the event that a route becomes unavailable. The Telesis X1 system realizes routing according to the dialed digits.
CIS RUSSIAN SIGNALING TYPES

Types:
Telesis X1 systems support the following types of signaling for Russian (and CIS) PSTN:

- Two-bit channel-associated signaling
- Single-bit channel-associated signaling
- Two-wire / four-wire analog signaling
- Single-frequency (1VF) signaling
- Multifrequency Signaling

Trunk types within Telesis X1 systems are local trunks, toll-connecting trunks, and toll-switched trunks for CIS-Russian PSTN. For all such trunk types, both two-bit and single-bit channel-associated signaling types are applicable when the interface is E1. If the interface is analog, then analog mapping of the single-bit channel-associated signaling type is applicable. For local (SL, CL) and toll-connecting (ZSL, OCL) trunks, available address- and register-signaling options are:

- Pulse
- Multifrequency shuttle (MFC R1.5)
- Multifrequency packet (1, 2, 3a, 3b)

For toll-switched trunks (SLM, TCL), available address- and register-signaling options are:

- Pulse
- Multifrequency packet (3a, 3b)

In addition, line signals can be transmitted/received as voice frequencies. Telesis X1 Switching Systems also provide 1VF signaling for local, toll-connecting, and toll-switched trunks.

Options:
The digital interface for two-bit and single-bit CAS is E1. 120 ohm balanced terminations are supported on any E1. The line code is HDB3 or AMI programmable. CRC4 may be enabled or disabled.

The analog interfaces for local, toll-connecting, and toll-switched trunks are:

- Two-wire/four-wire Type V analog E&M

The following options and parameters are available in CIS-Russian signaling interfaces of Telesis X1 systems:

- Automatic calling-party category translation for calls routed into and out of the CIS signaling system
- ANI generation (request)
- ANI response
- Programmable calling-party categories
- Several signaling periods and timers for signal generation and detection
- Both unilateral and bilateral call clearing

For single-bit CAS line signaling in the E1, the idle state of the A,B,C, and D bits can be set as desired. The signaling bit is also programmable as any of A,B,C, or D. The transmit and receive gains in any E1 voice channel may be increased or decreased.

Route, Routeset Configuration, Routing:
In a Telesis X1 system, each E1 channel may have its own route number. It is possible to define up to 128 distinct routes. A given route to a particular destination and its accompanying routes are grouped in a routeset. Each route in a routeset has a priority order. Routing to the next priority alternate route is possible in the event that a route becomes unavailable. A Telesis X1 system may realize routing on calls coming from CIS interfaces according to:

- Dialed digits
- Calling-party number
- Category of the calling party

H.323 VOIP PROTOCOL

Standards:
Applications:

Telesis X1 systems integrate both packet and circuit switching technology. A Telesis X1 system featuring an integrated gatekeeper provides an economical way for administrators to manage a central database of phone numbers without the expense of a separate-box gatekeeper solution. Telesis X1 systems support numerous H.323 users (entities) which can be terminals, gatekeepers or gateways. Number/IP translation is performed through Telesis advanced routing algorithm. Together with the integrated gatekeeper, call authorization, call management, enhanced billing functions, flexible routing algorithms and extensive business telephony features make a Telesis X1 system serve as a feature-rich communication platform. Integrated gatekeeper allows calls to be placed direct or gatekeeper routed between H.323 entities in various circumstances.

Furthermore, Telesis X1 systems are with media proxying capabilities. The integrated media proxy provides a transit point for media (audio) streams between H.323 entities. The media proxy operates only for the integrated gatekeeper routed calls in some circumstances. While voice bridging distant offices over the IP, security of a VoIP call is guaranteed with the encryption (optional) of voice according to 256 bit AES (AES-256).

While connecting to the long distance call operator over the IP, the Telesis X1 system may register to an external gatekeeper of the operator as an option. With the advanced routing algorithms and alternate routing capability, TDM calls from a terminal equipment of the system may be routed to a selected operator over the IP or PSTN. Alternate routing capability provides automatic fall back to the PSTN if the IP network is unaccessible.

Layers and Options:
The physical layer of H.323 protocol in Telesis X1 systems is 10/100 BaseT Ethernet. Several ethernet and other properties for H.323 are programmable.

Audio Codecs:
A Telesis X1 system is equipped with well-known audio codecs featuring audio compression as well. Audio codec preference list and properties such as silence suppression (VAD-Voice Activity Detection), frame length are programmable for the system. Currently available codecs for VoIP calls are:

- G.711 (A and µ)
- G.723.1 (5.3kbps, 6.4kbps)
- G.729
- G.729A

Echo Cancellation:
An AT&T certified G.168 echo canceler meets and exceeds G.168-2002 standards. The echo canceler can operate with delays as high as 128msec. It is better than industry standard cancelers under the most important and difficult conditions like double-talk and the presence of background noise.

Route and Routeset Configuration:
In a Telesis X1 system, an H.323 endpoint may have its own route number. It is possible to define numerous distinct routes. A given route to a particular destination and its accompanying alternate routes are grouped in a routeset. Each route in a routeset has a priority order. Alternate routes may be H.323 endpoints or TDM (PSTN) lines. Routing to the next priority alternate route is possible in the event that a route becomes unavailable.

Call Routing:
As an IP-TDM gateway, the Telesis X1 system routes a call from the TDM network to the IP network according to:

- Dialed digits (called number)
- DPC
- Calling party number and information elements whenever available, such that:
  - Category of calling party
  - NOA of calling party
  - Numbering plan of calling party
  - Presentation status of calling party
  - Screening status of calling party

H.450 Supplementary Services:
Telesis X1 systems support various H.450.x series supplementary services in H.323 protocol stack, such as:

- H.450.1 Generic functional protocol for the support of supplementary services in H.323
• H.450.2 Call transfer supplementary service for H.323
• H.450.4 Call hold supplementary service for H.323

SIP VOIP PROTOCOL

Standards:
The applied standard is RFC 3261 SIP: Session Initiation Protocol.

Applications:
Telesis X1 systems integrate both packet and circuit switching technology. A Telesis X1 system featuring an integrated SIP registrar provides an economical way for administrators to manage a central database of phone numbers without the expense of a separate-box registrar solution.

Telesis X1 systems support numerous SIP users (entities) which can be user agents or registrars. Number/IP translation is performed through an advanced routing algorithm. Together with the integrated registrar, call authorization, call management, enhanced billing functions, flexible routing algorithms, and extensive business telephony features make a Telesis X1 system serve as a feature-rich communication platform.

While connecting to the long distance call operator over the IP, the Telesis X1 system may register to an external registrar of the operator as an option. With the advanced routing algorithms and alternate routing capability, TDM calls from a terminal equipment connected to the system may be routed to a selected operator over the IP or PSTN. Alternate routing capability provides automatic fall back to the PSTN if the IP network is unaccessible.

Layers and Options:
The physical layer of SIP protocol in Telesis X1 systems is 10/100 BaseT Ethernet. Several ethernet and other properties for SIP are programmable.

Audio Codecs:
Telesis X1 systems are equipped with well-known audio codecs featuring audio compression as well. Audio codec preference list and properties such as silence suppression (VAD-Voice Activity Detection), frame length are programmable for the system. Currently available codecs for VoIP calls are:

• G.711 (A and µ)
• G.723.1 (5.3kbps, 6.4kbps)
• G.729
• G.729A

Echo Cancellation:
An AT&T certified G.168 echo canceler meets and exceeds G.168-2002 standards. The echo canceler can operate with delays as high as 128msec. It is better than industry standard cancelers under the most important and difficult conditions like double-talk and the presence of background noise.

Route and Routeset Configuration:
In a Telesis X1 system, a SIP user agent may have its own route number. It is possible to define numerous distinct routes. A given route to a particular destination and its accompanying alternate routes are grouped in a routeset. Each route in a routeset has a priority order. Alternate routes may be SIP user agents or TDM (PSTN) lines. Routing to the next priority alternate route is possible in the event that a route becomes unavailable.

Call Routing:
A Telesis X1 system routes a call from the TDM network to the IP network according to:

• Dialed digits (called number)
• DPC
• Calling party number and information elements whenever available, such that:
  • Category of calling party
  • NOA of calling party
  • Numbering plan of calling party
  • Presentation status of calling party
  • Screening status of calling party
AES 256 MEDIA ENCRYPTION FOR VOIP CALLS

Introduction:

All Telesis X1 systems are complete voice communication systems, which combine various TDM interfaces and IP components. They are all-in-one solutions with integrated gatekeeper, softswitch capability, IP-TDM routing (gateway) functions, and numerous IP and traditional system features. Even though the media encrypting algorithm explained here is applicable for H.323 endpoint-to-endpoint connection too, it is recommended for H.323 endpoint-to-gatekeeper connection for further security.

The following paragraphs demonstrate algorithms applied for site-to-site communication in brief, such that:

- Two Telesis systems in each site
- Both systems are provided with necessary licenses for the VoIP media security and their parameters are set accordingly.

Secure Gatekeeper Registration:

Two Telesis systems share an account name and a secret, which is the password. One system, as an H.323 endpoint, registers to the gatekeeper of the other with the shared account name and the password. For the registration, H.225 RAS messages are exchanged between the two Telesis systems according to the H.235 Baseline Security Profile with or without integrity check. The baseline security profile provides basic security for endpoint-to-gatekeeper registration using the secure password-based HMAC-SHA1-96 hashing algorithm.

Baseline authentication: For H.323 endpoint-to-gatekeeper registration, RAS message authentication is according to H.235 Baseline Security Profile standards. This security service supports authentication of selected fields only, but does not provide full message integrity. The authentication-only security profile may be preferable for the messages traversing NAT/firewall devices. Hashing algorithm is the password-based HMAC-SHA1-96.

Baseline integrity: For H.323 endpoint-to-gatekeeper registration, RAS message authentication and integrity is according to H.235 Baseline Security Profile standards. This is a security combining both message integrity and the authentication. Hashing algorithm is the password-based HMAC-SHA1-96.

Encrypting the Media:

For encrypting the media, 256-bit Advanced Encryption Standard (AES-256) is used. AES-256 specifies a cryptographic algorithm using a symmetrical block cipher that can process data blocks of 128 bits with 256-bit chipper key (crypto key) which is agreed by Diffie-Hellman procedure. Audio samples are collected from the codec, they are encrypted, and inserted into the RTP payloads.

When the receiving side gets RTP payloads, the decrypting occurs.

A secure contact would be by generating and exchanging shared Diffie-Hellman half-keys. Diffie-Hellman master key for the AES-256 encryption is generated from the combination of the two shared half keys exchanged by two Telesis systems involved in a call.

Diffie-Hellman key exchange:

Telesis systems exchange Diffie-Hellman half keys using authentication based on H.235 Baseline Security Profile with or without integrity check. This prevents Man-in-the-Middle (MIM) attacks and communicating systems can be sure with whom they share the Diffie-Hellman half keys. Hash algorithm for H.235 Baseline Security Profile or H.235 Baseline Security Profile with integrity check is HMAC-SHA1-96. Exchange of HMAC-SHA1-96 hashed Diffie-Hellman halfs keys provides additional security.

Key exchange occurs during H323 call signaling (H.225) messaging between two systems for end-to-end communication. First call signaling message in both direction are used in key exchange. Setup message is used in forward direction. Setup Acknowledge, Call proceeding, Alerting or Connect message can be used in reverse direction. Since, the authentication keyed by the password, which is a secret in two systems, it may be open to MIM attacks if simple passwords are chosen. Telesis systems allow Diffie-Hellman half key exchange provided that a sufficiently long password is selected. In the following cases, the call fails before connect.

- Authentication failure
- Authentication but missing half key in Setup message
- Authentication but missing half key in one of Setup Acknowledge, Call proceeding, Alerting or Connect messages

Summary:

Security of VoIP communication between two Telesis systems is ensured with:

- A sufficiently long password
• Baseline Security Profile for RAS messaging for H.323 endpoint-to-gatekeeper registration
• Baseline Security Profile for Call Signaling for secure Diffie-Hellman key exchange.
• Exchange of HMAC-SHA1-96 hashed Diffie-Hellman half keys
• Cipher AES-256

H.323 AND SIP INTEGRATION

Description:
Telesis X1 systems support for SIP and H.323 protocols. Both protocols coexist on the same Telesis X1 system. SIP and H.323 calls may originate and terminate in the same system. Furthermore, Telesis X1 systems allow calls from SIP based devices to be routed to H.323 based devices and vice versa. With this interoperability, enterprises may have the ability to use both protocols in the same network.

General Capabilities:
Telesis X1 systems can register to both SIP registrar and H.323 gatekeeper at the same time. This allows address resolution of a Telesis X1 system from either side and results in flexibility for multi-path VoIP access applications. Furthermore, Telesis X1 systems may have both integrated H.323 gatekeeper and external gatekeeper registration capability at the same time. With this interoperability, enterprises may have the ability to use both protocols in the same network.

• SIP users to call SIP users in private address space
• SIP users to call H.323 users in private address space
• SIP users to call SIP entities in public network
• SIP users to call H.323 entities in public network
• SIP users to call legacy PSTN via TDM interfaces
• H.323 users to call SIP users in private address space
• H.323 users to call H.323 users in private address space
• H.323 users to call SIP users in private address space
• H.323 users to call H.323 entities in public network
• H.323 users to call H.323 entities in public network

Telesis X1 systems try to have the media transport directly between the connecting IP entities. However, in some cases like IP-TDM, TDM-IP, SIP-H.323, H.323-SIP calls, this is not possible or efficient. In such cases, switching and media proxying capabilities in Telesis X1 systems route calls from one side to the other.

X SIP VOIP PROTOCOL

Development:
xSIP (eXtended SIP) protocol has been developed by Telesis. The main purpose of its development is to make some value-added services in Telesis X1 systems to be applicable for VoIP calls too. Presently, Telesis offers two products with xSIP protocol. One is the ITS821 IP Phone, and the other is the XPhone IP Softphone. Both products bring the comfort of Telesis digital sets over IP. Almost all the functions of Telesis digital telephones are also applicable over IP. To illustrate, ITS821 and XPhone are with a busy display panel showing the current status (idle, busy, ringing, etc.) of 120 pre-programmed users in a Telesis X1 system.

General Services with the xSIP:
Telesis X1 systems support numerous xSIP users. Number/IP translation is performed through an advanced routing algorithm. Together with the integrated xSIP registrar, call authorization, call management, enhanced billing functions, flexible routing algorithms, and extensive business telephony features make a Telesis X1 system serve as a feature-rich communication platform.

Extended Services with the xSIP:
Some of the extended services with the xSIP protocol and xSIP capable phones are:
• accessing to missed calls list, dialed numbers list, incoming calls list, and directory stored in the registered Telesis X1 system,
• large LCD display with lots of information about the user and registered Telesis X1 system,
• playing, deleting voice messages and recorded conversations stored in the registered Telesis X1 system,
• selecting the ringer melody available in the registered Telesis X1 system,
• many programmable function keys,
• many programmable quick access keys and busy display panel (BDP),
• setting call forward unconditional,
• setting call forward busy,
• setting call forward no-reply,
• setting hot-line,
• activating call waiting,
• activating do not disturb,
• activating wake-up (reminder) service,
• observing firmware version of the registered Telesis X1 system,
• deflecting calls,
• activating call back,
• holding calls on,
• retrieving calls,
• transferring calls,
• tracing transferred calls (up to 4 calls),
• activating conference,
• recording the conversation bi-directionally (both calling and called party voices).
### PART NUMBERS (ordering codes)

<table>
<thead>
<tr>
<th>Ordering Code</th>
<th>Short Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.x1g.kabinet</td>
<td>19inch cabinet to install racks.</td>
</tr>
<tr>
<td>1.x1g.ykpk000</td>
<td>Side cover for the 19inch cabinet.</td>
</tr>
<tr>
<td>1.x1g.kbnbagl</td>
<td>Part to fix cabinets to each other.</td>
</tr>
<tr>
<td>1.x1g.eklerxx</td>
<td>Installation accessories.</td>
</tr>
<tr>
<td>1.x1g.kontakr</td>
<td>Main power distribution frame with battery protection.</td>
</tr>
<tr>
<td>1.x1g.prizeki</td>
<td>220 VAC mains socket group.</td>
</tr>
<tr>
<td>1.x1g.gecis1u</td>
<td>Auxiliary power distribution frame.</td>
</tr>
<tr>
<td>1.x1g.rak0psm</td>
<td>19inch rack for 4 Amp. AC-DC converters.</td>
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<td>1.x1g.urntyc</td>
<td>19inch rack for 30 Amp. AC-DC converters.</td>
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<td>2.x1a.urnmps4</td>
<td>4 Amp. 220 VAC - 48 VDC converter.</td>
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<td>1.xpc.np01500</td>
<td>30 Amp. 220 VAC - 48 VDC converter.</td>
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<td>2.x1g.urn0cpm</td>
<td>19inch rack for main control boards with the backplane board and the power supply.</td>
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<td>1.x10.ipen000</td>
<td>Control unit mother board.</td>
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<td>2.x13.pagurn2</td>
<td>Multi purpose DSP board with 60 channels.</td>
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<tr>
<td>2.x13.pagurn4</td>
<td>Multi purpose DSP board with 120 channels.</td>
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<tr>
<td>2.x13.pagurn6</td>
<td>Multi purpose DSP board with 180 channels.</td>
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<td>1.x13.paax004</td>
<td>Group switch for 4 racks.</td>
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<td>1.x13.paax014</td>
<td>Group switch for 14 racks.</td>
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<td>1.x13.pae0000</td>
<td>Group switch for 30 racks.</td>
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<td>1.x13.paf0000</td>
<td>Mandatory daughter board for the 30-rack group switch.</td>
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<td>2.x13.pah0000</td>
<td>Ethernet interface board.</td>
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<td>2.x13.pahxes</td>
<td>Ethernet interface board with the DVR (Digital Voice Recorder).</td>
</tr>
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<td>Code</td>
<td>Description</td>
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<tr>
<td>1.kbl.rcpu000</td>
<td>Cable to interconnect main and redundant control &amp; switching units.</td>
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<td>19 inch rack for analog and E1 boards with the backplane board.</td>
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<td>Analog rack controller.</td>
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<td>E1 board with 1 PCM, 30 channels.</td>
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<td>E1 board with 2 PCMs, 60 channels.</td>
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<td>1.x13.fazx003</td>
<td>E1 board with 3 PCMs, 90 channels.</td>
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<td>1.x13.fazx004</td>
<td>E1 board with 4 PCMs, 120 channels.</td>
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<td>License for a single V5.2 link.</td>
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<td>w.pxf.lscxv52</td>
<td>License for a single SS7 E1 interface. More...</td>
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<td>w.pxf.lscxv52</td>
<td>License for a single ISDN E1 interface (DSS1 and QSIG).</td>
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<td>w.pxf.lscxv52</td>
<td>License for a single CAS E1 interface (R1, R2, CIS, EM emulation..)</td>
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w.pxf.lftx001 License for an ETSI-FSK transmitter.
w.pxf.lfrx001 License for an ETSI-FSK receiver.
w.pxf.lscxses License for additional 8 channels for the DVR.
w.x1f.lscxeth License for the Ethernet interface.
w.pxf.voip002 License for 2 (TWO) VoIP channels for IP to TDM and TDM to IP routing. Whenever a channel is free, it can be used by any TDM or VoIP user.
w.pxf.lscencr AES-256 encryption license for VoIP calls.
w.pxf.xmncmpn License for a single Xymphony-API client.
w.x1f.lscrcpu License for the redundant control&switching unit.
w.pxf.lscrerd Auto voice recording channel license.

**1.x1g.kabinet**

**Description**

1.x1g.kabinet is a 19 inch cabinet with 36U height. It includes, internal cabling, front and back covers. The dimensions together with all covers for a single 1.x1g.kabinet is 175 x 60 x 40 cm (h x w x d). When it is empty, the weight is 110kgs. A loaded 1.x1g.kabinet is 150-200kgs (depending on the racks and boards installed inside)
Number of units per system
Depends on the X1 configuration. Several units may be installed side by side, and each fixed to another.

Location
Mounted onto the floor.

1.x1g.ykpk000
Description
1.x1g.ykpk000 is the side cover for 19 inch X1 cabinets.

Number of units per system
Two per system.

Location
Attached to the open sides of the rightmost or the leftmost cabinet 1.x1g.kabinet.
**1.x1g.kbnbagl**

**Description**
1.x1g.kbnbagl is used for fixing the 1.x1g.kabinet cabinet to an adjacent cabinet.

![1.x1g.kbnbagl](image)

**Number of units per system**
Four pieces are used for fixing a cabinet to an adjacent one.

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**1.x1g.eklerxx**

**Description**
Some installation accessories like screws to complete the installation of a particular X1 system.

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**1.x1g.kontakr**

**Description**
1.x1g.kontakr is the main power (48 VDC) distribution frame and battery protection unit. 48 VDC output from the 19inch rack for 4 Amp. AC-DC converters (1.x1g.rak0psm) or from the 19inch rack for 30 Amp. AC-DC converters (1.x1g.urnrtyc) first connects to this unit and then distributed within the X1. Furthermore, it has a voltage monitoring relay and high-current contacts to disconnect the system from the batteries when the battery voltage is below the operational limit. That protects the batteries.

![1.x1g.kontakr](image)

**Number of units per system**
One per each system.

**Location**
Mounted to the back of the X1 cabinet 1.x1g.kabinet having the 19inch rack for 4 Amp. AC-DC converters (1.x1g.rak0psm) or the 19inch rack for 30 Amp. AC-DC converters (1.x1g.urnrtyc)
**1.x1g.prizeki**

**Description**
1.x1g.prizeki is a group of mains sockets for AC-DC rectifier units. It has six mains sockets and a switch to disconnect mains to these sockets.

**Number of units per system**
One per each X1 cabinet 1.x1g.kabinet having AC-DC rectifiers.

**Location**
Ceiling of the X1 cabinet 1.x1g.kabinet having AC-DC rectifiers.

---

**1.x1g.gecis1u**

**Description**
1.x1g.gecis1u is a 19inch bar having a group of power connectors for GND (ground) and -48VDC. It is used in auxiliary X1 cabinets not having rectifier units. Connects to the main power distribution frame 1.x1g.kontakr available in the X1 system and distributes 48VDC within the cabinet where it is located.

**Number of units per system**
One per each 1.x1g.kabinet, which does not have the main power distribution frame 1.x1g.kontakr.

**Location**
Mounted to the back of the X1 cabinet 1.x1g.kabinet not having the main power distribution frame 1.x1g.kontakr.
**1.x1g.rak0psm**

**Description**
1.x1g.rak0psm is a 19 inch rack, which may include up to 5 units of 4 Amp. capacity rectifiers (2.x1a.urmpsp4)

**Number of units per system**
Depends on the power requirements. Mostly, one unit per system is sufficient.

**Location**
In the main cabinet, as the topmost rack.

---

**1.x1g.urnrtyc**

**Description**
1.x1g.urnrtyc is a 19 inch rack, which may include up to 3 units of 30 Amp. capacity rectifiers (1.xpc.np01500)

**Number of units per system**
One unit per system.

**Location**
In the main cabinet, as the topmost rack.

---

**2.x1a.urmpsp4**

**Description**
The X1 may have two models of AC-DC rectifier units. 4Amp capacity one is the model RPS-48A (ordering code: 2.x1a.urmpsp4) and has the following power specifications:

- **Input (RPS-48A):**
  - 180 - 265 VAC at 50Hz
  - 0.95 Amp at 230VAC
- **Output (RPS-48A):**
  - -54 VDC, 4.2 Amp
**Number of units per system**
Depends on the X1 configuration and its power requirement.

**Location**
Within the rack 1.x1g.rak0psm (19inch rack for 4 Amp. AC-DC converters).

---

**1.xpc.np01500**

**Description**
The X1 may have two models of AC-DC rectifier units. 30Amp capacity one is the model NP1500 (ordering code: 1.xpc.np01500) and has the following power specifications:

**Input (NP1500):**
- 200 - 240 VAC at 50Hz
- 8.5 Amp at 230VAC

**Output (NP1500):**
- -54 VDC, 30 Amp

**Number of units per system**
Depends on the X1 configuration and its power requirement.

**Location**
Within the rack 1.x1g.urnrtyc (19inch rack for 30 Amp. AC-DC converters).
**2.x1g.urn0cpm**

**Description**

2.x1g.urn0cpm is a 19 inch rack with the backplane board and DC-DC converter for the control and switching boards. The 2.x1g.urn0bp has:

- One slot for the control unit mother board (1.x10.ipen000)
- One slot for the multi purpose DSP board (2.x13.pagurn2 or 2.x13.pagurn4 or 2.x13.pagurn6)
- One slot for the ethernet interface board (2.x13.pah0000)
- One slot for the group switch board (1.x13.pae0000 or 1.x13.pae0010)
- Some free slots for future applications

**Number of units per system**

One for an X1 without redundancy, two for an X1 with redundancy.

**Location**

Mounted into the main 1.x1g.kabinet (19inch cabinet to install racks).

---

**1.x10.ipen000**

**Description**

It is the main processing unit of the X1. The advanced operating system Xymphony runs on it.

**License**

1.x10.ipen000 together with the DSP board 2.x13.pagurnx stores the licences loaded into the X1. Never dismantle the original 2.x13.pagurnx and 1.x10ipen000 configuration without the assist of an authorized Telesis technical service point. Otherwise, the license data on the X1 is erased and the warranty is void.

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**Number of units per system**

One per systems without redundancy. Two per systems with redundancy.

**Location**

To the leftmost slot in the 2.x1g.urn0cpm, which is the 19inch rack for main control boards.
2.x13.pagurn2

**Description**

2.x13.pagurn2 board contains 60 channel capacity DSP (digital signal processor) for:

- HDLC operations
- MFR1, MFCR2, DTMF transmission and reception
- ANI tone generation and detection
- FSK tone generation and detection
- V32.bis modem (14400bps)
- conference

In addition, the board has:

- two RS232 serial I/O ports
- one RS485 serial I/O port for the operators/maintenance console connection

For each FAC controller board (2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1), one DSP channel is reserved for HDLC operations on the 2.x13.pagurn2. The number of DSP channels in a system must be greater than or equal to the total number of FAC controllers and CCS type E1 digital trunks. MF tone detection and generation require further DSP channels. Telesis and/or its authorized partners will decide on how many DSP channels are required for a particular X1 system and propose an appropriate PAG board.

A 9-conductor D-sub connector on the 2.x13.pagurn2 is for the maintenance console connection.

One of the RS-232 I/O ports on the 2.x13.pagurn2 board is for serial communications. The communication through this is of even parity, 8-bit data, 2-stop bits and adjustable rate from 2400 to 115200 baud. The baud rate can be set and modified by using the XMan (eXchange Manager). Another RS-232 I/O port on the 2.x13.pagurn2 is for interconnecting the main and the redundant control & switching racks for an X1 with redundancy option.

**License**

2.x13.pagurn2 together with the control unit motherboard 1.x10.ipen000 stores the licenses loaded into the X1. Never dismantle the original 2.x13.pagurn2 and 1.x10ipen000 configuration without the assist of an authorized Telesis technical service point. Otherwise, the license data on the X1 is erased and the warranty is void.

**Number of units per system**

One per systems without redundancy. Two per systems with redundancy.

**Location**

To a slot in between the control unit mother board and group switch board inside the 2.x1g.urn0cpm, which is the 19inch rack for main control boards.
**2.x13.pagurn4**

**Description**

2.x13.pagurn4 board contains 120 channel capacity DSP (digital signal processor) for:

- HDLC operations
- MFR1, MFCR2, DTMF transmission and reception
- ANI tone generation and detection
- FSK tone generation and detection
- V32.bis modem (14400bps)
- conference

In addition, the board has:

- two RS232 serial I/O ports
- one RS485 serial I/O port for the operators?/maintenance console connection

For each FAC controller board (2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1), one DSP channel is reserved for HDLC operations on the 2.x13.pagurn4. The number of DSP channels in a system must be greater than or equal to the total number of FAC controllers and CCS type E1 digital trunks. MF tone detection and generation require further DSP channels. Telesis and/or its authorized partners will decide on how many DSP channels are required for a particular X1 system and propose an appropriate PAG board.

A 9-conductor D-sub connector on the 2.x13.pagurn4 is for the maintenance console connection.

One of the RS-232 I/O ports on the 2.x13.pagurn4 board is for serial communications. The communication through this is of even parity, 8-bit data, 2-stop bits and adjustable rate from 2400 to 115200 baud. The baud rate can be set and modified by using the XMan (eXchange Manager). Another RS-232 I/O port on the 2.x13.pagurn4 is for interconnecting the main and the redundant control&switching racks for an X1 with redundancy option.

**License**

2.x13.pagurn4 together with the control unit motherboard 1.x10.ipen000 stores the licenses loaded into the X1. Never dismantle the original 2.x13.pagurn4 and 1.x10ipen000 configuration without the assist of an authorized Telesis technical service point. Otherwise, the license data on the X1 is erased and the warranty is void.

**Number of units per system**

One per systems without redundancy. Two per systems with redundancy.

**Location**

To a slot in between the control unit motherboard and group switch board inside the 2.x1g.urn0cpm, which is the 19inch rack for main control boards.
2.x13.pagurn6

Description
2.x13.pagurn6 board contains 180 channel capacity DSP (digital signal processor) for:

- HDLC operations
- MFR1, MFCR2, DTMF transmission and reception
- ANI tone generation and detection
- FSK tone generation and detection
- V32.bis modem (14400bps)
- conference

In addition, the board has:

- two RS232 serial I/O ports
- one RS485 serial I/O port for the operators'/maintenance console connection

For each FAC controller board (2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1), one DSP channel is reserved for HDLC operations on the 2.x13.pagurn6. The number of DSP channels in a system must be greater than or equal to the total number of FAC controllers and CCS type E1 digital trunks. MF tone detection and generation require further DSP channels. Telesis and/or its authorized partners will decide on how many DSP channels are required for a particular X1 system and propose an appropriate PAG board.

A 9-conductor D-sub connector on the 2.x13.pagurn6 is for the maintenance console connection.

One of the RS-232 I/O ports on the 2.x13.pagurn6 board is for serial communications. The communication through this is of even parity, 8-bit data, 2-stop bits and adjustable rate from 2400 to 115200 baud. The baud rate can be set and modified by using the XMan (eXchange Manager). Another RS-232 I/O port on the 2.x13.pagurn6 is for interconnecting the main and the redundant control&switching racks for an X1 with redundancy option.

License
2.x13.pagurn6 together with the control unit motherboard 1.x10.ipen000 stores the licenses loaded into the X1. Never dismantle the original 2.x13.pagurn6 and 1.x10ipen000 configuration without the assist of an authorized Telesis technical service point. Otherwise, the license data on the X1 is erased and the warranty is void.

Number of units per system
One per systems without redundancy. Two per systems with redundancy.

Location
To a slot in between the control unit mother board and group switch board inside the 2.x1g.urn0cpm, which is the 19inch rack for main control boards.

1.x13.paax004

Description
1.x13.paax004 board contains; PCM switching circuit, which allows switching of 512 x 512 64 kbps capacity voice and data channels, a system reference clock generator, which provides timing signals for the system, four 9-conductor D-sub female connectors, which provide interconnection between control&switching subsystem and blocks, and programming security button.

The serial counters on 1.x13.paax004 board generate synchronization signals for the data streams at 8.192 Mbit/sec incoming and outgoing to/from the 16.384 MHz clock and digital switching matrix. The serial data streams at 8.192 Mbit/sec on the serial telecommunication bus STB are composed of frames with a width of 125msec. There are 128 channels of 8 bits. The digital switching matrix switches the input STB channels to the output STB channels.

There are four 9-pin D-type female connectors on the 1.x13.paax004; P00, P01, P02, P03 providing the connection with the blocks 0, 1, 2, 3 respectively.
The connection between the connector P00 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by the this FAC controller to be the block-00 in the X1. Similarly, the connection between the connector P03 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by this FAC controller to be the block-03 in the X1.

If there is an E1 group within the first 4 blocks of the system then, the X1 may synchronize with the signal at those E1 interfaces by means of the reference-clock hardware. The operator can determine the synchronization channel by using some programming commands through the XMan (eXchange Manager).

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**Number of units per system**

One per systems without redundancy. Two per systems with redundancy.

**Location**

To the rightmost slot in the 2.x1g.urn0cpm, 19inch rack for main control boards.

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**1.x13.paax014**

**Description**

1.x13.paax014 board contains; PCM switching circuit, which allows switching of 2048 x 2048 64 kbps capacity voice and data channels, a system reference clock generator, which provides timing signals for the system, fourteen 9-conductor D-sub female connectors, which provide interconnection between control&switching subsystem and blocks, and programming security button.

The serial counters on 1.x13.paax014 board generate synchronization signals for the data streams at 8.192 Mbit/sec incoming and outgoing to/from the 16.384 MHz clock and digital switching matrix. The serial data streams at 8.192 Mbit/sec on the serial telecommunication bus STB are composed of frames with a width of 125msec. There are 128 channels of 8 bits. The digital switching matrix switches the input STB channels to the output STB channels.

There are fourteen 9-pin D-type female connectors on the 1.x13.paax004; P00, P01, P02,..., P13 providing the connection with the blocks 0, 1, 2,..., 13 respectively.

The connection between the connector P00 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by the this FAC controller to be the block-00 in the X1. Similarly, the connection between the connector P13 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by this FAC controller to be the block-13 in the X1.

If there is an E1 group within the first 4 blocks of the system then, the X1 may synchronize with the signal at those E1 interfaces by means of the reference-clock hardware. The operator can determine the synchronization channel by using some programming commands through the XMan (eXchange Manager).
**Number of units per system**

One per systems without redundancy. Two per systems with redundancy.

**Location**

To the rightmost slot in the 2.x1g.urn0cpm, 19inch rack for main control boards.

---

**1.x13.pae0000**

**Description**

1.x13.pae0000 board contains; PCM switching circuit, which allows switching of 4096 x 4096 64 kbps capacity voice and data channels, a system reference clock generator, which provides timing signals for the system, and programming security button. Its mandatory 1.x13.paf0000 daughter board has thirty 9-conductor D-sub female connectors, which provide interconnection between control&switching subsystem and blocks.

The serial counters on 1.x13.pae0000 board generate synchronization signals for the data streams at 8.192 Mbit/sec incoming and outgoing to/from the 16.384 MHz clock and digital switching matrix. The serial data streams at 8.192 Mbit/sec on the serial telecommunication bus STB are composed of frames with a width of 125msec. There are 128 channels of 8 bits. The digital switching matrix switches the input STB channels to the output STB channels.

There are thirty 9-pin D-type female connectors on its daughter board 1.x13.paf0000; P00, P01, P02,..., P29 providing the connection with the blocks 0, 1, 2,..., 29 respectively.

The connection between the connector P00 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by the this FAC controller to be the block-00 in the X1. Similarly, the connection between the connector P29 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by this FAC controller to be the block-29 in the X1.

If there is an E1 group within the first 4 blocks of the system then, the X1 may synchronize with the signal at those E1 interfaces by means of the reference-clock hardware. The operator can determine the synchronization channel by using some programming commands through the XMan (eXchange Manager).

**Number of units per system**

One per systems without redundancy. Two per systems with redundancy.

**Location**

To the rightmost slot in the 2.x1g.urn0cpm, 19inch rack for main control boards.
1.x13.paf0000

Description
1.x13.paf0000 is the mandatory daughter board of the 1.x13.pae0000. The 1.x13.paf0000 daughter board has thirty 9-conductor D-sub female connectors, which provide interconnection between control & switching subsystem and blocks.

There are thirty 9-pin D-type female connectors on the 1.x13.paf0000; P00, P01, P02,..., P29 providing the connection with the blocks 0, 1, 2,...,29 respectively.

The connection between the connector P00 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by the this FAC controller to be the block-00 in the X1. Similarly, the connection between the connector P29 and the 2.x1g.fac00ai, 2.x1g.fac00bi, or 2.x1g.fac00e1 board results in the analog, bri isdn, or E1 interfaces group controlled by this FAC controller to be the block-29 in the X1.

If there is an E1 group within the first 4 blocks of the system then, the X1 may synchronize with the signal at those E1 interfaces by means of the reference-clock hardware. The operator can determine the synchronization channel by using some programming commands through the XMan (eXchange Manager).

Number of units per system
One per each 1.x13.pae0000 board.

Location
1.x13.paf0000 is mounted over 1.x13.pae0000.

2.x13.pah0000

Description
2.x13.pah0000 board is with an ethernet interface for the IP Telephony and administration purposes

Number of units per system
One per systems without redundancy. Two per systems with redundancy.

Location
To a slot in between the control unit mother board and group switch board inside the 2.x1g.urn0cpm, which is the 19inch rack for main control boards.
**2.x13.pahxses**

**Description**
2.x13.pahxses board is with an ethernet interface for the IP Telephony and administration purposes, and The integrated DVR (digital voice recorder) hardware.

The standard DVR within the 2.x13.pahxses has 100 hours recording capacity, 8 channels for recording, and 8 channels for playing. The number of channels for recording and playing can be increased in multiples of 8 with purchasing additional w.pxf.lscxses licenses. More recording hour options are also available under request.

**Number of units per system**
One per systems without redundancy. Two per systems with redundancy.

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**Location**
To a slot in between the control unit mother board and group switch board inside the 2.x1g.urn0cpm, which is the 19inch rack for main control boards.

**1.kbl.rcpu000**

**Description**
RS-232 cable with 9-pin connectors. In X1 systems with duplicated control&switching units, RS232-2 connectors of PAG boards in both control racks are connected to each other with using a 1.kbl.rcpu000 cable.
**2.x1g.urn0xpm**

**Description**

2.x1g.urn0xpm is a 19 inch rack with the backplane board for analog and E1 interfaces. The 2.x1g.urn0xpm has:

- One slot for the power supply unit (DC-DC converter)
- One slot for the 2.x1g.fac00ai rack controller (when the rack would contain analog interfaces)
- 15 general purpose slots for the line cards; DASF or DATF or DEM or 2.x1g.fac00e1 (FAC E1 controller)

![An empty 2.x1g.urn0xpm, Height: 6U](image1)

![A loaded 2.x1g.urn0xpm](image2)

**Number of units per system**

Depends on the X1 configuration.

**Location**

Mounted into the 1.x1g.kabinet (19inch cabinet to install racks).

**2.x1g.fac00ai**

**Description**

2.x1g.fac00ai the controller board for the analog interfaces rack 2.x1g.urn0xpm. It can drive up to 15 analog interface boards of any type (1.px0.dasf016, 1.px0.dasfc16, 1.px0.datfr02, 1.px0.datfr06, 1.px0.datfc02, 1.px0.datfc06 or 1.px0.demx004). The two 9-conductor D-sub connectors on the board are used to connect group switch boards (1.x13.paax004, 1.x13.paax014, or 1.x13.pae0000+paf) in the control and switching rack 2.x1g.urn0cpm. The bottom connector connects to the main control and switching rack, whereas the top one connects to the redundant control and switching rack. A sequence of LEDs provides information about the status of the board and its rack.

![2.x1g.fac00ai front side](image3)
**Number of units per system**

One per each rack 2.x1g.urn0xpm.

**Location**

To the right slot of the power supply unit 2.x1g.urnps05 power supply unit within the rack 2.x1g.urn0xpm.

### 2.x1g.urn0bpm

**Description**

2.x1g.urn0bpm is a 19 inch rack with the backplane board for Basic Rate ISDN interfaces. The 2.x1g.urn0bpm has:

- One slot for the power supply unit (DC-DC converter)
- One slot for the 2.x1g.fac00bi rack controller
- 12 general purpose slots for the line cards; DDL or DDU

**Number of units per system**

Depends on the X1 configuration.

**Location**

Mounted into the 1.x1g.kabinet (19inch cabinet to install racks).
**2.x1g.fac00bi**

Description

2.x1g.fac00bi is the controller board for the BRI ISDN rack 2.x1g.urn0bpm. It can drive up to 12 BRI ISDN boards of any type (1.px0.ddlx008, 1.px0.ddlx004, 1.px0.ddux008, or 1.px0.ddux004). The two 9-conductor D-sub connectors on the board are used to connect group switch boards (1.x13.paa004, 1.x13.paa014, or 1.x13.pae0000+paf) in the control and switching rack 2.x1g.urn0cpm. The bottom connector connects to the main control and switching rack, whereas the top one connects to the redundant control and switching rack. A sequence of LEDs provides information about the status of the board and its rack.

**Number of units per system**

One per each rack 2.x1g.urn0bpm.

**Location**

To the right slot of the power supply unit 2.x1g.urnps05 power supply unit within the rack 2.x1g.urn0bpm.

**2.x1g.urnps05**

Description

The X1 may have two models of DC-DC converter units. The model RPS05A (ordering code: 2.x1g.urnps05) is used in analog and ISDN peripheral racks. It has the following power specifications:

Input (RPS05A):
- 54 VDC
Output (RPS05A):
- +5 VDC (3 Amp)
- -5 VDC (1 Amp)
- Ring (10w, 25Hz)

**Important Note:** This power supply unit can drive maximum one 2.x1g.fac00ai and one 2.x1g.fac00e1 in the same rack 2.x1g.urn0xpm. If it is required to install more than one 2.x1g.fac00e1 in a same rack, use 2.x1g.urnpcps with more power.

**Number of units per system**

One per each 2.x1g.urn0xpm (19inch rack for analog and E1 boards with the backplane board) and 2.x1g.urn0bpm (19inch rack for BRI ISDN interfaces with the backplane board).

**Location**

To the leftmost slot of the rack 2.x1g.urn0xpm or 2.x1g.urn0bpm.

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### 2.x1g.urnpcps

**Description**

The X1 may have two models of DC-DC converter units. The model PSPW05 (ordering code: 2.x1g.urnpcps) is used in E1 interfaces racks. It has the following power specifications:

**Input (PSPW):**
- 54 VDC

**Output (PSPW):**
- +5 VDC (25 Amp)
- -5 VDC (1 Amp)
- +12 VDC (8 Amp)
- -12 VDC (2 Amp)

**Important Note:** Since there is no ring generator in the 2.x1g.urnpcps power supply, never install analog subscriber interfaces in a rack 2.x1g.urn0xpm having this power supply unit. This power supply unit is required when the 2.x1g.urp05 is insufficient to drive E1 controller boards (2.x1g.fac00e1's) within the rack 2.x1g.urn0xpm.
Number of units per system
One per each 2.x1g.urn0xpm (19inch rack for analog and E1 boards with the backplane board).

Location
To the leftmost slot of the rack 2.x1g.urn0xpm.

2.x1g.fac00e1

Description
2.x1g.fac00e1 is the controller board for E1 interfaces. It can drive a single FAZ daughter board of any type (1.x13.fazx004, 1.x13.fazx003, 1.x13.fazx002, or 1.x13.fazx001). The two 9-conductor D-sub connectors on the board are used to connect group switch boards (1.x13.paxx004, 1.x13.paxx014, or 1.x13.pae0000+paf) in the control and switching rack 2.x1g.urn0xpm. The bottom connector connects to the main control and switching rack, whereas the top one connects to the redundant control and switching rack. A sequence of LEDs provides information about the status of the board and its FAZ board.

Number of units per system
Up to 30 per system.

Location
Any slot in the rack 2.x1g.urn0xpm (19inch rack for analog and E1 boards with the backplane board).
1.x13.fazx001 is a 2.048 Mbit E1 (ITU-T G.703) digital interface card providing a single E1. The E1 can connect to 120 ohm balanced (or optional 75 ohm unbalanced) terminations. The clock of the X1 can be programmed to synchronize with any of the E1s or it may run freely. The line code is programmable as AMI or HDB3. Cyclic redundancy check (CRC4) can be enabled or disabled for the E1 interface. The direction of each channel of the E1 can be programmed as incoming, outgoing, bothway, or unavailable and therefore yields fractional E1 connections. The types of signaling / protocol available for the E1 interface are:

- Channel-Associated Signaling (CAS)
  - Single-bit E&M emulation
  - Two-bit ITU-T R1
  - Two-bit ITU-T R2
  - Many variations of signaling types (such as CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM) widely employed in CIS countries

- Common-Channel Signaling (CCS)
  - DSS1 (Euro-ISDN in the TE direction)
  - DSS1 (Euro-ISDN in the NT direction)
  - ECMA QSIG (in the TE direction)
  - ECMA QSIG (in the NT direction)
  - ITU-T Signaling System No.7 ISUP
  - V5.2 LE and AN protocol

License
The E1 interface requires one of the following licenses for the signaling / protocol support.

- w.pxf.lscxv52
- w.pxf.lscxxs7
- w.pxf.lscxdss
- w.pxf.lscxcas
Number of units per system
Up to 30 FAZ cards in any type (1.x13.fazx004, 1.x13.fazx003, 1.x13.fazx002, or 1.x13.fazx001) per system.

Location
Mounted over E1 controller board (2.x1g.fac00e1).

1.x13.fazx002
Description
1.x13.fazx002 is a 2.048 Mbit E1 (ITU-T G.703) digital interface card providing two E1s. The E1s can connect to 120 ohm balanced (or optional 75 ohm unbalanced) terminations. The clock of the X1 can be programmed to synchronize with any of the E1s or it may run freely. The line code is programmable as AMI or HDB3. Cyclic redundancy check (CRC4) can be enabled or disabled for each E1 interface individually. The direction of each channel of each E1 can be programmed as incoming, outgoing, bothway, or unavailable and therefore yields fractional E1 connections. The types of signaling / protocol available for the E1 interfaces are:

- Channel-Associated Signaling (CAS)
  - Single-bit E&M emulation
  - Two-bit ITU-T R1
  - Two-bit ITU-T R2
  - Many variations of signaling types (such as CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM) widely employed in CIS countries

- Common-Channel Signaling (CCS)
  - DSS1 (Euro-ISDN in the TE direction)
  - DSS1 (Euro-ISDN in the NT direction)
  - ECMA QSIG (in the TE direction)
  - ECMA QSIG (in the NT direction)
  - ITU-T Signaling System No.7 ISUP
  - V5.2 LE and AN protocol

License
Each E1 interface requires one of the following licenses for the signaling / protocol support.

- w.pxf.lscxv52
- w.pxf.lscxss7
- w.pxf.lscxdss
- w.pxf.lscxcas

Number of units per system
Up to 30 FAZ cards in any type (1.x13.fazx004, 1.x13.fazx003, 1.x13.fazx002, or 1.x13.fazx001) per system.

Location
Mounted over E1 controller board (2.x1g.fac00e1).
1.x13.fazx003

Description
1.x13.fazx003 is a 2.048 Mbit E1 (ITU-T G.703) digital interface card providing three E1s. The E1s can connect to 120 ohm balanced (or optional 75 ohm unbalanced) terminations. The clock of the X1 can be programmed to synchronize with any of the E1s or it may run freely. The line code is programmable as AMI or HDB3. Cyclic redundancy check (CRC4) can be enabled or disabled for each E1 interface individually. The direction of each channel of each E1 can be programmed as incoming, outgoing, bothway, or unavailable and therefore yields fractional E1 connections. The types of signaling / protocol available for the E1 interfaces are:

- Channel-Associated Signaling (CAS)
  - Single-bit E&M emulation
  - Two-bit ITU-T R1
  - Two-bit ITU-T R2
  - Many variations of signaling types (such as CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM) widely employed in CIS countries

- Common-Channel Signaling (CCS)
  - DSS1 (Euro-ISDN in the TE direction)
  - DSS1 (Euro-ISDN in the NT direction)
  - ECMA QSIG (in the TE direction)
  - ECMA QSIG (in the NT direction)
  - ITU-T Signaling System No.7 ISUP
  - V5.2 LE and AN protocol

License
Each E1 interface requires one of the following licenses for the signaling / protocol support.

- w.p.x.lscxv52
- w.p.x.lscxss7
- w.p.x.lscxdss
- w.p.x.lscxcas

Number of units per system
Up to 30 FAZ cards in any type (1.x13.fazx004, 1.x13.fazx003, 1.x13.fazx002, or 1.x13.fazx001) per system.

Location
Mounted over E1 controller board (2.x1g.fac00e1).

1.x13.fazx004

Description
1.x13.fazx004 is a 2.048 Mbit E1 (ITU-T G.703) digital interface card providing four E1s. The E1s can connect to 120 ohm balanced (or optional 75 ohm unbalanced) terminations. The clock of the X1 can be programmed to synchronize with any of the E1s or it may run freely. The line code is programmable as AMI or HDB3. Cyclic redundancy check (CRC4) can be enabled or disabled for each E1 interface individually. The direction of each channel of each E1 can be programmed as incoming, outgoing, bothway, or unavailable and therefore yields fractional E1 connections. The types of signaling / protocol available for the E1 interfaces are:

- Channel-Associated Signaling (CAS)
  - Single-bit E&M emulation
  - Two-bit ITU-T R1
  - Two-bit ITU-T R2
• Many variations of signaling types (such as CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM) widely employed in CIS countries
• Common-Channel Signaling (CCS)
  • DSS1 (Euro-ISDN in the TE direction)
  • DSS1 (Euro-ISDN in the NT direction)
• ECMA QSIG (in the TE direction)
• ECMA QSIG (in the NT direction)
• ITU-T Signaling System No.7 ISUP
• V5.2 LE and AN protocol

License
Each E1 interface requires one of the following licenses for the signaling / protocol support.
  • w.pxf.lscxv52
  • w.pxf.lscxss7
  • w.pxf.lscxdss
  • w.pxf.lscxcas

Number of units per system
Up to 30 FAZ cards in any type (1.x13.fazx004, 1.x13.fazx003, 1.x13.fazx002, or 1.x13.fazx001) per system.

Location
Mounted over E1 controller board (2.x1g.fac00e1).

An FAZ card mounted over FAC

w.pxf.lscxv52
Description
In an X1, each E1 link in a V5.2 interface requires a w.pxf.lscxv52 license. The license is granted to a particular X1 system and it is non-transferable.

w.pxf.lscxss7
Description
In an X1, each SS7/E1 interface – that is an E1 with an Signaling System No.7 signaling path or sharing a common signaling path on any other E1- requires a w.pxf.lscxss7 license. The license is granted to a particular X1 system and it is non-transferable.
**w.pxf.lscxdss**

**Description**

In an X1, each E1 interface with the following Primary Rate ISDN signaling types requires a w.pxf.lscxdss license.

- DSS1 Euro ISDN TE (terminal side)
- DSS1 Euro ISDN NT (network side)
- ECMA QSIG TE (terminal side)
- ECMA QSIG NT (network side)

The license is granted to a particular X1 system and it is non-transferable.

**w.pxf.lscxcas**

**Description**

In an X1, each E1 interface with the following Channel Associated Signaling (CAS) types requires a w.pxf.lscxcas license.

- Single-bit E&M emulation
- Two-bit ITU-T R1
- Two-bit ITU-T R2
- ANI (Automatic Number Identification) detection and query widely employed in CIS countries
- Variations of CIS signaling types (such as CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM) widely employed in CIS countries

The license is granted to a particular X1 system and it is non-transferable.

**1.px0.dasf016**

**Description**

The X1's analog subscriber line card. 1.px0.dasf016 has 16 subscriber-loop interface circuits. The subscriber interface can drive loops up to 3,000 ohm resistance (including the terminal equipment) and is capable of on-hook transmission such as sending Caller ID information. The subscriber line card is for 600ohm resistive input impedance. The subscriber circuits have over-voltage protection, conforming to ITU-T K.20/K.21 recommendations.

**License**

Each 1.px0.dasf016 card requires a w.pxf.lscx16 license to operate in an X1 system.

For the 1.px0.dasf016 to insert Caller ID information to subscriber line, at least one Caller ID transmitter license w.pxf.lf001 must be ready in the X1.

**Number of units per system**

Up to 450 units of DASF card in any type (1.px0.dasf016 or 1.px0.dasfc16).

**Location**

In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.uml0xpm).
1.px0.dasfc16

**Description**
The X1's analog subscriber line card 1.px0.dasfc16 has 16 subscriber-loop interface circuits. The subscriber interface can drive loops up to 3,000 ohm resistance (including the terminal equipment) and is capable of on-hook transmission such as sending Caller ID information. The subscriber line card is for complex input impedance. The subscriber circuits have over-voltage protection, conforming to ITU-T K.20/K.21 recommendations.

**License**
Each 1.px0.dasfc16 card requires a w.pxf.lscx16 license to operate in an X1 system. For the 1.px0.dasfc16 to insert Caller ID information to subscriber line, at least one Caller ID transmitter license w.pxf.lftx001 must be ready in the X1.

**Number of units per system**
Up to 450 units of DASF card in any type (1.px0.dasf016 or 1.px0.dasfc16).

**Location**
In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.urn0xpm).

2.dds.krtx001

**Description**
The X1's digital subscriber line card 2.dds.krtx001 has 15 (fifteen) 4B+D ISDN digital subscriber circuits. An appropriate Telesis digital set may connect to the 2.dds.krtx001 over two wires (a single pair of wires).

**License**
Each 2.dds.krtx001 card requires a w.pxf.lscx16 license to operate in an X1 system. In other words, w.pxf.lscx16 license is for both analog and digital subscribers.

**Number of units per system**
Up to 32 units of DDS card in an X1 system.

**Location**
In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.urn0xpm).

w.pxf.lscx016

**Description**
Each 1.px0.dasf016 or 1.px0.dasfc16 or 2.dds.krtx001 card requires a w.pxf.lscx016 license to operate in an X1 system. The license is granted to a particular X1 system and it is non-transferable.
### 1.pxf.datfr02

**Description**

The X1’s analog DC loop trunk card 1.pxf.datfr02 has eight line circuits that can detect line DC feed, polarity reversal, and 12 kHz charge pulses and are capable of on-hook reception of tones such as receiving Caller ID information transmitted. The analog DC loop trunk card is for 600 ohm resistive reference impedance. Over-voltage and over-current protection on analog trunks conforms to ITU-T K.20/K.21 recommendations.

**License**

Each 1.pxf.datfr02 card requires a w.pxf.lscdatf license to operate in an X1 system.

For the 1.pxf.datfr02 to detect Caller ID information from the foreign exchange, at least one Caller ID receiver license w.pxf.lfrx001 must be ready in the X1.

**Number of units per system**

Up to 449 units of DATF card in any type (1.pxf.datfr02, 1.pxf.datfr06, 1.pxf.datfc02, or 1.pxf.datfc06)

**Location**

In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.urn0xpm).

### 1.pxf.datfr06

**Description**

The X1’s analog DC loop trunk card 1.pxf.datfr06 has eight line circuits that can detect line DC feed, polarity reversal, and 16 kHz charge pulses and are capable of on-hook reception of tones such as receiving Caller ID information transmitted. The analog DC loop trunk card is for 600 ohm resistive reference impedance. Over-voltage and over-current protection on analog trunks conforms to ITU-T K.20/K.21 recommendations.

**License**

Each 1.pxf.datfr06 card requires a w.pxf.lscdatf license to operate in an X1 system.

For the 1.pxf.datfr06 to detect Caller ID information from the foreign exchange, at least one Caller ID receiver license w.pxf.lfrx001 must be ready in the X1.

**Number of units per system**

Up to 449 units of DATF card in any type (1.pxf.datfr02, 1.pxf.datfr06, 1.pxf.datfc02, or 1.pxf.datfc06)

**Location**

In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.urn0xpm).
1.pxf.datfc02

**Description**

The X1’s analog DC loop trunk card 1.pxf.datfc02 has eight line circuits that can detect line DC feed, polarity reversal, and 12 kHz charge pulses and are capable of on-hook reception of tones such as receiving Caller ID information transmitted. The analog DC loop trunk card is for complex reference impedance. Over-voltage and over-current protection on analog trunks conforms to ITU-T K.20/K.21 recommendations.

**License**

Each 1.pxf.datfc02 card requires a w.pxf.lscdatf license to operate in an X1 system. For the 1.pxf.datfc02 to detect Caller ID information from the foreign exchange, at least one Caller ID receiver license w.pxf.lfrx001 must be ready in the X1.

**Number of units per system**

Up to 449 units of DATF card in any type (1.pxf.datfr02, 1.pxf.datfr06, 1.pxf.datfc02, or 1.pxf.datfc06)

**Location**

In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.urn0xpm).

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1.pxf.datfc06

**Description**

The X1’s analog DC loop trunk card 1.pxf.datfc06 has eight line circuits that can detect line DC feed, polarity reversal, and 16 kHz charge pulses and are capable of on-hook reception of tones such as receiving Caller ID information transmitted. The analog DC loop trunk card is for complex reference impedance. Over-voltage and over-current protection on analog trunks conforms to ITU-T K.20/K.21 recommendations.

**License**

Each 1.pxf.datfc06 card requires a w.pxf.lscdatf license to operate in an X1 system. For the 1.pxf.datfc06 to detect Caller ID information from the foreign exchange, at least one Caller ID receiver license w.pxf.lfrx001 must be ready in the X1.

**Number of units per system**

Up to 449 units of DATF card in any type (1.pxf.datfr02, 1.pxf.datfr06, 1.pxf.datfc02, or 1.pxf.datfc06)

**Location**

In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.urn0xpm).
w.pxf.lscdatf

Description
Each 1.pxf.datfr02, 1.pxf.datfr06, 1.pxf.datfc02 and 1.pxf.datfc06 card requires a w.pxf.lscdatf license to operate in an X1 system. The license is granted to a particular X1 system and it is non-transferable.

1.px0.demx004

Description
The analog E&M trunk card 1.px0.demx004 has four line circuits that satisfy conditions for AT&T Type-V connections. Each port may individually be set to either a two- or four-wire interface by appropriate configuration of the on-the-board jumpers. The software supports many different line and register signaling types over the E&M lines. An additional detector on the M-wire checks the DC feed on the line for faulty conditions at far-end equipment. The analog E&M trunk card is available for 600 ohm resistive reference impedance. Each E&M port can be programmed as incoming, outgoing, bothway trunk, or unavailable. Each E&M port can be programmed for one of the following signaling types:

- Pulsed line with decadic-address signaling
- Continuous line with DTMF, MFC-R2, MF-R1 register signaling
- Signaling types widely employed in CIS countries:
  - ANI (Automatic Number Identification) detection and query
  - Address signaling
    - Dial Pulse
    - MFC-R1.5
    - Pulse Packet 1
    - Pulse Packet 2
    - Pulse Packet 3a
    - Pulse Packet 3b

License
Each 1.px0.demx004 card requires a w.pxf.lscxem4 license to operate in an X1 system. There is no license requirement for any signaling type applicable for the 1.px0.demx004.

Number of units per system
Up to 449 units

Location
In any general purpose slot of a 19inch rack for analog interfaces (2.x1g.urn0xpm).
w.pxf.lscxem4

Description
Each 1.px0.demx004 card requires a w.p.xf.lscxem4 license to operate in an X1 system. The license is granted to a particular X1 system and it is non-transferable.

1.px0.ddlx008

Description
The 1.px0.ddlx008 card houses circuitry for eight S0 Basic Rate ISDN ports and supports both point-to-point and point-to-multipoint connections.

License
Each 1.px0.ddlx008 card requires a w.p.xf.lscxst8 license to operate in an X1 system.

Number of units per system
Up to 360 units of DDL card in any type (1.px0.ddlx004 or 1.px0.ddlx008)

Location
In any general purpose slot of a 19inch rack for BRI ISDN (2.x1g.urn0bpm)

w.p.xf.lscxst8

Description
Each 1.px0.ddlx008 card requires a w.p.xf.lscxst8 license to operate in an X1 system. The license is granted to a particular X1 system and it is non-transferable.

1.px0.ddlx004

Description
The 1.px0.ddlx004 card houses circuitry for four S0 Basic Rate ISDN ports and supports both point-to-point and point-to-multipoint connections.

License
Each 1.px0.ddlx004 card requires a w.p.xf.lscxst4 license to operate in an X1 system.

Number of units per system
Up to 360 units of DDL card in any type (1.px0.ddlx004 or 1.px0.ddlx008)

Location
In any general purpose slot of a 19inch rack for BRI ISDN (2.x1g.urn0bpm)
**w.pxf.lscxst4**

**Description**
Each 1.px0.ddlx004 card requires a w.pxf.lscxst4 license to operate in an X1 system. The license is granted to a particular X1 system and it is non-transferable.

**1.px0.ddux008**

**Description**
The 1.px0.ddux008 card houses circuitry for eight Uk0 Basic Rate ISDN ports. The line code is 2B1Q.

**License**
Each 1.px0.ddux008 card requires a w.pxf.lscxuk8 license to operate in an X1 system.

**Number of units per system**
Up to 360 units of DDU card in any type (1.px0.ddux004 or 1.px0.ddux008)

**Location**
In any general purpose slot of a 19inch rack for BRI ISDN (2.x1g.urn0bpm)

**w.pxf.lscxuk8**

**Description**
Each 1.px0.ddux008 card requires a w.pxf.lscxuk8 license to operate in an X1 system. The license is granted to a particular X1 system and it is non-transferable.

**1.px0.ddux004**

**Description**
The 1.px0.ddux004 card houses circuitry for four Uk0 Basic Rate ISDN ports. The line code is 2B1Q.

**License**
Each 1.px0.ddux004 card requires a w.pxf.lscxuk4 license to operate in an X1 system.

**Number of units per system**
Up to 360 units of DDU card in any type (1.px0.ddux004 or 1.px0.ddux008)

**Location**
In any general purpose slot of a 19inch rack for BRI ISDN (2.x1g.urn0bpm)
**w.pxf.lscxuk4**

**Description**

Each 1.px0.ddux004 card requires a w.pxf.lscxuk4 license to operate in an X1 system. The license is granted to a particular X1 system and it is non-transferable.

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**1.kbl.x1f9e9e**

**Description**

Cable to connect control and switching unit to the racks.

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**1.kbl.tstk2dd**

**Description**

Each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm) must connect to its adjacent interface racks with 1.kbl.tstk2dd cables. This interconnection allows all the interface racks in the X1 to be aware of which control and switching unit, the main or the redundant one, controls the X1.

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**Number of units per system**

One less than the total number of interface racks (2.x1g.urn0xpm or 2.x1g.urn0bpm) per system. For instance, if the X1 has five interface racks, four 1.kbl.tstk2dd cables are required to provide the continuity of the decision signal among all these racks.

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**w.pxf.lftx001**

**Description**

w.pxf.lftx001 license enables a Caller ID transmitter in the control and switching unit of the X1 for sending Caller ID information to the subscriber lines according to the ETSI ETS 300 659-1 standard as an FSK signal burst where the data transmission occurs during the first long silent period between two ring patterns. Whenever it is available, a licensed transmitter may serve to any subscriber interface in the X1. While inserting the Caller ID information between the first and second ring, the Caller ID transmitter is busy for approximately 2 seconds. The ring OFF time between the first and second ring is around 4 seconds. Consequently, a Caller ID transmitter may serve to two call attempts simultaneously.

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**w.pxf.lfrx001**

**Description**

w.pxf.lfrx001 license enables a caller ID receiver in the control and switching unit of the X1 for detecting Caller ID information from foreign exchange to the analog DC loop trunks according to the ETSI ETS 300 659-1 standard as an FSK signal burst where the data transmission occurs during the first long silent period between two ring patterns. Whenever it is available, a licensed receiver may serve to any DC loop trunk interface in the X1. While detecting the Caller ID information between the first and second ring, the Caller ID receiver is occupied approximately for five seconds. The ring OFF time between the first and second ring is around four seconds. Consequently, a Caller ID receiver may serve to one call at a time.

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**w.pxf.lscxses**

**Description**

The DVR (Digital Voice Recorder) hardware on the 2.x13.pahxses has initially 8 channels for recording and 8 channels for playing. The number of recording and playing channels can be increased in multiples of 8 with the license w.pxf.lscxses. Each w.pxf.lscxses adds 8 more channels to the DVR. The license is granted to a particular X1 system and it is non-transferable.
**w.x1f.lscxeth**

**Description**

10/100 BaseT ethernet hardware on the control and switching unit board PAH is activated with the license w.x1f.lscxeth. An activated ethernet interface may be used for maintenance and administration purposes and IP Telephony applications. However, for an IP Telephony application covering call routings from IP to TDM and/or TDM to IP network, VoIP gateway channel licenses are also required. The license is granted to a particular X1 system and it is non-transferable.

**w.pxf.voip002**

**Description**

IP to TDM and/or TDM to IP simultaneous call routing capacity (that is the gateway capacity) of a X1 can be increased in multiples of two by loading w.pxf.voip002 licenses into the system. The license is granted to a particular X1 system and it is non-transferable.

**w.pxf.lscencr**

**Description**

w.pxf.lscencr license allows VoIP calls between two Telesis systems to be encrypted according to 256 bit AES (Advanced Encryption Standard) while bridging distant offices. In an X1 with this license, encryption may be activated towards some routes and may be deactivated towards some others. A single license is sufficient to allow any number of encrypted VoIP calls. The license is granted to a particular X1 system and it is non-transferable.

**w.pxf.xmncmpn**

**Description**

The Xymphony-API server in the X1 system supports two Xymphony-API clients without needing any license. For each additional Xymphony-API client (like XCOM), a w.pxf.xmncmpn is required. To illustrate, for six Xymphony-API clients to operate for a particular X1, four w.pxf.xmncmpn licenses are required. The license is granted to a particular X1 system and it is non-transferable.

**w.x1f.lscrcpu**

**Description**

It is a license for X1 systems with duplicated control&switching units. w.x1f.lscrcpu license allows the redundant control&switching unit to share all the installed licenses in the main control&switching unit.

**w.pxf.lscrcrd**

**Description**

Auto voice recording channel license. Each w.pxf.lscrcrd licence allows a single port to be marked as auto recorded. That is, all conversations on this marked port will be recorded by the integrated DVR hardware. The license is granted to a particular X1 system and it is non-transferable.

**w.pxf.lscxsip**

License for a single VoIP phone user with xSIP protocol (that is ITS821 or XPhone). The license is granted to a particular X1 system and it is non-transferable.
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INSTALLATION

WARNING
This equipment should be installed and serviced only by qualified personnel who has the necessary training about electrical equipment, and who understands the hazards that can arise when working on this type of equipment.

FINDING SUITABLE SITE FOR INSTALLATION
The installation site should be bright enough for ease of operation to the maintenance personnel. The site should have enough space and ceiling clearance for system capacity expansion. The system should be installed in a room where there is not too much occupancy. Finding a quite location for the operators console will reduce occupational strain on the working operator. Batteries that are connected to the system should be in a well ventilated area not too far away from the system. The 220 V AC mains connection should be grounded, the power should not be interrupted except for power outages. The ambient environmental conditions should be within the range of:

• Temperature range: 0-40 °C
• Humidity range: %0-85 (non-condensing)

Ensure that the installation site does not contain:

• High voltage lines, smoke, dust, gas or radiation (such as a generator, photocopier etc.)
• Radio equipment that generates or emits high level signals
• Sever, pipes, or valves that could leak or cause condensation
• Vibration causing equipment
• Exposure heat sources or direct sunlight

GROUNDING
To ensure long operational life to your system and your safety, all electrical components should be suitably grounded.

• A protective, independent ground terminal should be connected to the system
• The resistance of the ground and neutral terminals should not exceed 0.5 ohm
• The 220 V AC outlet that connects to the system power supply unit should provide phase-ground and phase-neutral voltage difference no more that 5V
• Any auxiliary equipment that is connected to the system (such as a printer or a maintenance PC terminal) should use the same ground of the system
• Do not use UPS to drive the system

![Protective Grounding terminal](image)

**POWER SPECIFICATIONS**

The X1 may have two models of AC-DC rectifier units. The 4Amp capacity one is the model RPS-48A (ordering code: 2.x1a.urrnmps4) and has the following power specifications:

**Input (RPS-48A):**
- 180 - 265 VAC at 50Hz
- 0.95 Amp at 230VAC

**Output (RPS-48A):**
- -54 VDC, 4.2 Amp

The 30Amp capacity one is the model NP1500 (ordering code: 1.xpc.np01500) and has the following power specifications:

**Input (NP1500):**
- 200 - 240 VAC at 50Hz
- 8.5 Amp at 230VAC

**Output (NP1500):**
- -54 VDC, 30 Amp
The X1 may also have two models of DC-DC converter units. The model RPS05A (ordering code: 2.x1g.urnpso5) is used in analog and ISDN peripheral racks. It has the following power specifications:

Input (RPS05A):
- 54 VDC

Output (RPS05A):
- +5 VDC (3 Amp)
- -5 VDC (1 Amp)
- Ring (10w, 25Hz)

![RPS05A](2.x1g.urnpso5)

The model PSPW05 (ordering code: 2.x1g.urnpecps) is used in E1 interfaces racks. It has the following power specifications:

Input (PSPW):
- 54 VDC

Output (PSPW):
- +5 VDC (25 Amp)
- -5 VDC (1 Amp)
- +12 VDC (8 Amp)
- -12 VDC (2 Amp)

![PSPW05](2.x1g.urnpecps)

**WARNINGS**

This equipment should be installed and serviced only by qualified personnel who has the necessary training about electrical equipment, and who understands the hazards that can arise when working on this type of equipment.

Equipment needs to be grounded with a suitable ground terminal before installation. The ground cable should be routed to ensure that it is as short as possible. Protective ground should not be connected to the ground terminal of any other equipment (i.e., lightning or transformer ground).

External lines or outside lines that could be exposed to environmental conditions should be protected with secondary protective circuitry.
The MDF should be grounded. This ground should not be connected to the ground terminal of any other equipment (such as lightning ground, electrical transformer ground etc.)

The battery to equipment connection should be made with a single connector cable of specified cross sectional area:
- 0-8m: 4 mm²
- 8-12m: 6 mm²
- 12-20m: 10 mm²

To minimize the risk of electrostatic damage to the equipment, serviceman should discharge of electrostatic buildup by wearing a shielded bracelet or take other necessary before handling electrostatic sensitive parts such as cards.

This equipment is to be used in controlled environments where humidity and ambient temperature is maintained within working specifications and necessary precautions should be taken before leaving the equipment un-attendant.

Disconnect all power sources from the equipment before servicing. Do not remove the power supply unit from the equipment when the equipment is powered up.

Do not remove or install cards to the equipment when it is powered up.

Do not plug or unplug the maintenance PC connection from the equipment serial COM port while the equipment is in operation.

Leave the security/reset button in the secure position (status LED is off) after parameter changes are made.

Attach all covers in place after servicing equipment and before leaving the customer premises to avoid customer contact with damageable components.

Make sure to fix all the connectors securely in place. Screw any D-sub connectors that are attached to the equipment.

Any auxiliary equipment that is connected to the system should be set up according to its installation guide.
FIXING THE CABINETS

- In order to open the front, back, and side covers in comfort, choose a location, where there would be minimum one meter distance from any object (like walls) around the 1.x1g.kabinet cabinets of the X1.

- If the X1 would have more than two 1.x1g.kabinet cabinets, the main cabinet, which is the one having the rectifier units and the control&switching rack should be somewhere in the middle to make the internal wiring with less effort and shorter cables.

• The main cabinet should be in the middle

- If the X1 would have several 1.x1g.kabinet cabinets, each cabinet should be fixed to the adjacent one or ones with using 1.x1g.kbnbagl fixing parts.

• Never forget to fix all the cabinets onto the floor with using screws. Note that the X1 does not require raised floor. A flat surface floor might be sufficient. Each 1.x1g.kabinet cabinet comes with two pieces of floor mounting parts as shown below
• Screw 1.x1g.kabinet frame on to the floor mounting parts

• At the end of the installation screw the floor mounting parts on to the floor.

POWER CABBING

Wiring 30 Amp. Capacity Rectifiers (in the Main Cabinet)

- The GND connector (RED) on the 1.x1g.kontaktr is connected to the (+)RTN connector on the 1.x1g.urnrtyc rack. This wiring is done by using a 30cm long, red, multi-conductor wire with 16mm² conducting area.

- The blue, multi-conductor wire with 16mm² conducting area out from the 1.x1g.kontaktr is connected to the (-)NEG connector on the 1.x1g.urnrtyc.

- The 1.xpc.np01500 rectifiers slide into the slots of the 1.x1g.urnrtyc. They connect to the 48VDC power net through the backplane connectors on the 1.x1g.urnrtyc without any wiring.

- The X1 is shipped with internal cabling of the 1.x1g.kontaktr already done at the factory and it is ready to be connected to the rectifier units.
Detailed view of the 1.x1g.kontaktr and 1.x1g.urnrtc interconnection

Wiring 4 Amp. Capacity Rectifiers (in the Main Cabinet)

• The GND connector (RED) on the 1.x1g.kontaktr is connected to the GND connector on the (GND, -BAT) bar on the back of the 1.x1g.rak0psm rack. This wiring is done by using a 30cm long, red, multi-conductor wire with 10mm² conducting area.
• The gray, multi-conductor wire with 10mm² conducting area out from the 1.x1g.kontaktr is connected to the -BAT connector on the 1.x1g.rak0psm.
• Each 2.x1a.urnmps4 rectifier in the rack 1.x1g.rak0psm is connected to the (GND, -BAT) bar on the back. Red wire of the rectifier output cable is connected GND connector on the (GND, -BAT) bar and gray wire of the rectifier output cable is connected -BAT connector on the (GND, -BAT) bar.

• The X1 is shipped with internal cabling of the 1.x1g.kontaktr already done at the factory and it is ready to be connected to the rectifier units.

Distributing 48VDC Among the Racks (in the Main Cabinet)

• Sufficiently long, red, multi-conductor wires with 4mm² conducting area are used to carry GND to the racks within the same cabinet. For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm) and control rack (2.x1g.urn0cpm) within the same cabinet, one such wire is necessary. Each wire is connected to a red GND connector on the 1.x1g.kontaktr.

• Sufficiently long, blue, multi-conductor wires with 4mm² conducting area are used to carry -48VDC to the racks within the same cabinet. For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm) and control rack (2.x1g.urn0cpm) within the same cabinet, one such wire is necessary. Each wire is connected to a blue -48V connector on the 1.x1g.kontaktr.

• All red and blue wires are bundled and fixed to the cabinet.
• For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm), a red wire from the bundle is connected to the top power connector on the backplane board of the rack.

• For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm), a blue wire from the bundle is connected to the bottom power connector on the backplane board of the rack.

• For each control rack (2.x1g.urn0cpm), a red wire from the bundle is connected to the power supply unit of the rack as shown below.

• For each control rack (2.x1g.urn0cpm), a blue wire from the bundle is connected to the power supply unit of the rack as shown below.

Carrying 48VDC to the Auxiliary Cabinets

• To interconnect each auxiliary cabinet to the common 48VDC power net, 1.x1g.kontaktr power connectors on the Main Cabinet are wired to the 1.x1g.gecis1u auxiliary power distribution frames in Auxiliary Cabinets.

• The GND connector (RED) on the 1.x1g.kontaktr is connected to the GND connector (RED) on the 1.x1g.gecis1u of the auxiliary cabinet. This wiring is done by using a sufficiently long, red, multi-conductor wire with 10mm² conducting area.

• The -48V connector (BLUE) on the 1.x1g.kontaktr is connected to the -48V connector (BLUE) on the 1.x1g.gecis1u of the auxiliary cabinet. This wiring is done by using a sufficiently long, blue, multi-conductor wire with 10mm² conducting area.
Distributing 48VDC Among the Racks (in an Auxiliary Cabinet)

- Sufficiently long, red, multi-conductor wires with 4mm² conducting area are used to carry GND to the racks within the same auxiliary cabinet. For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm), one such wire is necessary. Each wire is connected to a red GND connector on the 1.x1g.gecis1u.

- Sufficiently long, blue, multi-conductor wires with 4mm² conducting area are used to carry -48VDC to the racks within the same cabinet. For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm), one such wire is necessary. Each wire is connected to a blue -48V connector on the 1.x1g.gecis1u.

- All red and blue wires are bundled and fixed to the auxiliary cabinet.

- For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm), a red wire from the bundle is connected to the top power connector on the backplane board of the rack.

- For each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm), a blue wire from the bundle is connected to the bottom power connector on the backplane board of the rack.

Connecting Batteries

- 48VDC batteries with the necessary AH (capacity) should be used for the back-up purpose in case the mains fails. The battery capacity depends on the X1 configuration and the required back-up duration.

- Always charged batteries should be connected to the X1 initially.

- Cables to connect batteries to the X1 should be carefully selected depending on the current to be drawn and the distance between the batteries and the X1.

- For an X1 system having 30 Amp rectifiers, Side-B (+)RTN and (-)NEG connectors on the 1.x1g.umrtyc rack should be connected to + and - poles of the batteries respectively.

- For an X1 system having 4 Amp rectifiers, GND and -BAT connectors on the bar at the back of the rack 1.x1g.rak0psm should be connected to + and - poles of the batteries respectively.
**SOME INTERCONNECTIONS**

**Voltage Detector Connection**

The voltage detector connector on the PAG board (2.x13.pagurn2, 2.x13.pagurn4, 2.x13.pagurn6) must be connected to the power supply unit of the rack 2.x1g.urn0cpm. This connection is necessary to close the operating system Xymphony in safe mode when the power switch on the rack power supply goes down. Without this connection, the X1 does not start up.
Group Switch Board to Rack Controller Connections

9-pin D-type female connectors on group switch board are connected to the rack controller boards

Four 9-pin D-type female connectors on the 1.x13.paax004; P00, P01, P02, P03 providing the connection with the blocks 0, 1, 2, 3 respectively.

Fourteen 9-pin D-type female connectors on the 1.x13.paax004; P00, P01, P02,...., P13 providing the connection with the blocks 0, 1, 2,..., 13 respectively.

Thirty 9-pin D-type female connectors on the 1.x13.paf0000 (the mandatory daughter board of the PAE); P00, P01, P02,...., P29 providing the connection with the blocks 0, 1, 2,..., 29 respectively.

These connectors connect to the FAC rack controllers of the relevant analog interface or BRI interface or E1 interface blocks.

9-pin D-type female connectors on the FAC rack controller board are connected to the main and redundant group switch boards

FAC rack controller of the Block-00. The bottom D-sub connector (0-A) connects to the group switch in the main 2.x1g.urn0cpm rack. The top D-sub connector (0-B)connects to the group switch in the redundant 2.x1g.urn0cpm rack.
FAC rack controller of the Block-01. The bottom D-sub connector (1-A) connects to the group switch in the main 2.x1g.urn0rpm rack. The top D-sub connector (1-B) connects to the group switch in the redundant 2.x1g.urn0rpm rack.

Group switch board in the main 2.x1g.urn0rpm rack. The bottom D-sub connector (0-A) connects to the FAC rack controller of the Block-00. The top D-sub connector (1-A) connects to the FAC rack controller of the Block-01.

Decision Logic Cable Connection

Each interface rack (2.x1g.urn0xpm or 2.x1g.urn0bpm) must connect to its adjacent interface racks with 1.kbl.tsk2dd cables.

1.kbl.tsk2dd decision cable

This interconnection allows all the interface racks in the X1 to be aware of which control and switching unit, the main or the redundant one, controls the X1.
Two interface racks interconnected with a 1. kbl. tst2dd.

WITH REDUNDANT CONTROL AND SWITCHING

Main and Redundant (Aux) Control&Switching Racks or Units

Both units are the same in terms of the hardware components (rack and the boards in the racks). The main unit is the one, which is connected to the bottom connectors of the FAC controller boards. The redundant (or auxiliary) one connects to the top connectors of the FAC controller boards.

Never forget to interconnect PAG boards in the main and redundant control units for license and parameter sharing. See the next page for the details.
Interconnecting PAG Boards in the Main and Redundant Control & Switching Racks

The PAG board has:

- two RS232 serial I/O ports. RS232-1 and RS232-2
- RS232-1 is for PC, printer, external dial-up modem, or Property Management System (Micros-Fidelio) connection
- RS232-2 is for main and redundant control & switching rack interconnection

In X1 systems with duplicated control & switching units, RS232-2 connectors of PAG boards in both control racks are connected to each other with using a 1.kbl.rcpu000 cable.

PAG in the main control & switching rack is connected to the PAG in the redundant control & switching rack.

PAA or PAE+PAF: Group Switch Board

9-pin D-type female connectors on the group switch board of the main control rack connect to the bottom connectors in FAC controllers of the relevant analog interface or BRI interface or E1 interface blocks.

9-pin D-type female connectors on the group switch board of the redundant (auxiliary) control rack connect to the bottom connectors in FAC controllers of the relevant analog interface or BRI interface or E1 interface blocks.
Licensing X1 Systems with Redundant Control&Switching Unit

There are two options. One is installing licenses to both units. If you choose this option, you should purchase licenses for both control&switching units. With this option:

- You should pay for licenses to be installed in both control&switching units
- Both control&switching units operate independently
- If any of the control&switching unit goes out of order (main or redundant), features that have been licensed do not expire

Second option is licensing the main control&switching unit only and allow the redundant control&switching unit to share those licenses. It is called License Sharing Option. With this option:

- You pay only for licenses to be installed in the main control&switching unit
- Operation of the redundant control unit does depend on the operation of the main control&switching unit
- If the redundant control&switching unit goes out of order, features that have been licensed will not expire
- If the redundant control&switching unit is switched off permanently or uninstalled, features that have been licensed will not expire
- If the main control&switching unit goes out of order, features that have been licensed expire after some weeks period following the failure. During this period, the main control&switching unit needs to be serviced and operated again.
- You can not switch off the main control&switching unit permanently
- You can not uninstall the main control&switching unit

In your original purchase order, you may indicate your exact configuration, requirement for the redundant operation and licenses to be installed for a particular X1 and we may ship the control and switching units of the X1 with the licenses installed at the factory accordingly. Or you may install an X1 without any licenses in its control and switching units on the field. Then you may send us the XID file prepared for the X1 for licensing. Then we return you with the XOP file to license this particular system.

License Sharing Option for X1 Systems with Redundant Control&Switching Unit

- The main control&switching unit should be operating in active mode
- The redundant control&switching should be operating in stand-by mode
- RS232-2 connectors of PAG boards in both control racks should be connected to each other with using a 1.kbl.rcpu000 cable
- Connect to the main control&switching unit of the X1
- Go to the licensing page
Then mark Redundant control unit
Mark or edit other features to be licensed. Save the XID file and send it to us.
When you get the XOP file, download it to the main control&switching unit again.

**Licensing Both Control&Switching Units (No License Sharing Option)**

- The main control&switching unit should be operating in active mode
- The redundant control&switching should be operating in stand-by mode or switched off
- Connect to the main control&switching unit of the X1
- Go to the licensing page

Redundant control unit should be gray, not editable (if the redundant control&switching unit is switched off). If it is not gray and editable, be sure that it is unmarked
Mark or edit other features to be licensed. Save the XID file and send it to us.
When you get the XOP file, download it to the main control&switching unit again.

Similarly,

- Power down the main control&switching unit to lead the redundant to become active
- The redundant control&switching unit should be operating in active mode
- The main control&switching should be operating in stand-by mode or switched off
- Connect to the redundant control&switching unit of the X1
- Go to the licensing page

Redundant control unit should be gray, not editable (if the main control&switching unit is switched off). If it is not gray and editable, be sure that it is unmarked
Mark or edit other features to be licensed. Save the XID file and send it to us.
When you get the XOP file, download it to the redundant control&switching unit again.

**INTERFACES AND CONNECTORS ON THE CONTROL UNITS**

**PAG: Multi purpose DSP Board**

The board has:
- two RS232 serial I/O ports. RS232-1 and RS232-2
  - RS232-1 is for PC, printer, external dial-up modem, or Property Management System (Micros-Fidelio) connection
  - RS232-2 is for main and redundant control&switching rack interconnection
- one RS485 serial I/O port for the operators?/maintenance console connection
- voltage detector connector

*Pin diagram of the RS232-1 port on the PAG for PC, printer, external dial-up modem and Property Management System (Micros-Fidelio) connection*
The voltage detector connector connects to the power supply unit of the rack 2.x1g.urn0cpm and used to close the operating system Xymphony in safe mode when the power switch on the rack power supply goes down.

**PAH: Ethernet Interface Board**

The only interface on this board is a 10/100 BaseT ethernet interface.

![Ethernet Interface Board Diagram](image)

**Pin diagram of the ethernet interface for IP Telephony and XMan access**

**PAA or PAE+PAF: Group Switch Board**

Four 9-pin D-type female connectors on the 1.x13.paax004; P00, P01, P02, P03 providing the connection with the blocks 0, 1, 2, 3 respectively. Fourteen 9-pin D-type female connectors on the 1.x13.paax004; P00, P01, P02,..., P13 providing the connection with the blocks 0, 1, 2,..., 13 respectively. Thirty 9-pin D-type female connectors on the 1.x13.paf0000 (the mandatory daughter board of the PAE); P00, P01, P02,..., P29 providing the connection with the blocks 0, 1, 2,..., 29 respectively. These connectors connect to the FAC controllers of the relevant analog interface or BRI interface or E1 interface blocks.

**FAC: Block Controller Board**

![FAC Controller Board](image)

**Connects to the main control&switching rack**

**Connects to the redundant control&switching rack**
The two 9-conductor D-sub connectors on the board are used to connect group switch boards (1.x13.paxx004, 1.x13.paxx014, or 1.x13.pae0000+1.x13.paf0000) in the control and switching rack 2.x1g.urn0cpm.

In addition, a sequence of LEDs provides some information about the status of the board and its rack.

**FAC leds when the board is installed into a rack**
Selection Jumper on an FAC Board

**Important Note:** In order for the decision and selection logic of the X1 system to operate properly, the selection jumper of the block-00 controller must be short. This selection jumper must be definitely open on the remaining FAC boards within the same X1.

**POWERING UP AND DOWN THE SYSTEM**

**Power Up**

Before powering up the X1:

- Be sure that all the power switches of the X1 is turned OFF,
- Check all power wiring and battery connections once more,
- Be sure that the program security switch is in insecure position (down position) to lead the system start with the factory defaults.

Program security switch is located on the 1.x13.paax004, 1.x13.paax014, and 1.x13.pae0000 boards.
Then, power up the X1 in the following sequence

- Set the switch on the 1.x1g.prizeki to ON position.

- Set all the switches of the 2.x1a.urnmps04 rectifiers to ON (I) position.
- Set the switch on the 1.x1g.kontaktr to ON (I) position.

- Set all the switches on the rack DC-DC converters (2.x1g.urnps05 and 2.x1g.urnpepcs) to ON (I) position.
As the last step, set the switch on the control and switching rack 2.x1g.urn0cpm to ON (I) position.

Wait for a short time for the start up the Xymphony operating system of the X1. When the operating system starts up, all the subscribers within the X1 would be able to get dial tone if they go off-hook.

Complete the installation by:

- downloading the license file into the X1, if the system came without license installed at the factory,
- setting the date and time
- programming the X1 according to the requirements,
- saving the programmed parameters into the non-volatile memories of the X1,
- setting the program security switch is in secure position (up position, its LED will go OFF),
- backing-up the programmed parameters to a file.

Power Down

For safe power down of the X1, turn on switches in sequence:

- Set the switch on the 2.x1g.urn0cpm to OFF (0) position. And wait for some minutes till the leds on the rack backplane turns off.
- Set all the switches on the rack DC-DC converters (2.x1g.urnps05 and 2.x1g.urnpcpcs) to OFF (0) position.
- Set the switch on the 1.x1g.kontaktr to OFF (0) position.
- Set the switch on the 1.x1g.prizeki to OFF position.

**Important Note:** NEVER SET THE SWITCH ON THE 1.x1g.kontaktr TO OFF BEFORE THE LEDS ON THE BACKPLANE OF THE 2.x1g.urn0cpm TURN OFF. Otherwise, you may damage the processing unit.

Be aware that turning off the switch on the 2.x1g.urn0cpm does not turn of the power completely and immediately to the control and switching boards. Voltage is still present in the system for some time.
REPLACING BOARDS AND POWER UNITS

Replacing Interface Boards

- Set the switch on the DC-DC converter (2.x1g.urmps05 or 2.x1g.urmpcpcs) to OFF (0) position for the rack where the interface board is located.

The LED on the DC-DC converter turns OFF,
- Replace the interface board,
- After the replacement, set the switch on the DC-DC converters (2.x1g.urmps05 and 2.x1g.urmpcpcs) to ON(I) position for the rack where the interface board is located.

Replacing Control Boards

- Set the switch on the 2.x1g.urmp0cpm to OFF (0) position. And wait for some minutes till the leds on the rack backplane turn off.

The power switch on the 2.x1g.urmp0cpm is ON (I) in normal operation. Set the switch from ON (I) to OFF (0) position and wait till the backplane leds turn off

Important Note: Be aware that turning off the switch on the 2.x1g.urmp0cpm does not turn of the power completely and immediately to the control and switching boards. Voltage is still present in the rack for some time. Board removal and replacement should be performed after the leds on the module backplane turn off.

- Replace the control board
- Set the switch on the 2.x1g.urmp0cpm to ON (I) position. Wait for a short time for the start up the Xymphony operating system of the X1.
Replacing 4Amp Capacity Rectifiers

- Set the power switch on the relevant 2.x1g.urnmps04 to OFF (0) position,

Power switch on the 2.x1a.urnmps04 rectifier is ON (I) in normal operation. Set the switch from ON (I) to OFF (0) position before the replacement.

- Disconnect the Mains plug from the 1.x1g.prizeki and the front of the rectifier,

First disconnect the mains plug from the 220 VAC mains socket group 1.x1g.prizeki then from the front of the 2.x1g.urnmps04.
• Disconnect the output wires of the rectifier from the connectors,

Output wires of the 2.x1a.urnmps04 are connected to GND, -BAT connectors at the back of the rectifier rack 1.x1g.rak0psm. Disconnect these wires before the replacement.

Make the replacement,

• Connect the output wires of the rectifier from the connectors,
• Connect the Mains plug at the front of the rectifier,
• Set the power switch to ON (I) position again.

**Important Note:** Board or unit removal and replacement must be always performed with disconnecting the power from slots or locations where the board or unit is located. Otherwise, you may damage the board or unit (if the board or unit is not an optionally protected one).
1.x1g.kabinet may have maximum 5 units of 6U racks in any type (2.x1g.urn0xpm or 2.x1g.urn0bpm or 2.x1g.urn0cpm or 1.x1g.rak0psm). Leave 2U space from the top and 1U space in between the racks.
MDF RECOMMENDATIONS

- Means of easy identification of the terminal connections should be provided on the main distribution frame (MDF).

- The use of over voltage protection is possible on all incoming subscriber pairs but not mandatory in case of fully buried subscriber cabling. The MDF should accept cable with a conductor diameter of 0.4 to 0.6 mm.

- Precautions should be taken to ensure that metal parts on which voltages higher than 60 V DC or 42 V AC can occur are not readily accessible.

- The MDF should be equipped with a copper ground bus bar of adequate cross-section area.

- All frames must have a smooth finish, are easy to clean and have no projections or sharp edges.

- MDF should provide the possibility for separating of the line side, for parallel measurement and for connection of an auxiliary measuring equipment to the line- and equipment-side.

- MDF should have high reliability, especially the parts subjected to mechanical actions.

- All cabling should be clearly labeled to enable easy identification.

- It may be preferred that Connections are made with a technique which does not require skills or extended training.

Important Note: The subscriber interfaces of the X1 have over-voltage protection, conforming to ITU-T K.20/K.21 recommendations. Additional primary protection devices to those residing in the main distribution frame may also be employed for further protection. Telesis A.S. recommends double sided, line side surge-protected, and good quality MDF.
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PHONES FOR THE X1:

Telesis DTS821 Executive Digital Telephone Set

DTS821 is an executive digital telephone set for X1 Switching Systems for the comfort of the user. DTS821 has easy-to-use access menu keys and numerous dedicated keys for various telephony tasks and personnel settings. It has a large back-lighted graphic LCD.

DTS821 is connected and powered by a single pair of wires. The set handles multiple calls simultaneously, such as receiving a new call while keeping another on hold.

A few of easy-to-use functions with the DTS821 are:

in idle state:
- access to missed calls list
- access to dialed numbers list
- access to incoming calls list
- access to directory
- playing, deleting voice messages and recorded conversations
- setting volume
- setting ringer volume
- selecting ringer melody
- programming function keys
- programming quick access keys
- setting call forward unconditional
- setting call forward busy
- setting call forward no-reply
- setting hot-line
- activating call waiting
- activating do not disturb
- activating wake-up (reminder) service
- observing firmware version
- in other states:
  - deflecting calls
  - activating call back
  - holding calls on
  - retrieving calls
  - transferring calls
  - tracing transferred calls (up to 4 calls)
  - activating conference
  - recording the conversation bi-directionally (both calling and called party voices)

Intelligent algorithms within the operating system of the X1 systems make the firmware upgrade of DTS821 digital sets automatic whenever new features are added in time. Furthermore, since all the user parameters are stored in the system, replacing and / or upgrading a DTS821 digital set does not require any re-programming.

Telesis ITS821 Executive IP Telephone Set

ITS821 is the IP version of the DTS821 digital set. It is specially designed for Telesis IP PBX systems. The applied VoIP protocol is Telesis xSIP (eXtended SIP) protocol, which allows value added services of the DTS set to be functional over IP.
Telesis Xphone IP Softphone

XPhone is the VoIP softphone with xSIP protocol and supports almost all the functions of the ITS821 on a Windows PC (XP or Vista). The XPhone may operate on a PC with internet connection and it does not need to tackle Firewall and NAT issues. It is an easy-to-install softphone and may operate anywhere with the internet connection. Thanks to intelligent algorithms and proprietary VoIP codecs within the XPhone, the bandwidth requirement is very small.

UTILITIES FOR THE X1:

Xymphony-API

Telesis proprietary Xymphony-API server in Telesis X1 systems performs various telephony tasks such as dialing, call transfer, call park, call retrieve, conference, conversation recording, and display of port status. The physical connection is between a user's personal computer, the same user's analog/digital telephone line and the Telesis X1 system. Both the Xymphony-API server and the PC with Xymphony-API client utility (software) must be in the same local area network. There can be multiple users or client PCs served by the same Xymphony-API server. XCom (Xymphony Companion), which is a Xymphony-API client software developed by Telesis, is an advanced screen-based telephone dialer and console with numerous features to integrate the computer and Telesis X1 system for telephony needs. Moreover, the XCom is a freeware utility.
CRM (Customers Relations Management) Integration

Telesis XCom, which is a freeware API client for Telesis X1 systems, is capable of searching automatically SugarCRM records in the common database according to the received Caller ID information. SugarCRM is the world's leading commercial-open source Customer Relations Management software. SugarCRM is available as a free download as well as a commercial package. The integrated solution is made up of three main components:

- A Telesis X1 system with the Xymphony-API server
- Another Server PC running SugarCRM,
- SugarCRM and XCom users with individual PCs

Xport Utility

A freeware utility software of the X1 for:

- Collecting CMDR (call records)
- Generating Call Reports
- Man-Machine-Interfacing for the integrated signaling analyzer
- Archiving conversation records of auto recorded ports

LICENSING:

Licencing Terms

Seller: Telesis CZ Ltd
Buyer: Reseller or Dealer or End User placing an order directly to Telesis CZ

TELESIS: Telesis CZ
Xymphony: Operating System software for TELESIS Systems
XMan, XPhone, XPort, XCom: Utility software for TELESIS Systems.
Xymphony, XMan, XPort, and XCom are registered trade marks and software products of TELESIS.

All license orders are subject to acceptance by - TELESIS - (Seller). Any acceptance by Seller of Buyer's order is expressly made conditional on Buyer's assent to any additional or different terms and conditions contained herein, and all sales and charges of the licenses listed herein shall be in the case of conflict between the terms and conditions of Buyer and Seller interpreted and governed exclusively by the terms and conditions contained herein. Seller shall not be bound by any terms and conditions proposed by Buyer, whether in its license purchase order or otherwise, which are additional to or different from the terms and conditions set forth herein, unless and only if accepted in writing by a principal officer of the Seller or his designated representative.

1. QUOTATIONS / PROFORMA INVOICES AND PUBLISHED PRICES

Quotations / Proforma Invoices are firm for fifteen (15) calendar days from the date issued unless otherwise stated in the quotation / proforma invoice, and are subject to price withdrawal by notice within that period. Seller reserves the right to unilaterally extend such quotation up to 2 months from the date of issuance. Prices shown on published price lists and other published literature provided by the Seller are not unconditional offers to sell, and are subject to change without notice.

The Seller's prices for the licenses, unless otherwise specified, do not include shipping, insurance, installation, training, maintenance service charges and any other taxes, which may arise.
2. TERMS OF PAYMENT
Except as otherwise provided in the quotation / proforma invoice, payment terms are IN ADVANCE. The USD account of the Seller is provided with the quotation / proforma invoice.

3. ISSUING THE XOP LICENCE FILES
XOP license files are generated and delivered to the Buyer via e-mail within TWO (2) working days after the receipt of the XID license request files and receipt of the payment, unless otherwise stated. The Seller will use reasonable efforts to meet the indicated delivery dates, but cannot be held responsible for its failure to do so. Title to the products and risk of loss or damage shall pass to Buyer upon transmitting the license files via e-mail to the address provided by the Buyer. E-mail address must be valid as it will be used for communication purposes too.

4. ELECTRONIC DATA INTERCHANGE/FACSIMILE (FAX)
Orders placed hereunder by Buyer may be transmitted electronically and in such event, such orders shall be subject to our standard terms and conditions, and shall reference to the Quotation / Proforma Invoice Number. Orders and other communications may be transmitted and confirmed by fax.

5. LIMITATION OF LIABILITY
Seller shall not in any event be liable for incidental, consequential or special damages of any kind resulting from any use or failure of the use of the licenses, even if seller has been advised of the possibility of such damage including, without limitation, liability for loss of use, loss of work in progress, loss of revenue or profits, failure to realize anticipated savings, loss of Buyer property or any liability of Buyer to a third party, or for any labor or any other expense, damage of loss occasioned by such products including personal injury or property damage.

6. REFUND POLICY
The Seller does not issue refunds after the XOP file is generated, which the Buyer is responsible for understanding upon purchasing licenses. All license payments are non-refundable.

7. LICENCING TERMS - LICENCE AGREEMENT
IMPORTANT: If the Buyer do not agree to the terms of this end-user license agreement, he must not upload the XOP license file which is provided to enable the optional parts of the Xymphony operating system software on the TELESIS Systems.

IMPORTANT: The parties to this agreement are:
The Buyer: Reseller or Dealer who purchased TELESIS Systems which incorporates the Xymphony operating system software and,
The Seller: TELESIS for supplying the operating system software Xymphony for the TELESIS Systems. The following optional parts of the Xymphony are licensed, not sold:

- Analog subscriber (FXS) port expansions and/or additions
- Analog DC loop trunk (FXO) port expansions and/or additions
- Digital subscriber port expansions and/or additions
- Analog E&M trunk port expansions and/or additions
- ISDN basic rate S/T interface expansions and/or additions
- ISDN basic rate U interface expansions and/or additions
- TDM-VOIP gateway channel expansions and/or additions
- DVR channel expansions and/or additions
- Enabling VoIP Media Encryption
- ETSI FSK encoders
- ETSI FSK decoders
- DVR auto voice recording channels
- V5.2 Link
- SS7 E1 interface
- ISDN E1 interface
- CAS E1 interface
- Ethernet interface
Companion (Xymphony-API client)
License sharing in system with redundant control and switching
xSIP (eXtended SIP) user
any others, which are not mentioned here

This license agreement describes the Buyer's rights with respect to the software product. The Seller grants the Buyer the following rights:

- use of Xymphony, with its options limited by the invoiced amounts, on a single TELESIS System that the Buyer have purchased,
- install and use of XMan, Xport, and XPhone on any number of computers.

Certain parts of the software which deliver particular functionality to a particular TELESIS System are technologically secured to prevent their unlicensed use. The licensed optional parts of Xymphony should be enabled as detailed in technical documents. Otherwise the optional parts of the software terminate a short time after the TELESIS System is turned on.

By enabling the options the Buyer agrees to be bound by the terms of this license agreement.

If the TELESIS System's control and switching module needs to be replaced for any reason than the optional parts of the Xymphony will have to be re-enabled for the new hardware configuration. The Seller employs these precautions to verify the Buyer's licensed software options.

The Buyer can buy and increase the number of licenses in time. The options that are licensed for a particular TELESIS System can not be transferred partly or wholly to other TELESIS Systems.

The TELESIS software products are protected by copyright laws and international treaty provisions. All title and copyrights in and to the TELESIS software products, the accompanying printed materials, and any copies of the software product are owned by the Seller. The Buyer may not reverse engineer, decompile or disassemble the TELESIS software products. All rights not expressly granted herein are reserved by the Seller.

8. SUPPORT
Licenses are delivered 'as is', with no implied meaning that they will function exactly as the Buyer wish or with all 3rd party extensions/products. Furthermore, the Seller offers no support via email or otherwise for installation, customization, administration of licenses. The Seller reserves the right to respond and answer questions.

9. VALIDITY OF ORDERS
The construction, validity and performance of the Buyer's order shall be governed by the laws of Czech republic.
Telesis CZ Ltd
## TECHNICAL SPECIFICATIONS:

### GENERAL
- **Operational software**: Xymphony X1
- **Maintenance and administrative software**: XMan or IP based
- **Operating voltage**: 110 V AC, 220 V AC, 48 V DC
- **CPU Type**: Half-size industrial
- **Switching Matrix**: 4096 x 4096
- **Redundancy**: Yes
- **Analog subscriber loop impedance**: 3,000 ohms
- **Ethernet interface**: 10/100 BaseT
- **Caller ID ETSI FSK modem**: Yes
- **Integrated CMDR buffer**: Yes
- **Integrated DVR (Digital Voice Recorder)**: Yes
- **DVR recording capacity**: 100 hours or more DVR recording channels 8 or more DVR playing channels 8 or more
  - **Xymphony API server**: Yes
  - **Conference hardware**: Yes
  - **DTMF transceivers**: Yes
  - **MFR1 transceivers, ITU-T Q.320**: Yes
  - **MFCR2 transceivers, ITU-T Q.441**: Yes
  - **HDLC transceivers**: Yes
  - **ANI transceivers**: Yes
  - **Pulse shuttle (R1.5) transceivers**: Yes
  - **Real-time charging**: Yes
  - **A-Party analysis**: Yes
  - **B-Party analysis**: Yes

### APPLICATIONS
- **Subscriber services**: Yes
- **Credited subscribers**: Yes
- **Remote access**: Yes
- **Signaling interworking**: Yes
- **Programmable tones**: Yes
- **APPLICATIONS**: Yes
  - **Central Office**: Yes
  - **Toll switch**: Yes
  - **Tandem switch**: Yes
  - **Transit switch**: Yes
  - **Rural switch**: Yes
  - **Signaling (protocol) converter**: Yes
  - **SSP, Service Switching Point for SS7**: Yes
  - **STP, Signal Transfer Point for SS7**: Yes
  - **PBX**: Yes
  - **IP Telephony**: Yes

### INTERFACES (MAX.)
- **Analog subscribers**: 15,360
- **Digital subscribers**: 480
- **Analog DC loop trunks**: 3840
- **Analog E&M (two- or four-wire)**: 3840
- **RDTT (Ring Down Tie Trunk, local battery)**: 3840
- **BRI ISDN S0**: 2880
- **BRI ISDN Uk0 (Line code 2B1Q)**: 2880
- **PRI ISDN**: 120
- **E1 interfaces (ITU-T G.703)**: 120
<table>
<thead>
<tr>
<th>Feature</th>
<th>Yes/No/Yes with Licensing</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP user agents</td>
<td>Thousands</td>
</tr>
<tr>
<td>H.323 endpoints</td>
<td>Thousands</td>
</tr>
<tr>
<td>Telesis xSIP users</td>
<td>numerous with licensing</td>
</tr>
</tbody>
</table>

### TDM SIGNALING

- Dial-pulse dialing from analog subscribers: Yes
- Dial-pulse dialing to analog CO trunks: Yes
- DTMF dialing from analog subscribers: Yes
- DTMF dialing to analog CO trunks: Yes
- Caller ID transmission on analog subscribers: Yes
- Caller ID detection on analog CO trunks: Yes
- Pulsed line/pulsed address on E&M: Yes
- Continuous line/DTMF address on E&M: Yes
- Single-bit pulsed line signaling types on E1: Yes
- Single-bit continuous line signaling on E1: Yes
- MFR1 signaling on E&M: Yes
- MFCR2 signaling on E&M: Yes
- MFR1 signaling on E1: Yes
- MFCR2 signaling on E1: Yes
- ISDN (Euro ISDN, DSS1), ETSI EN 300 403: Yes
- ISDN Supplementary services:
  - 3PTY, AOC, CCBS, CCNR, CFU, CFNR, CLIP, CLIR, COLP, COLR, ECT, DDI, HOLD, MCID, MSN, UUS: Yes
  - ISDN (QSIG), ECMA-143 PISN: Yes

### CIS COUNTRIES - RUSSIA

- Local trunks SL, Connection line CL: Yes
- Toll-connecting trunks ZSL, Ordered connection line OCL: Yes
- Toll-switched trunks SLM, Toll connection line TCL: Yes
- Two-bit CAS signaling: Yes
- Single-bit CAS signaling: Yes
- Two-, four-wire analog signaling: Yes
- Single-frequency signaling (1VF): Yes
- Multifrequency signaling:
  - Pulse packet 1, 2, 3a, 3b: Yes
  - Pulse shuttle, R1.5: Yes
- Pulse (decadic) signaling: Yes
- ANI request and reception: Yes
- ANI response (generation): Yes
- Unilateral call clearing: Yes
- Bilateral call clearing: Yes
- Calling party category translation: Yes

### IP TELEPHONY

- Interface: 2x 10/100 BaseT
- H.323 protocol, Version 5: Yes
<table>
<thead>
<tr>
<th>Feature</th>
<th>Support</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Session Initiation Protocol, RFC 3261</td>
<td>Yes</td>
</tr>
<tr>
<td>Telesis xSIP (eXtended SIP) protocol</td>
<td>Yes</td>
</tr>
<tr>
<td>G.711 audio codec</td>
<td>Yes</td>
</tr>
<tr>
<td>G.723.1 (5.3 and 6.4kbps) audio codec</td>
<td>Yes</td>
</tr>
<tr>
<td>G.729, G.729A audio codec</td>
<td>Yes</td>
</tr>
<tr>
<td>G.711 frame length</td>
<td>10 to 90msec</td>
</tr>
<tr>
<td>G.723.1 frame length</td>
<td>30 to 90msec</td>
</tr>
<tr>
<td>G.729, G.729A frame length</td>
<td>10 to 90msec</td>
</tr>
<tr>
<td>Silence Suppression (VAD)</td>
<td>Yes</td>
</tr>
<tr>
<td>Echo Canceller G.168-2002</td>
<td>Yes</td>
</tr>
<tr>
<td>QoS (Tos and Diffserv)</td>
<td>Yes</td>
</tr>
<tr>
<td>Integrated H.323 gatekeeper</td>
<td>Yes</td>
</tr>
<tr>
<td>Integrated SIP registrar</td>
<td>Yes</td>
</tr>
<tr>
<td>H.323 endpoints, which can register</td>
<td>Thousands</td>
</tr>
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<td>SIP user agents, which can register</td>
<td>Thousands</td>
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<tr>
<td>xSIP users, which can register</td>
<td>numerous (with licensing)</td>
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<tr>
<td>Programmable ports / sockets</td>
<td>Yes</td>
</tr>
<tr>
<td>MD5 authentication</td>
<td>Yes</td>
</tr>
<tr>
<td>H.235 Baseline Security Profile</td>
<td>Yes</td>
</tr>
<tr>
<td>H.235 Baseline Security Profile with integrity</td>
<td>Yes</td>
</tr>
<tr>
<td>Digest authentication</td>
<td>Yes</td>
</tr>
<tr>
<td>Audio (voice) encryption</td>
<td>AES-256</td>
</tr>
<tr>
<td>Softswitch capability</td>
<td>Yes</td>
</tr>
<tr>
<td>IP to TDM gateway capability</td>
<td>Yes</td>
</tr>
<tr>
<td>TDM to IP gateway capability</td>
<td>Yes</td>
</tr>
<tr>
<td>H.450 supplementary services</td>
<td>Yes</td>
</tr>
<tr>
<td>SIP supplementary services</td>
<td>Yes</td>
</tr>
</tbody>
</table>