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Stillink (r5)

Combined VoIP Gateway and Signaling Converter Systems

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Stillink 200

Stillink 800

COMBINED VOIP GATEWAY AND SIGNALING CONVERTER SYSTEMS

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IN BRIEF:

Thank you

... for choosing the Stillink 200 / 800 Combined VoIP Gateway and Signaling Converter System. It is manufactured to the highest quality standards and tested vigorously to comply with requirement for its product specifications.

In this manual, you will find basic information about the Stillink 200 / 800 and quickly setting it up. We hope that this manual will get you started and help you familiarize with your new system.

For the appropriate operation of your new system, it may need programming and/or licensing. Please contact us or your local Telesis Dealer for details.

We hope that your Stillink 200 / 800 System will serve you well and provide you with all your communication needs now and in the future.

SYSTEM OVERVIEW:

STATE OF THE ART TECHNOLOGY

Stillink 200 / 800 Combined VoIP Gateway and Signaling Converter Systems are designed in the beginning of 2000s, but many refinements have been incorporated since then.

Stillink 200 / 800 Systems have the following characteristics:

- all equipment is mounted on printed circuit boards,
- digital 30 channel PCM/E1 interfaces,
- fully non-blocking integrated circuit switch,
- all protocol/signaling switching (conversion) capability
- embedded and advanced VoIP (voice over IP) technology

The Stillink 200 / 800 is capable of inter-working between numerous digital 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. The possible signaling schemes that are supported by the Stillink 200 / 800 Systems are given in the following sections

STILLINK 200 INTRODUCTION

With dual E1 interfaces that allow connections to the PSTN with SS No. 7, V5.2, DSS1 Euro ISDN, QSIG, R1, R2, CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM, and other CAS signaling and an ethernet interface for VoIP calls, the Stillink 200 is the perfect solution if you are looking into updating an existing telecommunications network, migrating to the VoIP technology or building a network that is ready for the future.

The Stillink 200 is capable of making signaling conversion among any of E1 interfaces as well as routing TDM calls to VoIP calls and vice versa.

The Stillink 200 is a unique all-in-one solution. Notable features are:

- E1 signaling conversion capability
- Both H.323 and SIP protocol support
- Fully non-blocking integrated circuit switch
- All-protocol switching capability

- Advanced routing algorithms
- V5.2 protocol conversion
- Integrated TDM signaling analyzer
- Integrated H.323 gatekeeper
- Integrated SIP registrar
- Integrated media proxy
- Integrated web server for system management
- On-line help system embedded into the integrated web server

The Stillink 200 is ideal for customers who seek a cost effective solution for:

- Legacy equipment life extension
- VoIP connectivity
- Protocol support for older-generation switches
- V5.2 access network to be connected to Les that do not support V5.2
- Sharing and mixing PRI ISDN interfaces
- Easy and fast installation and integration without replacement of old switches
- Easy maintenance
- Adaptability to new protocols

STILLINK 200 APPLICATIONS

The Stillink 200 provides COST EFFECTIVE solutions for:

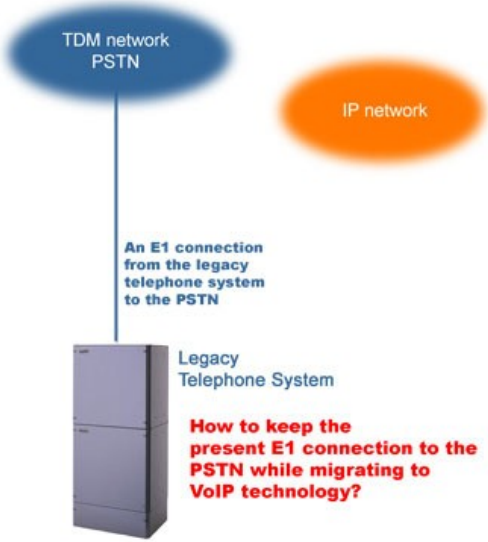
- Digital VoIP Gateway,
- E1 Signaling Converter,
- Combined Digital VoIP Gateway and E1 Signaling Converter,
- V5.2 Access Gateway
- V5.2 Protocol Converter

needs.

Stillink 200 Digital VoIP Gateway

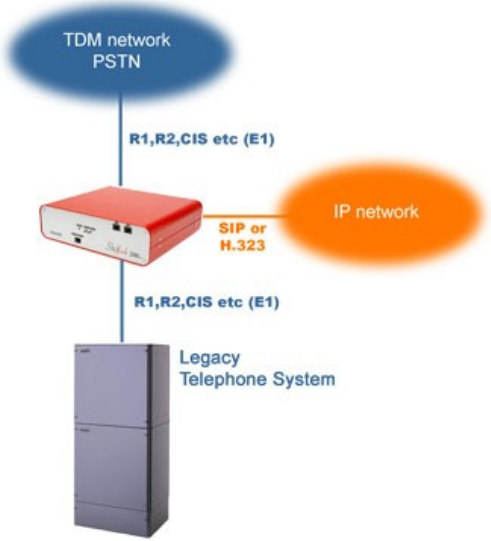
Stillink 200 digital VoIP gateways integrate both packet and circuit switching technology. A Stillink 200 digital VoIP gateway featuring an integrated gatekeeper and registrar provides an economical way for administrators to manage a central database of phone numbers without the expense of a separate-box gatekeeper or registrar solution.

Stillink 200 digital VoIP gateways are with the PSTN by-pass feature that operates not only for IP network failures but also for preselected routes. Alternate routing capability provides automatic fall back to the PSTN if the IP network is unaccessible. Furthermore, enterprises keep their present E1 connections to the PSTN while migrating to VoIP technology.



Stillink 200 digital ISDN-VoIP gateway with PSTN by-pass capability

Stillink 200 digital SS7-VoIP gateway with PSTN by-pass capability



Stillink 200 digital CAS-VoIP gateway with PSTN by-pass capability

Stillink 200 E1 Signaling Converter

The Stillink 200 gives enterprises and service providers the advantage of being able to add advanced services to their networks more quickly and at lower cost than would be the case if they upgraded their existing core network switches.

The Stillink 200 is fully non-blocking and can handle very high call volumes without congestion. The Stillink 200 has an integrated web server for the system management. An on-line help system is also embedded in to this web server. Internal routing tables are configured with any web browser, which also provides flexible E1 signaling conversion management.

The all-protocol switching capability and advanced routing features contained in the Stillink 200 represent a bridge to today`s heterogeneous networks.

The Stillink 200 with dual E1 interfaces provides single E1 span solutions for:

- SS No.7 - ISDN (NT or TE configurable DSS1, QSIG) signaling conversion
- SS No.7 - CAS (R1, R2, CIS, E&M) signaling conversion
- ISDN (NT or TE configurable DSS1, QSIG) - CAS (R1, R2, CIS, E&M) signaling conversion

Stillink 200 Combined VoIP Gateway and E1 Signaling Converter

The Stillink 200 is a unique all-in-one solution combining both digital VoIP gateway and E1 signaling conversion functions. It provides 30 simultaneous VoIP channels for IP to TDM call routing and vice versa. The Stillink 200 with dual E1 interfaces provides solutions for:

- 30 VoIP gateway channels as well as SS No.7 - ISDN (NT or TE configurable DSS1, QSIG) signaling conversion
- 30 VolP gateway channels as well as SS No.7 - CAS (R1, R2, CIS, E&M) signaling conversion
- 30 VoIP gateway channels as well as ISDN (NT or TE configurable DSS1, QSIG) - CAS (R1, R2, CIS, E&M) signaling conversion

The Stillink 200 is ideal for customers who seek a cost-effective solution for both signaling support for the PSTN connection and VoIP migration.

Stillink 200 digital VoIP gateway with E1 signaling conversion

Stillink 200 V5.2 VoIP Access Gateway

With the interworking capability, the Stillink 200 enables a V5.2 access network to connect to VoIP network directly as well as providing PSTN redundancy. This means the Stillink 200:

- allows V5.2 AN systems to be connected to VoIP network or LEs that do not support V5.2
- allows AN vendors to facilitate the connection of their V5.2 ANs to existing non-compliant LEs or VoIP network.

Setup, control, and diagnostics can be performed with a web browser easily. The Stillink 200 with dual E1 interfaces and 30 concurrent VoIP channels provides solutions for:

- V5.2 LE protocol to VoIP (SIP or H.323) conversion as well as ISDN E1 redundancy (alternate conversion to the ISDN protocol)
- V5.2 LE protocol to VoIP (SIP or H.323) conversion as well as SS7 E1 redundancy (alternate conversion to the SS7 protocol)

Stillink 200 V5.2 VoIP Access Gateway with protection protocol

Stillink 200 V5.2 VoIP Access Gateway with PSTN redundancy

STILLINK 800 INTRODUCTION

With a total capacity of eight E1 interfaces that allow connections to the PSTN with SS No. 7, V5.2, DSS1 Euro ISDN, QSIG, R1, R2, CL-1B, OCL-1B, TCL-1B, CL-1VF, OCL-1VF, TCL-1VF, SL/ZSL, SLM, and other CAS signaling and an ethernet interface for VoIP calls, the Stillink 800 is the perfect solution if you are looking into updating an existing telecommunications network, migrating to the VoIP technology or building a network that is ready for the future.

The Stillink 800 is capable of making signaling conversion among any of E1 interfaces as well as routing TDM calls to VoIP calls and vice versa.

The Stillink 800 is a unique all-in-one solution. Notable features are:

- E1 signaling conversion capability
- Both H.323 and SIP protocol support
- Fully non-blocking integrated circuit switch
- All-protocol switching capability
- Advanced routing algorithms
- V5.2 protocol conversion
- Integrated TDM signaling analyzer
- Integrated H.323 gatekeeper
- Integrated SIP registrar
- Integrated media proxy

- Integrated web server for system management
- On-line help system embedded into the integrated web server

The Stillink 800 is ideal for customers who seek a cost effective solution for:

- Legacy equipment life extension
- VoIP connectivity
- Protocol support for older-generation switches
- V5.2 access network to be connected to LEs that do not support V5.2
- Sharing and mixing PRI ISDN interfaces
- Easy and fast installation and integration without replacement of old switches
- Easy maintenance
- Adaptability to new protocols

STILLINK 800 APPLICATIONS

The Stillink 800 provides COST EFFECTIVE solutions for:

- E1 Signaling Converter,
- Combined Digital VoIP Gateway and E1 Signaling Converter,
- V5.2 Protocol Converter
- E1 PRI Switch

needs.

Stillink 800 E1 Signaling Converter

The Stillink 800 gives enterprises and service providers the advantage of being able to add advanced services to their networks more quickly and at lower cost than would be the case if they upgraded their existing core network switches.

The Stillink 800 is fully non-blocking and can handle very high call volumes without congestion. The Stillink 800 has an integrated web server for the system management. An on-line help system is also embedded in to this web server. Internal routing tables are configured with any web browser, which also provides flexible E1 signaling conversion management. The all-protocol switching capability and advanced routing features contained in the Stillink 800 represent a bridge to today`s heterogeneous networks.

The Stillink 800 with eight E1 interfaces provides solutions for:

- SS No.7 - ISDN (NT or TE configurable DSS1, QSIG) signaling conversion
- SS No.7 - CAS (R1, R2, CIS, E&M) signaling conversion
- ISDN (NT or TE configurable DSS1, QSIG) - CAS (R1, R2, CIS, E&M) signaling conversion

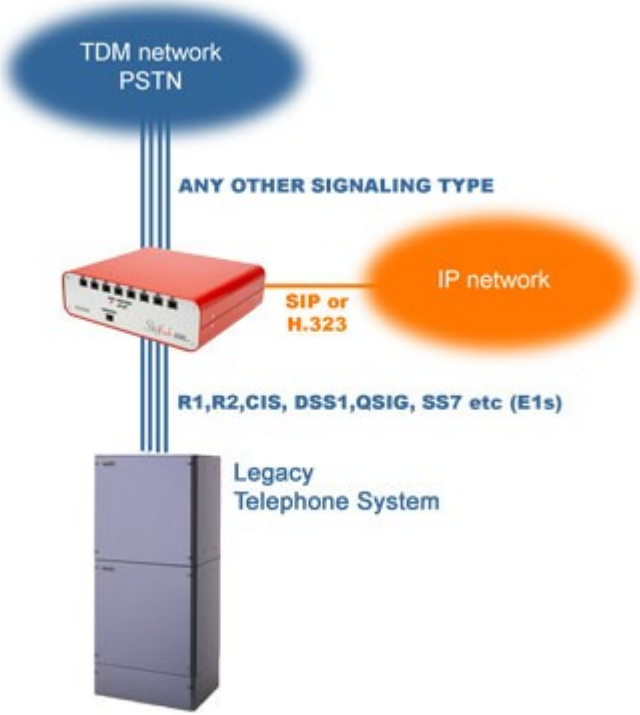
Stillink 800 E1 signaling converter

Stillink 800 Combined VoIP Gateway and E1 Signaling Converter

The Stillink 800 is a unique all-in-one solution combining both digital VoIP gateway and E1 signaling conversion functions. It provides 30 simultaneous VoIP channels for IP to TDM call routing and vice versa. The Stillink 800 with eight E1 interfaces provides solutions for:

- 30 VoIP gateway channels as well as SS No.7 - ISDN (NT or TE configurable DSS1, QSIG) signaling conversion
- 30 VolP gateway channels as well as SS No.7 - CAS (R1, R2, CIS, E&M) signaling conversion
- 30 VoIP gateway channels as well as ISDN (NT or TE configurable DSS1, QSIG) - CAS (R1, R2, CIS, E&M) signaling conversion

The Stillink 800 is ideal for customers who seek a cost-effective solution for both signaling support for the PSTN connection and VoIP migration.



Stillink 800 digital VoIP gateway with E1 signaling conversion

Stillink 800 V5.2 Protocol Converter

With the interworking capability, the Stillink 800 enables an exchange to adapt its existing equipment to the V5.2 protocol. This means the Stillink 800:

- allows V5.2 AN systems to be connected to LEs that do not support V5.2.
- allows AN vendors to facilitate the connection of their V5.2 ANs to existing non-compliant LEs.

Setup, control, and diagnostics can be performed with a web browser easily. The Stillink 800 with eight E1 interfaces provides solutions for:

- SS No.7 - V5.2 LE protocol conversion
- CAS (R1, R2, CIS, E&M) – V5.2 protocol conversion
- ISDN (NT or TE configurable DSS1, QSIG) – V5.2 protocol conversion

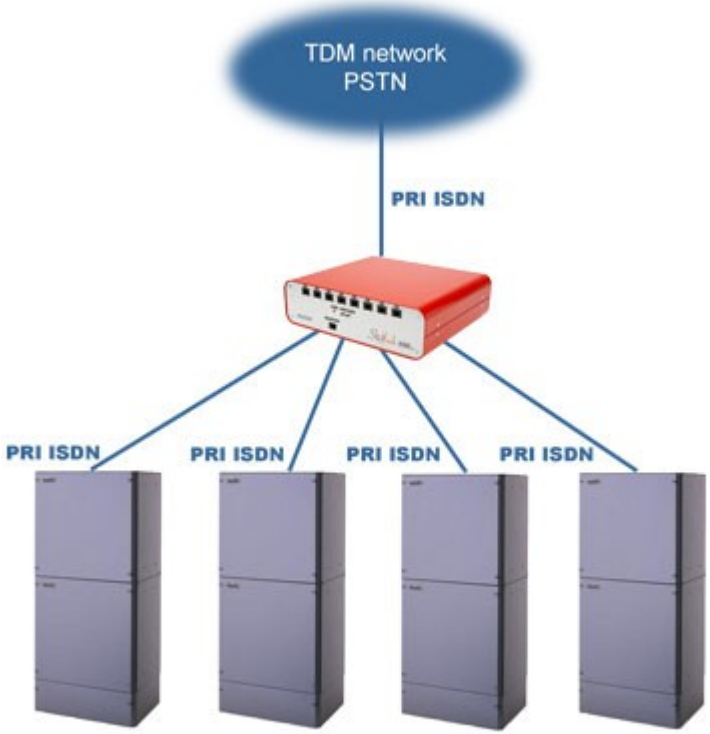
Stillink 800 V5.2 protocol converter

Stillink 800 E1 PRI Switch

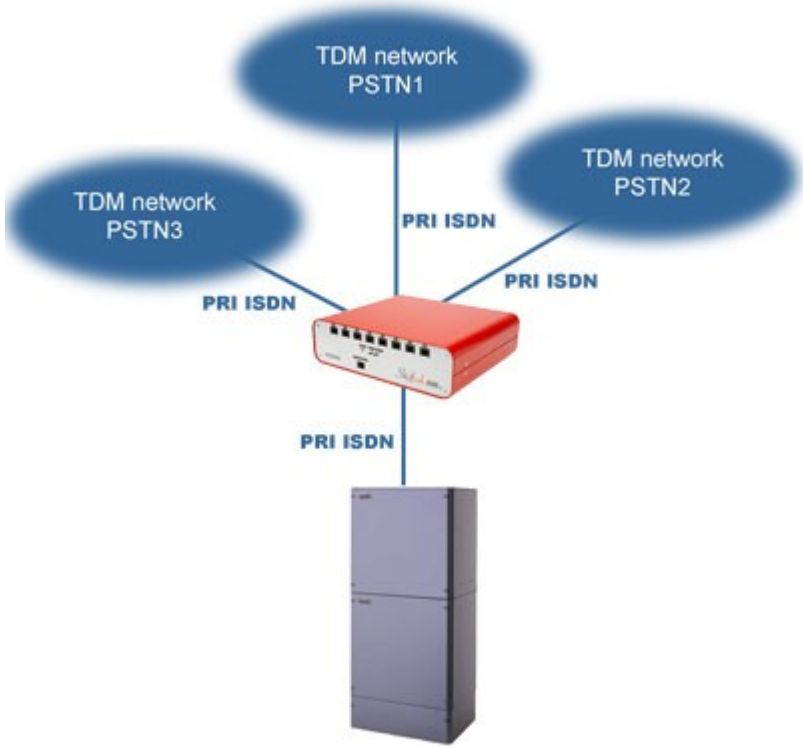
The Stillink 800 with eight DSS1 and QSIG signaling E1 (PRI - Primary Rate ISDN) interfaces provides solutions for mixing and sharing PRI interfaces.

The Stillink 800 is ideal for customers who seek a cost-effective solution for:

- Sharing a PRI PSTN line among local PRI devices. Many local PRI devices need less traffic than the capacity of a PRI interface. All local devices can be bundled for accessing to the existing PSTN PRI line.
- Accessing numerous telecom operators without adding PRI interfaces into the existing local device like the enterprise PABX. Enterprises may prefer to connect alternate operators to reduce the cost of calls for some destinations (LCR - Least Cost Routing).



Stillink 800 Sharing a PSTN PRI line



Stillink 800 Accessing alternate operators

Notable features of the Stillink 800 PRI switch are:

- Fully non-blocking integrated circuit switch
- Time-dependent advanced routing algorithms exceeding LCR requirements
- A-party (calling party) analysis. Modification and restriction functions
- B-party (called party) analysis
- Alternate routing according to the received Release Cause. Re-routing unavailable destination via a secondary, tertiary route and so on
- Integrated TDM signaling analyzer for DSS1 and QSIG
- Fractional E1. Blocking pre-selected voice channels

- Time-dependent tariff table for charging
- AOC-D and AOC-E generation
- CMDR (Call Message Detailed Record) generation
- Built-in ethernet interface for maintenance and billing functions
- Locally generated tones and messages

SYSTEM RESOURCES

The control and switching module within a Stillink 200 / 800 System includes digital signal processors, memory elements, switching matrix, HDLC handlers, and all other service circuits. 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 Stillink 200 / 800 and the essence of its technological sophistication is the control and switching unit, which includes:

- Powerful digital signal processors
- Program and parameter memories
- Operating memories
- Switching circuits (switching matrix)
- Tone generator/synthesizer
- HDLC circuits (for CCS and system control)
- Ethernet interface
- Integrated H.323 gatekeeper and SIP registrar
- Integrated voice cipher for VoIP calls (AES-256)
- Real-Time Clock (RTC)
- Other service components, such as timing circuits and signaling circuits

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).

MEMORIES

The non-volatile (NV) type memories in the Stillink 200 / 800 System are used for storing:

- Boot program,
- Operational software Xymphony,
- Programmed parameters,
- CMDR call records (tens of thousands of call records), and
- Statistics 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 Stillink.

ETHERNET INTERFACES

10/100 BaseT ethernet interfaces on the Stillink 200 / 800 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 data collection
- 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

SOFTWARE

The Stillink 200 / 800 is a Stored-Program-Controlled (SPC) system. Its leading-edge operational software, Xymphony, has been developed by Telesis specifically for these systems.

With many programmable features, Xymphony makes the traffic management, signaling conversion, and VoIP access highly efficient and versatile. Upgrading the Xymphony operating system in the field is accomplished simply by downloading a new version of it directly onto the Stillink's solid state disc over IP without interruption to any of the system's services.

Xymphony within the Stillink 200 / 800 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 features, to guarantee high reliability,

• introduction procedures for new software parts or repairs with minimum service interruptions. New software packages are introduced with a minimum service interruption 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 equipment set up of the Stillink 200 / 800, provided that the system allows the assigned data stored size per function to be changed by the staff.

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

The Stillink 200 / 800 System verifies the integrity of its software at regular basis.

TRAFFIC-HANDLING CAPACITY

The Stillink 200 / 800 has been tested with over 24,000 Busy Hour Call Attempts (BHCA) without any loss in the quality of service. The Stillink 200 / 800 employs a non-blocking switching matrix. The advanced design incorporates 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 Stillink 200 / 800 allows 512 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 Stillink 200 / 800 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 Stillink 200 / 800 provides the address resolution and some others services according to the predefined profile of every registered endpoint. For such calls, the Stillink 200 / 800 acts as a softswitch. There can be any number of simultaneous IP to IP, H.323 calls.

The Stillink 200 / 800 allows 512 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 Stillink 200 / 800 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 Stillink 200/ 800 acts as a softswitch. There can be any number of simultaneous IP to IP, SIP calls.

In the operation of the Stillink 200 / 800 System, there is a sufficient margin for overload situations. The capacity is sufficient to ensure the availability for future new functions or services to be introduced in the system.

Faulty circuits are automatically blocked by the system. Furthermore, each channel of an E1 interface may be:

• blocked for both-way calls

• blocked for incoming calls only

• blocked for outgoing calls only

manually.

TRAFFIC MEASUREMENTS

Traffic measurements of the Stillink 200 / 800 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 Stillink 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 Stillink 200 / 800 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 may collected and stored in a PC,

• Real time traffic measurements.

The Stillink 200 / 800 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

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
- # of successful seizure requests which rang (alerted) the destination
- # of successful seizure requests which connected to the destination
- Total # of received seizure requests from channels in the route
- Total # of received seizure requests which are replied with alerting
- Total # of received seizure requests which are connected
- Total # of channels in the route, which are set as Incoming only
- Total # of channels in the route which are set as Outgoing only
- Total # of channels in the route which are set as Bothway
- Min. # of busy incoming channels at a time in the route during the period
- Max. # of busy incoming channels at a time in the route
- Min. # of busy outgoing channels at a time in the route
- Max. # of busy outgoing channels at a time in the route
- Min. # of busy bothway channels at a time in the route
- Max. # of busy bothway channels at a time in the route
- Average Erlang for incoming channels in the route during the statistics period and calculated according to the occupation of the channels
- Average Erlang for incoming channels in the route during the statistics period and calculated whenever the channels are in conversation state
- Average Erlang for outgoing channels in the route during the statistics period and calculated according to the occupation of the channels
- Average Erlang for outgoing channels in the route during the statistics period and calculated whenever the channels are in conversation state
- Average Erlang for bothway channels in the route during the statistics period and calculated according to the occupation of the channels
- Average Erlang for bothway channels in the route during the statistics period and calculated whenever the channels are in conversation state

CALL ROUTING CAPABILITIES

The advanced routing algorithm within the Stillink 200 / 800 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 E1 interface or E1 channel or SIP entity or H.323 entity. In order to facilitate this analysis, an incoming A-party analysis table is assigned to the originating user (E1 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 Stillink 200 / 800 system services several operators at the same time. 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:

- An E1 interface or an E1 channel which the call is to be routed outward
- A remote H.323 gatekeeper
- A remote SIP registrar
- A tone or a system message stored in the Stillink

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 E1 interface or E1 channel or SIP entity or H.323 entity. In order to facilitate this analysis, an outgoing A-party analysis table is assigned to the called party. 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 CALL (CONVERSION) RECORDS

The Stillink 200 / 800 is 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. All system-management operations are performed without interrupting the functions of the system.

Furthermore, the Telesis XPort utility allows:

- Detailed monitoring of signaling on E1
- Traffic measurement including collecting detailed call-data records

A detailed record for each call may be generated by Xymphony (operating firmware of the Stillink) and is transmitted to Telesis XPort Software Utility (running on a PC). This option may be disabled or enabled. 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 Stillink system
- duration of the call in the record
- incoming port access code
- outgoing port access code
- digits received from the incoming port
- replaced digits after routing analysis
- calling party number
- 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
- Stillink system's given name

XPort can send call records to a printer or a text file in a user-defined line format.

The Stillink 200 / 800 has no hardware parts that require periodic maintenance. However, the call data accumulated in the Stillink's solid state disc should periodically be downloaded through XPort to its database to create space on the Stillink System's disc.

SYSTEM SYNCHRONIZATION

The synchronization algorithms in the Stillink 200 / 800 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 Stillink 200 / 800 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 $5x10^{-5}$, acc. to ITU-T Rec. G.812, MTBF for the internal clock (with stand-by module) is 100 years.

FAULT LOCALIZATION

The Stillink 200 / 800 is capable to perform detection, localization and reporting of failures in it, 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 Stillink 200 / 800 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 Stillink 200 / 800 Systems, it is possible to monitor any E1 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 Stillink 200 / 800 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)
- 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)
- SS7 (CCS no.7) MTP1 (Message Transfer Part 1)
- SS7 (CCS no.7) MTP2 (Message Transfer Part 2)
- SS7 (CCS no.7) MTP3 (Message Transfer Part 3)
- SS7 (CCS no.7) ISUP (ISDN User Part)
- Common control protocol for V5.2 LE and AN protocol
- Port control protocol for V5.2 LE and AN protocol
- Protection protocol for V5.2 LE and AN protocol

- Link control protocol for V5.2 LE and AN protocol
- BCC protocol for V5.2 LE and AN protocol
- ISDN messaging for V5.2 LE and AN protocol
- PSTN messaging for V5.2 LE and AN protocol

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 Stillink 200 / 800 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 Stillink 200 / 800 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.

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, etc.
    - DtmfReceiverTimeout
    - ReleaseConfirm
    - HoldIndication
    - RetrieveIndication
    - SuspendIndication
    - ResumeIndication
    - ReleaseIndication
    - ConnectIndication
    - AlertingIndication
    - ConnectAckIndication
    - ProceedingIndication
    - RoutingFailure
    - MoreInfoIndication
    - DisconnetIndication
    - InbandInfoIndication
    - StatusIndication
    - ProgressIndication
    - NotifyIndication
    - SuspendConfirm
    - SuspendRejectIndication
    - ResumeConfirm
    - ResumeRejectIndication
    - ChargePulseIndication
    - CamponIndication
    - MaliciousCallIndication
    - FacilityIndication
    - RingStartIndication
    - 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.

```
0042 2005/03/16 13:18:43.300 XMT PKT Sapi=00 Tei=40 C/R=1 P/F=0 RR    NR=03
       Hex Dump 02 81 01 07
0042 2005/03/16 13:18:43.318 RCV PKT Sapi=00 Tei=40 C/R=1 P/F=0 RR    NR=02
       Hex Dump 02 81 01 05
01E0 2005/03/16 13:20:54.520 RCV PKT Bib=1 Bsn=01 Fib=1 Fsn=02 msu IAM ISUP Initial address
       00011011 Spare field ok,Len=1B
       10000101 Subservice:National network-0,Spare bits:00,ISUP
       ******** Destination point code:0001
       ******** Originating point code:0000
       ******** Signalling link code:0001
       ******** Circuit identifiction code:0001
       0000---- Spare
       00000001 Message type:01
                Nature of connection indicators
       ------00 Satellite indicator:No satellite circuit in the connection
       ----00-- Continuity check indicator:Continuity check not required
       ---0---- Echo control device indicator:Outgoing echo control device not included
       000----- Spare
                Forward call indicators
       -------0 National/International call indicator:Call to be treated as a national call
       -----00- End-to-end method indicator:No end-to-end method available
       ----0--- Interworking indicator:No interworking encountered
       ---0---- End-to-end information indicator:No end-to-end information available
       --1----- ISDN user part indicator:ISDN user part used all the way
       00------ ISDN user part preference indicator:ISDN user part preferred all the way
       -------0 ISDN access indicator:originating access non ISDN
       -----00- SCCP method indicator:No indication
       ----0--- Spare
       0000---- Reserved
                Calling party's category
       00001010 Ordinary calling subscriber
                Transmission medium requirement
       00000000 Speech
       00000010 Pointer to parameter:Called party number
       00000110 Pointer to optional part
                Called party number
       00000100 Parameter length:04
       -0000010 Nature of address indicator:Unknown
       0------- Odd/even indicator:Even number of address signals
       ----0000 Spare
       -001---- Numbering plan indicator:ISDN/Telephony numbering plan (CCITT E.164)
       0------- Internal network number indicator:Routing to internal network number allowed
       Digits   1002
                Calling party number
```

A sample SS No.7 protocol / signaling analysis

E1 INTERFACES

The 2.048 Mbit E1 digital interfaces can connect to 120 ohm balanced terminations. The clock of a Stillink 200 / 800 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 protocol

SS No.7 SIGNALING

Standards

Stillink 200 / 800 VoIP gateway / signaling converter systems support ISUP ETSI EN 300 356 version 3 and ITU-T requirements.

Applications

Stillink 200 / 800 systems are integrated switches that include both a voice switch and an SS7 switch. A Stillink 200 / 800 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 Stillink 200 / 800 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 Stillink 200 / 800 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 Stillink 200 / 800 system.

As an STP, the Stillink 200 / 800 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 Stillink 200 / 800 '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 Stillink 200 / 800 systems is an E1 (ITU-T G.703) interface. 120 ohm balanced terminations are supported on E1 interfaces. 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. 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

Physical Links

Stillink 200 / 800 systems support both terrestrial and satellite links. For terrestrial links, a basic error detection/correction function is utilized. 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 Stillink 200 / 800 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 Stillink 200 / 800 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 Stillink 200 / 800 system, each link and/or linkset 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 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 Stillink 200 / 800 system converts received digits to SS7 signaling messages. As an STP, the Stillink 200 / 800 system 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

V5.2 PROTOCOL

Standards

V5.2 protocol implementation in Stillink 200 / 800 V5.2 protocol converters follows ETSI Standard EN 300 347-1 (Version 2) and ITU-T G.965. The implemented protocol covers LE (Local Exchange) protocols and supports analog telephone access (PSTN) and ISDN basic-rate access.

Capabilities

Stillink 200 / 800 V5.2 protocol converters support:

- V5.2 Local Exchange (LE) side protocol with
  - Common control protocol
  - Protection control protocol (provided that the V5.2 interface has two or more E1 links)
  - Link control protocol
  - Port control protocol
  - BCC protocol
  - ISDN messaging
  - PSTN messaging
- Protocol conversion between the V5.2 LE protocol and any other signaling type provided by a Stillink 200 / 800 V5.2 protocol converter
- ETSI FSK Caller ID transmission from Stillink 200 / 800 to an AN
- Charge-pulse information transmission from Stillink 200 / 800 to an AN

Protocol Conversion

With the interworking capability, Stillink 200 / 800 V5.2 protocol converters are used to enable an exchange to adapt its existing equipment to the V5.2 protocol. This means Stillink 200 / 800 V5.2 protocol converters:

- allow V5.2 AN systems to be connected to LEs that do not support V5.2.
- allow AN vendors to facilitate the connection of their V5.2 ANs to existing non-compliant LEs.

Setup, control, and diagnostics can be performed with a web browser easily.

VoIP Access Gateway

Stillink 200 V5.2-VoIP access gateways enable V5.2 Access Network (AN) systems to connect IP network. The Stillink 200 converts calls from AN systems to IP network and provides up to 30 simultaneous VoIP calls.

Supplementary Services

Stillink 200 / 800 systems are not non-intelligent converters and gateways. They are inherently switches with the Xymphony operating system, which also runs on other Telesis switches consisting of tens of thousands of PSTN ports. This being the case, Stillink 200 / 800 systems with the Xymphony operating system support for hundreds of PSTN subscribers services by default. This capability makes the Stillink 200 / 800 systems exceptional in V5.2 signaling converter or V5.2-VoIP access gateway applications. Many PSTN subscriber services are ready for use in the V5.2 side. All virtual subscribers defined in a V5.2 LE interface of a Stillink 200 / 800 may use the following subscriber services locally without any limitation:

- Call forwarding unconditional (to any number that the subscriber has authorization for, such as a mobile phone number)
- Call forwarding busy
- Call forwarding no reply
- Override diversions
- Delayed hot line
- Password login
- Password update
- User authorizations update
- Call hold
- Multiple hold
- Call transfer
- Three-party conference
- Call pickup
- Call suspend
- Call resume
- Forced release
- Call intrusion
- Callback
- Answer camped call
- Reject diverted call activation
- Reject diverted call deactivation
- Routed call pickup
- Last number redial
- Dial from user pool
- Dial from common pool
- Reminder service activation
- Reminder service deactivation
- Do not disturb activation
- Do not disturb deactivation
- Malicious call trace
- Voice mail send
- Voice mail receive
- Credit edit

Numbers to be dialed to activate such services can be easily set to any sequence of digits as well as * and # within the flexible routing tables of the Stillink 200 / 800 . Some services may be set as unaccessible for any preselected PSTN subscribers if desired.

Furthermore, subscriber services in a Stillink 200 / 800 may also be combined with the other subscriber services provided by the network side, which may be a TDM or IP type.

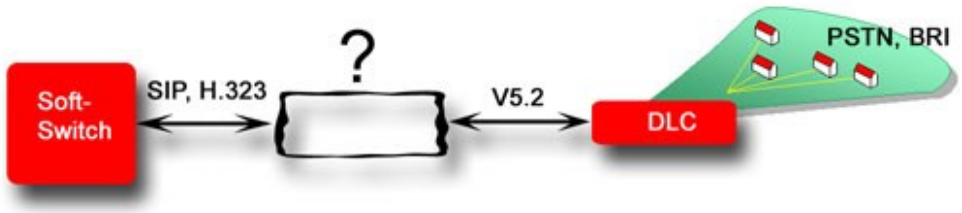
To illustrate the case for a V5.2-VoIP (SIP) access gateway application, many of the PSTN subscriber services (available in the Stillink 200 / 800) may be combined with the SIP calls with using the basic SIP supplementary services such as Invite (call hold), call forward and call transfer (with refer method). A V5.2 PSTN subscriber of a Stillink 200 / 800 may forward an incoming call from the SIP network to another local V5.2 PSTN subscriber or another destination in the same or other SIP network. Or, a V5.2 PSTN subscriber of a Stillink 200 / 800 may activate do not disturb service such that, whenever a call from another local V5.2 PSTN subscriber or SIP network comes in, the service functions.

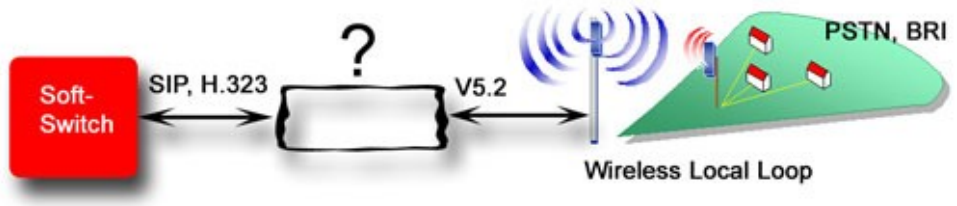
To remark, most of the PSTN subscriber services mentioned above are also valid for SIP user agents or H.323 endpoints, which are locally registered to the integrated SIP registrar or H.323 gatekeeper of the Stillink 200 / 800 . Any registered user is assumed to be the local user of the Stillink 200 / 800 as a subscriber or a trunk. If a PSTN service within the Stillink 200 / 800 is consistent for this local user then activation of the service is allowed.

To summary; professionals may use numerous supplementary subscriber services within the Stillink 200 / 800 for both TDM and IP calls with provisioning.

Migrating to Softswitch-Based Telephony with V5.2 Protocol

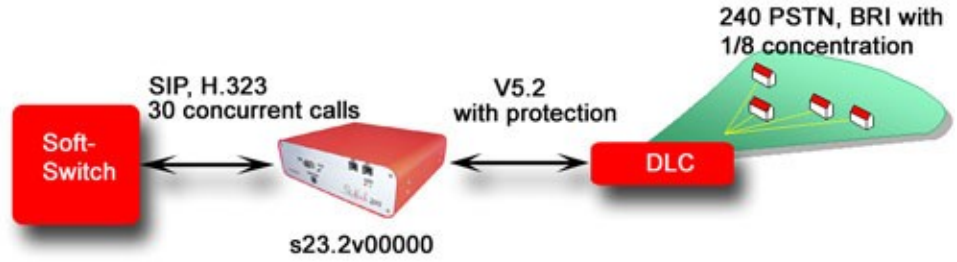
Some VoIP service providers may cut all circuit-switched line services. However, most of the providers are still in need to convert circuit-based remote terminals to packet-based network. Consequently, migrating to softswitch-based local telephony is becoming popular. This conversion or ' local telephony - IP network integration ' offers service providers the opportunity to deliver value-added services to a wide range of customers. Many service providers now focus on which platforms to use.





Stillink V5.2 protocol converters and V5.2 access gateways provide Access Network solutions for VoIP service providers. With the interworking capability, Stillink systems enable service providers to adapt their existing equipment to the V5.2 protocol. This means a Stillink system allows V5.2 AN systems to be connected to the existing network or a softswitch, which does not support V5.2 protocol stack.

For low density requirements, the Stillink 200 is an ideal and cost-effective choice. The system with the ordering code s23.2v00000 is a V5.2 - VoIP access gateway with 30 VoIP channels and a V5.2 interface with dual E1 links (primary and secondary) for the protection.



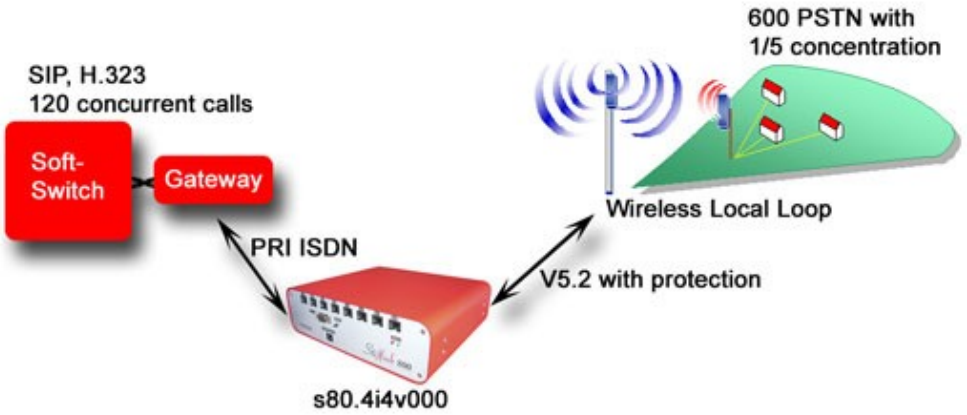
A sample solution with s23.2v00000

For high density requirements, the Stillink 800 with eight E1 interfaces provides solutions for:

- SS No.7 - V5.2 LE protocol conversion (ordering code s80.4s4v000)

- CAS (R1, R2, CIS, E&M) - V5.2 protocol conversion (ordering code s80.4c4v000)

- ISDN (DSS1, QSIG) - V5.2 protocol conversion (ordering code s80.4i4v000)



A sample solution with s80.4i4v000

The implemented protocol for the local telephony in Stilllink systems covers V5.2 LE (local exchange) protocol and supports both analog telephone access (PSTN) and Basic Rate ISDN access. For the softswitch side, both H.323 and SIP protocols are supported.

In addition, the integrated circuit switch in Stillink systems allows station-to-station calls within the same Access Network to physically pass through itself and, therefore, avoid consuming resources of the service providers' equipment. Locally generated call progress tones, voiceband data transmission tones like Caller ID to the stations, DTMF reception and generation provide additional savings in resources.

ISDN SIGNALING

Standards

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 Primary Rate (PRI), 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 Stillink 200 / 800 VoIP gateway / signaling converter systems is an E1 (ITU-T G.703) interface. 120 ohm balanced terminations are supported on E1 interfaces. The line code is HDB3 or AMI programmable. CRC4 may be enabled or disabled.

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 Stillink 200 / 800 system, each primary ISDN access 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. 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 Stillink 200 / 800 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

CIS SIGNALING TYPES

Types

Stillink 200 / 800 VoIP gateway / signaling converter systems support the following types of signaling for Russian (and CIS) PSTN:

- Two-bit channel-associated signaling
- Single-bit channel-associated signaling
- Single-frequency (1VF) signaling
- Multifrequency Signaling

E1 trunk types within Stillink 200 / 800 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. 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. Stillink 200 / 800 systems provide 1VF signaling for local, toll-connecting, and toll-switched trunks.

Options

The following options and parameters are available in CIS-Russian signaling interfaces of Stillink 200 / 800 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, 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 by MML commands.

Route, Routeset Configuration, Routing

In a Stillink 200 / 800 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 Stillink 200 / 800 system may realize routing on calls coming from CIS interfaces according to:

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

R2 SIGNALING

Standards

Stillink 200 / 800 VoIP gateway / signaling converter systems support signaling for two-bit R2 line / MFCR2 interregister according to ITU-T recommendations

Options

The following options and parameters are available on R2 signaling interfaces of Stillink 200 / 800 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 Stillink 200 / 800 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.

Routing

A Stillink 200 / 800 system may realize routing according to:

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

R1 SIGNALING

Standards

Stillink 200 / 800 VoIP gateway / signaling converter systems support signaling for two-bit R1 line / MFR1 interregister according to ITU-T recommendations

Options

The physical interface for single-bit and two-bit signaling is E1 (ITU-T G.703). 120 ohm balanced terminations are supported on E1 interfaces. 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.

Route, Routeset Configuration, Routing

In a Stillink 200 / 800 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 Stillink 200 / 800 system realizes routing according to the dialed digits.

CAS SINGLE BIT E&M EMULATION

Types

Stillink 200 / 800 VoIP gateway / signaling converter 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 (ITU-T G.703). 120 ohm balanced terminations are supported on E1 interfaces. 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 with MML commands.

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 Stillink 200 / 800 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 Stillink 200 / 800 system realizes routing according to the dialed digits.

H.323 PROTOCOL

Standards

Stillink 200 / 800 Digital VoIP Gateways support H.323 protocol. The applied standards are H.225.0 (07/2003) version 5, H.235 (05/2003) version 3, H.245 (07/2003) version 12, H.450 (09/1997), AES FIPS PUB 197.

Applications

Stillink 200 / 800 VoIP gateways integrate both packet and circuit switching technology. A Stillink 200 / 800 VoIP gateway 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.

Stillink 200 / 800 VoIP gateways are with the PSTN by-pass feature that operates not only for IP network failures but also for preselected routes. Furthermore, enterprises keep their present E1 connections to the PSTN while migrating to VoIP technology.

Stillink 200 / 800 systems are unique with their capabilities of VoIP gateway and signaling conversion functions at the same time.

Stillink 200 / 800 VoIP gateways support numerous H.323 entities which can be terminals, gatekeepers or gateways. Number/IP translation is performed through an advanced routing algorithm. Together with the integrated gatekeeper, call authorization, call management, enhanced billing functions, flexible routing algorithms make a Stillink 200 / 800 system serve as a unique Digital VoIP Gateway. Integrated gatekeeper allows calls to be placed direct or gatekeeper routed between H.323 entities in various circumstances.

While voice bridging distant offices over the IP, security of a VoIP call is guaranteed with the encryption of voice according to 256 bit AES (AES-256).

Furthermore, Stillink 200 and 800 VoIP 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 connecting to the long distance call operator over the IP, the Stillink 200 / 800 VoIP gateway 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 connected to the Stillink 200 / 800 VoIP gateway 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 Stillink 200 / 800 VoIP gateways is 10/100 BaseT Ethernet. Several ethernet and other properties for H.323 are programmable. These are:

- Equipment, Gateway, Gatekeeper, Call signaling IP and port addresses
- Reserved port block for the media
- H.245 parameters
- Audio codec frame lengths
- Manual selection of audio codecs and preferences per endpoint
- Silence suppression (or Voice Activity) activation and detection
- Gatekeeper identifier (for the integrated Gatekeeper)
- Port numbers for maintenance, RAS, and call signaling
- External gatekeeper discovery, and user authentication

Audio Codecs

Stillink 200 / 800 VoIP gateway 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 Stillink 200 / 800 VoIP gateway, an H.323 endpoint 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 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

Stillink 200 / 800 VoIP gateway routes a call from the TDM network to the IP network according to:

- Dialed digits (called number)
- DPC
- Calling party 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

Up to 30 simultaneous calls may be routed from the IP network to the PSTN network and vice versa based on the address resolution at the IP side and routing analysis.

SIP PROTOCOL

Standards

RFC 3261 SIP: Session Initiation Protocol.

Applications

Stillink 200 / 800 VoIP gateways integrate both packet and circuit switching technology. A Stillink 200 / 800 VoIP gateway 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.

Stillink 200 and 800 VoIP gateways are with the PSTN by-pass feature that operates not only for IP network failures but also for preselected routes. Furthermore, enterprises keep their present E1 connections to the PSTN while migrating to VoIP technology.

Stillink 200 and 800 are unique systems with their capabilities of VoIP gateway and signaling conversion functions at the same time.

Stillink 200 / 800 VoIP gateways support numerous SIP 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 make a Stillink 200 / 800 system serve as a unique Digital VoIP Gateway. Integrated gatekeeper allows calls to be placed directly between SIP user agents. IP-PSTN (IP-TDM) calls may be integrated registrar routed.

While connecting to the long distance call operator over the IP, the Stillink 200 / 800 VoIP gateway 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 Stillink 200 / 800 VoIP gateway 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 Stillink 200 / 800 VoIP gateways is 10/100 BaseT Ethernet. Several ethernet and other properties for SIP are programmable. These are:

- Equipment, Gateway, Registrar, IP and port addresses
- Reserved port block for the media
- Audio codec frame lengths
- Manual selection of audio codecs and preferences per agent
- Silence suppression (or Voice Activity) activation and detection

Audio Codecs

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- Calling party 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

Up to 30 simultaneous calls may be routed from the IP network to the PSTN network and vice versa based on the address resolution at the IP side and routing analysis.

AES 256 ENCRYPTION

Description

The optional AES 256 media encryption provides VoIP call security while bridging distant offices with Stillink 200 / 800 systems and H.323 protocol.

Stillink 200 / 800 systems are all-in-one solutions with integrated gatekeeper, softswitch capability, IP-TDM routing (gateway) functions, and numerous IP features. Even though the media encrypting algorithms are applicable for H.323 endpoint-to-endpoint connection too, the implemented algorithm in Stillink 200 / 800 systems operate for H.323 endpoint-to-gatekeeper connection for further security. The following sections demonstrate algorithms applied for site-to-site communication in brief, such that:

- Two Stillink 200 / 800 systems in each site
- Stillink 200 / 800 system in one side is a VoIP gateway
- Stillink 200 / 800 system in the other side is a VoIP gateway with the integrated gatekeeper
- Both systems are provided with necessary licenses for the VoIP media security and their parameters are set accordingly.

Networking

Thanks to the multiple gatekeeper registration capability of Stillink 200 / 800 systems, AES 256 encryption will also operate in a network of multiple Stillink 200 / 800 systems.

Secure Gatekeeper registration

Two Stillink 200 / 800 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 Stillink 200 / 800 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 256bit cipher (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 Stillink 200 / 800 systems involved in a call.

Diffie-Hellman key exchange

Stillink 200 / 800 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 half 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. Stillink 200 / 800 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 any of Setup Acknowledge, Call proceeding, Alerting or Connect messages

Interoperability

Telesis Stillink 200 / 800 , PX24, X1 systems. No other known.

Summary

Security of VoIP communication between two Stillink 200 / 800 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

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LICENSING

The following optional parts of the Stillink 200 / 800 are licensed:

- V5.2 Link
- SS7 E1 interface
- ISDN E1 interface
- CAS E1 interface
- VOIP channel
- Audio (voice) encryption AES 256

Certain parts of the Stillink software (Xymphony) which deliver particular functionality to the Stillink are technologically secured to prevent their unlicensed use.

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 TELESIS.

Licensing is done at the factory.

LICENCING TERMS - LICENCE AGREEMENT

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: Enabling VoIP Media Encryption, V5.2 Link, SS7 E1 interface, ISDN E1 interface, CAS E1 interface, and 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.

STILLINK 200 ORDERING CODES

s2x.AyBz0000

s2: Stillink 200

x:
- 0 No VoIP Channels
- 1 15 VoIP channels licensed
- 3 30 VoIP channels licensed

A: Number of E1s with the signaling y

y:
- c CAS licensed
- i ISDN licensed
- s SS7 ISUP licensed
- v V5.2 licensed

B: Number of E1s with the signaling z

z:
- c CAS licensed
- i ISDN licensed
- s SS7 ISUP licensed
- v V5.2 licensed

Example:

s23.1i1v000: Stillink 200 with 30 VoIP channels, 1 E1s is licensed for ISDN signaling, 1 E1s is licensed for V5.2 signaling

STILLINK 800 ORDERING CODES

s8x.AyBz0000

s8: Stillink 800

x:
- 0 No VoIP Channels
- 1 15 VoIP channels licensed
- 3 30 VoIP channels licensed

A: Number of E1s with the signaling y

y:
- c CAS licensed
- i ISDN licensed
- s SS7 ISUP licensed
- v V5.2 licensed

B: Number of E1s with the signaling z

z:
- c CAS licensed
- i ISDN licensed
- s SS7 ISUP licensed
- v V5.2 licensed

Example

s83.4i4v000: Stillink 800 with 30 VoIP channels, 4 E1s are licensed for ISDN signaling, 4 E1s are licensed for V5.2 signaling

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. The system should be installed in a room where there is not too much occupancy. The VAC 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 mains 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 should use the same ground of the system

The protective ground terminal is at the rear of the equipment and is marked with the protective ground sign as shown below.

Rear view of the Stilllink

WARNINGS

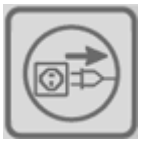
This equipment should be installed and serviced only by qualified personnel who has the necessary training about electrical equipment, and who understands the hazards that can arise when working on this type of equipment.

This 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.

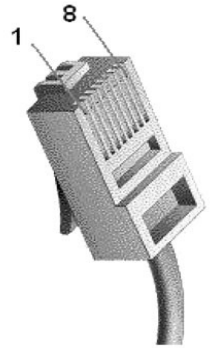
To minimize the risk of electrostatic damage to the equipment, serviceman should discharge of electrostatic buildup by wearing a shielded bracelet or take other necessary before handling electrostatic sensitive parts such as cards.

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.

E1 interface connectors in front of the unit are marked with the numbers 00, 01 for the Stillink 200 and 00, 01, 02, 03, 04, 05, 06, 07 for the Stillink 800. The cable to connect RJ45 E1 connectors on the front panel should be as follows:



Pins	Desc
1	Receive
2	Receive
4	Transmit
5	Transmit
The other pins are N/A	

THIS EQUIPMENT HAS NO SERVICEABLE COMPONENTS INSIDE. PLEASE CONTACT TO YOUR NEAREST AUTHORIZED TELESIS DEALER IN CASE OF FAILURE

PHYSICAL DIMENSIONS

- 255 x 220 x 77 mm (w x d x h)
- 3.25 kgs

POWER SPECIFICATIONS

- 90-264 VAC
- 0.25 A, 240 VAC
- 47-63 Hz

CONNECT TO THE SYSTEM

When the Stillink system starts up, its factory default

IP address is 10.0.0.12

port address is 80

There is no login ID or password for web browser connection.

Connect to your Stillink system with using any web browser (like Internet Explorer, Mozilla Firefox) for maintenance and programming.

ONLINE HELP PAGES AND DOCUMENTS

After you connect to your Stillink system with using a web browser, you may find online help pages and documents. Online help system will guide you programming the system. Whenever you need, you may just click on help icon near to an item.

Similarly, online documents embedded into the Stillink aim:

1.System administrators,

2.Maintenance personnel,

3.Software developers, and

4.Any other enthusiasts

to understand, maintain Stillink systems, and develop their applications.

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SYSTEM PROGRAMMING

Stillink systems are unique and all-in-one combined signaling converter - voip gateway systems. With using Stillink systems, dozens of applications are possible. Next pages intend to guide a Stillink user in configuring and programming its equipment. It aims the Stillink user to become familiar with the basic system programming before he can perform several other advanced programming to fulfill any complicated requirement. General topics of this document are not classified in application basis. However, the information given there may be used in any solution with some minor deviations only.

GENERAL TOPICS

THE BASIC TERMINOLOGY

Stillink is the common name of the product family. In this family, there are two models at present: Stillink 200 and Stillink 800. Stillink 200 has two E1 interfaces, whereas, Stillink 800 has eight E1 interfaces.

Xymphony is the operating system (or the firmware) of Stillink systems. The Xymphony is common for both models. Stillink systems are shipped out with the operating system Xymphony has already been installed at the factory.

Certain parts of the Xymphony which deliver particular functionality to the Stillink systems are technologically secured to prevent their unlicensed use. The licensing is done at factory based on the purchase ordering code. Ordering codes starting with s2 refers to the Stillink 200 model, whereas, s8 refers to the Stillink 800 model. Other characters of an ordering code refer to a particular application, that is, some specific licenses installed into the system at the factory.

Tips

Although the licenses are installed at the factory, systems are not pre-configured at the factory for any specific application. To illustrate, a Stillink may have been shipped with ISDN signaling licenses for four E1 interfaces, however the factory default configuration for E1s may be different. With some simple programming, signaling type of any four E1s can be selected as ISDN in the field.

INSTALLING THE STILLINK HARDWARE

Follow the previous sections of this manual for details. Hardware installation for both models are almost the same, except the number of E1 interfaces on these models differ.

Tips

E1 interface connectors in front of the unit are marked with the numbers 00, 01 for the Stillink 200 and 00, 01, 02, 03, 04, 05, 06, 07 for the Stillink 800.

00 refers to the lowest hardware addressed E1. In some documents including this one, the E1 interfaces may be also mentioned with the E1-prefix such that E1-00, E1-01...

The factory default access codes of the E1 interfaces, i.e., software addresses, which are frequently used in E1 programming are as follows:

The first 4 group of E1s, i.e., E1-00, -01, -02, -03

- For the E1 interface 00 or E1-00: Single factory default access code, which is 50000 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 00 or E1-00: Each channel has its own factory default access code, which are 50001 to 50031 if the signaling type is CAS (Channel Associated Signaling)
- For the E1 interface 01 or E1-01: Single factory default access code, which is 50100 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 01 or E1-01: Each channel has its own factory default access code, which are 50101 to 50131 if the signaling type is CAS (Channel Associated Signaling)
- For the E1 interface 02 or E1-02: Single factory default access code, which is 50200 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 02 or E1-02: Each channel has its own factory default access code, which are 50201 to 50231 if the signaling type is CAS (Channel Associated Signaling)
- For the E1 interface 03 or E1-03: Single factory default access code, which is 50300 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 03 or E1-03: Each channel has its own factory default access code, which are 50301 to 50331 if the signaling type is CAS (Channel Associated Signaling)

The second 4 group of E1s, ie., E1-04, -05, -06, -07

- For the E1 interface 04 or E1-04: Single factory default access code, which is 50400 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 04 or E1-04: Each channel has its own factory default access code, which are 50401 to 50431 if the signaling type is CAS (Channel Associated Signaling)
- For the E1 interface 05 or E1-05: Single factory default access code, which is 50500 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 05 or E1-05: Each channel has its own factory default access code, which are 50501 to 50531 if the signaling type is CAS (Channel Associated Signaling)
- For the E1 interface 06 or E1-06: Single factory default access code, which is 50600 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 06 or E1-06: Each channel has its own factory default access code, which are 50601 to 50631 if the signaling type is CAS (Channel Associated Signaling)
- For the E1 interface 07 or E1-07: Single factory default access code, which is 50700 if the signaling type is CCS (Common Channel Signaling)
- For the E1 interface 07 or E1-07: Each channel has its own factory default access code, which are 50701 to 50731 if the signaling type is CAS (Channel Associated Signaling)

Tips

Always use the lowest hardware addressed E1s to connect to network, where the Stillink would synchronize. To illustrate, in a Stillink 800 system with the ordering code s80.4i4s000 for quad E1 span SS7-ISDN conversion, it is almost the certain that the Stillink 800 would need to be in synchronous with the SS7 network. So that, the signaling on E1-00, E1-01, E1-02, E1-03 must be set to the SS7. The other E1s are set to the ISDN signaling and connected to the ISDN equipment. For further information about the synchronization, read the next sections.

CONNECT TO THE SYSTEM

When the Stillink system starts up, its factory default IP address is 10.0.0.12 and the port address is 80. There is no login ID or password for web browser connection.

Connect to your Stillink system with using any web browser (like Internet Explorer, Mozilla Firefox) for maintenance and programming.

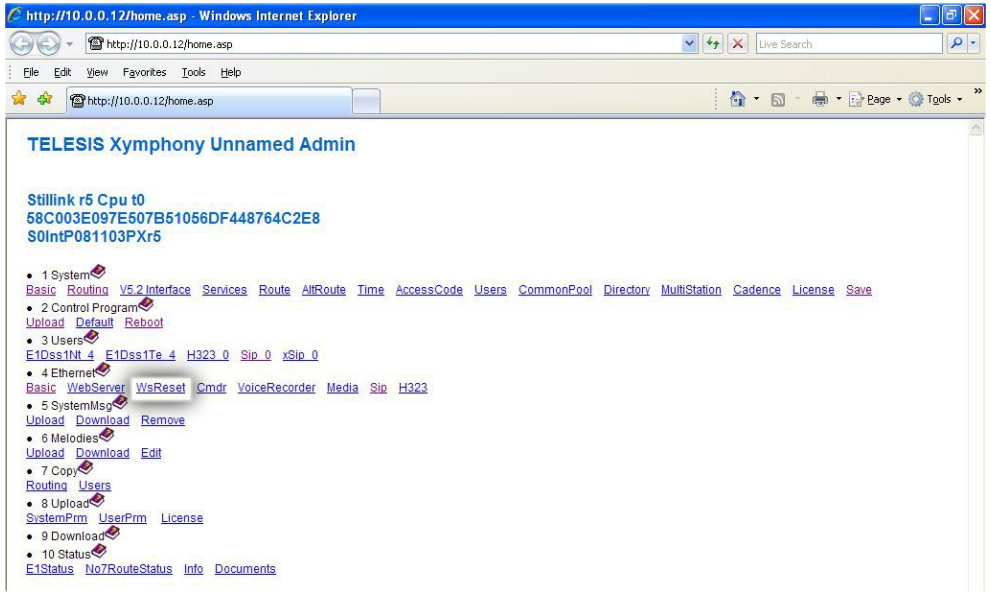
On-line help system embedded in the Stillink system will also guide you during the programming.

RESET THE WEB SERVER PARAMETERS

Assume that you have programmed the Stillink system by editing many parameters including web server settings and saved these. For any reason, Login ID, password and HTTP port number have been forgotten and it is not possible to reconnect to the web server of your Stillink system. And, you need to reconnect to the system (or reset the web server settings) without loosing any other parameters, which are previously set.

First step is to restart the Stillink system with factory default settings (see the next topic). When the system starts up, the factory default IP address of the system is 10.0.0.12 and the port address is 80. There is no login ID or password for web browser connection.

Connect to the system with a web browser (10.0.0.12, port 80, no login ID and password). Operate the *WsReset* command or item in Homepage.



IMPORTANT: NEVER OPERATE THE SAVE COMMAND AFTER WSRESET. OTHERWISE, PRE-PROGRAMMED PARAMETERS ARE LOST. YOU MUST REBOOT THE SYSTEM JUST AFTER CONFIRMING WSRESET.

Next step is to reboot (or power down and up) the system.

The system starts up with all the parameters, which are previously programmed, but:

- without any Login ID (name)
- without any password
- and HTTP port as 80

REBOOTING THE SYSTEM, PROGRAMMED PARAMETERS, FACTORY DEFAULTS

Memory Locations for Programmed Parameters:

The programmed parameters are stored in two locations in the Stillink system:

- the nonvolatile storage memory,or
- the volatile operational memory (RAM)

Any parameter, modification, except stated explicitly, is written to the volatile RAM and activated immediately. Without saving parameters to nonvolatile memory, resetting, restarting, rebooting the Stillink system will result in non-recoverable loss in modifications.

Once stored into the nonvolatile memory, the stored parameters are downloaded back to the RAM during a normal restart or power-up (except factory restart procedures, which download factory default values into the RAM).

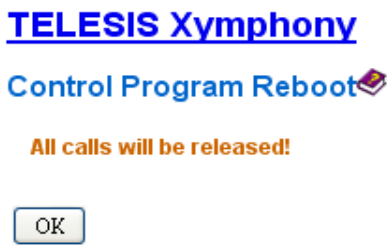
Restarting the System with Factory Defaults:

The parameter (**param**) button on the front panel of Stillink is to restart the system with factory default settings. To reset the parameters, the Stillink system is powered up with the parameter button pressed hold. Be aware that the parameter button must be hold pressed until the status led, which is also on the systems blinks.

Parameter (Param) button and Status (Status) led in the front panel

When the **Status** led blinks, i.e., the Stillink system starts up, the parameter button is released. The factory default IP address of the system is 10.0.0.12 and the port address is 80. There is no login name or password for web browser connection.

Similarly, if the system is rebooted with the web browser command *Control Program Reboot* and if the parameter button pressed hold until the status led blinks, again the system starts up with the factory defaults.



Note that the factory default parameters are restored into the operational memories when the above mentioned steps are applied. The previously programmed parameters (if any) are still in the nonvolatile memories until the web browser command *Save* is operated.

Restoring the Programmed Parameters:

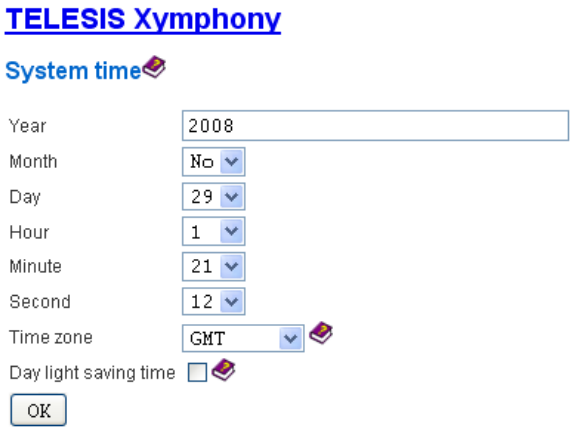
Whenever the *Save* command is operated, all the parameters in the operational memories are transferred into the nonvolatile memories.

This being the case, it is possible to restore the programmed parameters unless the *Save* command has been operated. To do this, just power up or reboot the system without any button pressed hold. Then, any parameters in the non-volatile memories are loaded into the operational memories and the system will start with these parameters.

SETTING SOME SYSTEM-BASED PARAMETERS

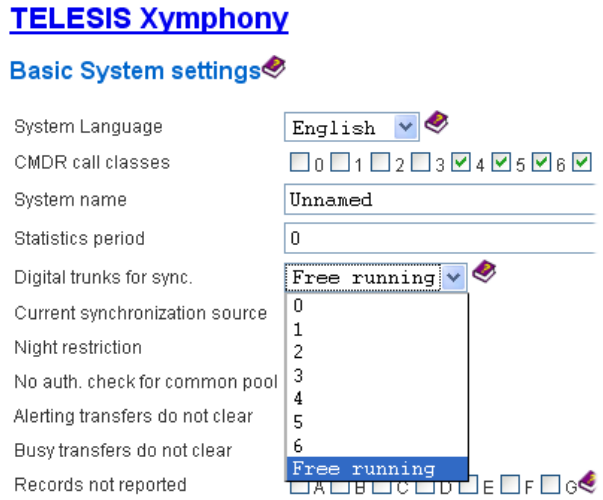
A name to a particular Stillink system may be given with editing *System name* field. This is optional. The name does not have any effect in functions or in operation. Naming may be preferred to distinguish individual systems in a multi-system environment.

The system time and date can be adjusted. This is also optional except encrypted VoIP communication applications.



For a proper operation of E1 interfaces, a Stillink system may be required to extract the clock information from the TDM network, where it is connected. A Stillink may be programmed from which E1 interface the clock has to be extracted. The programming is done with making a selection from *Digital trunks for synch* selection box. Options, which can be selected are:

- 0: Synchronize to the E1-00
- 1: Synchronize to the E1-01, if this synchronization cannot be established, try to synchronize to the E1-00
- 2: Synchronize to the E1-02, if this synchronization cannot be established, try to synchronize to the E1-01. If it again fails, try to synchronize to the E1-00
- 3: Synchronize to the E1-03. If any failure occurs, try the next lower hardware addressed E1.
- .....
- Free Running: The system generates its own clock, there is no clock extraction or synchronization.



Tips

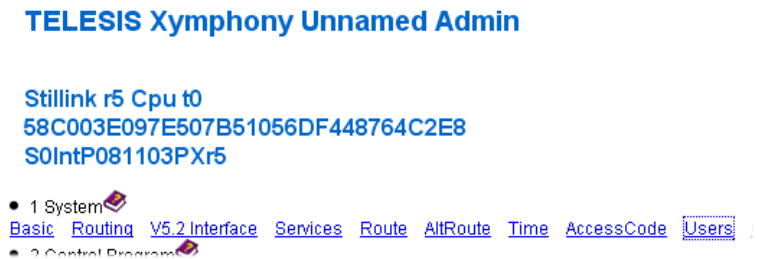
If a synchronization cannot be established in neither of the E1 interfaces, the system will be in *Free running* mode.

SETTING THE SIGNALING TYPE FOR A PARTICULAR E1

The factory default signaling type for E1 interfaces is DSS1 Euro ISDN. In Homepage *Users* paragraph, installed ports can be viewed.

Selecting another signaling type for an E1 interface with the default DSS1 signaling is so easy. For example, in order to set E1-00 as an SS7 signaling interface and E1-04 as an QSIG NT signaling interface in a Stillink 800 with the ordering code s80.4i4s000 , simply do the following:

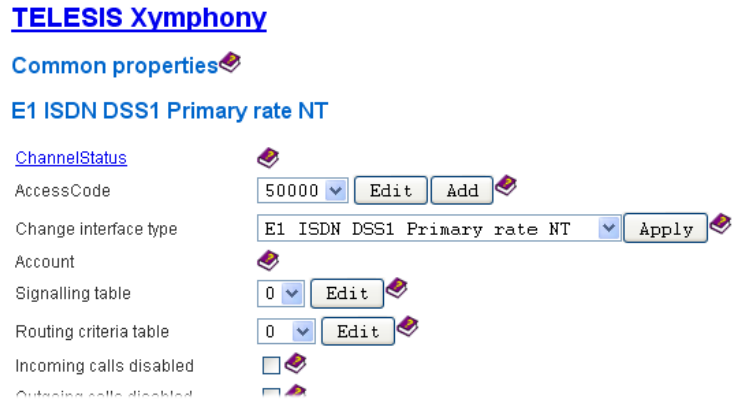
- In Homepage, click on *Users* item.



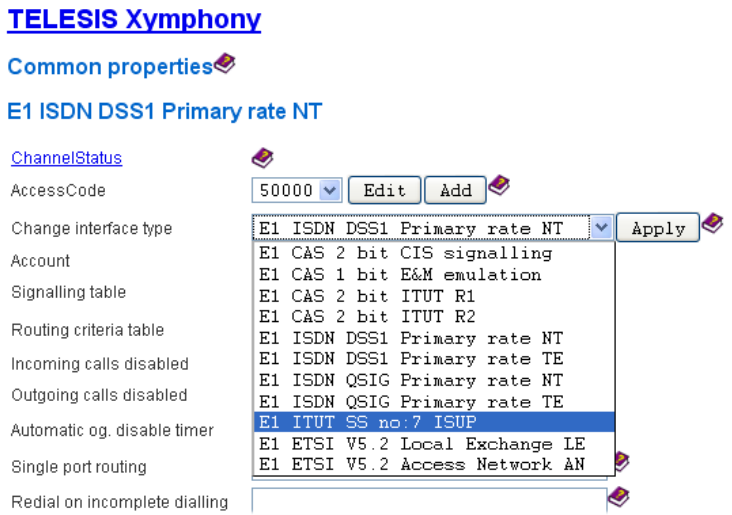
- *Select to edit* page is displayed.

- Since, all E1 interfaces by default are with DSS1 signaling, their access codes are 50000, 50100, 50200.... for E1-00, E1-01, E1-02..... respectively.
- For E1-00, edit 50000 into the textbox.

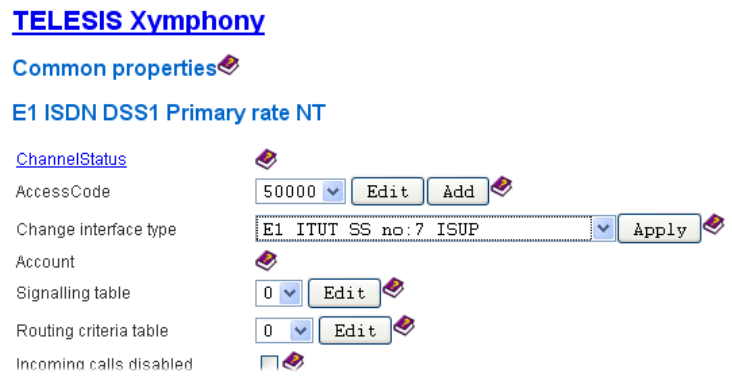
- Click on *EditCmn* button. *Common properties* page is displayed.



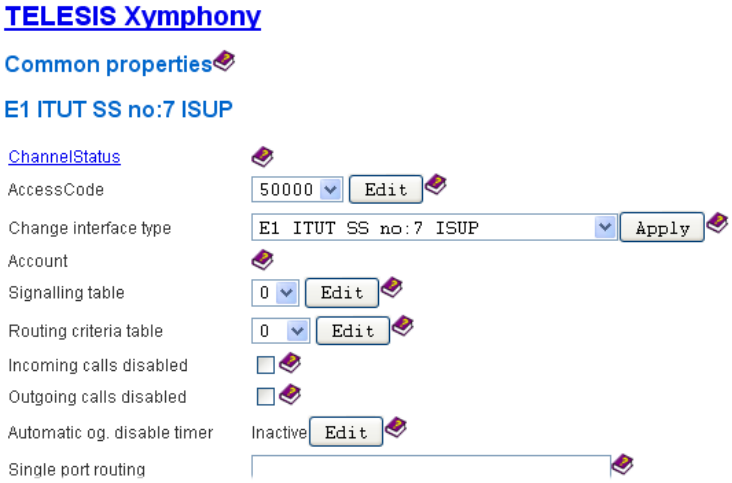
- Select *E1 ITUT SS no.7 ISUP* from the *Change interface type* selection box..



- Then click on the *Apply* button..



- The page will refresh and it is DONE.



Similarly, in order to set the interface E1-04, repeat the same.

- In Homepage, click on *Users* item.
- *Select to edit* page is displayed.
- For E1-00, edit 50400 into the textbox.
- Click on *EditCmn* button. *Common properties* page is displayed.
- Select *E1 ISDN QSIG Primary rate NT* from the *Change interface type* selection box.
- Then click on the *Apply* button..
- The page will refresh and it is DONE.

Tips

Setting the signaling type is not the last step for programming an E1 interface. Depending on the signaling type, further programming might be required for a proper communication with the TDM network connected.

Some of the signaling types may require E1 interfaces to be restarted. It can be simply done by disconnecting and connecting again the physical E1 wire (connector) in front of the Stillink system for the relevant E1 interface.

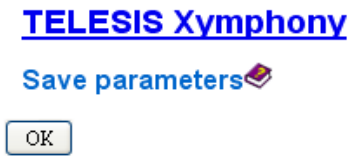
If a signaling type of an E1 interface is modified from a CCS type to a CAS type, the access code of this E1 will also change, such that each voice channel will have its own factory default access code, which are 50?01 to 50?31.

If a signaling type of an E1 interface is modified from a CAS type to a CCS type, the access code of this E1 will also change or be simplified, such that:

- For the E1-00, the default access code will be simplified to 50000.
- For the E1-01, the default access code will be simplified to 50100.
- For the E1-02, the default access code will be simplified to 50200.
- For the E1-03, the default access code will be simplified to 50300.
- For the E1-04, the default access code will be simplified to 50400.
- For the E1-05, the default access code will be simplified to 50500.
- For the E1-07, the default access code will be simplified to 50600.
- For the E1-07, the default access code will be simplified to 50700.

In other words, voice channels in an E1 interface with a CCS type signaling do not have individual access codes. Instead, the interface itself has an access code.

Whenever the *OK* or *Apply* button is clicked in a page, the modified parameters in the relevant pages are sent to the operational memories of the Stillink. Be aware that the operational memories are RAMs, so that, they can be lost if the the Stillink is powered down. In order to store these programmed parameters to the nonvolatile memories of the system *Save* command in Homepage should be operated. For further information, read the relevant sections of the online help system.



FURTHER PROGRAMMING FOR THE SIGNALING ON AN E1

After setting the signaling type for a particular E1, several other parameters may be also be needed to adjust for a proper communication with the network connected. To illustrate, the SS7 signaling would require the adjustment of point codes (origin and destination), link, CIC prefixes etc. It is expected that the Stillink user has the basic network-signaling knowledge and he is aware about the parameters already set in the network where the Stillink to be connected. The adjustable signaling parameters for an E1 depends on the signaling type selected. To access the adjustable signaling parameters, the user should go into the common properties pages of the relevant E1.

Tips

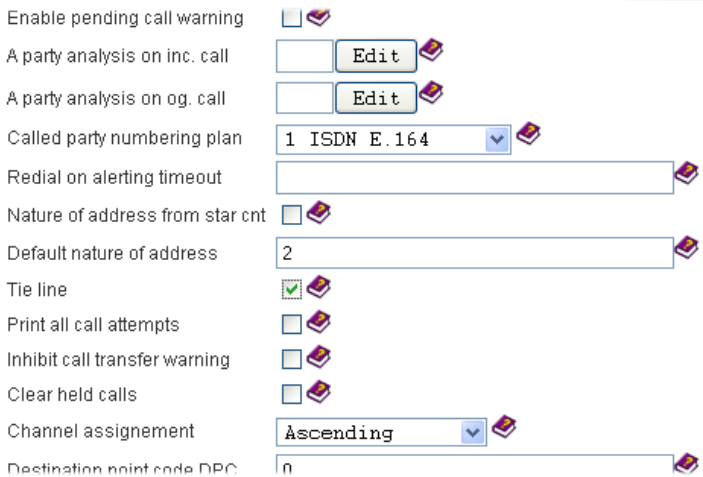
Be aware that for signaling types like DSS1, QSIG, the factory default settings in the Stillink might be sufficient and the proper communication might immediately start. However, for some other signaling types and special applications, some parameter adjustments may be necessary. It is strongly recommended to read the relevant sections in online help system.

In SS7 signaling, the point codes for the SS7 link should be entered as decimal numbers in the Stillink side.

SETTING CAS, SS7, and ISDN SIGNALING E1s as TIE-LINEs

Set all E1 interfaces with CAS, SS7, and ISDN (DSS1 or QSIG) signaling as Tie-Lines. To do this:

- Go to *Common properties* pages of the relevant access code.
- Mark the *Tie Line* checkbox.
- Click on the *OK* button.



Tips

Marking interfaces *Tie-Line* results in: the calling party addresses in INCOMING SETUP messages to be sent as the calling party addresses in OUTGOING SETUP messages. For some applications and signaling types, the Stillink may use the system access codes as the calling party addresses in outgoing setup messages unless interfaces are marked as tie-line.

VERIFYING THE COMMUNICATION / SIGNALING ON AN E1

Thanks to the advanced signaling analyzer embedded into the Stillink systems, it is possible to monitor or analyze the signaling (protocol) on a selected E1 interface without requiring any additional and costly external analyzer. The signaling (protocol) analyzer is capable of analyzing various events and signaling layers. After programming the signaling parameters, and establishing a physical connection, the analyzer may be utilized to verify the communication.

Even if there are no active calls on an E1 (since the routing parameters have not been programmed yet), some messages will be exchanged in the idle state for most of the signaling types. Furthermore, activation of some layers , especially for a CCS type signaling, may be a healthy indicator for a proper communication. To illustrate, exchange of RR messages in ISDN signaling or exchange of FISU messages in SS7 signaling would indicate that the basic programming for the signaling ended with success.

To monitor a selected E1 interface, you may use the freeware Telesis Xport Utility.

CALL ROUTING

After the verification of the signaling on E1 interfaces, the next step is to adjust call routing parameters within the system. This is necessary to route calls from one interface to another.

The advanced routing algorithm within the Stillink systems is carried out in four steps:

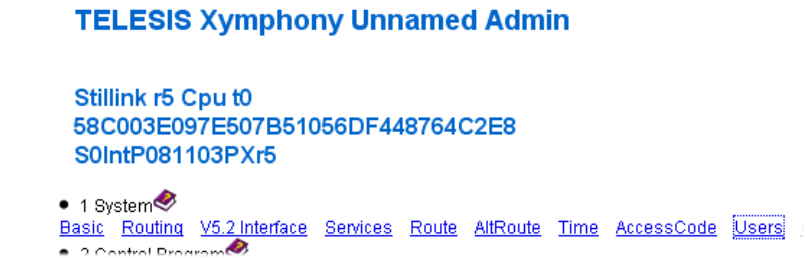
- First Step: Incoming A-Party Analysis
- Second Step: B-Party (called-party) Pre-Analysis
- Third Step: B-Party (called-party) Main Analysis
- Fourth Step: Outgoing A-Party Analysis

In most of the cases, the third step, which is the Main Analysis is sufficient to establish successful signaling conversions.

Transparent call routing (for beginners)

1:1 (one-to-one), or transparent call conversion may be achieved easily with using the Main analysis only. For example, in a Stillink 200 with the ordering code s20.1i1s000 (assume that you have already set the E1-00 as SS7 and E1-01 as ISDN DSS1 NT) do:

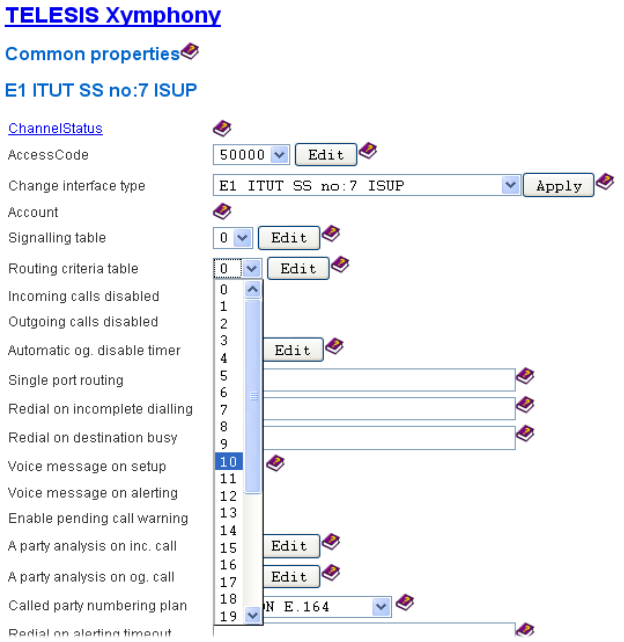
- In Homepage, click on *Users* item.

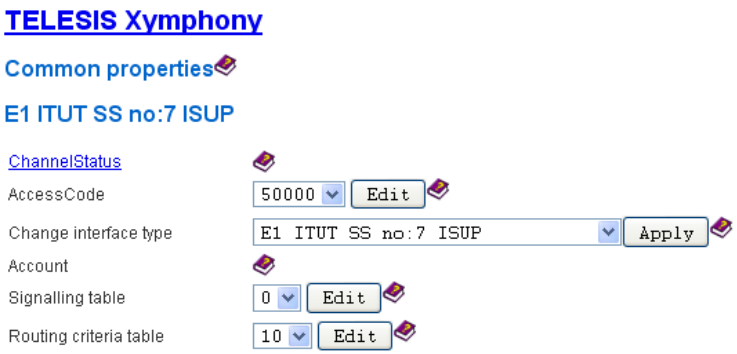


- *Select to edit* page is displayed.

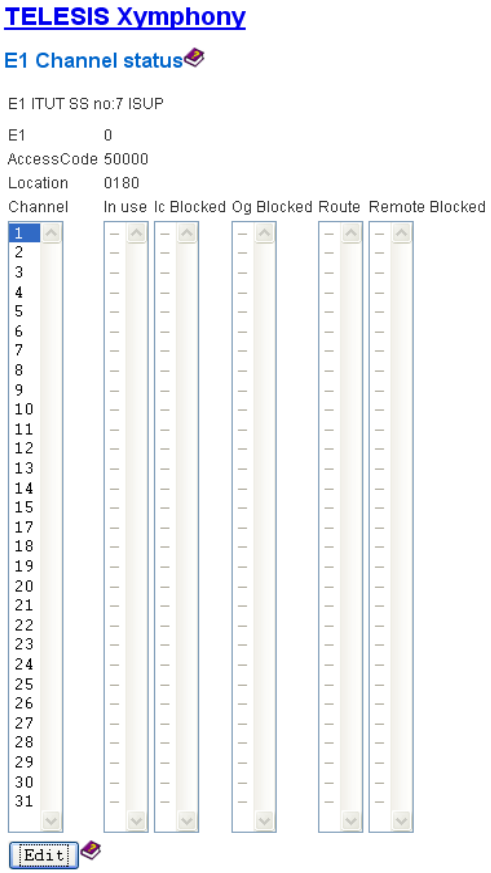
- For E1-00, edit 50000 into the textbox.

- Click on *EditCmn* button. *Common properties* page is displayed.
- Assign a routing table using *Routing criteria table* selector. Assume that 10 is assigned.

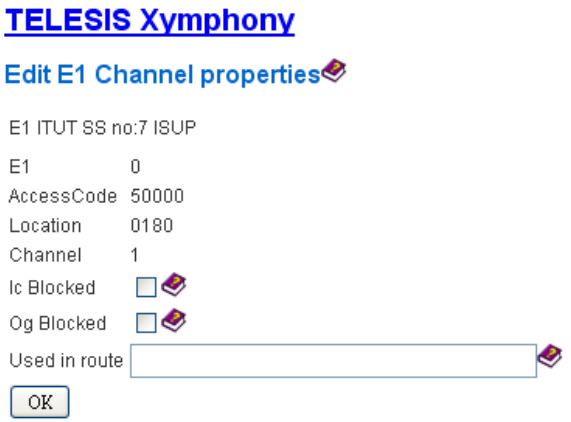




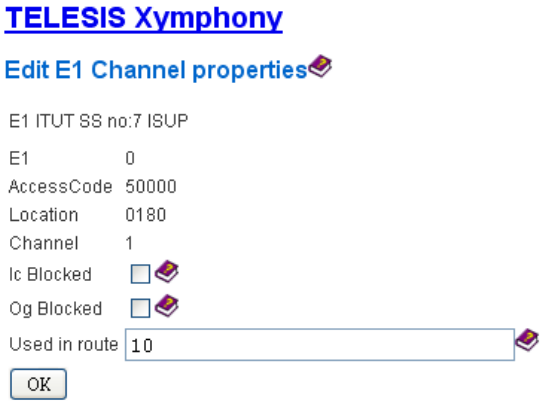
• Using the *Channelstatus* link in *Common properties* page, go to the page showing all the voice channels from 1 -31 for this E1-00. Select *Channel* 1 and then click on *Edit* button



• *Channel properties* of the voice channel 1 in E1-00 appears.



• Edit a route number for this voice channel into the *Used in route* textbox. Assume that 10 is edited. Then click on *OK* button.

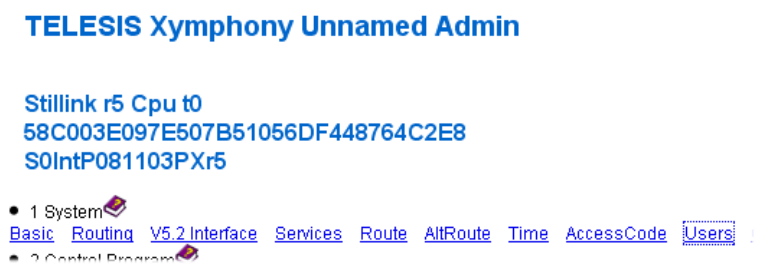


• Repeat the same for all the other voice channels.

WHAT IS DONE for E1-00? Its routing table is set to 10, and its route number is set to 10 for all of its voice channels.

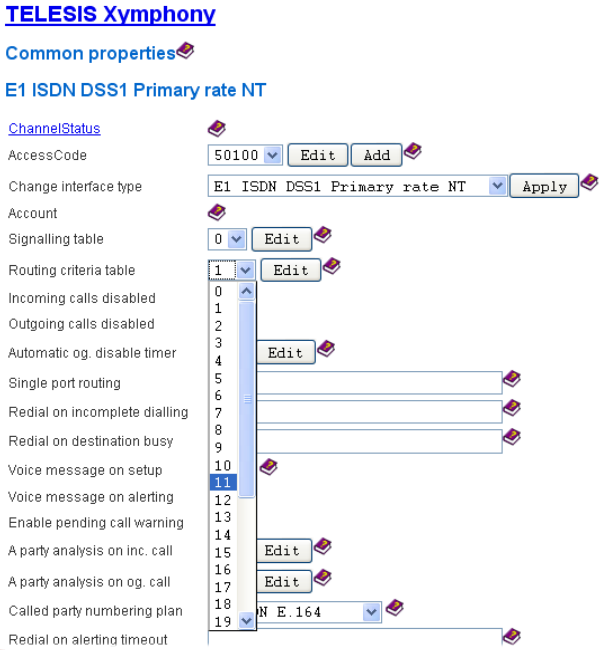
Similarly, in order to set routing parameters of the E1-01:

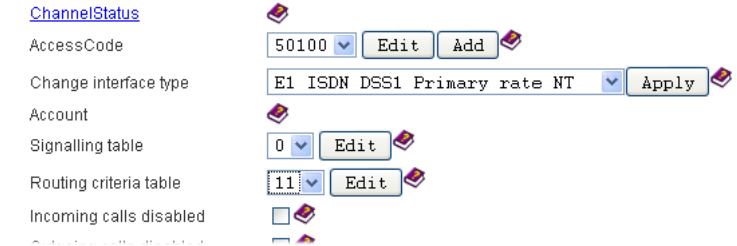
- In Homepage, click on *Users* item.



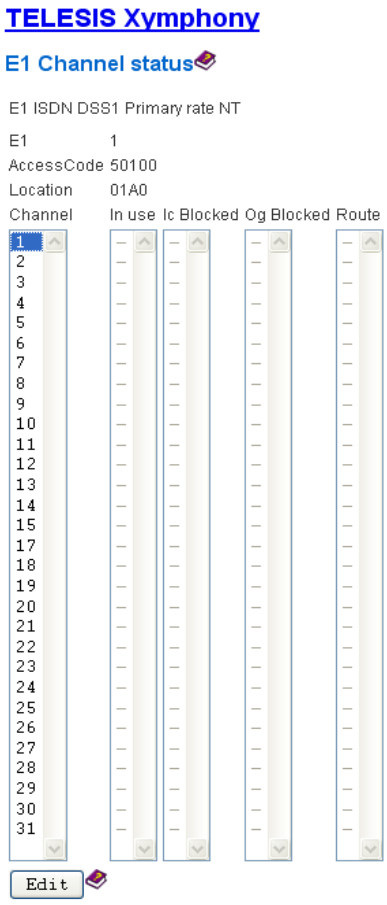
- *Select to edit* page is displayed. For E1-01, edit 50100 into the textbox..

- Click on *EditCmn* button. *Common properties* page is displayed. Assign a routing table using *Routing criteria table* selector. Assume that 11 is assigned.

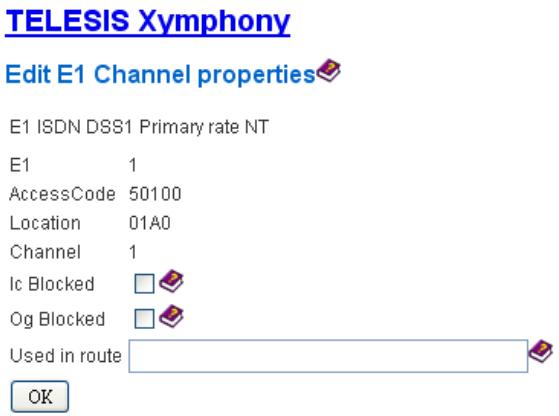




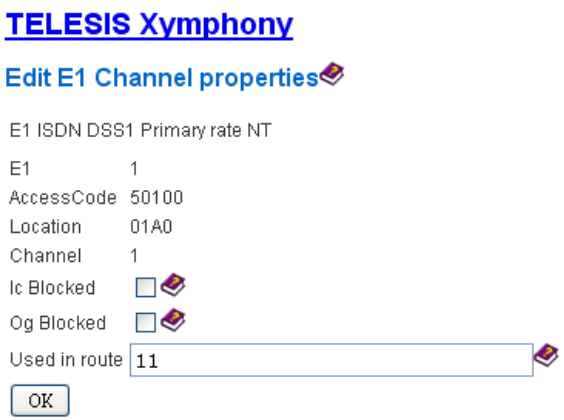
- Using the *Channelstatus* link, go to the page showing all the voice channels from 1 - 31 for this E1-01. Select *Channel* 1 and then click on *Edit* button



• *Channel properties* of the voice channel 1 in E1-01 appears.



• Edit a route number for this voice channel into the *Used in route* textbox.. Assume that 11 is edited. Then click on *OK* button.

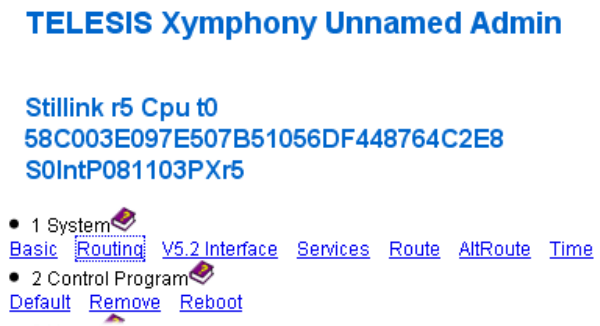


• Repeat the same for all the other voice channels.

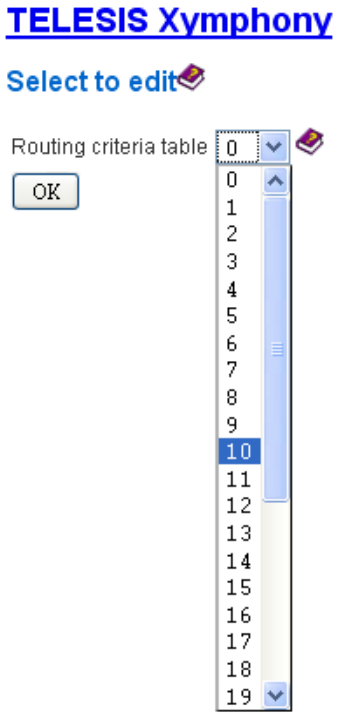
WHAT IS DONE for E1-01? Its routing table is set to 11, and its route number is set to 11 for all of its voice channels.

The next step is to fill out the routing tables 10 and 11. Start with programming the routing table 10.

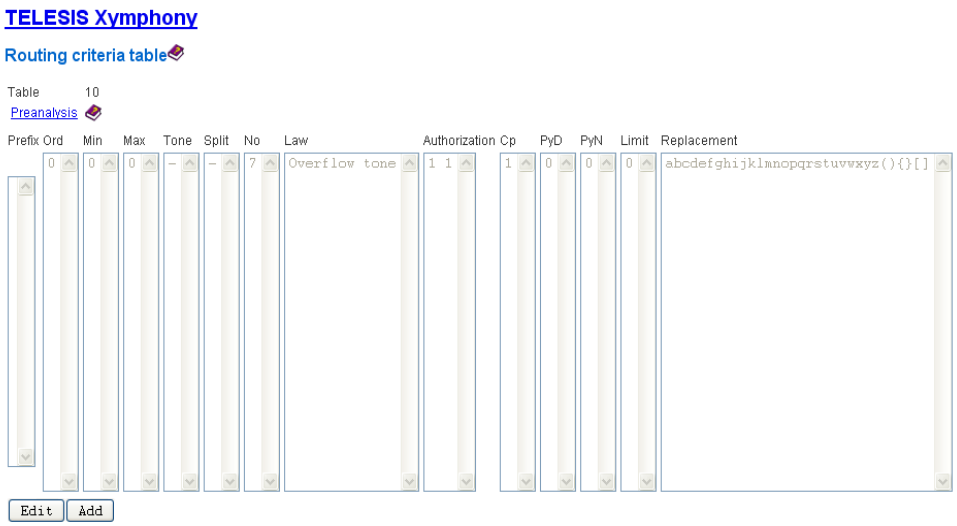
• In Homepage click on *Routing* item.



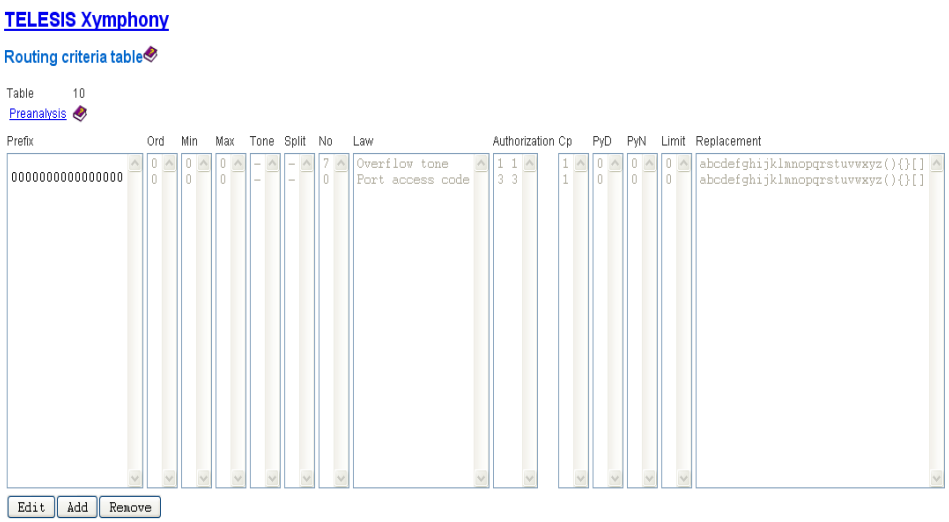
• *Select to edit* page is displayed.



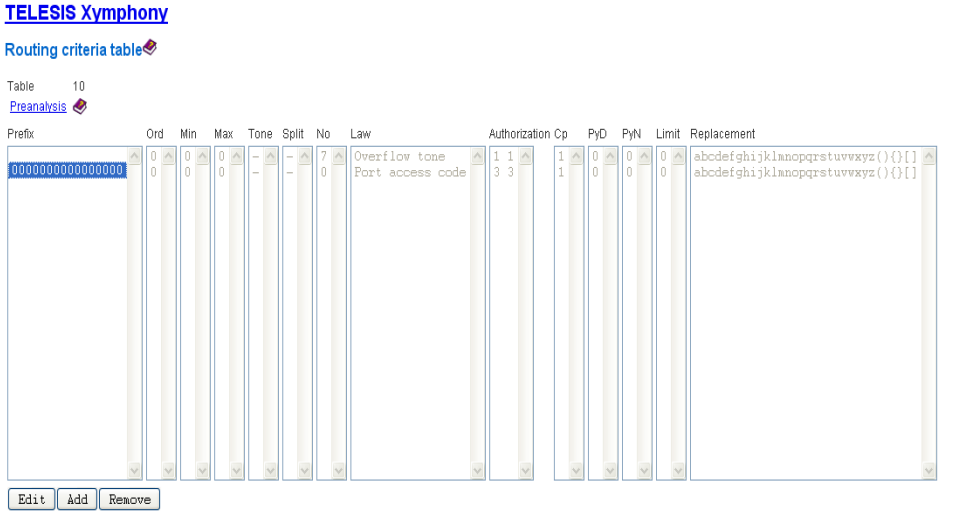
• Select *Routing criteria table* 10 and click on *OK* button. The table 10 is displayed.



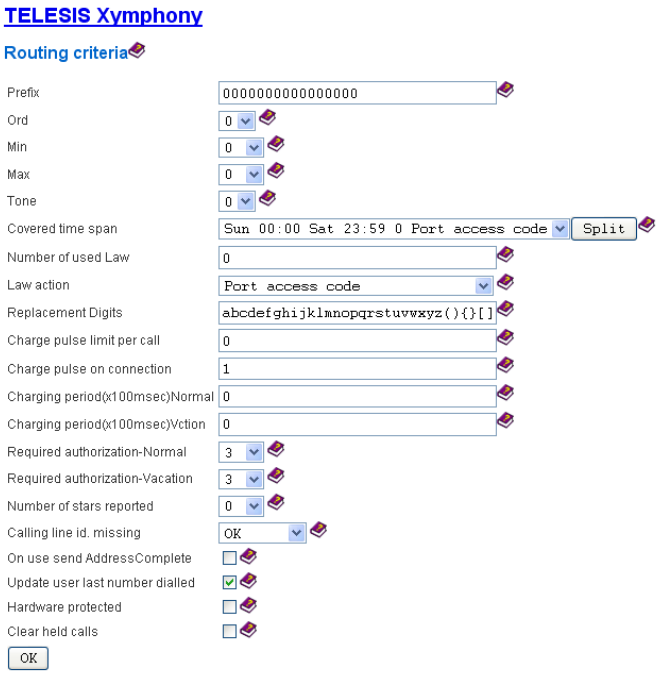
- Click on *Add* button to start editing an empty table.



- Mark the first row all with entries 0 and then click on *Edit* button.



- *Routing criteria* for this entry is displayed.



- Edit the *Prefix* as a single 0, *Min* as 3, *Max* as 31, *Number of used Law* as 300, *Law action* as R11 (route 11) , *Required authorization - Normal* as 0, *Required authorization - Normal* as 0.

TELESIS Xymphony

Routing criteria

Prefix	0
Ord	0
Min	3
Max	31
Tone	0
Covered time span	Sun 00:00 Sat 23:59 0 Port access code — Split
Number of used Law	300
Law action	R11
Replacement Digits	abcdefghijklmnopqrstuvwxyz(){}[]
Charge pulse limit per call	0
Charge pulse on connection	1
Charging period(x100msec)Normal	0
Charging period(x100msec)Vction	0
Required authorization-Normal	0
Required authorization-Vacation	0
Number of stars reported	0
Calling line id. missing	OK
On use send AddressComplete	☐
Update user last number dialled	☑
Hardware protected	☐
Clear held calls	☐

OK

- Click on *OK* button.
- *Routing criteria table* 10 is displayed with this new entry.

TELESIS Xymphony

Routing criteria table

Table 10

Preanalysis

Prefix	Ord	Min	Max	Tone	Split	No	Law	Authorization	Cp	PyD	PyN	Limit	Replacement
0	0	0	0	–	–	7	Overflow tone	1 1	1	0	0	0	abcdefghijklmnopqrstuvwxyz(){}[]
	0	3	31	–	–	300	R11	0 0	1	0	0	0	abcdefghijklmnopqrstuvwxyz(){}[]

Edit Add Remove

- Then click on *Add* button again.

TELESIS Xymphony

Routing criteria table

Table 10

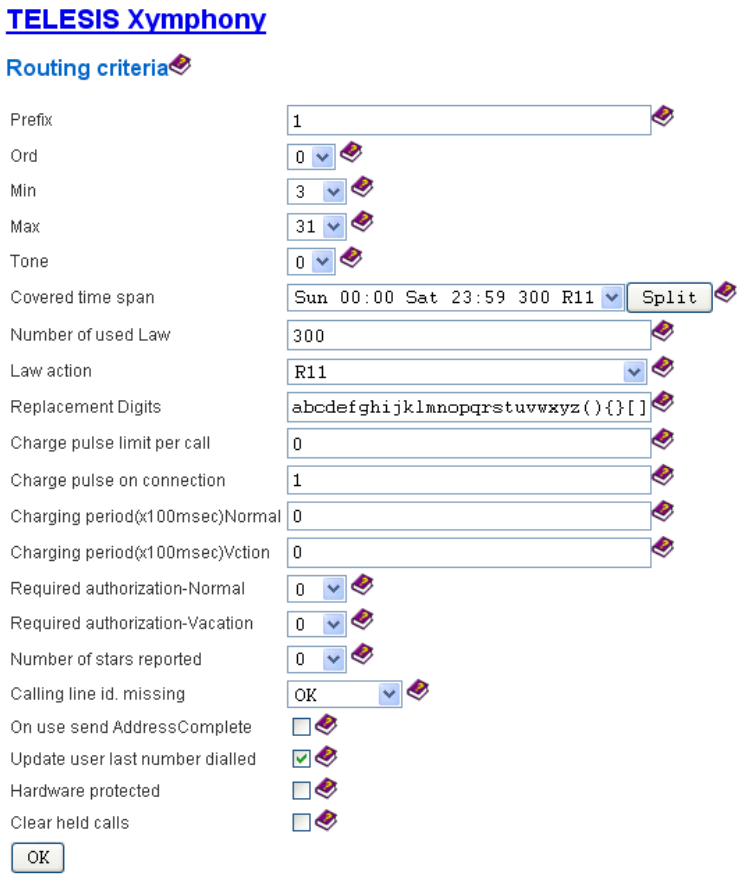
Preanalysis

Prefix	Ord	Min	Max	Tone	Split	No	Law	Authorization	Cp	PyD	PyN	Limit	Replacement
0	0	0	0	–	–	7	Overflow tone	1 1	1	0	0	0	abcdefghijklmnopqrstuvwxyz(){}[]
	0	3	31	–	–	300	R11	0 0	1	0	0	0	abcdefghijklmnopqrstuvwxyz(){}[]
0000000000000000	0	0	0	–	–	0	Port access code	3 3	1	0	0	0	abcdefghijklmnopqrstuvwxyz(){}[]

Edit Add Remove

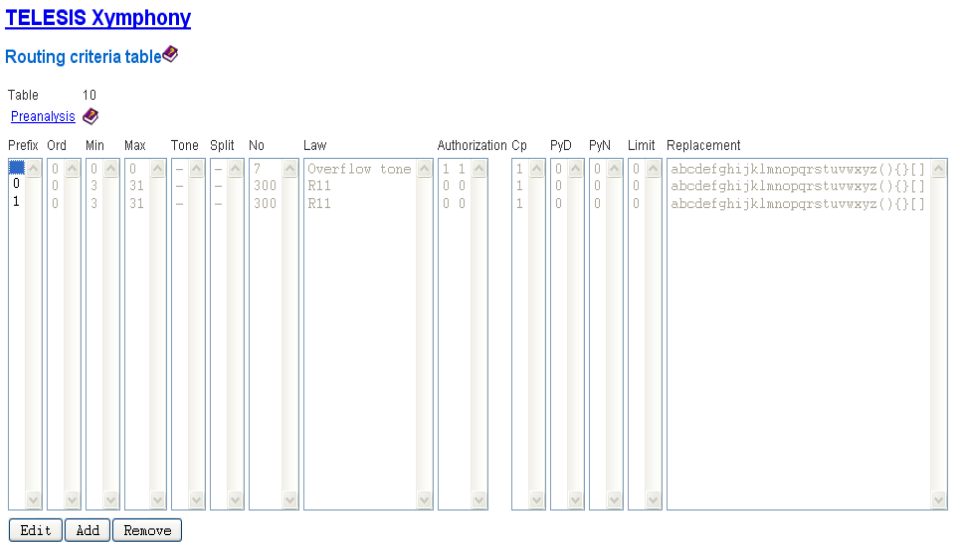
- Mark the new row with all zeros and then click on *Edit* button.
- *Routing criteria* for this entry is displayed.
- Edit *Number of used Law* as 300, then click on *OK* button
- The page will refresh

- Then edit the *Prefix* as a single 1, *Min* as 3, *Max* as 31,
- Note that *Law action* as R11 (route 11) , *Required authorization - Normal* as 0, *Required authorization - Normal* as 0 will be automatically set when *Number of used Law* is selected as 300

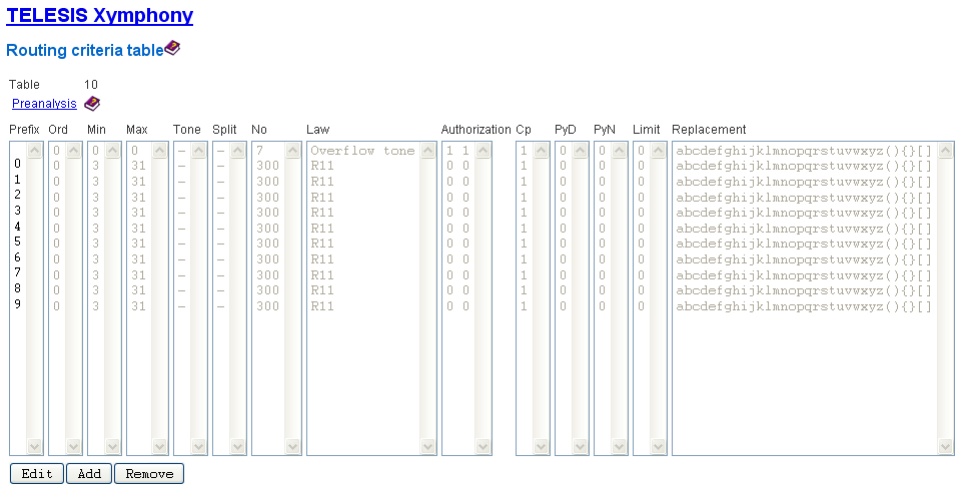


- Click on *OK* button.
- *Routing criteria table* 10 is displayed with this new entry.

- Repeat the same procedures to add prefixes 2,3,4,5,6,7,8,9 in to the *Routing criteria table* 10.

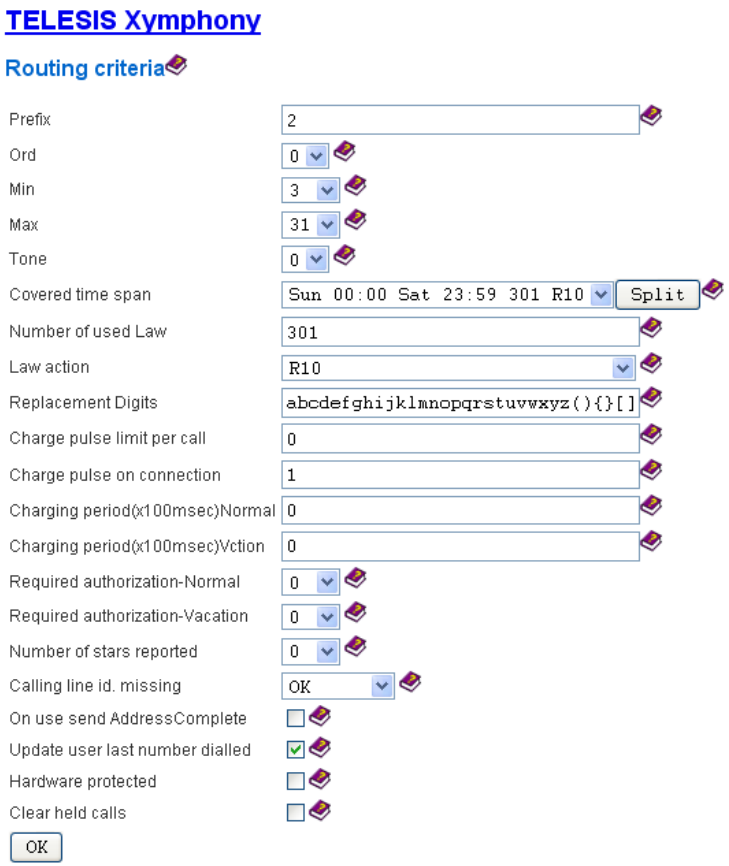


- At the end, the *Routing criteria table* 10 looks like below.

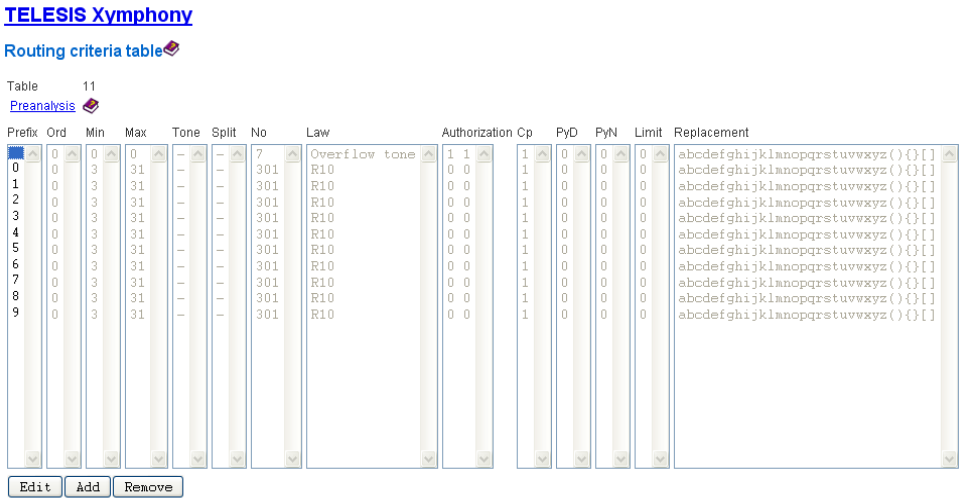


WHAT IS DONE for routing table 10? If any call with any called party address (minimum 3 digit and maximum 31 digit long) is received from an interface using this table, this call will be routed to the Port Route 11.

Next is to program the *Routing criteria table* 11 in a similar way. Repeat the same procedure mentioned for the *Routing criteria table* 10, except *Number of used Law* as 301, *Law action* as R10 (route 10). To illustrate, for the *Prefix* 2, the *Routing criteria* will look like below:



At the end of programming, the *Routing criteria table* 11 will look like below.



Tips

Since the *Routing criteria table* 11 will be very similar to the already programmed *Routing criteria table* 10, programming the table 11 may be started with copying all the contents of the table 10 over the empty table 11 at the beginning. This may save time in programming. To do this select *Copy Routing* items in Homepage.

Copy page is displayed. In this page, select 10 as the *Source* and 11 as the *Destination* and then click on *OK* button. Without doing any other modification, first set *Number of used Law* as 301 for all prefixes (0 to 9).

As the last step, edit the *Law action* as R10 (route 10) in one of the *Routing criteria* page (with the *Number of used Law* 301). Since all the other prefixes also use this Law 301, their *Law action* will be automatically set to R10.

WHAT IS DONE for routing table 11? If any call with any called party address (minimum 3 digit and maximum 31 digit long) is received from an interface using this table, this call will be routed to the Port Route 010

Summary for this sample programming:

- E1-00, that is SS no:7 ISUP signaling E1 interface with the access code 50000, is programmed such that
  - it uses the Routing Table 10.
  - all of its voice channels are set to Port Route 10.
- E1-01, that is ISDN DSS1 Primary rate NT interface signaling E1 interface with the access code 50100, is programmed such that
  - it uses Routing Table 11.
  - all of its voice channels are set to Port Route 11.
- Routing Table 10 routes all incoming calls to the Port Route 11
- Routing Table 11 routes all incoming calls to the Port Route 10

Result with this sample programming:

- Any incoming call with any called party address (minimum 3 digit and maximum 31 digit long) to the E1-00 is routed to the E1-01.
- Any incoming call with any called party address (minimum 3 digit and maximum 31 digit long) to the E1-01 is routed to the E1-00.

The above mentioned programming takes only several minutes after understanding the logic behind the routing algorithms in Stillink systems.

For further information, read the routing sections of the online help system.

Tips

Min and *Max* fields refer to minimum and maximum number of digits required prior to the conversion analysis. These fields may take values between 0 and 31. The analysis does not start before receiving the *Min* number of digits. If the entry in the *Max* field is greater than or equal to that of the *Min* field, the Stillink

- starts the main analysis after receiving the Min number of digits and
- routes the call immediately after the receipt of Max number of digits or the time-out occurrence.
- *Min* and *Max* numbers should be selected carefully for a proper call routing (conversion).

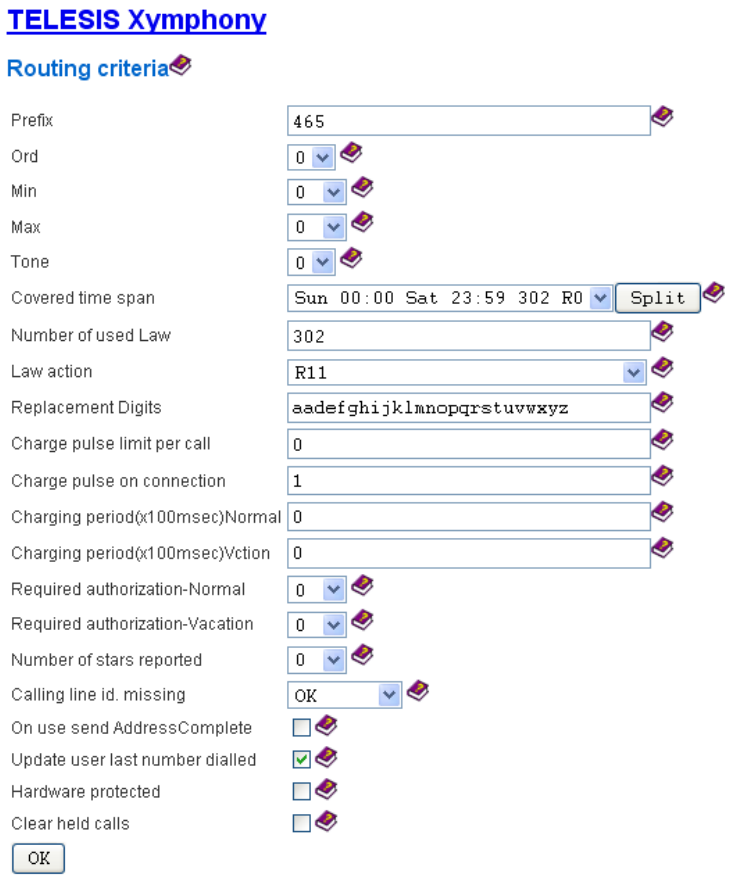
Routing criteria tables, *Law actions*, and *Number of used Laws* are common and unique within the same Stillink system. In other words, there is only one *Number of used Law* 300 in a system, which can be used by a single interface or shared by many other interfaces. So that, when creating or modifying *Routing criteria tables*, *Law actions*, and *Number of used Laws*, **BE SURE** that is not used by some other interfaces and modifications will not have an effect on other interfaces, which are already under the operation.

Replacing the called party address in conversion (for advanced users and applications)

Called party address digits received from an interface can be modified (or replaced) in routing this call to another interface. This replacement is done in *Replacement Digits* field of the *Routing Criteria* page. Each alphabetical character in *Replacement Digits* field represents a digit received from the incoming interface. For example, **a** is the first, **b** is the second and **z** is the twentysixth digit. (,) , { , } , [,] characters represent the digits between twentyseven and thirtytwo. A repeated character means, that particular digit will be repeated too, and if a character is deleted, this digit will be eliminated. By default, the Replacement Digits field is filled with complete series of abcdefghijklmnopqrstuvwxyz(){}[] . That means any called party address received from an incoming interface will be transparently routed.

Example-1:

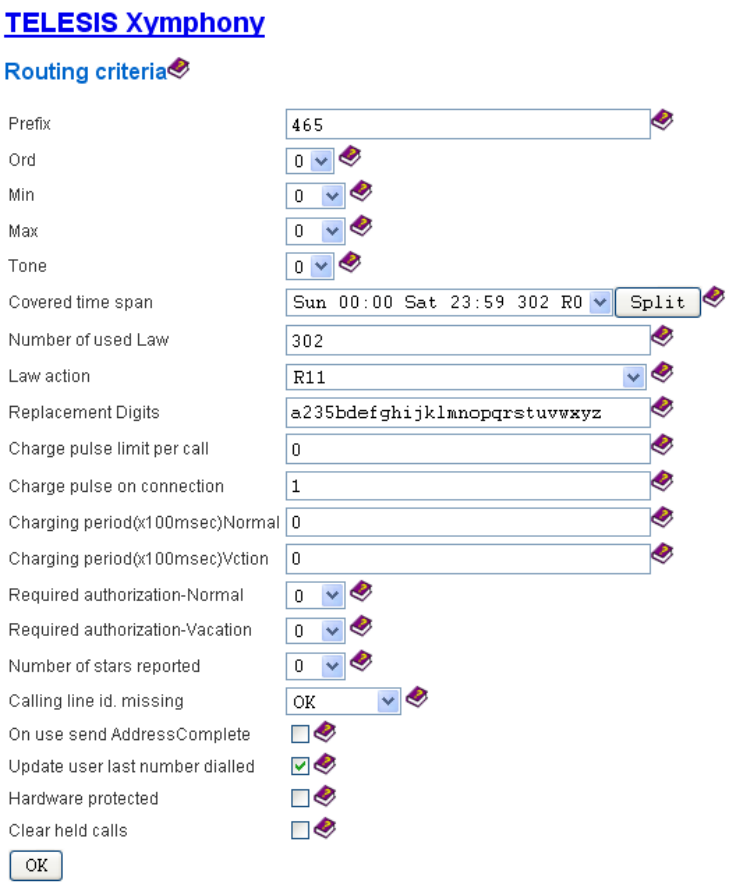
The *Prefix* field in the *Routing criteria* page, which is used by the E1-00 has the digit sequence of 465. The *Replacement Digits* field has the character sequence of aadefghijklmnopqrstuvwxyz , and the *Law action* R11 refers to E1-01.



When a call comes to the E1-00 with the called party address 4652293, this call is routed to the E1-01 but the called party address is modified as 442293. To analyze in detail, received called party address is 4652293, where 4=a, 6=b, 5=c, 2=d, 2=e, 9=f, 3=g. The routing law orders to repeat a, eliminate b and c, keep the others.

Example-2:

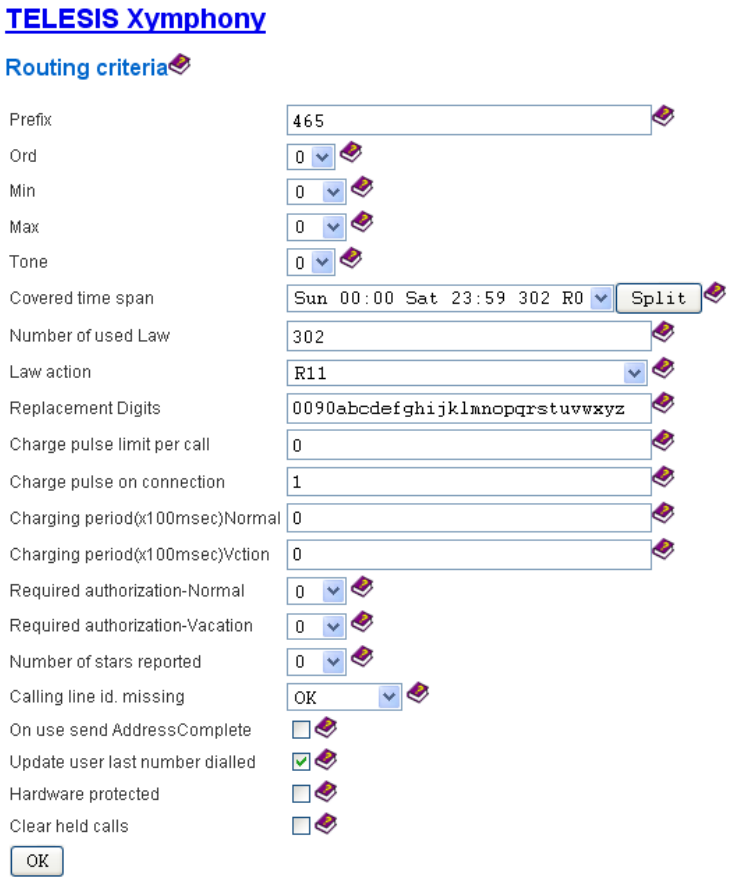
The *Prefix* field in the *Routing criteria* page, which is used by the E1-00 has the digit sequence of 465. The *Replacement Digits* field has the character sequence of a235bdefghijklmnopqrstuvwxyz, and the *Law action* R11 refers to E1-01.



When a call comes to the E1-00 with the called party address 4652293, this call is routed to the E1-01 but the called party address is modified as 423562293.

Example-3:

The *Prefix* field in the *Routing criteria* page, which is used by the E1-00 has the digit sequence of 465. The *Replacement Digits* field has the character sequence of 0090abcdefghijklmnopqrstuvwxyz, and the *Law action* R11 refers to E1-01.



When a call comes to the E1-00 with the called party address 4652293, this call is routed to the E1-01 but the called party address is modified as 00904652293.

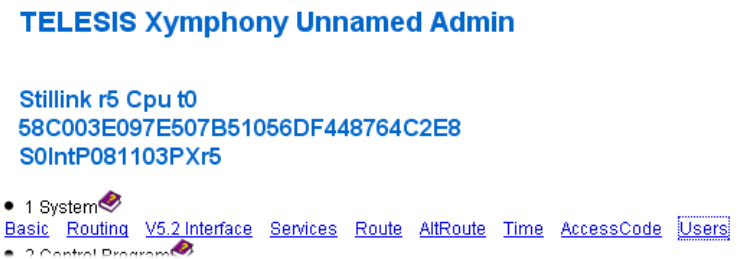
Replacing the calling party address in conversion (for advanced users and applications)

Calling party address digits and the other calling party information elements (if any) received from an interface can be modified (or replaced) in routing this call to another interface. This replacement is done in the incoming interface or in the outgoing interface.

Replacing the calling party addresses in the outgoing interface (just before routing out the call) is more usual and sufficient in most of the applications. This replacement is done in the A party analysis table of the relevant interface.

To illustrate the programming, assume a Stillink 200 with the ordering code s20.1i1s000 (the E1-00 is set to SS7 and E1-01 is set to ISDN DSS1 NT). The request is to add the digit sequence 0090 in front of every calling party address starting with the digit 4 in making conversion from ISDN DSS1 NT to SS7. In other words, if a SETUP message with the calling party address 4..... is received from the ISDN E1-01, this call will be routed to the SS7 E1-00 but the IAM message will include the calling party address modified such 00904..... To do this:

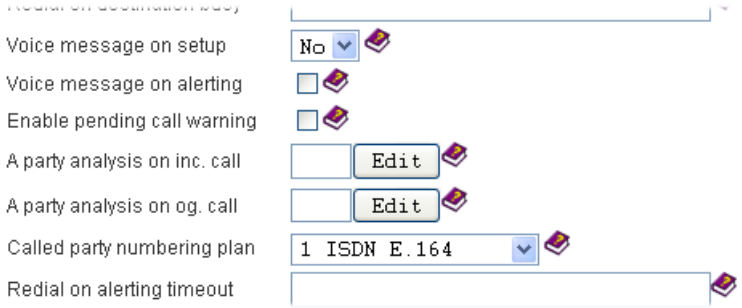
- In Homepage, click on *Users* item.



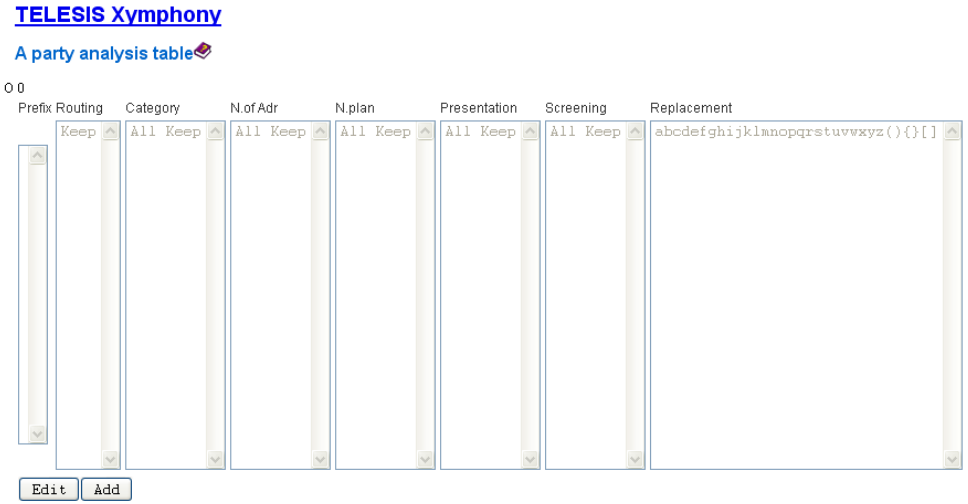
- *Select to edit* page is displayed

- For E1-00, edit 50000 into the textbox.

•Click on *EditCmn* button. *Common properties* page is displayed. *Common properties* page has a row of *A party analysis on og. call.*

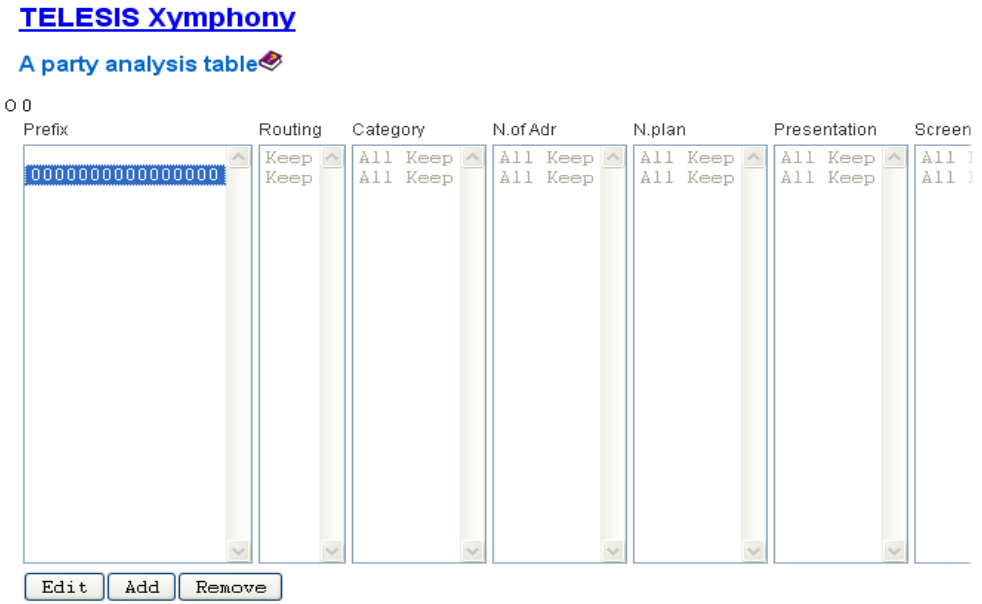


• Edit 0 into he textbox right to the *A party analysis on og. call* .Then click on *Edit* button on the right. The table 0 opens for editing as shown below

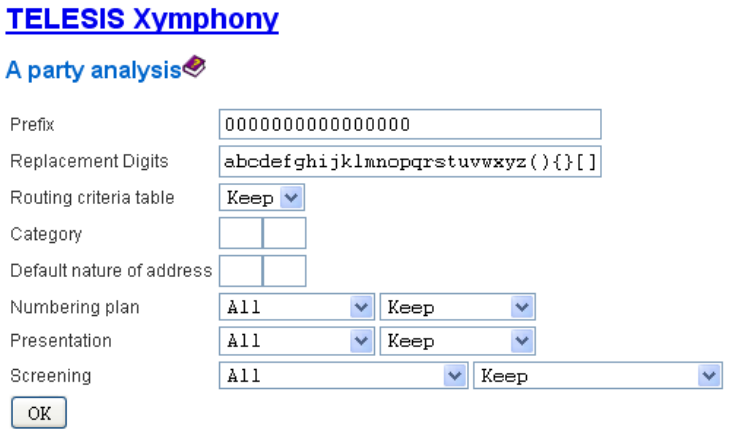


• Click on *Add* button to start editing an empty table. The page will refresh and have a row with all zeros.

• Mark this row with entries 0 and then click on *Edit* button.

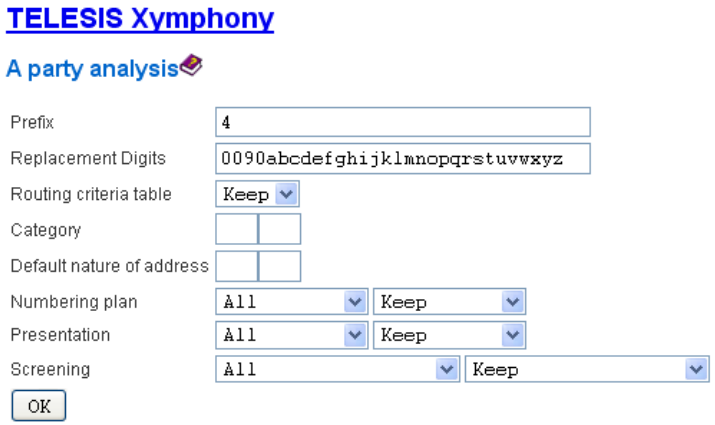


• *A party analysis page* is displayed.



• Edit 4 in *Prefix* field

• Add 0090 in the front of the character sequence abcdefghijklmnopqrstuvwxyz(){}[] in *Replacement Digits* field.



• Click on *OK* button.

WHAT IS DONE for E1-00? Its outgoing A party analysis table is set to 0, and whenever a call is routed out the E1-00, the digit sequence 0090 will be added in front of the calling party address if the original calling party starts with the digit 4.

As seen above, there is a similarity between the called party and calling part address analysis. In both analysis, there is a prefix field where the digit sequence to analyzed. The main result of the analysis is the replacement of the digits. The default replacement field is filled with complete series of abcdefghijklmnopqrstuvwxyz(){}[] . That means no replacement by default.

Tips

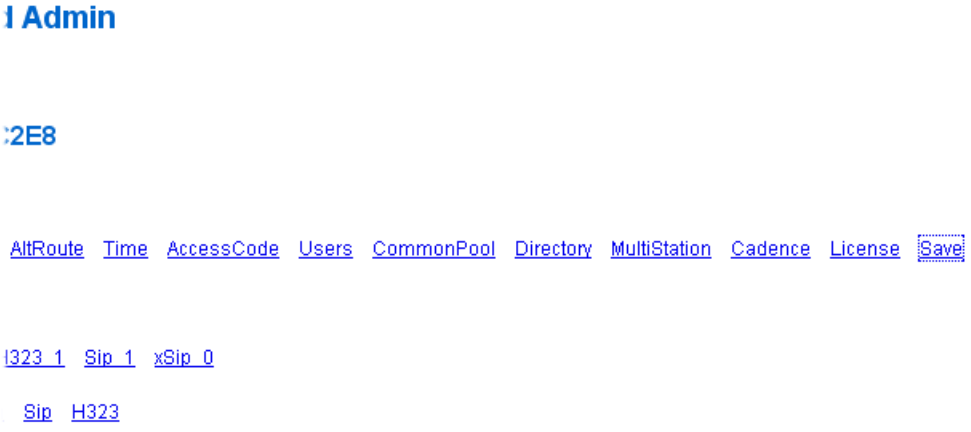
In addition to the calling party address analysis, A party analysis tables enable the analysis of the following calling party information elements if any:

- Category: received A-party category
- N. of Address: received A-party nature of address
- N.plan: received A-party Numbering plan
- Presentation: received A-party presentation status
- Screening: received A-party screening status

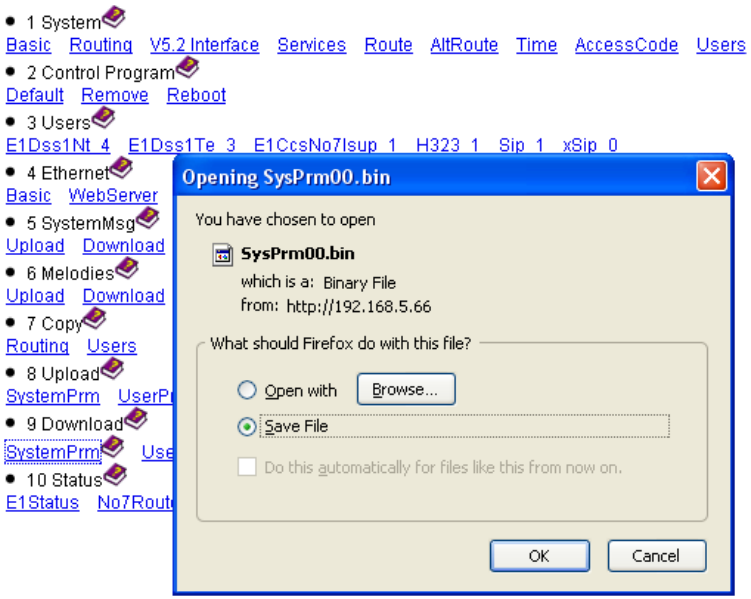
Note that these information elements in the tables may be used, used with a different notation, or not used in a particular signaling type. It is assumed that advanced users are aware about these information elements. After the analysis, the original calling party information elements may be modified. By default, the A party analysis tables Keep the original elements for All received information elements, i.e., transparent operation.

STORING AND BACKING UP PARAMETERS, FINAL STEPS

Whenever *OK* or *Apply* button is clicked in a page, the modified parameters in the relevant pages are sent to the operational memories of the Stillink. Be aware that the operational memories are RAMs, so that, they can be lost if the the Stillink is powered down. In order to store these programmed parameters to the nonvolatile memories of the system *Save* command in Homapage should be operated.



When the system programming is completed with success, it is recommended to make a back-up copy of all the system parameters. In order to back up the parameters to a file in the PC, select *Download* - *SystemPrm* command in Homepage.



COLD RESTART (RESETTING THE PARAMETERS TO FACTORY DEFAULTS)

Cold restart is rarely needed for a Stillink system. To illustrate such a rare case, the user may be confused after many mistakes done in programming. Consequently, starting with factory defaults may be easier and preferred.

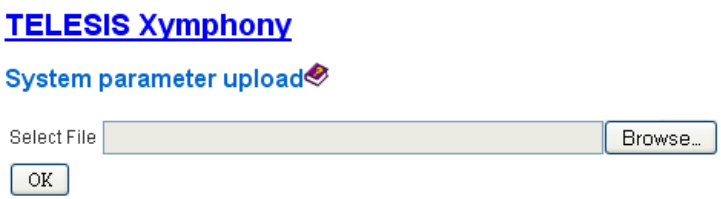
The parameter button on the front panel of the Stillink is to restart the system with factory default settings. To reset the parameters, the system is powered up with the parameter button pressed hold. Be aware that the parameter button must be hold pressed until the status led, which is also on the systems blinks. When the status led blinks, i.e., the system starts up, the parameter button is released. The factory default IP address of the system is 10.0.0.12 and the port address is 80. There is no login name or password for web browser connection.

Tips

Before the cold restart, it is recommended to make a back-up copy of all the system parameters for the possibility of reverting to the previously stored parameters.

Reverting to the previously stored parameters is possible with:

- powering down and up again the Stillink if and only if the *Save* command (in Homepage) has never been operated after the cold restart, or
- Select *Upload - SystemPrm* command in Homepage. to upload back-up file. After uploading, powering down and up again the Stillink.



Do not make any unnecessary cold restarts.

STILLINK SERVICE MODE

When is the Service Mode Necessary

- To upload a new Xymphony operational system (that is, operational software or operational firmware) into the Stillink
- To mark which of pre-installed Xymphony runs in normal operation mode
- To delete or remove a previously installed Xymphony from memories

These operations (except uploading a new Xymphony) may also be done (and preferred) in the normal Programming Mode. However,

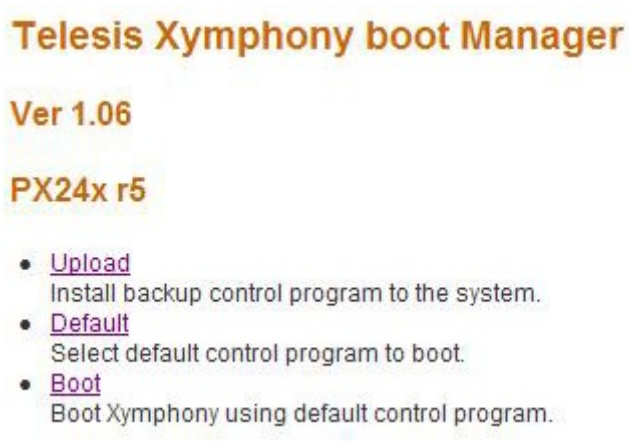
- If the system does not start up for any reason such as; an invalid Xymphony has been selected as the default boot program
- If Xymphony upload is required

then the Service Mode is a must.

Starting the Stillink in Service Mode

The **Service** button on the front cover of the Stillink is to start the system in service mode. To activate the service mode, the Stillink is powered up with the service button pressed hold. Be aware that the service button must be hold pressed until the status led, which is also on the front cover, blinks fast.

When the status led blinks fast, the service button is released. The IP address of the Stillink in service mode is the factory default address, which is 10.0.0.12 and the port address is 80. The web server of the Stillink waits for a web browser connection within 2 minutes in service mode. If there will be no connection in this period, the Stillink starts up in normal operation mode. So that, connect to the Stillink within 2 minutes with using a web browser (Internet Explorer, Mozilla Firefox, etc.) to access Homepage of the service mode. There is no login name or password for the service mode web browser connection.



Tips

Before upgrading the firmware Xymphony, it is strongly recommended to make a back-up copy of all the system parameters for future compatibility concerns, and for the possibility of reverting to the stored parameters.

Never power down the Stillink during the *Upload* operation, otherwise operational memories storing the firmware may be damaged.

Do not make any unnecessary firmware upgrades.

SOME APPLICATION SPECIFIC TOPICS

We recommend you to read first General Topics before proceeding with Application Specific Topics below.

PROGRAMMING ISSUES FOR THE V5.2 PROTOCOL CONVERSION

Setting the V5.2 protocol over E1

Assume an SS7 - V5.2 protocol conversion application with using a Stillink 800 (ordering code s80.4s4v000). To begin with, set E1-00, E1-01, E1-02, and E1-03 interfaces to CCITT signaling type no:7 ISUP and select the Digital trunks for synchronization. Adjust SS7 signaling parameters. Monitor the SS7 E1 interface (especially the one having the signaling channel D) and be sure about the communication there. Then, store the parameters into the non-volatile memories. Refer to General Topics for these programming.

Then, E1-04, E1-05, E1-06, and E1-07 should be set to V5.2 LE . In this Stillink system:

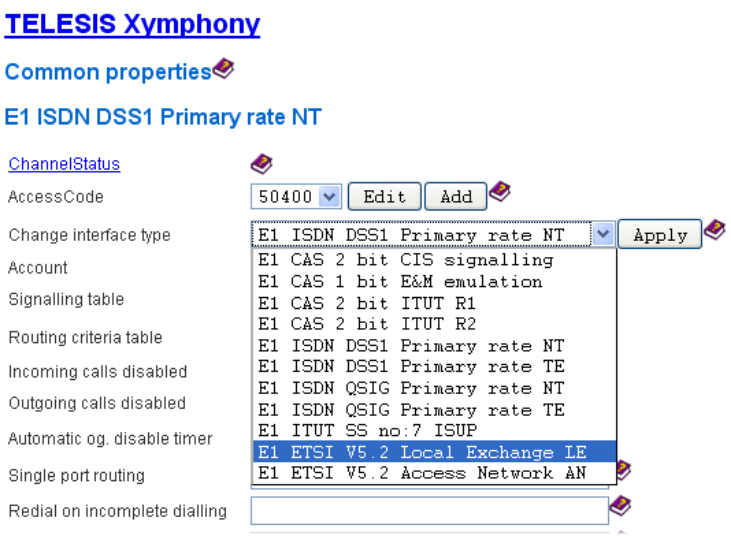
- All four E1s can be bundled into a single V5.2 interface. That is a single V5.2 interface having four E1 links, which are E1-04, E1-05, E1-06, and E1-07.
- Independent V5.2 interfaces can be created such as:
- Two V5.2 interfaces, each with 2 E1 links (total number of E1s equals to four), or
- Four V5.2 interfaces, each with a single E1 link (total number of E1s again equals to four), or
- Some other configurations.

However, the most frequent application is a conversion from a single V5.2 LE interface to the SS7. So as to bundle E1-04, E1-05, E1-06, and E1-07 into a single V5.2 interface, do the following:

Step-1

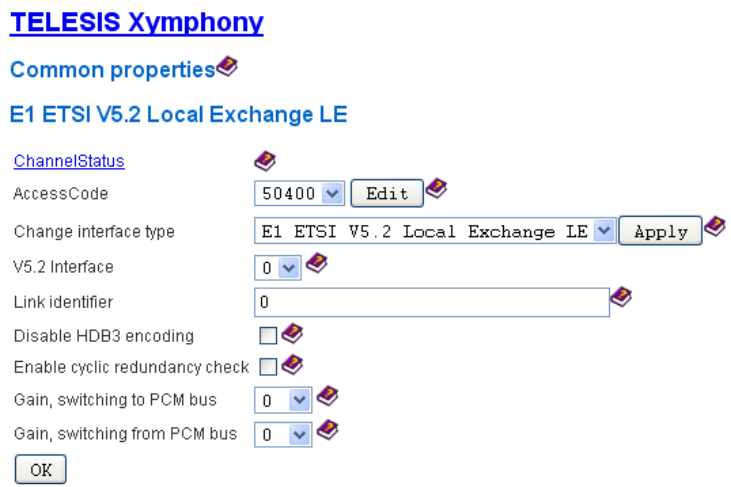
- In Homepage, click on *Users* item.
- *Select to edit* page is displayed.
- For E1-04, edit 50400 into the textbox .
- Click on *EditCmn* button. *Common properties* page is displayed.

• *Change interface type* to *E1 ETSI V5.2 Local Exchange LE* .



• Click on *Apply* button.

• *Common properties* page will refresh and become a simple one.



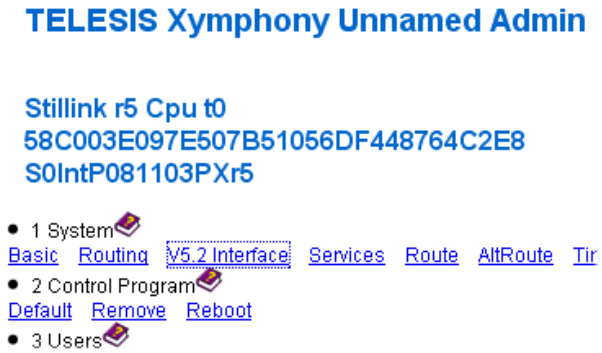
•Repeat the same for E1-05 (50500), E1-06 (50600), and E1-07 (50700).

Tips

For each V5.2 signaling E1, the *V5.2 Interface* name (shown in *Common Properties* page) is 0 by default. This name is only used during programming a Stillink system. This name is not used in any messages of the V5.2 protocol. In programming, the E1s sharing the same interface name are in the same V5.2 interface. In order to check, access to the *Common Properties* page for E1-04, E1-05, E1-06, and E1-07, see that *V5.2 Interface* name is set to 0 for each of them. Do not modify this number, keep it.

Step-2

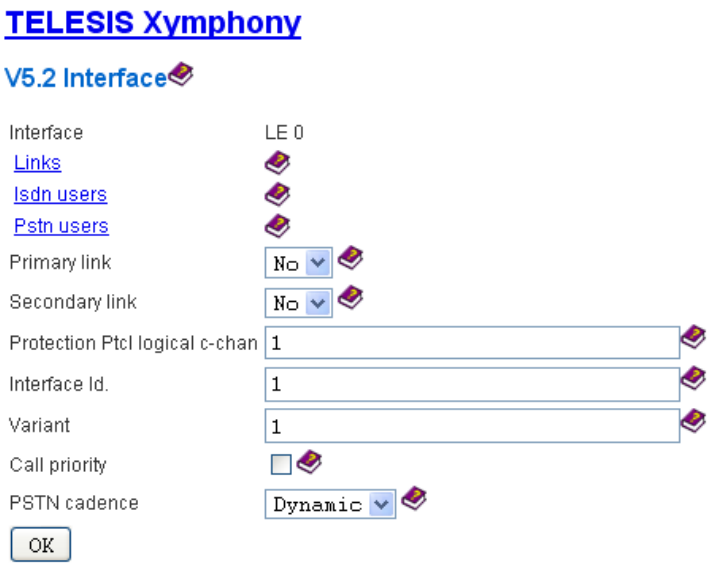
• In Homepage click on *V5.2 Interface* item .



• Select to edit page is displayed

• Select *LE 0* and then click on *OK* button.

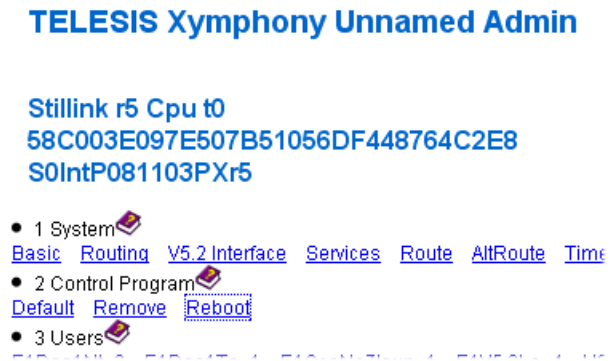
• *V5.2 interface* properties page for the LE 00 is displayed.



- In this page, set the interface parameters in order to match similar parameters in the AN side. The adjustable parameters are:
  - Primary link (E1-04, E1-05, E1-06, or E1-07)
  - Secondary link (E1-04, E1-05, E1-06, or E1-07)
  - Protection Ptcl logical c channel
  - Links (for Link IDs of E1-04, E1-05, E1-06, and E1-07
  - Page links for PSTN and ISDN users
  - Variant number
  - Logical address
  - Call priority
  - PSTN cadence
- Click on *OK* button after adjusting the parameters.

Step-3

- Store the programmed parameters to non-volatile memories with using *Save* command in Homepage
- Reset the system with using *Control Program - Reboot* command in Homepage



- After the restart, monitor the primary and secondary links to see the links are up

Adding V5.2 PSTN or BRI ISDN users

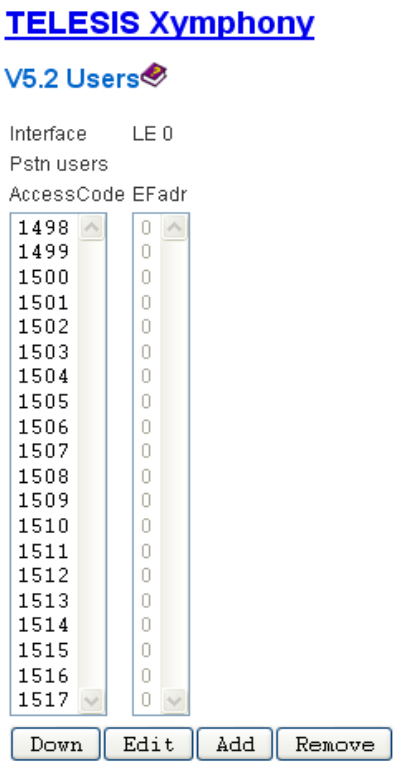
After setting the V5.2 interface up and seeing the communication on the primary and secondary links, PSTN and/or BRI ISDN users can be added for the interface.

Go with the above mentioned example. An SS7 - V5.2 protocol conversion application with using a Stillink 800 (ordering code s80.4s4v000). E1-00, E1-01, E1-02, and E1-03 are set to CCITT signaling type no:7 ISUP . E1-04, E1-05, E1-06, and E1-07 are bundled into the V5.2 LE 00 interface.

To add a PSTN user, do the following:

- Go to *V5.2 interface* properties page for the LE 00.
- Click on *Pstn Users* item.
- *V5.2 Users* page is displayed.
- Click on the *Add* button.
- *V5.2 users* page will refresh with a new user.

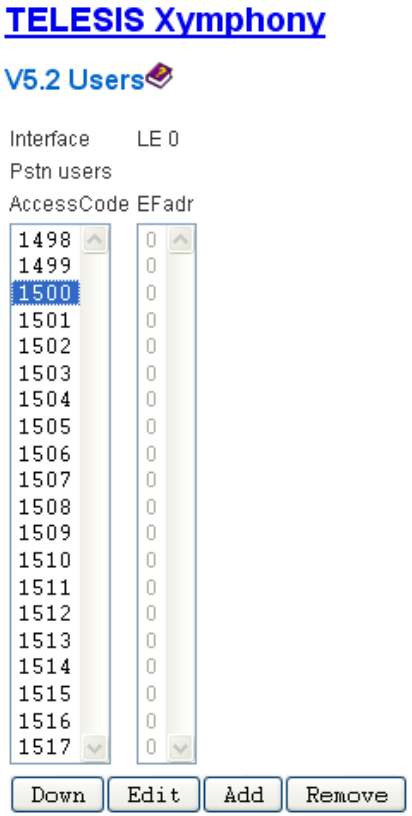
• Repeat this operation to add any number of PSTN users. The Stillink will assign access codes for users automatically



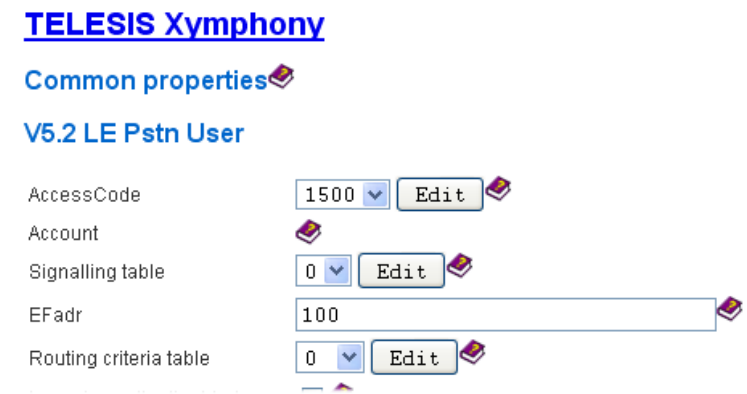
The result is several PSTN users in V5.2 Interface Users LE 00 Pstn page. In this page, for each PSTN user, there are two fields;

- access code field (like 1498, 1499, 1500, 1501....), and
- port ID or EF address field (by default, it is 0 for all access codes). EF address of the port as ETSI EN 300 347. This address will be used by both AN and LE side while referencing to this user. Valid values are 1-16383. If address is set to 0, provisioning of the user with the peer will not be performed (ie. user will not be available).

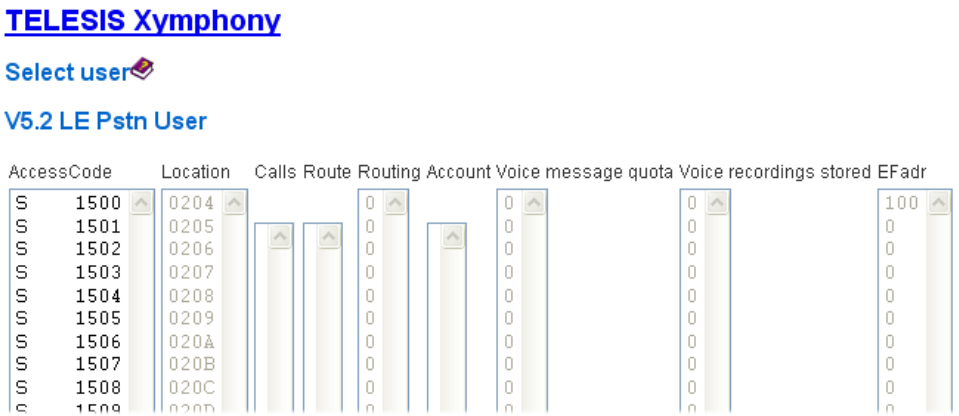
In order to change the port ID or EF address, select a user by marking it and then click on *Edit* button.



In *Common properties* page of the Pstn user, edit the *EFadr* field accordingly



• Click on *OK* button.



•Change the port ID or EF addresses of other users in their *Common properties* pages.

Next step is to activate the added PSTN users. This can be done by restarting the Stillink or the AN equipment, or by disconnecting and connecting again the physical E1 wire (connector) in front of the Stillink system for the primary and secondary links.

Setting the routing parameters

Consider the Stillink 800 programmed above. Assume that 100 PSTN users have been defined in this Stillink. The PSTN user access codes are from 1500 to 1599, and these access codes have the Port Ids (EF addresses) programmed according to the connected AN. Assume EF addresses from 100 to 199 in sequence. Let the requirement to be the following:

• All physical PSTN users in the AN side should be transparently routed to the SS7 network. The dialed numbers may be minimum 3 digit and maximum 20 digit long.

• From the SS7 network, called party addresses to be received are from 4448600 to 4448699. The called party address from 4448600 to 4448699 should ring on the AN subscribers with Port IDs 100 to 199 respectively.

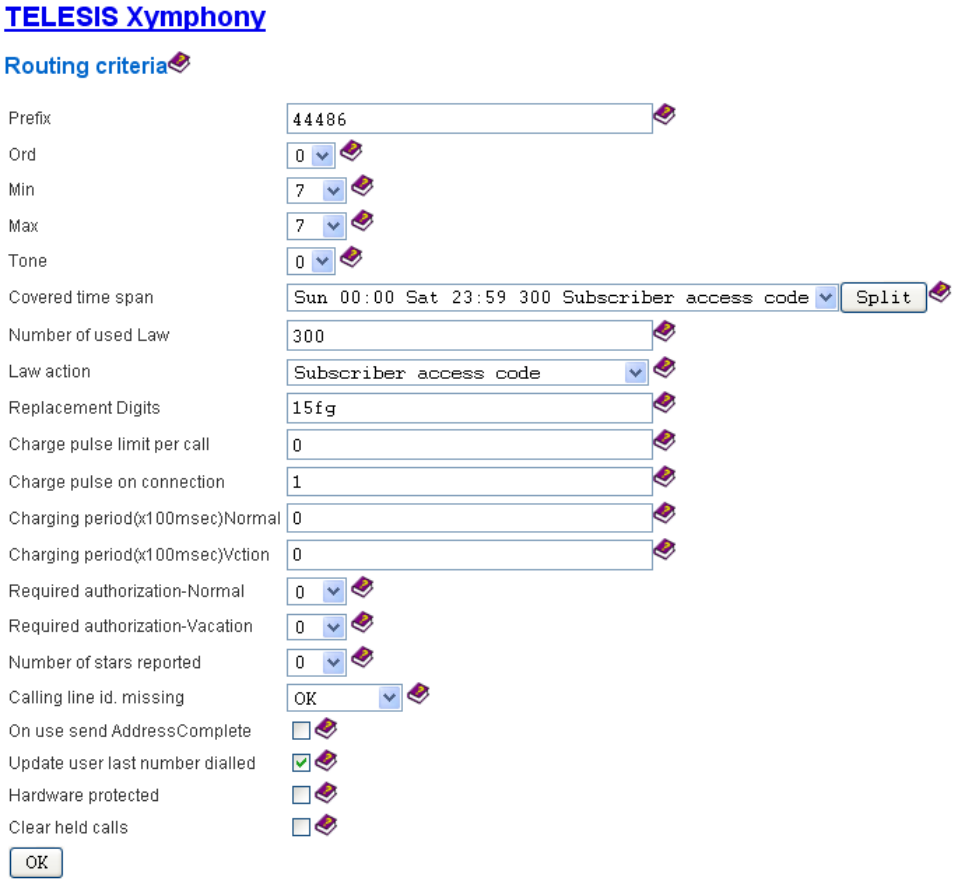
Do this programming as follows:

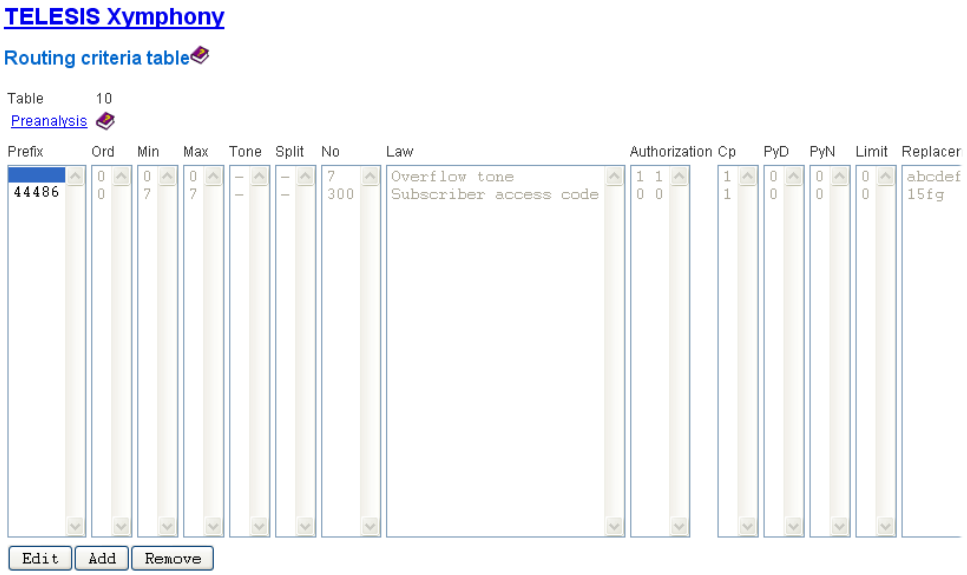
Step-1

Program all SS7 interfaces (E1-00, E1-01, E1-02, E1-03) such that

• Their routing tables are set to 10, and their route numbers are set to 10 for all of their voice channels (see General Topics section about how to achieve this).

• Program the *Routing criteria table* 10 such that *Prefix* field has 44486, *Min* field has 7, *Max* field has 7, *Replacement digits* field has 15fg , *Law action* field has Subscriber access code, *Required authorization* fields have 0.





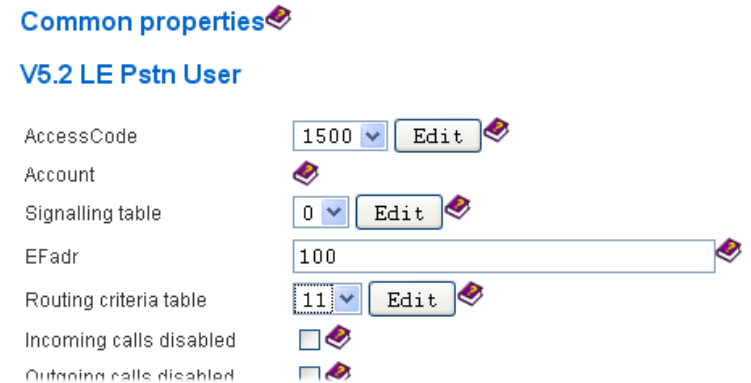
WHAT IS DONE for *Routing criteria table* 10? If any call with the called party address number like 44486XX is received from the SS7 network, this call will be routed to the Subscriber access code with some digit replacement such that:

- the first 5 digits will be deleted, that is 44486
- 15 will be added in the front
- the other digits will be kept
- consequently, 44486XX will be replaced to 15XX
- Note that 15XX refers to the EF address 1XX

Step-2

Program the routing table of the V5.2 PSTN users. Do the following:

- Open the *Common properties* page for the user 1500
- Assign a *Routing criteria table*. Assume that 11 is assigned.
- Click on the *OK* button.

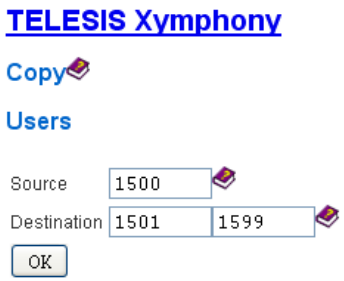


- Repeat the same, i.e., assignment of Routing table 11 for the other access codes 1501 to 1599.

Tips

After selecting the Routing table 11 for the access code 1500, this parameter can be copied onto the other access codes (1501 to 1599) with a simple command. This will make the programming easy and short. To do this select *Copy-Users* item in Homepage.

Copy page is displayed. In this page, select 1500 as the *Source* and 1501 to 1599 as the *Destination* and then click on *OK* button.

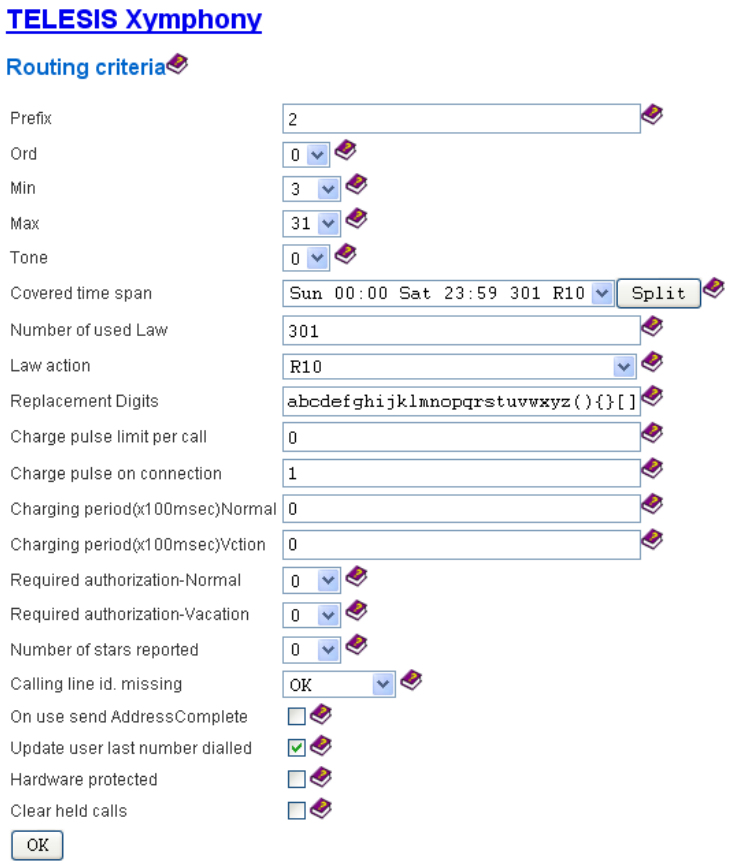


It is possible to adjust several parameters of the V5.2 users such as CLIP, Charge Pulse, and Credit. Some of these parameter adjustments can be done in *Common properties*, but most of these can be done in *User properties* pages. Both common and user properties can be accessed with using *Users* item in Homepage.

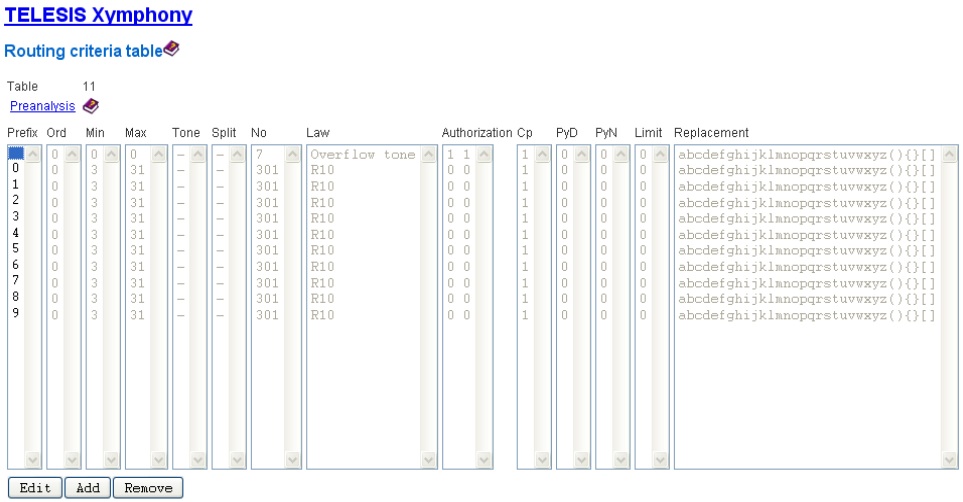
Next is to program the *Routing criteria table* 11 such that

- 10 rows
- *Prefix*es single digit from 0 to 9
- *Min* is 3 for all prefixes
- *Max* is 31 for all prefixes
- *Number of used Law* is 301
- *Law action* as R10 (route 10)

To illustrate, for the *Prefix* 2, the *Routing criteria* will look like below:



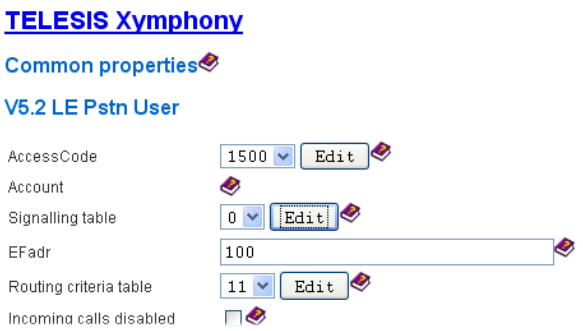
At the end of programming, the *Routing criteria table* 11 will look like below.



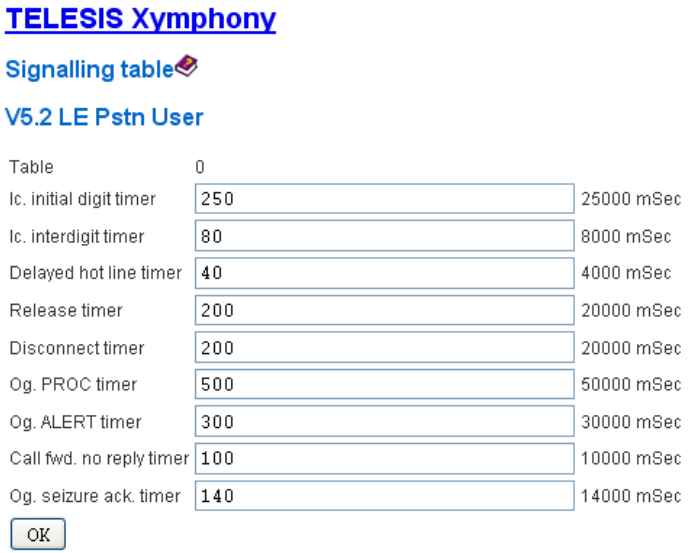
WHAT IS DONE for routing table 11? If any call with any called party address (minimum 3 digit and maximum 31 digit long) is received from a V5.2 user, this call will be routed to the Port Route 10 that is SS7 network. Remember that all SS7 signaling E1 have been in Port Route 10.

Tips

The Stillink will route the call of a V5.2 user immediately after the receipt of Max number of digits or the time-out occurrence. The default value of this timer is 8 seconds. This timer may be changed, however it is not recommended to make it less than 4 seconds. To change the timer, *Edit* the *Signaling table* of the V5.2 user. The *Signaling table* can be accessed in *Common properties* page.



- By default, Table 0 is assigned to all users, click on the *Edit* button.



- The relevant timer in this page is *Ic. Interdigit timer* By default, it is 80, which indicates 8 seconds.
- Edit the new value and click on the *OK* button..

Step-3

- Store the programmed parameter to non-volatile memories with using *Save* command in Homepage..

Tips

Before storing the programmed parameters, be sure that all the programming done is correct and according to the requirements. So that, it is recommended to create some test calls from SS7 network to the V5.2 LE, and vice versa.

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TECHNICAL SPECIFICATIONS:

GENERAL

Operational software	Xymphony Stillink
Maintenance and administrative software	Web Browser (IP based)
Operating voltage	Universal AC
CPU Type	High Speed DSPs
Switching Matrix	512 x 512
Ethernet interface	10/100 BaseT
Integrated CMDR buffer	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
Remote access	Yes
Signaling interworking	Yes

APPLICATIONS

E1 Digital VoIP Gateway	Yes
Combined VoIP Gateway and Converter	Yes
Pure E1 Signaling Converter	Yes
V5.2 Protocol Converter	Yes
V5.2-VoIP Access Gateway	Yes
E1 PRI Switch	Yes

INTERFACES

E1 interfaces (ITU-T G.703)	Two for the Stillink 200, Eight for the Stillink 800
SIP user agents	512
H.323 endpoints	512
Telesis xSIP users	single for maintenance and administration

TDM SIGNALING

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 (QSIG), ECMA-143 PISN	Yes
V5.2 LE protocol, ETSI EN 300 347	Version 2
V5.2 AN protocol, ETSI EN 300 347	Version 2
SS7 ISUP (CCS no.7),	
ETSI EN 300 356, ITU-T	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
Multifrequency shuttle signaling:	
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	10/100 BaseT
H.323 protocol, Version 5	Yes
SIP Session Initiation Protocol, RFC 3261	Yes
Telesis xSIP (eXtended SIP) protocol	Yes
G.711 audio codec	Yes
G.723.1 (5.3 and 6.4kbps) audio codec	Yes
G.729, G.729A audio codec	Yes
G.711 frame length	10 to 90msec
G.723.1 frame length	30 to 90msec
G.729, G.729A frame length	10 to 90msec
Silence Suppression (VAD)	Yes
Echo Canceler G.168-2002	Yes
QoS (Tos and Diffserv)	Yes
Integrated H.323 gatekeeper	Yes
Integrated SIP registrar	Yes
H.323 endpoints, which can register	512
SIP user agents, which can register	512
xSIP users, which can register	single
Programmable ports / sockets	Yes
MD5 authentication	Yes
H.235 Baseline Security Profile	Yes
H.235 Baseline Security Profile with integrity	Yes
Digest authentication	Yes
Audio (voice) encryption	AES-256
Softswitch capability	Yes
IP to TDM gateway capability	Yes
TDM to IP gateway capability	Yes
H.450 supplementary services	Yes
SIP supplementary services	Yes