SmartNode SN9000 Series

SS7 Gateways

(Digital VoIP Gateway, TDM Switch, Signaling Converter)

User Manual

This is a Class A device and is not intended for use in a residential environment.

REGULATORY MODEL NUMBER: SN9000RD4-001

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About this guide

This guide describes the SmartNode SN9000 Series SS7 Gateway hardware, installation and basic configuration.

This guide is intended for the following users:

- Operators
- Installers
- Maintenance technicians

Read this User Manual carefully before you start operating the product.

All connections must be made with the equipment fully powered off!

Do not operate the equipment without proper grounding!

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1. Conventions and Abbreviations

1.1. Designations

The document uses conventional symbols (icons) located on the left side of the page to highlight critical information. The list of the conventional symbols used in this document can be found below:

Indicates a warning that special attention should be paid to a particular section of the document.

Indicates a warning about critical information to which special attention should be paid.

Indicates a note or a piece of explanatory information.

Indicates example text from the system console, report or other source.

Indicates a tip that saves time and helps the user to work more efficiently.

Indicates a reference to an external document (e.g. specification or other resource) where more detailed information or description can be found.

Indicates a screenshot demonstrating a respective part of a text.

1.2. Abbreviations

Table 1. Abbreviations

Abreviation	Description
SIP	The Session Initiation Protocol (SIP) is a communications protocol for signaling, for the purpose of controlling multimedia communication sessions. Internet telephony, business IP telephone systems, service providers and all of the carriers use SIP.
VoIP	Voice over IP is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.
TDM	Time-division multiplexing (TDM) is a method of transmitting and receiving independent signals over a common signal path by means of synchronized switches at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern. It is used when the data rate of the transmission medium exceeds that of signal to be transmitted.
E1	Standard of digital transmission of data
D-Channel	D channel (delta channel) is a telecommunications term which refers to the ISDN channel in which the control and signaling information is carried.
PSTN	The Public Switched Telephone Network (PSTN) is the aggregate of the world's circuit-switched telephone networks that are operated by national, regional, or local telephone operators, providing infrastructure and services for public telecommunication. The PSTN consists of telephone lines, fiber optic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables, all interconnected by switching centers, thus allowing most telephones to communicate with each other.
TE	Telephone Exchange
SNMP	Simple Network Management Protocol
CDR	Call Detail Record
DLU	Digital Line Unit
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Service
DTMF	Dual-Tone Multi Frequency (tones)
HTTP	Hypertext Transfer Protocol. Refer to IETF RFC 1945 and RFC 2068
IP	Internet Protocol. An Internet network-layer protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part. A protocol within the SS7 suite of protocols that is used for call signaling within an SS7 network
IVR	Interactive Voice Response system

Abreviation	Description
MG	Media Gateway. Provides the bearer circuit interfaces to the PSTN and transcodes the media stream.
MTP	The Message Transfer Part. A set of two protocols (MTP 2, MTP 3) within the SS7 suite of protocols that are used to implement physical, data link, and network-level transport facilities within an SS7 network.
RADIUS	Remote Authentication Dial-In User Service. An internet protocol (IETF RFC 2865 and RFC 2866) originally designed for allowing users dial-in access to the internet through remote servers. Its flexible design has allowed it to be extended well beyond its original intended use.
RTP	Real-time Transport Protocol. A protocol for encapsulating encoded voice and video streams. Refer to IETF RFC 1889.
RTCP	Real-Time Control Protocol
SS7	Signaling System number 7. An architecture and set of protocols for performing out-of-band call signaling with a telephone network.
STP	Signal Transfer Point. A node within an SS7 network that routes signaling messages based on their destination address. This is essentially a packet switch for SS7. It may also perform additional routing services such as Global Title Translation.
TCP	Transmission Control Protocol
UDP	User Datagram Protocol. A connectionless protocol built upon Internet Protocol (IP).

2. System Description

This manual will make you familiar with the basics of operating the SIP/E1 Gateway configurator software included with the unit. The contents of the document cover all items required to start operating the product immediately.

Read the manual in a cursory manner before you start operating the product and then refer to required sections for help using the Table of Contents.

2.1. Product Specification Summary

- 16x E1 ports, 2x Ethernet ports, 1x RS-232 port
- Signaling and control protocols:
  - IP - SIP, RTP, RTCP, TCP, UDP
  - PSTN - SS7 (MTP2, MTP3, ISUP), ISDN PRI (DSS1), V5.2 (LE/AN mode), R2 MFC, R1,5.

- The following services are provided with 480 channels occupied at a time:
  - G.711, GSM, G.723, G.726, G.729 compression
  - Fax - fax over G.711, T.38 fax relay
  - DTMF - RFC2833, SIP INFO, RFC 2976
  - Comfort Noise Generation (CNG)
- Power Supply
  - Voltage: DC 48/60V (+/- 20%) and/or AC 220V (voltage range AC 100V - 240V)
  - Power consumption: max. 28W
- Design Specifications
  - EUROPACK 19", 1U form factor
  - Weight - 2.5 kg
- Call routing over all directions
- Signaling and SIP settings

3. System in Brief

The Patton SmartNode SN9000 Series Gateway is a flexible digital telecom switch for inter-networking TDM and SIP networks. The unit supports T1/E1 multiplexing, signaling conversion, digital switching, and media gateway functions.

As an SIP-to-SS7 and SS7-to-SIP Media Gateway, the unit delivers the ability to integrate TDM equipment, trunks and inter-connects with SS7 into IP, IMS, and NGN networks. Similarly, the unit can act as a V5.2-to-SIP and DSS1-to-SIP VoIP gateway.

As a Signaling Converter the unit can be used to adapt and interwork between SS7, ISDN-PRI (QSIG), DSS1, V5.2, R2/CAS and Ethernet transferring media across PSTN and/or IP networks. Integrated TDM Multiplexing and Switching provides the ability to cross connect and groom E1 channels between several E1 links.

The unit is equipped with 4, 8 or 16 T1/E1 ports, 2x Ethernet ports, and 1x RS-232 console port. It offers a full suite of routing, switching, signaling, and control protocols including IP SIP, RTP, RTCP, TCP, and UDP. On the TDM side SS7 (MTP2, MTP3, ISUP), ISDN PRI (DSS1), V5.2 (LE/AN mode), and MFC-R2.

The unit offers G.711, GSM, G.723, G.726, G.729 compression codecs, fax-over-IP using G.711 or T.38 fax relay. DTMF (via RFC2833) and SIP INFO (via RFC 2976) are supported as well as Comfort Noise Generation.

The unit is rack mountable in 1U of space and comes with either dual DC (48/60V) and/or AC (100 to 240 V) internal power supplies.

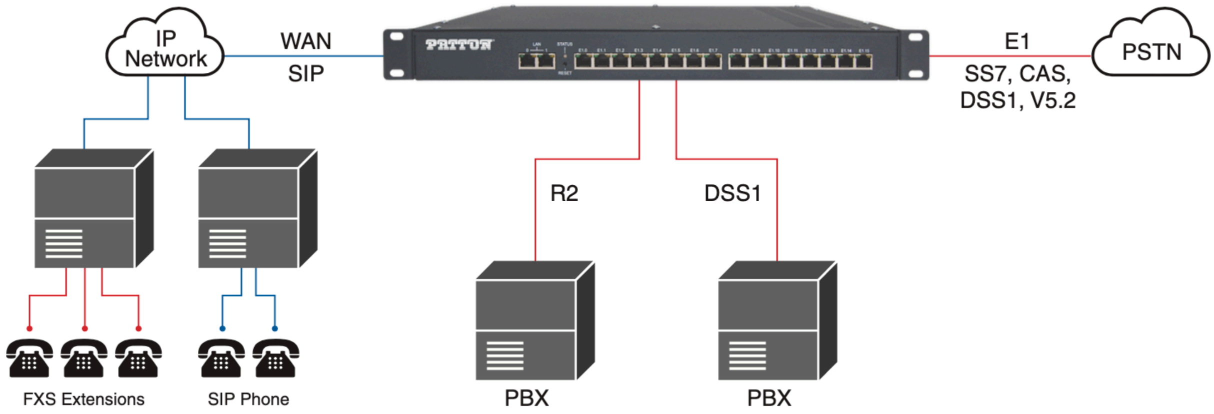
Applications

The SN9000 is a flexible device used for adaptation of VoIP and TDM telephony networks and equipment. As a Media Gateway, the devices are used to adapt legacy TDM trunks to SIP/VoIP. The unit supports all signaling types used with the PSTN network and is useful for converting signaling protocols or multiplexing TDM channels.

The following figures illustrate typical device applications.

Figure 1. Device Applications - A typical use case of the device in a VoIP network

SIP/E1 Gateway



Signaling Converter

Media Gateway

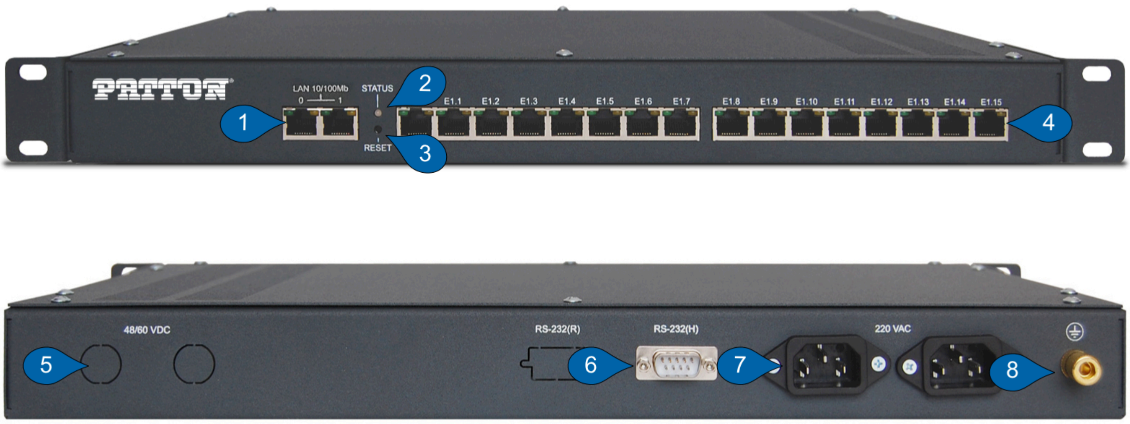
Table 2. Specifications

VoIP	SIP 2.0 (RFC 3261)
Network Protocols	IP, TCP, UDP, FTP, TFTP, RTP, RTCP, ARP, ICMP, NTP, Telnet, IEEE 802.1Q, IEEE 802.1P
Signaling Protocols	SS7 (ITU-T Q.700 series), 24bit/14bit PC, ISUP/TUP, ISDN-PRI (ITU-T Q.931,Q.921), CAS R2 Q.400-Q.490, CAS DTMF BellCore TR-TSV-002275, V5.2 (ETS 300 347-1)
Sound	G.711u-Law and A-Law, G.711 Appendix 1, G.723.1 and G.723.1 Annex A, G.729 Annex A and Annex B, G.726, GSM, ARM, ILBC, DTMF, adaptive jitter buffer, VAD, CNG, G.165, G.168, silence suppression, tone scheme in accordance with standard, ITU v.152
Network	WAN interface, 10 Mbit/100 Mbit, static IP address
QoS	802.1Q, 802.1P
Interfaces	E1 G.703; Ethernet 10/100/1000; RS-232
E1 port impedance	Support for the connection of balanced and unbalanced cables with impedances 75 / 120Ω. If the E1 interface is connected with an unbalanced cable, it is necessary to use the RJ-45 - BNC adapter
Security	Filtering by IP addresses, 802.1Q
Applications	Out-band Proxy, Reregister, Voice/Modem/Fax Calls, Prefix Number Routing, Dial Plan, Number Modification, 1+1 SS Protection
Control	GUI client, Telnet, SNMP v1/v2c Traps, updating software over FTP/TFTP, storing/recovering configuration, checking E1 ports/signaling status
Power Supply	Single/Dual-redundant 220 VAC or 48/60 VDC Input
Power Consumption	28W max
Environment	Operation temperature: 0°C to +60°C, humidity: 0% to 90% (non-condensing)
Dimensions	485 mm x 286 mm x 44 mm

3.1. General view

Structurally, the SIP/E1 Gateway is designed for horizontal installation in a 19 inch shelf. The device is 1U in height. The device's front and back panels with connectors are shown below (Figure 2).

Figure 2. SIP/E1 Gateway – Front and back panels



Descriptions of connectors can be found in table below.

Table 3. Front and back panel - Component Description

Tag number	Element	Description
1	LAN 0 / 1	RJ-45 - network interface connector (Ethernet). Dual redundant 10/100 Base-T ports. To connect to the equipment and the initial settings you want to use the connector: **LAN 10/100 Mb – «1»**
2	STATUS	Indicator of electricity
3	RESET	Device reset button (short press) / Reset settings to default (press for more than 5 seconds)
4	E1 0-7 (0-15)	The E1 connector ports 0-7 (0-15)
5	48/60 VDC	Electricity switching connector (48/60V DC)
6	RS-232 (R/H)	Serial port is a serial communication physical interface
7	220VAC	Electricity switching connector (220V AC)
8	GND	Grounding device

Table 4. STATUS - LED Description

	Indicator Color	Color	Operating Status	
			Booting	Operating
	Blinking red		Booting U-boot bootloader	Timeout
	Red		Booting device OS *	-
	Blinking yellow		Booting the main device application *	Failure of one of E1 ports or failure of D-channel
	Green		-	Normal operation
	* Yellow + blinking red		The combined mode of booting OS and application	-

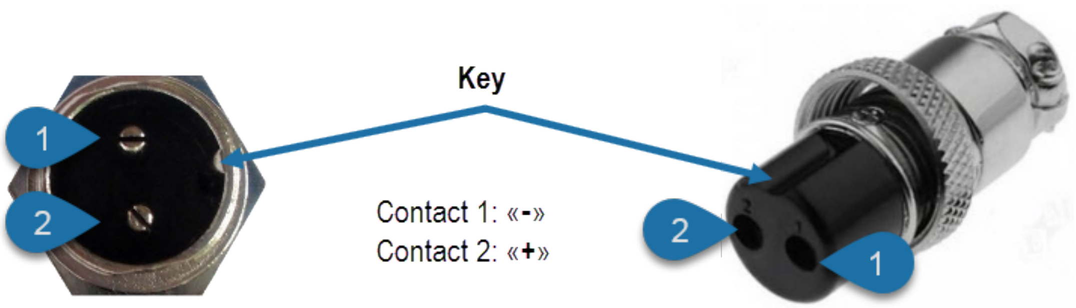
3.2. Connecting the power supply of the device

Depending on the model, the equipment supports various power supply options:

- 220V AC
- 48/60V DC
- 220V AC + 48/60V DC
- 220V AC + 220V AC
- 48/60V DC + 48/60V DC

Table 5. Power supply options

AC Power	• Single universal power supply 100-240V 1.5A max, 50-60 Hz • Redundant power supply (optional)
DC Power	• 36 – 72VDC (48VDC nominally), 4A max (optional) • Redundant power supply (optional)
Power Consumption	• 1-4 E1: 16W • 8E1: 18W • 16E1: 30W



The color of the wire for the "-" contact, in the DC power supply cord supplied with the equipment, is always "blue".

The color of the wire for the "+" contact, depending on the delivery, can be different.

Incorrect connection of the polarity of the input voltage for direct current does not lead to failure of the equipment. In this case, the equipment does not turn on.

3.3. Connecting to Ground

The device must be permanently connected to ground (earth), using an equipment - grounding conductor.

To ground the device:

1. Connect an electrically grounded strap of 1.65 mm wire (minimum) to the chassis grounding screw (located on the rear panel).

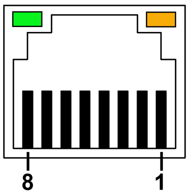
2. Connect the other end of the strap to a protective grounding. This should be in accordance with the regulations enforced in the country of installation.

Attention!

For safe use of the equipment it is necessary to carry out grounding of the case. The image below shows the device with a connected grounding wire.

Figure 3. SIP/E1 Gateway – Grounding device

3.4. Ethernet Control Connector



Type: RJ-45. An RJ-45 is an 8-pin connection used for Ethernet network adapters.

The table below describes the status LED indicators.

Table 6. Ethernet (RJ-45) Connector Pinouts - 10/100Mb

Contact number (10/100 Mb)	Signal	Description
1	TX+	Ethernet interface. Transmit Data+
2	TX-	Ethernet interface. Transmit Data-
3	RX+	Ethernet interface. Receive Data+
4	NC	Not connected
5	NC	Not connected
6	RX-	Ethernet interface. Receive Data-
7	NC	Not connected
8	NC	Not connected

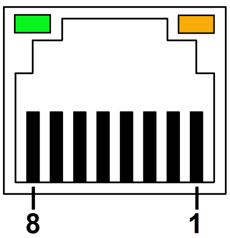
Table 7. Ethernet (RJ-45) Connector Pinouts - 1000Mb

Contact number (1000 Mb)	Signal	Description
1	BI_DA+	Bi-directional pair A, +
2	BI_DA-	Bi-directional pair A, -
3	BI_DB+	Bi-directional pair B, +
4	BI_DC+	Bi-directional pair C, +
5	BI_DC-	Bi-directional pair C, -
6	BI_DB-	Bi-directional pair B, -
7	BI_DD+	Bi-directional pair D, +
8	BI_DD-	Bi-directional pair D, -

Table 8. Ethernet (RJ-45) Connector – LED Description

LED Color	Color	LED State	Description
Green (Left)		Off	No connection
		On	Link Up
		Blinking	Link activity
Amber (Right)		On	1000 Mb
		Off	10/100 Mb

3.5. Connecting E1 Digital Ports 0-15



Type: RJ-45. An RJ-45 is an 8-pin connector used for connecting the E1 digital port.

The table below describes the status LED indicators.

Table 9. E1 (RJ-45) Connector Pinouts

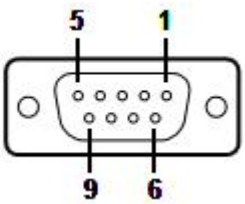
Contact number	Description
1	Output PCM
2	Output PCM
3	Not connected
4	Input PCM
5	Input PCM
6	Not connected
7	Not connected
8	Not connected

Table 10. E1 Connector – LED Description

LED Color	Color	LED State	Description
Green (Left)		Off	Port E1 is out of service
Amber (Right)		Off	
Green (Left)		Off	Port E1 configured but physical link is down
Amber (Right)		On	
Green (Left)		On	Physical link is up. Signaling link is ready
Amber (Right)		Off	
Green (Left)		On	Physical link is up. Signaling link is not ready
Amber (Right)		On	

If the E1 interface is connected with an unbalanced cable, it is necessary to use the RJ-45 - BNC adapter.

3.6. Interface Connector RS-232 (Serial port)



Type: DB-9. Recommended Standard-232. A RS-232 is a 9-pin connector used to connect to the console port of the device.

Table 11. Serial Port (RS-232) Connector Pinouts

Contact number	Signal	Description
1	NC	Not connected
2	TXD	Transmitted Data
3	RXD	Received Data
4	NC	Not connected
5	GND	Signal Ground
6	NC	Not connected
7	NC	Not connected
8	NC	Not connected
9	NC	Not connected

3.6.1. Connecting to the device via the Serial port

Connection via the serial port is used as a spare connection used to diagnose and restore operability of the equipment.

Connect your PC to the unit by connecting the communication serial port of the PC to the serial port of the equipment using the console cable.

The settings of the serial port of the device are as follows:

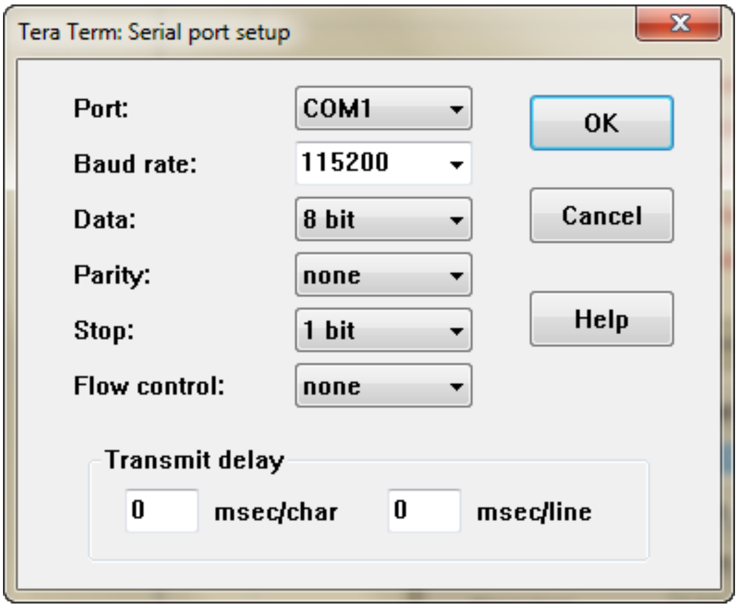
- Serial port rate (Baud Rate): 115200
- Quantity of data bits (Data Bits): 8
- Parity (Parity Bits): None
- Quantity of stop bits (Stop Bit): 1
- Flow control: None

The PC that will be used for connecting to the device should be equipped with an RS-232 interface. You can use an integrated serial port (if it is available) or an RS-232 USB interface converter that can be connected.

You can use any terminal software to connect to the device. For example, you can use the Hyper Terminal integrated into Windows OS, or Tera Term, or PuTTY applications.

Prior to using the terminal software, the application should be configured (see Figure 4 below) based on the initial settings specified above.

Figure 4. Configuring the serial port - Tera Term



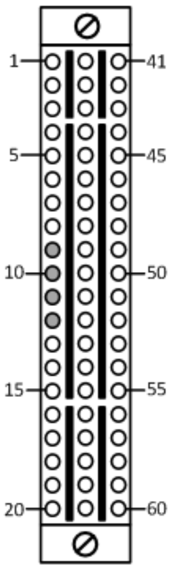
3.7. Connecting to block DLU

The VoIP Gateway connects to EWSD DLUs using the E1 interface.

In the VoIP Gateway, the E1 interfaces are output through the RJ-45 connectors on the front panel with the corresponding pinout (Table 9).

The E1 interface of the DLU is output via the corresponding interface module. For example, in the DLU-B version, the connection is made to the DIUD module connector (Table 12).

The upper connector connects the E1 link with signal and conversational channels, and the lower connector-E1 link contains only the conversational channels.



DIUD-30 connector view on the connection side.

Table 12. DLU block DIUD-30 Connector Pinouts

Contact number	Description
1-8	Not connected
9-10	Output PCM (or PDC primary digital carrier)
11-12	Input PCM (or PDC primary digital carrier)
13-60	Not connected

To connect to other types of governors/controllers, a connection point interface E1 DLU block must be defined according to the manufacturer's documentation.

3.8. General Implementation Guidelines

The device features the following functionalities:

- full access switching of any digital channel with any digital channel (load of 1 Erlang)
- converting signaling protocols of each of 480 digital TDM channels to SIP, RTP, RTCP G.711, GSM, G.729, fax - fax over G.711, T.38 fax relay, DTMF - RFC2833, SIP INFO, RFC 2976
- analyzing translated (transmitted) digits with automatic selection (generation) of outgoing communications directions
- modifying the digits of a number for both the call source and call destination as needed
- routing in digital trunks

The equipment supports the following signaling protocols:

- SS7 (Signaling System 7)
- ISDN PRI DSS1 (telephony signaling protocol in ISDN digital network)
- V5.2 interface (used to connect the access networks (AN) to local exchanges (LE))
- R2 MFC digital
- R2 DTMF
- SIP (VoIP protocol)

Some equipment releases can support the following protocols:

- R 1.5

3.8.1. Signaling Converter and Protocol

The signaling and protocol converter is an optimum and efficient tool used to organize interaction between different types of telecommunication equipment that uses different signaling types.

The signaling converter provides:

- Transparent conversion of signaling protocols of E1 ports between local exchanges,
- Creation of semipermanent connections, i.e. it is possible to forward any timeslot of a single E1 port to any timeslot of the different E1 port, and
- Converting the subscriber number when the connection is being established over the assigned masks.

The equipment is set up and configured through the web interface. The semipermanent connections are created using the separate configuration file that is configured using Telnet. The configuration file is saved on the device and it is used any time when the equipment is switched on. The file is saved on the device until it is deleted or replaced with another file.

The example configuration of the signaling converter and protocol converter is described in the Configuration Examples section of this manual.

3.8.2. TDM Digital Switch

The TDM digital switch provides 3 types of TDM channel switching:

- Semipermanent switching
- Dynamic switching
- Combined switching

Semipermanent Switching

Semipermanent switching corresponds to the configuration loaded in the switch and this type of switching is performed directly after the power is turned on. Since the switching scheme is a static scheme this type of switching is not accompanied by converting and analyzing switched data.

Its main purpose is to create semipermanent connections, i.e. to forward any timeslot of a single E1 port to any timeslot of the different E1 port.

A file is created with a text editor where Telnet commands used for switching a channel to another channel are recorded. This file is saved on the device and it is run any time when the device is switched on. The file is saved on the device until it is deleted or replaced with another file.

Dynamic Switching

Dynamic switching means connection of channels that meet the conditions determined by the configuration. Such conditions may include:

- Unconditional switching in accordance with the channel numbers
- Conditional switching in accordance with the number of the called subscriber
- System switch time (channel operating schedule)

Combined Switching

Combined switching allows using the combination of semipermanent and dynamic connections.

The example configuration of the E1 port switch is described in the Configuration Examples section of this manual.

3.8.3. Digital VoIP Gateway

The VoIP gateway is a dedicated industrial controller that provides SIP connections and access to PSTN (Public Switched Telephone Network).

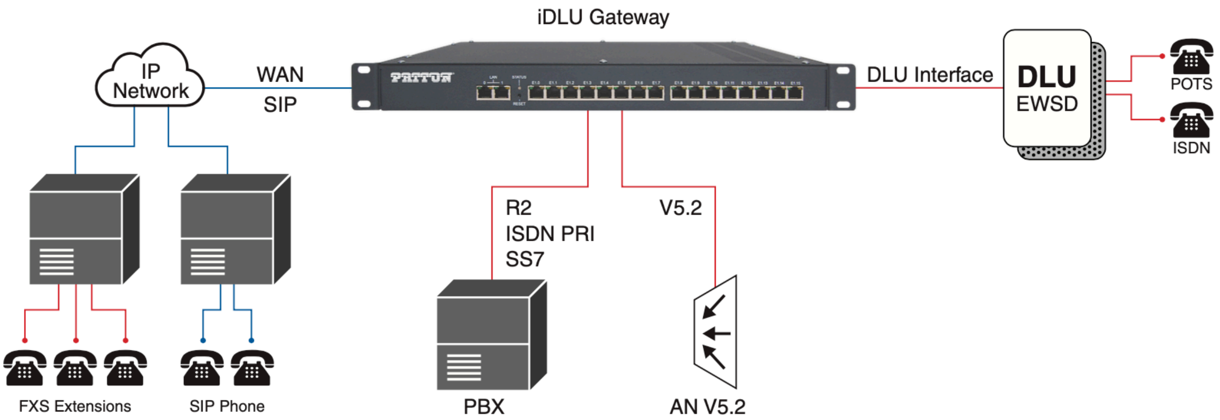
The VoIP gateway provides integration of TDM and IP networks by converting signaling protocols of each of the 480 digital E1 channels to SIP supporting RTP, RTCP, GSM, G.711, G.723, G.726, G.729, fax over G.711, T.38 fax relay, DTMF – RFC2833, SIP INFO, RFC 2976.

The example configuration of the VoIP gateway is described in the Configuration Examples section of this manual.

3.8.4. iDLU VoIP gateway for integration of digital line units DLU EWSD system in IP networks

The iDLU VoIP Gateway is designed to integrate all types of EWSD DLUs with IP/NGN/IMS packet switched networks.

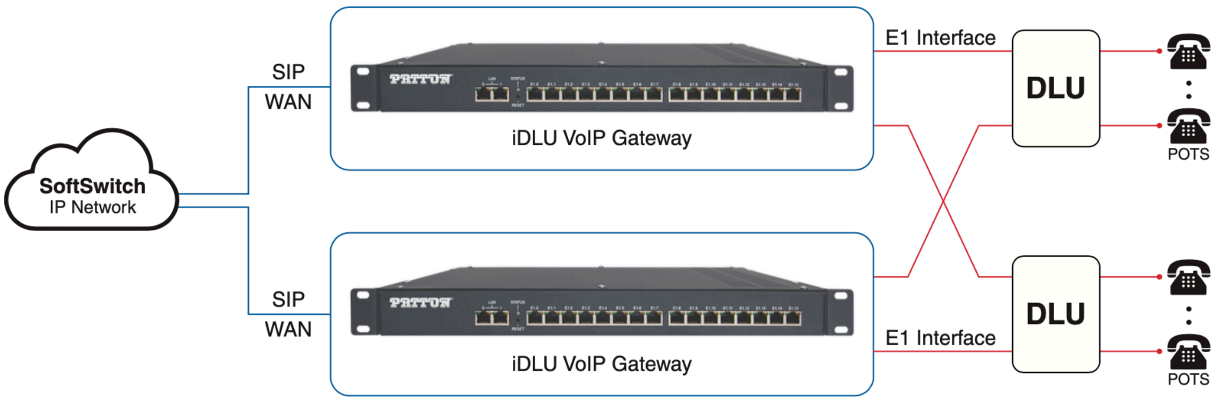
Figure 5. A typical DLU integration pattern in an IP network



This solution provides for upgrading EWSD systems to DLUs that connect to a iDLU VoIP Gateway access media gateway over an internal DLU protocol and to an IP network over SIP protocol.

Redundancy support.

Figure 6. A typical scheme for providing system redundancy



In order to protect against failures, each DLU is connected to two iDLU VoIP gateways that work simultaneously like two LTGs.

There is an option of converting DLUs to AN subscriber remote node supporting V5.2 or SS7 signaling.

The objective of this solution is to switch subscribers from a TDM segment to SoftSwitch control over SIP, and RTP/RTCP access protocols.

All subscribers of DLUs will obtain the full package of services from NGN/IMS service servers.

3.9. Software Functions

The product software supports connection over HTTP (web interface) that can be used to:

- configure switching directions
- set up switching parameters
- create a backup and restore from a backup for configurations of the product and main software

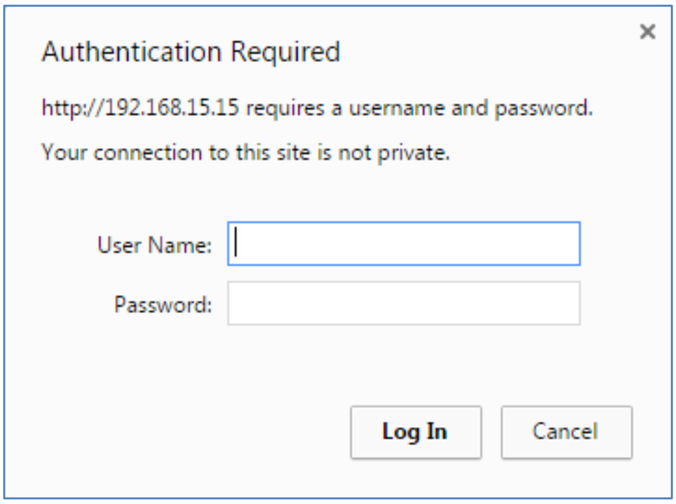
To operate the web interface you will need any web browser.

4. Operating Basics

To connect to the web interface, type the IP address of the product in the address line of your web browser and press Enter.

The browser will display the user authorization dialog.

Figure 7. User authorization form



It contains user name and password input fields.

Table 13. Default data user authorization

User Name	Password
admin	admin
user	user

Table 14. Network equipment settings for port 1

Parameter	Value
IP address	192.168.5.5
Network mask	255.255.0.0

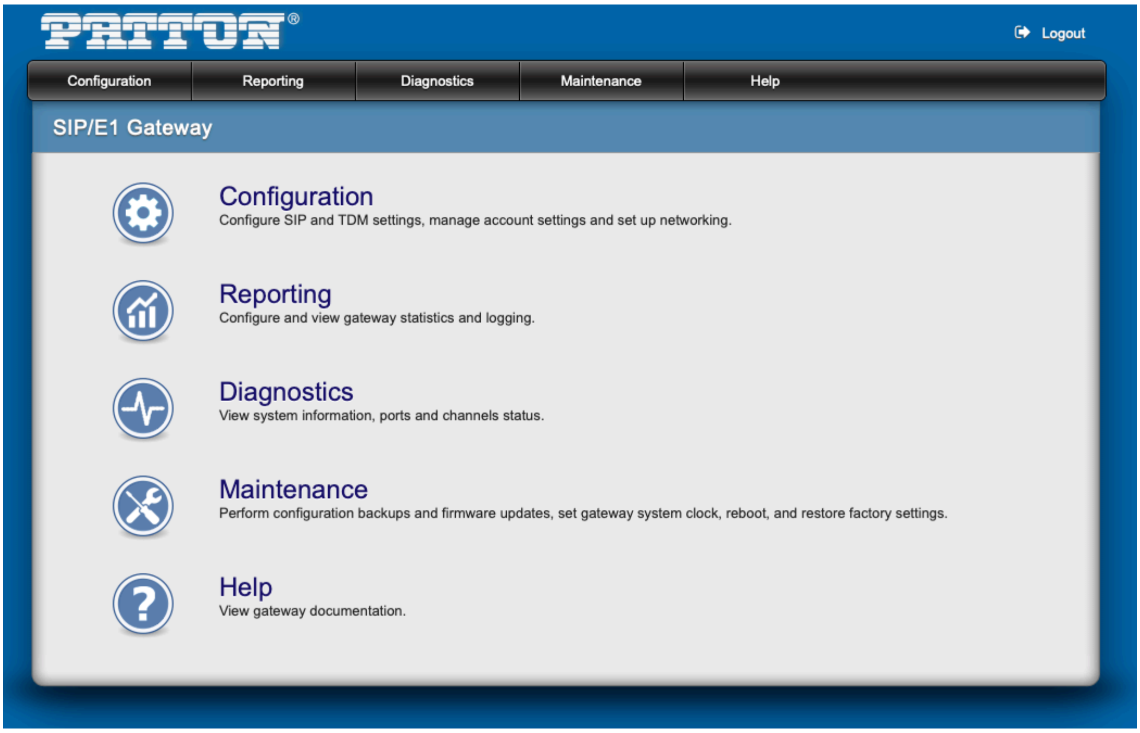
The above IP address is only for port 1.

The web interface of the product supports 2 types of user accounts - Administrator and User. See below for a description of each type of the account as well as how to change user names and passwords.

Warning! To prevent unauthorized access to the device, it's recommended that you change the default login user name and password after you initially access the Web interface (refer to 'Users').

If authorization is successful the browser will display the main page of the product control interface (Figure 8).

Figure 8. Web Interface - Device Management



Use the Logout link to exit the web interface.

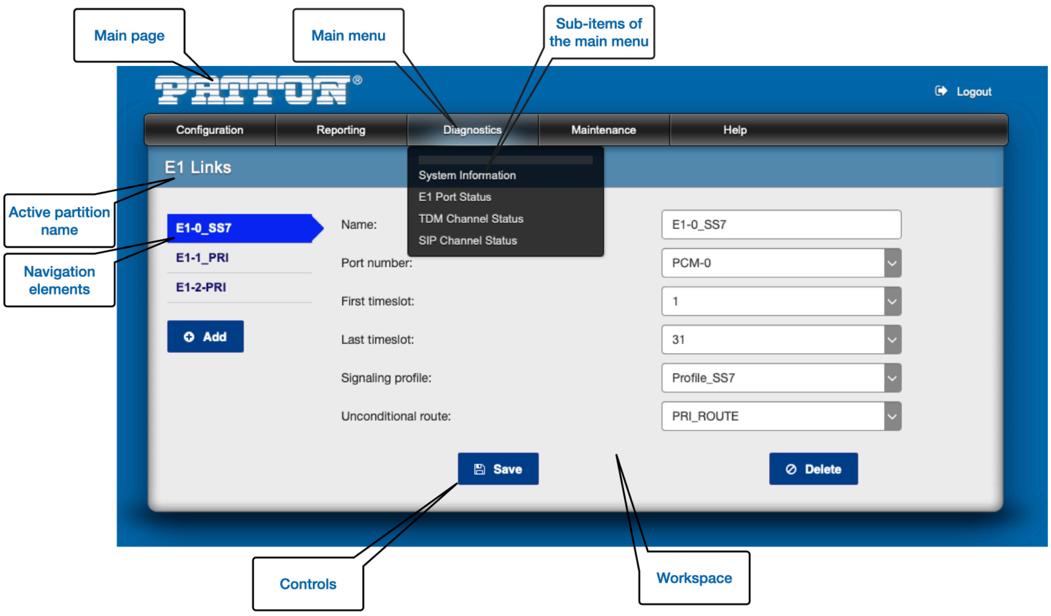
Warning! It is recommended that you back up the product configuration files ("Configuration Files") and product firmware file ("Device Firmware Files") before you start changing any product parameters. See the Maintenance – Backup Files subsection on how to create backups.

Warning! All software subsections contain the Save button. The Save button is used to save applied changes in the configuration file of the equipment. To start operating the product with the changes applied switch to Maintenance - Reboot Device subsection and click Reboot. This will reload the product with the saved parameter changes.

4.1. User interface areas

The figure below shows the basic layout of the Web user interface.

Figure 9. Main areas of the Web user interface



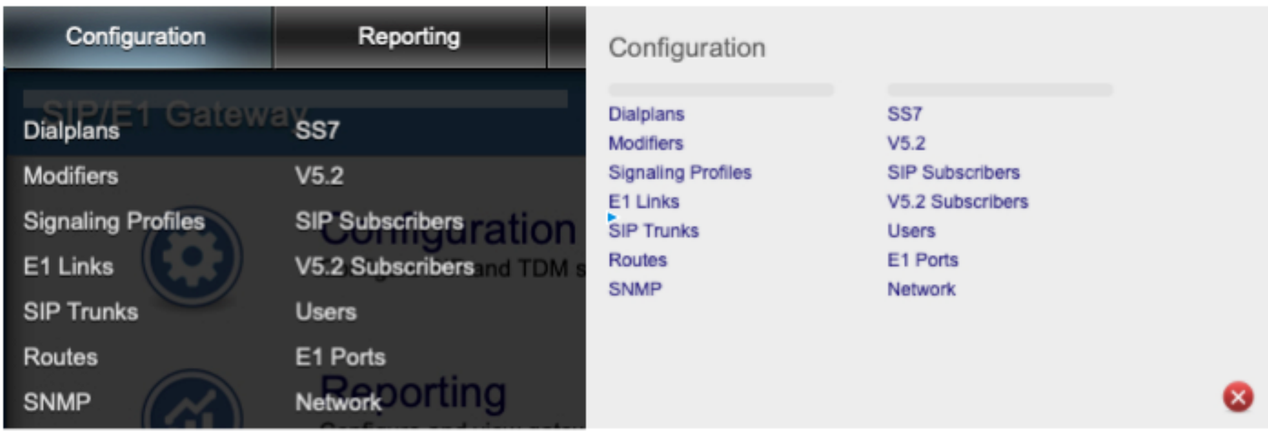
All sections of the Web-based user interface contain control buttons. The list of controls and a brief description are given in the table below:

Table 15. Buttons are controls.

Control element	Description
Add	Add a new configuration.
Save	Saves the changes made to the current configuration.
Delete	Deletes the active (current) configuration.
Apply	Saves values for parameters that do not belong to any of the configuration subkeys.
Download	Downloads a hardware configuration file(s) to the PC.
Select file	Select the configuration files to download.
Upload	Upload the selected configuration file to the SN9000.
Sync	Synchronization of data.
Reboot	Restarts the equipment.

The product control web interface contains the main menu with menu items that appear when you hover over a menu item in the main menu bar. The main screen duplicates the items of the main menu (Figure 10).

Figure 10. Web Interface - Menu Items



The main menu of the web interface includes the following items:

- Configuration
- Reporting
- Diagnostics
- Maintenance
- Help

4.2. Configuration

The Configuration menu items provide access to the control web interface forms that can be used to configure the SIP/E1 Gateway and save data in the product configuration file. This menu contains the following subitems:

- Dialplans
- Modifiers
- Signaling Profiles
- E1 Links
- SIP Trunks
- Routes
- SNMP
- SS7
- V5.2
- DLU
- SIP Subscriber
- V5.2 Subscribers
- DLU Subscribers
- Users
- E1 Port
- Network

Please Note! The number of menu items depends on the specific model of the product.

4.2.1. Dialplans

The Dialplan is a list of number templates that can be used for the following purposes:

- select any direction to establish a connection with the called subscriber (based on the number of the calling or called subscriber)
- identify the indicator that indicates when dialing is complete
- identify when it is required to modify the subscriber's number

The Dialplans subsection is used to control dialplans (number templates).

Figure 11. Web Interface - Dialplans

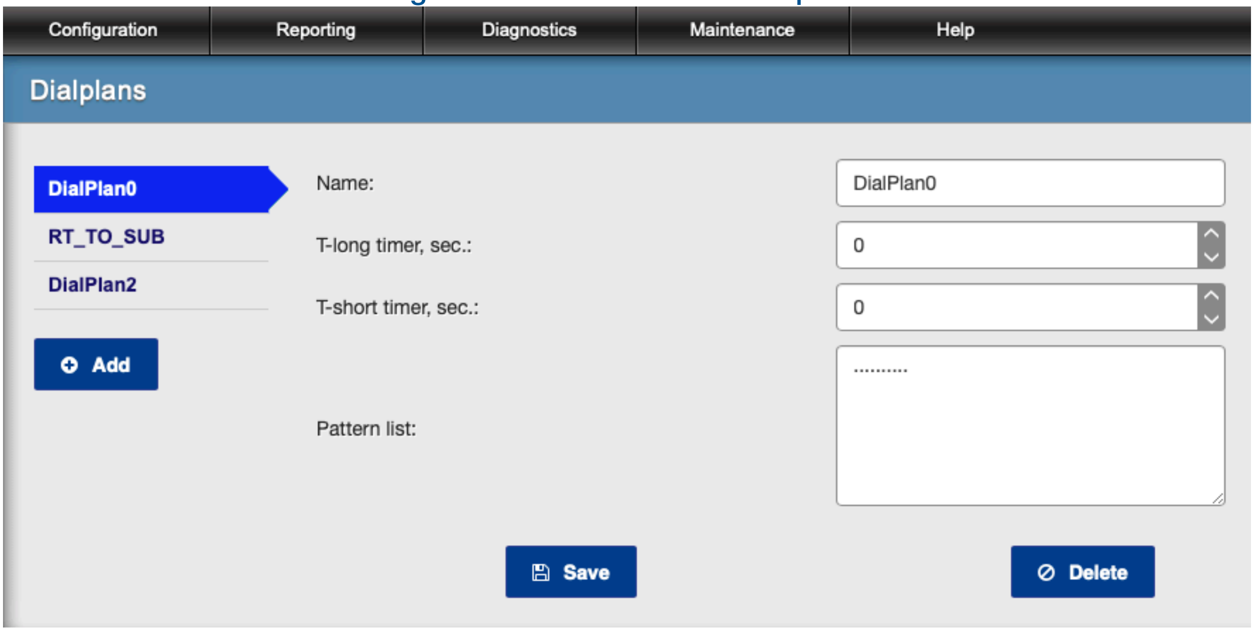


Table 16. Settings for number templates

Parameter	Description
Name	Dialplan name (name of the template set).
T-long timer	Maximum time (in seconds) that is allowed between dialed digits when no template matches the dialed number to the full extent.
T-shot timer	Maximum time (in seconds) that is allowed between dialed digits when at least one template matches the dialed number.

Parameter	Description
Pattern list	List of templates for the current dial plan. Each new template for the current dial plan is assigned in a new line. The following symbols can be used to create the template: **1234567890*#|[] .,xSL-** 1) Digits from 1 to 9. 2) The asterisk (*) and hash (#) characters are used to control additional services. 3) The sequence of digits within square brackets [ ]. It means that any digit displayed within the square brackets can be used. For example the [13579] sequence means that digits 1, 3, 5, 7 and 9 can be used. 4) x. It means any digit. For example the xxxx sequence means that any 4-digit number can be used. 5) Intervals. For example the 1-9 sequence means that any digit from 1 to 9 can be used. The interval of values is used within square brackets, i.e. [1-9]. 6) The comma (,) character. It is used to apply the dial tone in the line. For example the 2,3 sequence means that when 2 is dialed the dial tone is applied in the line till 3 is dialed. 7) The period (.) character. If the period is placed following the certain digit this means that it can be included in the number any number of times. For example the 8x. sequence means that a number can be dialed that starts with 8 and has unlimited length and contains any digits. 8) S - it allows to change the value of the "T-shot Timeout" parameter for the current template that is predefined above. 9) L - it allows to change the value of the "T-long Timeout" parameter for the current template that is predefined above. Example: 012345679]x.# 8,[346789]xxxxxxxxx 8,10xxxxxxx 8,1[123] 8,1[489]x 8,[25][1-5]xxxxxxxxx 8,80xxxxxxxx 8,5[6789]xxxxxxxxxxxx 8,9998[068]1 [012345679]x.# 8,15xxx

Dial Plan Creation

1. Switch to the Dialplans subsection.
2. Click Add to add a new dial plan.
3. Assign a name to the dial plan, set up time slots and specify dialing templates.
4. Click Save to save applied changes.
5. Click Delete to delete the current dial plan.

4.2.2. Modifiers

A modifier is a block of parameters that contain data required to change the subscriber's number. The Modifiers subsection is used to manage rules for number changing.

Figure 12. Web Interface - Modifiers

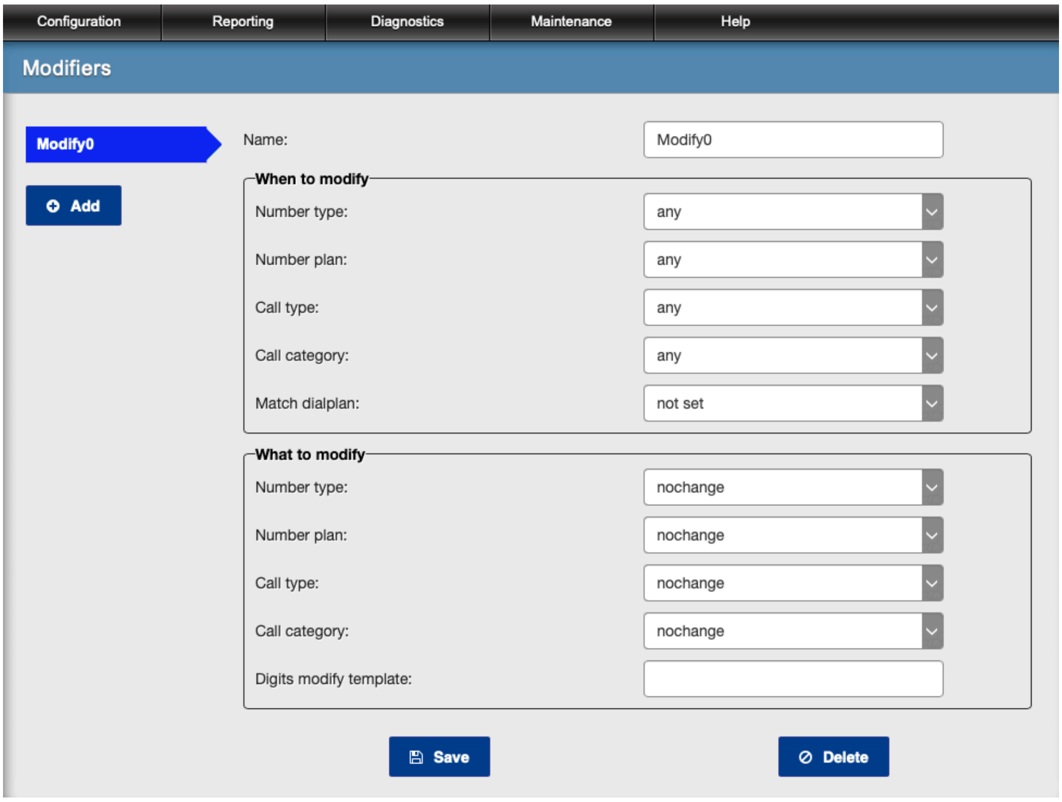


Table 17. Parameters for changing a subscriber number

Parameter	Description
Name	Name of the rule for changing the number.
When to Modify	
Number type	If the value is specified that differs from "any" value or from the value of the number to be checked then the current modifier will not be applied. Otherwise the process will proceed to checking the remaining conditions.
Number plan	If the value is specified that differs from "any" value or from the value of the number to be checked then the current modifier will not be applied. Otherwise the process will proceed to checking the remaining conditions.
Call type	Classifying calls by their type allows a more flexible routing configuration. The calls that are received by the gateway at the input can be classified by the following types after they undergo number filtering using the corresponding modifiers: local, long-distance and general. After the direction is selected this classifier will be used to select the specific link within the direction. There are directions that can include separate links for local and long-distance calls. If the direction contains only local links and the call is classified as a long-distance call this call will then be rejected. If the direction contains long-distance and general links and the call is classified as a long-distance call then the search for the free channel will initially be performed within long-distance links. If the free channel is not found there then the search will be performed within general links. By default all calls are classified as "general".
Call category	0 - 9 Category. It is used to identify various services for specific groups of telephone numbers. Please note that assignment of categories varies depending on specific city/country. The new value of the category from 0 to 9 when all specified conditions for selecting the modifier are met.
Match dialplan	It is selected from the list of the available dial plans and it is used to identify whether the number to be checked matches the templates of the selected dial plan. If the dial plan is assigned and the number does not match the template then the current modifier will not be applied. Otherwise the number to be checked will be modified with the values specified below.

Table 17. Parameters for changing a subscriber number - continued

Parameter	Description
What to modify	
Number type	The new value of the subscriber number parameter when all specified conditions for selecting the modifier are met.
Number plan	The new value of the subscriber number parameter when all specified conditions for selecting the modifier are met.
Call type	The new value of the subscriber number parameter when all specified conditions for selecting the modifier are met.
Call category	The new category value from 0 to 9, when all given conditions the choice of modifier.
Digits modify template	The template for modifying digits of the number to be checked. The template for modifying the number allows the following symbols: "x" - omit one digit of the number "." - delete one digit of the number "+" - start adding digits to the number "0..9" - digits of the number Let us consider a specific example of using mask symbols. Assume that the received number of the calling subscriber should be changed as follows: add digits 28 to the front of the number and replace the second received digit with digit 7. The mask would look like this: "+28x.+7".

Modifier Creation

1. Switch to the Modifiers subsection.
2. Click Add to add a new modifier.
3. Indicate conditions for selecting the modifier and specify the values modified by the modifier.
4. Click Save to save applied changes.
5. Click Delete to delete the current modifier.

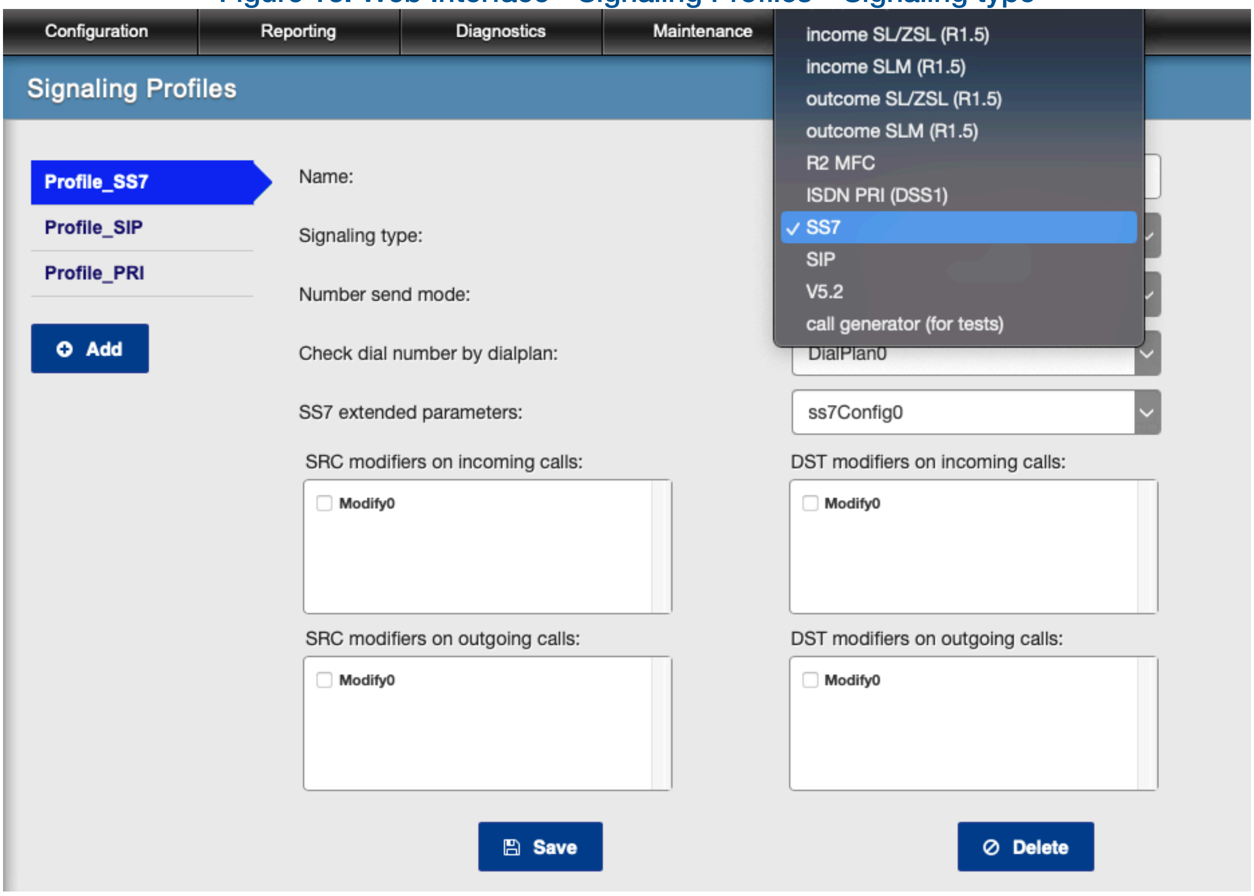
4.2.3. Signaling Profiles

To establish a connection for each of the channels depending on its type uses a specific set of exchange parameters. To make configuration of this set of parameters easier, the parameter set is consolidated into a logical structure that is called a signaling profile.

A single link signaling profile consolidates channels of the same type with the same set of exchange parameters. To do this, create in the Signaling Profiles subsection as many link configuration blocks as required to describe all the various parameters for existing communication channels.

> Warning! Each profile created in the Signaling Profiles section contains its own set of parameters depending on the type of selected signaling.
>
> Access to all signaling profile settings when a new profile is created or the signaling type of the existing profile is changed will be provided only after "Save" is pressed (Figure 13).

Figure 13. Web Interface - Signaling Profiles – Signaling type



Descriptions of all available configuration block types and their parameters are provided below.

Table 18. Settings for various types of signaling systems

Parameter	Description
Name	The name of the signaling profile.
Signaling type	Selected from the list of available signaling types.
Register exchange type	Assigns the type of register signaling to be used for local and/ or long-distance/international calls. This parameter is used for CAS family signaling only.
Number send mode	Controlling the mode of "complete/incomplete address information".
Check dial number by dialplan	Selected from the list of the available dial plans and used to identify the indicator that indicates when dialing is complete.
SRC (DST) number modifiers on incoming calls	Selected from the list of available rules for modifying a number (modifiers).
SRC (DST) number modifiers on outgoing calls	Similar to the parameter for incoming calls.
ANI settings	These parameters define the following methods for requesting information from the Automatic Number Identifier for each type of connection (local or long-distance): request information from the Automatic Number Identifier either before the called subscriber responds or immediately after the subscriber responds; use the 500Hz signal generator when requesting information from the Automatic Number Identifier; number of repeated attempts of requesting the Automatic Number Identifier when a previous attempt failed. It is also possible to specify a condition for issuing the Automatic Number Identifier information either only when 500Hz frequency request is available or always when there is a response from the subscriber or never.
MFS set for register exchange	This parameter is designed to assign the individual option of exchanging dual-tone pulses and it is used with signaling where shuttle transfer of address information is applied. (see the table below - Multifrequency Shuttle - multifrequency exchange over circuits between TEs). Example: request 6 digits - 122222, request 3 following digits - 222
Expected decadic digits count	This parameter can be used to identify the indicator of completion of dialing the called subscriber number based on the specified number of expected digits. It is used only when the number is received using the decade (battery) method.
Index of the first sending digit from DST number	Used to specify the index of the first digit of the called subscriber number issued during the address exchange.
CAS bits inversion	Controls signal channel inversion.
ISDN PRI interface type	Selected from the list - Network or Terminal Equipment.

Table 19. MFC code

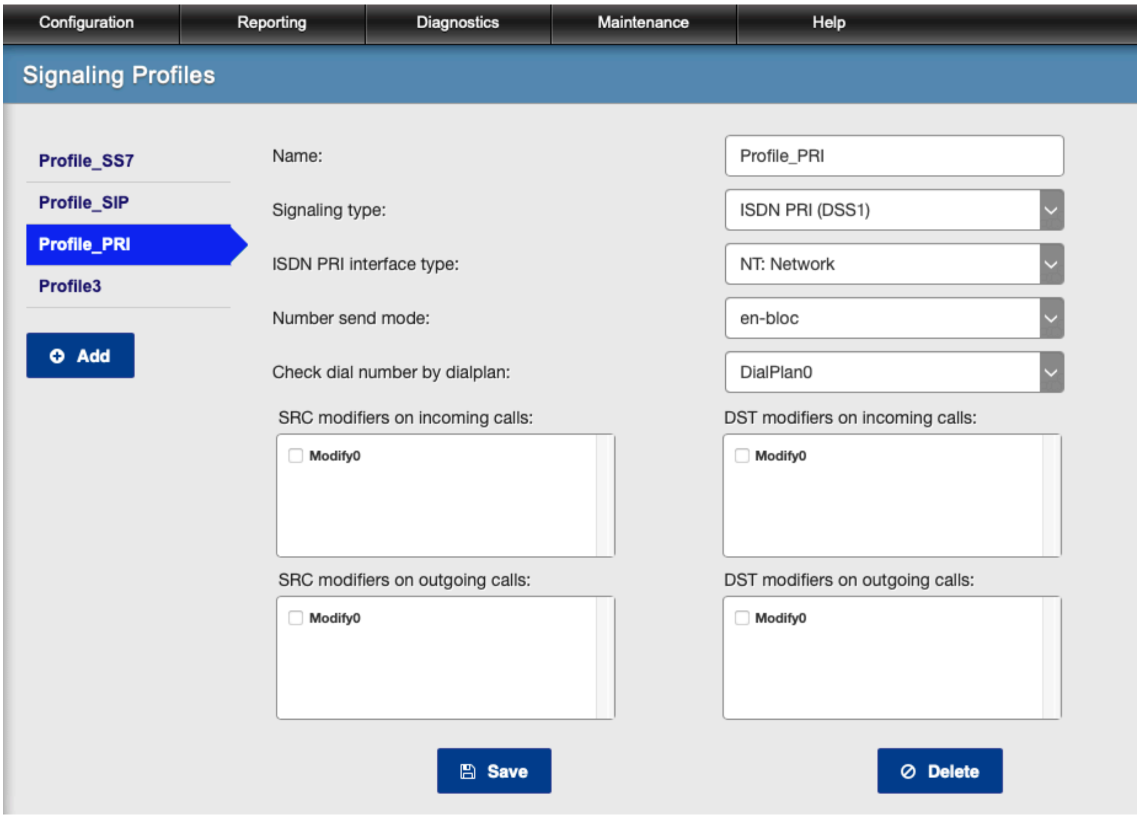
Frequency Pattern		Pattern No.	Transmit Signal Value	Receive Signal Value	Signal Value in Automatic Number Identifier Package
f0 f1	700+900	1	Digit "1"	Request for "frequency transmission of the first digit"	Digit "1"
f0 f2	700+1100	2	Digit "2"	Request for "frequency transmission of the next digit"	Digit "2"
f1 f2	900+1100	3	Digit "3"	Request for "frequency transmission of the previous digit"	Digit "3"
f0 f4	700+1300	4	Digit "4"	"number receiving completed"	Digit "4"
f1 f4	900+1300	5	Digit "5"	"disconnect" (for unavailability or other reasons)	Digit "5"
f2 f4	1100+1300	6	Digit "6"	Request "repeat the previous signal that was received with distortion"	Digit "6"
f0 f7	700+1500	7	Digit "7"	"no free ways"	Digit "7"
f1 f7	900+1500	8	Digit "8"	Request for "battery transmission of all digits starting with the first digit"	Digit "8"
f2 f7	1100+1500	9	Digit "9"	Request for "battery transmission of the next digit"	Digit "9"
f4 f7	1300+1500	0 (A)	Digit "0"	Request for "battery transmission of the previous digit"	Digit "0"
f0 f11	700+1700	B	---	---	---
f1 f11	900+1700	C	Confirmation of receiving 4,5,8,9,A signals	---	---

Frequency Pattern		Pattern No.	Transmit Signal Value	Receive Signal Value	Signal Value in Automatic Number Identifier Package
f2 f11	1100+1700	D	Request to repeat the signal that was received with distortion	---	Start of the bundle
f4 f11	1300+1700	E	---	---	Instead of the second of 2 same successive digits
f7 f11	1500+1700	F	---	"no frequency information"	---

Data Entry

1. Switch to the Signaling Profiles subsection of the configuration setup menu.
2. Click Add to add a new profile.
3. Specify the profile name and select the signaling type. Click Save to save the profile.
4. Select the added profile from the list of profiles, move the mouse cursor over the profile name and click the left mouse button to load the signaling profile parameters available for editing.
5. Edit the profile parameters based on the type of selected signaling.
6. Click Save to save applied changes.
7. Click Delete to delete the profile.

Figure 14. Web Interface - Signaling Profiles



Common parameters for all signaling types (excluding V5.2):

- Number send mode
- Check dial number by dialplan

4.2.3.1. Inbound SL/ZSL/SLM (R1.5 Signaling)

System R1.5 is an asymmetrical protocol that is widely used in East Europe over E1 trunks. It has provisions to interface with analog and digital switches.

Protocols R1.5 uses three different register signaling types:

- Decadic
- Multi-Frequency Pulse Shuttle (MFS)
- Multi-Frequency Continuous Packet (MFCP) (implemented specifically for the transmission of ANI information).

When organizing inbound lines over R1.5 protocol, the corresponding signaling type should be selected: Inbound SL/ZSL (R1.5) or Inbound SLM (R1.5).

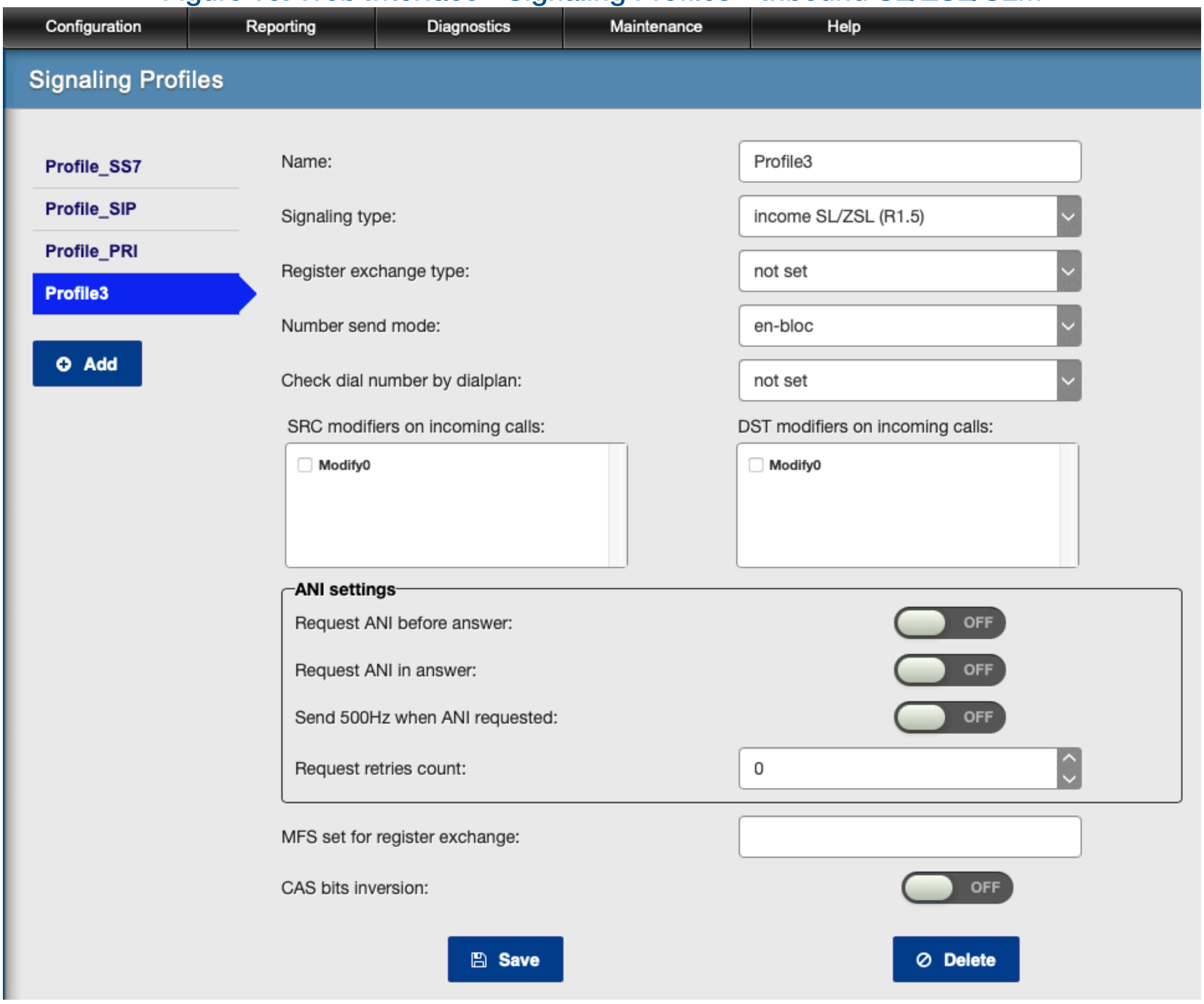
In addition to setting parameters common to all signaling types these signaling types should also have the parameters specific to R1.5 protocol set to their required values:

- Register exchange type
- CAS bits inversion.

You should also set the following additional parameters to their required values:

- Request ANI before answer
- Request ANI in answer
- Send 500Hz when ANI requested
- Request retries count
- MFS set for register exchange.

Figure 15. Web Interface - Signaling Profiles – Inbound SL/ZSL/SLM



R1.5 (SL/ZSL/SLM) – these are versions of CAS signaling used in Eastern Europe.

SL - Direct connection lines (local inbound and outbound side)

ZSL – Requested connection lines (long distance outbound side)

SLM - Incoming inter-city direct connection lines (long distance inbound side only)

ANI – specific CID (Caller ID) version for Eastern Europe.

4.2.3.2. Outbound SL/ZSL/SLM (R1.5 Signaling)

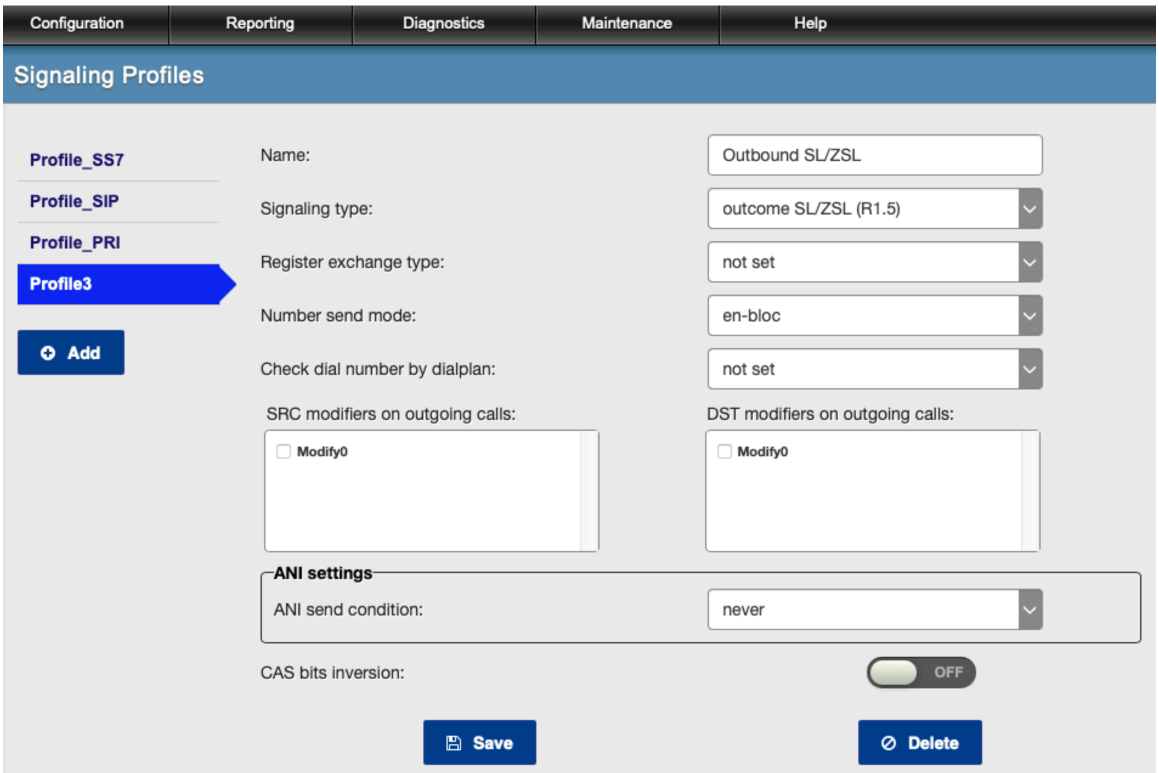
When organizing outbound lines using the R1.5 protocol, the corresponding signaling type should be selected: Outbound SL/ZSL (R1.5) or Outbound SLM (R1.5).

In addition to setting parameters common to all signaling types, these signaling types should have the parameters specific to R1.5 protocol set to their required values:

- Register exchange type
- CAS bits inversion

You should also set the additional parameter ANI Send Condition to its required value.

Figure 16. Web Interface - Signaling Profiles – Outbound SL/ZSL/SLM



ANI – specific CID (Caller ID) version for Eastern Europe.

4.2.3.3. R2 MFC Digital

R2 MFC is a signaling protocol (type CAS) that operates on E1 connections and is used internationally between switches. R2 MFC is an ITU-T Q400 series standard. The SIP/E1 Gateway supports the digital version of line signals and inter-register signals. R2 signaling from switch to switch supports the following MFC call types:

- Simple calls (Incoming and outgoing calls): This is the signaling used between a PBX and a Central Office for local calls and toll free calls.
- Calls with Caller ID: This is used between central offices for tracking the call to report the number to the switch being called and to apply billing charges when necessary.

When organizing two-way lines with the R2 MFC signaling type, you should set the parameters specific to CAS protocol to their required values in addition to setting parameters common to all signaling types:

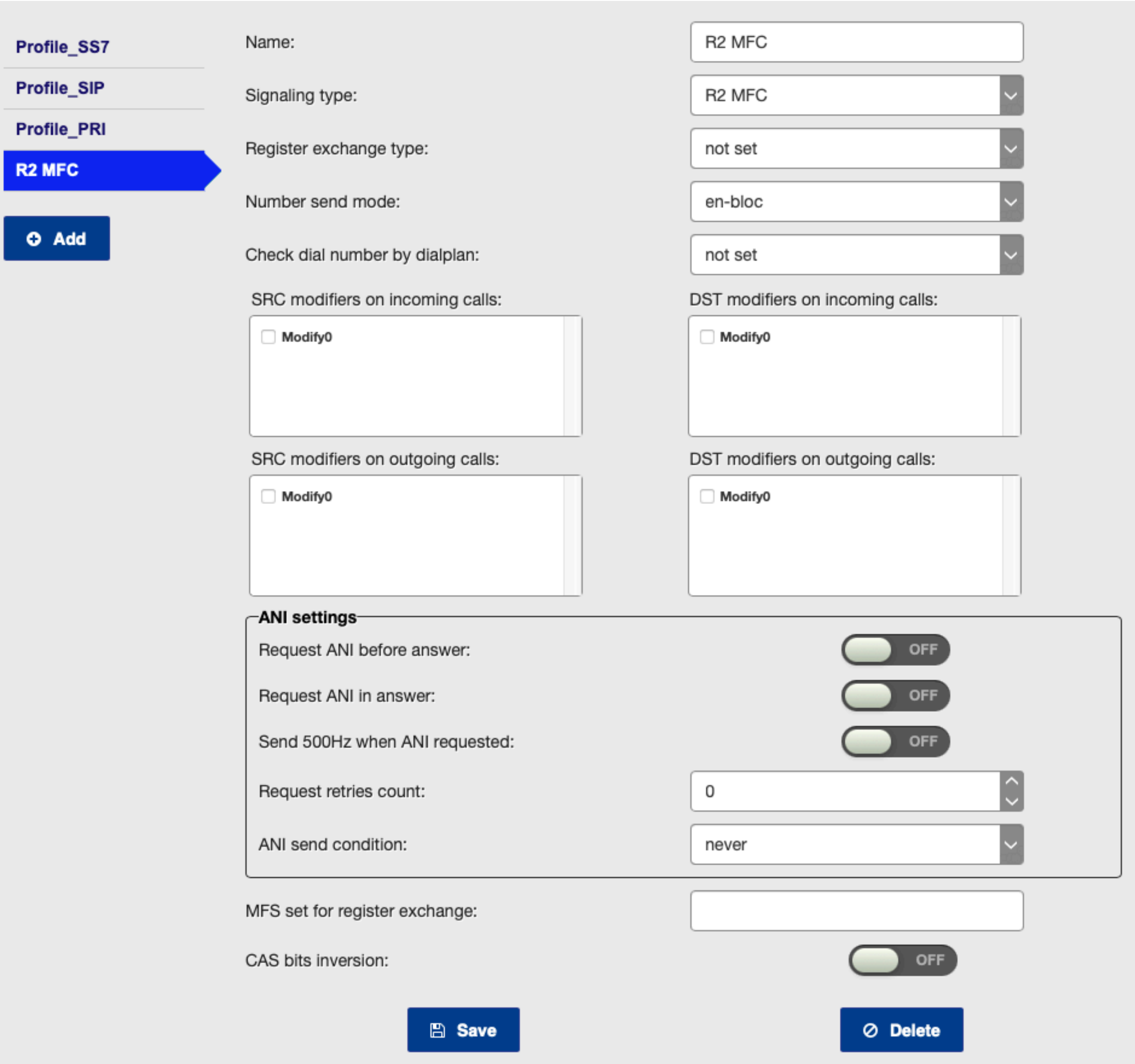
- Register exchange type
- CAS bits inversion

For an outbound connection you should set the following additional parameters to their required values:

- Request ANI before answer
- Request ANI in answer
- Send 500Hz when ANI requested
- Request retries count
- MFS set for register exchange

For an inbound connection you should set the additional parameter ANI Send Condition to its required value.

Figure 17. Web Interface - Signaling Profiles – R2 MFC Digital

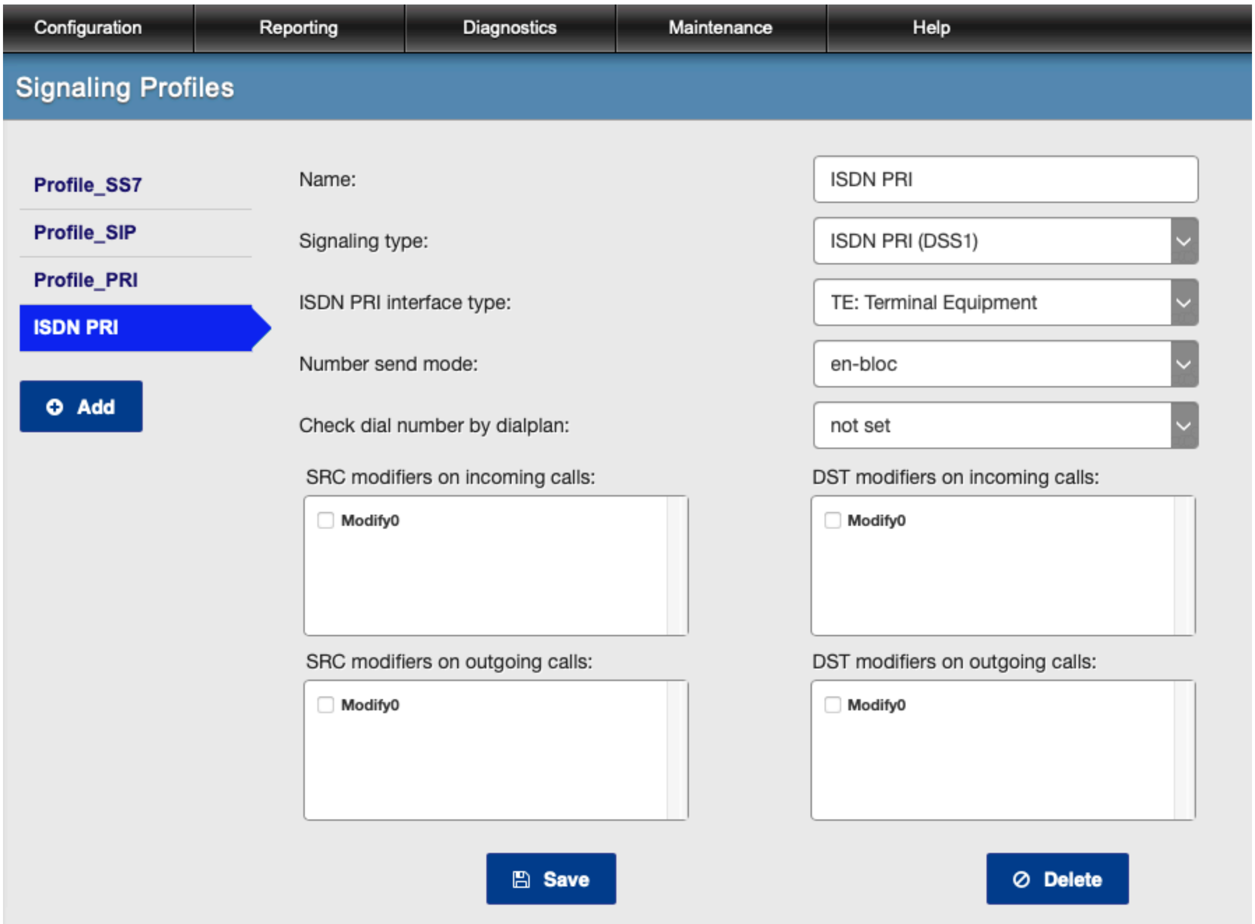


ANI – specific CID (Caller ID) version for Eastern Europe.

4.2.3.4. ISDN PRI (DSS1)

For DSS1 signaling, in addition to setting parameters common to all signaling types, you should set the ISDN PRI Interface Type parameter to the value that would match the configuration: TE (Terminal Equipment) or NT (Network Termination).

Figure 18. Web Interface - Signaling Profiles – ISDN PRI (DSS1)



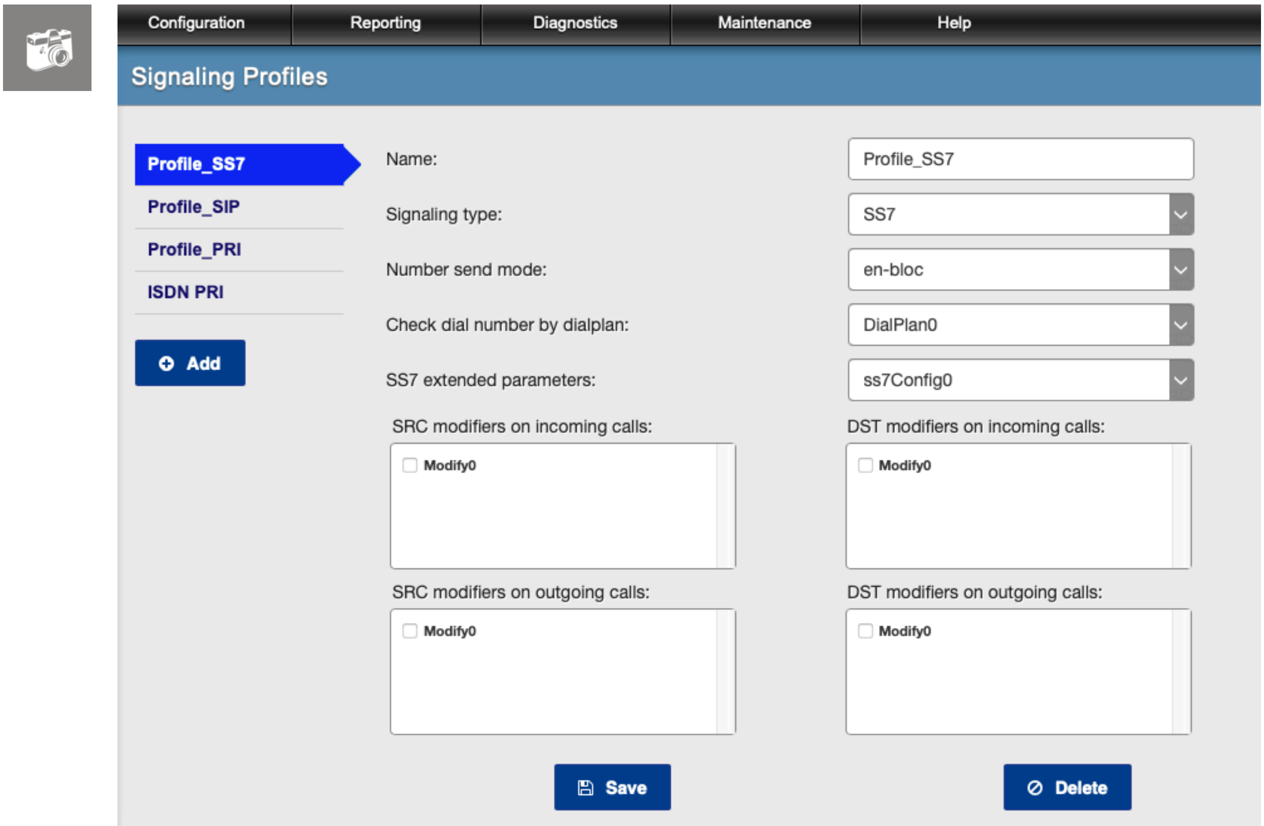
4.2.3.5. SS7

Signaling System 7 (SS7) is an out-of-band signaling system that provides fast call setup using circuit-switched connections and transaction capabilities for remote database interactions.

SS7 is the carrier signaling protocol used for call control and providing Intelligent Networking (IN) and Advanced IN (AIN) services. SS7 is often used for applications such as IP signaling gateways, wireless infrastructure, and a wide variety of in-network enhanced services, including voice and fax messaging, one-number/follow-me, number portability, and pre-paid services.

For SS7 signaling, in addition to setting parameters common to all signaling types, you should select the SS7 Extended Parameters that were created earlier (see the SS7 Configuration section).

Figure 19. Web Interface - Signaling Profiles – SS7

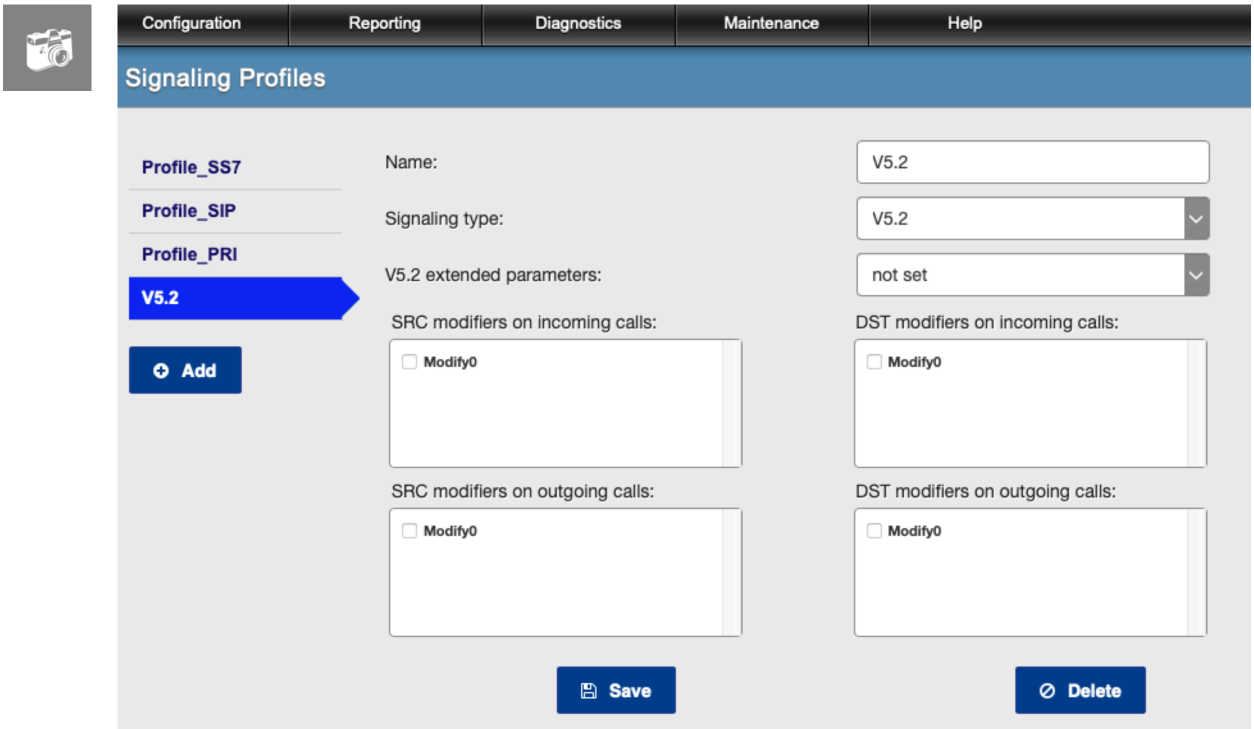


4.2.3.6. V5.2

A protocol that defines the switching and signaling protocol between access network (AN) and the local exchange (LE) to support analog telephone access, ISDN (BRI and PRI) access, and other digital or analog access for semi-permanent connections.

When creating a V5.2 signaling profile you should select the V5.2 Extended Parameters that were created earlier (see the V5.2 Configurations section).

Figure 20. Web Interface - Signaling Profiles – V5.2

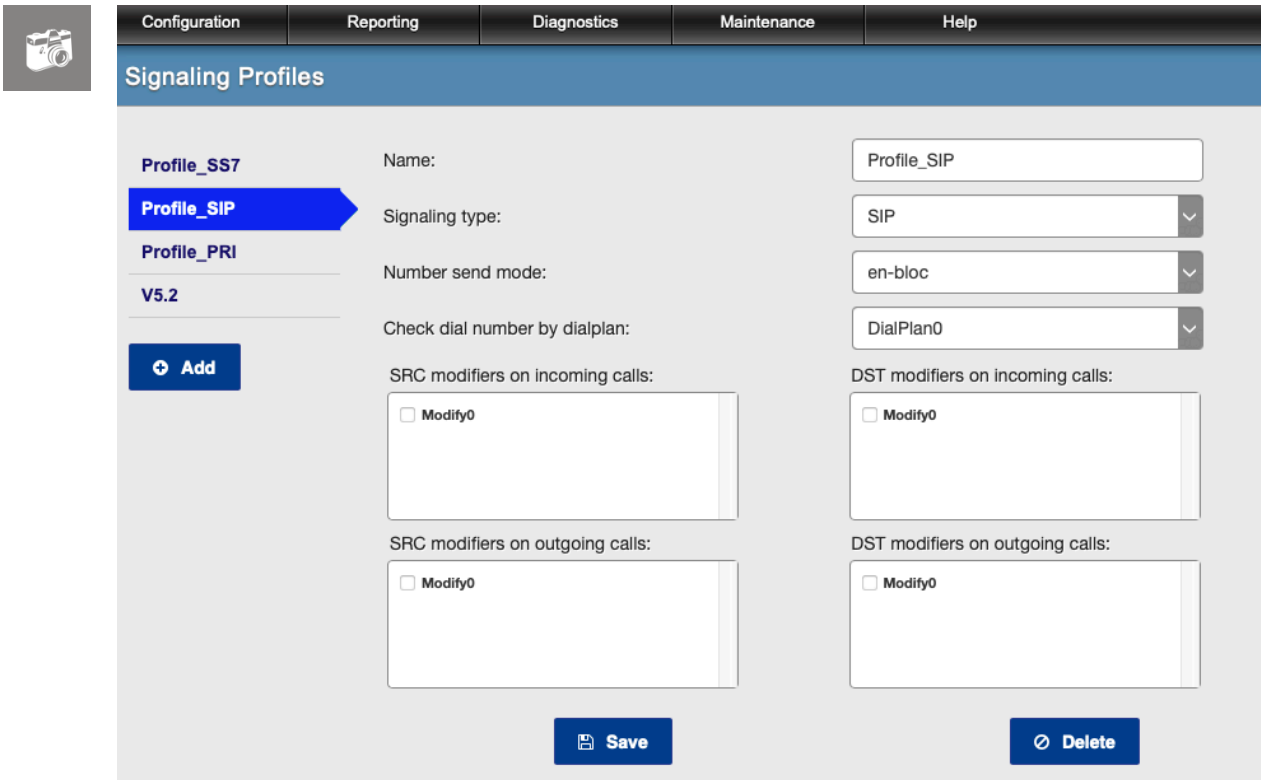


4.2.3.7. SIP

SIP (Session Initiation Protocol) is a signaling protocol used to create, manage and terminate sessions in an IP based network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. This makes it possible to implement services like voice-enriched e-commerce, web page click-to-dial or Instant Messaging with buddy lists in an IP based environment.

A SIP signaling profile requires setting parameters common to all signaling types only since it does not have any specific settings.

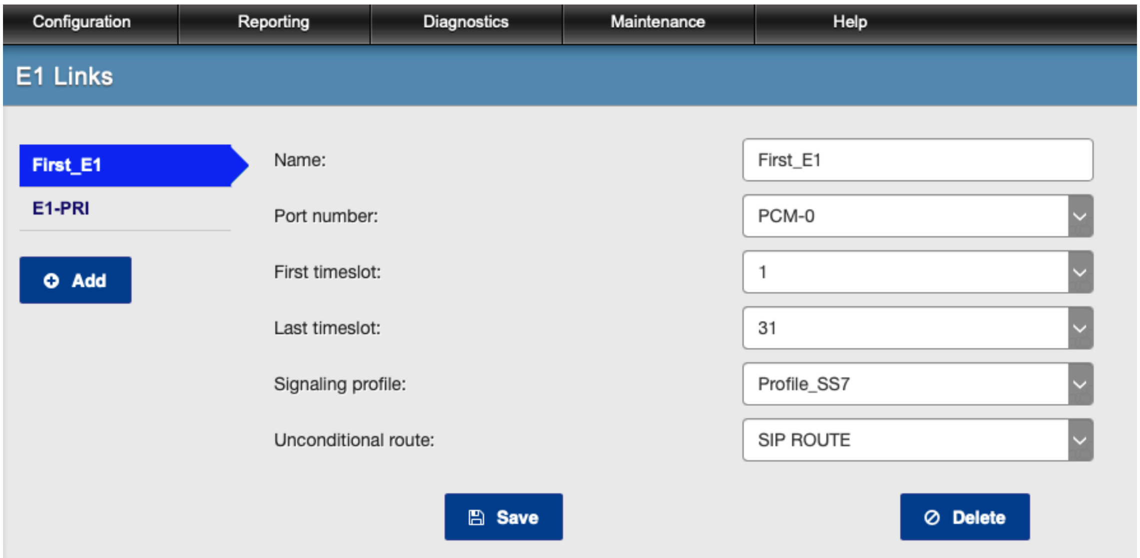
Figure 21. Web Interface - Signaling Profiles – SIP



4.2.4. E1 Links

Each E1 port contains 32 channels numbered from 0 to 31. The SN9000 implements a logical consolidation of channels with common protocol specifications within trunks. These specifications include the line signaling type in use, direction, register exchange type, etc. Such a consolidation is called a link. As an example, all channels of incoming connections of the specific E1 port can be consolidated within a single link. Each link consolidates channels that belong to one E1 port only.

Figure 22. Web Interface - E1 Links



Each link has its own set of parameters.

Table 20. E1 link parameters

Parameter	Description
Name	E1 link name.
Port number	E1 link port number (from 0 to 15).
First timeslot	The starting boundary of channel numbers within the given link.
Last timeslot	The ending boundary of channel numbers within the given link.
Signaling profile	Indicates the configuration profile for the given link and it is selected from the list of the previously created signaling profiles.
Unconditional route	Allows the user to assign one of the available routes that is used for further routing of the call that was received within the current link without the need to perform procedures of selecting the route based on numbers of the calling and/or called subscribers.

Link Creation

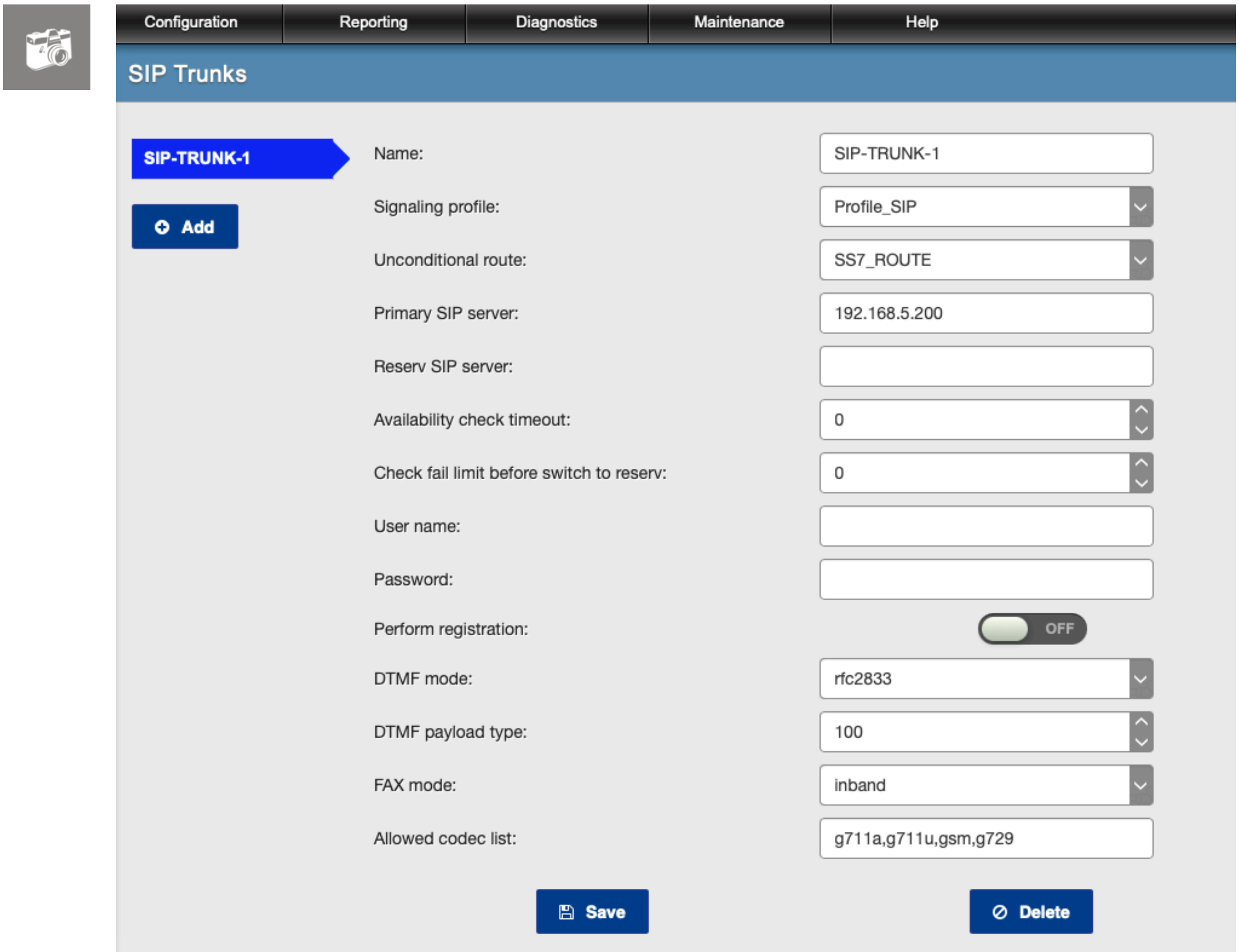
1. Switch to the E1 Links subsection of the configuration setup menu.
2. Click Add to add a new link.
3. Edit the link parameters or select a value from the drop-down lists.
4. Click Save to save applied changes.
5. Click Delete to delete the current E1 link.

4.2.5. SIP Trunks

SIP (Session Initiation Protocol) is the data transfer protocol that describes the method to establish and terminate the user session including multimedia contents exchange.

The "SIP Trunks" section is designed to control groups of IP lines combined within the same SIP authorization.

Figure 23. Web Interface – SIP Trunks



Each SIP trunk has its own set of primary and additional parameters.

Table 21. SIP Trunk Parameters

Parameter	Description
Name	The name of the created SIP trunk.
Signaling profile	Indicates the configuration profile for the given trunk and it is selected from the list of the previously created signaling profiles.
Unconditional route	Allows you to assign one of the available routes for further routing of the call that was received within the current account without the need to perform procedures of selecting the route based on numbers of the calling and/or called subscribers.
SIP settings	
Primary SIP server	SIP server IP address.
Reserve SIP server	Reserved SIP server IP address.
Availability check timeout	Time interval (in seconds) - at the end of each interval the primary SIP server is checked for availability.
Check failure limit before switch to reserve	The number of unsuccessful checks for availability of the primary SIP server allowed before switching to the reserved SIP server.
User Name	SIP user name.
Password	SIP user password.
Perform registration	Enables/disables the registration procedure on the SIP server.
DTMF mode	DTMF signal transfer mode.
DTMF payload type	This parameter is used in headers when transferring DTMF signals.
FAX mode	Fax message transfer mode.
Allowed codec list	Comma separated list of codecs used to code voice data. The first item in the list has the highest priority.

Trunk Creation

1. Switch to the SIP Trunks subsection of the configuration setup menu.
2. Click Add to add a new trunk.
3. Edit the SIP trunk parameters or select the value from the drop-down lists.
4. Click Save to save applied changes.
5. Click Delete to delete the current trunk.

Table 22. Codecs for encoding voice data

	Codec	Description
	g711 (g711a, g711u)	The standardized ITU-T codec that is used in ISDN devices. The required bit rate is 64 kbit/s. There are 2 types of the codec - a-law (G711a or PCMA) and µ-law (G711µ or PCMU). They differ in coding algorithms. The codec is supported by practically all IP-telephony devices.
	g729	The standardized ITU-T codec designed to transmit voice with "good quality" using a small bit rate (8 kbit/s). It is supported by practically all vendors.
	gsm	The voice codec used for voice coding in GSM phones with 13 kbit/s bit rate.

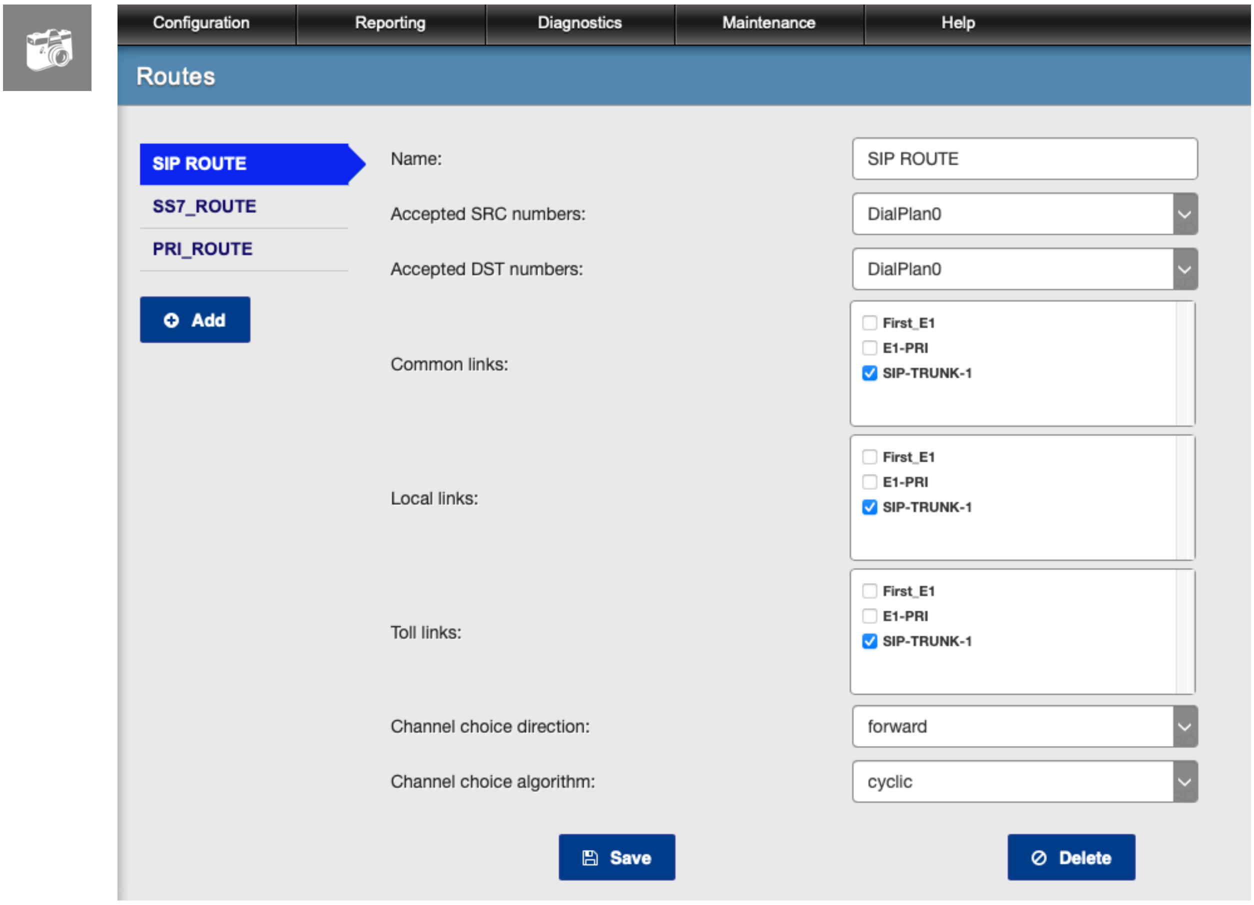
4.2.6. Routes

Each utilized E1 port is connected to the external TE (Telephone Exchange) or other exchange equipment. These connections are the basis for creating routes.

The Routes subsection is used to combine the links to a directional route. The links within their route are divided conditionally into 3 groups:

- General links that can be used for both local and long-distance/international calls;
- Local links that are used for local calls only;
- Long-Distance links that are used for long-distance/international calls only.

Figure 24. Web Interface - Routes



Each route has its own set of parameters.

Table 23. Route parameters

Parameter	Description
Name	Name of the route to be created.
Accepted SRC numbers	Indicates the dial plan that is used to select this route based on the number of the calling subscriber (subscriber A).
Accepted DST numbers	Indicates the dial plan that is used to select this route based on the number of the called subscriber (subscriber B).
Common links	Selected links that are used for all types of calls (local, long-distance/international). They are selected from the general list (E1 links plus SIP links). To use the links correctly, assign them clear and accurate names when creating E1 and SIP links.
Local links	Selected links used for local calls only. They are selected from the general list (E1 links plus SIP links). To use the links correctly, assign them clear and accurate names when creating E1 and SIP links.
Toll links	Selected links used for long-distance/international calls only. They are selected from the general list (E1 links plus SIP links). To use the links correctly, assign them clear and accurate names when creating E1 and SIP links.
Channel selection direction	Assigns the direction to bypass channels of the given route. When the forward selection is used the link channel with the lowest sequence number will be selected the first while when the backward selection is used the first number to be selected will be the number with the highest sequence number.
Channel selection algorithm	Assigns the algorithm to search for a free channel. When "sequential cyclic marker" is selected the search will be performed sequentially over the loopback list starting with the value of the last occupied channel and further on in accordance with the Selection Direction parameter. Two other cyclic algorithms that account for even priority/odd priority operate in a similar way. When the parameter has the "reset marker to start" value, the search for the free channel will start from the beginning of the list every time. Controlling channel selection algorithms minimizes risk when "bidirectional occupation" of the channel occurs.

Route Creation

1. Switch to the Routes subsection.
2. Click Add to add a new route.
3. Edit route parameters, select the value from the drop-down list and mark the required position(s) in the open lists.
4. Click Save to save applied changes.
5. Click Delete to delete the active route.

4.2.7. SS7 Configurations

SS7 configuration is used to set up the parameters of SS7 signaling points. The following parameters should correspond with the remote side to establish connections using the SS7 protocol:

- OPC - Origination Point Code. The address that provides unambiguous identification of the product in the SS7 network. When the connection is established this address should be coordinated with the opposite TE.
- DPC - Destination Point Code. Address of the TE to which the product is connected.
- NI - Network Indicator. Indicator of the SS7 network with which the connection will be established. This can be a national or international or any other network. The parameter should be coordinated with the opposite TE.
- Signaling link position. It specifies the E1 port and channel interval where the signaling link is located.
- CIC - numbers of speech time slots. When a connection is established over SS7 the time slot number is specified for creating the speech path. If numbers on different sides do not match the connection will not be established.

Data Entry

1. Switch to the SS7 Configurations subsection.
2. Click Add to add a new configuration.
3. Edit the SS7 configuration parameters.
4. Click Save button to the right of the configuration name to save applied changes.
5. Click Apply button to the right of the configuration name to save and apply the changes.
6. Click Delete to delete the current configuration.

The configuration page is divided into three dialog tabs (Figure 25) that contain parameters to configure MTP2, MTP3 and ISUP levels.

Figure 25. SS7 Configurations - Dialog Tabs of SS7 Setup and Configuration Levels

Figure 26. SS7 Configurations - MTP2 Terminals

Table 24. MTP2 Settings

Parameter	Description
Name	The logical name.
E1 port	The number of the E1 port where the signaling channel is located.
Timeslot	The number of the time slot used for the signaling channel (link). Any channel time slot can be used for signaling. As a rule time slot 1 or 16 is selected for this purpose. This value should be coordinated with the opposite exchange.
FISU period	Assigns (in milliseconds) the period for sending fill-in packets when there is no payload. If the parameter is set to 0 (the default value) then the hardware engine will be applied to send/receive such packets.

Data Editing

1. Click Add to add a new item to the Terminals section.
2. Move the mouse cursor over the line and click the left button of the mouse to edit values of the selected item. Items to be edited are highlighted in one color and vertical editing cursor is set up (Figure 26).
3. Click Save in the Terminals section to save applied changes.
4. Click Delete to delete the highlighted item in the Terminals section.

Figure 27. SS7 Configurations - MTP3 Settings

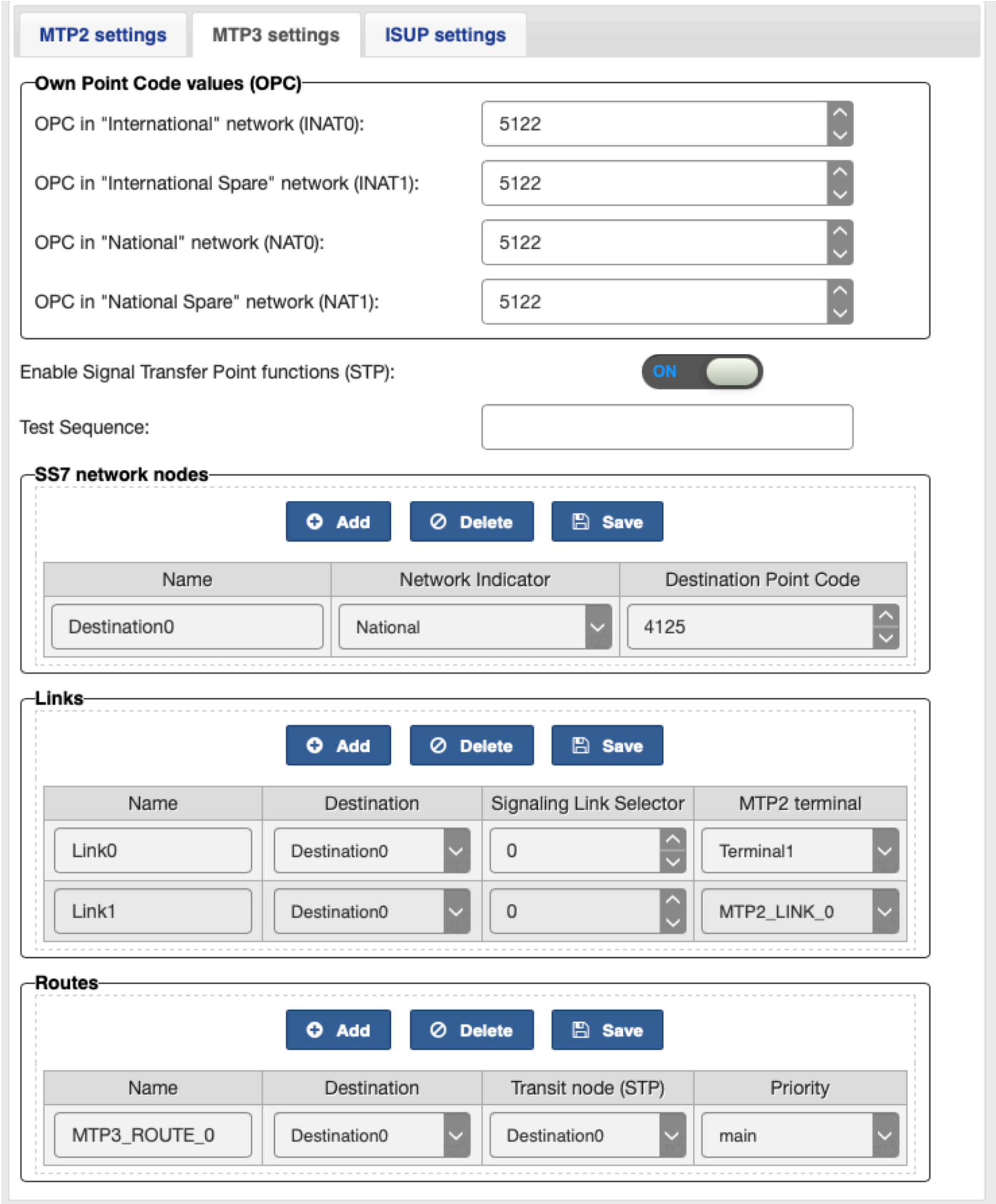


Table 25. MTP3 Settings

Parameter	Description
Own Point Code values (OPC)	
OPC in "International" network (INAT0)	The OPC (Origination Point Code) value is assigned that is used within the international network.
OPC in "International Spare" network (INAT1)	The OPC value is assigned for the spare network (used within the international network only).
OPC in "National" network (NAT0)	The OPC value is assigned for the national network.
OPC in "National Spare" network (NAT1)	The OPC value is assigned for the spare network (used within the national network only).
Enable Signal Transfer Point functions (STP)	If enabled, the configured SS7 signaling node will be a transit node too, i.e. it can perform functions of "transparent" routing of the signaling traffic.
Test Sequence	The set of test data used when testing the signaling link.
SS7 network nodes	
Name	The logical name of the destination.
Network Indicator	The network indicator. Indicates the network operated for this destination.
Destination Point Code	The address of the remote signaling point for the given destination.
Links	
Name	The logical name of the link set.
Destination	The destination is selected that will be used by the link set.
Signaling Link Selector	The SLS value of the given signaling link. To improve connection reliability one link set can contain several signaling channels. The SLS parameter serves to number these signaling channels. Generally, when only one signaling link is used this parameter does not have a value and equals 0 as a rule.
MTP2 Terminal	Indicates the terminal assigned on MTP2 tab.
Routes	
Name	The logical name of the route.
Destination	Destination route.
Transit node (STP)	The route used to transit the signaling traffic.
Priority	Route priority.

Data Editing

1. Click Add in the corresponding subsection to add a new item to Destinations, Links, and Routes sections.
2. Move the mouse cursor over the line and click the left button of the mouse to edit values of the selected item. The items to be edited are highlighted in one color. Edit the required value or select the value from the drop-down list.
3. Click Save in the required SS7 network nodes, Links, or Routes section to save applied changes.

Figure 28. SS7 Configurations - ISUP Settings

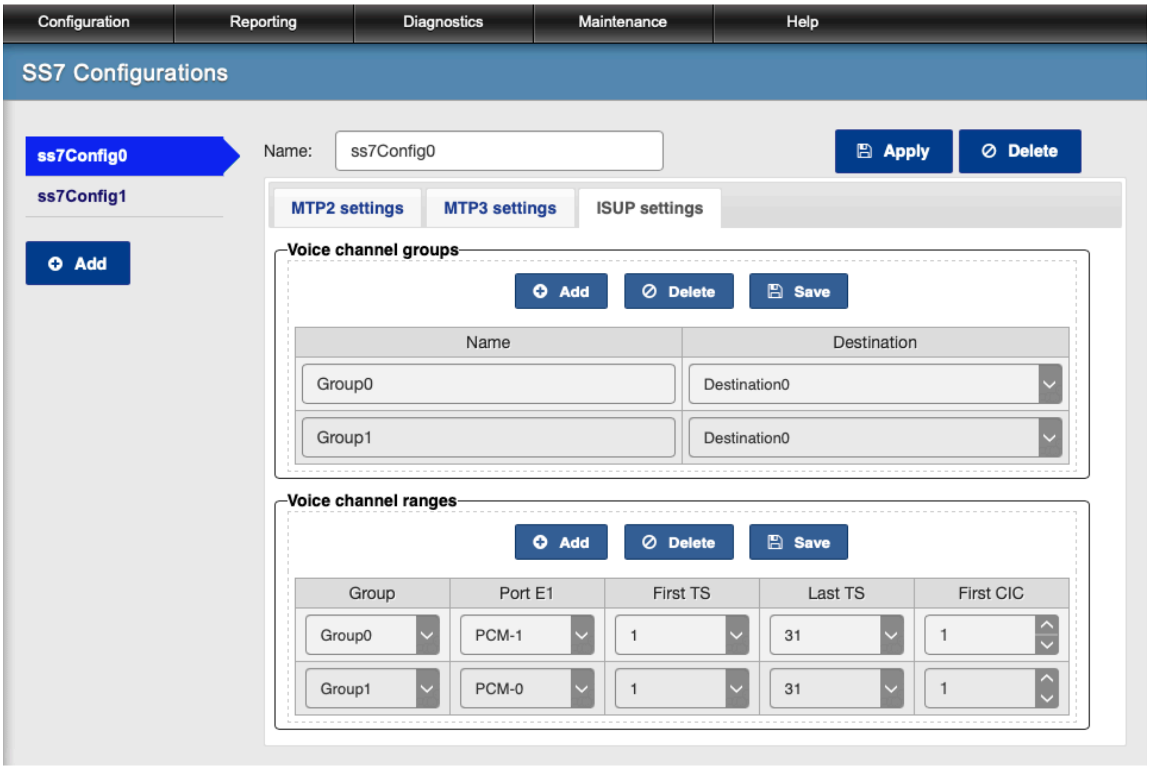


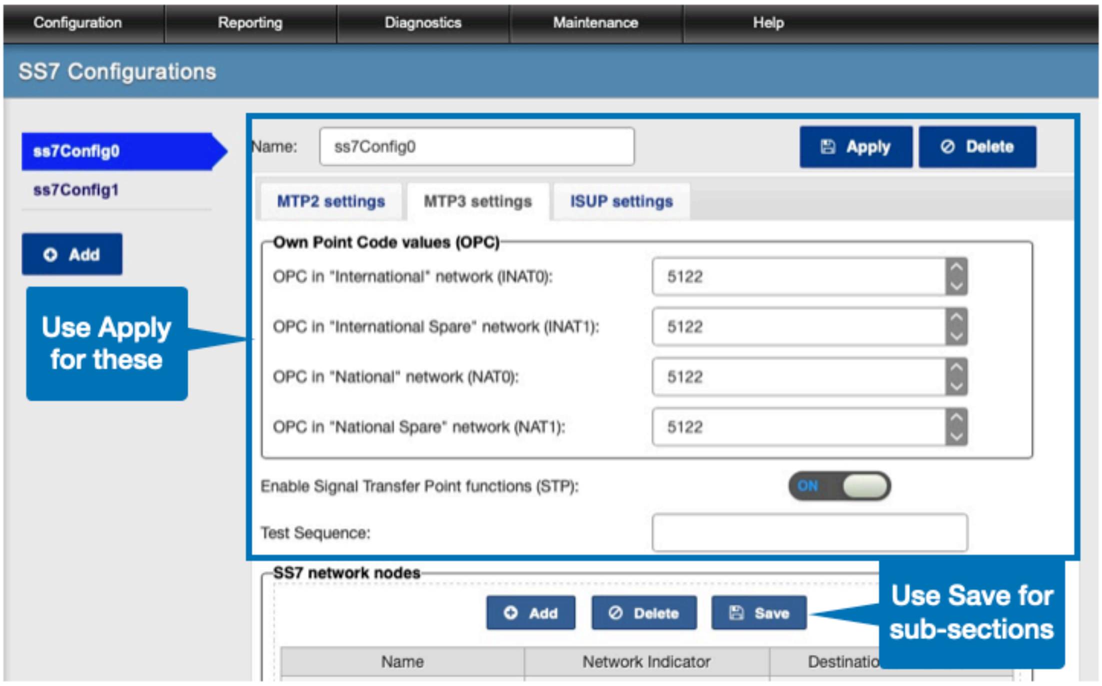
Table 26. ISUP Settings

Parameter	Description
Voice channel groups	
Name	The logical name of the channel group.
Destination	Indicates the destination assigned on MTP3 tab (SS7 network nodes).
Voice channel ranges	
Group	Indicates that a channel belongs to a group of channels that was defined earlier.
Port E1	The number of the E1 port where the given voice channel range is located.
First TS	The number of the first time slot within the range.
Last TS	The number of the last time slot within the range.
First CIC	The starting value of CIC (Circuit Identification Code) numbering that is used for this range. This parameter should correspond clearly with the opposite exchange.

Data Editing

1. Click Add to add a new item in Voice Channel Groups, or Voice Channel Ranges sections.
2. Move the mouse cursor over the line and click the left button of the mouse to edit values of the selected item. The items to be edited are highlighted in one color. Edit the parameter value or select the value from the drop-down list.
3. Click Save in the required Voice Channel Groups, or Voice Channel Ranges section to save applied changes.

Figure 29. SS7 Configurations – Save and Apply buttons



Warning! Various subsections within SS7 Configurations section have their own Save buttons. The Save button designed for a specific subsection is effective for that subsection only.

To save the parameter values that do not belong to any subsection the global Apply button is used that is located to the right of the configuration name (see the figure above).

4.2.8. V5.2 Configurations

The V5.2 configuration is used to configure the parameters of interfaces used for servicing subscriber nodes connected over V5.2 signaling systems.

Figure 30. Web-interface - V5.2 Configurations

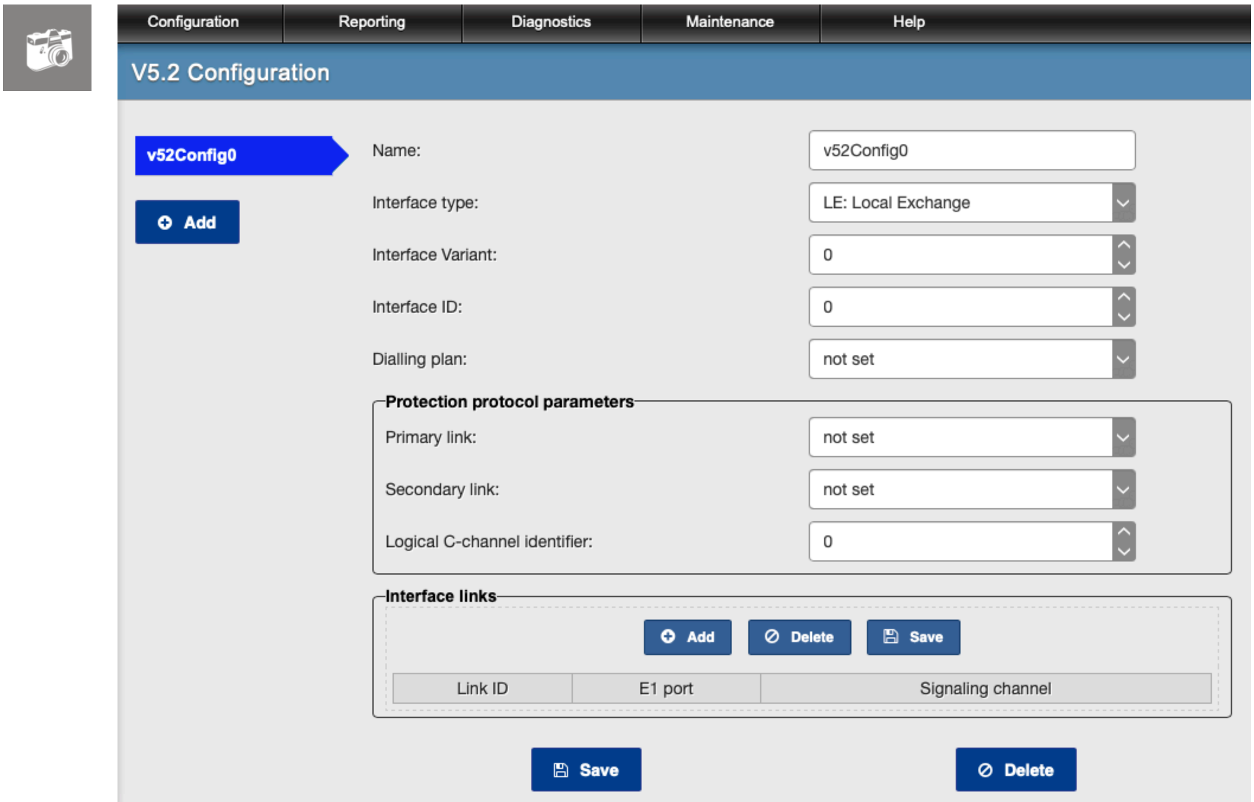


Table 27. V5.2 configuration settings

Parameter	Description
Name	The logical name.
Interface type	Indicates the type of the used interface (“LE Local Exchange”, “AN Access Network”).
Interface Variant	Assigns the value of the Provision Variant parameter. This parameter should be the same on both sides at the stage of installing the equipment.
Interface ID	Assigns the value of the Provision Interface ID parameter. This parameter should be the same on both sides at the stage of installing the equipment.
Dialing plan	Defines rules for number dialing for the subscribers of this interface.
Primary link	The main link of the protection protocol is assigned from the list of the available interface links.
Secondary link	The backup link of the protection protocol is assigned from the list of the available interface links.
Logical C-channel identifier	The unique 16-bit long number of the logical channel of AN and LE interface Allowed number values are from 0 to 65535.

Data Entry

1. Switch to the V5.2 Configurations subsection.
2. Click Add to add a new configuration.
3. Edit V5.2 configuration parameters.
4. Click Save at the bottom of the form to save applied changes.
5. Click Delete to delete the current configuration.

Each V5.2 interface can include from 1 up to 16 E1 ports (links) whose time slots can be used to carry both signaling data and speech channels.

Table 28. E1 link parameters

Name	Description
Link ID	Assigns a unique link identifier within the interface. The value of this parameter is used by the link control protocol.
E1 port	Indicates the physical E1 port used to place the link channels.
Signaling channel	Assigns the number of the channel interval used to carry the signaling information of the selected interface.

Data Entry

1. Click Add within the Interface Links group to add the interface link.
2. Edit parameters of the selected link. Move the mouse cursor over the line and click the left button of the mouse to edit the parameter values of the selected item. The items to be edited are highlighted in one color. Click Save within the Interface Links group to save applied changes.
3. Click Delete within the Interface Links group to delete the current interface link line.

4.2.9. DLU Configurations

DLU configuration is used to configure the transport layer parameters for a DLU interface.

Data Entry

1. Switch to the 'DLU Configuration' subsection.
2. Click Add to add a new configuration.
3. Edit DLU configuration parameters.
4. Click Save button in configuration level subsection to save applied changes.
5. Click Apply button to the right of the configuration name to save and apply the applied changes.
6. Click Delete button to delete the current configuration.

Table 29. MTP2 Settings

Parameter	Description
Name	The logical name.
E1 port	The number of the E1 port where the signaling channel is located.
Timeslot	The number of the time slot used for the signaling channel (link). Any channel time slot can be used for signaling. As a rule time slot 1 or 16 is selected for this purpose. This value should be coordinated with the opposite exchange.
FISU period	Assigns (in milliseconds) the period for sending fill-in packets when there is no payload. If the parameter is set to 0 (the default value) then the hardware engine will be applied to send/receive such

Data Editing

1. Click Add to add a new item to the 'MTP2 Terminals' section.
2. Move the mouse cursor over the line and click the left button of the mouse to edit values of the selected item. Items to be edited are highlighted in a darker color and a vertical editing cursor is set up (Figure 32).
3. Click Save button in 'MTP2 Terminals' section to save the applied changes.
4. Click Delete button to delete the highlighted item in 'MTP2 Terminals' section.

4.2.10. SIP Subscribers

The product allows you to configure the registration of SIP subscribers and make voice calls subsequently within both SIP and PSTN networks using E1 port channels. In other words, it features the functionalities of a small IP exchange. If SIP network subscribers are tied to V5.2 subscriber ports (V5.2 interface and port number are specified) the product will then perform functions of V5.2 remote subscriber node where SIP subscribers will serve as analog ports.

Figure 31. Web-interface – SIP Local Subscribers

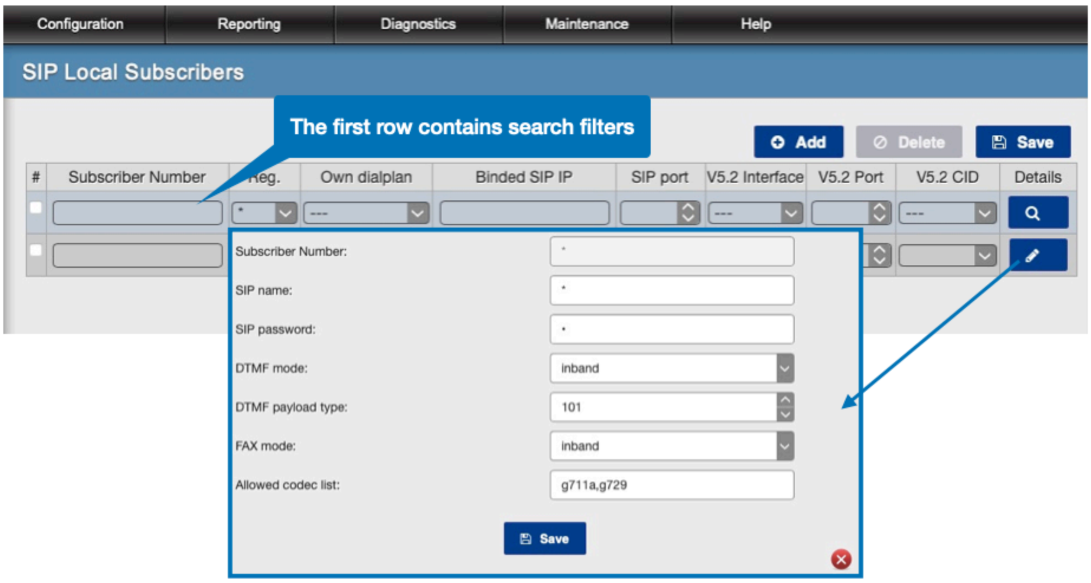


Table 30. SIP Subscriber Settings

Name	Description
Subscriber Number	Assigns the subscriber number that will be used when processing outgoing and incoming calls.
Reg.	Information parameter displaying information about the current registration status. It is active when the subscriber is registered currently with the product.
Own dialplan	Allows to assign specific dialing rules for a specific subscriber.
Binded SIP IP	Assigns the specific IP address that the subscriber can use to register with the product. If 0.0.0.0 value is specified for this parameter then the subscriber can register with the product using any IP address (dynamic registration).
SIP port	Assigns the specific IP port that the subscriber can use to register with the product. If 0 value is specified for this parameter then the subscriber can register with the product using any IP port (dynamic registration).
V5.2 Interface	Ties the subscriber to V5.2 interface.
V5.2 Port Number	Assigns the V5.2 port number of the specified interface used to tie the subscriber.
V5.2 CID	Specifies the dispense mode CID (Caller ID).
Details 	Activates an additional form for editing parameters.
Subscriber Number	Specifies the current IP address that was used to register the subscriber (when dynamic registration is applied).
Connected IP address	Specifies the current IP port that was used to register the subscriber (when dynamic registration is applied).
SIP name	SIP user name used to authenticate the subscriber.
SIP password	SIP user password used to authenticate the subscriber.
DTMF mode	DTMF signal transfer mode selection.
DTMF payload type	This parameter is used in headers when transferring DTMF signals.
FAX mode	Fax message transfer mode.
Allowed codec list	Comma separated list of codecs used to code voice data. The first item in the list is the highest priority.

Data Entry

1. Click Add to add a new SIP subscriber.
2. Edit parameters or select the available value in the drop-down list.
3. Use the button in Details column to activate the additional form of parameters.
4. Move the mouse cursor over the line and click the left button of the mouse to edit parameter values of the selected item. The items to be edited are highlighted in one color. Click Save to save applied changes.
5. Click Delete to delete the current subscriber line.

The table with the list of SIP subscribers contains the filter line to filter the data in the table. The filter line is the first line below the table header.

To use the filter line specify the required value in the required box and press Enter or select the required filter value in the drop-down list.

4.2.11. V5.2 Subscribers

The 'V5.2 Subscribers' subsection allows you to add records to the subscriber table for each connected subscriber remote node and tie it to the assigned interface and specify the number of the subscriber port. In addition, it is possible to create the specific SIP account for each created V5.2 subscriber. In this case the gateway will register all subscribers of V5.2 network remote node on the SIP server specified in the subscriber settings and transmit calls transparently from TDM network to SIP network and backwards.

Figure 32. Web interface - V5.2 Subscribers

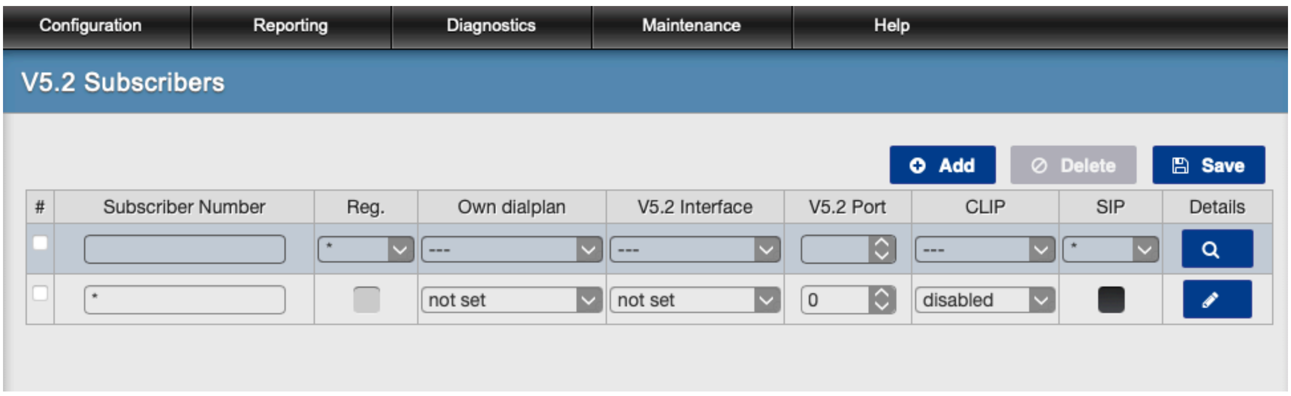


Table 31. V5.2. Subscriber Settings

Name	Description
Subscriber Number	Assigns the subscriber number that will be used when processing outgoing and incoming calls.
Reg.	Information parameter displaying information about the current registration status. It is active when the subscriber is registered currently with the remote server.
Own dialplan	Allows to assign specific dialing rules for a specific subscriber.
V5.2 Interface	Ties the subscriber to V5.2 interface.
V5.2 Port	Ties the subscriber to V5.2 interface port number.
CLIP	Assigns the value of the Caller Line Identification Protocol parameter (provides information about the calling subscriber number).
SIP	Indicates whether it is required to create the specific SIP account for the specified subscriber,
Details 	Activates the additional form for editing parameters.
Primary SIP server	Assigns the IP address of the main SIP server to register the subscriber and make calls in SIP network.
Reserv SIP server	Assigns the IP address of the backup SIP server to register the subscriber and make calls in SIP network (when the main server is unavailable).
SIP name	The name used to authenticate the subscriber in SIP network.
SIP password	The password used to authenticate the subscriber in SIP network.
Register timeout	Subscriber registration update timeout.
DTMF mode	DTMF signal transfer mode selection.
DTMF payload type	This parameter is used in headers when transferring DTMF signals.
FAX mode	Fax message transfer mode.
Allowed codec list	Comma separated list of codecs used to code voice data. **The first item in the list has the highest priority.**

Data Entry

1. Click Add to add a new V5.2 subscriber.
2. Edit parameters or select the available value in the drop-down list.
3. Use the button in the Details column to activate the additional form of parameters.
4. Move the mouse cursor over the line and click the left button of the mouse to edit parameter values of the selected item. The items to be edited are highlighted in color. Click Save to save applied changes.
5. Click Delete to delete the current V5.2 subscriber line.

The table with the list of V5.2 subscribers contains a filter line to filter the data in the table. The filter line is the first line below the table header.

To use the filter line specify the required value in the required box and press Enter or select the required filter value in the drop-down list.

4.2.12. DLU Subscribers

The 'DLU Subscribers' subsection allows you to add records in the subscriber table for each connected DLU subscriber remote node and tie it to the assigned interface and specify the number of the subscriber port.

In addition, it is possible to create the specific SIP account for each created DLU subscriber. In this case the gateway will register all subscribers of the DLU network remote node on the SIP server specified in the subscriber settings and transmit calls transparently from the TDM network to the SIP network and backwards.

Table 32. DLU Subscriber Settings

Name	Description
Subscriber Number	Assigns the subscriber number that will be used when processing outgoing and incoming calls.
Reg.	Information parameter displaying information about the current registration status. It is active when the subscriber is registered currently with the remote server.
Own dialplan	Allows to assign specific dialing rules for a specific subscriber.
DLU Interface	Ties the subscriber to DLU interface.
DLU Port Number	Ties the subscriber to DLU interface port number.
CLIP	Assigns the value of the Caller Line Identification Protocol parameter (provides information about the calling subscriber number).
SIP	Indicates whether it is required to create the specific SIP account for the specified subscriber,
Details 	Activates the additional form for editing parameters.
Primary SIP server	Assigns the IP address of the main SIP server to register the subscriber and make calls in SIP network.
Reserve SIP server	Assigns the IP address of the backup SIP server to register the subscriber and make calls in SIP network (when the main server is unavailable).
SIP name	The name used to authenticate the subscriber in SIP network.
SIP password	The password used to authenticate the subscriber in SIP network.
Register expire timeout	Subscriber registration update timeout.
DTMF mode	DTMF signal transfer mode selection.
DTMF payload type	This parameter is used in headers when transferring DTMF signals.
FAX mode	Fax message transfer mode.
Allowed codec list	Comma separated list of codecs used to code voice data. **The first item in the list has the highest priority.**

Data Entry

1. Click the Add button to add a new DLU subscriber.
2. Edit parameters or select the available value in the drop-down list.
3. Use button in 'Details' line to activate the additional form of parameters.
4. Move the mouse cursor over the line and click the left button of the mouse to edit parameter values of the selected item. The items to be edited are highlighted in a darker color. Click Save button to save the applied changes.
5. Click Delete button to delete the current DLU subscriber line.

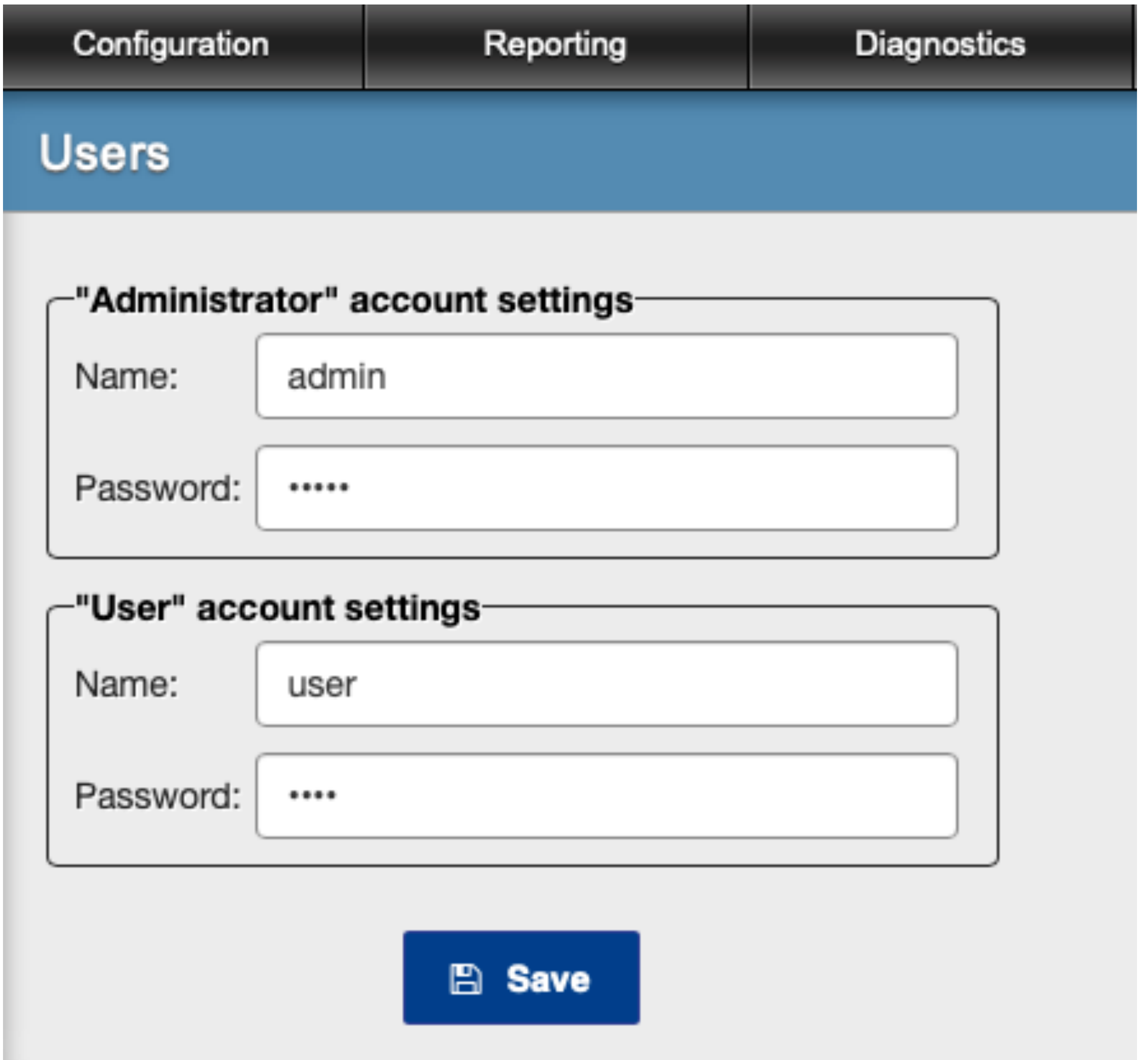
The table with the list of DLU subscribers contains a filter line to filter the data in the table. The filter line is the first line below the table header.

To use the filter line specify the required value in the required box and press 'Enter' or select the required filter value in the drop-down list.

4.2.13. Users

The Users subsection is designed to edit the user name and password that are used for the authorization of Administrator and User accounts. The operator with the Administrator account type can view and edit all available parameters with all forms of the web user interface. The operator with the User account type can only view all the forms.

Figure 33. Web Interface – Users



User Settings

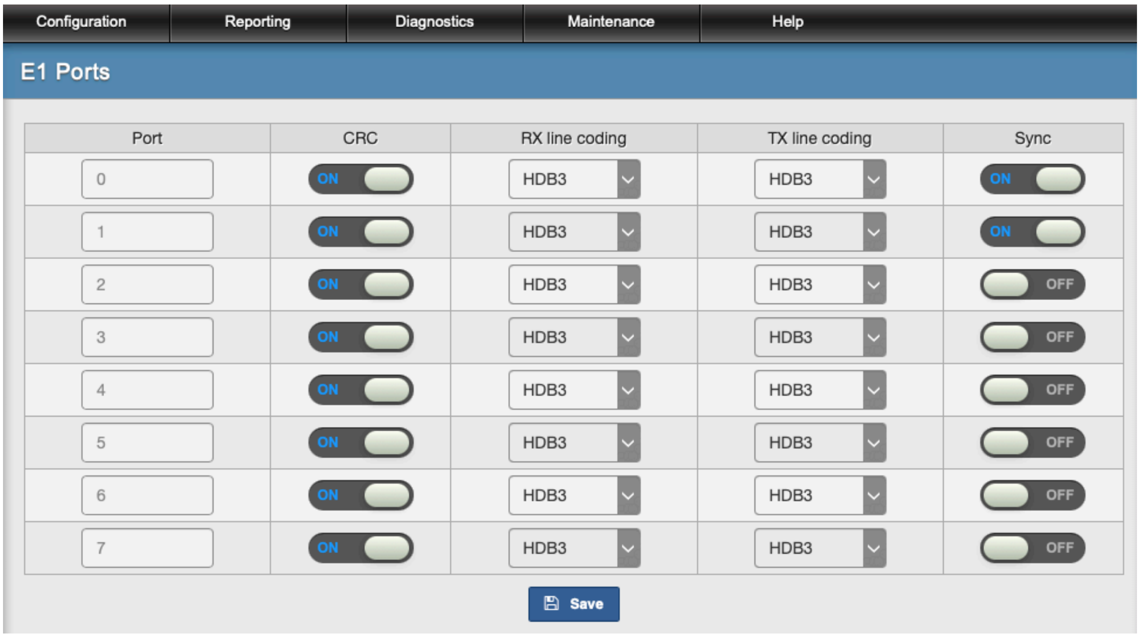
1. Switch to the Users subsection.
2. Move the mouse cursor over the required box and edit the Name and Password values for the Administrator or User accounts. Warning! The length of the text boxes should not exceed 32 characters. Text boxes are case-sensitive.
3. Click Save to save applied changes.

4.2.14. E1 Port

The E1 Ports subsection specifies the physical parameters of the 2Mbit E1 port. These parameters are coordinated with the telephone network operator.

For each port the algorithm for checking the data transfer integrity (CRC - Cyclic Redundancy Check) can be enabled/disabled, the method for coding data reception and transfer (AMI - Alternate Mark Inversion or HDB3 - High-Density Bipolar order 3 encoding) can be assigned and the port(s) to be used as a source for receiving synchronization pulses can be selected.

Figure 34. Web Interface - E1 Ports



Parameter Editing

1. Switch to the E1 Ports subsection.
2. Set up the required value using the switch for CRC, Sync in the line of the required port and select the required value from the drop-down list to code reception/transfer.
3. Click Save to save applied changes.

4.2.15. Synchronization E1 ports

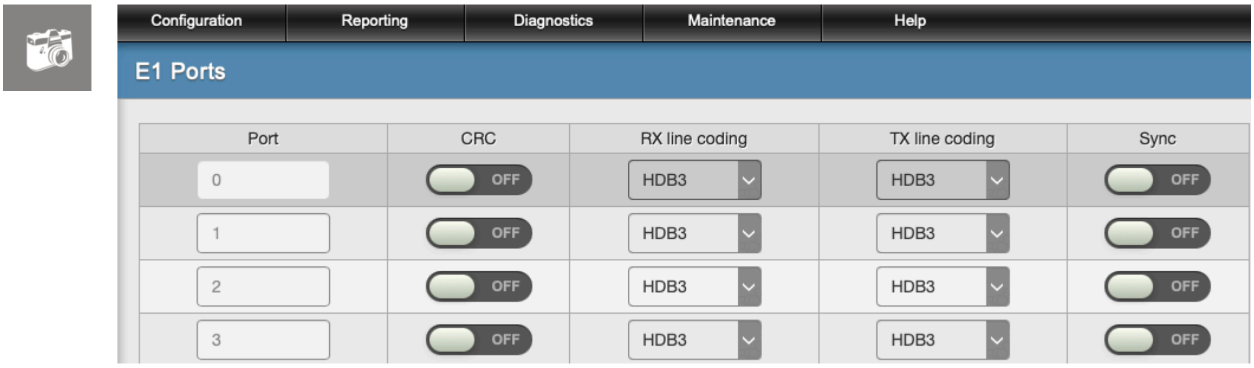
To synchronize E1 ports the device supports the hierarchic source-sink method (Master-Slave). The device features a precise, internal generator that can synchronize all E1 ports. The device can also generate clock pulses from the signal received from any E1 port and synchronize the remaining ports using this signal.

The following operating modes are supported:

1. The stand-alone mode (Master)

All E1 ports are synchronized from a single internal generator. All devices connected to this gateway over E1 should be set to synchronization mode over the E1 port received from the gateway (Slave). The gateway is set to this mode by disabling synchronization for all ports. To do this set all switches to the OFF position (Figure 35).

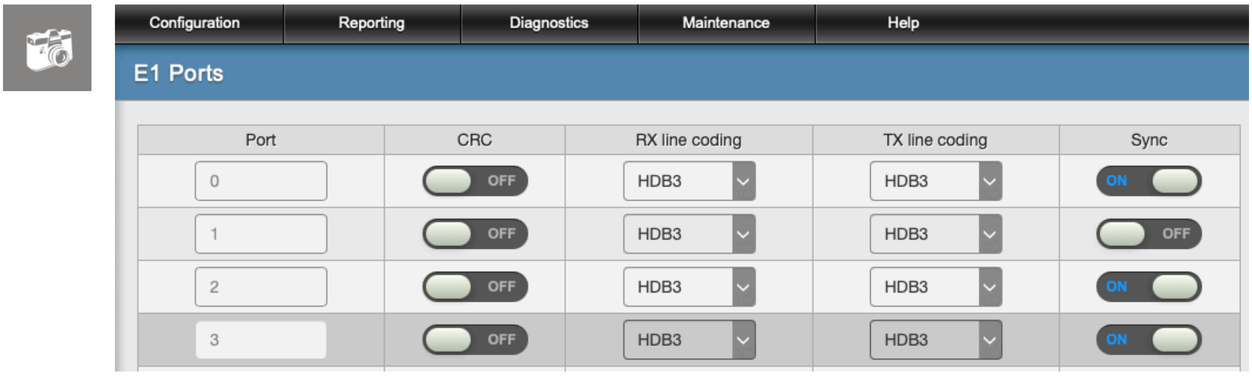
Figure 35. Web Interface - E1 Ports - Master



2. Synchronization mode over E1 port (Slave)

To enable synchronization over the selected E1 port set the switch to ON position. You can specify several ports as synchronization sources. If all E1 ports selected for synchronization have the signal then the E1 port with the least number will be the synchronization source. When the signal within the given E1 port is lost then the selected E1 port with the next sequential number will be used for synchronization. Because of this, when the network is built it is recommended to connect the higher-level node that was selected as the main node for synchronization to E1 gateway ports with a lower number. When the signal is lost in all the ports selected for synchronization the device will switch to the stand-alone synchronization mode. Below is an example selecting ports 0, 2, 3 for synchronization (Figure 36).

Figure 36. Web Interface - E1 Ports - Slave

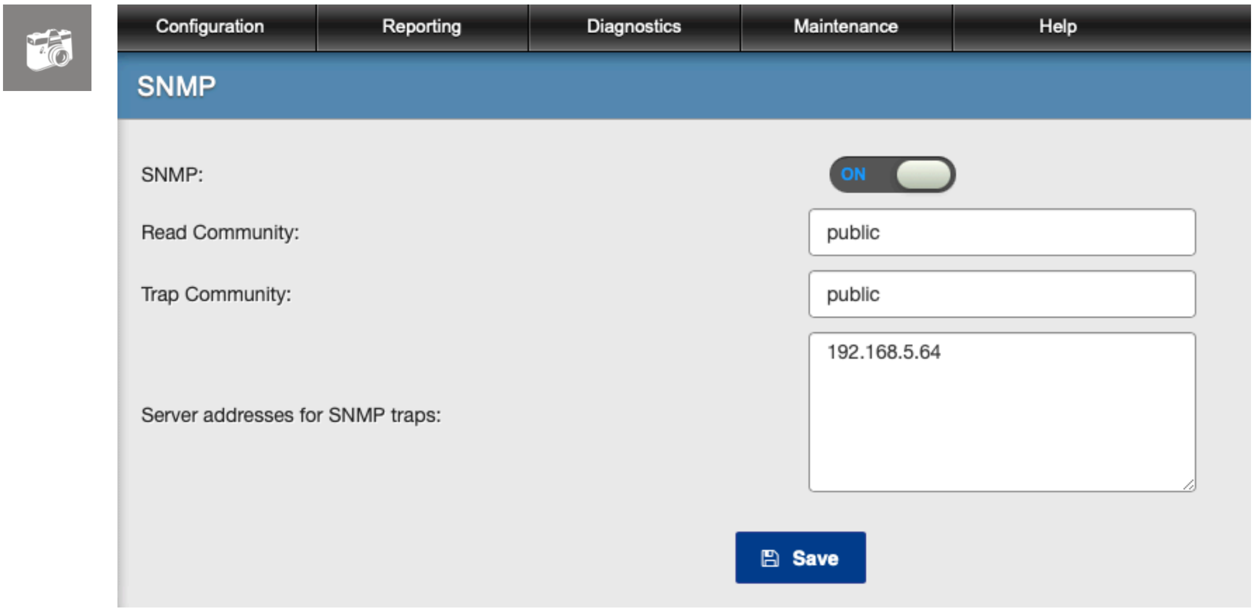


When the signal is present in all E1 ports, then port 0 will be the synchronization source. When the signal is lost in port 0 then port 2 will become the synchronization source etc.

4.2.16. SNMP

The SIP/E1 Gateway provides an embedded SNMP v1/v2c agent. The SNMP agent sends alarm messages to the controlling SNMP manager in real time. The SNMP agent also supports monitoring of equipment status upon request from the SNMP manager.

Figure 37. Web Interface - SNMP



Parameter Editing

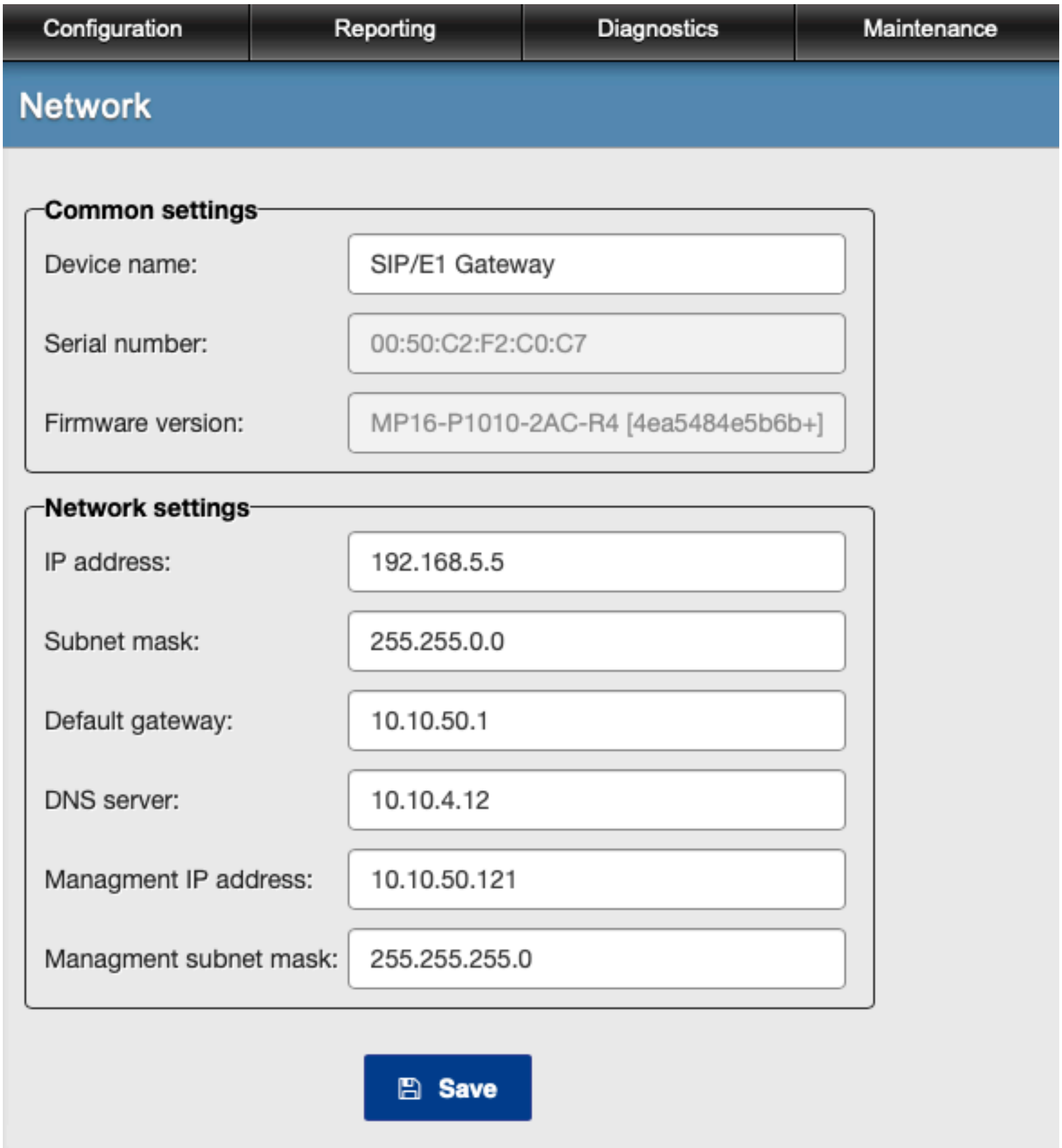
1. Switch to the SNMP subsection.
2. Enable SNMP support.
3. Move the mouse cursor over the required box and edit parameter values: Read Community, Trap Community, Server Addresses for SNMP Traps (each server should be entered in a new line).
4. Click Save to save applied changes.

4.2.17. Network configuration

The Network subsection is designed to configure the parameters of the product 's operation with Ethernet. The following parameters are available for editing:

- Device Name – network name of the product
- IP Address – network address of the product
- Subnet Mask – subnet mask of the product
- Default Gateway – network gateway address for the product
- DNS Server – main DNS server
- Management IP Address – product network address for managing and monitoring
- Management Subnet Mask – product subnet mask for managing and monitoring

Figure 38. Web Interface – Network



Parameter Editing

1. Switch to the Network subsection.
2. Move the mouse cursor over the required box and edit its parameter values. Serial Number and Firmware Version boxes are read-only.
3. Click Save to save applied changes.

Warning! The LAN-1 connector is required to connect to the equipment and perform initial configuration. In this case, the Management IP Address and Management Subnet Mask parameters shall not be specified and all control traffic and Telnet, Web, and SIP/RTP traffic goes through LAN-1.

To divide traffic between LAN-0 and LAN-1, the Management IP Address and Management Subnet Mask parameters shall be specified.

Important! IP Address and Management IP Address values should be located in different subnetworks. In this case control traffic and Telnet and Web traffic is directed through LAN – 0. Then SIP/RTP traffic is directed through LAN-1.

Attention! After setting up the network parameters, the equipment must be reloaded.

4.2.18. Firewall

A Firewall is used to prevent unauthorized access by blocking packets that do not comply with specified rules. The device uses the standard Firewall Linux OS.

Edit the file **/mnt/firewall.sh** to turn on Firewall and set up the rules.

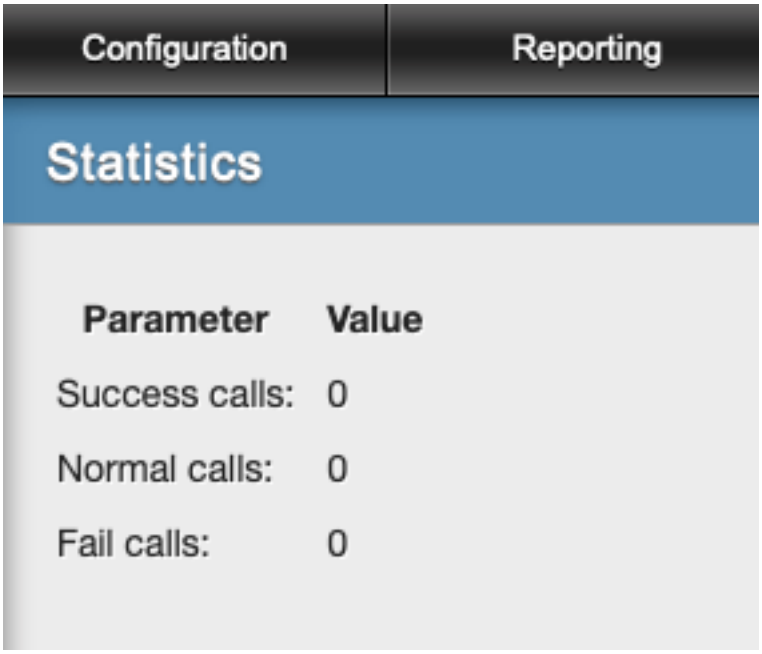
In this file, you can find a typical example of a Firewall setting.

4.3. Reporting

4.3.1. Statistics

The Statistics subsection is designed to obtain statistical information in terms of completed calls.

Figure 39. Web Interface - Reporting – Statistics

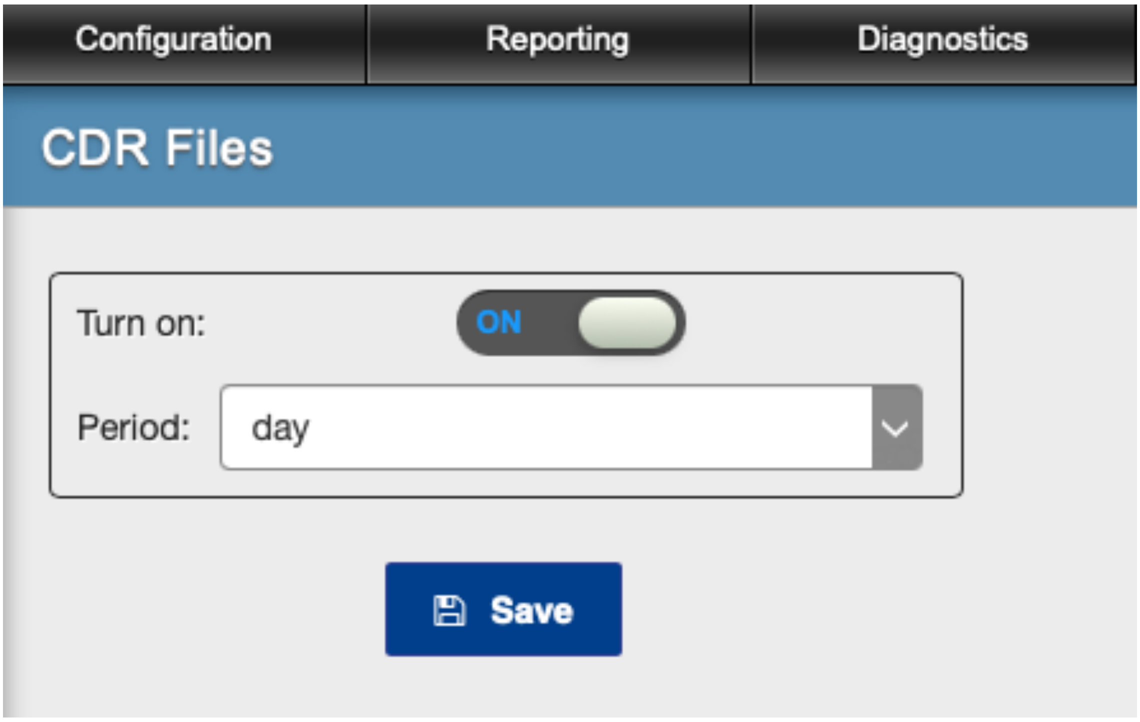


4.3.2. CDR Files

The CDR Files subsection (Call Detail Record) is used to configure options for saving detailed call records. Inclusion of this option allows keeping data about calls in the csv-file in the memory of the device.

CDR files are intended for assessment of use of traffic, service needs, etc.

Figure 40. Web Interface - Reporting – CDR Files



Parameter Editing

1. Switch to the CDR Files subsection
2. Turn the switch to the "On / Off"
3. Click Save to save applied changes
4. Choose the period of formation of CDR records
5. Click Save to save applied changes

Access to CDR files

1. Enter in an address line of the browser:
   http:// ip_device_address/cdr/
2. In the list of saved files choose the file for viewing.

Figure 41. CDR Files - the list of the saved files

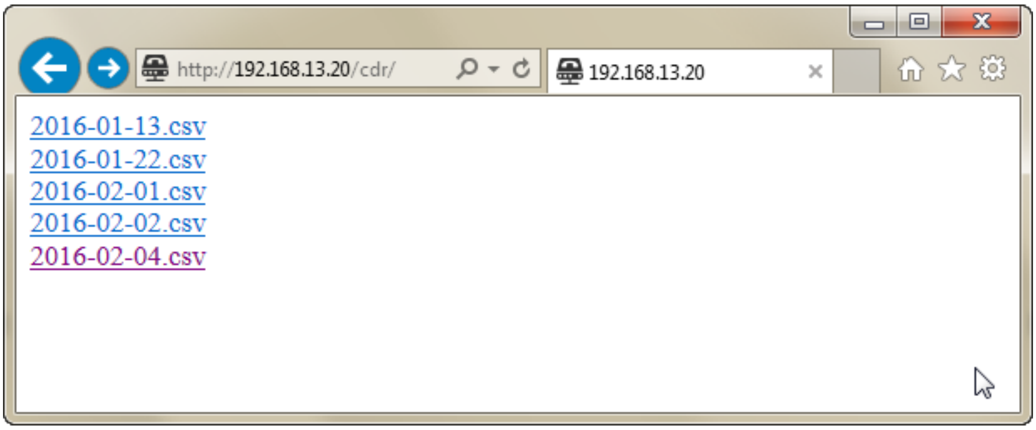
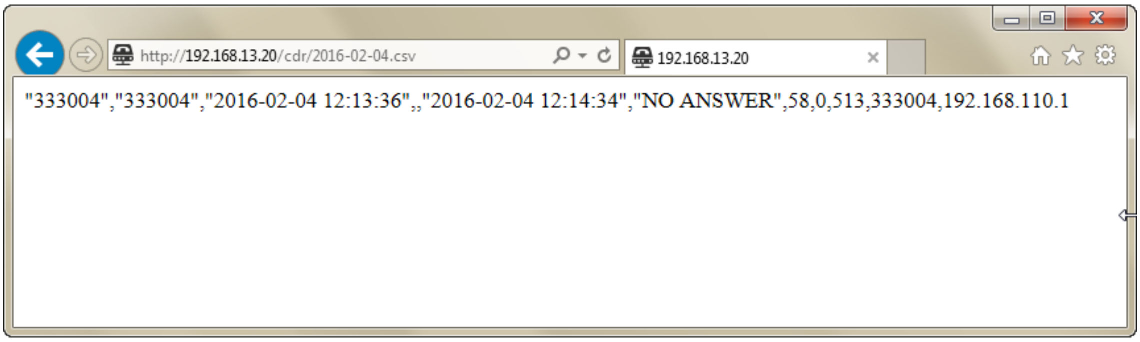


Figure 42. CDR Files - contents of the chosen file



4.4. Diagnostics

The Diagnostics subsection is designed to perform real-time monitoring of the product status, TDM and SIP channel status and E1 port status. It contains the following subitems:

System Information

E1port status

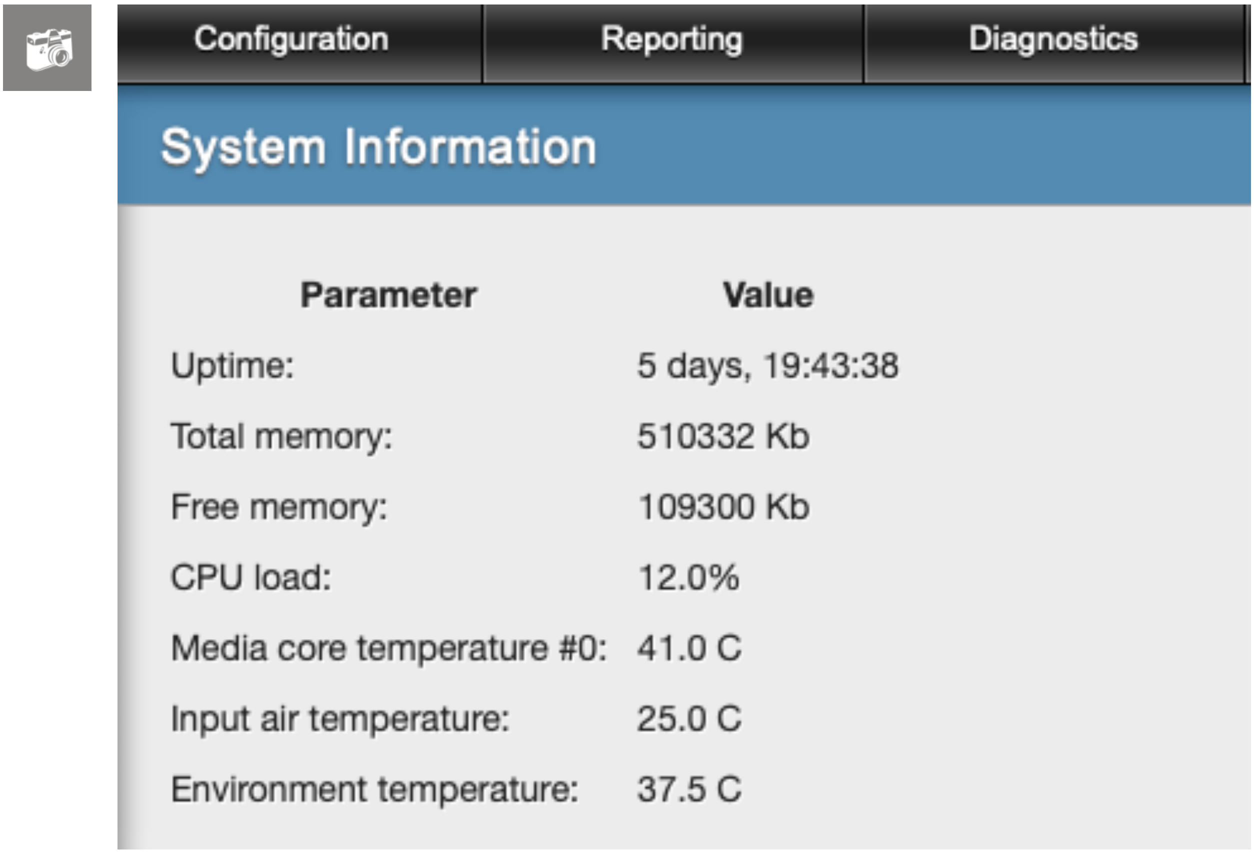
TDM Channel status

SIP Channel status

4.4.1. System Information

The System Information subitem displays the system status in real time including: Operation Time, Free Memory, CPU Load, Processor Temperature.

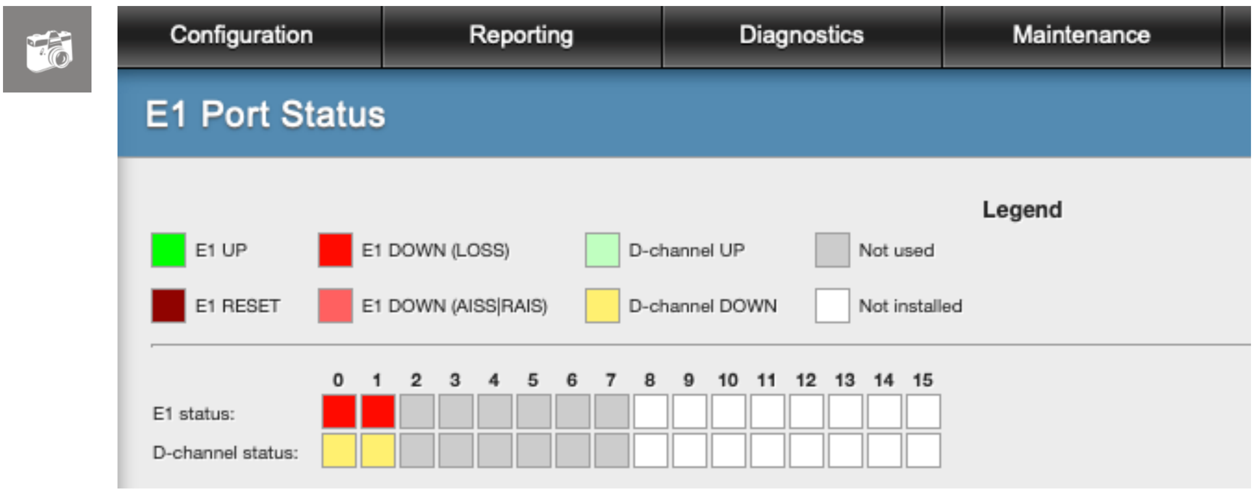
Figure 43. Web Interface - Diagnostics - System Information



4.4.2. E1 port Status

The E1 Port Status subitem displays E1 port map and current status of these ports.

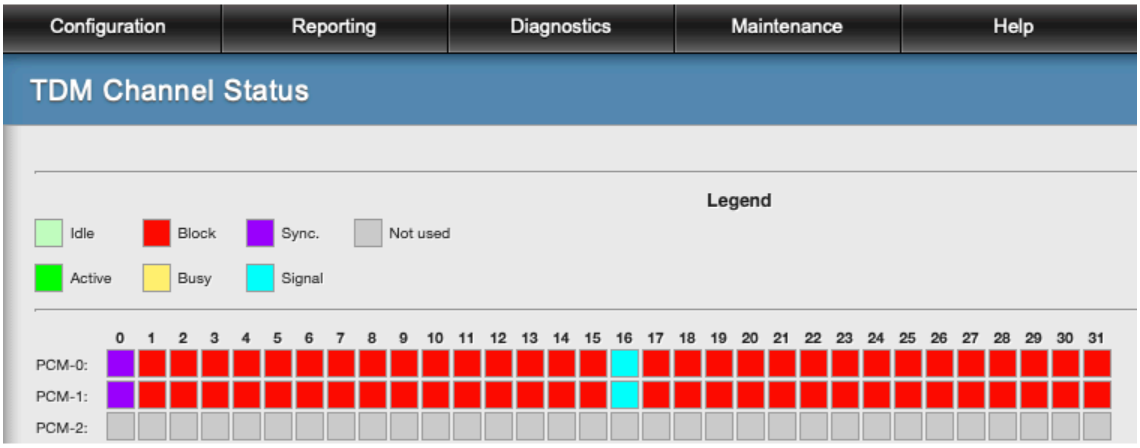
Figure 44. Web Interface - Diagnostics - E1 Port Map



4.4.3. TDM Channel Status

The TDM Channel Status subitem displays the TDM channel map and current status of these channels.

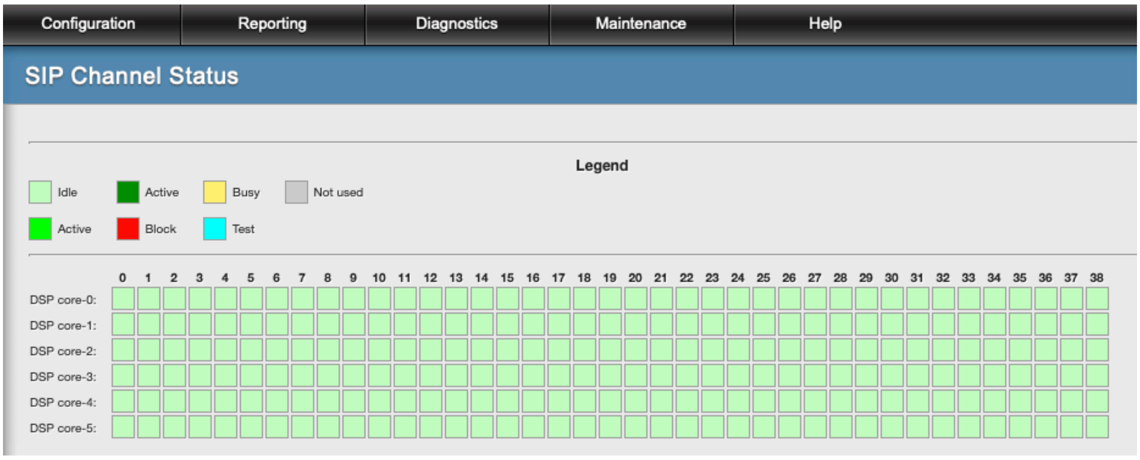
Figure 45. Web Interface - Diagnostics - TDM Channel Map



4.4.4. SIP Channel Status

The SIP Channel Status subitem displays the SIP channel map and current status of these channels.

Figure 46. Web Interface - Diagnostics - SIP Channel Map



4.4.5. Measurement of parameters of a subscriber line of the DLU block

The software-based measuring unit is used to measure parameters of the subscriber line.

The unit for measuring parameters of the subscriber line is fully adapted to standard DLU EWSD commands thus allowing the unit to emulate (replace) line test PC (LT-PC) operation. This unit is used for measuring and testing analog subscriber lines.

The unit for measuring parameters of the subscriber line simulates the interface that is equivalent to the subscriber line.

The commands for testing and measuring are entered through Telnet interface such as Hyper Terminal (integrated into Windows OS) or Tera Term or PuTTY (freeware).

To access the unit for measuring parameters of the subscriber line the following command is used:

telnet [ip] 7001

where [ip] is SIP E1 Gateway address and 7001 is the connection port.

To perform measurements and tests in the DLU (local connection) the following Telnet protocol command is used: **dlu test**

Command format:

dlu test [phone number]

where **dlu test** is a command that launches measurements and tests;

[phone number] is a phone line number.

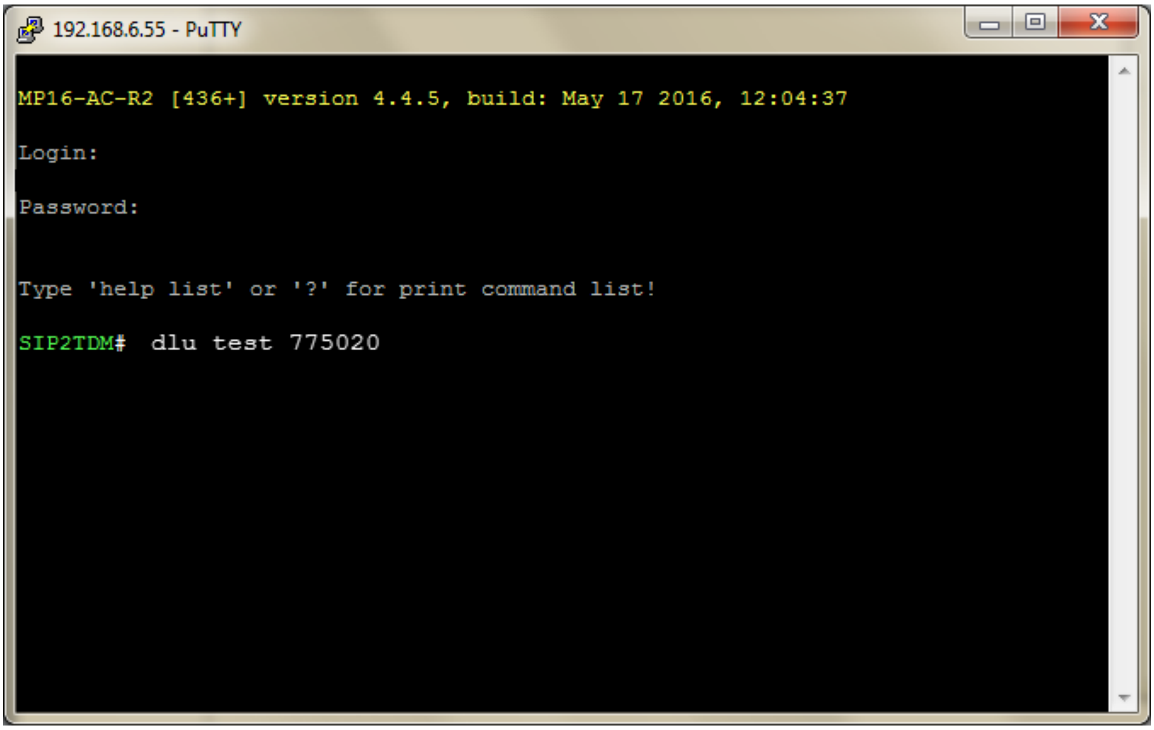
Example: **dlu test 775020**

The following measurements are possible:

- insulation resistance measurement (for A/G, B/G);
- loop resistance measurement (for A/B);
- capacity measurement (for A/G, B/G, A/B);
- stray voltage measurement (for DC, AC, A/G, B/G, A/B).

The following pictures demonstrate example test launches and their results.

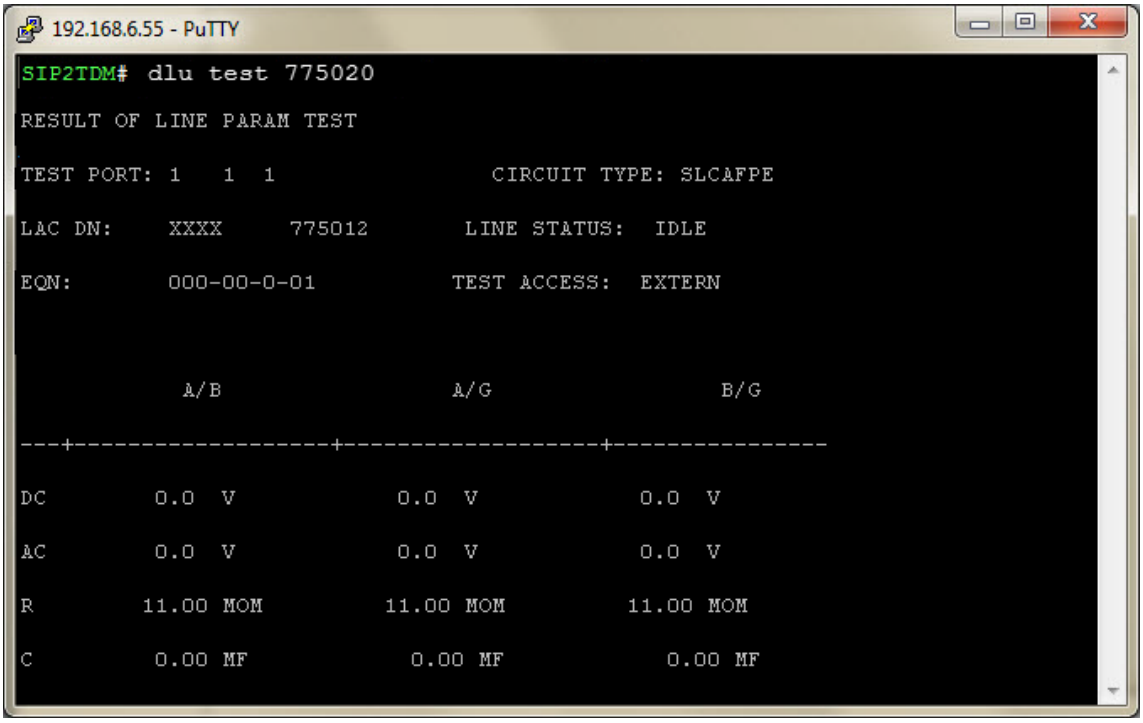
Figure 47. The command to start measurements and tests



The results of measurements and tests are summarized in the summary table (Figure 48) where:

- DC – DC voltages;
- AC – AC voltages;
- R – resistances;
- C – capacities;
- A/B – values between A and B wires;
- A/G – values between A wire and ground;
- B/G – values between B wire and ground.

Figure 48. Result of measurements and tests



4.5. Maintenance

The Maintenance subsection is designed to back up configuration files (Configuration Files) and firmware files (Device Firmware Files) on the PC hard drive, to upload configuration files or firmware file to the device from the PC hard drive and to reload the device after changes are applied to its parameters. The subsection contains the following subitems:

Backup Files

Upload Files

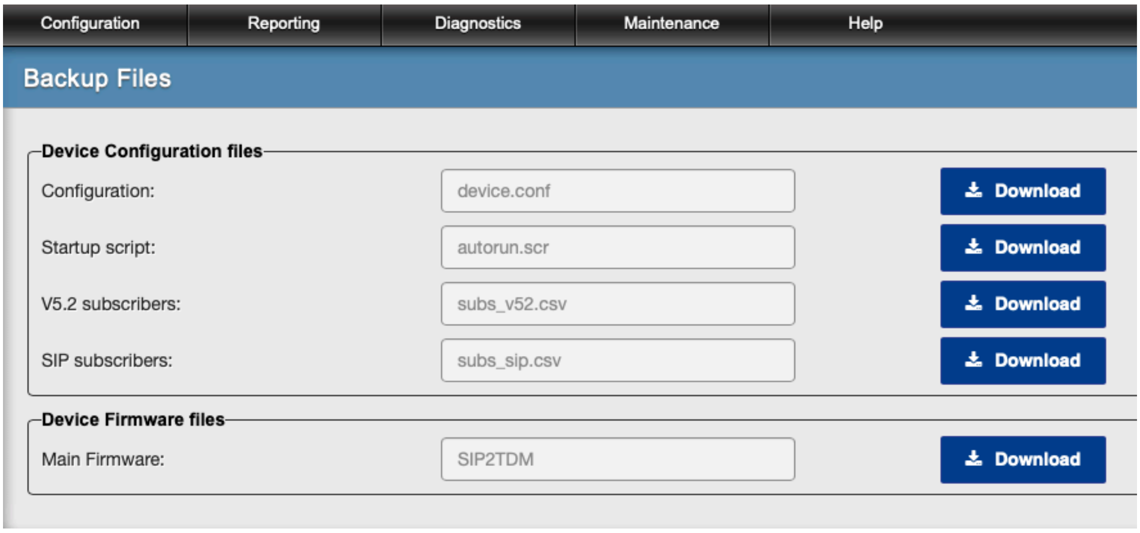
System Clock

Reboot Device

4.5.1. Backup Files

It is recommended to back up the product configuration files and product firmware file before you start changing any product parameters. The current Backup Files subitem is used to create backups.

Figure 49. Web Interface - Maintenance - Backup Files



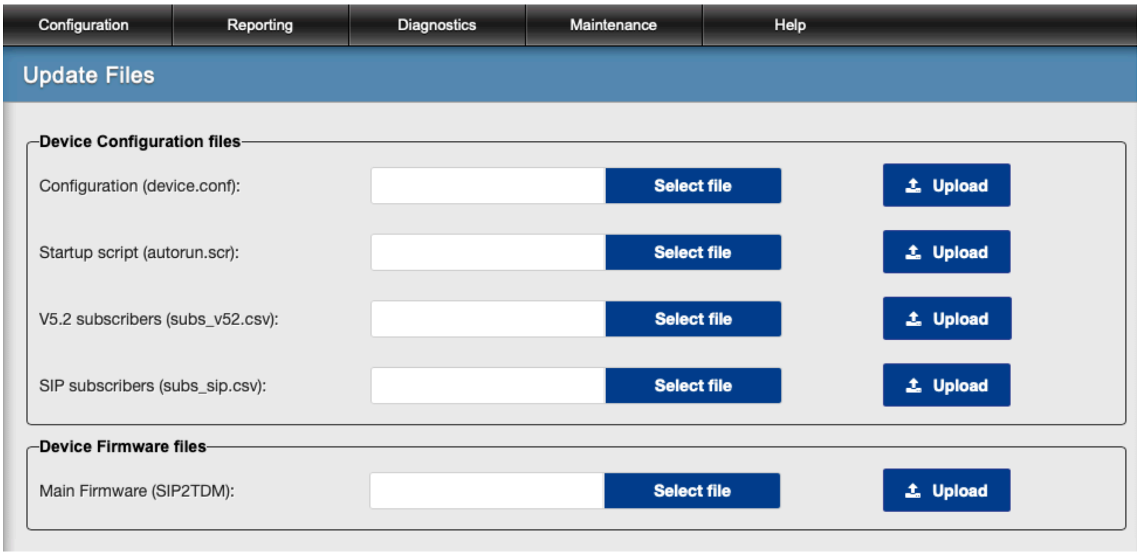
Backup Creation

1. Switch to the Backup Files subitem.
2. Save configuration files and device firmware file on the PC hard drive using the Download buttons.

4.5.2. Upload Files

The Upload Files subitem is used to upload the configuration files and firmware file from the PC hard drive to the device.

Figure 50. Web Interface - Maintenance - Update Files



Uploading a File

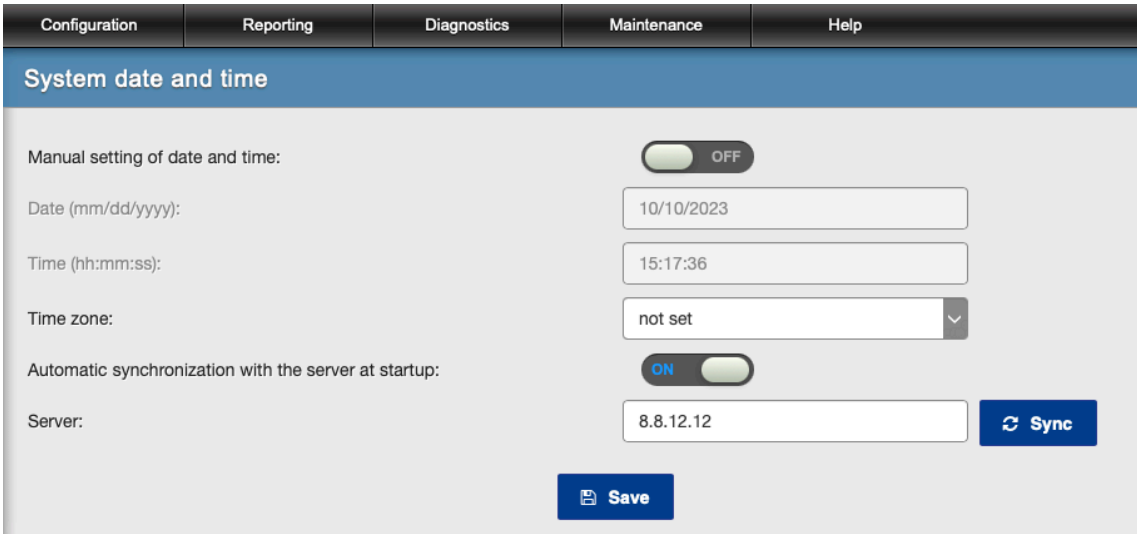
1. Switch to the Update Files subitem.
2. Select the file to be uploaded using the Select File button.
3. Click the Upload button to upload files to the device.
4. To start the device with updated configuration files or software switch to the Reboot Device subsection and click Reboot.

Be careful when selecting files to be uploaded to the device! Selecting the wrong file for uploading can damage the device!

4.5.3. System Clock

The System Clock subsection displays the current date and time of the product and allows you to synchronize date and time parameters manually or automatically.

Figure 51. Web Interface – System Date and Time



Date and Time Setup

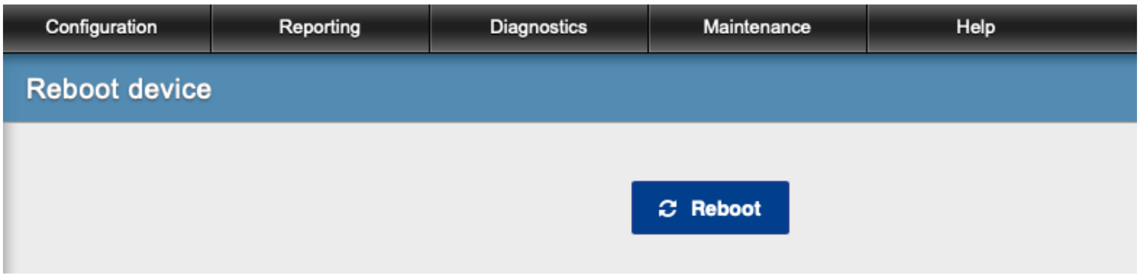
1. Switch to the System Clock subsection.
2. Select the required date and time synchronization option: Manual Setting of Date and Time, or Automatic Synchronization at Startup.
3. Select the time zone and specify the synchronization server.
4. Click Save to save applied changes.
5. Click Sync for quick synchronization of date and time with the specified server.

4.5.4. Reboot Device

After changes are saved during the device configuration process it is required to reboot the device so that all changes take effect.

The Reboot Device subitem is used to reboot the device.

Figure 52. Web Interface - Maintenance - Reboot Device



Rebooting Device

1. Switch to the Reboot Device subitem.
2. Click Reboot.

4.6. Help

The Help menu item is designed to get brief help on configuring the product using the web interface.

5. Contacting Patton for Assistance

5.1 Contact Information

Patton, LLC offers a wide array of free technical services. If you have questions about any of our other products we recommend you begin your search for answers by using our technical knowledge base. Here, we have gathered together many of the more commonly asked questions and compiled them into a searchable database to help you quickly solve your problems.

5.1.1 Contacting Patton Technical Services for Free Support

Table 33. Contacting Patton Technical Services

REGION	North America	Western Europe	Central & Eastern Europe
Location	Maryland, USA	Bern, Switzerland	Budapest, Hungary
Time Zone	EST/EDT UTC/GMT - 4/5 hours	CET/CEDT UTC/GMT + 1/2 hours	CET/CEDT UTC/GMT + 1/2 hours
Business Hours	Monday-Friday 8:00am to 5:00pm	Monday-Friday 09:00 to 12:00 13:30 to 17:30	Monday-Friday 8:30 to 17:00
Email	support@patton.com	support@patton.com	support@patton.com
Phone	+ 1 301 975 1007	+41 31 985 25 55	+36 439 3835
Fax	+1 301 869 9293	+41 31 985 2526	

5.2 Warranty Service and Returned Merchandise Authorizations (RMAs)

Patton, LLC is an ISO-9001 certified manufacturer and our products are carefully tested before shipment. All of our products are backed by a comprehensive warranty program.

Note If you purchased your equipment from a Patton, LLC reseller, ask your reseller how you should proceed with warranty service. It is often more convenient for you to work with your local reseller to obtain a replacement. Patton services our products no matter how you acquired them.

5.2.1 Warranty coverage

Our products are under warranty to be free from defects, and we will, at our option, repair or replace the product should it fail within one year from the first date of shipment. Our warranty is limited to defects in workmanship or materials, and does not cover customer damage, lightning or power surge damage, abuse, or unauthorized modification.

5.2.2 Out-of-warranty service

Patton services what we sell, no matter how you acquired it, including malfunctioning products that are no longer under warranty. Our products have a flat fee for repairs. Units damaged by lightning or other catastrophes may require replacement.

5.2.3 Returns for credit

Customer satisfaction is important to us, therefore any product may be returned with authorization within 30 days from the shipment date for a full credit of the purchase price. If you have ordered the wrong equipment or you are dissatisfied in any way, please contact us to request an RMA number to accept your return. Patton is not responsible for equipment returned without a Return Authorization.

5.2.4 Return for credit policy

- Less than 30 days: No Charge. Your credit will be issued upon receipt and inspection of the equipment.
- 30 to 60 days: We will add a 20% restocking charge (crediting your account with 80% of the purchase price).
- Over 60 days: Products will be accepted for repairs only.

5.3 RMA numbers

RMA numbers are required for all product returns. You can obtain an RMA by doing one of the following:

- Completing a request on the RMA Request page in the *Support* section at **www.patton.com**
- By calling **+1 (301) 975-1007** and speaking to a Technical Support Engineer
- By sending an e-mail to **returns@patton.com**

All returned units must have the RMA number clearly visible on the outside of the shipping container. Please use the original packing material that the device came in or pack the unit securely to avoid damage during shipping.

5.3.1 Shipping instructions

The RMA number should be clearly visible on the address label. Our shipping address is as follows:

Patton, LLC
RMA#: xxxx
7622 Rickenbacker Dr.
Gaithersburg, MD 20879-4773 USA

Patton will ship the equipment back to you in the same manner you ship it to us. Patton will pay the return shipping costs.

6. End User License Agreement

By opening this package, operating the Designated Equipment or downloading the Program(s) electronically, the End User agrees to the following conditions:

6.1 Definitions

A) "Effective Date" shall mean the earliest date of purchase or download of a product containing the Patton LLC Program(s) or the Program(s) themselves.

B) "Program(s)" shall mean all software, software documentation, source code, object code, or executable code.

C) "End User" shall mean the person or organization which has valid title to the Designated Equipment.

D) "Designated Equipment" shall mean the hardware on which the Program(s) have been designed and provided to operate by the End User.

6.2 Title

Title to the Program(s), all copies of the Program(s), all patent rights, copyrights, trade secrets and proprietary information in the Program(s), worldwide, remains with Patton LLC or its licensors.

Patton does not convey any intellectual property title or rights in the Licensed Products to Licensee. All Licensed Products furnished by Patton, and all copies thereof, and compilations, programmatic extension, and all Patches, Updates, Upgrades and Platform Releases, are and shall remain the property of Patton or Patton's licensors, as applicable. Further, the Licensed Products provided under this Agreement are not custom software but are standard commercial software. Except for the license use rights otherwise expressly provided in this Agreement, no right, title or interest in Patton Licensed Products is granted hereunder. Licensee shall not use any proprietary information of Patton to create any computer software program or user documentation, which is substantially similar to the Licensed Products.

6.3 Term

The term of this Agreement is from the Effective Date until title of the Designated Equipment is transferred by End User or unless the license is terminated earlier as defined in section "6.6. Termination".

6.4 Grant of License

A) During the term of this Agreement, Patton LLC grants a personal, non-transferable, non-assignable and non-exclusive license to the End User to use the Program(s) only with the Designated Equipment at a site owned or leased by the End User.

B) The End User may copy licensed Program(s) as necessary for backup purposes only for use with the Designated Equipment that was first purchased or used or its temporary or permanent replacement.

C) The End User is prohibited from disassembling; decompiling, reverse-engineering or otherwise attempting to discover or disclose the Program(s), source code, methods or concepts embodied in the Program(s) or having the same done by another party.

D) Should End User transfer title of the Designated Equipment to a third party after entering into this license agreement, End User is obligated to inform the third party in writing that a separate End User License Agreement from Patton LLC is required to operate the Designated Equipment.

6.5 Warranty

The Program(s) are provided "as is" without warranty of any kind. Patton LLC and its licensors disclaim all warranties, either express or implied, including but not limited to the implied warranties of merchantability, fitness for a particular purpose or non-infringement. In no event shall Patton LLC or its licensors be liable for any damages whatsoever (including, without limitation, damages for loss of business profits, business interruption, loss of business information, or other pecuniary loss) arising out of the use of or inability to use the Program(s), even if Patton LLC has been advised of the possibility of such damages. Because some states do not allow the exclusion or limitation of liability for consequential or incidental damages, the above limitation may not apply to you.

If the Program(s) are acquired by or on behalf of a unit or agency of the United States Government, the Government agrees that such Program(s) are "commercial computer software" or "computer software documentation" and that, absent a written agreement to the contrary, the Government's rights with respect to such Program(s) are limited by the terms of this Agreement, pursuant to Federal Acquisition Regulations 12.212(a) and/or DEARS 227.7202-1(a) and/or sub-paragraphs (a) through (d) of the "Commercial Computer Software—Restricted Rights" clause at 48 C.F.R. 52.227-19 of the Federal Acquisition Regulations as applicable.

6.6 Termination

A) The End User may terminate this agreement by returning the Designated Equipment and destroying all copies of the licensed Program(s).

B) Patton LLC may terminate this Agreement should End User violate any of the provisions of section "6.4. Grant of License" on page 92.

C) Upon termination for **A** or **B** above or the end of the Term, End User is required to destroy all copies of the licensed Program(s).

6.7 Notices

Patton devices may log, collect and report data related to installed software, licenses, feature utilization, product performance, device management, service quality and other parameters which is used for quality control, product improvement, license management, service level management and technical support. Collected data may be reported to Patton or a service provider delivering its services connected to the device.

Patton may use this information for other business purposes, such as to alerting you to updated products or services, securing access to software updates, and assisting in order processing.

Any and all information collected by Patton or its assigns will be kept strictly confidential and will not be sold, rented, loaned, or otherwise disclosed to any third party except as required by law.

6.8 Other Licenses

The Program may be subject to licenses extended by third parties. Accordingly, Patton LLC licenses the Programs subject to the terms and conditions dictated by third parties. Third party software identified to the Programs includes the LGPL (Lesser General Public License) open source license distributed to you pursuant to the LGPL license terms (**http://www.gnu.org/licenses/lgpl.html**).

6.9 Unenforceable Provisions

If any part of these terms and conditions are found to be invalid or unenforceable under applicable law, such part will be ineffective to the extent of such invalid or unenforceable part only, without in any way affecting the remaining parts of these terms and conditions.

6.10 Governing Law

The rights and obligations of the parties pursuant to these terms and conditions are governed by, and shall be construed in accordance with, the laws of the State of Maryland, USA.

User may be subject to other local, provincial or state and national laws. User hereby irrevocably submits to the exclusive jurisdiction of the courts of the State of Maryland, USA for any dispute arising under or relating to this agreement and waives user's right to institute legal proceedings in any other jurisdiction. Patton shall be entitled to institute legal proceedings in connection with any matter arising under this agreement in any jurisdiction where User resides, does business, or has assets.

6.11 Waiver

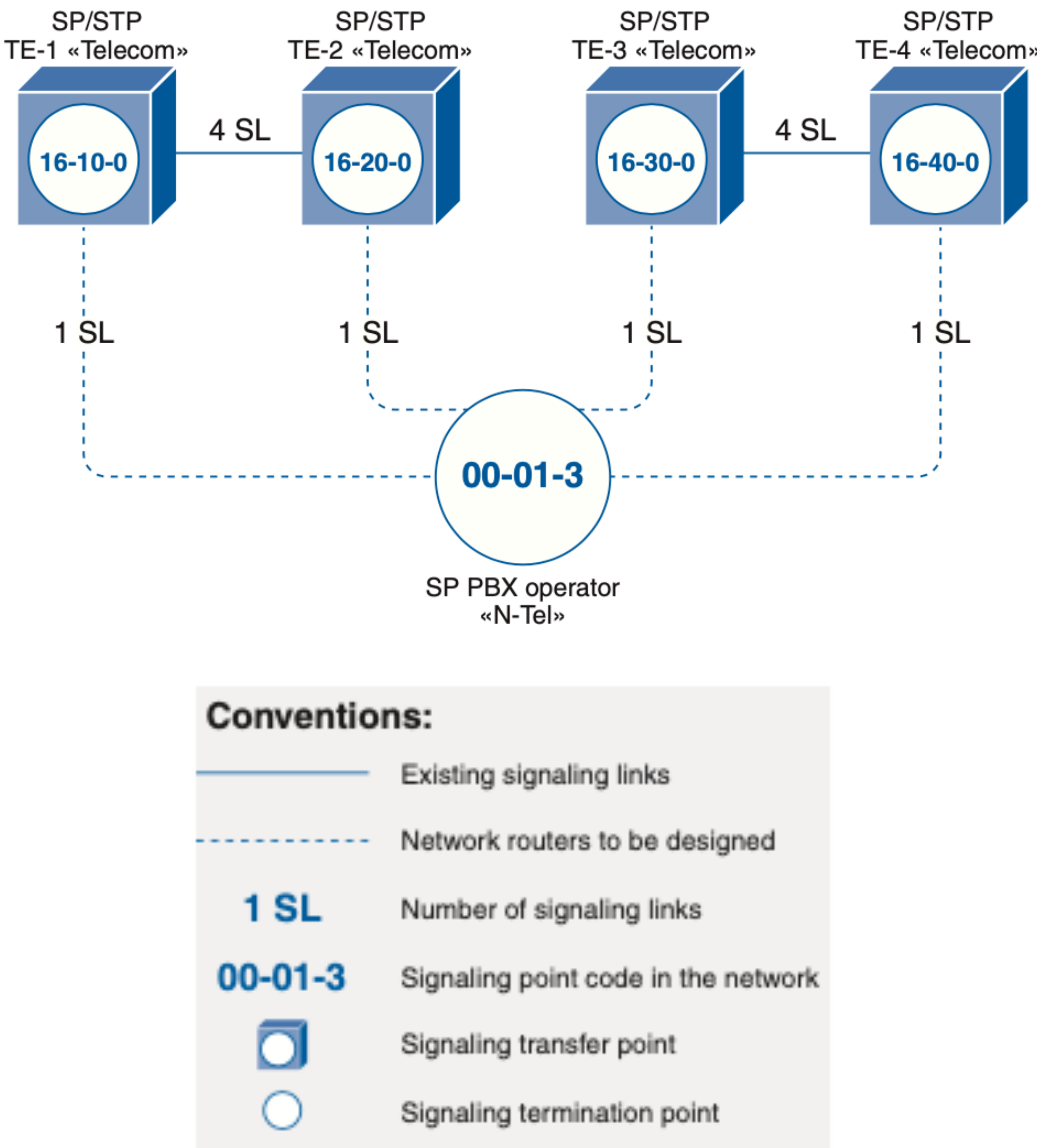
No waiver of any of the provisions of these terms and conditions will be deemed to constitute a waiver of any other provision nor shall such a waiver constitute a continuing waiver unless otherwise expressly provided in writing duly executed by the party to be bound thereby. Any other terms and conditions of sale, to the extent not inconsistent herein, regarding a Patton device, program, license or service remain in full force and effect.

7. Examples of setup and configuration

7.1. SIP Gateway in SS7. Connecting VoIP Network Subscribers to PSTN Number Capacity

Objective: Connect a VoIP network with the number capacity of 1000 subscribers to PSTN over 4E1 ports with SS7 signaling. Each E1 path is connected to different operator's switch nodes (TC1 - TC4) but the outgoing traffic from the customer's side does not need to be separated based on the number of the called subscriber, i.e. any PSTN subscriber can be accessed over any E1 path.

Figure 53. Diagram for Connecting Customer's Gateway to PSTN Operator Exchanges



Operator's prerequisites: 4E1, SS7 signaling, HDB3 coding method, CRC parameter is disabled, the main synchronization port is E1 path connected to the operator's TC-1 node, the backup synchronization port is E1 of TC-3.

Additional Data

Table 34. Allocation of Signaling Channels and Links

Direction	Signaling link code	Channel Interval (CI) for signaling link	Channel Interval (CI) for information channels	CIC
00-01-3⇔ 16-10-0	0	CI 1	CI 2…31	2…26 circuit 27…31 preordered circuit, long-distance circuit
00-01-3⇔ 16-20-0	0	CI 1	CI 2…31	2…26 circuit 27…31 ordered connection line, long-distance connection line
00-01-3⇔ 16-30-0	0	CI 1	CI 2…31	2…26 circuit 27…31 preordered circuit, long-distance circuit
00-01-3⇔ 16-40-0	0	CI 1	CI 2…31	2…26 circuit 27…31 preordered circuit, long-distance circuit

Table 35. Signaling Message Routing

Direction	OPC	DPC	Route 1	Route 2
TE «N-Tel» → TE-1 «Telecom»	00-01-3	16-10-0	Direct	16-20-0
TE-1 «Telecom» → TE «N-Tel»	16-10-0	00-01-3	Direct	16-20-0
TE «N-Tel» → TE-2 «Telecom»	00-01-3	16-20-0	Direct	16-10-0
TE-2 «Telecom» → TE «N-Tel»	16-20-0	00-01-3	Direct	16-10-0
TE «N-Tel» → TE-3 «Telecom»	00-01-3	16-30-0	Direct	16-40-0
TE-3 «Telecom» → TE «N-Tel»	16-30-0	00-01-3	Direct	16-40-0
TE «N-Tel» → TE-4 «Telecom»	00-01-3	16-40-0	Direct	16-30-0
TE-4 «Telecom» → TE «N-Tel»	16-40-0	00-01-3	Direct	16-30-0

According to the connection diagram (Figure 53) and routing table the signaling links are created between TC-1 and TC-2 as well as between TC-3 and TC-4. These signaling links make it possible for the signaling traffic to reach the assigned node when the main link is down. For example, when the signaling link between N-Tel and TE-1 is down, the signaling traffic can go to TE-1 using the transit route through TE-2 and when the link is down the transit route through TE-1 can be used to reach TE-2, respectively. The reservation of signaling routes within the TE-3 and TE-4 pair is implemented in the same way.

As the channel allocation table suggests the operator is provided with the option of dividing the interface channels into local and long-distance/international, i.e. all calls from N-Tel side depending on the phone number of the called subscriber should be divided into local and long-distance. The operator is also provided with some rules relating to the formats of the sent numbers of the calling and called subscribers:

- The number of the calling subscriber should always start with the "44" prefix, its "Number Type" parameter should be "National", the "Numbering Plan" parameter should be "ISDN Telephony" and it should also have the call category of “1".
- When the calls are made to the numbers that start with the "044" prefix, such calls should be classified as local, the "044" prefix should be removed from the number, the "Number Type" parameter should be set to "Subscriber" and the "Numbering Plan" parameter should be set to "ISDN Telephony”.
- When the calls are made to the numbers that start with the "00" prefix, such calls should be classified as long-distance/international, the "00" prefix should be removed from the number, the "Number Type" parameter should be set to "International" and the "Numbering Plan" parameter should be set to "ISDN Telephony”.

The customer's side provides a SIP TRUNK account that is used for calls in the SIP network. The following parameters are used for the specified account:

- IP address of the SIP server: **94.86.221.115**
- Data for authorizing login/password: **Tera/Tera**
- Method for transmitting DTMF tones: **rfc2833**
- Fax message transfer mode: **T.38**
- List of used voice codecs: **G.711(a-law, u-law), GSM, G.729**

When the parameters of the SS7 signaling system are configured using the WEB interface, the decimal system is used to specify the values of signaling point codes. The data received from the operator uses a 6-6-2 format, i.e. 14 code bits are divided into 3 groups that contain 6, 6 and 2 bits, respectively.

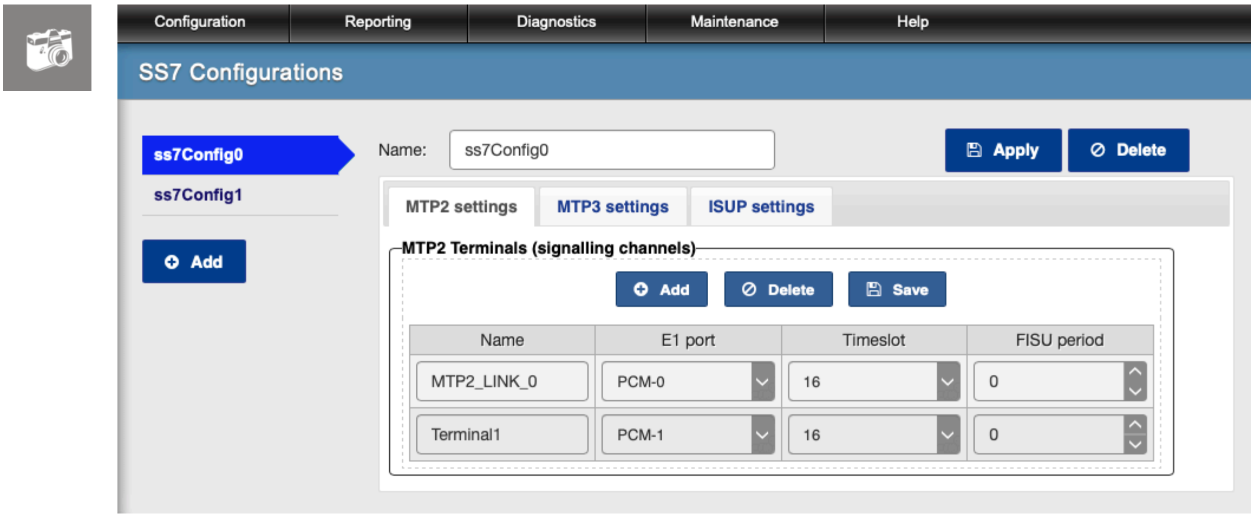
For example the TC-1 node code is represented as "16-10-0". This code will be converted to decimal notation as follows. The 6 first bits are equal to 16 (binary notation: 010000), the 6 following bits are equal to 10 (binary notation: 001010) and the 2 last bits are equal to 0 (binary notation: 00). The "glueing" of all bits results in 14-bit binary number «01000000101000» that is equal to 4136 in decimal notation. The remaining signaling codes are converted in the same way. This results in the following values:

- 16-10-0 → 4136
- 16-20-0 → 4176
- 16-30-0 → 4216
- 16-40-0 → 4256
- 0-1-3 → 7

Now we can start creating the configuration.

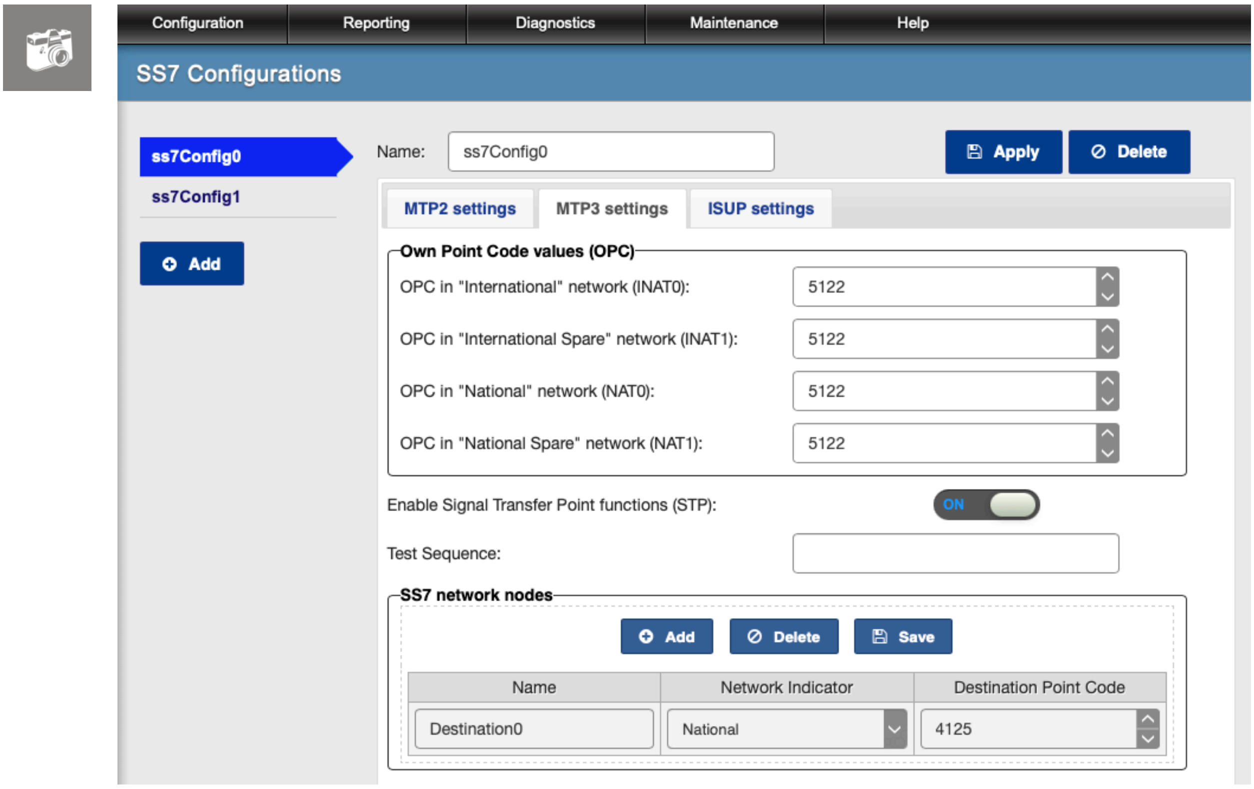
Let's create the SS7 configuration block. We add the terminals in the "MTP2 settings" tab (Figure 54). In accordance with the configuration the channel intervals (CI) 1 are used for signaling channels. The Channel Interval is equivalent to "Timeslot" parameter in our case.

Figure 54. SS7 Configuration – MTP2 Terminals



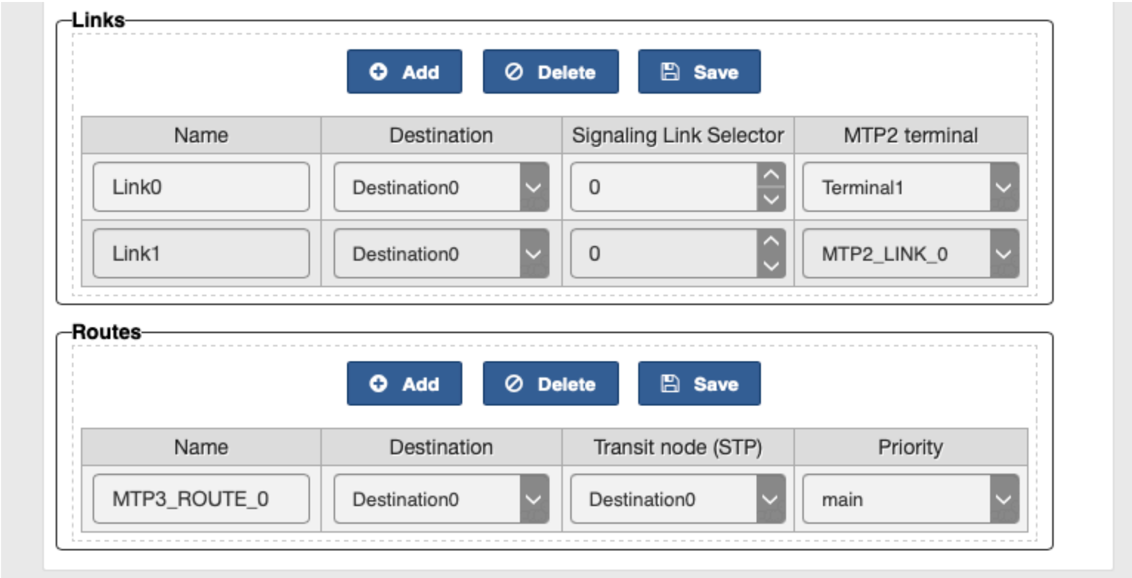
Specify "National" network code in the "MTP3 Parameters" tab and create records in the "SS7 Network Nodes" table for the connected operator nodes (Figure 55).

Figure 55. SS7 Configuration – MTP3 settings (1)



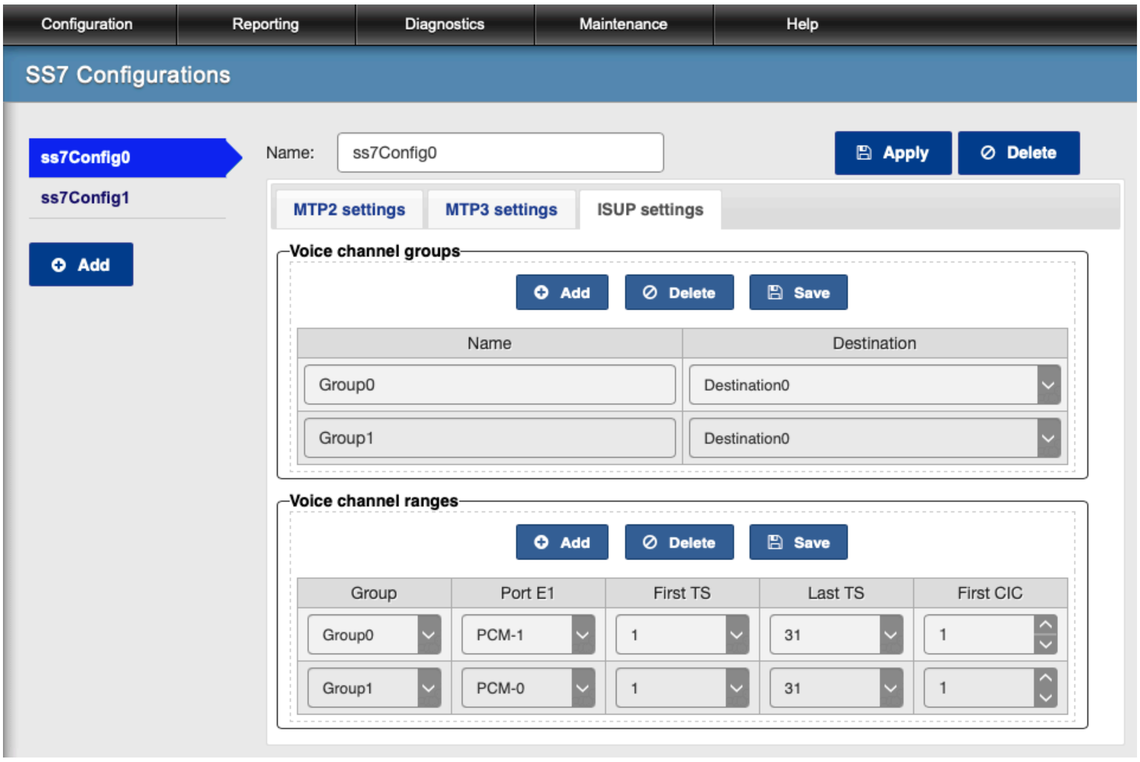
The links and routes are added then. Reservation should be taken into account when routes are created (Figure 56).

Figure 56. SS7 Configuration – MTP3 settings (2)



Create groups and ranges of voice channels in the "ISUP settings" tab (Figure 57) using division into groups by the channel type (local and long-distance) as well as CIC numbering.

Figure 57. SS7 Configuration – ISUP settings



The next step is to create dial plans and modifiers to provide conversion of subscriber numbers into the format assigned by the operator.

The dial plans for selecting the numbers that start with "044" and "00" prefixes are shown below (Figure 58, Figure 59).

Figure 58. 044 Dial Plan

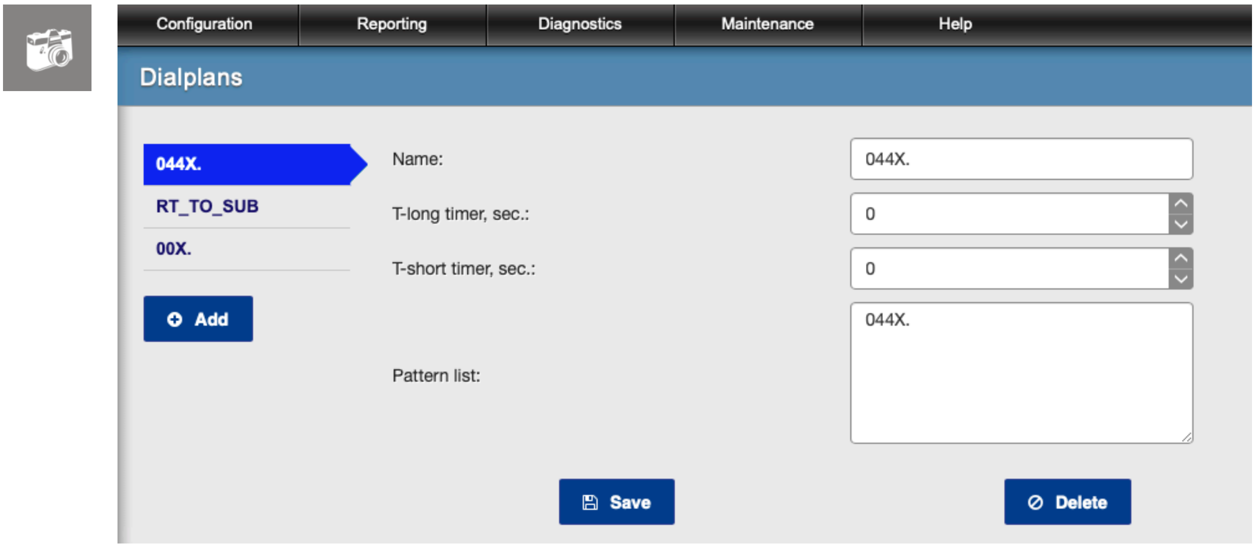
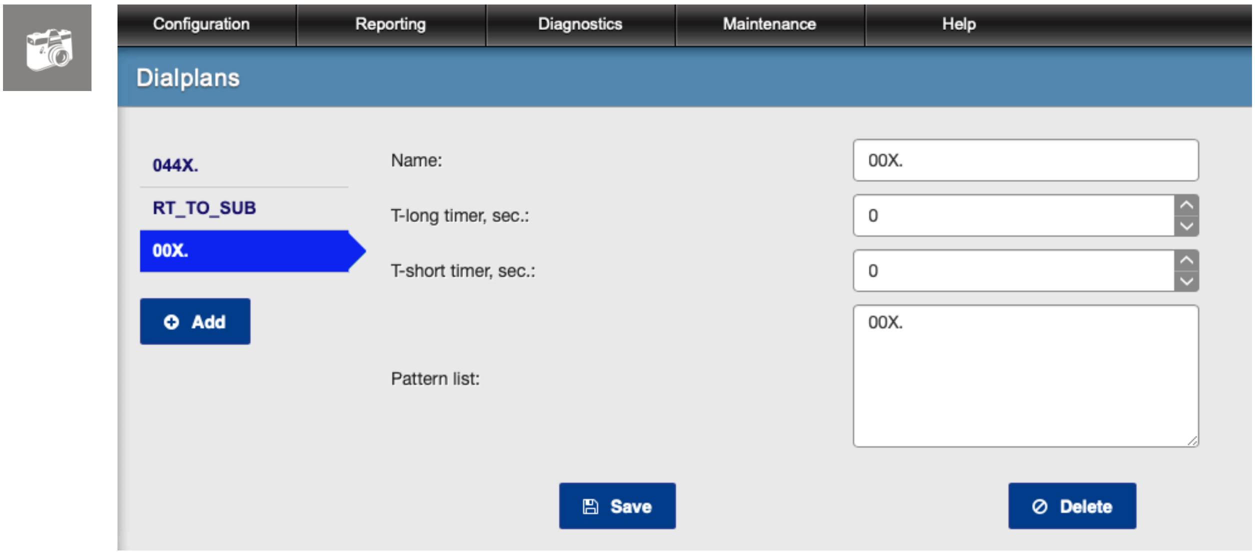


Figure 59. 00 Dial Plan



Modifier configuration:

- The modifier for converting the number of the calling subscriber (Figure 60)
- The modifier for converting the number of the called subscriber for local calls (Figure 61)
- The modifier for converting the number of the called subscriber for long-distance/international calls (Figure 62)

Figure 60. Modifier for Converting the Calling Subscriber Number

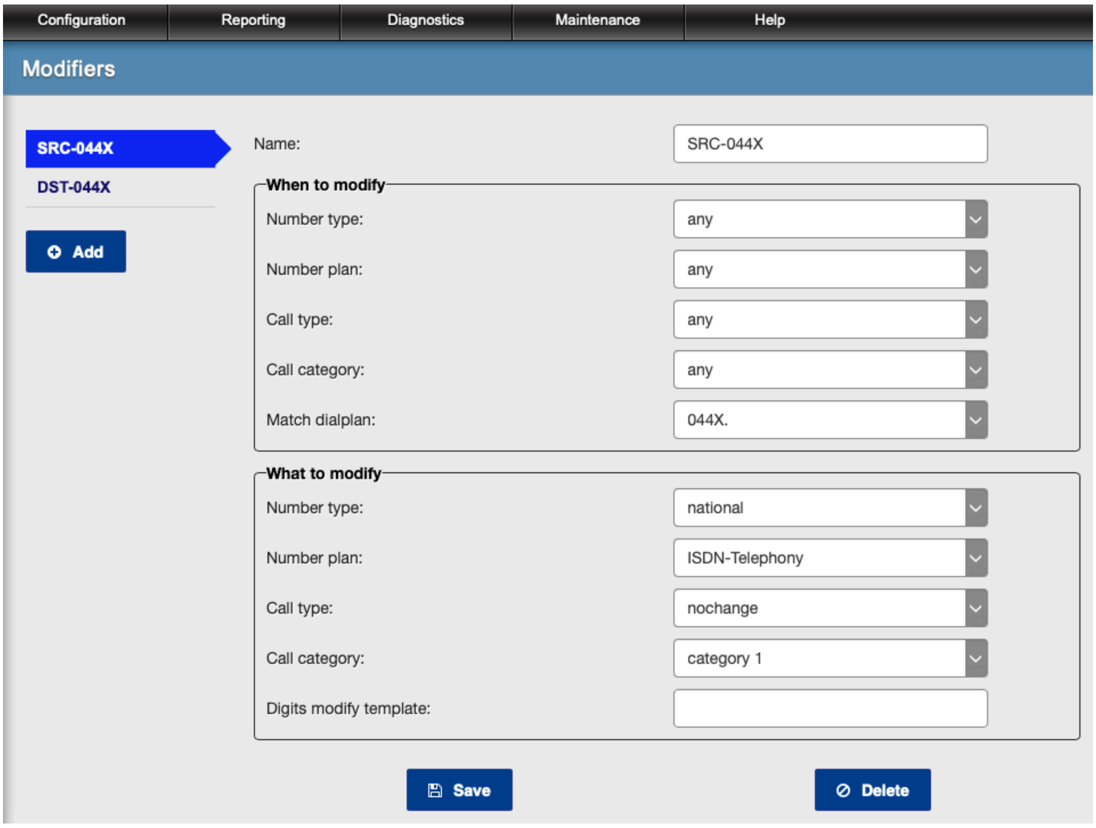


Figure 61. Modifier for Converting the Called Subscriber Number for Local Calls

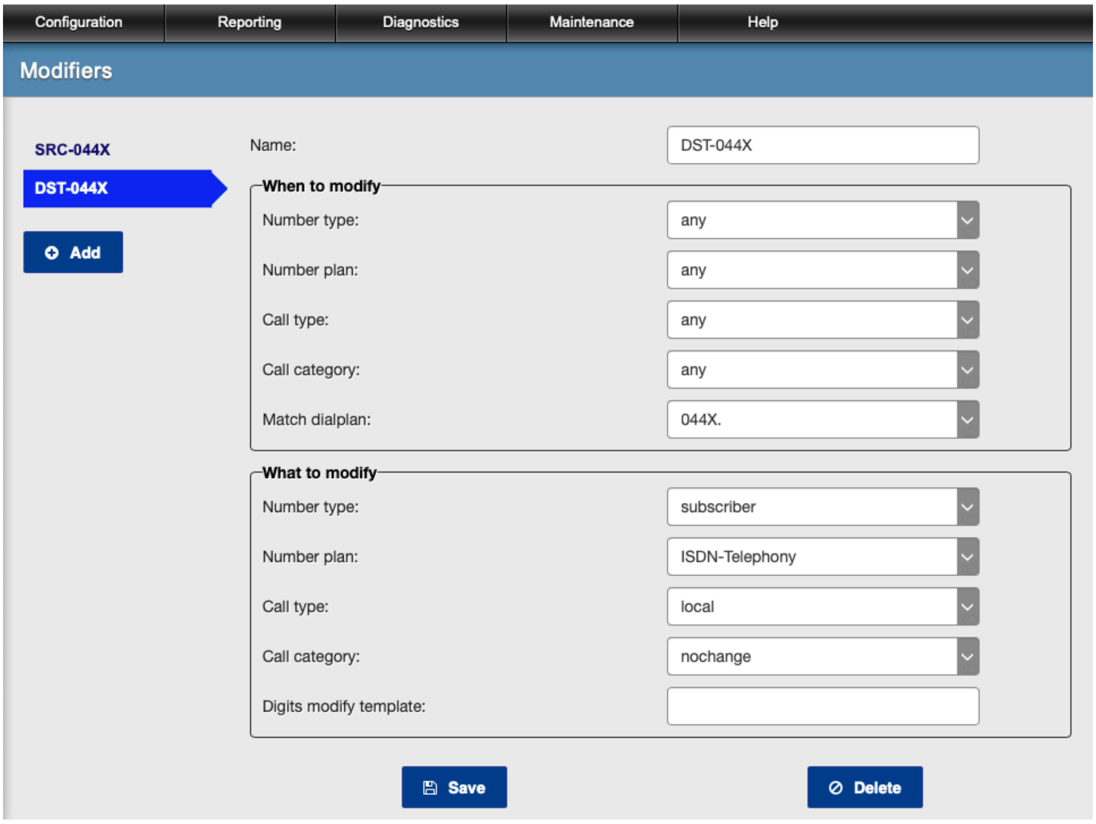
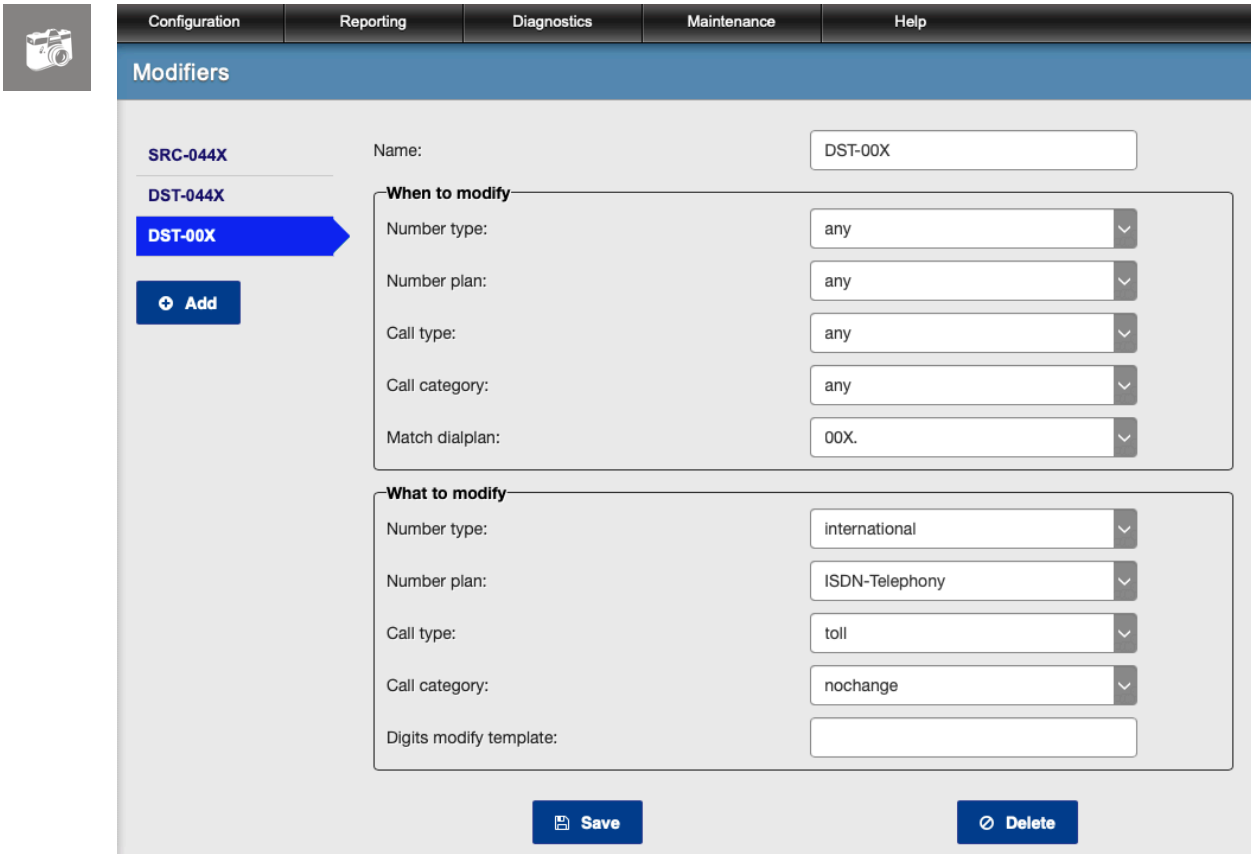


Figure 62. Modifier for Converting the Called Subscriber Number for Long-Distance/International Calls

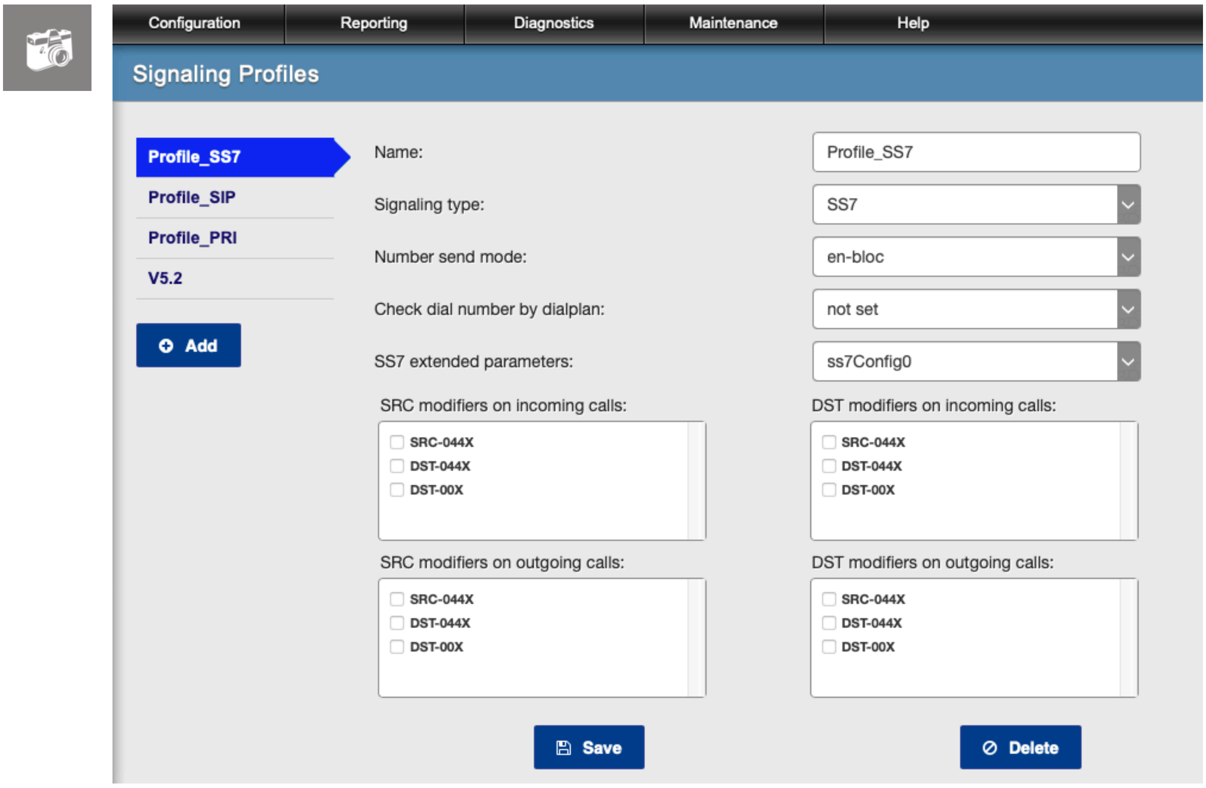


The next step is to create the signaling profiles and then to link them subsequently to the specific channels.

The "SS7 Additional Parameter Block" parameter is available in the profile for the "SS7" signaling type. This parameter is used to link the specific profile with one of the SS7 configurations that were created earlier.

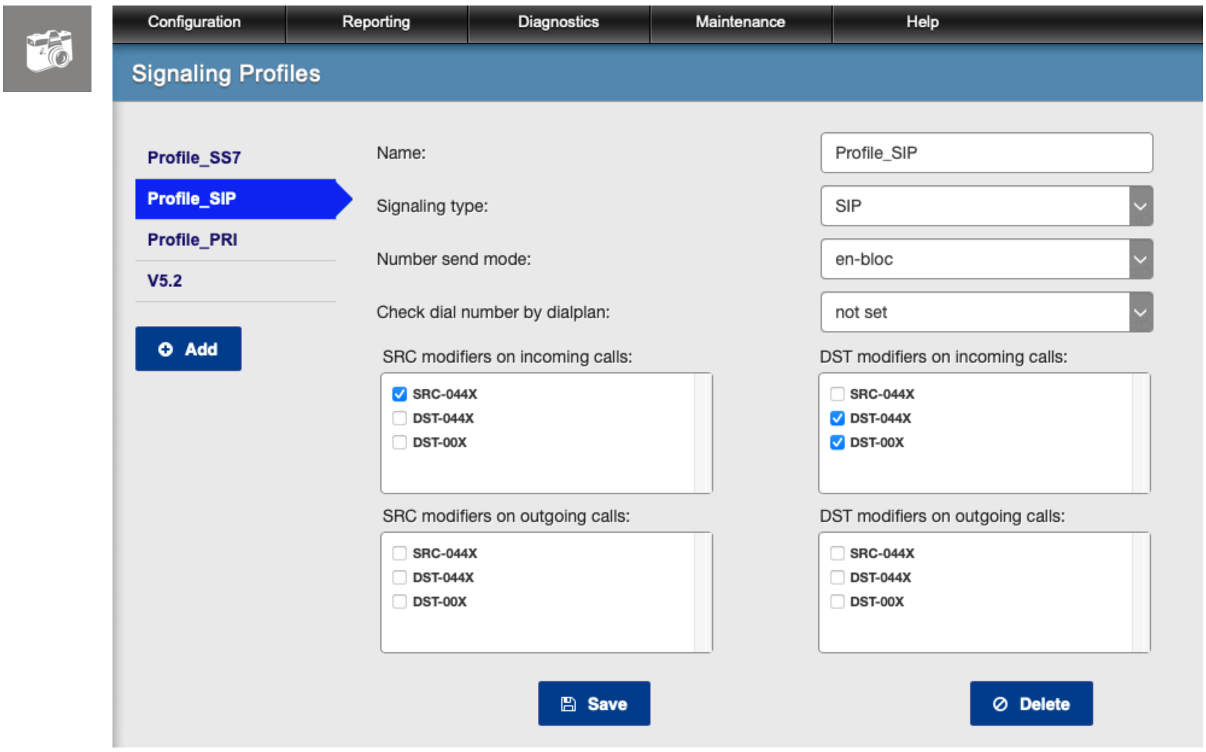
Let's create the SS7 signaling profile for this task. The profile parameters are shown in the Figure below (Figure 63).

Figure 63. Signaling Profile – SS7 Type



Then we create the SIP signaling profile and select the number modifiers for incoming calls that were created earlier (Figure 64).

Figure 64. Signaling Profile – SIP Type

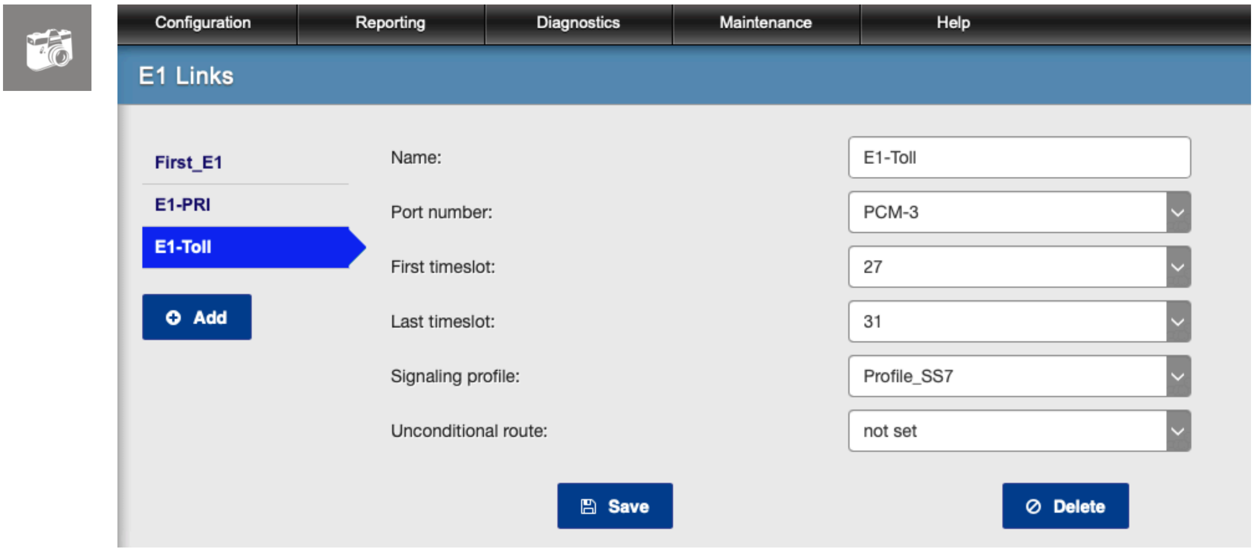


Consequently, when the call reaches the gateway from the SIP network it will be classified as the local or long-distance call and all modifications of the numbers will be performed before the call is routed to the SS7 network by the gateway.

Then we create E1 links and voice channel groups located within one E1 path that uses the single signaling profile.

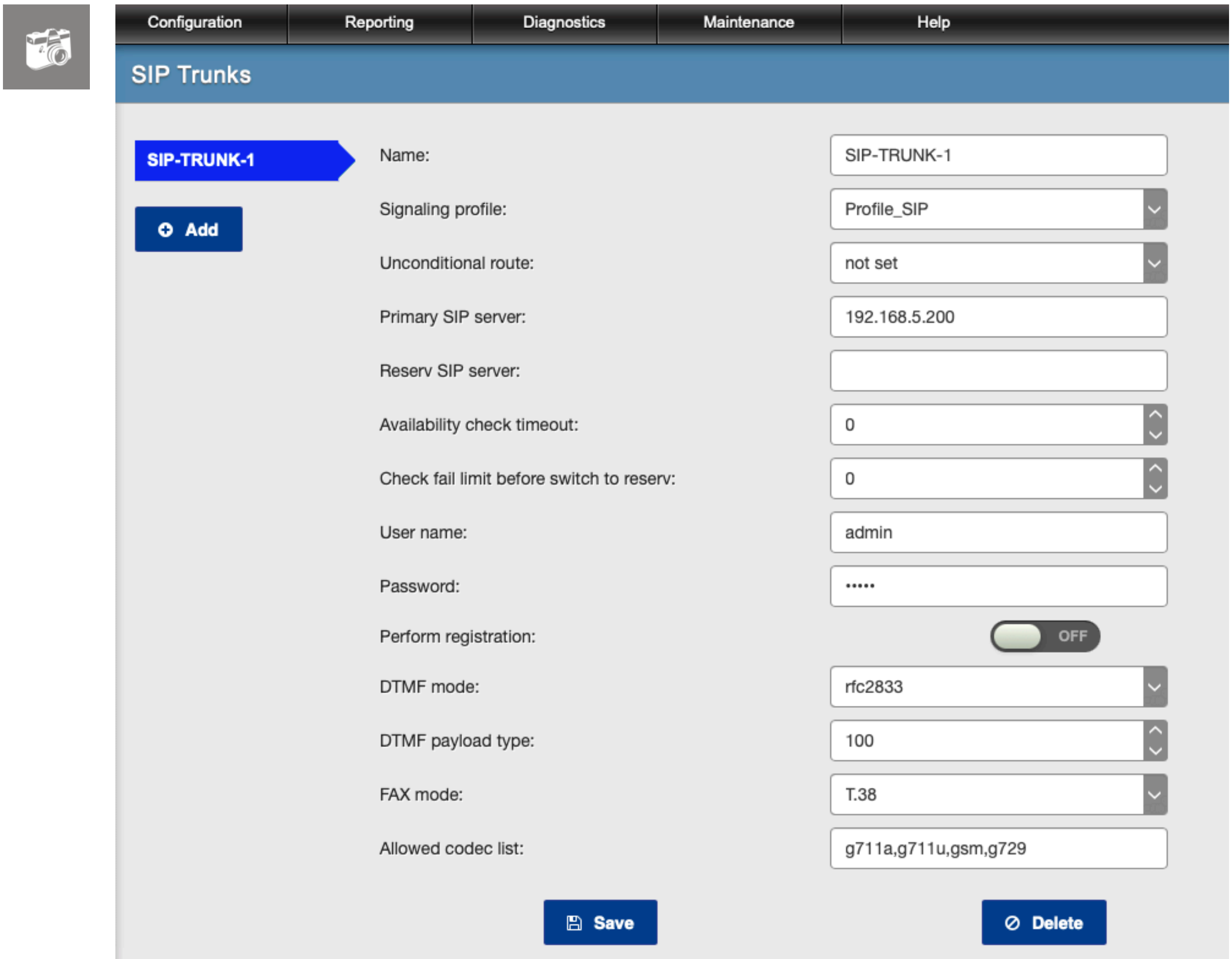
Groups for local and long-distance calls are created separately. All links use the same signaling profile and the link timeslot allocation meets the requirements for allocating the voice channels received from the operator. The Figure below (Figure 65) shows the list of all created links and parameters of the last link.

Figure 65. E1 Links



The next step is to create the SIP trunk account (Figure 66).

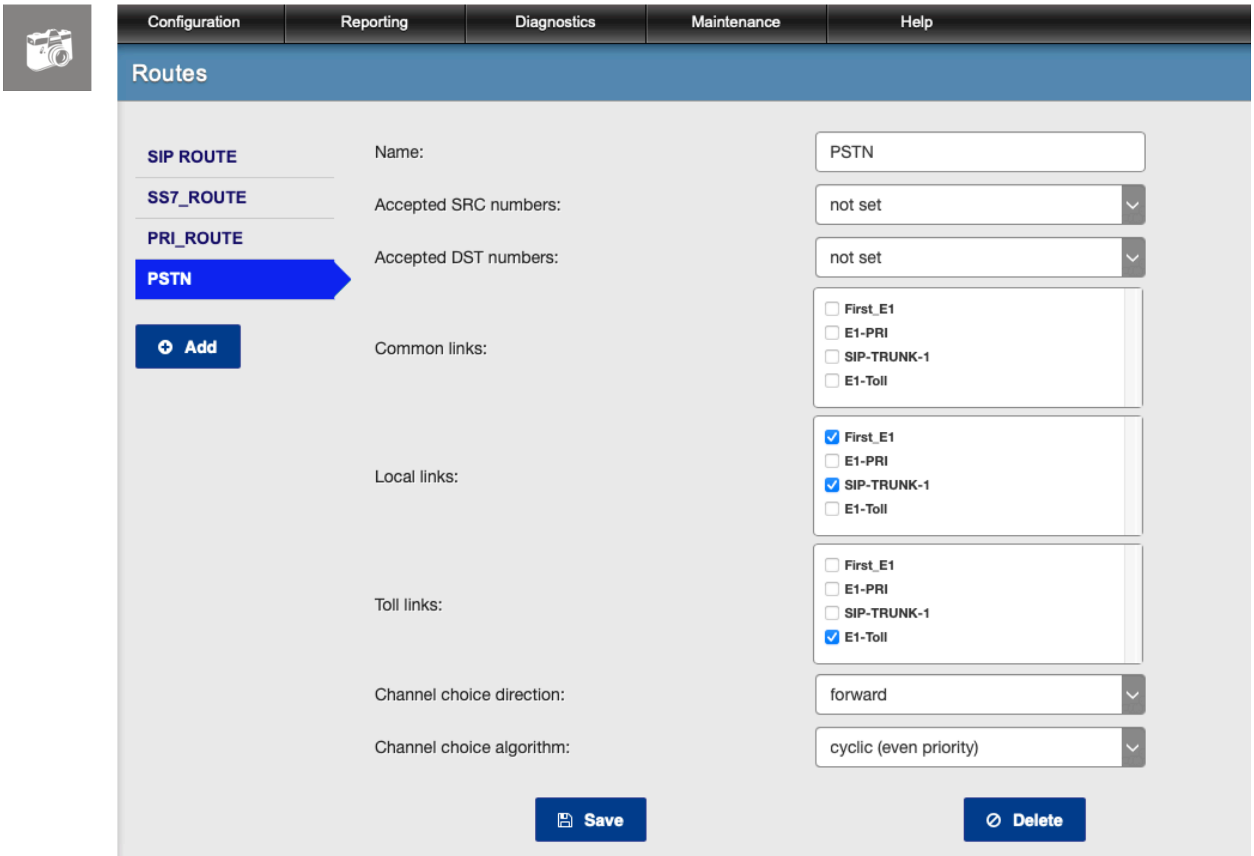
Figure 66. SIP Trunks



Routes are created then to link the link and trunk groups into a single route.

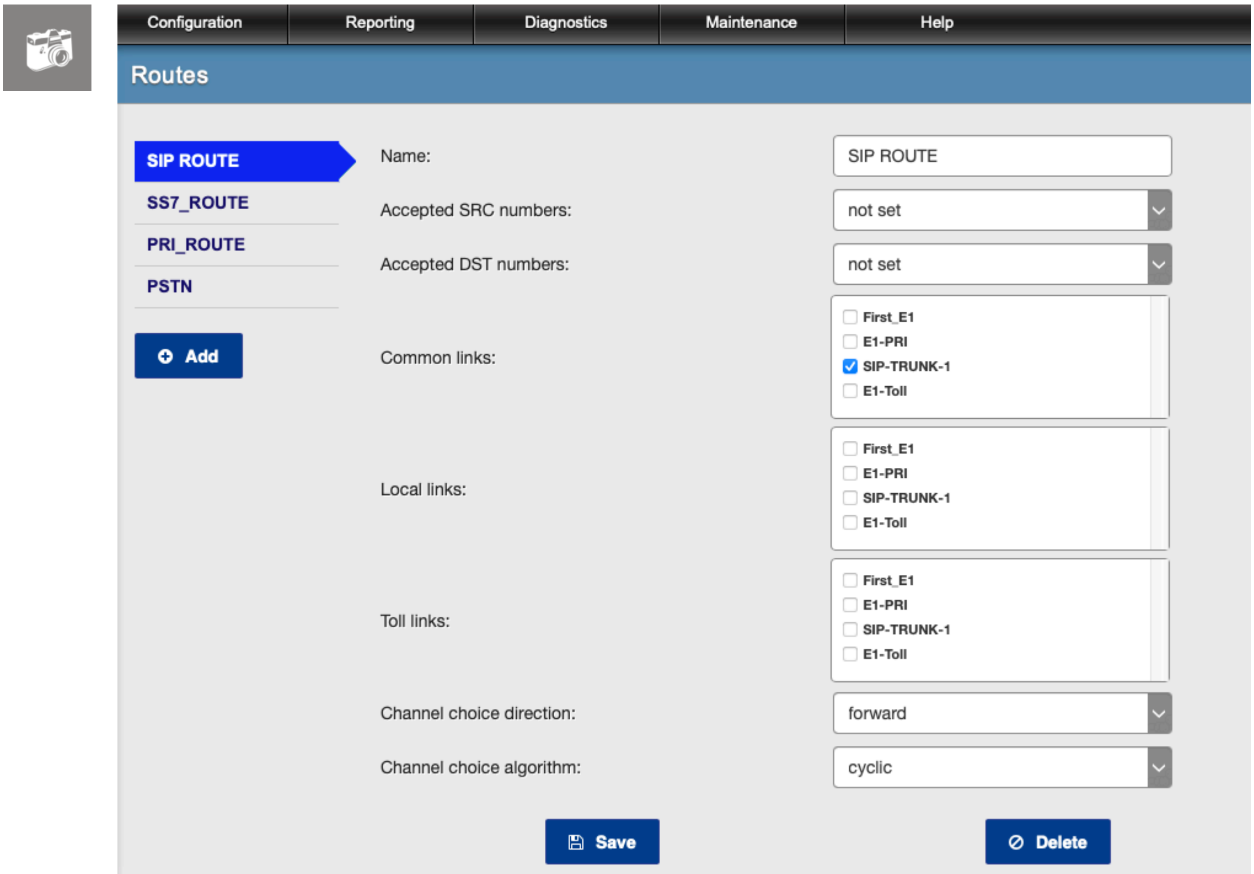
The algorithm for selecting the next free channel can be also specified in the route parameters. Based on the operator's requirements the gateway should select free channels using the cyclic algorithm and prioritizing the even channels. The following route is created accordingly for the PSTN operator network (Figure 67).

Figure 67. Routes for PSTN Operator Network



The figure below (Figure 68) shows the route for the SIP network on the customer's side.

Figure 68. Routes on the Customer's Side

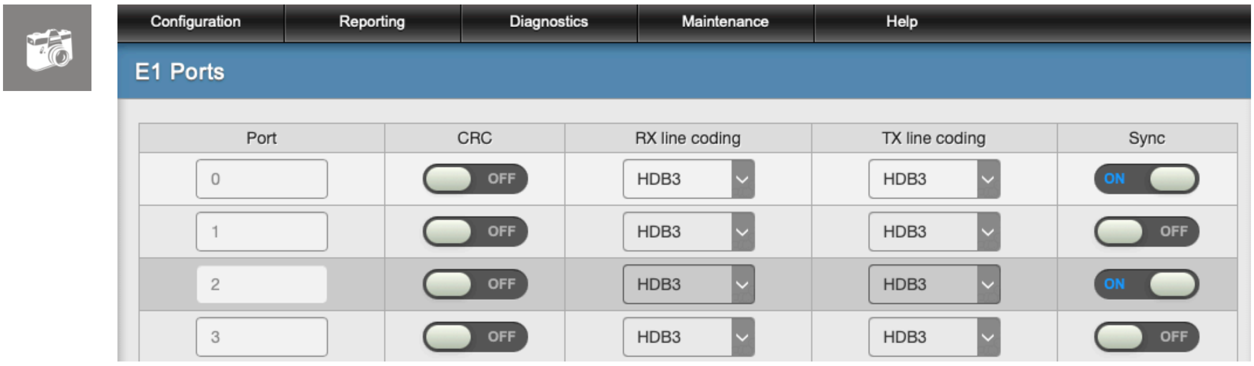


The current task of using the gateway uses preassigned call routing, i.e. it has been defined that all calls that come from the operator's PSTN should be routed to the customer's SIP network. The same rule is effective for the opposite direction - all calls that are received from the customer's SIP network are routed by the gateway to the operator's PSTN.

Taking the above into consideration, the rule for call routing can be defined. To do this we should select the "SIP-Net" SIP network in all created E1 links in the "Unconditional Routing Direction" parameter and the "PSTN-Net" PSTN network should be selected in the SIP trunk for the similar parameter.

Finally, we configure synchronization and parameters of E1 ports in accordance with the operator's requirements (Figure 69).

Figure 69. Port Parameters



To apply the configured parameters properly restart the gateway.

7.2. SIP Gateway in V5.2. Connecting VoIP Ethernet Network Subscribers to PSTN Number Capacity.

Objective: Organize an IP phone network for 100 numbers and integrate it into a PSTN. To provide connection to the PSTN the operator should provide one E1 path with a V5.2 interface.

Operator's data: 1E1, HDB3 coding method, CRC parameter is disabled, the customer's gateway is synchronized from the operator's E1 path, V5.2 interface, operator's side - LE (local exchange), customer's side - AN (access network).

Additional data for V5.2 interface and E1 path:

- Interface option - 0
- Interface identifier - 4
- Link identifier (E1 port) – 1
- Signaling channel - 16
- Quantity of V5.2 ports (subscribers) - 100
- Numbers of ports (subscribers) in the interface - 0-99
- Numbering of subscribers in PSTN - 220000-220099

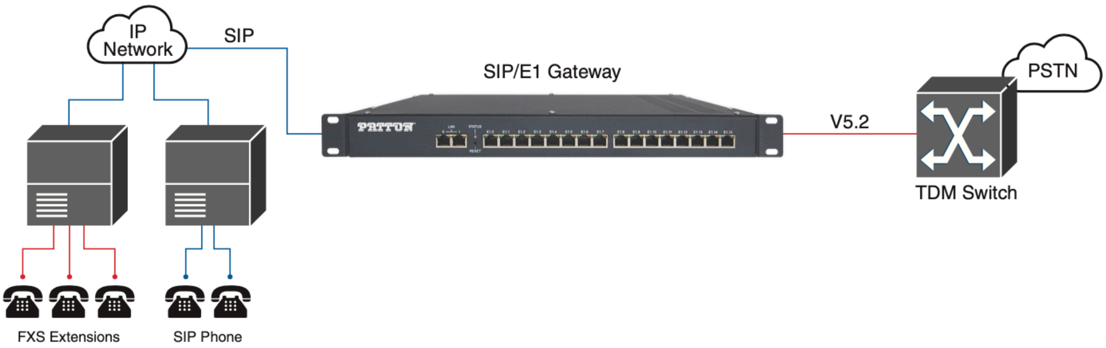
Protection protocol parameters:

- Main link - 1 (since it is the only link)
- Backup link - not used (since there is only 1 link in the configuration)
- Logical channel identifier - 4

Numbering allocated by the PSTN operator for these subscribers is not used in the configuration process since the gateway here is the remote V5.2 node i.e. it does not analyze the dialed numbers. Numbers in the PSTN and V5.2 interface are correlated serially: 220000 – 0, 220001 – 1 etc.

Schematically this task looks as follows:

Figure 70. Connection diagram of SIP/V5.2 Gateway in AN mode



A SIP/E1 Gateway directed towards a TDM network looks like a remote V5.2 subscriber node (AN side) and a SIP/E1 Gateway directed towards IP network looks like a VoIP server where SIP subscribers are registered.

Now we can start creating the configuration.

Create the V5.2 block configuration (Configuration → V5.2 Configuration) (Figure 71).

Figure 71. V5.2 Configuration

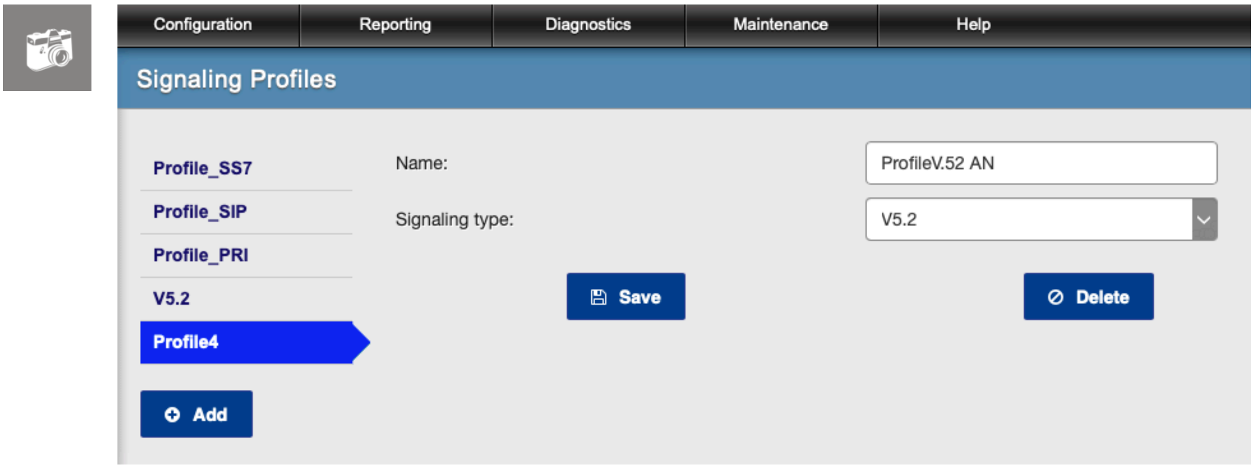
Fill in fields in block 1 (Figure 71) - Interface Type - AN: Access Network, Option and Interface Identifier based on the operator's source data. Do not select Number Dial Plan since the gateway will not analyze the digits dialed by the subscribers.

Fill in block 2 (Figure 71) - Adding the link to the interface (E1 port). The Link Identifier should be set in accordance with the operator's data. Select the physical E1 gateway port - PCM0. Set the signaling channel number (mandatory). Save the links (Button 2.1). Save the entire configuration (Button 2.2). You will still see the current configuration editing form.

Set the protection protocol parameters in the block 3 (Figure 71) - Main Link: select the link created in the previous block. Leave the Backup Link blank since the configuration contains only 1 link. Set the number of the logical channel. Save the configuration (Button 2.2).

Next step. Create the signaling profile of V5.2 interface (Configuration -> Signaling Profiles) (Figure 72).

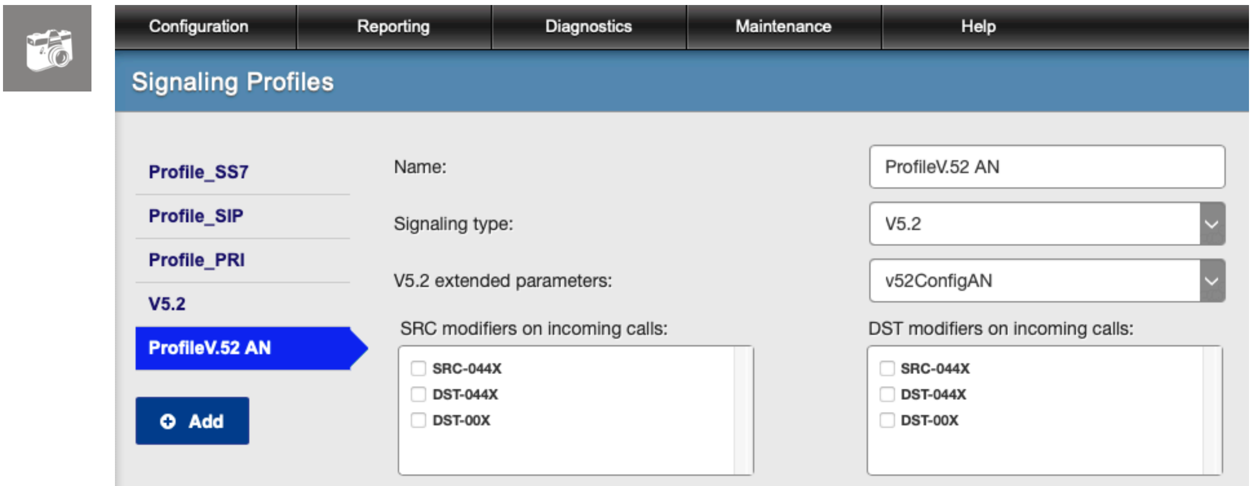
Figure 72. Signaling Profiles



Add the new profile, specify a meaningful name for the created profile, and select V5.2 signaling type. Save the profile.

After the profile is saved it will be updated and a new item will be added - Block of V5.2 extended parameters (Figure 73).

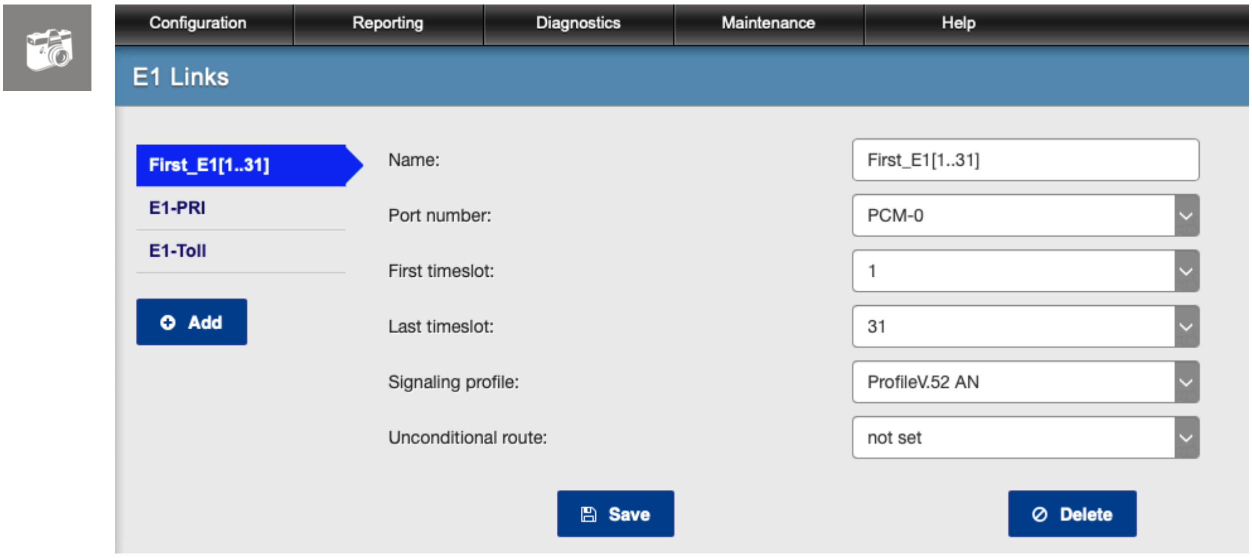
Figure 73. Signaling Profiles – Extended parameters



In the Block of V5.2 extended parameters field select the V5.2 configuration that was created earlier and save the changes.

Next step. Create E1 port link (Configuration → E1 Links) (Figure 74).

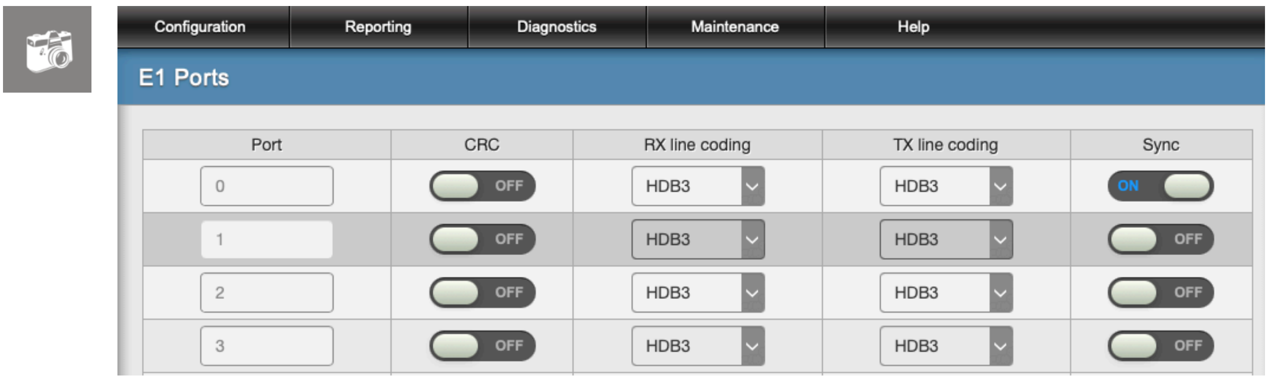
Figure 74. E1 Links



E1 Port Number - select PCM0 (it should match the port that was selected when the V5.2 configuration had been created). Use the entire E1 port - set the start and end time slot to 1 and 31. Select the Signaling Profile that was configured during the previous step. Do not select direction of unconditional routing since we'll create strict SIP-V5.2 binding later when SIP subscribers are created.

To complete the TDM configuration, synchronization and parameters of E1 ports should be configured in accordance with the operator's requirements (Configuration → E1 Port Configuration) (Figure 75).

Figure 75. E1 Port Configuration



Next step. It is required to add SIP subscribers to the configuration. Their number should match the number capacity allocated by the operator over V5.2 (i.e. 100 numbers).

Each SIP subscriber should be assigned the unique SIP number and the name and password should be set up. This configuration can have any SIP numbers and names. They will not be used for routing, i.e. when the subscriber whose logical number is 2 has SIP number 202 then the other SIP subscriber whose number is 201 will not be able to call to this subscriber using 202 number since the gateway does not analyze dialed numbers and sends them to the higher-level exchange over V5.2. As such it will be possible to make a call using number 220002 that is assigned by the operator. The SIP number, name and password are required to register the SIP subscriber with the gateway.

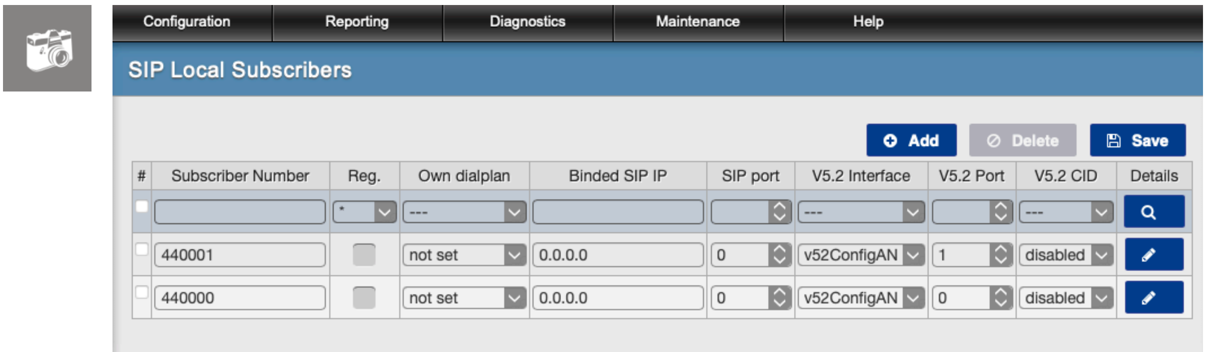
To simplify the example, assume that the SIP numbers are within the 440000 – 440099 range and the name and password of each subscriber match the number. Each created SIP subscriber should be bound to the V5.2 interface port. SIP number, V5.2 port and PSTN number are correlated linearly.

Table 36. Configuring SIP Subscribers

SIP-number	SIP-name	SIP-password	The port number V5.2 interface	PSTN-number
440000	440000	440000	0	220000
440001	440001	440001	1	220001
…	…	…	…	…
440099	440099	440099	99	220099

Create SIP subscribers (Configuration → SIP Subscribers) (Figure 76).

Figure 76. SIP Subscribers



Create the first subscriber with 440000 number. Specify the number, select the V5.2 interface to be bound (this interface was created earlier) and set up V5.2 port. Set up the bound IP address and SIP port if required (if it is required to determine whether it is possible to register the subscriber with the specific address or port only). Press Save. The extended parameters of the added record that are available by pressing Details button - are automatically set to default values. SIP name and SIP password match the Subscriber Number by default (Figure 77).

Figure 77. SIP Subscribers - extended parameters

If it is not required to change the default values of the additional parameters then there is no need to open the Parameters window ( ). This can make the process of entering subscribers faster.

Adding a second subscriber.

When there is at least one subscriber, then you can make the process of entering the next subscriber faster: highlight the subscriber with the highest number (put the cursor in any field of this subscriber) and the subscriber's line will be highlighted in dark grey. Press Add button then. The added subscriber will have the Interface V5.2 value matching the same value of the highlighted subscriber and the value of V5.2 Port will increase by 1 (next in sequence).

Enter all subscribers in the same way and save them.

This will complete configuring the gateway. SIP subscribers can now register with the gateway using the IP address of the gateway with their login and password.

Reload the gateway to apply all settings properly.

If it is required to enter very many subscribers then to make the process faster the list of all subscribers can be created with any available table editor and imported to the gateway configuration.

Let's consider the process of adding the subscribers using Microsoft® Excel™.

First, create 2 subscribers in the Web interface as described above. Then select Maintenance → Backup Files (Figure 49). Click the Download button on the SIP Subscribers line and save the subs_sip.csv file to the PC's disk. Open this file with Excel (Figure 78).

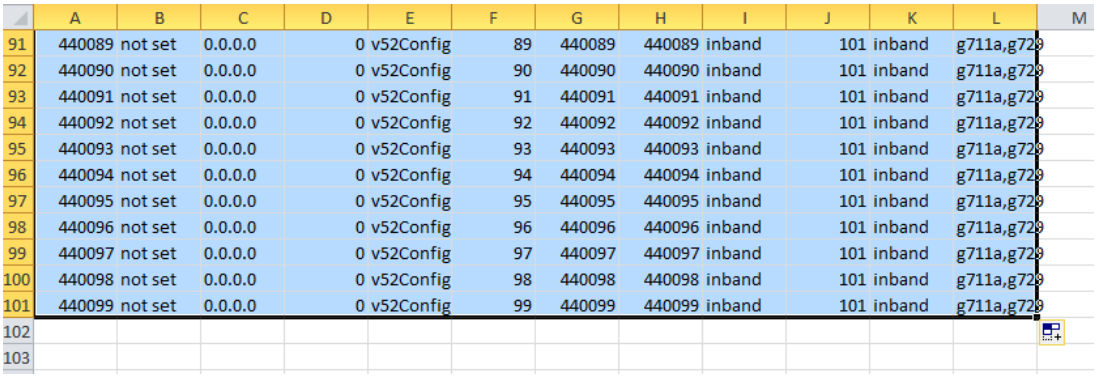
Figure 78. subs_sip.csv

	A	B	C	D	E	F	G	H	I	J	K	L
1	#Number	Dialplan	Bind SIP IF	Bind SIP P	V5.2 If	V5.2 Port	SIP name	SIP pass	DTMF mod	Payload ty	FAX mode	Codecs list
2	440000	not set	0.0.0.0	0	v52Config	0	440000	440000	inband	101	inband	g711a,g729
3	440001	not set	0.0.0.0	0	v52Config	1	440001	440001	inband	101	inband	g711a,g729

Highlight all fields in the second and third rows (do not highlight the first row with the titles). Grab the highlighted lower right corner and drag it down. When you drag it down the new rows will be added that are filled automatically. All modified fields will be automatically incremented in each new row.

When the required number of the subscribers is reached (number 440099 or row 101 in our example) then stop dragging down. (Figure 79).

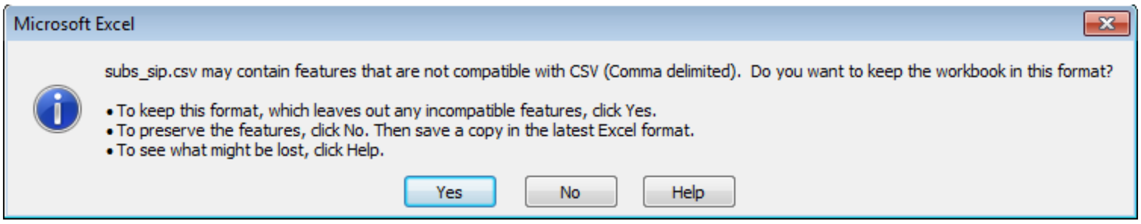
Figure 79. subs_sip.csv - add rows



	A	B	C	D	E	F	G	H	I	J	K	L	M
91	440089	not set	0.0.0.0	0	v52Config	89	440089	440089	inband	101	inband	g711a,g729	
92	440090	not set	0.0.0.0	0	v52Config	90	440090	440090	inband	101	inband	g711a,g729	
93	440091	not set	0.0.0.0	0	v52Config	91	440091	440091	inband	101	inband	g711a,g729	
94	440092	not set	0.0.0.0	0	v52Config	92	440092	440092	inband	101	inband	g711a,g729	
95	440093	not set	0.0.0.0	0	v52Config	93	440093	440093	inband	101	inband	g711a,g729	
96	440094	not set	0.0.0.0	0	v52Config	94	440094	440094	inband	101	inband	g711a,g729	
97	440095	not set	0.0.0.0	0	v52Config	95	440095	440095	inband	101	inband	g711a,g729	
98	440096	not set	0.0.0.0	0	v52Config	96	440096	440096	inband	101	inband	g711a,g729	
99	440097	not set	0.0.0.0	0	v52Config	97	440097	440097	inband	101	inband	g711a,g729	
100	440098	not set	0.0.0.0	0	v52Config	98	440098	440098	inband	101	inband	g711a,g729	
101	440099	not set	0.0.0.0	0	v52Config	99	440099	440099	inband	101	inband	g711a,g729	
102													
103													

Edit the required fields if required. Save the file and press Yes when you are prompted to confirm that you are going to save the file in csv format.

Figure 80. Changes saved confirmation

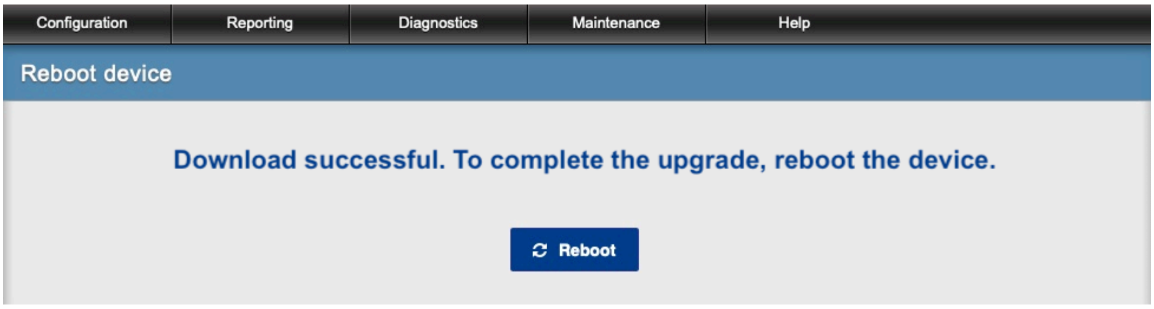


Return to the Web interface of the gateway: Maintenance → Update Download (Figure 50).

Press Select File in the SIP Subscribers line and find the edited file. Press Download and confirm the operation when prompted. After you press OK, the Subscribers file will be downloaded. After the file is downloaded you will be prompted to reload the device (Figure 81).

Reload the gateway.

Figure 81. Device Reboot Confirmation



7.3. V5.2 Gateway in SIP. Connecting TDM Remote Node with V5.2 Interface to SoftSwitch (to VoIP Network).

Objective. There is a TDM remote node for 100 numbers with V5.2 interface, 1E1 port. It is required to connect the remote node to the operator's VoIP network.

Source data for V5.2 interface and E1 path:

- HDB3 coding method is used for E1 path, CRC parameter is disabled, the customer's remote node is synchronized from the operator's E1 path.
- Subscriber remote node - AN (access network)
- SIP/E1 Gateway - LE (local exchange)
- Interface option - 0
- Interface identifier - 4
- Link identifier (E1 port) – 1
- Signaling channel - 16

Protection protocol parameters:

- Main link - 1 (since it is the only link)
- Backup link - not used (since there is only 1 link in the configuration)
- Logical channel identifier - 4
- Quantity of V5.2 ports (subscribers) - 100
- Numbers of ports (subscribers) in the interface - 0-99

Source SIP data:

- IP address of SIP server: 192.168.110.1;
- Assigned number capacity: 220000 – 220099;
- SIP name and password of each subscriber matches his number.

Existing network directions:

- 1xx - 3 characters
- 22xxxx – 6 characters
- 0xxx. - unlimited number of characters (the minimum number of characters is 4).

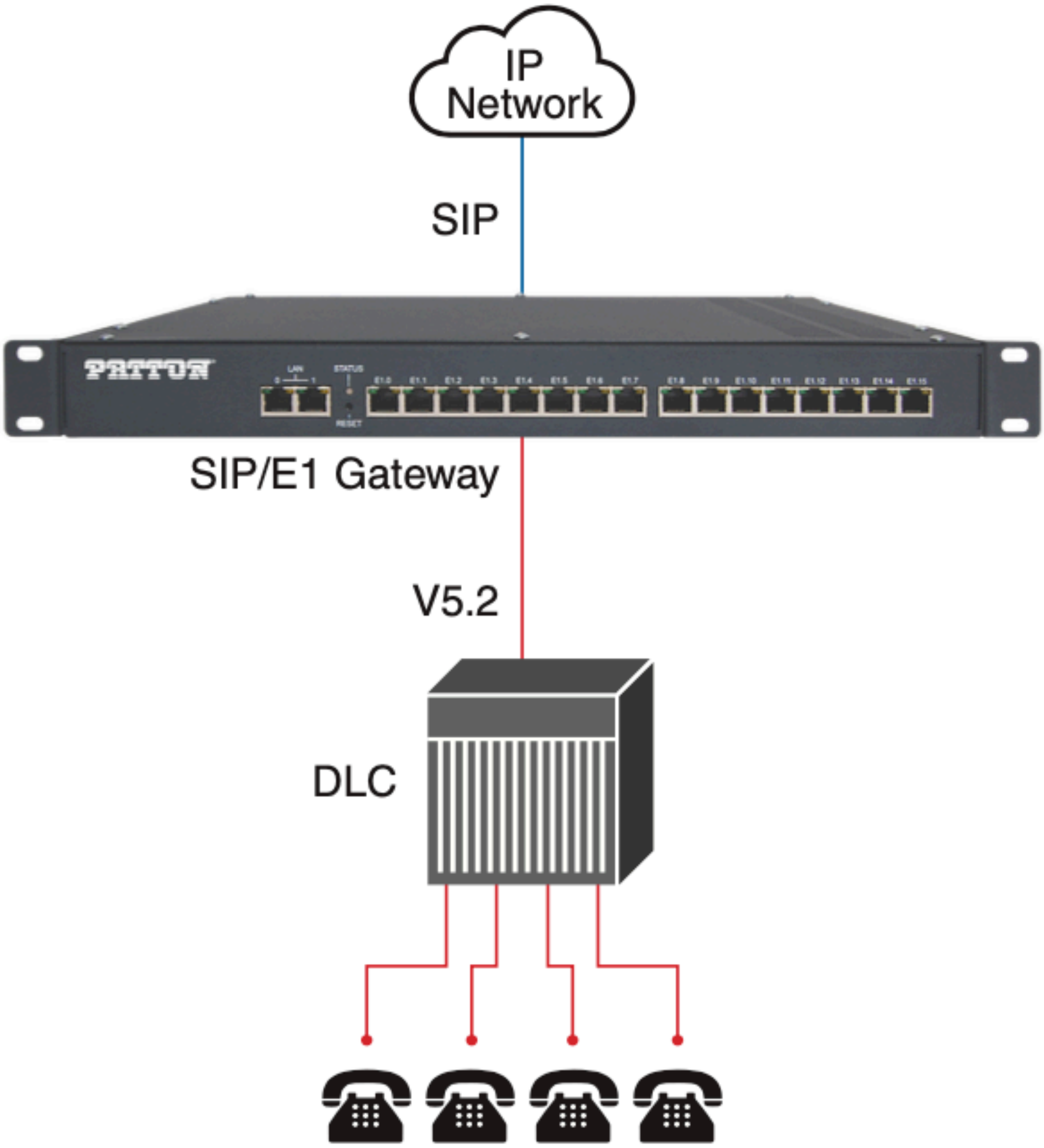
V5.2 interface ports and SIP numbers are correlated serially.

Table 37. Example table of V5.2 interface ports and SIP numbers

Number	SIP-name	SIP-password	The port number V5.2 interface
220000	220000	220000	0
220001	220001	220001	1
…	…	…	…
220099	220099	220099	99

Schematically, the task is presented in the figure below.

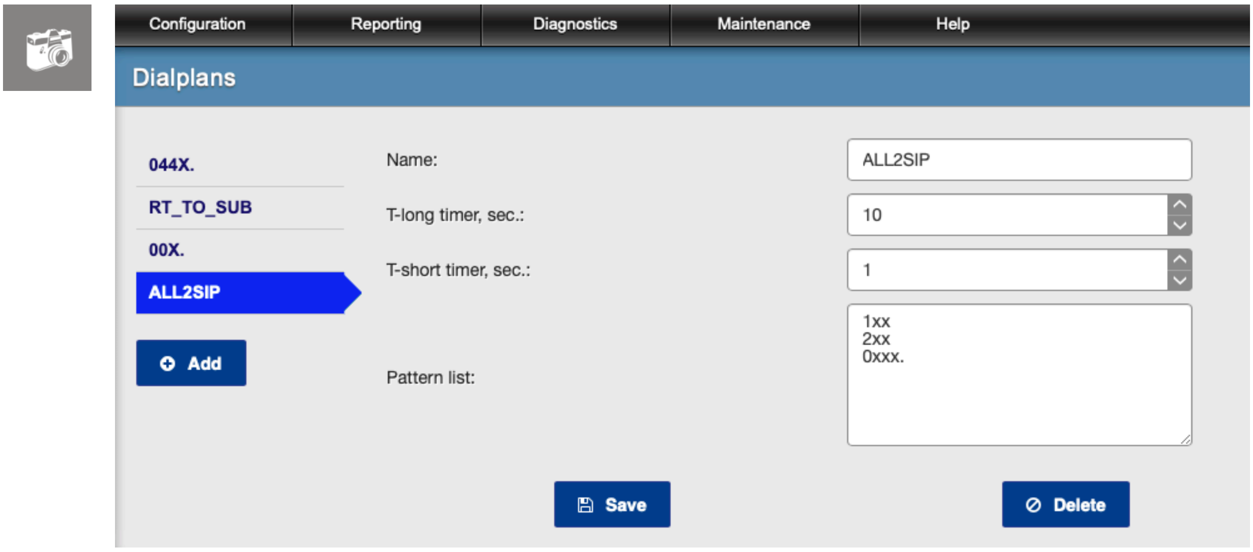
Figure 82. Connection diagram of V5.2/SIP Gateway in LE mode



Now we can start creating the configuration.

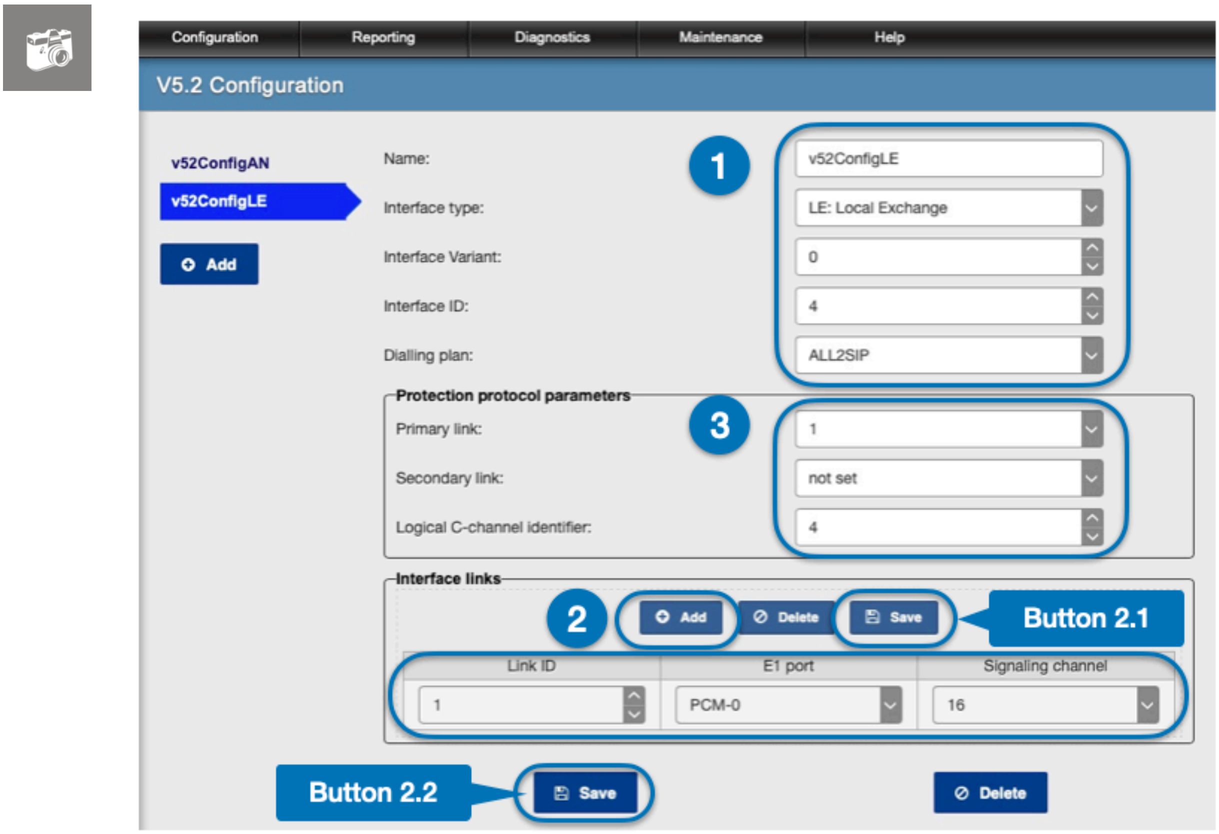
Create the dial plan with assigned numbering (Configuration → Dialplans) (Figure 83).

Figure 83. Dialplans



Create the V5.2 block configuration (Configuration → V5.2) (Figure 84).

Figure 84. V5.2 Configuration



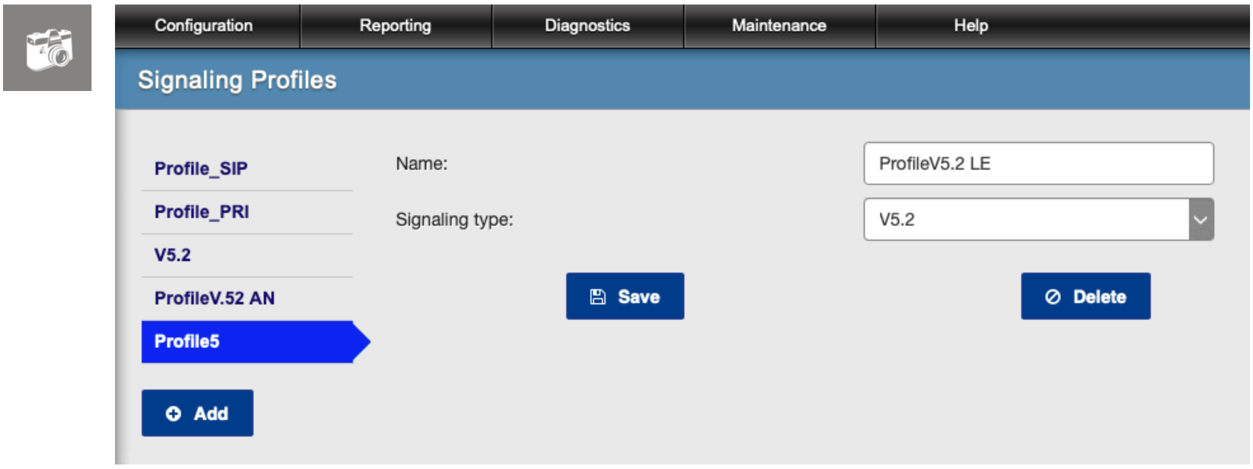
Fill in fields in block 1 (Figure 84) - Interface Type - LE: Local Exchange, Option and Interface Identifier based on the source data. Select the Dialplan that was created earlier (Figure 83).

Fill in block 2 (Figure 84) - add the link to the interface (E1 port). Set the Link identifier based on the source data. Select the physical port of the E1 gateway - PCM-0. Set the number of the signaling channel (mandatory). Save the links (Button 2.1). Save the entire configuration (Button 2.2). You will still see the current configuration editing form.

Set the protection protocol parameters (block 3) - Main Link: select the link created in the previous block. Leave the Backup Link blank since the configuration contains only 1 link. Set the number of the logical channel. Save the configuration (Button 2.2).

Next step. Create the signaling profile of the V5.2 interface (Configuration → Signaling Profiles) (Figure 85).

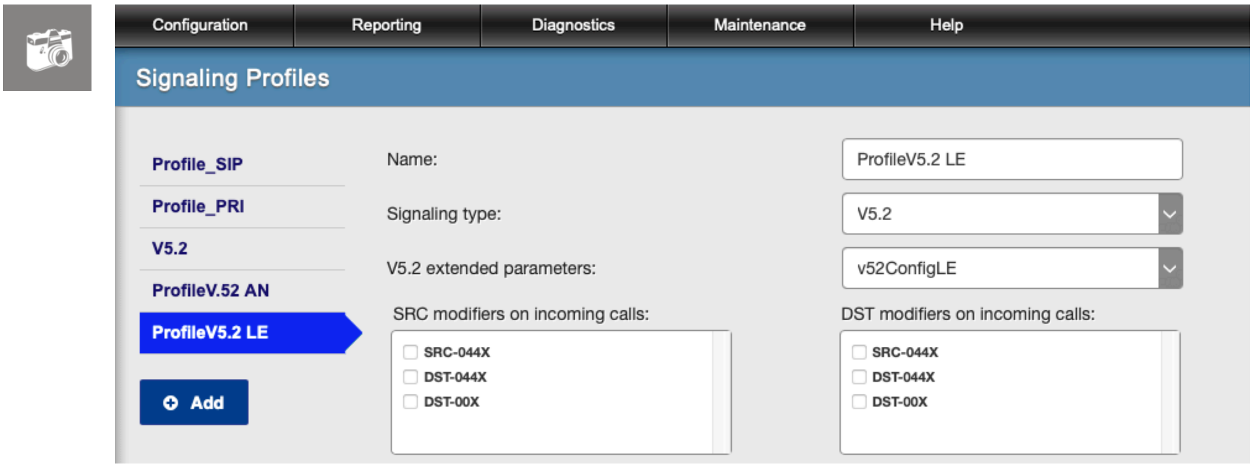
Figure 85. Signaling Profiles



Add the new profile, specify a meaningful name for the created profile, select V5.2 signaling type. Save the profile.

After the profile is saved it will be updated and a new item will be added - Block of V5.2 Extended Parameters (Figure 86).

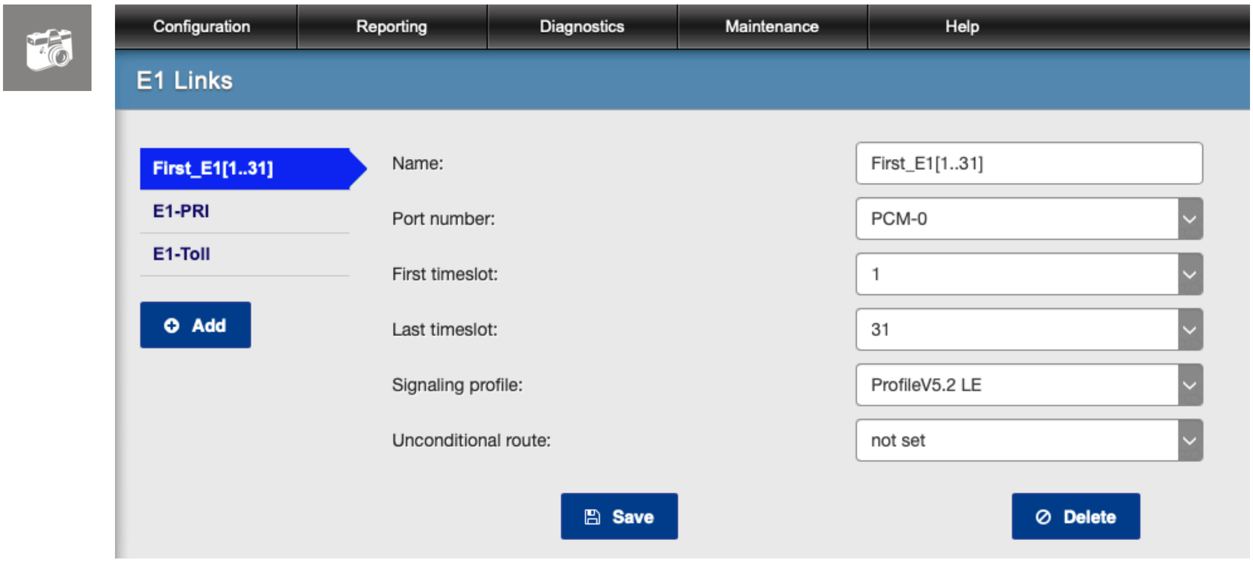
Figure 86. Signaling Profiles – extended parameters



In the Block of V5.2 extended parameters field select the V5.2 configuration that was created earlier and save the changes.

Next step. Create E1 port link (Configuration → E1 Links) (Figure 87).

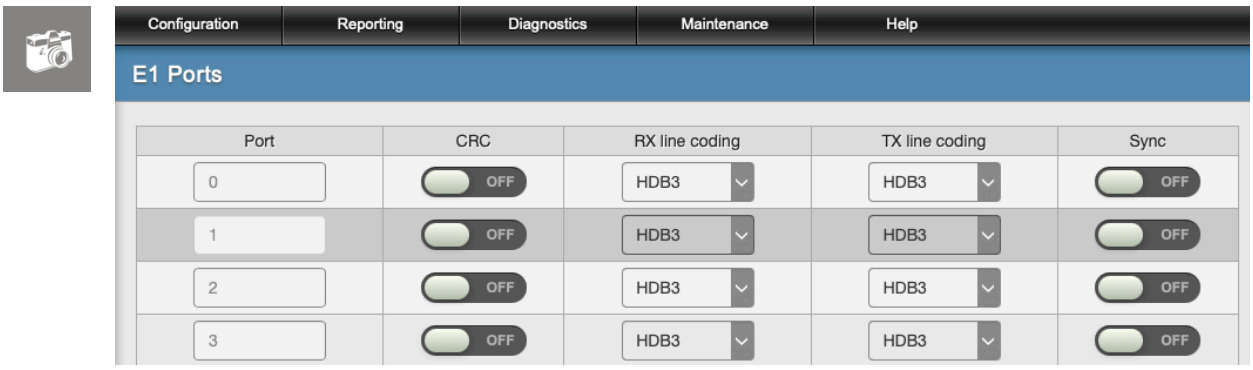
Figure 87. E1 Links



E1 Port Number - select PCM-0 (it should match the port that was selected when the V5.2 configuration had been created) (Figure 84). Use the entire E1 port - set the start and end time slots to 1 and 31. Select the Signaling Profile that was configured during the previous step (Figure 85). Do not select direction of unconditional routing since we'll create a strict V5.2-SIP binding later when V5.2 subscribers are created.

To complete the TDM configuration, synchronization and parameters of E1 ports should be configured (E1 gateway port uses the internal generator and the TDM remote node will be synchronized from E1 gateway port) (Figure 88).

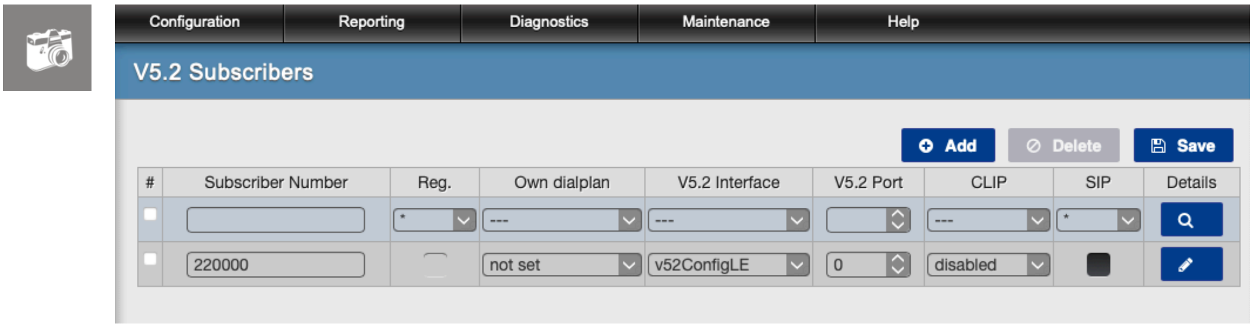
Figure 88. E1 Port Configuration



Next step. Add 100 V5.2 subscribers to the configuration and bind them to SIP.

Create subscribers (Configuration → V5.2 Subscribers) (Figure 89).

Figure 89. V5.2 Subscribers

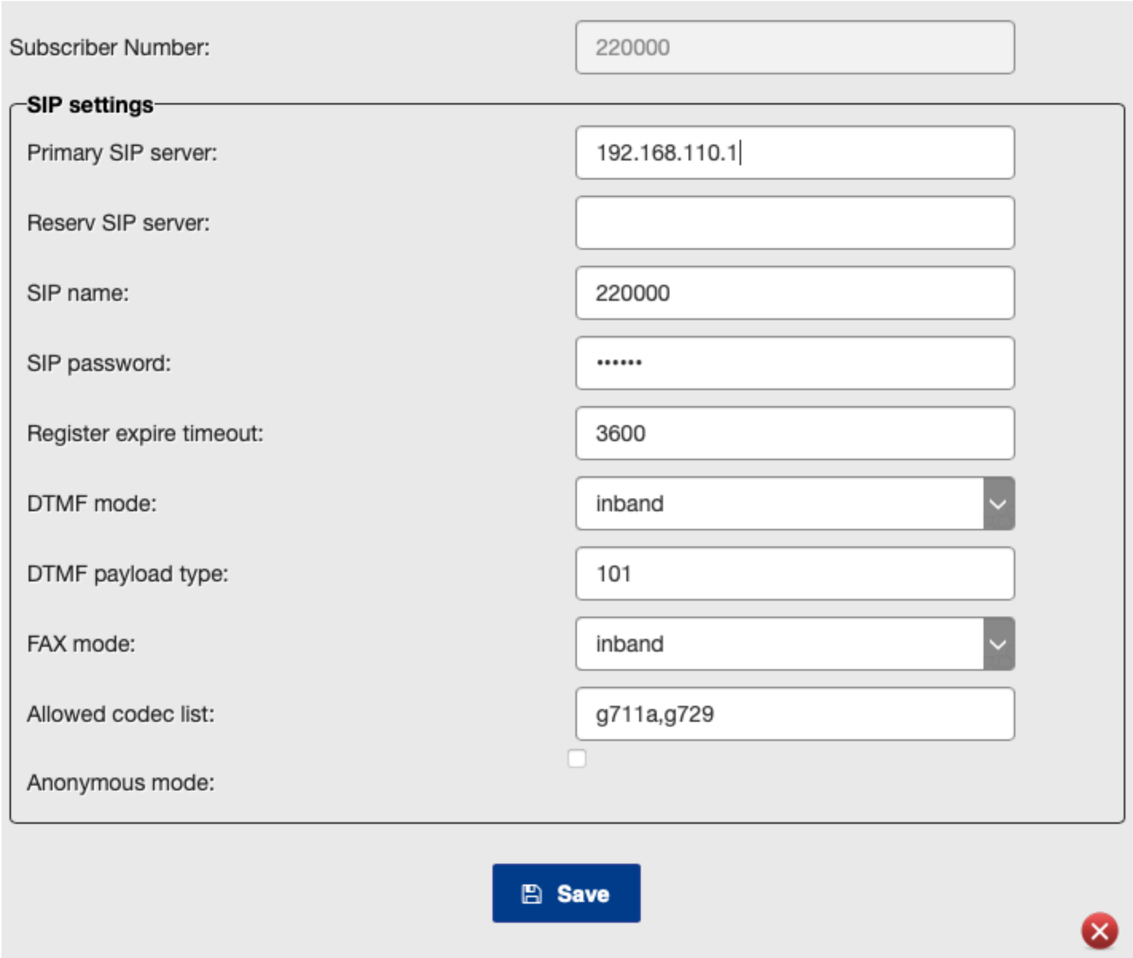


Create the first subscriber with 220000 number. Enter the number, select V5.2 Interface (that was created earlier - Figure 84), set V5.2 Port and check SIP option. This results in binding the created SIP subscriber to SIP. Press Save.

The extended parameters of the added record are automatically set to default values. SIP name and SIP password match the Subscriber Number by default.

Press the Details button (). Enter Main SIP Server Address and modify the remaining fields if required (Figure 90). Save the applied changes.

Figure 90. SIP Subscribers - extended parameters



Adding a second subscriber.

When there is at least one subscriber then you can make the process of entering the next subscriber faster: highlight the subscriber with the highest number (put the cursor in any field of this subscriber) and the subscriber's line will be highlighted in dark grey. Then press the Add button. The added subscriber will have the Interface V5.2 value matching the same value of the highlighted subscriber and the value of V5.2 Port will increase by 1 (next in sequence). The remaining parameters such as SIP server addresses will be added automatically also.

If the additional parameters that are added by default match the configuration then there is no need to open Details window.

Enter all subscribers in the same way and save them.

If it is required to enter very many subscribers then you can make the process faster by creating a list of all the subscribers with any available table editor and importing it into the gateway configuration.

The creation process is identical to the same process described in the previous example (7.2) with the only exclusion that you will need to process the V5.2 subs_v52.csv subscriber file rather than the SIP subs_sip.csv subscriber file.

This will complete configuring the gateway. Reload the gateway to apply all settings properly.

Warning! It is required to configure V5.2 interface of the AN subscriber remote node.

8. Specifications

Table 38. Specifications

Physical Interfaces	4, 8 or 16 Telephony Interfaces (E1 G.703 balanced and unbalanced cables with 75Ω and 120Ω. Unbalanced interfaces require an RJ-45 to Coax BNC adapter.); Dual RJ-45 Copper Ethernet (10/100/1000); One Console (RS-232).
IP/VoIP Protocols	SIP (RFC 3261), SIP-T, SIP-I, RTP, RTCP, TCP, and UDP.
Network Protocols	IP, TCP, UDP, FTP, TFTP, RTP, RTCP, ARP, ICMP, NTP, Telnet, IEEE 802.1Q, and IEEE 802.1P.
PSTN Signaling Protocols	SS7 (ITU-T Q.700 series), 24bit/14bit PC, ISUP/TUP, ISDN-PRI (ITU-T Q.931,Q.921), CAS R2 Q.400-Q.490, CAS DTMF BellCore TR-TSV-002275, and V5.2 (ETS 300 347-1).
Voice Codecs	G.711 u-Law and A-Law, G.711 Appendix 1, G.723.1 and G.723.1 Annex A, G.729 Annex A and Annex B, G.726, GSM, ARM, ILBC, DTMF, adaptive jitter buffer, VAD, CNG, G.165, G.168, silence suppression/echo cancellation, tone scheme in accordance with standard, and ITU v.152.
WAN Interfaces	Dual/Two 10/100 Ethernet, static IP address.
QoS and Security	802.1Q, 802.1P. Filtering by IP addresses, 802.1Q.
Applications	Out-band Proxy, Reregister, Voice/Modem/Fax Calls, Prefix Number Routing, Dial Plan, Number Modification, 1+1 SS Protection
Configuration and Management	Web GUI, Telnet, SNMP v1/v2c Traps, updating software over FTP/TFTP, storing/recovering configuration, checking E1 ports/signaling status.
Power	Single/Dual-redundant 220 VAC or 48/60 VDC Input. 28W max Power Consumption.
Operating Temperature	0°C to +60°C, humidity: 0% to 90% (non- condensing).
Dimensions	485 mm x 286 mm x 44 mm.