

# TAU-32M.IP

Operation Manual

Version 2.13.1 (November 2015)

**Universal Network Terminal**

**Firmware version: 2.13.1**

**Linux version: 291 Tue Aug 25 10:15:38 NOV 2015**

**Firmware version: v10\_23\_03\_15**

**BPU version: v20150810** date: 2015 Aug 10 time 13:33:40

**Factory default IP address 192.168.1.2**

**username: admin**

**password: rootpasswd**

| Document version | Issue date        | Revisions  |
|------------------|-------------------|--|
| Version 2.13.1   | 10 November 2015  | Added: <ul style="list-style-type: none"> <li>– MTU configuration</li> <li>– Configuration of ports for access via Telnet, SSH, HTTPS</li> <li>– Changeover to redundant proxy for INVITE request type only</li> </ul>   |
| Version 2.13     | 28 January 2015   | Additions: <ul style="list-style-type: none"> <li>– Fixed incorrect RTP/SAVP processing</li> <li>– Fixed call release on receiving response 500 to SIP INFO request</li> <li>– Fixed 'accept' header incorrect utilization in SIP replies</li> <li>– Fixed display issues with SIP headers in web interface log</li> <li>– Fixed autocompletion in login and password fields on web interface pages</li> <li>– Russian language for Web interface</li> <li>– Disabled '%' character utilization in 'username', 'hot number', 'alt number', 'cf_no_answer', 'cf_busy', 'cf_unconditional', 'cf_out_of_service' parameters</li> <li>– Changed response used to switch to redundant proxy from 408 to 505</li> <li>– Username and password entry field length extended to 50 characters in SIP profile</li> <li>– Implemented MWI service for SIP</li> <li>– Implemented option to change static/dynamic address obtaining method in factory configuration</li> <li>– Implemented option to change factory MAC address</li> <li>– Updated time zone files for NTP due to reversion to standard time in Russian Federation</li> <li>– Implemented early channel forwarding for calls to call group</li> <li>– Maximum number of simultaneous web interface users increased to 4</li> <li>– Implemented SIP domain transmission in request URI</li> <li>– Implemented 'Wait answer timeout' utilization for incoming calls</li> <li>– DHCP option 82 formation</li> </ul> |
| Version 2.12     | 18 September 2014 | Additions: <ul style="list-style-type: none"> <li>– 'alert-info' header processing</li> <li>– 'multihoming' mode support</li> <li>– Support for operation behind NAT—STUN, PublicIP</li> <li>– CgPN/CdPN modification support for outgoing communications</li> <li>– RURI check depth is optional for incoming communications</li> <li>– Support for configuration and firmware update via FTP/HTTP/HTTPS</li> <li>– Local log</li> <li>– Support for configurable daylight saving</li> <li>– Configuration of Speed/Duplex modes on switch ports</li> </ul>   |
| Version 1.17     | 20 June 2014      | Additions: <ul style="list-style-type: none"> <li>– SNMP Support for new block reason (Receiver offhook)</li> <li>– WEB 'regexp' dialplan support:</li> <li>– Processing of ABCD characters in 'regexp' routing plan</li> <li>– Option to replace S-timer with L-timer for rules with varying character count in 'regexp' routing plan</li> <li>– SNMP, WEB Number of call groups increased up to 32</li> <li>– Number of FXO call groups increased up to 32</li> <li>– H323 'status enquiry' message processing</li> </ul>  |

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| Version 1.16 | 19 May 2014      | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– APPENDIX K. Processing of INFO requests containing application/broadsoft and application/sscc and used for Supplementary Services</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.1 Network</li> <li>– 5.1.1.5.1 SNMP Monitoring</li> <li>– 5.1.2.1 Basic Configuration (Main)</li> <li>– 5.1.2.2 SIP/H323 Profiles</li> <li>– 5.1.2.4 Configuration of Subscriber Ports (Ports)</li> <li>– 7.3 3-way Conference</li> <li>– 9.1 Configuration file—CFG.YAML</li> </ul>   |
| Version 1.15 | 15 January 2013  | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.7. Firewall configuration</li> <li>– 5.1.4.5. IMS SS Status Monitoring</li> <li>– 5.1.4.6. Serial Group Registration Status Monitoring</li> <li>– 5.1.4.7. FXO Group Registration Status Monitoring</li> <li>– 5.4. SUPERVISOR Access</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– TAU-32M.IP<br/>Firmware Update Instructions, version 2.9.0</li> <li>– 3.1.1 General Guidelines</li> <li>– 5.1. TAU-32M.IP Configuration via WEB Interface. Administrator Access</li> <li>– 5.1.1.1. Network</li> <li>– 5.1.1.5.1. SNMP Monitoring</li> <li>– 5.1.1.6. Syslog Protocol Configuration</li> <li>– 5.1.2.1. Basic Configuration (Main)</li> <li>– 5.1.2.2.3. SIP Custom Parameters (SIP Custom)</li> <li>– 5.1.2.2.4. Codecs Configuration (SIP Custom)</li> <li>– 5.1.2.4. Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.2.8. Configuration of FXO Groups (FXO groups)</li> <li>– 5.1.4.2. Board Parameter Status Monitoring (Status)</li> <li>– 5.1.6.4. Encryption Features (Security)</li> <li>– 5.1.6.6. Changing Access Passwords using Web Configurator (Password)</li> <li>– 7. Supplementary Services Usage</li> <li>– 9.1. Configuration file—CFG.YAML</li> <li>– Appendix 3. Calculation of phone line length</li> </ul> |
| Version 1.14 | 2 September 2013 | <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– Appendix 3. Calculation of phone line length</li> </ul>   |
| Version 1.13 | 24 July 2013     | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– TAU-32M.IP<br/>Firmware Update Instructions, version 2.6.0</li> <li>– 5.1.5.2 Route</li> <li>– 5.1.5.3 ARP</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.1 Network</li> <li>– 5.1.1.5.1 SNMP Monitoring</li> <li>– 5.1.2.1 Basic Configuration (Main)</li> <li>– 5.1.2.2 SIP/H323 Profiles</li> <li>– 5.1.2.4 Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.4.2 Board Parameter Status Monitoring (Status)</li> <li>– 5.1.5.1 Service Status Monitoring (Device info)</li> <li>– 9. Description of Configuration Files</li> </ul>   |

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| Version 1.12 | 10 May 2013      | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.2.10 'Distinctive Ring' Service Configuration</li> <li>– Appendix G. Automatic Configuration Procedure and Gateway Firmware Version Check</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.1 Network</li> <li>– 5.1.2.4 Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.2.1 Basic Configuration (Main)</li> <li>– 5.1.2.2.4 Codecs Configuration</li> <li>– 6.1 Basic Commands</li> <li>– 9 Description of Configuration Files</li> </ul>  |
| Version 1.11 | 21 December 2012 | <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 2.7 'F' Function Button Operation</li> <li>– 6.5 Reset to Factory Defaults</li> </ul>   |
| Version 1.10 | 18 December 2012 | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– Appendix I. Device firewall configuration—iptables</li> </ul>  |
| Version 1.9  | 27 November 2012 | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.2.2 SIP/H323 Profiles</li> <li>– APPENDIX F. Configuration Example for PABX Connected via FXO Lines</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 2.1 Purpose</li> <li>– 2.2 Typical Application Diagrams</li> <li>– 2.4 Main specifications</li> <li>– 5.1 TAU-32M.IP Configuration via WEB Interface. Administrator Access.</li> <li>– 5.1.1.1 Network</li> <li>– 5.1.1.5 SNMP Protocol Configuration</li> <li>– 5.1.2.2.1 SIP Common Parameters (SIP Common)</li> <li>– 5.1.2.2.2 H.323 Protocol</li> <li>– 5.1.2.2.3 SIP Custom Parameters (SIP Custom)</li> <li>– 5.1.2.2.4 Codecs Configuration (Codecs)</li> <li>– 5.1.2.2.5 Routing and Pickup Code Configuration (Dialplan)</li> <li>– 5.1.2.4 Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.2.6 Configuration of Supplementary Service Codes (Suppl. Service Codes)</li> <li>– 5.1.2.8 Configuration of FXO Groups (FXO groups)</li> <li>– 5.1.4.1 Subscriber Port Monitoring (Port)</li> <li>– 5.1.4.4 Supplementary Service Status Monitoring (Suppl.Service)</li> <li>– 5.1.6.1 Firmware Upgrade (Firmware upgrade)</li> <li>– 5.2 TAU-32M.IP Configuration via Web Interface. Operator Access</li> <li>– 9 Description of Configuration Files</li> <li>– Appendix C. General Device Setup / Configuration Procedure</li> <li>– Appendix E. Example of PABX Configuration with TAU-32M.IP</li> </ul> |
| Version 1.8  | 06 July 2012     | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.2.8 Configuration of Supplementary Service Codes (Suppl. Service Codes)</li> <li>– 5.1.2.11 Configuration of FXO Groups</li> <li>– 5.1.4.4 Supplementary Service Status Monitoring (Suppl.Service)</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.1 Network</li> <li>– 5.1.1.6.2 Device Configuration via SNMP</li> <li>– 5.1.2.1 Basic Configuration (Main)</li> <li>– 5.1.2.6 Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.2.9 Routing and Pickup Code Configuration (Dialplan)</li> <li>– 5.1.4.1 Subscriber Port Monitoring (Port)</li> <li>– 5.1.6.2 Download/Upload Configuration (Backup/Restore)</li> </ul>   |

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| Version 1.7 | 13 March 2012     | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.4.3 Switch Port Status Monitoring</li> <li>– 6.2.3 Port-specific Statistics</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.3 Static Routes</li> <li>– 5.1.1.4 Local DNS (Hosts)</li> <li>– 5.1.2.5 Codecs Configuration (Codecs)</li> <li>– 5.1.2.6 Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.6.4 Encryption Features (Security)</li> <li>– 6.1 Basic Commands</li> <li>– 9 Description of Configuration Files</li> </ul>   |
| Version 1.6 | 07 December 2011  | <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 3.2.7 SFP Transceiver Installation and Removal</li> <li>– 5.1.2.8 Configuration of Supplementary Service Codes—DVO</li> <li>– 5.1.2.9.4 Configuration of Routing Rules with Regular Expressions</li> <li>– 9. Description of Configuration Files</li> <li>– Appendix F. Automatic Configuration Procedure and Gateway Firmware Version Check</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 2.6 LED indication</li> <li>– 5.1.1.1 Network</li> <li>– 5.1.1.5 SNMP Protocol Configuration</li> <li>– 5.1.1.6.2 Device Configuration via SNMP</li> <li>– 5.1.2.3.1 'SIP' Submenu</li> <li>– 5.1.2.5 Codecs Configuration (Codecs)</li> <li>– 5.1.2.6 Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.2.9 Routing and Pickup Code Configuration (Dialplan)</li> <li>– 5.1.4.1 Subscriber Port Monitoring (Port)</li> <li>– 5.1.5 System Information</li> </ul> |
| Version 1.5 | 12 September 2011 | <p>Added chapter:</p> <ul style="list-style-type: none"> <li>– 5.1.2.7 Simultaneous Call Limits</li> </ul> <p>Changes in chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.6.1 SNMP Monitoring</li> <li>– 5.1.2.3.1 'SIP' Submenu</li> <li>– 6.2.1 Command Line Mode</li> </ul>  |
| Version 1.4 | 29 August 2011    | <p>Web Management Interface Update:</p> <ul style="list-style-type: none"> <li>– 5.1.1.1 Network—added 'Get GW via DHCP' setting</li> <li>– 5.1.1.2 VLAN conf—added 'DHCP for VLAN' and 'Get GW via DHCP' settings</li> <li>– 5.1.2.3 SIP Protocol Configuration—added 'Home server test' control using 'REGISTER', 'Registration retry interval', 'Inbound', 'Remote ringback', 'SIP hash URI' messages</li> <li>– 5.1.2.5 Codecs Configuration—added settings for 'RTCP timer', 'TCP control period', 'G.711A RFC3108', and 'G.711U RFC3108'</li> <li>– 5.1.2.6 Configuration of Subscriber Ports—added 'Play music on hold' setting</li> </ul> <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.1.6.1 SNMP Monitoring</li> <li>– 5.1.1.6.2 Device Configuration via SNMP</li> </ul>   |
| Version 1.3 | 12 April 2011     | <p>Web Management Interface Update:</p> <ul style="list-style-type: none"> <li>– 5.1.1.6 SNMP Protocol Configuration</li> <li>– 5.1.2.3 SIP Protocol Configuration</li> <li>– 5.1.2.4 TCP/IP Configuration</li> <li>– 5.1.2.6 Configuration of Subscriber Ports (Ports)</li> <li>– 5.1.3.2 802.1Q</li> <li>– 5.1.4.1 Subscriber Port Monitoring (Port)</li> </ul> <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 5.1.3.3. QOS &amp; Bandwidth control</li> <li>– 7.3 3-way Conference</li> </ul>   |

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| Version 1.2 | 26 January 2011   | <p>Web Management Interface Update:</p> <ul style="list-style-type: none"> <li>– 5.1.1.12 Codecs Configuration Added 'Jitter Buffer Configuration' subsection</li> <li>– 5.1.1.13 Configuration of Subscriber Ports</li> </ul>   |
| Version 1.1 | 13 January 2011   | <p>Web Management Interface Update.</p> <p>Added chapters:</p> <ul style="list-style-type: none"> <li>– 3.1 Safety instructions</li> <li>– 4 General Switch Operation Guidelines</li> <li>– 5.1.2.3.2 Configuration of Internal Switching for SIP-proxy Connection Loss</li> <li>– 5.1.2.3.3 Configuration of Internal Switching for SIP-proxy Connection Loss</li> <li>– 5.1.2.7.2 Configuration of Prefix with Varying Number Count</li> <li>– 5.1.2.7.3 Pickup Code Configuration</li> <li>– 5.1.2.9 Pickup Group Configuration (Pickup Groups)</li> <li>– 5.1.3.1.2 Tracing, Network Traffic Mirroring</li> <li>– 5.1.6.5 Encryption Features</li> <li>– 8 Connection Establishment Algorithms</li> <li>– Appendix D. Example of Switch Configuration using VLAN</li> <li>– Appendix E. Example of PABX Configuration with TAU-32M.IP</li> </ul> |
| Version 1.0 | 27 September 2010 | First issue.   |

## TAU-32M.IP

### FIRMWARE UPDATE INSTRUCTIONS, UPGRADING TO THE LATEST VERSION



**New firmware versions utilize new update method and new firmware file format. Exercise caution and follow instructions while performing firmware update.**

Steps necessary to update the firmware to the latest version (2.13.1).



Verify that the name of the firmware **version 2.13.1** file is ***firmware.img32***.

**If the current switch firmware version is 1.9.0 or earlier (including legacy versions with 4-digit version number):**

1. Download the firmware **version 1.11.4** file:  
[http://eltex.nsk.ru/upload/files/tau32m/tau32m/1\\_11\\_4/firmware.tar.gz](http://eltex.nsk.ru/upload/files/tau32m/tau32m/1_11_4/firmware.tar.gz)
2. Download the firmware **version 2.13.1** file:  
<http://eltex.nsk.ru/upload/files/tau32m/2.13.1/firmware.img32m>
3. In the device web configurator, select 'Service' menu, and then 'Firmware upgrade' tab. In the dialog window, browse to the firmware **version 1.11.4** file. To do this, click the 'Browse' button in 'Universal firmware upgrade' section and then click 'Upgrade firmware' button. Verify that the name of the file is **firmware.tar.gz**.
4. Wait until firmware upgrade process is completed; after that TAU-32M.IP should automatically restart.
5. When the device is rebooted, open web configurator and click 'Save' button shown next to any configuration menu item, e.g. on 'Network' tab.
6. When configuration is saved, update firmware in a similar manner using the firmware **version 2.13.1** file. Verify that the name of the firmware version 2.13.1 file is **firmware.img32m**.

**If the current switch firmware version is 2.1.0 through 2.1.4:**

1. Download the firmware **version 2.1.4** file:  
[http://eltex.nsk.ru/upload/files/tau32m/tau32m/2\\_1\\_4/firmware.tar.gz](http://eltex.nsk.ru/upload/files/tau32m/tau32m/2_1_4/firmware.tar.gz)
2. Download the firmware **version 2.13.1** file:  
<http://eltex.nsk.ru/upload/files/tau32m/2.13.1/firmware.img32m>
3. In the device web configurator, select 'Service' menu, and then 'Firmware upgrade' tab. In the dialog window, browse to the firmware **version 2.1.4** file. To do this, click the 'Browse' button in 'Universal firmware upgrade' section and then click 'Upgrade firmware' button. Verify that the name of the file is **firmware.tar.gz**.
4. When the device is rebooted, update firmware in a similar manner using the firmware **version 2.13.1** file. Verify that the name of the firmware version 2.13.1 file is **firmware.img32m**.

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**If the current switch firmware version is 2.2.0 through 2.5.0:**

1. Download the firmware **version 2.5.0** file:  
[http://eltex.nsk.ru/upload/files/tau32m/tau32m/2\\_5\\_0/firmware.img32m](http://eltex.nsk.ru/upload/files/tau32m/tau32m/2_5_0/firmware.img32m)
2. Download the firmware **version 2.13.1** file:  
<http://eltex.nsk.ru/upload/files/tau32m/2.13.1/firmware.img32m>
3. In the device web configurator, select 'Service' menu, and then 'Firmware upgrade' tab. In the dialog window, browse to the firmware **version 2.5.0** file. To do this, click the 'Browse' button in 'Universal firmware upgrade' section and then click 'Upgrade firmware' button. Verify that the name of the file is **firmware.img32m**.
4. When the device is rebooted, update firmware in a similar manner using the firmware **version 2.13.1** file. Verify that the name of the firmware version 2.13.1 file is **firmware.img32m**.

**If the current switch firmware version is 2.5.0 or later:**

1. Download the firmware **version 2.13.1** file:  
<http://eltex.nsk.ru/upload/files/tau32m/2.13.1/firmware.img32m>
2. Before firmware update, restart the device to free up cache in RAM.
3. In the equipment web interface, select 'Service' menu, and then 'Firmware upgrade' tab. In the dialog window, browse to the firmware **version 2.13.1** file. To do this, click the 'Browse' button in 'Universal firmware upgrade' section and then click 'Upgrade firmware' button. Verify that the name of the file is **firmware.img32m**.











**If you are unable to update the firmware using web configurator or other means, use the alternative firmware update method. The alternative firmware update method is described in *Appendix B* of this Operation Manual. Necessary files are located in reserve\_soft.zip archive.**



## TARGET AUDIENCE

This operation manual is intended for technical personnel that performs switch installation, configuration, monitoring, and maintenance using web configurator. Qualified technical personnel should be familiar with the operation basics of TCP/IP & UDP/IP protocol stacks and Ethernet networks design concepts.

## SYMBOLS

| Symbol  | Description  |
|---|--|
| <b>Bold font face</b>   | Notes, warnings, chapter headings, titles, table titles are written in bold.                                   |
| <i>Calibri Italic</i>   | Important information is written in Calibri Italic.  |
| Courier New   | Command entry examples, command execution results and program output data are written in Courier New semibold. |
| <KEY>   | Keyboard keys are written in upper-case and enclosed in angle brackets.  |
|    | Analogue phone unit icon   |
|    | Gatekeeper icon  |
|    | TAU-32M.IP Universal Network Terminal icon   |
|    | MES3124F Ethernet switch icon  |
|  | Softswitch ECSS-10 hardware-software switch icon   |
|  | Digital subscriber PBX icon  |
|  | Network Connection icon  |
|  | Optical transmission medium  |

### Notes and warnings



Notes contain important information, tips or recommendations on device operation and setup.



Warnings are used to inform the user about harmful situations for the device and the user alike, which could cause malfunction or data loss.

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## 1. INTRODUCTION

TAU-32M.IP Universal Network Terminal allows to connect analogue phone units to packet-based data networks accessible through copper-wire or optical Ethernet interfaces.

TAU-32M.IP could be used as a subscriber access point utilizing SIP/SIP-T and H.323 protocols and provides a perfect telephone communication solution for underpopulated areas, offices, dwellings and geographically dispersed facilities.

This operation manual describes intended use, key specifications, configuration, and firmware update methods for TAU-32M.IP network terminal (hereinafter the "device").

## 2. PRODUCT DESCRIPTION

### 2.1. Purpose

TAU-32M.IP is a subscriber VoIP gateway with integrated Layer 2 Ethernet switch that uses copper-wire and optical Gigabit Ethernet interfaces to establish connection to provider's IP network. In order to transfer data via IP networks, device converts analogue voice signals to digital data packets.

TAU-32M.IP has a modular design. Device features the basic unit and can accommodate up to 4 various submodules. When utilized at the stage of transition from TDM to NGN networks, the terminal allows you to keep the existing network infrastructure and analogue subscribers to access IP networks.

#### *Interface types:*

- Analogue ports are executed as extension modules. Each extension module contains 8 ports:
  - TAU32M-M8S submodule: 8 x FXS ports.
  - TAU32M-M8O submodule: 8 x FXO ports.
  - TAU32M-M4S4O submodule: 4 x FXS ports—4 x FXO ports<sup>1</sup>.
  - TAU32M-M4S4OR submodule: 4 x FXS ports—4 x FXO ports with the ability to reroute FXS lines to redundant PBX lines (FXO) in case of failure/power loss<sup>1</sup>.
- Three Ethernet 10/100/1000BaseT electrical interfaces.
- Two Mini-Gbic (SFP) Ethernet 1000BaseX optical interfaces.

#### *Device features:*

- Integrated Layer 2 Ethernet switch.
- VoIP protocols: H.323, SIP/SIP-T<sup>2</sup>.
- Static address and DHCP support.
- Echo cancellation (G.168 recommendation).
- Packet loss concealment (PLC).
- Voice activity detector (VAD).
- Silence suppression.
- DTMF tone detection and generation.
- Fax transmission: upspeed/pass-through; T.38 UDP Real-Time Fax.
- Cisco NSE support.
- V.152 support.
- Flexible numbering scheme.
- Operation with and without external gatekeeper (H.323/RAS).
- Operation with multiple SIP servers in various SIP profiles.
- Support for VoIP operation in the switch in case of SIP server connection loss.
- Active session support for SIP protocol operations through NAT.

---

<sup>1</sup> TAU32M-M4S4O, TAU32M-M4S4OR submodules are available on specific request only.

<sup>2</sup> SIP-T only supports basic call establishment, additional types of service are not implemented



- 
- Transmission of cpc-rus subscriber category via SIP protocol.
  - Configuration file download/upload via FTP/FTPS, TFTP, HTTP/HTTPS.
  - Firmware update via TFTP, HTTP/HTTPS.
  - Automatic configuration and firmware update via FTP, TFTP, HTTP/HTTPS.
  - STP support.
  - LLDP support.
  - iptables network-level firewall.
  - STUN support.
  - Service (simulation service) management using IMS (3GPP TS 24.623).
  - Remote monitoring, configuration and setup:
    - Web interface.
    - telnet.
    - SSH.
    - SNMP.
    - TR-069.
    - User authentication with RADIUS server.
  - Supported supplementary services:
    - Call Hold/Retrieve.
    - Call Transfer.
    - Call Waiting notification.
    - Call Forward Busy
    - Call Forward No Answer.
    - Call Forward Unconditional.
    - Call Forward Out Of Service.
    - Caller ID with ETSI FSK type 1, type 2.
    - Caller ID in DTMF format.
    - 'Russian Caller ID'.
    - Calling without Caller ID broadcasting.
    - Hotline/warmline.
    - Call Hunt.
    - Call PickUp.
    - 3-way conference (local or using conference server).
    - Voice message waiting indicator—MWI.
    - Do Not Disturb.

*SIP, supported recommendations:*

- RFC 3261 SIP 2.0;
- RFC 3262 SIP PRACK;
- RFC 4566 Session Description Protocol (SDP);
- RFC 3263 Locating SIP servers for DNS lookup SRV and A records;

- RFC 3264 SDP Offer/Answer Model;
- RFC 3265 SIP Notify;
- RFC 3311 SIP Update;
- RFC 3515 SIP REFER;
- RFC 3891 SIP Replaces Header;
- RFC 3892 SIP Referred-By Mechanism;
- RFC 4028 SIP Session Timer;
- RFC 2976 SIP INFO Method;
- RFC 2833 RTP Payload for DTMF Digits, Flash event;
- RFC 3108 Attributes ecan and silenceSupp in SDP;
- RFC 4579 SIP. Call Control - Conferencing for User Agents;
- RFC 3372 SIP for Telephones (SIP-T);
- RFC 3398 ISUP/SIP Mapping;
- RFC 3204 MIME Media Types for ISUP and QSIG (ISUP support);
- RFC 3361 DHCP Option 120;
- SIP OPTIONS Keep-Alive (SIP Busy Out);
- NAT support.

## 2.2. Typical Application Diagrams

This manual covers the following TAU-32M.IP connection methods:

**1. Subscriber access point** In this case, the device acts as a gateway between analogue phone units and remote PBX, see Fig. 1. Gateway subscriber ports are registered at the software switch—Softswitch. Supplementary services in this method are provided by the software switch.

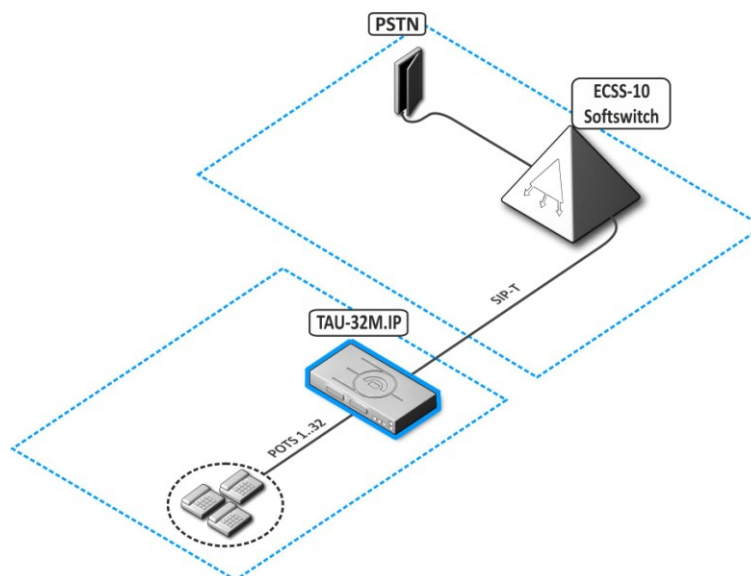


Fig. 1—TAU-32M.IP subscriber access point

**2. Distributed mini-PABX mode** In this case, the device acts as a mini-PABX that is able to access other gateways (TAU-32M.IP, TAU-72M.IP, etc.) and Softswitch using SIP/H.323 protocols. The device allows for unassisted processing of supplementary services, call routing, see Fig. 2.

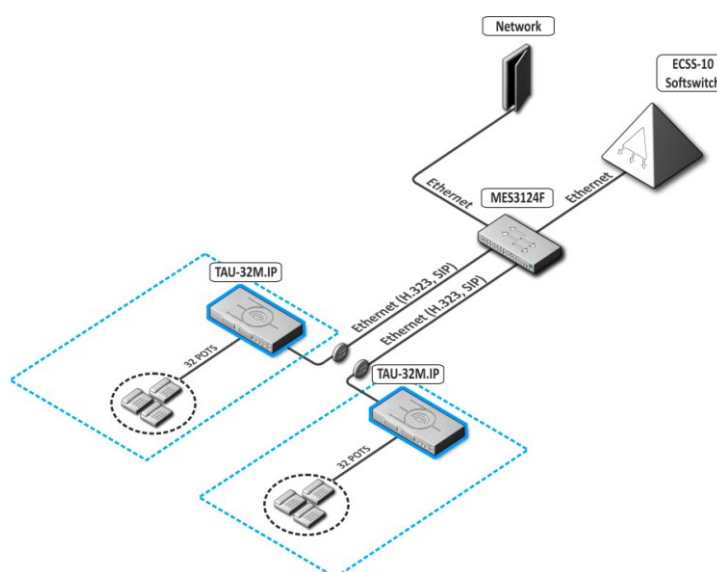


Fig. 2—TAU-32M.IP distributed mini-PABX

**3. TAU-32M.IP operation in point-to-point mode** In this case, the device acts as a line stretcher through IP network.

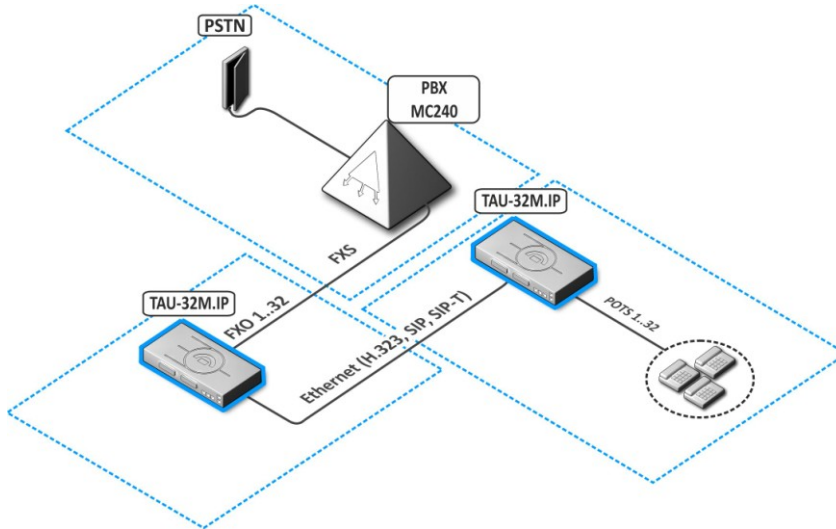


Fig. 3—TAU-32M.IP operation in point-to-point mode

**4. TAU-32M.IP operation in analogue trunk mode** In this case, the device acts as a gateway between double-wire analogue PBX lines and subscriber IP network devices. This configuration allows you to use additional SIP servers, gatekeepers or similar flexible switching equipment.

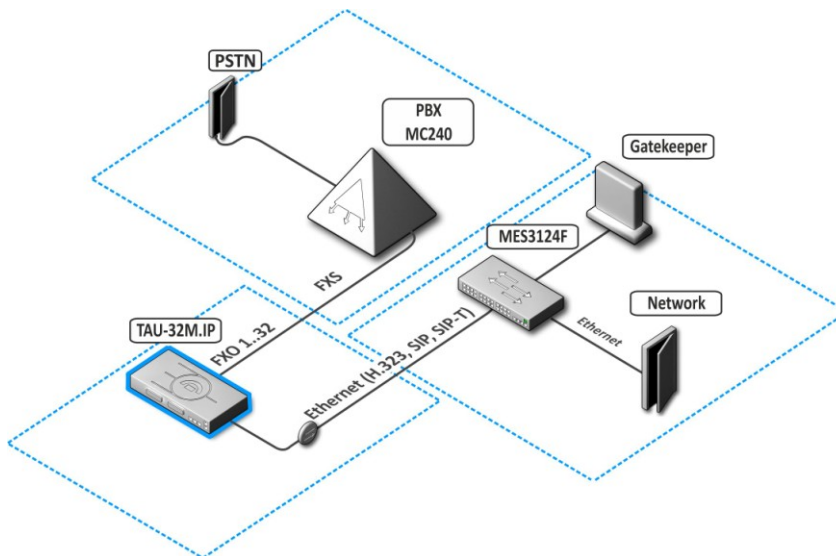


Fig. 4—TAU-32M.IP operation in analogue trunk mode

**5. FXS+FXO operation** In this case, the device acts as a gateway/mini-PABX with the ability to reroute FXS lines to redundant PBX lines (FXO) in case of failure/power loss 1.

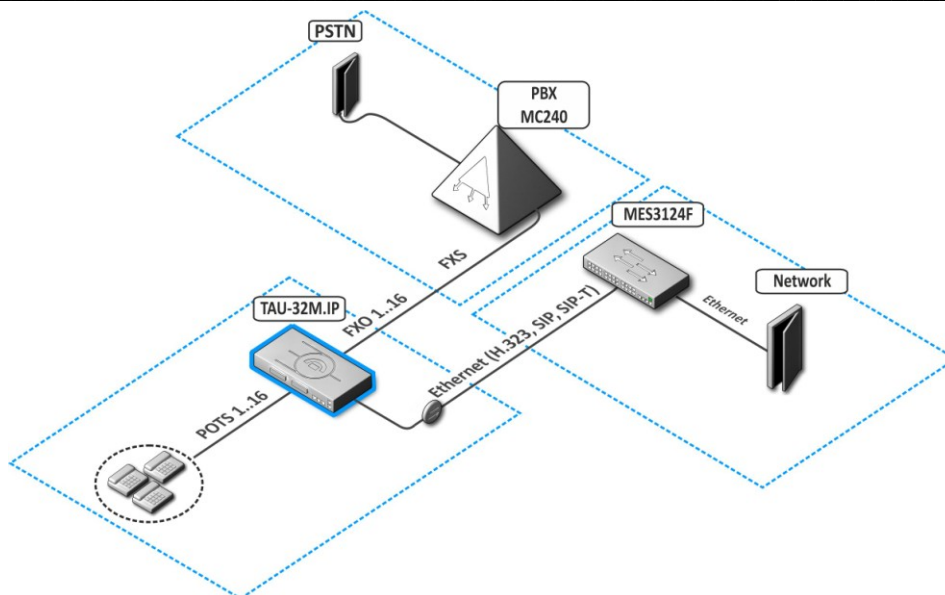


Fig. 5—TAU-32M.IP operation as FXS+FXO

### 2.3. Product Design and Operating Principle

Subscriber voice signals are served to audio codecs of subscriber units, where they are encoded using one of the selected standards, and then sent as digital packets to the controller via internal backbone. In addition to voice signals, digital packets contain control and interaction signals.

Fig. 6 shows TAU-32M.IP functional chart.

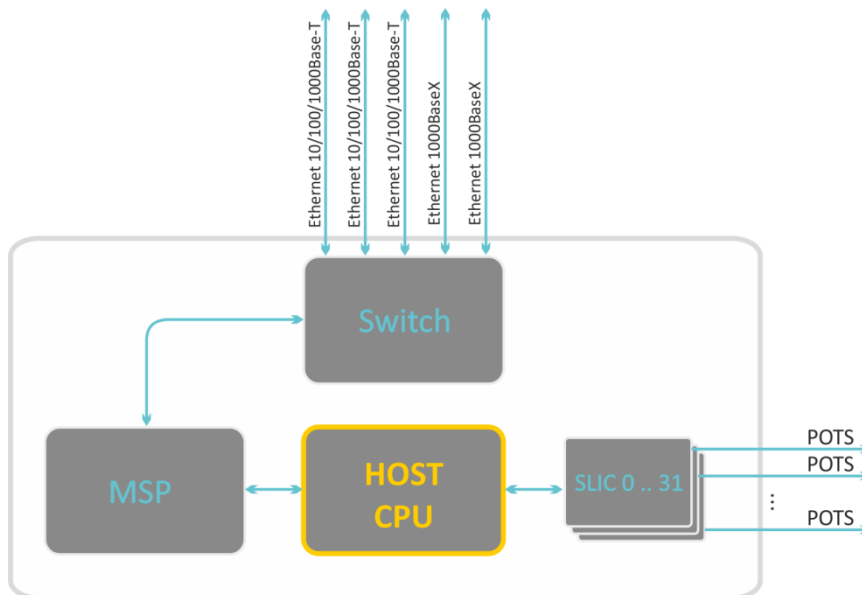


Fig. 6—TAU-32M.IP functional chart

## 2.4. Main Specifications

Table 1 lists main specifications of the terminal.

Table 1—Main specifications of the terminal

| <b>Protocols and Standards</b>   |  |
|--|--|
| Protocol stack   | ITU-T H.323 v3/v4/v5   |
| Communication protocol for session initiation, monitoring and cancellation | SIP  |
| Voice codecs   | G.729AB<br>G.711(A/U)<br>G.723.1 (6.3 Kbps, 5.3 Kbps)<br>G.726 32 kbps (for SIP only)  |
| Fax support  | T.38 UDP Real-Time Fax<br>pass-through (G.711A/U)  |
| Modem support  | V.152<br>CISCO NSE   |
| Voice standards  | VAD (voice activity detector)<br>AEC (echo cancellation, G.168 recommendation)<br>CNG(comfort noise generator)   |
| <b>Interfaces</b>  |  |
| Ethernet optical state   | 1000BaseX E Mini-Gbic (SFP) port x 2   |
| Ethernet electrical interface  | 10/100/1000Base-T Ethernet RJ-45 port x 3  |
| Analogue interfaces  | No. of ports: 32;<br>Port types: FXS, FXO;<br>Programmable unit parameters<br>Loop resistance: up to 2.8kΩ<br>FSK (ITU-T V.23, Bell 202), DTMF, Russian Caller ID<br>Caller ID detection (for FXO ports)<br>Dialling reception: pulse/frequency (DTMF)<br>Comprehensive protective circuit (current and voltage) |
| <b>Network and Configuration</b>   |  |
| Connection types   | Static IP, DHCP client   |
| Management   | WEB, RS-232 console, Telnet, SSH   |
| Security   | User name and password verification, HTTPS, FTPS   |
| <b>Physical specifications and ambient conditions</b>                      |  |
| Power voltage  | AC: 220V, 50Hz   |

|                             |  |
|-----------------------------|--|
|                             | DC: <ul style="list-style-type: none"> <li>• Rated voltage: -48V</li> <li>• Permissible voltage: -38V to -72V</li> </ul> |
| Power consumption           | Up to 50W (for 32 simultaneously active units)   |
| Operating temperature range | From 0 to 40°C   |
| Relative humidity           | Up to 80 %   |
| Dimensions (W x H x D)      | 430x191x44mm, 19" form-factor, 1U size   |
| Weight                      | 3.2kg  |

## 2.5. Design

TAU-32M.IP network terminal has a metal case available for 19" form-factor rack-mount 1U shelf installation.

The front panel of the device is shown in Fig. 7.

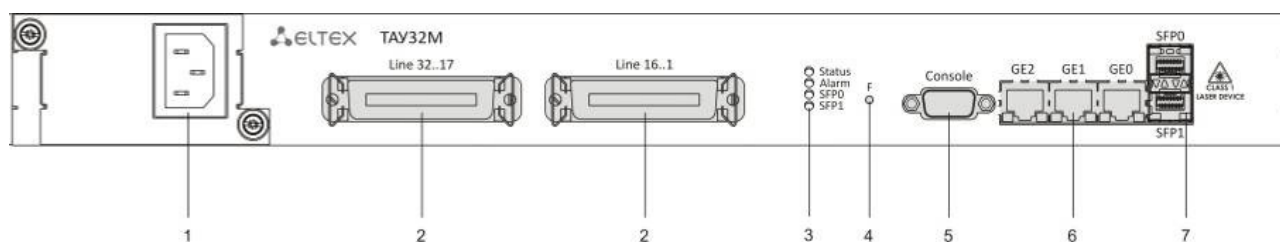


Fig. 7—TAU-32M.IP front panel appearance

Connectors, LEDs and controls located on the front panel of the device are listed in Table 2.

Table 2—Description of connectors, LEDs, and controls located on the front panel

| No | Front panel elements               | Description   |
|----|------------------------------------|---|
| 1  | 48VDC/~150 .. 250VAC, 50Hz, max 1A | Connector for DC power supply with rated voltage -48VDC (permissible voltage from -38VDC to -72VDC) or AC power supply with voltage 150–250VAC, 50Hz (depending on the order) |
| 2  | Line 1..16/ Line 17..32            | CENC-36M connectors (for contact pin assignment, see Appendix A)  |
| 3  | Status                             | Device operation indicator  |
|    | Alarm                              | Alarm indicator Shows three types of alarms   |
|    | SFP0                               | SFP0 optical interface activity indicator   |
|    | SFP1                               | SFP1 optical interface activity indicator   |
| 4  | F                                  | Function button   |
| 5  | Console                            | RS-232 console port for local control of the device   |

|   |             |   |
|---|-------------|---|
| 6 | GE0/GE1/GE2 | 3 x RJ-45 ports of Ethernet 10/100/1000 Base-T interfaces   |
| 7 | SFP0/SFP1   | 2 chassis for optical SFP modules of 1000Base-X Gigabit uplink interface used for IP network connection |

The rear panel of the device is shown in Fig. 8.



Fig. 8—TAU-32M.IP rear panel appearance

An earthing bolt is located on the rear panel of the device.

For contact pin assignment, see Appendix A.

## 2.6. LED Indication

**Alarm, Status, SFP0, SFP1** LEDs located on the front panel indicate the current state of the device. Table 3 lists possible states of the LEDs.

Table 3—Device status LED indication

| Indicator                                  | Indicator State                                  | Device State  |
|--|--|---|
| Status                                     | off  | operating system loaded, board type identified  |
|  | solid yellow                                     | device initialization in progress, subscriber ports are not initialized yet,<br>address is not obtained through DHCP (if dynamic address obtaining method is enabled) |
|  | solid green                                      | subscriber ports are initialized, device is in operation  |
| Alarm                                      | solid on   | Warning—port blocking, operating system loading   |
|  | flashes slowly (once per second)                 | Error (failure)—module sensor failure (SFP module installed, but there is no link)  |
|  | flashes rapidly (once per 200ms)                 | Fatal (critical failure)—connection of the main application to subscriber ports is lost   |
| simultaneous operation<br>Alarm and Status | Alarm—solid red<br>Status—solid red              | operating system is not loaded  |
|  | Alarm—flashes once per 200ms<br>Status—solid red | main application is not running   |



|           |   |                         |
|-----------|---|-------------------------|
|           | Alarm—flashes once per 200ms<br>Status—flashes red, yellow, and green | factory Safemode        |
| SFP0/SFP1 | solid green   | optical link is present |
|           | off   | no optical link         |

Ethernet interface state is shown by 1000/100 socket built-in LED indicators.

Table 4—Light indication of Ethernet 10/100/1000 interfaces

| LED/Status                |                          | LED/Status  |
|---------------------------|--------------------------|---|
| Yellow LED<br>10/100/1000 | Green LED<br>10/100/1000 |   |
| solid on                  | solid on                 | Port operates in 1000Base-T mode, data transfer is inactive |
| solid on                  | flashes                  | Port operates in 1000Base-T mode, data transfer is active   |
| off                       | solid on                 | Port operates in 10/100Base-TX, data transfer is inactive   |
| off                       | flashes                  | Port operates in 10/100Base-TX, data transfer is active     |

## 2.7. 'F' Function Button Operation

To reboot the operating device, press and hold 'F' button located on the front panel of the device for 5 seconds. **Alarm** LED will become solid red. Also, this button allows you to reset the device to factory settings when you forget or don't know device IP address or password used for login. To do this, turn the device on while holding 'F' button. Hold the button until **Status** LED begins to flash yellow, green and red alternatively, and **Alarm** LED becomes solid red. After that, you can access the device by IP address **192.168.1.2**. When connecting with web configurator, default password for **admin** user is **rootpasswd**.

For detailed description of the factory reset procedure, see Section 6.6 'Reset to factory settings in protected mode'.

## 2.8. Delivery Package

TAU-32M.IP standard delivery package includes:

- TAU-32M.IP universal network terminal.
- CENC-36M connector—2pcs (if 18 pairs of UTP CAT5E cable were not included in order).
- Power cable (if equipped with 220V power supply).
- PVS cable (if equipped with 48V power supply).
- RS-232 DB9(F) – DB9(F) cable for console port connection.
- Earthing cable (if equipped with 48V power supply).
- A mounting set for 19" rack.

- Operation manual.
- Declaration of conformity.

If ordered, delivery package may also include:

- 1000Base-T/Mini-Gbic (SFP) optical interface—2pcs.
- 18 pairs of UTP CAT5E cable with CENC-36M connectors—1pcs.

### 3. INSTALLATION ORDER AND SAFETY MEASURES

This section describes safety measures and installation of the equipment into a rack and connection to a power supply.

#### 3.1. Safety instructions

##### 3.1.1. General Guidelines

Any operations with the equipment should comply to the Safety Rules for Operation of Customers' Electrical Installations.



**Operations with the equipment should be carried out only by personnel authorised in accordance with the safety requirements.**

- 1 Before operating the device, all engineers should undergo special training.
- 2 The device should be connected only to properly functioning supplementary equipment.
- 3 TAU-32M.IP terminal could be permanently used provided the following requirements are met:
  - Ambient temperature from 0 to +40°C.
  - Relative humidity up to 80% at +25°C.
  - Atmosphere pressure from  $6,0 \times 10^4$  to  $10,7 \times 10^4$  Pa (from 450 to 800 mm Hg).
- 4 The device should be not be exposed to mechanical shock, vibration, smoke, dust, water, and chemicals.
- 5 To avoid components overheating which may result in device malfunction, do not block air vents or place objects on the equipment.
- 6 Electrostatic Discharge Safety Measures In order to avoid failures caused by electrostatic discharge, we strongly recommend to wear ESD belt, shoes and wrist strap which prevent electrostatic charge accumulation (for wrist strap, make sure that it has a secure fit against the skin) and connect the cable to earthing prior to operation.

##### 3.1.2. Electrical Safety Requirements

- 1 Prior to connecting the device to a power source, ensure that the equipment case is grounded with an earth bonding point. The earthing wire should be securely connected to the earth bonding point. The resistance between the earth bonding point and earthing busbar should be less than 0.1 Ohm.
- 2 PC and measurement instruments should be grounded prior to connection to the device. The potential difference between the equipment case and the cases of the instruments should be less than 1 V.
- 3 Prior to turning the device on, ensure that all cables are undamaged and securely connected.

- 4 Make sure the device is off, when installing or removing the case.
- 5 Power supply modules should be changed only when the device is off. Follow the replacement order given in Section 3.6.
- 6 Power supply modules installation and removal should be conducted only when the device is powered off according to the procedure described in Section 3.2.5.

## 3.2. TAU-32M.IP Installation

Check the device for visible mechanical damage before installing and turning it on. In case of any damage, stop the installation, fill in a corresponding document and contact your supplier.

If the device was exposed to low temperatures for a long time before installation, leave it for 2 hours at ambient temperature prior to operation. If the device was exposed to high humidity for a long time, leave it for at least 12 hours in normal conditions prior to turning it on.

Mount the device. The device is intended to be installed into 19" rack using the mounting set or mounted on the horizontally oriented perforated shelf.



**If the device is being installed into a closed non-ventilated cabinet with volume less than 180l per device, device performance will not exceed 0.8 Erlang per subscriber unit.**

Ground the case of the device after installation. This should be done prior to connecting the device to the power supply. An insulated multiconductor wire should be used for earthing. The device grounding and the earthing wire section should comply with Electric Installation Code. The earth bonding point is located at the right bottom corner of the side panel, see Fig. 8.

### 3.2.1. Startup sequence

Connect subscriber lines, optical and electrical Ethernet cables to corresponding switch connectors.



**For subscriber unit overvoltage protection, the linear side of the distribution cross should be equipped with triple pole arresters with trip voltage 230V. We recommend to use KRONE arresters—MK, 230V—with a thermal protection spring.**

Connect the power supply cable to the device. Depending on the provided sources, the device could be powered from grounded power outlet 220/110VAC, 50/60Hz, or from -38...-72VDC power supply. To connect the device to 220VAC electrical network, use the cable provided with the delivery package. To connect the device to DC power supply, use the cable with cross-section not less than 1mm<sup>2</sup>.

If a PC is supposed to be connected to TAU-32M.IP console port, connect TAU-32M.IP COM port to PC COM port. PC should be powered off and grounded at the same point with the switch.

Ensure that all cables are undamaged and securely connected.

Turn the device on and check the front panel LEDs to make sure the terminal is in normal operating conditions.

### 3.2.2. Support brackets mounting

The delivery package includes support brackets for rack installation and mounting screws to fix the device case on the brackets.

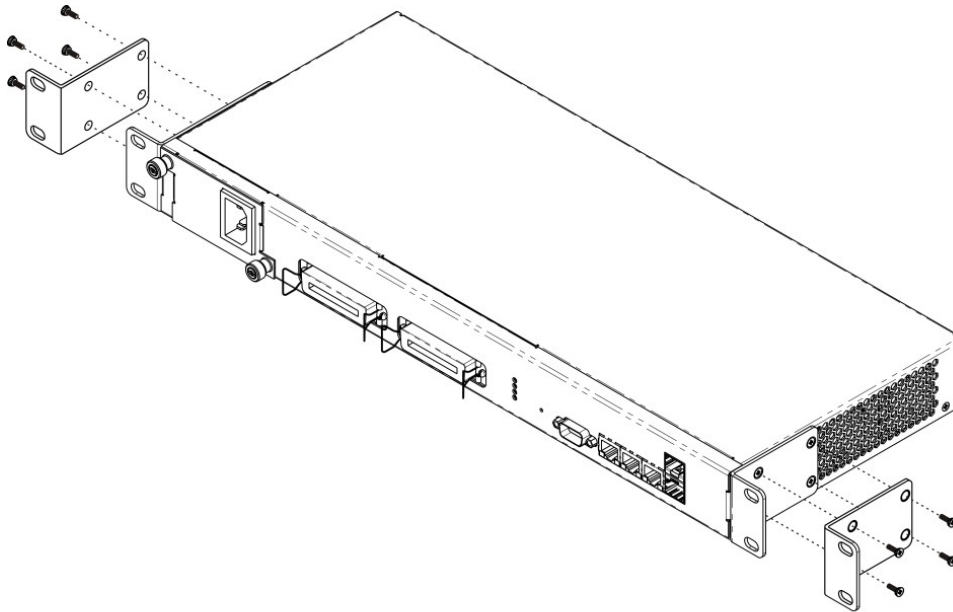


Fig. 9—Support brackets mounting

To install the support brackets:

- 1 Align four mounting holes in the support bracket with the corresponding holes in the side panel of the device, see Fig. 9.
- 2 Use a screwdriver to screw the support bracket to the case.

Repeat steps 1 and 2 for the second support bracket.

### 3.2.3. Device rack installation

To install the device to the rack:

- 1 Attach the device to the vertical guides of the rack.
- 2 Align mounting holes in the support bracket with the corresponding holes in the rack guides. Use the holes of the same level on both sides of the guides to ensure the device horizontal installation.
- 3 Use a screwdriver to screw the device to the rack.

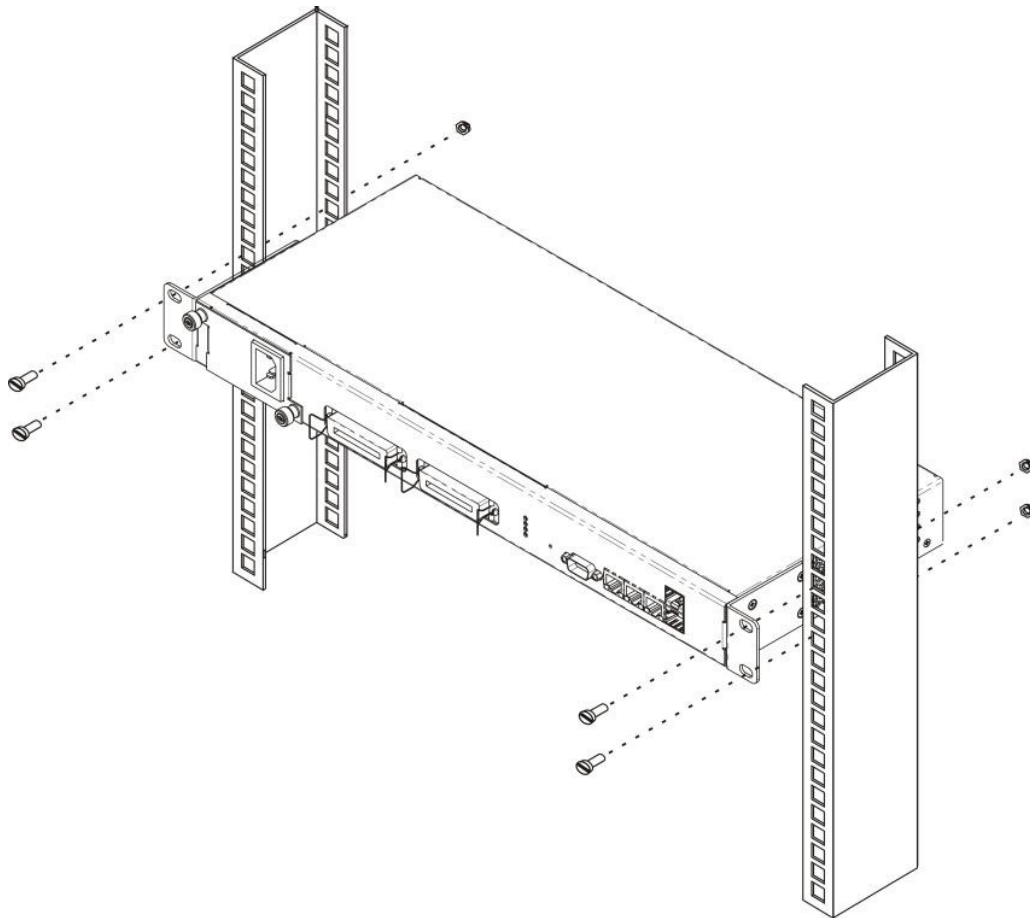


Fig. 10—Device rack installation

### 3.2.4. Power Supply Module Installation

Depending on power supply requirements, TAU-32M.IP could be equipped with either 220VAC 50Hz power supply module, or a DC power supply module with rated voltage -48V (permissible voltage -38...-72VDC). The location of the power supply module is shown in Fig. 11.



**Power supply module installation and removal should be conducted when the device is de-energized.**

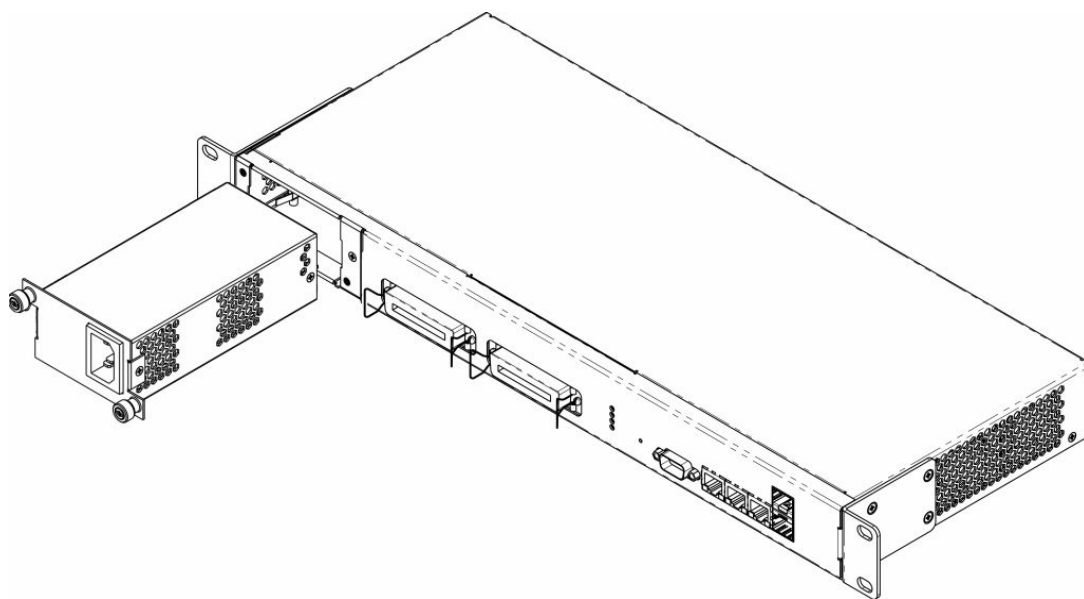


Fig. 11—Power module installation

To install the power supply module:

- 1 Install the module to the opening as shown in Fig. 11.
- 2 Screw the module to the case.
- 3 Follow the instructions in *Section 3.2* to power on.

To replace power supply modules:

- 1 Check the voltage on module.
- 2 If the voltage is present, disconnect the power supply.
- 3 Remove the module.

### 3.2.5. Submodule Installation

The device has a modular design and can accommodate up to 4 various submodules.

- TAU32M-M8S submodule: 8 x FXS ports.
- TAU32M-M8O submodule: 8 x FXO ports.
- TAU32M-M4S4O submodule: 4 x FXS ports—4 x FXO ports.
- TAU32M-M4S4OR submodule: 4 x FXS ports—4 x FXO ports with the ability to reroute FXS lines to redundant PBX lines (FXO) in case of failure/power loss.

In minimal configuration, the device contains a single submodule installed in Position 1, Fig. 12.

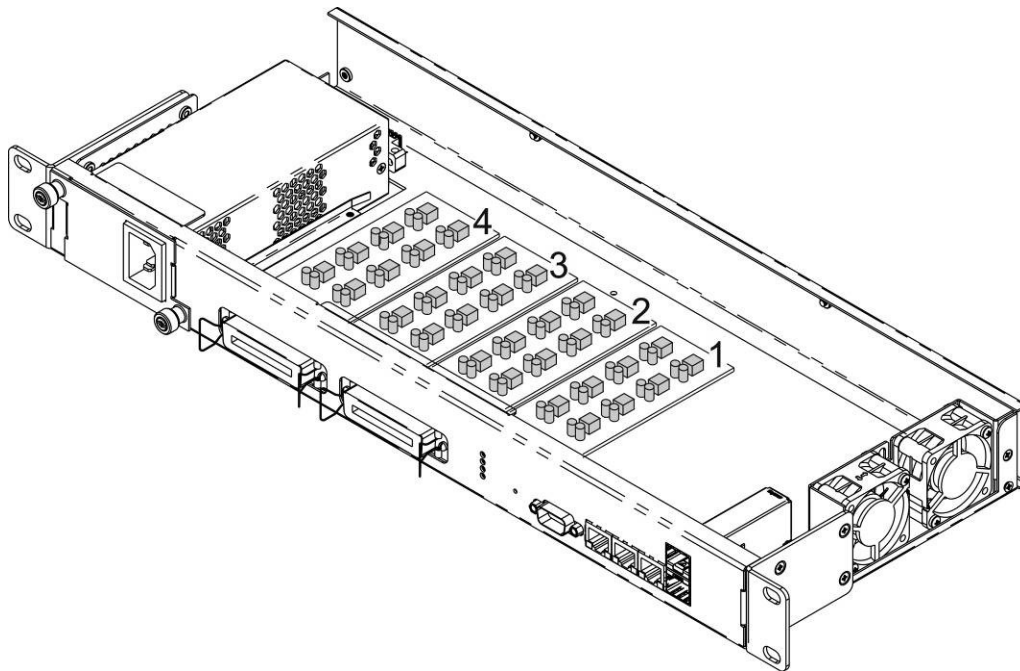


Fig. 12—TAU-32M.IP submodule location

TAU32M submodule installation order:

- 1 Check if the device is energized.
- 2 If the voltage is present, disconnect the power supply.
- 3 Install module into a free Position—2, 3, or 4 (see Fig. 12).

### 3.2.6. Fan Location

Side panels of the device have air vents for heat removal. Two fans are installed on the inside of the right-side panel, see Fig. 12.

Air flows in through the perforated left-side panel, circulates through all internal components, cools them down, and then is removed with the help of ventilators located on the perforated right-side panel. Other device



panels are free from air vents in order to keep up the necessary internal airflow pressure.



**Do not block air vents. This may cause components overheating which may result in terminal malfunction.**

### 3.2.7. SFP transceiver installation and removal



**Optical modules can be installed when the terminal is turned on or off.**

Transceiver installation:

1. Insert the top SFP module into a slot with its open side down, and the bottom SFP module with its open side up.
2. Press the module until it fits with a click.

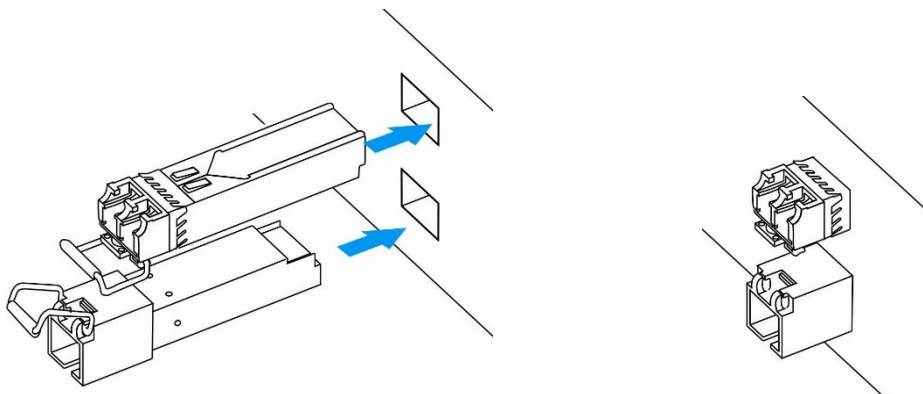


Fig 13—SFP transceiver installation

Transceiver removal:

1. Unlock the module's latch.
2. Remove the module from the slot.

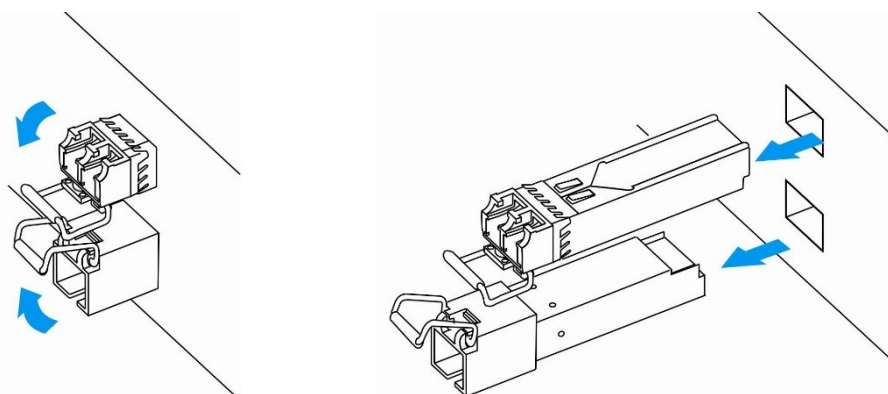


Fig. 14—Opening SFT transceiver latch and SFT transceiver removal

---

## 4. GENERAL SWITCH OPERATION GUIDELINES

The easiest way to configure and monitor the device is to use the web interface, so we recommend you to use it for these purposes.

In order to prevent an unauthorized access to the device, we recommend to set password for access via telnet and ssh (password is not defined by default), and also change administrator, operator and non-privileged user passwords to access the web interface. For setting password for access via telnet and ssh, see Section 6.4 'Setting password for 'admin' user' For setting password for access via web interface, see Section 5.1.6.5 'Password'. We recommend to write down and store defined passwords in a safe place, inaccessible by intruders.

In order to prevent device configuration data loss, e.g. after reset to factory settings, we recommend making configuration backup copies and storing them on a PC each time significant changes are made.

## 5. DEVICE CONFIGURATION

You can connect to the device using the following methods: via web interface, via telnet/ssh protocols, or using RS-232 cable.

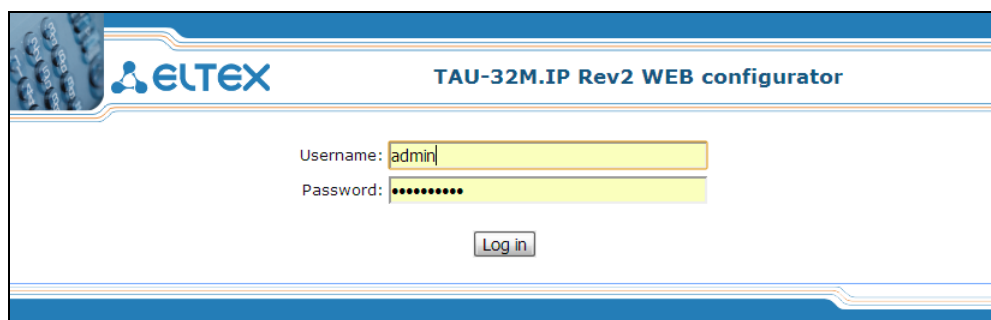
### 5.1. TAU-32M.IP Configuration via WEB Interface. Administrator Access.

To configure the device, establish connection in the *web browser*, e.g. Firefox, Internet Explorer, etc. Enter device IP address into address bar of web browser.



**TAU-32M.IP factory default IP address—192.168.1.2, network mask—255.255.255.0**

After entering IP address the device will request username and password.




**Initial startup username: *admin*, password: *rootpasswd*.**



**For security reasons, duration of authorized access session is limited for 20 minutes, i.e. if you are inactive after establishing connection to the device interface for the stated amount of time, the gateway will be forced to end the session. This restriction is not effective in cases when you leave 'Monitoring' or 'System info' pages open, as these pages perform periodic polling of the device data.**



**Up to 4 users may connect to the device web interface simultaneously.**

The following menu will appear on the administrator's terminal: To prevent unauthorized access to device in the future, it's recommended to change password (see Section 5.1.6.6).



**In all tabs, 'Save' button stores configuration into the non-volatile (flash) memory of the device.**

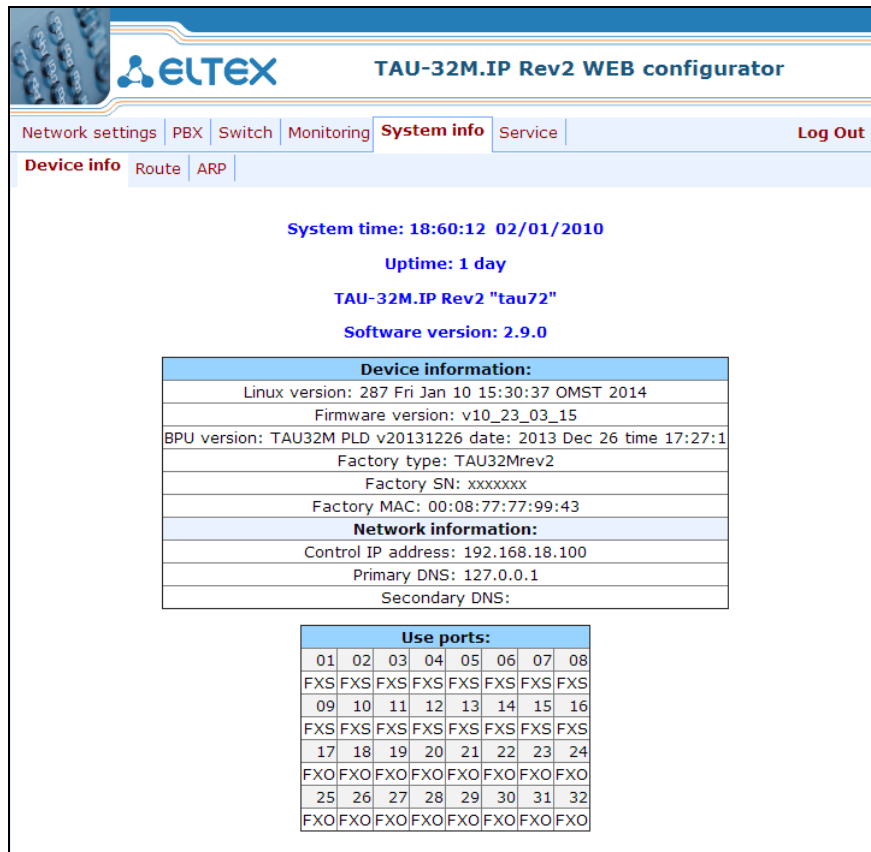


Table 5 lists description of configuration menu windows.

Table 5—Description of configuration menu, administrator access

| Menu (en)                | Description  |
|--------------------------|--|
| <b>Network settings</b>  | Adjustment of the device network settings              |
| <i>Network</i>           | Configuration of network settings                      |
| <i>VLAN conf</i>         | VLAN configuration                                     |
| <i>Route</i>             | Static route configuration for WAN and VLAN interfaces |
| <i>Hosts</i>             | Local DNS server configuration                         |
| <i>SNMP</i>              | SNMP agent configuration                               |
| <i>Syslog</i>            | Syslog server configuration                            |
| <i>Firewall</i>          | Configuration of denied/allowed IP server addresses    |
| <i>NTP</i>               | NTP configuration                                      |
| <i>ACS</i>               | TR-069 monitoring and management protocol settings     |
| <i>Uatoupdate</i>        | Automatic update configuration                         |
| <b>PBX</b>               | VoIP (Voice over IP) configuration                     |
| <i>Main</i>              | Device basic settings                                  |
| <i>SIP/H323 Profiles</i> | Configuration of SIP/H323 profiles                     |
| <i>SIP Common</i>        | SIP common settings                                    |
| <i>H323</i>              | H323 protocol settings (works in profile 1 only)       |

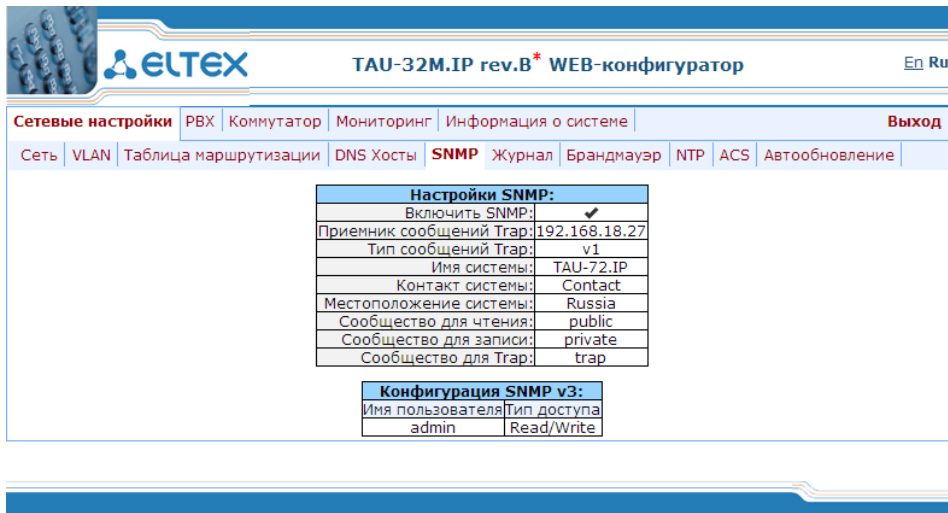
|                                    |  |
|------------------------------------|--|
| <i>Profile 1..8</i>                | Profile configuration  |
| <i>SIP Custom</i>                  | SIP custom settings for a profile  |
| <i>Codecs</i>                      | Codec settings for a profile   |
| <i>Dialplan</i>                    | Routing settings for a profile   |
| <i>Alert-Info</i>                  | Configuration of a distinctive ring, formed by Alert-Info value                              |
| <i>TCP/IP</i>                      | Configuration of network port range for various protocols                                    |
| <i>Ports</i>                       | Configuration of device subscriber ports and subscriber profiles                             |
| <i>Call limits</i>                 | Configuration of simultaneous call limits  |
| <i>Suppl. Service Codes</i>        | Configuration of supplementary service codes   |
| <i>Serial groups</i>               | Configuration of serial groups   |
| <i>FXO groups</i>                  | Configuration of FXO groups  |
| <i>PickUp groups</i>               | Configuration of pickup groups   |
| <i>Distinctive ring</i>            | 'Distinctive ring' service administration  |
| <i>Modifiers</i>                   | Configuration of number modifiers  |
| <b>Switch</b>                      | Configuration of switch settings   |
| <i>Switch ports settings</i>       | Configuration of integrated Ethernet switch ports  |
| <i>802.1q</i>                      | Configuration of packet routing rules for switch operation in 802.1q mode                    |
| <i>QoS &amp; Bandwidth control</i> | Quality of service functions and bandwidth limits configuration                              |
| <b>Monitoring</b>                  | Device monitoring  |
| <i>Port</i>                        | Device subscriber ports status information   |
| <i>Status</i>                      | Gateway hardware platform status information—voltages, temperature sensors, fans, SFP data   |
| <i>Switch</i>                      | Switch port state monitoring   |
| <i>Suppl. Service</i>              | Information on the current status of supplementary services on subscriber port               |
| <b>System info</b>                 | System information   |
| <i>Device info</i>                 | View the device and network settings information   |
| <i>Route</i>                       | Routing table configuration  |
| <i>ARP</i>                         | ARP table configuration  |
| <b>Service</b>                     | Firmware update, configuration file operations, rebooting device, setting/changing passwords |
| <i>Firmware upgrade</i>            | Firmware update of subscriber units  |
| <i>Backup/Restore</i>              | Download/upload configuration files to/from PC   |
| <i>Reboot</i>                      | Rebooting device   |
| <i>Security</i>                    | Encryption feature   |
| <i>MOH</i>                         | Download/upload audio file for call hold service   |
| <i>Password</i>                    | Management of passwords used to access the device via web interface                          |
| <b>Logout</b>                      | Finish the device administration session for the current user                                |

## Web Configurator Language

Web configurator allows you to select from two interface languages: 'Russian (Ru)' and 'English (En)'.

Firmware version '-ru' postfix means that the default interface language is Russian, and '-en' postfix stands for English. To change the interface language, select the respective link in the web configurator header bar (on the right side).

Example of web configurator menu in Russian:

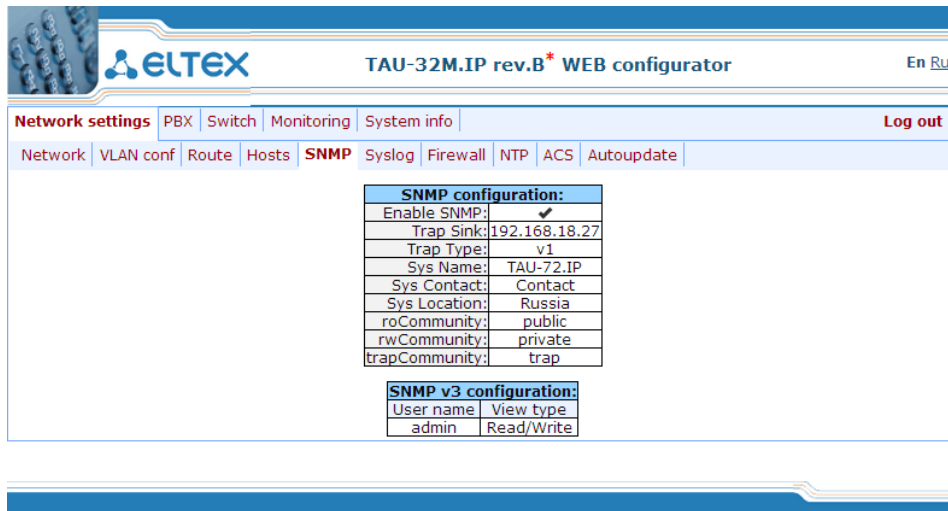


| Настройки SNMP:          |                                     |
|--------------------------|-------------------------------------|
| Включить SNMP:           | <input checked="" type="checkbox"/> |
| Приемник сообщений Trap: | 192.168.18.27                       |
| Тип сообщений Trap:      | v1                                  |
| Имя системы:             | TAU-72.IP                           |
| Контакт системы:         | Contact                             |
| Местоположение системы:  | Russia                              |
| Сообщество для чтения:   | public                              |
| Сообщество для записи:   | private                             |
| Сообщество для Trap:     | trap                                |

| Конфигурация SNMP v3: |             |
|-----------------------|-------------|
| Имя пользователя      | Тип доступа |
| admin                 | Read/Write  |

Example of web configurator menu in English:



| SNMP configuration: |                                     |
|---------------------|-------------------------------------|
| Enable SNMP:        | <input checked="" type="checkbox"/> |
| Trap Sink:          | 192.168.18.27                       |
| Trap Type:          | v1                                  |
| Sys Name:           | TAU-72.IP                           |
| Sys Contact:        | Contact                             |
| Sys Location:       | Russia                              |
| roCommunity:        | public                              |
| rwCommunity:        | private                             |
| trapCommunity:      | trap                                |

| SNMP v3 configuration: |            |
|------------------------|------------|
| User name              | View type  |
| admin                  | Read/Write |

## Indication of Changes in Web Configurator

Web configurator supports indication of configuration changes, that is shown in the header bar of configuration interface (TAU-32M.IP WEB configurator.) Table 6 lists indicator states ('\*' character in the header bar of configuration interface).

Table 6—Indicator state \*

| Indicator State          | Description   |
|--------------------------|---|
| * character is red       | Changes has been made to the configuration, but it has not been saved to flash memory yet   |
| * character is not shown | No changes has been made to the configuration<br>Changes has been successfully saved to flash memory<br>Gateway IP address has been changed |



**When network settings are changed, web service on the device restarts, and when the connection is established using new address, '\*' character will not be shown, but the configuration will still contain changes that are not saved to the flash memory.**

### 5.1.1. Network settings

In **Network settings** menu, you can define network settings of the device.

#### 5.1.1.1. Network

In the '*Network*' submenu, you may specify the device name, IP address, subnet mask, network broadcast address, DNS server address, device access rules, etc.



**You don't have to reboot the gateway in order to apply network settings. When applying settings, all current calls will be terminated.**

- **DHCP** is a protocol that allows to automatically obtain IP address and other settings required for operation in TCP/IP network. Allows the gateway to obtain all necessary network settings from DHCP server.
- **SNMP** is a simple network management protocol. Allows the gateway to send real-time messages on occurred failures to controlling SNMP manager. Also, gateway SNMP agent supports monitoring of gateway sensors' status on request from SNMP manager.
- **DNS** is a protocol that allows to obtain domain information. Allows the gateway to obtain IP address of the communicating device by its network name (hostname). It may be necessary, e.g. when specifying hosts in the routing plan or using network name of the SIP server as its address.
- **TELNET** is a protocol that allows to establish mechanisms of control over the network. Allows you to remotely connect to the gateway from a computer for configuration and management purposes. For TELNET protocol operation, the data transfer process is not encrypted.
- **SSH** is a protocol that allows to establish remote control over the network. Serves the similar purpose as TELNET protocol, but unlike the latter provides encryption of the transferred data.
- **LLDP (Link Layer Discovery Protocol)** is a data-link level protocol that allows network equipment to notify the neighbouring devices located in a local network on their capabilities and gather such notifications from the neighbouring devices.
- **STP (Spanning Tree Protocol)** is a network protocol that allows to eliminate loops in the arbitrary Ethernet network topology, containing one or multiple network bridges connected with redundant links.
- **TR-069** is a technical specification that defines the Internet protocol for management of network equipment—CWMP (CPE WAN Management Protocol). The protocol allows for comprehensive device configuration, software updates, reading device information (software version, model, serial number, etc.), complete configuration file downloading/uploading, remote device restart (TR-069, TR-098, TR-104 specifications are supported).
- **STUN** is a network protocol that allows the client located behind a network address translation server (NAT) to discover its external IP address.



**TAU-32M.IP WEB configurator**
En

**Network settings**
PBX
Switch
Monitoring
System info
Service
Log out

**Network**
VLAN conf
Route
Hosts
SNMP
Syslog
Firewall
NTP
ACS
Autoupdate

Attention! Changing of these parameters will lead to aborting of all calls!

| Network Settings:             |                                     |
|-------------------------------|-------------------------------------|
| Enable DHCP:                  | <input checked="" type="checkbox"/> |
| Get GW via DHCP:              | <input type="checkbox"/>            |
| Default gateway:              |                                     |
| Primary DNS IP:               | 127.0.0.1                           |
| Secondary DNS IP:             |                                     |
| DHCP Options:                 |                                     |
| Alternative option 60 enable: | <input type="checkbox"/>            |
| Alternative option 60 value:  |                                     |
| Option 82. Agent Circuit ID:  |                                     |
| Option 82. Agent Remote ID:   |                                     |
| WAN Settings:                 |                                     |
| IP address:                   | 192.168.1.32                        |
| Netmask:                      | 255.255.255.0                       |
| Broadcast:                    |                                     |
| MTU:                          | 1500                                |
| Services:                     |                                     |
| Enable TELNET:                | <input checked="" type="checkbox"/> |
| TELNET port:                  | 23                                  |
| Enable SSH:                   | <input checked="" type="checkbox"/> |
| SSH port:                     | 22                                  |
| Enable STP:                   | <input type="checkbox"/>            |
| Enable WEB:                   | <input checked="" type="checkbox"/> |
| HTTP port:                    | 80                                  |
| HTTPS port:                   | 443                                 |
| PPPoE Settings:               |                                     |
| Use PPPoE:                    | <input type="checkbox"/>            |
| Username:                     |                                     |
| Password:                     |                                     |
| VLAN:                         | <input type="checkbox"/>            |
| VLAN ID:                      | 0                                   |
| MTU:                          | 1400                                |
| LLDP Settings:                |                                     |
| Enable LLDP:                  | <input type="checkbox"/>            |
| LLDP transmit period:         | 30                                  |

Undo all changes
Submit changes
Save

Network settings:

– *Enable DHCP*—when checked, use DHCP protocol to obtain device network settings, otherwise fixed settings (WAN settings) will be used.

Supported options:

- 1—network mask.
- 3—default network gateway address.
- 6—DNS server address.
- 12—device network name.
- 15—domain name.

- 28—network broadcast address.
  - 42—NTP server address.
  - 43—specific vendor information (for option usage, see subsection 'TR-069 Monitoring and Management Protocol Settings' below).
  - 60—specific vendor information (for option usage, see subsection 'DHCP Options' below).
  - 66—TFTP server address (for option usage, see subsection 'Autoupdate Settings' below).
  - 67—name of the file with firmware versions and configurations (for option usage, see subsection 'Autoupdate Settings' below).
  - 82—agent informational parameter (Agent Circuit ID and Agent Remote ID suboptions).
  - 120—outbound SIP servers (for option usage, see Section 5.1.2.2.4).
  - 121—classless static routes (for option usage, see Section 5.1.1.3).
- *Get GW via DHCP*—when checked, use default gateway obtained via DHCP.
  - *Default gateway*—default address of a network gateway. I.e. the address of a gateway that receives all the traffic falling outside the scope of every static routing rule.
  - *Primary DNS IP*—primary DNS server address. To use a local DNS, enter IP address 127.0.0.1 into the field.
  - *Secondary DNS IP*—secondary DNS server address.

DHCP Options configuration:

- *Enable alternative Option 60 value*—when checked, use alternative Option 60 value, specified by user. Otherwise, in Option 60 DHCP request the device will send specific vendor information in the following format:

**[VENDOR:vendor][DEVICE:device type][HW:hardware version][SN:serial number][WAN:MAC address][VERSION:firmware version]**

where

- Vendor—**Eltex**.
- Device type—depends on factory settings.
- Serial number—depends on factory settings.
- MAC address—depends on factory settings.



**You may check factory settings and firmware version in 'System info' tab (5.3.2) of the web interface.**

Example:

```
[VENDOR:Eltex] [DEVICE:TAU32M] [HW:0x21] [SN:MS5370043] [WAN:00:01:09:44:33:22] [VERSION:2.10.0]
```

- *Alternative Option 60 value*—alternative Option 60 value (format: string), specified by user.
- *Option 82. Agent Circuit ID* —allows to add Option 82, Suboption 1 — Agent Circuit ID, into DHCP request.

– *Option 82. Agent Remote ID* —allows to add Option 82, Suboption 2 — Agent Remote ID, into DHCP request.

#### WAN Settings:

- *IP address*—device IP address.
- *Netmask*—device network mask.
- *Broadcast address*—device subnet broadcast address.
- *MTU*—maximum transmission unit, that could be transferred through WAN interface without fragmentation.

#### Device Access Settings (Services):

- *Enable TELNET*—when checked, enable device access via Telnet protocol, otherwise it is disabled.
- *TELNET port*—TCP port (23 by default) for Telnet protocol operation.
- *Enable SSH*—when checked, enable device access via SSH protocol, otherwise it is disabled.
- *SSH port*—TCP port (22 by default) for SSH protocol operation.
- *Enable STP*—when checked, STP is enabled.
- *Enable WEB*—when checked, enable device access via web interface, otherwise it is disabled.
  - *HTTP port*—web server port (80 by default) for HTTP protocol operation.
  - *HTTPS port*—web server port (443 by default) for HTTPS protocol operation.

#### PPPoE Settings—PPPoE Connection Settings:

- *Use PPPoE*—when checked, enable PPPoE connection.
- *Username*—username for PPP server authentication.
- *Password*—password for PPP server authentication.
- *VLAN*—when checked, use separate VLAN for PPPoE access.
- *VLAN ID*—VLAN identifier.
- *MTU*—maximum transmission unit, that could be transferred through PPP interface without fragmentation.



**If the network is managed through PPPoE, do not click the 'Submit Changes' button after you finish PPPoE connection configuration as it may lead to connection loss. Go to 'VLAN conf' tab first, set the setting for 'RTP/signaling/control traffic transmission via PPPoE', and then apply configuration changes using 'Submit Changes' button.**

#### LLDP Settings:

- *Enable LLDP*—when checked, enable LLDP protocol.
- *LLDP transmit period*—LLDP message transmission period. Default value: 30 seconds.

To apply changes, click 'Submit Changes' button. To discard all changes made to configuration, click 'Undo All Changes' button.

To store changes to non-volatile memory of the device, click 'Save' button.

### 5.1.1.2. VLAN conf

In 'VLAN conf' submenu, you will be able to configure VLAN network settings and transmission of signals and voice traffic, and also set up device management through various VLAN networks.



**You don't have to reboot the gateway in order to apply VLAN settings.**

**When applying settings, all current calls will be terminated.**

- **VLAN** is a virtual local area network. VLAN consist of a group of hosts combined into a single network regardless of their location. Devices grouped into a single VLAN will have the same VLAN ID.

Gateway software allows to set up device management (via web interface, TELNET, or SSH), transmission of signals (SIP, H.323/RAS protocol data) and voice traffic (RTP) through a single or multiple virtual local area networks. This feature may become useful, when a separate network is used for device management in organization.



**IP addresses assigned to WAN interface as well as VLAN interfaces should belong to different subnets. For example, if you use a mask 255.255.240.0, IP addresses 192.168.1.6 and 192.168.2.199 will belong to a single network, and if you use a mask 255.255.255.0, they will belong to different networks.**

The screenshot shows the 'VLAN conf' section of the ELTEX TAU-32M.IP WEB configurator. It features a navigation menu with 'Network settings' selected, and sub-menus for 'PBX', 'Switch', 'Monitoring', 'System info', 'Service', and 'Log out'. A warning message states: 'Attention! Changing of these parameters will lead to aborting of all calls!'. The configuration is organized into three sections: VLAN 1, VLAN 2, and VLAN 3. Each section contains the following parameters: 'Enable' (checkbox), 'VLAN ID' (text field), 'DHCP for VLAN' (checkbox), 'Get GW via DHCP' (checkbox), 'IP address' (text field), 'VLAN netmask' (text field), 'VLAN broadcast' (text field), 'MTU' (text field, set to 1496), and 'Class of service' (dropdown menu, set to 0). Below these sections is a 'Traffic Type - VLAN Number' section with three rows: 'RTP' (dropdown, set to 'no VLAN'), 'Signaling (SIP/H.323)' (dropdown, set to 'no VLAN'), and 'Control (Web/Telnet)' (dropdown, set to 'no VLAN'). At the bottom, there are buttons for 'Undo all changes', 'Submit changes', and 'Save'.

In sections **VLAN1**, **VLAN2**, **VLAN3**, you may configure from one to three VLAN networks.

- *Enable*—when checked, enable VLAN.
- *VLAN ID*—VLAN identifier (1- 4095).
- *DHCP for VLAN*—when checked, VLAN network settings will be obtained via DHCP.
- *Get GW via DHCP*—when checked, use default gateway obtained via DHCP protocol.
- *IP address*—VLAN interface IP address.
- *VLAN netmask*—network mask used for VLAN interface.
- *VLAN broadcast*—subnet broadcast address of VLAN interface.
- *MTU*—maximum transmission unit, that could be transferred through VLAN interface without fragmentation.
- *Class of service (802.1p)*—802.1p priority for the current VLAN.

In section '**Traffic Type – VLAN Number**', you can assign one of three configured VLANs (**VLAN1**, **VLAN2**, **VLAN3**) or PPPoE interface to the specific traffic type:

- *RTP*—VLAN, PPPoE assignment for voice traffic.

- *Signaling (SIP/H.323)*—VLAN, PPPoE assignment for SIP/H323 signal traffic.
- *Control (Web/Telnet)*—VLAN, PPPoE assignment for gateway management via web interface, telnet, and SSH.



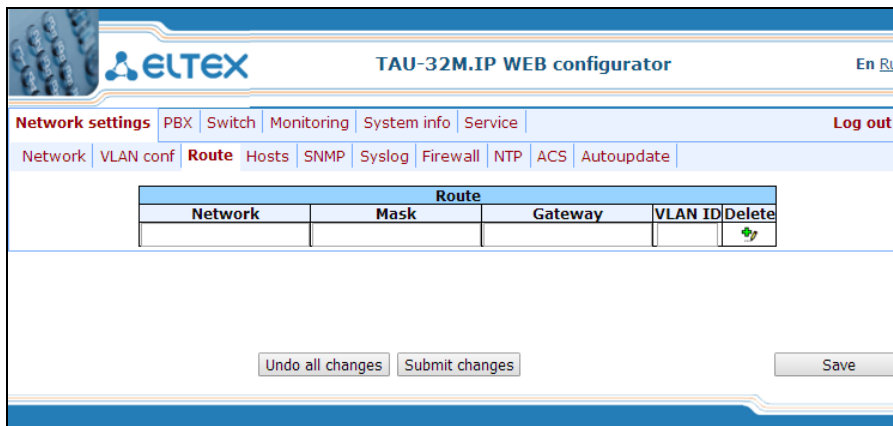
**Voice traffic will be transmitted via PPPoE only after the device is restarted!**

To apply changes, click *'Submit Changes'* button. To discard all changes made to configuration, click *'Undo All Changes'* button.

### 5.1.1.3. Static Routes

In *'Route'* submenu, you can configure static routes for WAN and VLAN interfaces.

Static routing allows you to route packets to defined IP networks or IP addresses through the specified gateways. Packets sent to IP addresses not belonging to the gateway IP network and falling outside the scope of static routing rules will be sent to the default gateway.



- *Network*—destination IP network or address.
- *Mask*—network mask If IP address is specified in the Network field, use the following mask: 255.255.255.255.
- *Gateway*—address of a network gateway that will be used for packet routing to the defined network (or IP address).
- *Vlan*—virtual local area network identifier (VLAN ID). Use it when destination IP network or IP address belong to virtual local area network, otherwise leave this field blank.

To add/apply a new route, enter the data in the field with icon, and click *'Submit Changes'* button. To remove the route, select *'Delete'* checkbox and click *'Submit Changes'* button.

To discard all changes made to configuration, click *'Undo All Changes'* button. To store changes to non-volatile memory of the device, click *'Save'* button.



**Apart from configuration performed via web configurator, the gateway is able to receive static route settings via Option 121 of DHCP protocol. Routes in this option are sent as a list of 'destination description/gateway' pairs, the format is described in RFC 3442.**

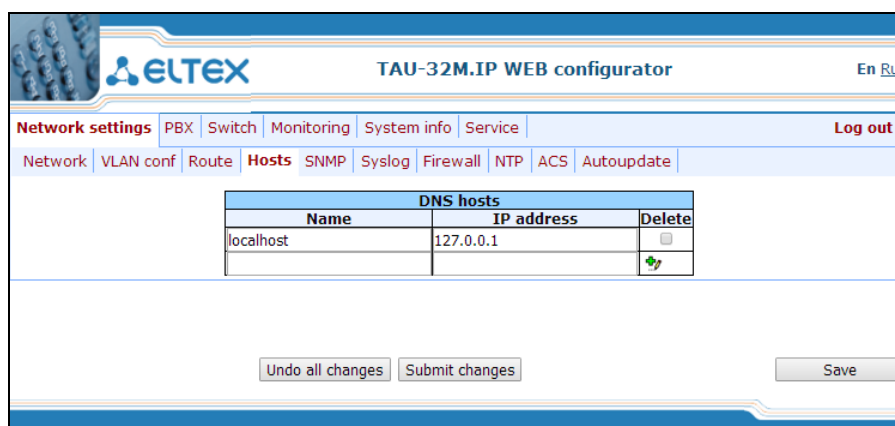
#### 5.1.1.4. Local DNS (Hosts)

In 'Hosts' submenu, you can configure settings required for local DNS operation.

- **Local DNS**—allows the gateway to obtain IP address of the communicating device by its network name (hostname). You may use local DNS in cases when DNS server is missing from the network segment that the gateway belongs to, and you need to establish routing using network names, or when you have to use SIP server network name as its address. Although, you have to know matches between hostnames and their IP addresses. Also, local DNS allows you to configure SIP domain on a gateway (see Section 5.1.2.2.4.3).


Local DNS configuration involves definition of matches between hostnames and their respective IP addresses.

To enable local DNS, enter 127.0.0.1 into 'Primary DNS IP' field in the 'Network' tab. Also, local DNS will be used when configured DNS servers are not available.



#### Table of domain names (DNS hosts):

- *Name*—name of a host.
- *IP-address*—IP address of a host.

To add/apply a new route, enter the data in the field with  icon, and click 'Submit Changes' button. To remove the route, select 'Delete' checkbox and click 'Submit Changes' button.

After implementation of changes, click 'Submit Changes' button; to discard all changes, click 'Undo All Changes' button; to save changes, click 'Save' button.

#### 5.1.1.5. SNMP protocol configuration

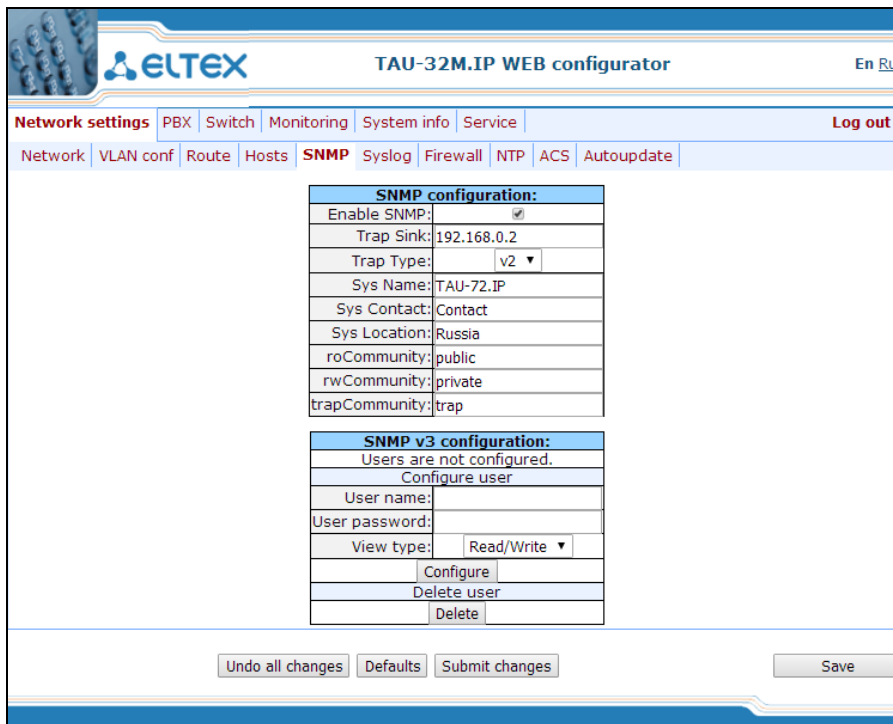
TAU-32M.IP software allows to monitor status of the device and its sensors via SNMP protocol. In SNMP submenu, you can configure settings of SNMP agent. Device supports SNMPv1, SNMPv2c, SNMPv3 protocol versions.



For detailed monitoring parameters and Traps description, see MIBs on disk shipped with the gateway.

SNMP agent sends a message (SNMP-trap or SNMPv2-trap), when the following events occur:

- Port is blocked.
- Port is unblocked.
- Unit power supply voltage is changed from 48V to 60V or in reverse.
- Fans turned on/off.
- Fans malfunction.
- One of the following parameters falls outside of allowable limits:
  - Board power supply voltage should fall within the limits:  $8V < V_{input} < 16V$ .
  - Temperature on a sensor should not exceed  $90^{\circ}\text{C}$ .
- Successful/unsuccessful firmware update.
- Successful/unsuccessful configuration download/upload.



The screenshot shows the 'SNMP configuration' page in the ELTEX TAU-32M.IP WEB configurator. The page has a navigation menu with 'SNMP' selected. The configuration table is as follows:

| SNMP configuration: |                                     |
|---------------------|-------------------------------------|
| Enable SNMP:        | <input checked="" type="checkbox"/> |
| Trap Sink:          | 192.168.0.2                         |
| Trap Type:          | v2 ▼                                |
| Sys Name:           | TAU-72.IP                           |
| Sys Contact:        | Contact                             |
| Sys Location:       | Russia                              |
| roCommunity:        | public                              |
| rwCommunity:        | private                             |
| trapCommunity:      | trap                                |

| SNMP v3 configuration:                   |                          |
|--|--------------------------|
| Users are not configured.                |                          |
| Configure user                           |                          |
| User name:                               | <input type="text"/>     |
| User password:                           | <input type="password"/> |
| View type:                               | Read/Write ▼             |
| <input type="button" value="Configure"/> |                          |
| Delete user                              |                          |
| <input type="button" value="Delete"/>    |                          |

At the bottom of the page, there are three buttons: 'Undo all changes', 'Defaults', and 'Submit changes'. A 'Save' button is located at the bottom right.

After implementation of changes, click 'Submit Changes' button; to discard all changes, click 'Undo All Changes' button; to save changes, click 'Save' button.

SNMP configuration:

- *Trap Sink*—IP address of a trap recipient (manager server or proxy agent server).
- *Trap Type*—SNMP trap type (SNMP-trap or SNMPv2-trap).
- *SysName*—device system name.



- *SysContact*—device vendor contact information.
- *SysLocation*—device location.
- *roCommunity*—password for parameter reading (common: *public*).
- *rwCommunity*—password for parameter writing (common: *private*).
- *trapCommunity*—password located in traps.

### SNMP v3 configuration:

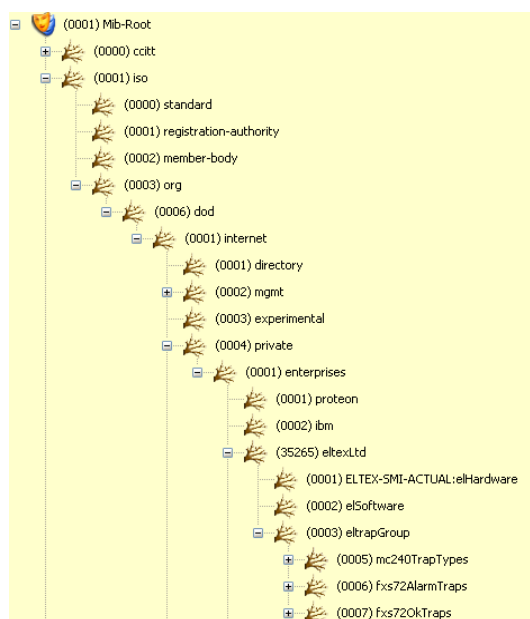
The system employs a single SNMPv3 user that executes SORM commands. SORM feature implementation is based on rfc3924 recommendation—Cisco Architecture for Lawful Intercept in IP Networks. To perform the pickup, the following MIBs are used: CISCO-IP-TAP-MIB.my and CISCO-TAP2-MIB.my.

- *User name*—account username.
- *User password*—access password Password should contain 8 characters or more.
- *View type*—account access mode selection:
  - *Read/Write*—read/write mode.
  - *Read only*—read-only mode.
- *Delete*—click this button to delete all accounts for access via SNMP v3.

Click '*Configure*' button to apply SNMPv3 user configuration. Settings will be applied immediately. Click '*Delete*' button to delete the record.

To discard all changes made to configuration, click '*Undo All Changes*' button. To set the default parameters, click '*Defaults*' button. To apply changes, click '*Submit Changes*' button.

### **MIB Tree**



## SNMP TRAP

SNMP agent sends a message (SNMP-trap or SNMPv2-trap), when the following events occur:

- Port is blocked.
- Port is unblocked.
- Unit power supply voltage is changed.
- Fans turned on/off.
- Fans malfunction.
- SFP module is installed, but there is no optical link.
- BPU connection lost/resumed.
- One of the following parameters falls outside of allowable limits:
  - Board power supply voltage should fall within the limits:  $8V < V_{bat} < 16V$
  - Temperature on a sensor should not exceed  $90^{\circ}\text{C}$ .
- Successful/unsuccessful firmware update.
- Successful/unsuccessful configuration download/upload.

### 5.1.1.5.1. SNMP Monitoring

Parameter settings listed in a second column correspond to actual parameter names used in the web interface and their descriptions are listed in the respective Sections of this document.

The gateway supports monitoring of the following parameters via SNMP:

#### General Gateway Data

Object identifier enterprises.35265.1.9.

|                |                        |
|----------------|------------------------|
| fxsDevName     | Gateway name           |
| fxsDevType     | Gateway type (TAU-32M) |
| fxsDevCfgBuild | Firmware version       |
| fxsFreeSpace   | Free disk space        |
| fxsFreeRam     | Free RAM               |
| fxsCpuUsage    | CPU utilization (%)    |

Object identifier enterprises.35265.4.

|                    |  |
|--------------------|--|
| omsSerialNumber    | Device serial number (factory setting) |
| omsLinuxVersion    | Linux version                          |
| omsFirmwareVersion | Media processor version                |
| omsBPUVersion      | Subscriber unit firmware version       |
| omsFactoryType     | Device type (factory setting)          |
| omsFactoryMAC      | Factory default MAC address            |
| omsProductClass    | Hardware platform version              |

### Platform Sensor Parameters

Object identifier enterprises.35265.1.9.10.

|                          |  |
|--------------------------|--|
| fxsMonitoringTemp1       | Temperature measured by submodule 1 sensor |
| fxsMonitoringTemp2       | Temperature measured by submodule 2 sensor |
| fxsMonitoringTemp3       | Temperature measured by submodule 3 sensor |
| fxsMonitoringTemp4       | Temperature measured by submodule 4 sensor |
| fxsMonitoringFanState    | Fan status (on or off)                     |
| fxsMonitoringFan1Rotate  | Fan health 1, if it's on                   |
| fxsMonitoringFan2Rotate  | Fan health 2, if it's on                   |
| fxsMonitoringDevicePower | Type of power supply installed             |
| fxsMonitoringVinput      | Board power supply voltage, V              |

### Call Monitoring

Object identifier enterprises.35265.1.9.12.1.1.

|                         |  |
|-------------------------|--|
| fxsPortPhoneNumber      | Subscriber number  |
| fxsPortState            | Port status  |
| fxsPortUserName         | Subscriber name  |
| fxsPortTalkingNum       | Number(s) of the remote subscriber or two subscribers in conference mode |
| fxsPortTalkingStartTime | Call start time  |
| fxsPortSipConnected     | Last known successful registration on SIP server                         |
| fxsPortH323Connected    | Gatekeeper registration time   |
| fxsPortSipConnecteNext  | Amount of time until next SIP server registration                        |
| fxsPortSipConnecteState | SIP server registration status   |
| fxsPortSipConnectHost   | Registration SIP server address  |

List of possible port states:

- *hangdown*—phone is offhook
- *hangup*—phone is onhook
- *FXO hangdown*—FXO port is busy
- *FXO hangdup*—FXO port is free
- *dial*—dialling number
- *ringback*—send 'ringback' tone
- *ringing*—send 'ringing' tone
- *talking*—call in progress
- *conference*—3-way conference
- *busy*—sending 'busy' tone
- *hold*—port is on hold
- *testing*—port is in testing mode

List of possible registration states:

- *off*—registration disabled
- *ok*—successful registration
- *failed*—registration failed

### **Call Group Monitoring**

Object identifier enterprises.35265.1.9.41.

|                                  |  |
|----------------------------------|--|
| serialGroupPhone                 | Group sequential number                            |
| serialGroupRegistrationState     | SIP server registration status                     |
| serialGroupRegistrationHost      | Registration SIP server address                    |
| serialGroupLastRegistrationAt    | Last known successful registration on SIP server   |
| serialGroupNextRegistrationAfter | Remaining time for SIP server registration renewal |
| serialGroupH323GK                | H.323 gatekeeper registration time                 |

### **FXO Group Monitoring**

Object identifier enterprises.35265.1.9.42.

|                               |  |
|-------------------------------|--|
| fxoGroupPhone                 | Group sequential number                            |
| fxoGroupRegistrationState     | SIP server registration status                     |
| fxoGroupRegistrationHost      | Registration SIP server address                    |
| fxoGroupLastRegistrationAt    | Last known successful registration on SIP server   |
| fxoGroupNextRegistrationAfter | Remaining time for SIP server registration renewal |
| fxoGroupH323GK                | H.323 gatekeeper registration time                 |

#### **5.1.1.5.2. Device Configuration via SNMP**

Gateway supports data readout and configuration via SNMP for the following settings:

### **Custom Settings for FXS Ports**

Object identifier enterprises.35265.1.9.12.2.1.

|                          |   |
|--------------------------|---|
| fxsPortConfigPhone       | Phone (up to 20 characters)                   |
| fxsPortConfigUserName    | User Name (up to 20 characters)               |
| fxsPortConfigAuthName    | Authentication name (up to 20 characters)     |
| fxsPortConfigAuthPass    | Authentication password (up to 20 characters) |
| fxsPortConfigCustom      | Custom  |
| fxsPortConfigPlaymoh     | Play music on hold                            |
| fxsPortConfigAON         | CallerID                                      |
| fxsPortConfigAONHideDate | Hide date                                     |
| fxsPortConfigAONHideName | Hide Name                                     |
| fxsPortConfigTaxophone   | Taxophone—operation in payphone mode          |

|                                 |   |
|---------------------------------|---|
| fxsPortConfigMinFlashtime       | Min Flashtime (70 to 1000)  |
| fxsPortConfigMaxFlashtime       | Max Flashtime (minflashtime to 1000)                                      |
| fxsPortConfigGainr              | Gain receive (-230 to 20)   |
| fxsPortConfigGaint              | Gain transmit (-170 to 60)  |
| fxsPortConfigCategory           | SS7 category (SIP-T)  |
| fxsPortConfigCallTransfer       | Process flash   |
| fxsPortConfigCallWaiting        | Call Waiting  |
| fxsPortConfigHotLine            | Hot Line  |
| fxsPortConfigHotNumber          | Hot Number (up to 20 characters)  |
| fxsPortConfigHotTimeout         | Hot Timeout (0 to 300)  |
| fxsPortConfigDisabled           | Disabled  |
| fxsPortConfigCtBusy             | CF Busy   |
| fxsPortConfigCtUnconditional    | CF Unconditional  |
| fxsPortConfigCtNoanswer         | CF No answer  |
| fxsPortConfigCtTimeout          | CFNR Timeout (0 to 300)   |
| fxsPortConfigClir               | CLIR  |
| fxsPortConfigStopDial           | Stop dial at #  |
| fxsPortConfigAltNumber          | Alt.Number (up to 20 characters)  |
| fxsPortConfigUseAltNumber       | Use Alt.Number  |
| fxsPortConfigPickUp             | Membership in PickUp groups (up to 86 characters)                         |
| fxsPortConfigSipPort            | SIP Port (0 to 65535)   |
| fxsPortConfigCfgPriOverCw       | CFB has priority over CW  |
| fxsPortConfigRowStatus          | Row status (required in SNMP SET). Value for storing data in a file:<br>1 |
| fxsPortConfigDvoCwEn            | Call waiting enable   |
| fxsPortConfigDvoCtAttendedEn    | Call transfer attended enable   |
| fxsPortConfigDvoCtUnattendedEn  | Call transfer unattended enable   |
| fxsPortConfigDvoUnconditionalEn | Call forward unconditional enable   |
| fxsPortConfigDvoCfBusyEn        | Call forward on busy enable   |
| fxsPortConfigDvoCfAnswerEn      | Call forward on no answer enable  |
| fxsPortConfigDvoCfServiceEn     | Call forward on out of service enable                                     |
| fxsPortConfigDvoDoDisturbEn     | Do not disturb enable   |
| fxsPortConfigCtOutofservice     | CF Out Of Service   |
| fxsPortConfigCfuNumber          | CF Unconditional Number (up to 20 characters)                             |
| fxsPortConfigCfbNumber          | CF Busy Number (up to 20 characters)                                      |
| fxsPortConfigCfnrNumber         | CF No Reply Number (up to 20 characters)                                  |
| fxsPortConfigCfoosNumber        | CF Out Of Service Number (up to 20 characters)                            |
| fxsPortConfigDnd                | DND   |
| fxsPortConfigEnableCpc          | CPC   |

|                                    |   |
|------------------------------------|---|
| fxsPortConfigCpcTime               | CPC time(ms)  |
| fxsPortConfigSipProfileID          | SIP/H323 profile  |
| fxsPortConfigPortProfileID         | Subscriber profile  |
| fxsPortConfigUseAltNumberAsContact | Use alternative number as contact<br>(only for serial groups members)                           |
| fxsPortConfigRowStatus             | Row status This parameter is mandatory for SNMP SET. To store data in a file, set '1' as value. |
| fxsPortConfigCpcRus                | Category  |
| fxsPortConfigModifier              | Modifier  |
| fxsPortConfigMwiDialtone           | MWI   |
| fxsPortConfigRowStatus             | Row status This parameter is mandatory for SNMP SET. To store data in a file, set '1' as value. |



**These settings match ones described in Section 5.1.2.4.**

### ***Custom Settings for FXO Ports***

Object identifier enterprises.35265.1.9.12.2.1.

|  |                                       |
|--|---------------------------------------|
| fxsPortConfigFxoFlashTime                    | Flashtime                             |
| fxsPortConfigFxoDelTdm                       | Interdigit delay                      |
| fxsPortConfigFxoRingtdm                      | Ring detection                        |
| fxsPortConfigPstnNumberprefix                | PSTN number prefix                    |
| fxsPortConfigPstnNameprefix                  | PSTN name prefix                      |
| fxsPortConfigUsePstnCid                      | Use PSTN CallerID                     |
| fxsPortConfigtdmhotline                      | Use Hotline to PSTN                   |
| fxsPortConfigtdmhottimeout                   | Hotline Timeout to PSTN               |
| fxsPortConfigtdmhotnumber                    | Hotline Number to PSTN                |
| fxsPortConfigDontDetectDT                    | Dialtone detection                    |
| fxsPortConfigDelayDialingTimeout             | Dialing delay, sec                    |
| fxsPortConfigDialing                         | Dialing                               |
| fxsPortConfigTransmitNumber                  | TransmitNumber                        |
| fxsPortConfigDontTransmitPrefix              | Don't transmit prefix                 |
| fxsPortConfigDialToneDetectionParameters     | Dial tone detection parameters        |
| fxsPortConfigBusyToneDetectionParameters     | Busy tone detection parameters        |
| fxsPortConfigDtDetectTime                    | Dial tone time detect(s)              |
| fxsPortConfigDecadePulseTime                 | Pulse time(ms)                        |
| fxsPortConfigDecadePauseTime                 | Pause time(ms)                        |
| fxsPortConfigNoOffhookAtRinging              | No offhook at ringing                 |
| fxsPortConfigFxoCallBusy                     | 503 Service unavailable on busy (SIP) |
| fxsPortConfigRingBackToneDetectionParameters | Ringback tone detection parameters    |

|                                  |  |
|----------------------------------|--|
| fxsPortConfigPstnActivity        | PSTN activity                            |
| fxsPortConfigPstnRbDetectTimeout | Ringback detect timeout (s)              |
| fxsPortConfigGainr               | Gain receive (-230 to 20)                |
| fxsPortConfigGaint               | Gain transmit (-170 to 60)               |
| fxsPortConfigFxoMinLevelDetect   | Minimum level of detectable signal (dBm) |



**These settings match ones described in Section 5.1.2.4.**

### ***FXS settings of subscriber profiles***

Object identifier *enterprises.35265.1.9.30.3.1.1.*

|                          |   |
|--------------------------|---|
| profilePortsPlaymoh      | Play music on hold  |
| profilePortsAON          | CallerID  |
| profilePortsAONHideDate  | Hide date   |
| profilePortsAONHideName  | Hide Name   |
| profilePortsTaxophone    | Taxophone   |
| profilePortsMinFlashtime | Min Flashtime (70 to 1000)  |
| profilePortsMaxFlashtime | Max Flashtime (minflashtime to 1000)  |
| profilePortsGainr        | Gain receive (-230 to 20)   |
| profilePortsGaint        | Gain transmit (-170 to 60)  |
| profilePortsCategory     | SS7 category (SIP-T)  |
| profilePortsCfgPriOverCw | CFB has priority over CW  |
| profilePortsEnableCpc    | CPC enable  |
| profilePortsCpcTime      | CPC time  |
| profilePortsRowStatus    | Row status This parameter is mandatory for SNMP SET. To store data in a file, set '1' as value. |
| profilePortsCpcRus       | Category  |



**These settings match ones described in Section 5.1.2.4.**

### ***FXO settings of subscriber profiles***

Object identifier enterprises.35265.1.9.30.3.1.1.

|   |  |
|---|--|
| profilePortsDontDetectDT                    | Dialtone detection                       |
| profilePortsDelayDialingTimeout             | Dialing delay, sec                       |
| profilePortsDialing                         | Dialing                                  |
| profilePortsTransmitNumber                  | TransmitNumber                           |
| profilePortsGainr                           | Gain receive (-230 to 20)                |
| profilePortsGaint                           | Gain transmit (-170 to 60)               |
| profilePortsDontTransmitPrefix              | Don't transmit prefix                    |
| profilePortsDialToneDetectionParameters     | Dial tone detection parameters           |
| profilePortsBusyToneDetectionParameters     | Busy tone detection parameters           |
| profilePortsFxoCallBusy                     | 503 Service unavailable on busy (SIP)    |
| profilePortsRingBackToneDetectionParameters | Ringback tone detection parameters       |
| profilePortsPstnActivity                    | PSTN activity                            |
| profilePortsPstnRbDetectTimeout             | Ringback detect timeout (s)              |
| profilePortsFxoMinLevelDetect               | Minimum level of detectable signal (dBm) |



**These settings match ones described in Section 5.1.2.4.**

### ***Configuration of common SIP parameters***

Object identifier enterprises.35265.1.9.30.1.1.

|                                |                                      |
|--------------------------------|--------------------------------------|
| sipCommonEnablesip             | Enable SIP                           |
| sipCommonShortmode             | Short mode                           |
| sipCommonTransport             | Transport                            |
| sipCommonSipMtu                | SIP UDP MTU                          |
| sipCommonInviteTotalT          | Invite total timeout (1000 to 39000) |
| sipCommonInviteInitT           | Invite initial timeout (100 to 1000) |
| sipCommonPortRegistrationDelay | Port registration delay (ms)         |
| stunEnable                     | Use STUN                             |
| stunServer                     | STUN server                          |
| stunInterval                   | STUN interval                        |
| sipPublicIp                    | PublicIP                             |



**These settings match ones described in Section 5.1.2.2.1.**



### **Configuration of common parameters**

Object identifier enterprises.35265.1.9.37.

|                          |   |
|--------------------------|---|
| fansForceEnable          | Fans force enable <sup>1</sup>          |
| fansThresholdTemperature | Fans threshold temperature <sup>1</sup> |
| deviceName               | Device name                             |
| startTimer               | Start timer                             |
| durationTimer            | Duration timer                          |
| waitAnswerTimer          | Wait answer timer                       |
| powerMode                | Extended range loop <sup>2</sup>        |
| siptUsePrefix            | Use prefix (SIP-T)                      |
| siptPrefix               | Prefix (SIP-T)                          |

### **Configuration of port TCP/UDP parameters**

Object identifier enterprises.35265.1.9.45.

|                          |                             |
|--------------------------|-----------------------------|
| rtpSipMin                | RTP SIP min                 |
| rtpSipMax                | RTP SIP max                 |
| interceptPortMin         | Intercept port min          |
| interceptPortMax         | Intercept port max          |
| diffservForSip           | Diffserv for SIP            |
| diffservForRtp           | Diffserv for RTP            |
| verifyRemoteMediaAddress | Verify remote media address |

### **Configuration of call limits**

Object identifier enterprises.35265.1.9.46.1.

|                          |   |
|--------------------------|---|
| clType                   | Host of neighbour gateway radiobutton   |
| clHostOfNeighbourGateway | Host of neighbour gateway area  |
| clSimultaneousCallsCount | Simultaneous calls count  |
| clRowStatus              | Row status This parameter is mandatory for SNMP SET. To store data in a file, its value should be as follows: to change the limit record, set value 1, to add a record—value 4, to remove a record—value 2. |

<sup>1</sup> Not used for TAU-32M.IP rev. B

<sup>2</sup> Used only for TAU-32M.IP rev. B

### ***Distinctive ringing service configuration***

Object identifier enterprises.35265.1.9.47.1.

|                      |   |
|----------------------|---|
| drRule               | Rule  |
| drRing               | Ring, msec  |
| drPause              | Pause, msec   |
| drSubscriberProfiles | Subscriber profiles   |
| drRowStatus          | Row status This parameter is mandatory for SNMP SET. To store data in a file, its value should be as follows: to change the service record, set value 1, to add a record—value 4, to remove a record—value 2. |

### ***Automatic update configuration***

Object identifier enterprises.35265.1.9.35.

|                                |                               |
|--------------------------------|-------------------------------|
| fxsEnableAutoupdate            | Enable autoupdate             |
| fxsSource                      | Source                        |
| fxsTFTPServer                  | Autoupdate server             |
| fxsConfigurationFile           | Configuration file            |
| fxsFirmwareVersion             | Firmware versions file        |
| fxsConfigurationUpdateInterval | Configuration update interval |
| fxsFirmwareUpdateInterval      | Firmware update interval      |
| autoupdateProtocol             | Autoupdate protocol           |
| autoupdateAuth                 | Autoupdate auth               |
| autoupdateUser                 | Username                      |
| autoupdatePassword             | Password                      |

### ***System Log Configuration***

Object identifier enterprises.35265.1.9.38.

|              |                       |
|--------------|-----------------------|
| runSyslog    | Run syslog on startup |
| syslogAddr   | Syslog server         |
| syslogPort   | Syslog port           |
| appErr       | Error                 |
| appWarn      | Warning               |
| appInfo      | Info                  |
| appDbg       | Debug                 |
| sipLevel     | SIP Log Level         |
| h323Level    | H323 Log Level        |
| vapiEnabled  | Enabled               |
| vapiLibLevel | Lib Level             |

|              |           |
|--------------|-----------|
| vapiAppLevel | App Level |
| appAlarm     | Alarm     |



**These settings match ones described in Section 5.1.1.6.**

### ***Specific SIP parameters' configuration***

Object identifier enterprises.35265.1.9.30.1.3.1.

|                                 |  |
|---------------------------------|--|
| sipProfileObttimeout            | Dial timeout (0 to 300)                  |
| sipProfileMode                  | Proxy mode                               |
| sipProfileOptions               | Home server test                         |
| sipProfileKeepalivet            | Keepalive time (10000 to 3600000), ms    |
| sipProfileDomainToReg           | Use domain to Register                   |
| sipProfileDomain                | SIP-Domain (up to 20 characters)         |
| sipProfileRegisterRetryInterval | Registration Retry Interval (10 to 3600) |
| sipProfileOutbound              | Outbound                                 |
| sipProfileInboundProxy          | Inbound                                  |
| sipProfileExpires               | Expires (10 to 345600)                   |
| sipProfileAuthentication        | Authentication                           |
| sipProfileUsername              | Username (up to 20 characters)           |
| sipProfilePassword              | Password (up to 20 characters)           |
| sipProfileDtmfmime              | DTMF MIME Type                           |
| sipProfileHfmime                | Hook flash MIME Type                     |
| sipProfileCtWithReplaces        | CT with replaces                         |
| sipProfile100Rel                | 100rel                                   |
| sipProfileUserPhone             | User=Phone                               |
| sipProfileUriEscapeHash         | Escape hash uri                          |
| sipProfileCwRingback            | Ringback at callwaiting                  |
| sipProfileRingbackSdp           | Remote ringback                          |
| sipProfileRingback              | Ringback at answer 183                   |
| sipProfileProxy0                | Proxy (up to 40 characters)              |
| sipProfileProxy1                |  |
| sipProfileProxy2                |  |
| sipProfileProxy3                |  |
| sipProfileProxy4                |  |
| sipProfileRegrar0               | Registrar (up to 40 characters)          |
| sipProfileRegrar1               |  |
| sipProfileRegrar2               |  |
| sipProfileRegrar3               |  |
| sipProfileRegrar4               |  |

|   |   |
|---|---|
| sipProfileRegistration0                   | Use registration  |
| sipProfileRegistration1                   |   |
| sipProfileRegistration2                   |   |
| sipProfileRegistration3                   |   |
| sipProfileRegistration4                   |   |
| sipProfilePRTpstat                        | P-RTP-Stat  |
| sipProfileRowStatus                       | Row status This parameter is mandatory for SNMP SET. To store data in a file, set '1' as value. |
| sipProfileKeepAliveInterval               | NAT Keep Alive Interval (s)   |
| sipProfileKeepAliveMode                   | NAT Keep Alive Msg  |
| sipProfileConferenceMode                  | Conference mode   |
| sipProfileConferenceServer                | Conference server   |
| sipProfileEnableIMS                       | Enable IMS  |
| sipProfileXCAPNameForThreePartyConference | XCAP name for three-party conference  |
| sipProfileXCAPNameForHotline              | XCAP name for hotline   |
| sipProfileXCAPNameForCallWaiting          | XCAP name for call waiting  |
| sipProfileXCAPNameForCallHold             | XCAP name for call hold   |
| sipProfileXCAPNameForExplicitCallTransfer | XCAP name for explicit call transfer  |
| sipProfileUseAlertInfo                    | Alert-Info  |
| sipProfileFullRuriCompliance              | Full RURI compliance  |
| sipProfileChangeover                      | Change-over   |
| sipProfileChangeoverBy408                 | Changeover by timeout   |



**These settings match ones described in Section 5.1.2.2.4.**

### ***Configuration of the distinctive type ring with alert-info header***

Object identifier *enterprises.35265.1.9.30.1.5.1.*

|                  |   |
|------------------|---|
| cadenceNumber    | Rule number   |
| cadenceName      | Alert-Info string   |
| cadenceRingRule  | Distinctive Ring rule   |
| cadenceRowStatus | Row status This parameter is mandatory for SNMP SET. To store data in a file, its value should be as follows: to change the service record, set value '1', to add a record—value '4', to remove a record—value '2'. |

### ***Codecs Configuration***

Object identifier *enterprises.35265.1.9.30.7.1.1.*

|          |            |
|----------|------------|
| useG711A | Use G.711U |
|----------|------------|

|                         |   |
|-------------------------|---|
| useG711U                | Use G.711A  |
| useG726to32             | Use G.726-32  |
| useG723                 | Use G.723   |
| useG729B                | Use G.729B  |
| useG729A                | Use G.729A  |
| g711Ptime               | G.711 Ptime   |
| g729Ptime               | G.729 Ptime   |
| g723Ptime               | G.723 Ptime   |
| g726to32Ptime           | G.726-32 Ptime  |
| g726to32PT              | G.726-32 PT   |
| dtmfTransfer            | DTMF Transfer   |
| flashTransfer           | Flash Transfer  |
| faxDetectDirection      | Fax Detect Direction  |
| faxTransferCodec        | Fax Transfer Codec  |
| slaveFaxTransferCodec   | Slave Fax Transfer Codec  |
| modemTransfer           | Modem Transfer  |
| rfc2833PT               | rfc2833 PT  |
| silenceSuppression      | Silence suppression   |
| echoCanceller           | Echo canceller  |
| nlpDisable              | NLP disable   |
| comfortNoise            | Comfort noise   |
| rtcpTimer               | RTCP timer  |
| rtcpControlPeriod       | RTCP control period   |
| rtcpXR                  | RTCP-XR   |
| ciscoNsePT              | NSE PT  |
| t38MaxDatagramSize      | Max Datagram Size   |
| t38Bitrate              | Bitrate   |
| modemFaxDelay           | Delay (modem/fax)   |
| voiceMode               | Mode  |
| voiceDelayMin           | Delay   |
| voiceDelayMax           | Delay max   |
| voiceDeletionThreshold  | Deletion threshold  |
| voiceDeletionMode       | Deletion mode   |
| profilesCodecsRowStatus | Row status This parameter is mandatory for SNMP SET. To store data in a file, set '1' as value. |
| rfc3264PtCommon         | Decoding rfc2833 with PT from answer SDP  |
| rtcpXR                  | RTCP-XR   |



**These settings match ones described in Section 5.1.2.2.5.**

### Configuration of routing and pickup groups

Object identifier enterprises. 35265.1.9.30.5.1.1.

Data readout performed for enterprises.35265.1.9.30.5.1.1.fxsDialPlanNext.n identifier allows you to get the number of the next free record in SIP profile routing table. You can configure up to 300 records in total.

|                           |  |
|---------------------------|--|
| profileDialPlanHost       | IP address (up to 40 characters)   |
| profileDialPlanDigits     | Prefix (up to 20 characters)   |
| profileDialPlanTimeout    | Timeout (0 to 20)  |
| profileDialPlanMinDigits  | Min Digits (up to 20)  |
| profileDialPlanType       | Protocol&Target  |
| profileDialPlanAccessMask | Ingress (up to 108 characters)   |
| profileDialPlanDialtone   | Dial tone  |
| profileDialPlanModifier   | Modifier (up to 8 characters)  |
| profileDialPlanDelnum     | Number of digits to delete (0 to quantity of digits in a number)   |
| profileDialPlanPtime      | Ptime (0, 10, 20,... 90)   |
| profileDialPlanRowStatus  | Row status This parameter is mandatory for SNMP SET. To store data in a file, its value should be as follows: to change the dialplan record, set value 1, to add a record—value 4, to remove a record—value 2. |



**These settings match ones described in Section 5.1.2.2.6.**

### Configuration of a Routing Plan Based on Regular Expressions

Object identifier enterprises.35265.1.9.30.5.3.1.

|                            |   |
|----------------------------|---|
| profileRegExpDialOn        | Regular expression dialplan   |
| profileRegExpDialProtocol  | Protocol  |
| profileRegExpDialText      | Expressions   |
| profileRegExpDialRowStatus | Row status This parameter is mandatory for SNMP SET. To store data in a file, set '1' as value. |



**These settings match ones described in Section 5.1.2.2.6.4.**

### Call Group Configuration

Object identifier enterprises.35265.1.9.18.1.1.

Data readout performed for enterprises.35265.1.9.18.fxsSerialGroupsNext identifier allows you to get the number of the next free group. You can configure up to 8 groups in total.

|                        |                             |
|------------------------|-----------------------------|
| fxsSerialGroupsPhone   | Phone (up to 20 characters) |
| fxsSerialGroupsEnabled | Enabled                     |

|                           |  |
|---------------------------|--|
| fxsSerialGroupsSerialType | Type   |
| fxsSerialGroupsBusyType   | Busy   |
| fxsSerialGroupsTimeout    | Timeout (0 to 99)  |
| fxsSerialGroupsSipPort    | SIP Port (0 to 65535)  |
| fxsSerialGroupsAuthName   | Group name (up to 20 characters)   |
| fxsSerialGroupsAuthPass   | Password (up to 20 characters)   |
| fxsSerialGroupsPorts      | Ports (up to 48 characters)  |
| fxsSerialGroupsSipProfile | Sip Profile  |
| fxsSerialGroupsRowStatus  | Row status This parameter is mandatory for SNMP SET. To store data in a file, its value should be as follows: to change the serial group record, set value 1, to add a record—value 4, to remove a record—value 2. |



**These settings match ones described in Section 5.1.2.7.**

### **Configuration of FXO groups**

Object identifier enterprises.35265.1.9.34.1.1.

|                                   |  |
|-----------------------------------|--|
| fxoSerialGroupsPhone              | Phone  |
| fxoSerialGroupsEnabled            | Enabled  |
| fxoSerialGroupsBusyType           | Busy mode  |
| fxoSerialGroupsSipPort            | SIP port   |
| fxoSerialGroupsAuthName           | Group name   |
| fxoSerialGroupsAuthPass           | Password   |
| fxoSerialGroupsPorts              | SIP port   |
| fxoSerialGroupsSipProfile         | SIP/H323 profile   |
| fxoSerialGroupsTransmitNumber     | Transmit number  |
| fxoSerialGroupsDontTransmitPrefix | Don't transmit prefix  |
| fxoSerialGroupsRowStatus          | Row status This parameter is mandatory for SNMP SET. To store data in a file, its value should be as follows: to change the serial group record, set value 1, to add a record—value 4, to remove a record—value 2. |
| fxoSerialGroupsSend503OnBusy      | 503 Service unavailable on busy (SIP)  |
| fxoSerialGroupsType               | Group type   |



**These settings match ones described in Section 5.1.2.8.**

### ***SNMP Settings Configuration***

Object identifier enterprises.35265.1.9.31.

|                   |                                |
|-------------------|--------------------------------|
| tauTrapSink       | Trap Sink                      |
| tauTrapType       | Trap Type                      |
| tauSysName        | Sys Name                       |
| tauSysContact     | Sys Contact                    |
| tauSysLocation    | Sys Location                   |
| tauRoCommunity    | roCommunity                    |
| tauRwCommunity    | rwCommunity                    |
| tauTrapCommunity  | trapCommunity                  |
| tauUserV3Name     | User name                      |
| tauUserV3Password | User password                  |
| tauViewV3Type     | View type                      |
| tauRestartSnmp    | Allows to restart SNMP client. |



**These settings match ones described in Section 5.1.1.5.**

### ***Configuration of Supplementary Service Codes***

Object identifier enterprises.35265.1.9.20.

|                           |                                |
|---------------------------|--------------------------------|
| tauVoipDvoCallwaiting     | Call waiting                   |
| tauVoipDvoCtAttended      | Call transfer attended         |
| tauVoipDvoCtUnattended    | Call transfer unattended       |
| tauVoipDvoCfUnconditional | Call forward unconditional     |
| tauVoipDvoCfBusy          | Call forward on busy           |
| tauVoipDvoCfNoanswer      | Call forward on no answer      |
| tauVoipDvoCfService       | Call forward on out of service |
| tauVoipDvoDoDisturb       | Do not disturb                 |



**These settings match ones described in Section 5.1.2.6.**



## Firewall Settings Configuration

Object identifier enterprises.35265.1.9.44.

|                           |   |
|---------------------------|---|
| startingSourceIpAddress   | Starting source IP address  |
| numberOfSourceIpAddresses | Number of source IP addresses   |
| allSourceIpAddresses      | All source IP addresses   |
| ruleprotocol              | Protocol  |
| typeOfMessageICMP         | Type of message (ICMP)  |
| startingSourcePort        | Starting source port  |
| numberOfSourcePorts       | Number of source ports  |
| allSourcePorts            | All source ports  |
| startingDestinationPort   | Starting destination port   |
| numberOfDestinationPorts  | Number of destination ports   |
| allDestinationPorts       | All destination ports   |
| ruleTarget                | Target  |
| ruleMoveTo                | Moves the rule in the table; specify a row to move the rule into (1 to 30).   |
| ruleRowStatus             | Row status This parameter is mandatory for SNMP SET. To store data in a file, its value should be as follows: to change the rule, set value 1, to add a rule—value 4, to remove a rule—value 2. |
| firewallApply             | Apply rules   |
| firewallConfirm           | Confirm applied rules   |



**These settings match ones described in Section 5.1.1.7.**

### Service features

Object identifier enterprises.35265.1.9.

|               |   |
|---------------|---|
| fxsConfigSave | Save configuration into non-volatile memory |
| fxsReboot     | Reboot gateway                              |

#### 5.1.1.5.3. Device Firmware Update

To do this, send 'set' request to OID 1.3.6.1.4.1.35265.1.9.25.0

Request format: <TFTP server IP address> <Firmware file name>

Example: 192.168.16.44 firmware.img32m

SNMP trap message will be sent to notify you on success or failure of firmware update operation.

---

#### 5.1.1.5.4. *Device configuration download/upload*

##### ***Device configuration upload***

To do this, send 'set' request to OID .1.3.6.1.4.1.35265.4.10.2.0

Request format: <TFTP server IP address> <Configuration file name> upload.

Example: 192.168.16.44 cfgTau32.crypt upload

##### ***Device configuration download***

To do this, send 'set' request to OID .1.3.6.1.4.1.35265.4.10.2.0

Request format: <TFTP server IP address> <Configuration file name> download

Example: 192.168.16.44 cfgTau32.crypt download

##### ***Apply loaded changes***

To do this, send 'set' request to OID .1.3.6.1.4.1.35265.4.10.2.0

Request format: <TFTP server IP address> <Configuration file name> apply

Example: 192.168.16.44 cfgTau32.crypt apply

### 5.1.1.5.5. Description of events sent in Trap, Trapv2, Inform messages

Table 7—Description of events sent in Trap, Trap2, Inform messages

| Event                     | Severity | Description  | OID                      | Note  |
|---------------------------|----------|--|--------------------------|---|
| fxs72VbatAlarmTrap        | MAJOR    | Voltage Vbat = %1\$d V falls outside of allowable limits (38–72V)        | 1.3.6.1.4.1.35265.3.6.1  | Parameter 1: Voltage  |
| fxs72VringAlarmTrap       | MAJOR    | Voltage Vring %2\$d = %1\$d falls outside of allowable limits (100–120V) | 1.3.6.1.4.1.35265.3.6.2  | Parameter 1: Voltage<br>Parameter 2: Magneto generator number (1 or 2)      |
| fxs72TemperatureAlarmTrap | MAJOR    | Sensor temperature %2\$d = %1\$d exceeds the permitted value (90°C)      | 1.3.6.1.4.1.35265.3.6.3  | Parameter 1: Temperature<br>Parameter 2: Temperature sensor number (1 of 4) |
| fxs72FanAlarmTrap         | MAJOR    | Fan %1\$d is on, no rotation   | 1.3.6.1.4.1.35265.3.6.4  | Parameter 1: Fan number   |
| fxs72SSwAlarmTrap         | MAJOR    | No registration at MGC/SSW   | 1.3.6.1.4.1.35265.3.6.5  | Used in Megaco firmware version   |
| fxs72PortAlarmTrap        | MINOR    | Port %1\$d is blocked  | 1.3.6.1.4.1.35265.3.6.6  | Parameter 1: Port number  |
| fxs72VbatOkTrap           | CLEAR    | Vbat voltage is OK   | 1.3.6.1.4.1.35265.3.7.1  |   |
| fxs72VringOkTrap          | CLEAR    | Vring %2\$d voltage is OK  | 1.3.6.1.4.1.35265.3.7.2  | Parameter 2: Magneto generator number (1 or 2)                              |
| fxs72TemperatureOkTrap    | CLEAR    | Sensor temperature %2\$d is OK   | 1.3.6.1.4.1.35265.3.7.3  | Parameter 2: Temperature sensor number (1 of 4)                             |
| fxs72FanOkTrap            | CLEAR    | Fan %1\$d is in normal operation   | 1.3.6.1.4.1.35265.3.7.4  | Parameter 1: Fan number   |
| fxs72SSwOkTrap            | CLEAR    | Registration at MGC/SSW  | 1.3.6.1.4.1.35265.3.7.5  | Used in Megaco firmware version   |
| fxs72PortOkTrap           | CLEAR    | Port %1\$d is unblocked  | 1.3.6.1.4.1.35265.3.7.6  | Parameter 1: Port number  |
| fxs72VmodeSwitchTrap      | INFO     | Power mode changed – %1\$d V   | 1.3.6.1.4.1.35265.3.7.10 | Parameter 1: new mode: 1 – 60V, 2 – 48V                                     |

|                     |          |                               |                          |                              |
|---------------------|----------|-------------------------------|--------------------------|------------------------------|
| fxs72FansSwitchTrap | INFO     | Fan status changed            | 1.3.6.1.4.1.35265.3.7.11 | Parameter 1: 0 – off, 1 – on |
| fxs72updateFwFail   | MINOR    | Firmware update error         | 1.3.6.1.4.1.35265.3.6.20 | Parameter 1: error type      |
| fxs72updateFwOk     | INFO     | Firmware successfully updated | 1.3.6.1.4.1.35265.3.7.20 |                              |
| fxs72BpuAlarmTrap   | CRITICAL | No BPU connection             | 1.3.6.1.4.1.35265.3.6.12 |                              |
| fxs72BpuOkTrap      | CLEAR    | BPU connection restored       | 1.3.6.1.4.1.35265.3.7.12 |                              |

### 5.1.1.6. Syslog Protocol Configuration

In 'Syslog' menu, you may configure system log settings.

**SYSLOG** is a protocol, designed for transmission of messages on current system events. Gateway software generates system data logs on operation of system applications and signalling protocols, as well as occurred failures and sends them to SYSLOG server.



**High debug levels may cause delays in operation of the device.  
IT IS NOT RECOMMENDED to use system log without due cause.**



**System log should be used only when problems in gateway operation occur, and you have to identify the reason. To define the necessary debug levels, consult a Eltex Service Centre Specialist.**

#### Syslog configuration:

- *Run syslog on startup*—when checked, run Syslog on device startup.
- *Syslog to file*—when checked, save Syslog into file to view it later via web interface.
- *Syslog server*—Syslog server IP address.
- *Syslog Port*—port for Syslog server incoming messages (514 by default).

#### Record type (APPLICATION):

- *Error*—send application failure messages to Syslog server.
- *Warning*—send application warning messages to Syslog server.
- *Info*—send application Info messages to Syslog server.
- *Debug*—send application debug messages to Syslog server.
- *Alarm*—send alarm event messages to Syslog server.

SIP:

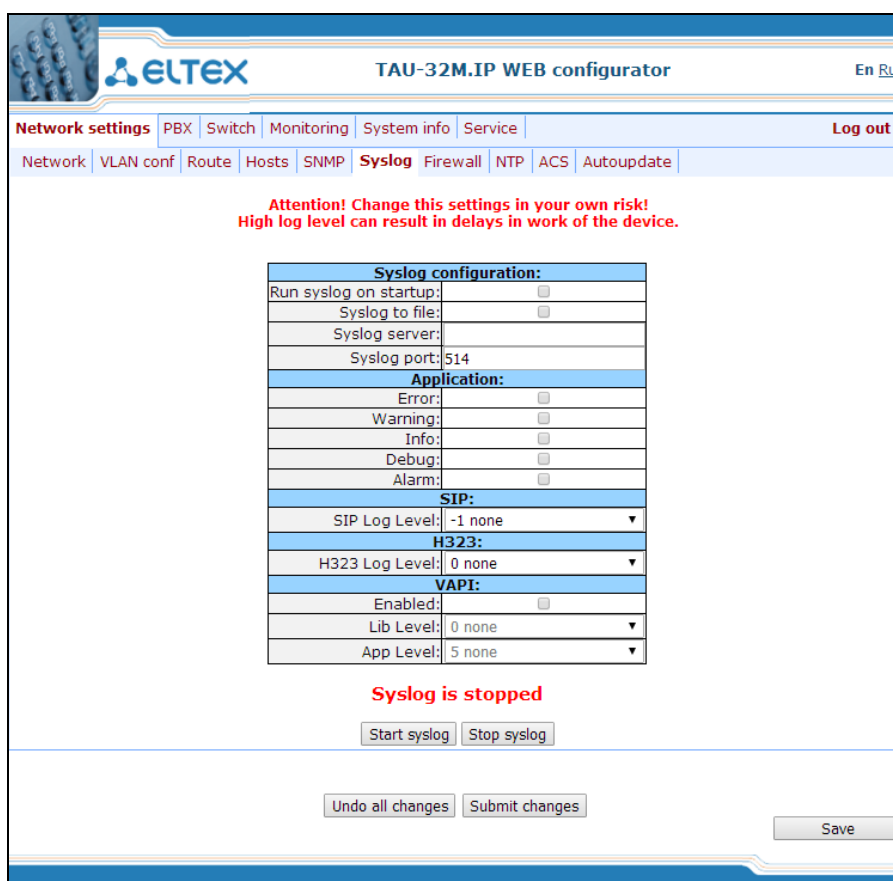
- *SIP Log Level*—SIP protocol log level.

H.323:

- *H.323 Log Level*—H.323 protocol log level.

VAPI:

- *Enabled*—when checked, VAPI library logging is enabled, otherwise it is disabled.
- *Lib Level*—VAPI library log level.
- *App Level*—VAPI log level from the application side.



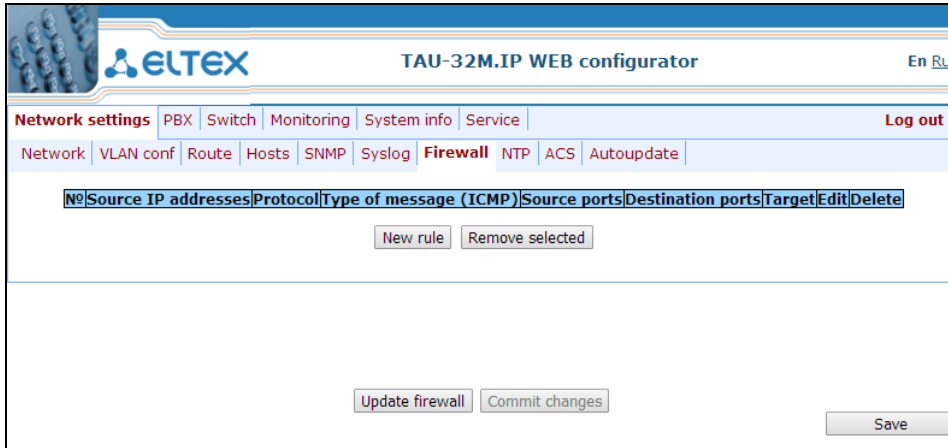
Use 'Start' and 'Stop' buttons to start and stop the output of logging information to the system log.

Use 'Show' and 'Clear' buttons available in syslog file saving mode to view the log via web interface and clear the log on the device.

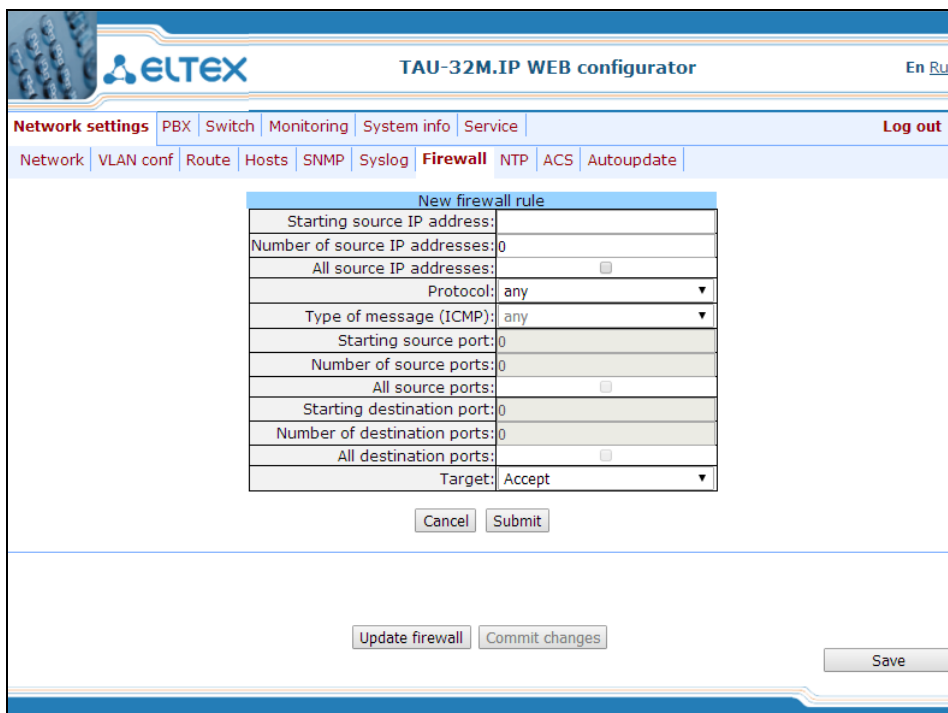
To discard all changes made to configuration, click 'Undo All Changes' button. To apply changes, click 'Submit Changes' button.

### 5.1.1.7. Firewall configuration

In 'Firewall' submenu, you may configure black and white lists of IP addresses to allow or deny them access to the device.



To add a new rule, click 'New rule' button.



#### New firewall rule:

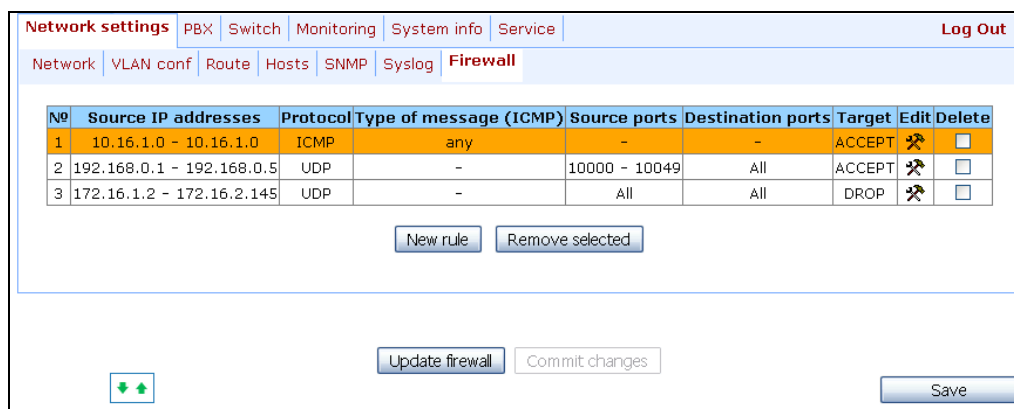
- *Starting source IP address*—starting IP address of the range of packet sources.
- *Number of source IP addresses*—number of IP addresses in the range of packet sources.
- *All source IP addresses*—when checked, the rule applies to all packet source IP addresses.
- *Protocol*—type of incoming packets' protocol that the rule to be applied to:
  - *Any*—for UDP and TCP

- *UDP*—for UDP
- *TCP*—for TCP
- *ICMP*—for ICMP

- *Type of message (ICMP)*—type of ICMP message that the rule is created for.
- *Starting source port*—starting TCP/UDP port of the source port range.
- *Number of source ports*—number of ports in the source port range.
- *All source ports*—when checked, the rule applies to packets with any source port value.
- *Starting destination port*—starting TCP/UDP port (on the device) of the packet destination port range.
- *Number of destination ports*—number of ports in the packet destination port range.
- *All destination ports*—when checked, the rule applies to packets with any destination port value.
- *Target*—action to be performed on packets falling under this rule:

- *Accept*
- *DROP*
- *REJECT*

To apply a new rule, click '*Submit*' button.



To edit the rule, click icon in '*Edit*' column for the respective rule.

To change the rule sequence, select the necessary rule and move it to the desired position with buttons.

After all necessary rules has been added, click '*Update firewall*' button to apply the rules. Next, you should click '*Commit changes*' button in two minute interval after approving new rules, otherwise previous settings will be restored.

To discard all changes made to configuration, click '*Undo All Changes*' button. To store changes to non-volatile memory of the device, click '*Save*' button.

### 5.1.1.8. NTP Configuration

**NTP** is a protocol designed for synchronization of real-time clock of the device. Allows to synchronize date and time used by the gateway against their reference values.

| NTP Settings:                |                                 |
|------------------------------|---------------------------------|
| Enable NTP:                  | <input type="checkbox"/>        |
| NTP server:                  |                                 |
| Enable synchronization:      | <input type="checkbox"/>        |
| Synchronization period, sec: | 0                               |
| Zone info:                   | GMT0   Default DST              |
| DST enable:                  | <input type="checkbox"/>        |
| DST start:                   | -   -   in   -   at   -   -   - |
| DST end:                     | -   -   in   -   at   -   -   - |
| DST offset, min:             | 60                              |

- *Enable NTP*—when checked, enable the synchronization of the device time with an external server via NTP protocol. Given that TAU is not equipped with real-time clock, in order to use the real time in monitoring and statistics tasks you should enable time synchronization with an external server.
- *NTP server*—NTP server address
- *Enable synchronization*—when checked, perform periodic synchronization of the device with NTP server.
- *Synchronization period*—period of synchronization with NTP server (permissible value: 30 to 100000s).
- *Zone info*—timezone. Given that NTP server sends the time in a zero timezone, this setting allows to set local time on the device. If you need help on timezones, see **Appendix K**.



**Exclamation mark symbol means that DST settings are not used for this timezone!**



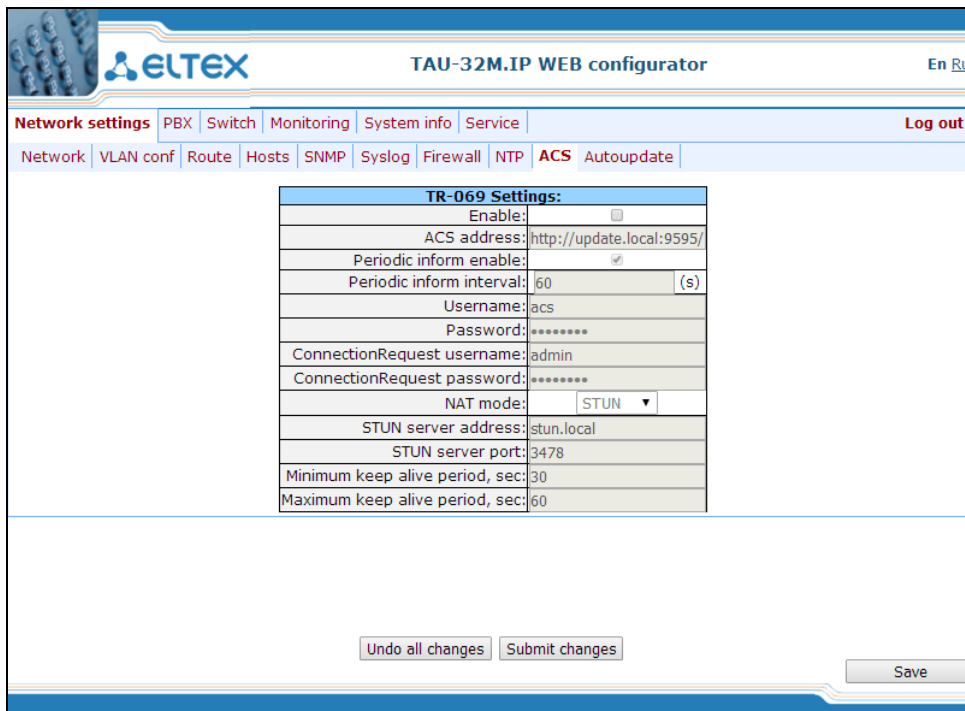
**DST settings will be applied only after the device is restarted!**

- *DST enable*—when checked, device will perform daylight saving change and the set back process.
- *Default DST*—allows to set standard DST periods for the current timezone.
- *DST start*—defines the moment of daylight saving change.
- *DST end*—defines the moment of set back process.
- *DST offset, min*—time adjustment amount used in transition.

To discard all changes made to configuration, click 'Undo All Changes' button. To apply changes, click 'Submit Changes' button.



### 5.1.1.9. TR-069 Monitoring and Management Protocol Configuration (ACS)



| TR-069 Settings:                |                                     |
|---------------------------------|-------------------------------------|
| Enable:                         | <input type="checkbox"/>            |
| ACS address:                    | http://update.local:9595/           |
| Periodic inform enable:         | <input checked="" type="checkbox"/> |
| Periodic inform interval:       | 60 (s)                              |
| Username:                       | acs                                 |
| Password:                       | *****                               |
| ConnectionRequest username:     | admin                               |
| ConnectionRequest password:     | *****                               |
| NAT mode:                       | STUN                                |
| STUN server address:            | stun.local                          |
| STUN server port:               | 3478                                |
| Minimum keep alive period, sec: | 30                                  |
| Maximum keep alive period, sec: | 60                                  |

Buttons: Undo all changes, Submit changes, Save

#### TR-069 Monitoring and Management Protocol Settings (TR-069 Settings):

- *Enable*—when checked, enable device management via TR-069 protocol.
- *ACS address*—ACS server address. Enter address in the following format: http://<address>:<port> (<address>—ACS server IP address or domain name, <port>—ACS server port, 10301 by default).
- *Periodic inform enable*—when checked, integrated TR-069 client will periodically poll ACS server at intervals equal to 'Periodic inform interval' value in seconds. Goal of the polling is to identify possible changes in the device configuration.
- *Periodic inform interval*—ACS server polling interval.
- *Username*—username used by client to access the ACS server.
- *Password*—password used by client to access the ACS server.
- *ConnectionRequest username*—username used by ACS server to access the TR-069 client. Server sends ConnectionRequest notifications.
- *ConnectionRequest password*—password used by ACS server to access the TR-069 client. Server sends ConnectionRequest notifications.

If there is a NAT (network address translation) between the client and ACS server, ACS server may not be able to establish the connection to client without specific technologies intended to prevent such situations. These technologies allow the client to identify its so called public address (NAT address or in other words external address of a gateway, that covers the client.) When public address is identified, the client reports it to the server that uses this public address for establishing connection to the client in the future.

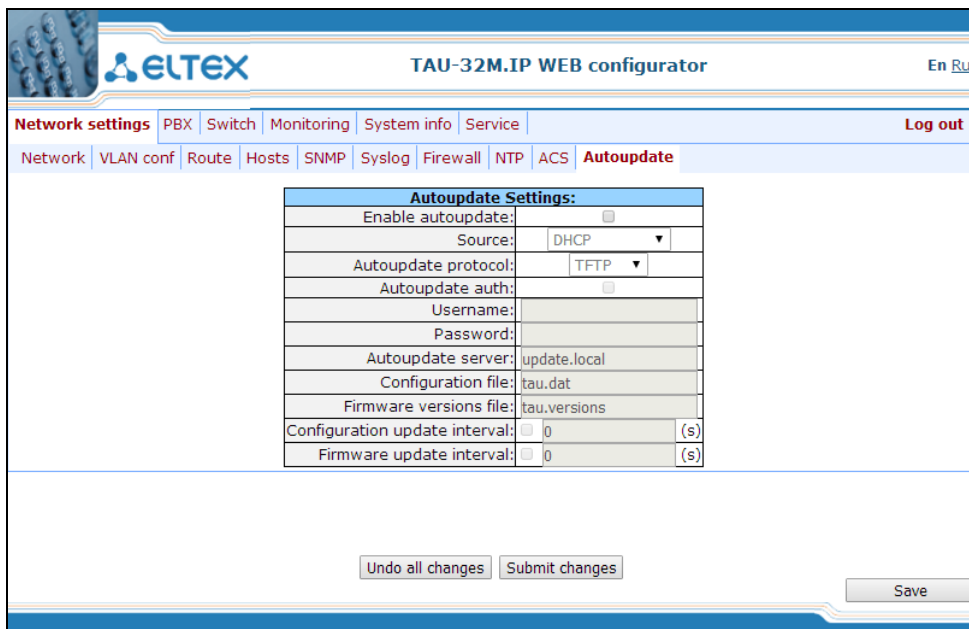
- *NAT mode*—TR-069 client operation mode in the presence of NAT; identifies the method, that will be

used by client for obtaining its public address information. Available modes:

- *STUN*—use STUN protocol for public address identification. When choosing STUN client operation mode, you should define the following settings:
  - *STUN server address*—STUN server IP address or domain name.
  - *STUN server port*—STUN server UDP port (3478 by default).
  - *Minimum keep alive period, seconds and Maximum keep alive period, seconds*—define the time interval in seconds for periodic transmission of messages to STUN server for public address discovery and modification.
- *Public address (Manual)*—manual mode, when public address is explicit in configuration; in this mode, you should add a forwarding rule on a device that acts as a NAT for TCP port used by TR-069 client. When the manual mode client ('Manual') is selected, the public client address should be specified manually.
  - *NAT address*—IP address of a public NAT.
- *Off*—NAT will no be used—this mode is recommended only when the device is directly connected to ACS server without network address translation. In this case public address will match local client address.

To discard all changes made to configuration, click '*Undo All Changes*' button. To apply changes, click '*Submit Changes*' button.

#### 5.1.1.10. Automatic update configuration (Autoupdate)



| Autoupdate Settings:           |                          |
|--------------------------------|--------------------------|
| Enable autoupdate:             | <input type="checkbox"/> |
| Source:                        | DHCP                     |
| Autoupdate protocol:           | TFTP                     |
| Autoupdate auth:               | <input type="checkbox"/> |
| Username:                      |                          |
| Password:                      |                          |
| Autoupdate server:             | update.local             |
| Configuration file:            | tau.dat                  |
| Firmware versions file:        | tau.versions             |
| Configuration update interval: | 0 (s)                    |
| Firmware update interval:      | 0 (s)                    |

#### Automatic update settings (Autoupdate)

- *Enable autoupdate*—when checked, device configuration and firmware will be updated automatically.
- *Source*—parameter obtaining method for autoupdate procedure.

- *DHCP (VLAN 1, VLAN 2, VLAN 3)*—receive autoupdate parameters via DHCP Options 66 and 67.
- *Static*—use autoupdate parameters specified in TAU-36.IP/TAU-72.IP configuration.
- *Autoupdate protocol*—a protocol, which will be used for autoupdate (TFTP/FTP/HTTP/HTTPS).
- *Autoupdate auth*—when checked, authentication settings will be used during autoupdate procedure.
- *Username*—login to access the autoupdate server.
- *Password*—password to access the autoupdate server.
- *Autoupdate server*—autoupdate server IP address or network name.
- *Configuration file*—name of the configuration file located on autoupdate server and its path.
- *Firmware versions file*—name of the firmware versions file located on autoupdate server and its path.
- *Configuration update interval*—when checked, automatically update configuration with the specified period in seconds.
- *Firmware update interval*—when checked, automatically update firmware with the specified period in seconds.

For autoupdate system operating procedure, see **Appendix F. Automatic Configuration Procedure and Gateway Firmware Version Check.**

To discard all changes made to configuration, click '*Undo All Changes*' button. To apply changes, click '*Submit Changes*' button.

In addition to static configuration of TR-069 client, the device supports DHCP Option 43 processing in the following format:

**<suboption number><suboption length><suboption value>,**

where

- Suboption number and length are passed in a numeric (Hex) format
- Suboption value is passed as ASCII code

Gateway recognizes the following suboptions:

- 1—*ACS URL*—ACS server URL.

Address should be received in the following format: **http://<address>:<port>**,

where

<address>—ACS server IP address or domain name

<port>—ACS server port number, 10301 by default (optional parameter)

- 2—*Provisioning code*—identifier that allows ACS server to identify specific configuration parameters.
- 3—*Login*—username used by client to access the ACS server.

- 4—*Password*—password used by client to access the ACS server.
- 5—autoupdate server address.

Address should be received in the following format: **<proto>://<address>[:<port>]**,

where

<proto>—protocol (FTP, TFTP, HTTP, HTTPS).

<address>—autoupdate server IP address or domain name.

<port>—autoupdate server port (optional parameter).

- 6—autoupdate configuration file name.
- 7—autoupdate firmware file name.

Upon receiving Option 43, suboption 1, device launches management via TR-069 protocol.

Example of the option record:

```
01:10:68:74:74:70:3A:2F:2F:61:63:73:2E:72:75:3A:38:30:02:02:31:39:03:03:61:63:73:04:06:61:63:73:61:63:73
```

where

01—*ACS URL* suboption number.

10—length, 16bytes (0x10 = 16 dec)

68:74:74:70:3A:2F:2F:61:63:73:2E:72:75:3A:38:30—suboption value (<http://acs.ru:80>).

02—*Provisioning code* suboption number.

02—length, 2bytes.

31:39—suboption value (19).

03—Login suboption value.

03—length, 3bytes.

61:63:73—suboption value (acs).

04—Password suboption value.

06—length, 6bytes.

61:63:73:61:63:73—suboption value (acsacs).

### 5.1.2. VoIP Configuration (PBX)

In 'PBX' menu, you can configure VoIP (Voice over IP): SIP/H.323 protocol configuration, TCP/IP protocol configuration, FXS interface configuration, installation of codecs, numbering schedule, etc.

### 5.1.2.1. Basic Configuration (Main)

In 'Basic Configuration' ('Main') submenu, you can configure basic device settings: set the device name, device prefix, and global timers.

| General configuration:      |  |
|-----------------------------|--|
| Device name:                | <input type="text" value="tau72"/>                     |
| Start timer:                | <input type="text" value="300"/> (sec, from 10 to 300) |
| Duration timer:             | <input type="text" value="300"/> (sec, from 10 to 300) |
| Wait answer timer:          | <input type="text" value="180"/> (sec, from 40 to 300) |
| Fans threshold temperature: | <input type="text" value="55"/> (°C, from 35 to 55)    |
| Fans force enable:          | <input type="checkbox"/>                               |

For rev. B:

The screenshot shows the 'TAU-32M.IP WEB configurator' interface. The 'General configuration' section is highlighted, showing the following settings:

| General configuration:      |  |
|-----------------------------|--|
| Device name:                | tau32  |
| Use prefix (SIP-T):         | <input type="checkbox"/>                               |
| Prefix (SIP-T):             |  |
| Start timer:                | <input type="text" value="15"/> (sec, from 10 to 300)  |
| Duration timer:             | <input type="text" value="10"/> (sec, from 10 to 300)  |
| Wait answer timer:          | <input type="text" value="180"/> (sec, from 40 to 300) |
| Fans threshold temperature: | <input type="text" value="55"/> (°C, from 35 to 55)    |
| Fans force enable:          | <input type="checkbox"/>                               |

#### General configuration:

- *Device name*—name of the device. Used for sending messages to SYSLOG server, enables device identification.
- *Use prefix (SIP-T)*—when checked, *Prefix (SIP-T)* parameter value will be used as a PBX prefix. This prefix will be added before the subscriber's number and will affect the number type: if the prefix is present, subscriber's number will be 'national'; if it is absent, then the number will be 'subscriber' (passed in CgPN parameter).
- *Prefix (SIP-T)*—PBX prefix (numeric string).



***Use prefix (SIP-T) and Prefix (SIP-T) parameters are used only in gateway operation via SIP-T protocol. SIP-T protocol operation mode is defined by: in incoming communications—the presence of ISUP attachment in initializing SIP INVITE request, in outgoing communications—SIP-T protocol configuration in routing prefix (see Section 5.1.2.2.6.1 Configuration of Routing Rules).***

- Start timer—*dialling timeout for the first digit of a number; when there is no dialling during the specified time, 'busy' tone will be sent to the subscriber, and the dialling will end.*
- *Duration timer*—complete number dialling timeout. Takes effect after the first digit of a number has been dialled, and specifies the time for dialling the full number.
- *Wait answer timer*—subscriber's response timeout for incoming and outgoing calls. If the subscriber fails to answer in the specified time, the call will be cleared back.
- *Fans threshold temperature*—device threshold temperature that causes cooling fans to start, permitted values from 35°C to 55°C.
- *Fans force enable*<sup>1</sup>—when checked, threshold temperature measurement feature will be disabled, and fans will operate continuously.
- *Extended range loop*<sup>2</sup>—enable extended range mode. If 'Extended range loop' option is not set, power supply voltage of subscriber units equals to 34V, current in a closed loop—22mA. Maximum loop resistance is 1.5kΩ. Fans will start only when submodule sensor temperature exceeds 95°C (ambient temperature 43-46°C), also they will start at their minimum speed. If 'Extended range loop' option is set, power supply voltage of subscriber units equals to 54V, current in a closed loop—25mA. Maximum loop resistance is 2.1kΩ. In this mode, fans will start using the following algorithm:

- When temperature of any submodule exceeds the threshold value ('*Fans threshold temperature*'), fans will start at 1/2 speed.
- When temperature of any submodule exceeds the threshold value by 5°C, fans will start at 5/8 speed.
- When temperature of any submodule exceeds the threshold value by 10°C, fans will start at 6/8 speed.
- When temperature of any submodule exceeds the threshold value by 15°C, fans will start at 7/8 speed.
- When temperature of any submodule exceeds the threshold value by 20°C, fans will start at maximum speed.

To apply changes, click '*Submit Changes*' button. To discard all changes made to configuration, click '*Undo All Changes*' button. To store changes to non-volatile memory of the device, click '*Save*' button.

#### **5.1.2.2. SIP/H323 Profiles**

In '*SIP/H323 Profiles*' submenu, you may configure SIP profiles and H.323 protocol. You may organize gateway operation with multiple carriers by configuring various SIP profiles on subscriber ports.

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<sup>1</sup> This parameter is effective for TAU-32M.IP rev. 1 only (revision version for this board type is not shown in the web interface header bar)

<sup>2</sup> This parameter is effective for TAU-32M.IP rev. B only (revision version for this board type is shown in the web interface header bar)

### 5.1.2.2.1. SIP Common Parameters (SIP Common)

In 'SIP Common' tab, you may configure common SIP protocol parameters applied to all profiles.

**SIP** (Session Initiation Protocol) is a signalling protocol, used in IP telephony. It performs basic call management tasks such as starting and finishing session.

Addressing in SIP network based on SIP URI scheme:

**sip:user@host:port;uri-parameters**

where:

**user**—number of a SIP subscriber.

**@**—separator located between the number and domain of a SIP subscriber.

**host**—domain or IP address of a SIP subscriber.

**port**—UDP port used for subscriber's SIP service operation.

**uri-parameters**—additional parameters.

One of the additional SIP URI parameters: user=phone. When this parameter is used, SIP subscriber number syntax should match TEL URI syntax described in RFC 3966. In this case, TAU-32M.IP will not clear-back calls, if SIP subscriber's number contains the following characters: '+', ';', '=', '?'. Allows you to receive calls from numbers beginning with '+' character.

The screenshot shows the 'SIP Common' configuration page in the TAU-32M.IP WEB configurator. At the top, there is a warning: "Attention! Changing of these parameters will lead to aborting of all calls!". Below the warning is a table of configuration parameters:

| SIP configuration:                           |                                     |
|--|-------------------------------------|
| Enable SIP:                                  | <input checked="" type="checkbox"/> |
| Invite initial timeout (ms):                 | 500                                 |
| Invite total timeout (ms):                   | 32000                               |
| Short mode:                                  | <input type="checkbox"/>            |
| Transport:                                   | UDP(preffered),TCP ▾                |
| SIP UDP MTU (for "udp(preffered),tcp" mode): | 1300                                |
| Port registration delay (ms):                | 500                                 |
| Work through NAT:                            |                                     |
| Use STUN:                                    | <input type="checkbox"/>            |
| STUN server:                                 |                                     |
| STUN interval:                               | 300                                 |
| PublicIP:                                    |                                     |

At the bottom of the table, there are three buttons: "Undo all changes", "Defaults", and "Submit changes". A "Save" button is located at the bottom right of the page.



**You don't have to reboot the gateway in order to apply SIP settings. When applying settings, all current calls will be terminated.**

SIP configuration:

- *Enable SIP*—when checked, SIP is enabled.
- *Invite initial timeout (ms)*—time interval between first and second INVITEs, when there is no response to the first one, in ms; the interval will be doubled for subsequent INVITEs (third, fourth, etc.) (e.g. for 300ms, the second INVITE will be sent in 300ms, the third is in 600ms, the fourth is in 1200ms, etc).
- *Invite total timeout (ms)*—total timeout for INVITE message transmission, in milliseconds. When this timeout expires, the direction is deemed to be unavailable. Allows to limit INVITE message retransmission, including messages used for SIP proxy availability identification.

*Invite total timeout* parameter is calculated depending on the required number of INVITE message retransmissions and the time interval between first and second INVITEs—*Invite initial timeout*—using the following equation:

$$\text{Invite total timeout} = 100 + \sum_{n=0}^{N-1} (2^n) \cdot \text{Invite\_initial\_timeout}$$

where N is a number of INVITE message retransmissions. For example, in order to switch to redundant SIP-proxy, when there is no response to three INVITE messages and *Invite initial timeout* parameter value equals to 300ms, Invite total timeout should be:  $100 + 300 \cdot 1 + 300 \cdot 2 + 300 \cdot 4 = 2200\text{ms}$ .

- *Short mode*—when checked, use shortened field names in SIP protocol header, otherwise use complete names. Also, spaces will be removed from parameter strings in this mode.
- *Transport*—select transport layer protocol, used for SIP message transmission:
  - *udp(preferred),tcp*—use both UDP and TCP protocols, but UDP priority will be higher.
  - *tcp(preferred),udp*—use both UDP and TCP protocols, but TCP priority will be higher.
  - *udp only*—use UDP protocol only.
  - *tcp only*—use TCP protocol only.
- *SIP UDP MTU (for "udp(preffered),tcp" mode)*—maximum SIP protocol data size in bytes, sent with UDP transport protocol (according to RFC3261, recommended value is 1300). If SIP protocol data size exceeds specified value (it is possible, e.g. when qop authentication is used), TCP will be used as a transport protocol. This example applies to *udp(preferred), tcp* mode only.
- *Port registration delay (ms)*—delay between successive registrations of neighbouring gateway ports. Default value is 500ms. Longer delay may be necessary when the gateway operates through SBC that can temporarily block the reception of messages from gateway IP address or blacklist the gateway in case of large numbers of REGISTER queries.

Work through NAT:

When TAU gateway is located behind a NAT, it is necessary to discover an external NAT IP address for voice and signal traffic delivery to the gateway.





If NAT is used for incoming calls to the gateway, NAT address may be specified in request URI. Therefore, in order to process calls, you should set '*Full RURI compliance*' option in SIP profile!

- *Use STUN*—use STUN protocol for public NAT address discovery.



This setting is available only if the gateway operates via SIP protocol with UDP transport, i.e. the value of *Transport* parameter should be *udp only*.

- *STUN server*—STUN server IP address.
- *STUN interval*—STUN server polling period.
- *Public IP*—this setting contains a public NAT address to be used in cases, when it cannot be obtained via STUN protocol. This setting cannot be used in cases, when NAT dynamically obtains its external IP address.

Use '*Defaults*' button to set default parameters (the figure below shows default values).

To apply changes, click '*Submit Changes*' button; to discard all changes, click '*Undo All Changes*' button; to save changes, click '*Save*' button.

#### 5.1.2.2.2. **SIP-T Protocol Configuration**

Configure the following parameters to utilize SIP-T protocol:

1. If you need to define a '*national*' value for subscriber number type, configure the following parameters: *Use prefix (SIP-T)* and *Prefix (SIP-T)*. For description of parameters, see Section 5.1.2.1 Basic Configuration (Main).
2. To route outgoing calls via SIP-T protocol, you should configure prefixes with the corresponding protocol (Protocol & Target: SIP-T Direct IP) and the type of the number fetched by the prefix (Number type). For description of parameters, see Section 5.1.2.2.6.1.
3. To assign Caller ID category to the subscriber, use SS7 category (SIP-T) parameter in subscriber port configuration or subscriber profile. For description of parameters, see Section 5.1.2.4 Configuration of Subscriber Ports (Ports).
4. To receive international calls with '+' symbol preceding the number, you should configure '*User=Phone*' option, see Section 5.1.2.2.4).

### 5.1.2.2.3. H.323 Protocol

In 'H.323' submenu, you can configure H.323 protocol settings.



**H.323 protocol operation is possible only when Profile 1 is used. Use Profile 1 to configure codecs and routing when H.323 protocol is used.**

H.323 standard states specifications for audio and video data transmission via data networks and includes standards for video and voice codecs, public domain applications, call and system management.

H.323 stack of TAU-32M.IP gateway supports the following protocols:

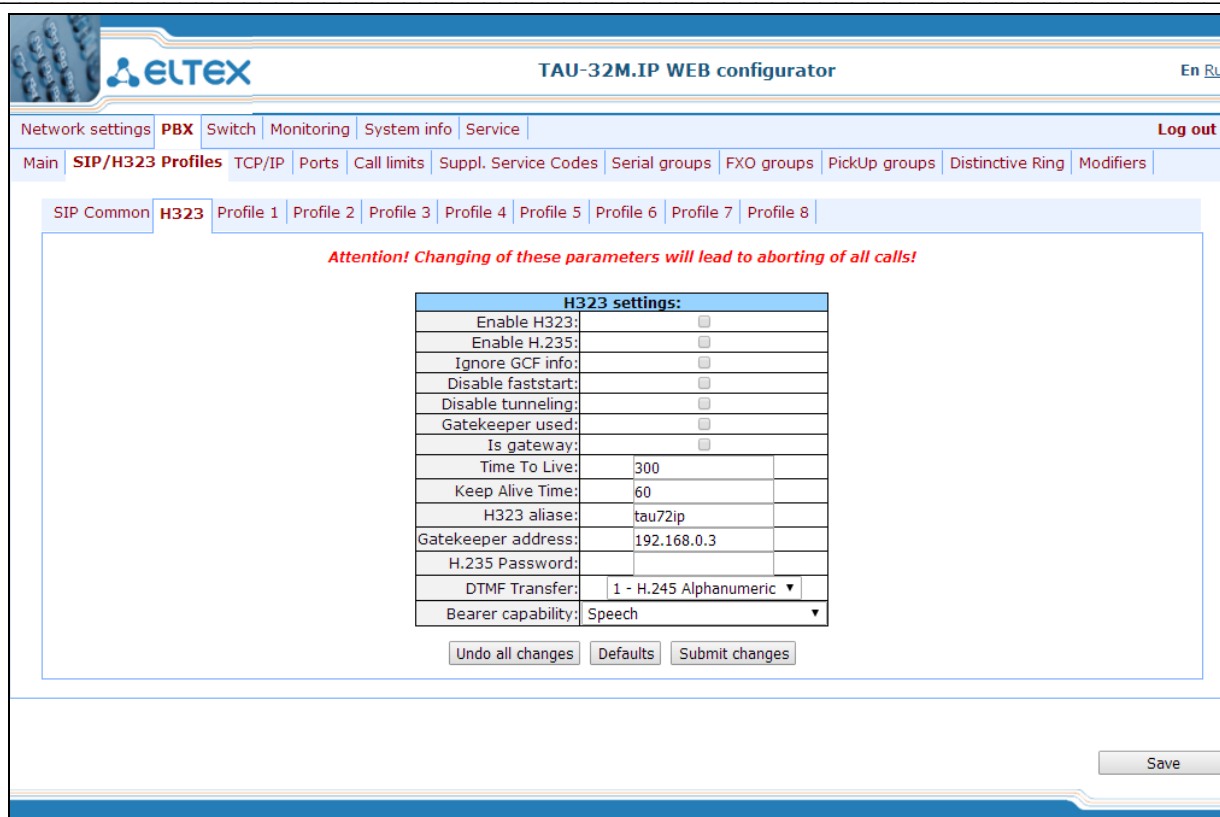
- *H.245*—allows for codec matching and opening of voice connection when faststart procedure is not used.
- *Q.931/H.225*—allows to establish and control a connection.
- *RAS*—allows for gatekeeper interactions.
- *H.235*—authenticates calls during gatekeeper interactions.
- *H.450.1*—used during put on/remove from hold.

**Gatekeeper** allows for call processing inside its zone and interaction with other zones as well as call management. During gatekeeper operations, the gateway should register on the gatekeeper and perform authorization using login and password (H.235) depending on the local network policy. Only after successful registration gateway subscribers will be able to perform calls through the gatekeeper. Gateway registers on the gatekeeper for a limited amount of time—*Time to live* (TTL)—during which it should renew its registration. Keep alive timer is used for this purpose; upon expiration, the gateway sends a renewal request.

Faststart procedure enables fast establishment of a voice connection. In this case, channel will be established before the start of capability coordination with H.245 protocol. Tunnelling procedure allows to transfer H.245 signalling via Q.931 signal channels. As a result, no additional TCP connection (or TCP port) is required for capability coordination.



**You don't have to reboot the gateway in order to apply H.323 settings. When applying settings, all current calls will be terminated.**



After implementation of changes, click 'Submit Changes' button; to discard all changes, click 'Undo All Changes' button; to save changes, click 'Save' button.

Use 'Defaults' button to set default parameters (the figure below shows default values).

### H323 settings:

- *Enable H323*—when checked, H.323 protocol is enabled.
- *Enable H.235*—when checked, use authentication on the gatekeeper with H.235 protocol.
- *Ignore GCF info*—when checked, output authentication data in RRQ message via H.235 protocol in any events, otherwise—only in case of reception of supported hash method in GCF message. This setting applies to operations with gatekeepers that do not send used hash method in a response to GRQ request. In this case, the gateway will transfer MD5-encrypted authentication data for all RRQs, even if supported hash method is not received from the gatekeeper.
- *Disable faststart*—when checked, *faststart* feature will be disabled.
- *Disable tunneling*—when checked, H.245 signal tunnelling through Q.931 signal channels will be disabled.
- *Gatekeeper used*—when checked, use gatekeeper registration option.
- *Is gateway*—when checked, device registers on a gatekeeper as a gateway, otherwise—as a terminal device. When registered as a terminal device, the gateway registers all configured subscribers' numbers and a gateway name—H.323 alias—on a gatekeeper. When registered as a gateway, the gateway registers its name—H.323 alias—only. To simplify the gatekeeper configuration, we recommend using registration as a terminal device.
- *Time To Live*—time period in seconds, for which the device will keep its registration on a gatekeeper.

- *Keep Alive Time*—time period in seconds, after which the device will renew its registration on a gatekeeper.
- *H.323 alias*—name for registration on a gatekeeper.
- *Gatekeeper address*—IP address of a gatekeeper.
- *H.235 password*—password used for H.235 protocol authentication.
- *DTMF Transfer*—select transfer method for flash and DTMF tones via H.323 protocol (H.245 Alphanumeric, H.245 Signal, Q931 Keypad IE). Transfer of DTMF tones enables extension dialling feature.
  - H.245 Alphanumeric—basicstring *compatibility is used for DTMF transmission, and hookflash compatibility for flash transmission (flash is transferred as '!' symbol)*.
  - H.245 Signal—*dtmf compatibility is used for DTMF transmission, and hookflash compatibility for flash transmission (flash is transferred as '!' symbol)*.
  - Q931 Keypad IE—*for DTMF and flash transmission (flash is transferred as '!' symbol), Keypad information element is used in INFORMATION Q931 message*.
- *Bearer capability*—select information transfer service (*Speech, Unrestricted Digital, Restricted Digital, 3.1 kHz Audio, unrestricted Digitals with Tones*). We recommend using value '3.1 kHz Audio'. All other values used only for compatibility with communicating gateways.



**'DTMF Transfer' item will be used only if there is an item 2—INFO— is selected in DTMF Transfer item of the Codecs conf.**



**To ensure the successful renewal of device registration on gatekeeper, specify *Keep Alive Time* renewal period equal to 2/3 of *Time To Live* registration period. Moreover, for *Time To Live* parameter, we recommend specifying the same value as for the gatekeeper, so the registration renewal period—*Keep Alive Time*—of the gateway was less or equal to *Time To Live* value (transferred in responses). Otherwise, invalid configuration may lead to situations, where gatekeeper will void the gateway registration before the renewal, which in turn may lead to termination of all active connections, established through the gatekeeper.**

To apply changes, click '*Submit Changes*' button. To discard all changes made to configuration, click '*Undo All Changes*' button.

### 5.1.2.2.4. SIP Custom Parameters (SIP Custom)

In 'Profile n/SIP Custom' tab, you may configure SIP protocol parameters for each profile.



**You don't have to reboot the gateway in order to apply SIP settings. When applying settings, all current calls will be terminated.**

ELTEX TAU-32M.IP WEB configurator

Network settings **PBX** Switch Monitoring System info Service Log out

Main **SIP/H323 Profiles** TCP/IP Ports Call limits Suppl. Service Codes Serial groups FXO groups PickUp groups Distinctive Ring Modifiers

SIP Common H323 **Profile 1** Profile 2 Profile 3 Profile 4 Profile 5 Profile 6 Profile 7 Profile 8

**SIP Custom** Codecs Dialplan Alert-Info

*Attention! Changing of these parameters will lead to aborting of all calls!*

| SIP configuration:                       |   |  |
|--|---|--|
| Proxy mode:                              | Off                                       | ▼  |
| Proxy / Registrar / Use registration 1:  | 192.168.10.101                            | 192.168.10.101 <input checked="" type="checkbox"/> |
| Proxy / Registrar / Use registration 2:  |   | <input type="checkbox"/>                           |
| Proxy / Registrar / Use registration 3:  |   | <input type="checkbox"/>                           |
| Proxy / Registrar / Use registration 4:  |   | <input type="checkbox"/>                           |
| Proxy / Registrar / Use registration 5:  |   | <input type="checkbox"/>                           |
| Home server test:                        | invite                                    | ▼  |
| Changeover:                              | changeover on failure of INVITE request ▼ |  |
| Changeover by timeout:                   | <input checked="" type="checkbox"/>       |  |
| Keepalive time (s):                      | 60  |  |
| Full RURI compliance:                    | <input type="checkbox"/>                  |  |
| SIP-Domain:                              | 192.168.10.101                            |  |
| Use domain to Register:                  | <input checked="" type="checkbox"/>       |  |
| Registration Retry Interval (s):         | 30  |  |
| Inbound:                                 | <input type="checkbox"/>                  |  |
| Outbound:                                | with busytone ▼                           |  |
| Dial timeout:                            | 10  |  |
| Expires:                                 | 1800                                      |  |
| Authentication:                          | user defined ▼                            |  |
| Username:                                | TAU-72.IP                                 |  |
| Password:                                | *****                                     |  |
| Alert-Info:                              | <input type="checkbox"/>                  |  |
| Ringback at answer 183:                  | <input type="checkbox"/>                  |  |
| Ringback at callwaiting:                 | 180 Ringing ▼                             |  |
| Remote ringback:                         | don't send ringback in RTP ▼              |  |
| DTMF MIME Type:                          | application/dtmf-relay ▼                  |  |
| Hook flash MIME Type:                    | application/hook-flash ▼                  |  |
| Escape hash uri:                         | <input type="checkbox"/>                  |  |
| User=Phone:                              | <input type="checkbox"/>                  |  |
| Remove inactive media:                   | <input type="checkbox"/>                  |  |
| P-RTP-Stat:                              | <input type="checkbox"/>                  |  |
| CT with replaces:                        | <input checked="" type="checkbox"/>       |  |
| 100rel:                                  | off ▼                                     |  |
| Enable timer:                            | <input type="checkbox"/>                  |  |
| Min SE:                                  | 120                                       |  |
| Session expires (0 - unlimited session): | 0   |  |
| NAT settings:                            |   |  |
| NAT Keep Alive Msg:                      | off ▼                                     |  |
| NAT Keep Alive Interval (s):             | 30  |  |
| Conference settings:                     |   |  |
| Conference mode:                         | Local ▼                                   |  |
| Conference server:                       | conf                                      |  |
| IMS settings:                            |   |  |
| Enable IMS:                              | off ▼                                     |  |
| XCAP name for three-party conference:    | three-party-conference                    |  |
| XCAP name for hotline:                   | hot-line-service                          |  |
| XCAP name for call waiting:              | call-waiting                              |  |
| XCAP name for call hold:                 | call-hold                                 |  |
| XCAP name for explicit call transfer:    | explicit-call-transfer                    |  |

Undo all changes Re-registration Defaults Submit changes

Save

SIP configuration:

- *Proxy mode*—select SIP server (SIP-proxy) operation mode form the drop-down list:
  - Off—*disabled*.
  - Parking—*SIP-proxy redundancy mode without main SIP-proxy management*.
  - Homing—*SIP-proxy redundancy mode with main SIP-proxy management*.

Gateway may operate with a single main SIP-proxy and up to four redundant SIP-proxies. For exclusive operations with the main SIP-proxy, 'Parking' and 'Homing' modes are identical. In this case, if the main SIP-proxy fails, it will take time to restore its operational status.

For operations with redundant SIP-proxies, 'Parking' and 'Homing' modes will work as follows: the gateway sends INVITE message to the main SIP-proxy address when performing outgoing call, and REGISTER message when performing registration attempt. If on expiration of 'Invite total timeout' there is no response from the main SIP-proxy or response 408 (when 'changeover by timeout' option is enabled), 503, or 505 is received, the gateway sends INVITE (or REGISTER) message to the first redundant SIP-proxy address, and if it is not available, the request is forwarded to the next redundant SIP-proxy and so forth. When available redundant SIP-proxy is found, registration will be renewed on that SIP-proxy.

Next, the following actions will be available depending on the selected redundancy mode:

- 1 In the 'parking' mode, the main SIP-proxy management is absent, and the gateway will continue operation with the redundant SIP-proxy even when the main proxy operation is restored. If the connection to the current SIP-proxy is lost, querying of the subsequent SIP-proxies will be continued using the algorithm described above. If the last redundant SIP-proxy is not available, the querying will continue in a cycle, beginning from the main SIP-proxy:
  - 2 In the 'homing' mode, three types of the main SIP-proxy management are available: periodic transmission of OPTIONS messages to its address, periodic transmission of REGISTER messages to its address, or transmission of INVITE request when performing outgoing call. First of all, INVITE request is sent to the main SIP-proxy, and if it is unavailable, then to the next redundant one, etc. Regardless of the management type, when the main SIP-proxy operation is restored, gateway will renew its registration and begin operation with the main SIP-proxy.
- *Proxy/ Registrar address 1..5*—SIP-proxy/registration server network address; you may define the port after the colon; if it is not specified, 5060 will be taken as the default port value.
  - *Use registration 1..5*—when checked, register on server, otherwise registration server will not be used.
  - *Home server test*—depending on the selected configuration, test the main proxy using OPTIONS, REGISTER, or INVITE messages in 'homing' redundancy mode.
  - *Change-over*—this setting defines the request transmission error that will be used for redundant proxy changeover: INVITE and REGISTER, INVITE only or REGISTER only.
  - *Changeover by timeout*—when enabled, redundant proxy changeover will be performed when response 408 is received, in addition to standard responses 503 and 505.

- *Keepalive time (s)*—period of time between OPTIONS or REGISTER management message transfers, in seconds.
- *Full RURI compliance*—when checked, all URI elements (*user, host and port*—subscriber number, IP address and UDP/TCP port) will be analyzed upon receiving an incoming call. If all URI elements match, the call will be assigned to the subscriber port. When unchecked, only subscriber number (*user*) will be analyzed, and if the number matches, the call will be assigned to the subscriber port.
- *SIP Domain*—SIP domain. Used when you need to pass *from* and *to* fields in the '*host*' parameter of SIP URI scheme.
- *Use domain to Register*—use a domain in Request URI. In this case, domain will be sent in 'REGISTER', 'INVITE', 'SUBSCRIBE', 'NOTIFY', 'OPTIONS' Request URI. Does not apply in 'OPTIONS' requests, used for the main SIP server management (Home server test).
- *Registration Retry Interval (s)*—retry interval for SIP server registration attempts, when the previous attempt was unsuccessful (e.g., if response '*403 forbidden*' was received from the server).
- *Inbound*—when checked, receive all incoming calls from SIP-proxy, otherwise receive incoming calls from all hosts. When enabled, the routing to the proxy address will be created for all calls originated by addresses that differ from SIP-proxy (response '*305 Use proxy*' will be used with the address of the required server).
- *Outbound*—defines the mode for outgoing calls via SIP-proxy:
  - *off*—outgoing calls routed is performed according to the routing plan.
  - *on*—SIP-proxy will be used for outgoing calls in all cases.
  - *with busy tone*—SIP-proxy will be used for outgoing calls in all cases. If subscriber port is not registered for some reason, busy tone will be played on this port, when the phone is offhook.



**In addition to static Outbound SIP server configuration, you may define dynamic configuration with DHCP Option 120. When this option is received, the gateway will use it in the first SIP profile (Profile 1) only; at that, '*Proxy/Registrar address*' settings will remain in effect and will still be used as SIP-proxy and registration server addresses. If you want to use addresses specified in Option 120 as SIP-proxy and registration server addresses, leave '*Proxy/Registrar address*' settings blank. As this option allows to send addresses of a multiple outbound SIP servers, *Proxy redundancy modes* described above will also work in this case.**

- *Dial timeout (for Outbound)*—dialling timeout for the next digit (in 'Outbound' mode), in seconds. To dial without a timeout, you should use prefixes with the definite quantity of digits or use '*Stop dial at #*' setting separately for subscriber ports.



**This setting is effective for 'Dialplan table' routing plan only.**

- *Expires*—registration renewal time period.
- *Authentication*—defines device authentication mode:

- Global—enable SIP server authentication with common user name and password for all subscribers.
  - User defined—enable SIP server authentication with different user names and passwords for each subscriber, user name and password for ports could be defined in 'PBX/Ports' settings.
  - Username—username for 'global' mode authentication.
  - Password—password for 'global' mode authentication ('password', by default).
  - Alert-Info—process INVITE request 'Alert-Info' header to send a non-standard ringing to the subscriber port. Cadence for a non-standard ringing may be configured in 'Alert-Info' tab of the corresponding SIP profile.
  - Ringback at answer 183—when checked, 'ringback' tone will be sent upon receiving '183 Progress' message. When this setting is used, the gateway will not generate a ringback tone to the local subscriber, if the voice frequency path is already forwarded at the time when the message 183 is received, or if message 183 contains SDP session description for the frequency path forwarding.
  - Ringback at callwaiting—send 180 or 182 message, when the second call is received on the port with an active Call waiting service. Used to notify the caller (with a ringback tone of specific tonality) that their call is queued and waiting for response. Depending on the received message (180 Ringing or 182 Queued), the caller gateway generates either a standard ringback (180 Ringing) or a non-standard one (182 Queued).
  - Remote ringback—parameter defines, whether the gateway should send a ringback tone upon receiving an incoming call:
    - Don't send ringback in RTP—when an incoming call is received, the gateway will not generate a ringback tone.
    - Ringback with 180 ringing—when an incoming call is received, the gateway will generate a ringback tone and send it to the communicating gateway in the voice frequency path. Voice frequency path forwarding will be performed along with '180 ringing' message transmission via SIP protocol.
    - Ringback with 183 progress—when an incoming call is received, the gateway will generate a ringback tone and send it to the communicating gateway in the voice frequency path. Voice frequency path forwarding will be performed along with '183progress' message transmission via SIP protocol.
  - DTMF MIME Type—MIME extension type used for DTMF transmission in SIP protocol INFO messages:
    - Application/ dtmf—DTMF is sent in application/dtmf extension ('\*' and '#' are sent as digits 10 and 11).
    - Application/ dtmf-relay—DTMF is sent in application/dtmf-relay extension ('\*' and '#' are sent as symbols '\*' and '#').
    - Audio/telephone-event—DTMF is sent in audio/telephone-event extension ('\*' and '#' are sent as digits 10 and 11).
- DTMF transmission performed during the established session allows for extension dialling.**
- Hook Flash MIME Type—MIME extension type used for Flash transmission in SIP protocol INFO messages:



- *As DTMF*—send in MIME extension configured in DTMF 'MIME Type' parameter. If *application/dtmf-relay* is used, then the flash will be sent as 'signal=hf'; if *application/dtmf* or *audio/telephone-event* is used, then the flash will be sent as the digit '16'.
- *Application/Hook Flash*—flash is sent in Application/ Hook Flash extension (as 'signal=hf').
- *Application/Broadsoft*—flash is sent in Application/ Broadsoft extension (as 'event flashhook').
- *Application/sscc*—flash is sent in Application/ sscc extension (as event flashhook).  
Used when you have to send the flash impulse to the opposite device without update of session parameters.



**For detailed information on operations with flash in application/broadsoft and application/sscc used for supplementary services, see Appendix K.**

- *Escape hash uri*—when checked, send hash symbol (#) in SIP URI as escape sequence '%23', otherwise— as '#' symbol. When option *user=phone* is checked, hash symbol is always sent as '#' symbol regardless of '*Escape hash uri*' setting.
- *User=Phone*—when checked, use 'User=Phone' tag in SIP URI, otherwise it will not be used. Tag usage is described in the beginning of this section.
- *Remove inactive media*—when checked, remove inactive media streams during SDP session modification. Enables interaction with gateways that incorrectly handle rfc3264 recommendation (according to recommendation, the number of streams should not decrease during session modifications.)
- *P-RTP-Stat*—use 'P-RTP-Stat' header in BYE request or in its reply to transfer RTP statistics.
- *CT with replaces*—when checked, use '*replaces*' tag while performing '*Call Transfer*' service, otherwise it will not be used. When the checkbox is selected, the gateway performing the service generates '*refer-to*' header, which—in addition to the address of a subscriber the call being transferred to—adds '*replaces*' tag that contains DIALOG ID (Call-ID, to-tag, from-tag) of a replaced call. It is recommended to use '*replaces*' tag in operations with SIP server, as this option mostly does not require the establishment of a new dialogue between SIP server and the subscriber that the call is being forwarded to.
- *100rel*—use reliable provisional responses (RFC3262):
  - *supported*—reliable provisional responses are supported.
  - *required*—reliable provisional responses are mandatory.
  - *off*—reliable provisional responses are disabled.
- *Enable timer*—when checked, enables support of SIP session timers (RFC 4028). During the voice session, UPDATE requests (if the opposite gateway supports them) or re-INVITE requests should be sent for connection management purposes.
- *Min SE*—minimal time interval for connection health checks (90 to 1800s, 120s by default.)
- *Session expires*—period of time in seconds that should pass before the forced session termination if the session is not renewed in time (90 to 80000s, recommended value—1800s, 0—unlimited session).

#### NAT settings:

- *NAT Keep Alive Msg*—selection of an active session support mode for operations through NAT.

- *off*—disabled.
- *options*—use OPTIONS request as an active session support message.
- *notify*—use NOTIFY notification as an active session support message.
- *CRLF*—use CRLF special request as an active session support message.
- *NAT Keep Alive Interval (s)*—active session support message transmission period. Permitted values—30 to 120 seconds.

Conference settings:

- *Conference mode*—conference assembly mode selection.
  - *Local*—conference assembly is performed locally at the gateway. Voice packets are mixed at the gateway.
  - *Remote (REFER to Focus)*—conference assembly is performed at the conference server. Voice packets are mixed at the server. In this mode, gateway sends to server the information on gateways which should be added to the conference. Next, conference server will add these gateways to the conference.
  - *Remote (REFER to User)*—conference assembly is performed at the conference server. Voice packets are mixed at the server. In this mode, gateway sends to subscribers the identifier of a conference, that they should connect to at the conference server. Next, gateways will add themselves to the conference.



**For conference operation algorithms in various modes, see Section: 7.3 3-way Conference**

- *Conference server*—conference server name in Remote mode operation.

IMS settings:

- *Enable IMS*—enable service (simulation service) management using IMS (3GPP TS 24.623).

Gateway supports:

- *Implicit subscription to IMS services*—in this subscription option, gateway will not send SUBSCRIBE requests after subscriber registration, and will only process NOTIFY requests received from IMS, which are used for service management.
- *Explicit subscription to IMS services*—in this subscription option, gateway will send SUBSCRIBE requests after subscriber registration, and upon successful subscription, will process NOTIFY requests received from IMS, which are used for service management.



**When 'Enable IMS' setting is enabled, 'Process flash', 'Call waiting' and 'Hot line' parameters will not be processed in subscriber port settings, as these services are managed by IMS server.**

- *XCAP name for three-party conference*—a name sent in XCAP attachment for '3-party conference' service management.
- *XCAP name for hotline*—a name sent in XCAP attachment for 'Hotline' service management.
- *XCAP name for call waiting*—a name sent in XCAP attachment for 'Call waiting' service management.

- *XCAP name for call hold*—a name sent in XCAP attachment for 'Call hold' service management.
- *XCAP name for explicit call transfer*—a name sent in XCAP attachment for 'Explicit call transfer' service management.

For forced registration renewal of subscriber ports with the current SIP profile, click *Re-registration* button.

Use '*Defaults*' button to set default parameters (the figure below shows default values).

To apply changes, click '*Submit Changes*' button; to discard all changes, click '*Undo All Changes*' button; to save changes, click '*Save*' button.

#### 5.1.2.2.4.1. *Provisional response setting operation*

SIP protocol defines two types of responses for connection initiating request (INVITE)—provisional and final. 2xx, 3xx, 4xx, 5xx and 6xx-class responses are final and their transfer is reliable, with ACK message confirmation. 1xx-class responses, except for '100 Trying' response, are provisional, without confirmation (rfc3261). These responses contain information on the current INVITE request processing step, therefore loss of these responses is unacceptable. Utilization of reliable provisional responses is also stated in SIP (rfc3262) protocol and defined by '100rel' tag presence in the initiating request. In this case, provisional responses are confirmed with PRACK message.

Setting operation for outgoing communications:

- supported—send the following tag in 'INVITE' request—supported: 100rel. In this case, communicating gateway may transfer provisional responses reliably or unreliably—as it deems fit.
- required—send the following tags in 'INVITE' request—supported: 100rel and required: 100rel. In this case, communicating gateway should perform reliable transfer of provisional replies. If communicating gateway does not support reliable provisional responses, it should reject the request with message 420 and provide the following tag—unsupported: 100rel. In this case, the second INVITE request will be sent without the following tag—required: 100rel.
- off—do not send any of the following tags in INVITE request—supported: 100rel and required: 100rel. In this case, communicating gateway will perform unreliable transfer of provisional replies.

Setting operation for incoming communications:

- supported, required—when the following tag is received in 'INVITE' request—supported: 100rel, or required: 100rel—perform reliable transfer of provisional replies. If there is no supported: 100rel tag in INVITE request, the gateway will perform unreliable transfer of provisional replies.

- off—when the following tag is received in 'INVITE' request—required: 100rel, reject the request with message 420 and provide the following tag—unsupported: 100rel. Otherwise, perform unreliable transfer of provisional replies.

#### 5.1.2.2.4.2. *Configuration of Internal Switching for SIP-proxy Connection Loss*

In order to perform intra-office calls when connection to SIP-proxy is lost, you should specify TAU-32M.IP gateway IP address as the last SIP-proxy. At that, '*Proxy mode*' must be set to '*homing*', otherwise, when the connection to the main SIP-proxy is restored, it will not be used afterwards.

#### 5.1.2.2.4.3. *SIP domain configuration via local DNS*

In the current firmware version, it is possible to configure SIP domain using a local DNS. This option may become useful, for example, when you use redundant SIP-proxies in different domains.

*SIP domain configuration order for 'n' profile:*

1. To use a local DNS, leave DNS field in '*Network/Network settings*' tab blank or enter the value 127.0.0.1.
2. In '*Network/Hosts*' tab, enter the mapping of a host (SIP domain) to actual IP addresses of SIP proxy/SIP registrar.
3. In '*PBX/SIP-H323 Profiles/Profile n/SIP Custom*' tab, specify domains for each pair of SIP proxy and SIP registrar.
4. Enable routing via SIP proxy by selecting *outbound* checkbox in '*PBX/SIP-H323 Profiles/Profile n/SIP Custom*' tab, or entering prefixes in '*PBX/SIP-H323 Profiles/Profile n/Dialplan (Dialplan table)*' tab. If you configure prefixes, select SIP proxy protocol in '*Protocol&Target*' field.

### 5.1.2.2.5. Codecs Configuration—Profile N Codecs

In 'Profile n/Codecs' submenu, you may configure codecs used in the current profile.

TAU-32M.IP signal processor encodes analogue voice traffic and fax/modem data into digital signal and performs its reverse decoding. Gateway supports the following codecs: G.711A, G.711U, G.729, G723.1, G.726-32.

G.711 is PCM codec that does not employ a compression of voice data. This codec must be supported by all VoIP equipment manufacturers. G.711A and G.711U codecs differ from each other in encoding law (A-law is a linear encoding and U-law is non-linear). The U-law encoding is used in North America, and the A-law encoding—in Europe.

G.723.1 is a voice data compression codec, allows for two operation modes: 6.3kbps and 5.3kbps. G.723.1 codec has a voice activity detector and performs comfort noise generation at the remote end during period of silence (Annex A).

G.726-32 is a voice data compression codec that uses ADPCM compression algorithm at the rate of 32kbps.

G.729 is also a voice data compression codec with the rate of 8kbps. As with G.723.1, G.729 codec supports voice activity detector and performs comfort noise generation (Annex B).

T.38 is a standard for sending facsimile messages in real time over IP networks. Signals and data sent by the fax unit are copied to T.38 protocol packets. Generated packets may feature redundancy data from previous packets that allows to perform reliable fax transmissions through unstable channels.



**You don't have to reboot the gateway in order to apply codec settings. When applying settings, all current calls will be terminated!!!**

**TAU-32M.IP WEB configurator**

Network settings | **PBX** | Switch | Monitoring | System info | Service | Log out

Main | **SIP/H323 Profiles** | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups | Distinctive Ring | Modifiers

SIP Common | H323 | **Profile 1** | Profile 2 | Profile 3 | Profile 4 | Profile 5 | Profile 6 | Profile 7 | Profile 8

SIP Custom | **Codecs** | Dialplan | Alert-Info

**Attention! Changing of these parameters will lead to aborting of all calls!**

| Codecs configuration:              |                                     |
|------------------------------------|-------------------------------------|
| List of codecs in preferred order: |                                     |
| G.711A                             | <input checked="" type="checkbox"/> |
| G.711U                             | <input checked="" type="checkbox"/> |
| G.726-32                           | <input type="checkbox"/>            |
| G.723                              | <input type="checkbox"/>            |
| G.729A                             | <input type="checkbox"/>            |
| G.729B                             | <input type="checkbox"/>            |

↕

| Packet coder time:                        |                                     |
|---|-------------------------------------|
| G.711 Ptime:                              | 20 ms                               |
| G.729 Ptime:                              | 20 ms                               |
| G.723 Ptime:                              | 30 ms                               |
| G.726-32 Ptime:                           | 20 ms                               |
| Features:                                 |                                     |
| G.726-32 PT:                              | 102                                 |
| DTMF Transfer:                            | rfc2833                             |
| Flash Transfer:                           | INFO                                |
| Fax Detect Direction:                     | Caller and Callee                   |
| Fax Transfer Codec:                       | G.711U                              |
| Slave Fax Transfer Codec:                 | Off                                 |
| Modem Transfer:                           | G.711A VBD                          |
| rfc2833 PT:                               | 101                                 |
| Decoding rfc2833 with PT from answer SDP: | <input type="checkbox"/>            |
| Silence suppression:                      | <input type="checkbox"/>            |
| Echo canceller:                           | <input checked="" type="checkbox"/> |
| NLP disable:                              | <input type="checkbox"/>            |
| Comfort noise:                            | <input type="checkbox"/>            |
| RTCP Configuration:                       |                                     |
| RTCP timer:                               | <input type="checkbox"/>            |
| RTCP control period:                      | <input type="checkbox"/>            |
| RTCP-XR:                                  | <input type="checkbox"/>            |
| Cisco NSE Configuration:                  |                                     |
| NSE PT:                                   | 100                                 |
| T.38 Configuration:                       |                                     |
| Max datagram size:                        | 512                                 |
| Bitrate:                                  | 14400                               |
| Jitter buffer Configuration:              |                                     |
| Modem/Fax pass-thru:                      |                                     |
| Delay:                                    | 0 ms                                |
| Voice:                                    |                                     |
| Mode:                                     | Adaptive                            |
| Delay min:                                | 0 ms                                |
| Delay max:                                | 200 ms                              |
| Deletion threshold:                       | 500 ms                              |
| Deletion mode:                            | Soft                                |

In **'Codecs configuration'** section, you may select codecs and an order of their usage on connection establishment. Codec with the highest priority should be placed in top position. Click the left mouse button to highlight the row with the selected codec. Use arrow buttons (up, down) to change the codec priority.



**G.723.1 codec is used together with 'Silence compression' setting. When the setting is enabled, Annex A support is enabled, otherwise it is disabled.**

- Use G.711A—use G.711A codec
- Use G.711U—use G.711U codec
- Use G.723—use G.723.1 codec

- Use *G.729A*—use G.729 annexA codec (when defining codec compatibility, non-standard codec description is sent via SIP: a=rtpmap:18 G729A/8000 a=fmtp:18 annexb=no).
- Use *G.729B*—use G.729 annexB codec.
- Use *G.726-32*—use G.726-32 codec.



**G.726-32 codec used only in SIP protocol operations.**

In '**Packet coder time**' section, you should define packetization time, i.e. amount of voice data in milliseconds (ms), transmitted in a single RTP protocol voice packet:

- *G711 Ptime*—for G711 codec (permitted values: 10, 20, 30, 40, 50, 60).
- *G729 Ptime*—for G729 codec (permitted values: 10, 20, 30, 40, 50, 60, 70, 80).
- *G723 Ptime*—for G723 codec (permitted values: 30, 60, 90).
- *G.726-32 Ptime*—for G.726-32 codec (permitted values: 10, 20, 30).
- *G.726-32 PT*—G.726-32 codec payload type (permitted values: 96 to 127).

In '**Features**' section:

- *DTMF Transfer*—DTMF tone transmission method During established session, DTMF transmission is used for extension dialling.
  - *Inband*—inband, in RTP voice packets.
  - *RFC2833*—according to RFC2833 recommendation, as a dedicated payload in RTP voice packets.
  - *INFO*—outbound. For SIP protocol, INFO messages are used; the type of transmitted DTMF tones depends on MIME extension type (for detailed description, see Section 5.1.2.2.4). When H.323 protocol is used, DTMF transmission method depends on 'DTMF Transfer' parameter in H.323 tab (see Section 5.1.2.2.3).



**In order to be able to use extension dialling during the call, make sure that the similar DTMF tone transmission method is configured on the opposite gateway.**

- *Flash Transfer*—short clearback Flash transmission method. Flash transmission by the subscriber's port via IP network is possible only when Flash function operation mode 'Transmit flash' is configured on this port (see Section 5.1.2.4).
  - *Disabled*—Flash transmission is disabled.
  - *RFC2833*—Flash transmission is performed according to RFC2833 recommendation, as a dedicated payload in RTP voice packets.
  - *INFO*—Flash transmission is performed with SIP/H323 protocol methods. For SIP protocol, INFO messages are used; the type of transmitted Flash tones depends on MIME extension type (for detailed description, see Section 5.1.2.2.4).

When H.323 protocol is used, Flash transmission method depends on 'DTMF Transfer'

parameter in H.323 tab (see Section 5.1.2.2.3).

- *Fax Detect Direction*—defines the call direction for fax tone detection and subsequent switching to fax codec:
  - *no detect fax*—disables fax tone detection, but will not affect fax transmission (switching to fax codec will not be initiated, but such operation still may be performed by the opposite gateway).
  - *Caller and Callee*—tones are detected during both fax transmission and receiving. During fax transmission, CNG FAX signal is detected from the subscriber's line. During fax receiving, V.21 signal is detected from the subscriber's line.
  - *Caller*—tones are detected only during fax transmission. During fax transmission, CNG FAX signal is detected from the subscriber's line.
  - *Callee*—tones are detected only during fax receiving. During fax receiving, V.21 signal is detected from the subscriber's line.
  
- *Fax Transfer Codec*—master protocol/codec used for fax transmissions:
  - *fax transfer G.711A*—use G.711A codec for fax transmissions. Switching to G.711A codec will be performed when the corresponding tones are detected.
  - *fax transfer G.711U*—use G.711U codec for fax transmissions. Switching to G.711U codec will be performed when the corresponding tones are detected.
  - *T.38 mode*—use T.38 protocol for fax transmissions. Switching to T.38 will be performed when the corresponding tones are detected.
  
- *Slave Fax Transfer Codec*—slave protocol/codec used for fax transmissions: This codec is used when the opposite device does not support the priority:
  - *fax transfer G.711A*—use G.711A codec for fax transmissions. Switching to G.711A codec will be performed when the corresponding tones are detected.
  - *fax transfer G.711U*—use G.711U codec for fax transmissions. Switching to G.711U codec will be performed when the corresponding tones are detected.
  - *T.38 mode*—use T.38 protocol for fax transmissions. Switching to T.38 will be performed when the corresponding tones are detected.
  - *Off*—disable slave protocol/codec.



**Master and slave protocols/codecs should differ from each other.**

- *Modem Transfer*—defines switching into 'Voice band data' mode (according to V.152 recommendation). In VBD mode, the gateway disables the voice activity detector (VAD) and comfort noise generator (CNG), this is necessary for establishing a modem connection.



- Off—disable modem signal detection.
- G.711A VBD—use G.711A codec to transfer data via modem connection. Switching to G.711A codec in VBD mode will be performed when the CED tone is detected.
- G.711U VBD—use G.711U codec to transfer data via modem connection. Switching to G.711U codec in VBD mode will be performed when the CED tone is detected.
- G.711A RFC3108—use G.711A codec to transfer data via modem connection. When entering modem data transfer mode via SIP protocol, echo cancellation and VAD are disabled with attributes described in RFC3108 recommendation:
  - a=silenceSupp:off - - - -
  - a=ecan:fb off -;
- G.711U RFC3108—use G.711U codec to transfer data via modem connection. When entering modem data transfer mode via SIP protocol, echo cancellation and VAD are disabled with attributes described in RFC3108 recommendation:
  - a=silenceSupp:off - - - -
  - a=ecan:fb off -;
- G.711A NSE—CISCO NSE support, G.711A codec is used to transfer data via modem connection.
- G.711U NSE—CISCO NSE support, G.711U codec is used to transfer data via modem connection.

**Cisco NSE support: when NSE 192 packet is received, gateway will switch to the selected codec and disable VAD; when NSE 193 packet is received, echo canceller will be disabled.**

- *RFC2833 PT*—type of payload used to transfer packets via RFC2833. Permitted values: 96 to 127. RFC2833 recommendation describes the transmission of DTMF and Flash tones via RTP protocol. This parameter should conform to the similar parameter of a communicating gateway.
- *Decoding rfc2833 with PT from answer SDP*—when performing outgoing call, receive DTMF tones in rfc2833 format with payload type proposed by a communicating gateway. When unchecked, tones will be received with the payload type, configured on the gateway. Enables compatibility with gateways that incorrectly handle rfc3264 recommendation.
- *Silence suppression*—when checked, use voice activity detector (VAD) and silence suppression (SSup), otherwise they will not be used. Voice activity detector disables transmission of RTP packets during periods of silence, reducing loads in data networks.
- *Echo canceller*—when checked, use echo cancellation (tail length is up to 128ms).
- *NLP disable*—when checked, use echo cancellation with disabled non-linear processor (NLP). When signal levels on transmission and reception significantly differ, useful signal may become suppressed by the NLP. Use this echo canceller operation mode to prevent the signal suppression.
- *Comfort noise*—when checked, use comfort noise generator. Used together with 'Silence compression (VAD)' setting, as comfort noise packets are generated only upon voice pauses detection.

In '**RTCP configuration**' section, you may configure basic settings for device operation via RTCP protocol:

- *RTCP timer*—time period in seconds (5-65535), after which the device send control packets via RTCP protocol. When unchecked, RTCP will not be used.
- *RTCP control period*—control function of a voice frequency path status. Defines the period of time (RTCP timer), during which the opposite side will wait for RTCP protocol packets. When there is no packets in the specified period of time, established connection will be terminated due to loss of connection—cause 3 no route to destination. Control period value is calculated using the following equation: RTCP timer\* RTCP control period, seconds. When unchecked, control feature will be disabled.
- *RTCP-XR*—when checked, generate 'RTCP Extended Reports' control packets according to RFC 3611.

In '**Cisco NSE configuration**' section, you may configure codec payload type for modem transmission using CISCO NSE method:

- *NSE PT*—type of payload used to transfer packets via NSE. Permitted values: 96 to 127.

In '**T38 configuration**' section, you may configure T.38 protocol parameters:

- *Max Datagram Size*—maximum datagram size. (Zero value means that T38MaxDatagram attribute will not be transferred via SIP, and the gateway will support the reception of datagrams up to 512bytes. Use zero value in interactions with gateways that do not support datagrams from 272bytes and higher.) This parameter defines the maximum quantity of bytes that will be sent in T.38 protocol packet.
- *Bitrate*—maximum fax transfer rate (9600, 14400). This setting affects the ability of a gateway to work with high-speed fax units. If fax units support data transfer at 14400 baud, and the gateway is configured to 9600 baud, the maximum speed of connection between fax units and the gateway will be limited at 9600 baud. And vice versa, if fax units support data transfer at 9600 baud, and the gateway is configured to 14400 baud, this setting will not affect the interaction, maximum speed will be defined by the performance of fax units.

In '**Jitter buffer configuration**' section, you may configure jitter buffer parameters.

Due to various factors, e.g. network overload, voice data packets may be served to the gateway at different speeds, and their arrival order may change. Such event is called 'jitter'.

In order to compensate the jitter effect, the jitter buffer has been implemented. In jitter buffer, packets are saved as soon as they are received. Voice packets that came out of sequence (earlier or later) have their sequential number analyzed. After that, they are positioned into their respective places in a queue and sent further in the right order that allows to improve call quality for unstable communication channels.

Jitter buffer may be fixed or adaptive. The size of adaptive jitter buffer changes along with the average identified delay in voice packets' reception. When delay rises, the size of adaptive jitter buffer grows instantaneously, when delay lowers, buffer size shrinks in 10 seconds after the delay has been steadily reduced.

In '**Modem/Fax pass-thru**' section, you may configure the jitter buffer in fax/modem data transfer mode.

- *Delay*—the size of a fixed jitter buffer, used in fax or modem data transfer mode. Permitted value range is from 0 to 200ms.
  - '**Voice**'—*jitter buffer voice connection settings*.
- *Mode*—jitter buffer operation mode: fixed or adaptive.
- *Delay*—size of fixed jitter buffer or lower limit (minimum size) of adaptive jitter buffer. Permitted value range is from 0 to 200ms.
- *Delay max*—upper limit (maximum size) of adaptive jitter buffer, in milliseconds. Permitted value range is from 'Delay' to 200ms.
- *Deletion threshold*—threshold for immediate deletion of a packet, in milliseconds. When buffer size grows and packet delay exceeds this threshold, packets will be deleted immediately. Permitted value range is from 'Delay max' to 500ms.
- *Deletion mode*—buffer adjustment mode. Defines the method of packet deletion during buffer adjustment to lower limit. In 'SOFT' mode, device uses intelligent selection pattern for deletion of packets that exceed the threshold. In 'HARD' mode, packets which delay exceeds the threshold will be deleted immediately.

To discard all changes made to configuration, click '*Undo All Changes*' button. To set default parameters, click '*Defaults*' button (the figure below shows default values). To apply changes, click '*Submit Changes*' button.

To store changes to non-volatile memory of the device, click '*Save*' button.

#### 5.1.2.2.6. *Routing and Pickup Code Configuration (Dialplan)*

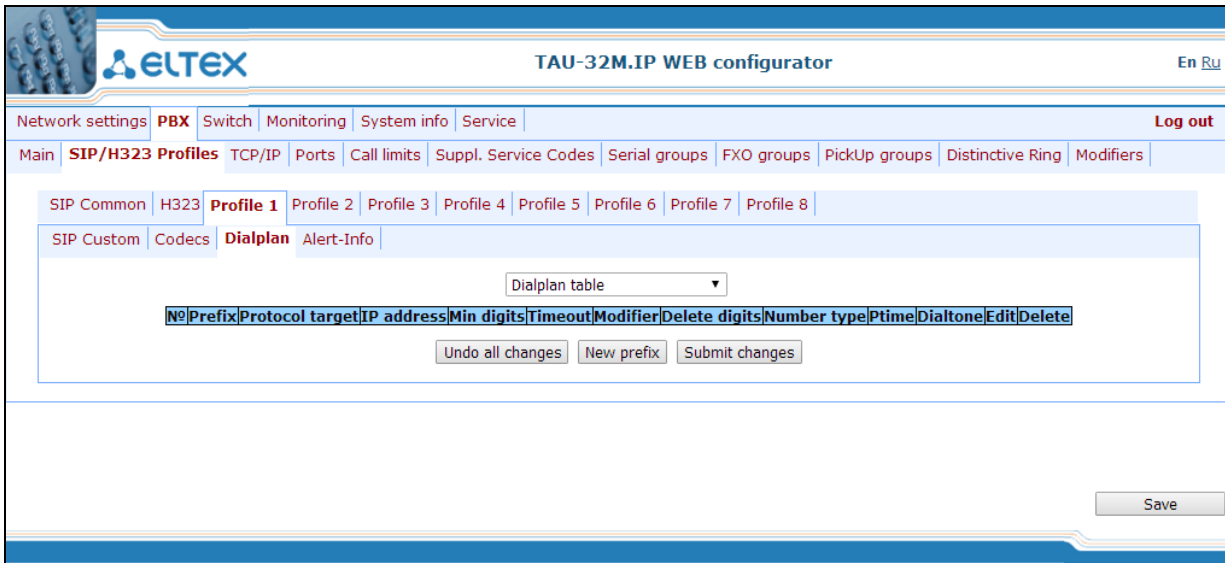
In '*Profile n/Dialplan*' submenu, you may configure prefixes for routing and pickup groups for each profile.

TAU-32M.IP gateway routing is built on prefixes. Prefix is the first part of the callee number, and when it is combined with the quantity of digits of a dialled number and the dialling timeout, it comprises the routing rule. If a number dialled by the subscriber falls within the scope of a single rule, the call will be routed by this rule. If a dialled number falls within the scope of multiple rules, the call will be routed by the rule with the highest priority. When dialled number does not match any rules, busy tone will be played to the subscriber.

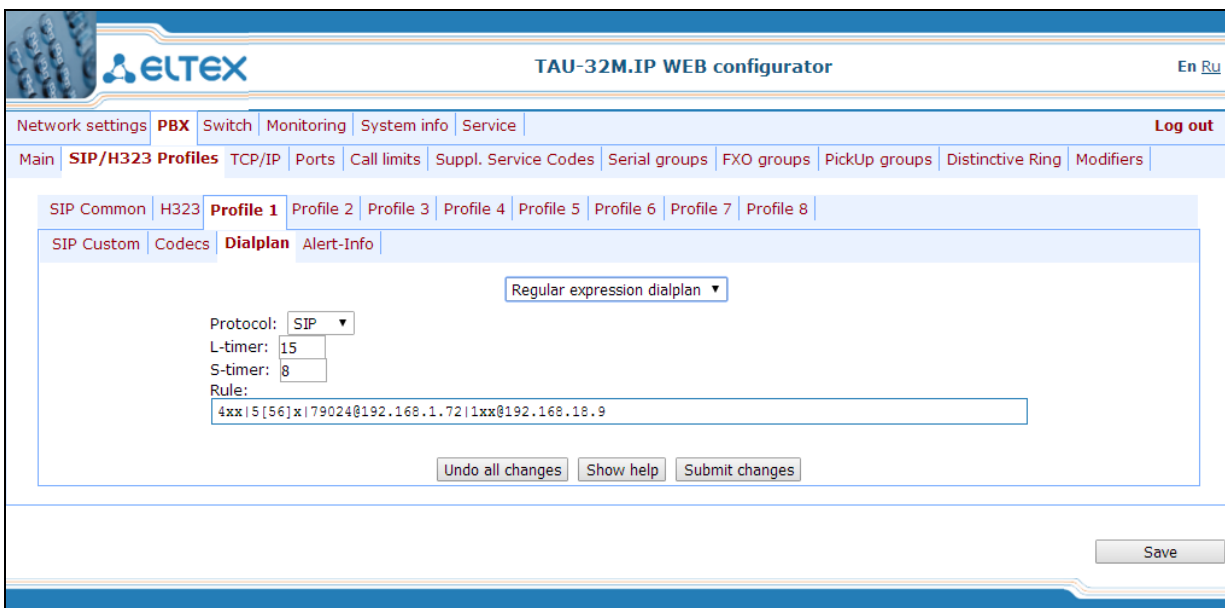
When SIP-proxy operates in *outbound* mode, all calls are routed via SIP-proxy; configuration of prefixes is optional in this case. In the absence of prefixes, the quantity of digits in the dialled number is not limited, and the end of dialling occurs on the expiration of 'outbound' timer, or on '#' button pressed. If you have to use *outbound* mode without the wait for the end of dialling on 'outbound' timer, you will have to configure prefixes.

*Pickup group*—subscriber group, authorized to receive (or intercept) any calls directed at another subscriber of the group.

*Dialplan Table*—table of routing prefixes' settings; for parameter description, see Section 5.1.2.2.6.1.



*Regular Expression Dialplan*—configuration of routing prefixes using regular expressions; for description of regular expression format, see Section 5.1.2.2.6.4.



### 5.1.2.2.6.1. Configuration of Routing Rules


Hover the mouse cursor over a row and left-click it to highlight with orange and make it active (available for moving). Use arrow buttons (up, down) to change the prefix sequence order. The higher the prefix row in configuration, the higher its priority.

To add a new prefix, click 'New prefix' button:

| New dialplan entry  |                          |
|---|--------------------------|
| Prefix:   |                          |
| Min digits:   | 0                        |
| Timeout:  | 0                        |
| Protocol & Target:  | SIP Proxy                |
| Address:  |                          |
| Modifier:   |                          |
| Number of digits to delete:   | 0                        |
| Number type:  | Unknown                  |
| Ptime:  | <input type="checkbox"/> |
| Dialtone:   | <input type="checkbox"/> |
| <input type="button" value="Cancel"/> <input type="button" value="Submit changes"/> |                          |

| Ingress  |                          |                          |                          |                          |                          |                          |                          |                          |
|--|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|
| Port   | 1                        | 2                        | 3                        | 4                        | 5                        | 6                        | 7                        | 8                        |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| Port   | 9                        | 10                       | 11                       | 12                       | 13                       | 14                       | 15                       | 16                       |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| Port   | 17                       | 18                       | 19                       | 20                       | 21                       | 22                       | 23                       | 24                       |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| Port   | 25                       | 26                       | 27                       | 28                       | 29                       | 30                       | 31                       | 32                       |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="button" value="Enable all"/> <input type="button" value="Disable all"/> |                          |                          |                          |                          |                          |                          |                          |                          |

- *Prefix*.
- *Min digits*—minimum length of a number dialled by the prefix.
- *Timeout*—dialling timeout for the next digit of a number, in seconds. Begins operation, when the minimum length of a number dialled by the prefix is achieved. If the minimum length of a dialled number is already achieved, and no digits have been dialled during this timeout, the call is routed by the prefix. In order to route the call immediately on dialling the minimum length of a number, specify 0 as a dialling timeout for the next digit of a number.
- *Protocol&Target*—signalling protocol, used in prefix operations:
  - *H.323 Gatekeeper*—H.323 protocol operation through the gatekeeper (possible for profile 1 only).
  - *H.323 Direct IP*—H.323 point-to-point protocol operation (possible for profile 1 only).
  - *SIP Proxy*—SIP protocol operation via SIP-proxy.
  - *SIP Direct IP*—SIP point-to-point protocol operation.
  - *SIP-T Direct IP*—SIP-T point-to-point protocol operation.
  - *PickUp Group*—pickup group.
- *Address*—IP address of a communicating gateway in point-to-point operation mode (specified when H.323 Direct IP /SIP Direct IP is used).
- *Modifier*—dialling modifier, enables translation of a callee number. Modifier is added at the beginning of a dialled number.
- *Number of digits to delete*—dialling modifier, enables translation of a callee number. Defines the number of digits to be deleted from a dialled number for outgoing calls (the most significant digits of a number will be removed).
 



**When outgoing call is performed using a prefix, the digit deletion modifier ('Number of digits to delete') is applied first to the dialled number, followed by the digit addition modifier ('Modifier').**
- *Number type*—callee number type. Used only in SIP-T and H.323 protocol operations. Transferred in CdPN parameter.
- *Ptime*—when checked, defines the packetization time for the current direction, in seconds.
- *Dial tone*—send 'PBX response' tone when the first prefix digit is dialled. Usually, used with a prefix beginning with '8' to send the 'PBX response' tone for a long-distance direction. If there are multiple prefixes beginning with the same digit, but having different configurations of this setting, then a prefix with the highest priority will be responsible for determining whether the 'PBX response' tone will be sent or not.

To apply changes, click 'Submit Changes' button; to discard all changes, click 'Cancel'.

To edit parameters of existing prefix, you may directly modify data in fields, or call the edit menu by clicking button in the respective row. To delete a prefix, click button.

To discard all changes made to configuration, click 'Undo All Changes' button. To apply changes, click 'Submit Changes' button. To store changes to non-volatile memory of the device, click 'Save' button.

#### 5.1.2.2.6.2. Configuration of Prefix with Varying Number Count

Enables dialling by a single prefix with various quantity of digits using *Dialplan Table*.

Prefix should be configured as follows:

- 1 In 'Min digits' field, enter a minimum quantity of digits for routing with this prefix.
- 2 In 'Timeout' field, dialling timeout for the next digit of a number should be greater than zero. In this case, when user dials the number with length that matches the minimum quantity of digits, gateway will wait for the next digit dialling during the specified timeout. If the digit is not dialled, prefix call will be performed with the minimum quantity of digits; if the digit is dialled, the timer will restart, and the gateway will wait again for the next digit dialling.
- 3 If dialling timeout for the next digit is zero, the call will be routed immediately when the length of a number equal to minimum quantity of digits is achieved.
- 4 'Stop dial at #' function allows to perform a call after the necessary quantity of digits are dialled without the wait for a timeout. It may be configured separately for each port in 'PBX/Ports/Edit/Custom' tab. If this function is enabled for the port, user upon dialling a necessary number, the port may press # button on the phone unit (provided that the unit is configured for DTMF dialling mode), and after that the call will be routed immediately.

#### 5.1.2.2.6.3. Configuration of pickup codes

| New dialplan entry  |                |
|---|----------------|
| Prefix:   |                |
| Min digits:   | 0              |
| Timeout:  | 0              |
| Protocol & Target:  | Pickup Group ▼ |
| Address:  |                |
| Modifier:   |                |
| Number of digits to delete:   | 0              |
| Number type:  | Unknown ▼      |
| Ptime:  |                |
| Dialtone:   |                |
| <input type="button" value="Cancel"/> <input type="button" value="Submit changes"/> |                |

| Pickup Group   |                          |                          |                          |                          |                          |                          |                          |                          |
|--|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|
| #  | 1                        | 2                        | 3                        | 4                        | 5                        | 6                        | 7                        | 8                        |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| #  | 9                        | 10                       | 11                       | 12                       | 13                       | 14                       | 15                       | 16                       |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| #  | 17                       | 18                       | 19                       | 20                       | 21                       | 22                       | 23                       | 24                       |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| #  | 25                       | 26                       | 27                       | 28                       | 29                       | 30                       | 31                       | 32                       |
| Enable   | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="button" value="Enable all"/> <input type="button" value="Disable all"/> |                          |                          |                          |                          |                          |                          |                          |                          |

Configuration of pickup groups affects the following settings:

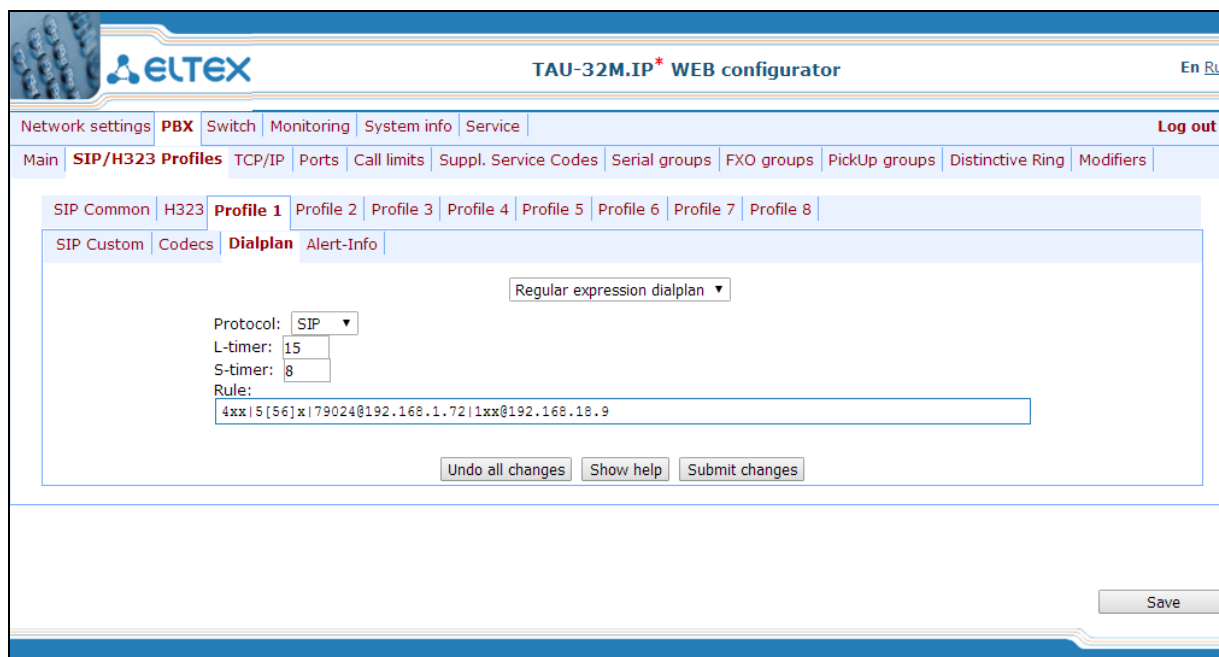
- Prefix—pickup code. Sequence of digits (for example, \*8) that, when dialled, allows any subscriber of the group to pickup the call received by another subscriber of the group.
- Protocol&Target—it's necessary to select a pickup group—PickUp.
- PickUp Group—defines the list of groups, that will use this code for the call pickup. Thus, a single code

may be used for call pickups in different groups.

To enable this pickup code for all groups, click 'Enable all' button. To disable this pickup code for all groups, click 'Disable all' button.

#### 5.1.2.2.6.4. Configuration of Regular Expression Routing Rules

This section describes the configuration of regular expression routing rules.



To open the configuration page for regular expression routing rules, select 'Regular Expression Dialplan' from the 'Dialplan' drop-down list.

- Protocol—VoIP protocol name: H.323, SIP (H.323 may be used in profile 1 only).
- L-timer—activates, when the gateway detects the necessity of dialling of at least one more digit in order to achieve the compliance with any of the dialplan rules.
- S-timer—activates, when the dialling complies with one of the rules, but there is a possibility that further dialling will achieve compliance with another rule.
- Rule—field for routing rules written with regular expressions (up to 1000 characters). The structure and format of regular expressions that enable different dialling features are listed below.

#### Regular expression routing plan record rule ('Rule'):

**Rule1| Rule2|..| RuleN**

**Rule= L{value} S{value} prefix@optional**

where

L – L-timer (optional parameter)

S – S-timer(optional parameter)

Timers inside rules could be dropped; in this case, global timer values, defined before the parentheses, will be

used.

*prefix*—prefix part of the rule

*@optional*—optional part of the rule (may be skipped)

## Regular expressions' syntax

*Prefix part of the rule*

- |—logical **OR**—used to separate rules.
- **X** or **x**—any number from 0 to 9, equal to a range [0-9].
- **0 - 9**—numbers from 0 to 9.
- **"A", "B", "C", "D"**—'A', 'B', 'C', 'D' characters.
- **\*—\*** character.
- **#—#** character.
- **[ ]**—define ranges (with a hyphen), or enumeration (w/o spaces, commas, and other characters between the digits), e.g.

Range: **[1-5]**—1,2,3,4, or 5.

Enumeration: **[138]**—1,3, or 8.

Range and enumeration **[0-9\*#]**—0 to 9, and also \* and #.

- **{min,max}**—define the repetition count for a character located outside the parentheses, a range or \*# symbols.

*min*—minimum repetition count, *max*—maximum repetition count.

**{,max}**—equal to {0,max}.

**{min,}**—equal to {min,inf}.

Example:

**5{2,5}**—'5' could be dialled up to 5 times.

Equal to the following record: 55|555|5555|55555

- **.** — 'dot' special symbol means that a preceding digit, range, or '\*', '#' characters may be repeated from one to infinity times. Equivalent to a record {0,}.

Example:

**5x.\*** —'x' in this rule may be completely absent or may be present any number of times.

Equivalent to a record 5\*|5x\*|5xx\*|5xxx\*|...

- **+**—digit, range, or '\*', '#' characters preceding the '+' symbol may be repeated from one to infinity times. Equivalent to a record {1,}.



- **<:>**—modification of a number. Digits and '\*', '#' characters preceding the colon will be replaced with those after the colon. Modification allows to remove (<xx:>), add (<:xx>), or replace (<xx:xx>) digits and symbols.
- **!** —dial block. Specified at the end of a rule and means that the dialling of numbers corresponding to the template will be blocked.
- **,**—send 'PBX response' tone. For long-distance access (for city access in case of office PBX), it is common to hear a ringback, that may be implemented by inserting comma in a sequence of digits.

**8,x.** —after dialling '8' subscriber will hear 'PBX response' tone.

- **'S', 'T'**—short (S) or long (T) timers are used in rules containing special repetition characters '{min,max}', '.', or '+' and are specified right after them. They define, which timer will work for the current rule when it is already possible to perform the the routing for the dialled number. If the timer is not specified, S-timer will be used by default. Allows to replace S-timer with L-timer in the current profile.

*Optional part of the rule (may be skipped)*

- **host:port**—routing to IP address. Usage of a port is effective for SIP protocol only. If @host:port is not specified, calls will be routed via SIP-proxy or H.323 gatekeeper.

Example:

**1xxxx@192.168.16.13:5062**—all five-digit dials, beginning with 1, will be routed to IP address 192.168.16.13 to port 5062

- **{pickup:x,xx}**—pickup group code dialling. You may specify multiple pickup groups using comma.

Example:

**\*8@{pickup:1}**—'\*8' code is used for the first pickup group

- **{local}**—routing inside the gateway to a local IP address. Must be used for internal routing, when the device receives its network settings dynamically (via DHCP protocol).

*Timers*

- **S-timer**—activates, when the dialling complies with one of the rules, but it is possible that further dialling will achieve compliance with another rule.

- **L-timer**—activates, when the gateway detects the necessity of dialling of at least one more digit in order to achieve the compliance with any of the dialplan rules.

Timer values may be specified for a complete routing plan, as well as for the specific rule. Timer values may be specified for all templates in a routing plan; in this case values are listed before the opening parenthesis.

If these values are listed in one sequence only, they are effective only for this sequence.

### Example of the dialplan record

```
L20 8,x.|520001@192.168.16.150:5061|52xxx[02-9]|1xxxx|<53:70>xxxx@192.168.16.13|
26x{,5}|*8@{pickup:1,6,32}|3[0-3]x+|34*{1,3}|35#x{0,}|36x.*|37[0-2]x+T
```

#### 5.1.2.2.7. Alert-Info distinctive ring

In 'Alert-Info' submenu, you may configure a distinctive ring, generated by the value from Alert-Info header received in INVITE request. 16 various Alert-Info values may be processed for each profile.

The screenshot shows the 'Alert-Info' configuration page in the ELTEX TAU-32M.IP WEB configurator. The page has a navigation menu at the top with options like 'Network settings', 'PBX', 'Switch', 'Monitoring', 'System info', 'Service', and 'Log out'. Below the navigation menu, there are tabs for 'SIP/H323 Profiles', 'TCP/IP', 'Ports', 'Call limits', 'Suppl. Service Codes', 'Serial groups', 'FXO groups', 'PickUp groups', 'Distinctive Ring', and 'Modifiers'. The 'Alert-Info' tab is selected, and it shows a table with 16 rows for configuring Alert-Info strings and Distinctive Ring rules. The table has two columns: 'Alert-Info string' and 'Distinctive Ring rule'. The first five rows are pre-filled with values, and the remaining 11 rows are empty. Below the table, there is a green text box with instructions: 'An Alert-Info string must contain the filename only but no full path to it. A Distinctive Ring rule must contain no more than 6 "pulse-pause" pairs separated by commas or semicolons. The duration of each pulse and pause must belong to the range [200; 16000] milliseconds and must be the multiplier of 100.' At the bottom of the page, there are buttons for 'Undo all changes', 'Defaults', 'Submit changes', and 'Save'.

|    | Alert-Info string | Distinctive Ring rule |
|----|-------------------|-----------------------|
| 1  | Bellcore-dr1      | 1000,4000             |
| 2  | Bellcore-dr2      | 1000,3000             |
| 3  | Bellcore-dr3      | 1000,2000             |
| 4  | Bellcore-dr4      | 1000,1000             |
| 5  | Bellcore-dr5      | 700,700,700,3000      |
| 6  |                   |                       |
| 7  |                   |                       |
| 8  |                   |                       |
| 9  |                   |                       |
| 10 |                   |                       |
| 11 |                   |                       |
| 12 |                   |                       |
| 13 |                   |                       |
| 14 |                   |                       |
| 15 |                   |                       |
| 16 |                   |                       |

- *Alert-Info string*—signal name sent in Alert-Info header.

Alert-Info header appears as follows: <<http://ipaddr/signal>>, where:

- *ipaddr*—IP address of a device, that the signal should be played from (not processed at TAU).

- *signal*—signal name that should be used for generation of non-standard ringing.
- *Distinctive Ring rule*—non-standard ringing generation rule. Ringing tone is cyclic.

The rule includes up to 6 pairs of impulse/pause values; all values are comma-separated. Each value must be divisible by 100 and fall within the range from 200 to 16000ms.

For example, a record '700,700,700,3000' means that 700ms impulse will be sent first, followed by 700ms pause, then again 700ms impulse, 3s pause; after that, this sequence will be repeated.

### 5.1.2.3. Configuration of network ports (TCP/IP)

In *TCP/IP* submenu, you may configure network port range for various protocols.



**You don't have to reboot the gateway in order to apply TCP/IP settings. When applying settings, all current calls will be terminated.**

| TCP/IP configuration:        |                          |
|------------------------------|--------------------------|
| TCP port range (H.245/H.225) |                          |
| TCP port min:                | 10000                    |
| TCP port max:                | 11920                    |
| UDP port range (RAS)         |                          |
| UDP port min:                | 12000                    |
| UDP port max:                | 13920                    |
| RTP port range (RTP)         |                          |
| RTP H323 min:                | 30000                    |
| RTP H323 max:                | 35000                    |
| RTP SIP min:                 | 35002                    |
| RTP SIP max:                 | 40000                    |
| Intercept port range         |                          |
| Intercept port min:          | 50000                    |
| Intercept port max:          | 50100                    |
| TOS configuration            |                          |
| Diffserv for SIP:            | 104                      |
| Diffserv for RTP:            | 184                      |
| Other                        |                          |
| Verify remote media address: | <input type="checkbox"/> |

- *TCP port range (H.245/H.225)*—range of network ports used for H.323 - H.245/H.225 stack protocols' operation:
  - *TCP port min*—the lower limit of a TCP port range.
  - *TCP port max*—the upper limit of a TCP port range.
- *UDP port range (RAS)*—range of network ports used for H.323 stack RAS protocol operation (RAS protocol is used during gatekeeper interactions):

- *UDP port min*—the lower limit of a UDP port range.
- *UDP port max*—the upper limit of a UDP port range.
- *RTP port range (RTP)*—range of network ports used for voice data protocol (RTP) operation:
  - *RTP H323 min*—the lower limit of a range of RTP ports used for H.323 protocol operation.
  - *RTP H323 max*—the upper limit of a range of RTP ports used for H.323 protocol operation.
  - *RTP SIP min*—the lower limit of a range of RTP ports used for SIP protocol operation.
  - *RTP SIP max*—the upper limit of a range of RTP ports used for SIP protocol operation.
- *Intercept port range*—range of network ports used for pickup traffic transmission (SORM):
  - *Intercept port min*—the lower limit of a range of ports used for pickup traffic transmission (SORM feature).
  - *Intercept port max*—the upper limit of a range of ports used for pickup traffic transmission (SORM feature).



**SORM feature implementation is based on rfc3924 recommendation—Cisco Architecture for Lawful Intercept in IP Networks. To perform the pickup, the following MIBs are used: CISCO-IP-TAP-MIB.my and CISCO-TAP2-MIB.my.**

*Diffserv configuration—configuration of Diffserv:*

- *Diffserv for SIP*—type of service for SIP packets. You may configure all 8-bit Diffserv fields (DSCP bits—6 high bits) sent in IP protocol header; parameter value should be specified decimally. For utilized values, see Table 7.
- *Diffserv for RTP*—type of service for RTP packets. You may configure all 8-bit Diffserv fields (DSCP bits—6 high bits) sent in IP protocol header; parameter value should be specified decimally. For utilized values, see Table 7.

*Other:*

- *Verify remote media address*—when checked, apply control to the media traffic received, otherwise it will not be controlled. This function controls the received media traffic (voice traffic, T38 fax) for established connection. If this traffic comes in from the host or port not specified in SIP/H.323 signalling exchange, it will be rejected.



**To avoid the conflicts, ports used by H.225/H.245/RAS signalling and RTP should not overlap the ports used by SIP signalling (5060 by default, and also ports configured in 'ports' and 'serial groups' tabs.)**

Table 8—'Type of service for RTP packets' (Diffserv) field value:

| Diffserv field value | Description                               |
|----------------------|---|
| 0 (0x00)             | (DSCP 0x00) – Best effort – default value |
| 32 (0x20)            | (DSCP 0x08) – Class 1                     |

|                       |  |
|-----------------------|--|
| 40 (0x28)             | (DSCP 0x0A)– assured forwarding, low drop precedence (Class1, AF11)    |
| 48 (0x30)             | (DSCP 0x0C)– assured forwarding, medium drop precedence (Class1, AF12) |
| 56 (0x38)             | (DSCP 0x0E)– assured forwarding, high drop precedence (Class1, AF13)   |
| 64 (0x40)             | (DSCP 0x10) – Class 2  |
| 72 (0x48)             | (DSCP 0x12)– assured forwarding, low drop precedence (Class2, AF21)    |
| 80 (0x50)             | (DSCP 0x14)– assured forwarding, medium drop precedence (Class2, AF22) |
| 88 (0x58)             | (DSCP 0x16)– assured forwarding, high drop precedence (Class2, AF23)   |
| 96 (0x60)             | (DSCP 0x18) – Class 3  |
| 104 (0x68)            | (DSCP 0x1A)– assured forwarding, low drop precedence (Class3, AF31)    |
| 112 (0x70)            | (DSCP 0x1C)– assured forwarding, medium drop precedence (Class3, AF32) |
| 120 (0x78)            | (DSCP 0x1E)– assured forwarding, high drop precedence (Class3, AF33)   |
| 128 (0x80)            | (DSCP 0x20) – Class 4  |
| 136 (0x88)            | (DSCP 0x22)– assured forwarding, low drop precedence (Class4, AF41)    |
| 144 (0x90)            | (DSCP 0x24)– assured forwarding, medium drop precedence (Class4, AF42) |
| 152 (0x98)            | (DSCP 0x26)– assured forwarding, high drop precedence (Class4, AF43)   |
| 160 (0xA0)            | (DSCP 0x28) – Class 5  |
| 184 (0xB8)            | (DSCP 0x2E) – expedited forwarding (Class5, Expedited Forwarding)      |
| <b>IP Precedence:</b> |  |
| 0 (0x00)              | IPP0 (Routine)   |
| 32 (0x20)             | IPP1 (Priority)  |
| 64 (0x40)             | IPP2 (Immediate)   |
| 96 (0x60)             | IPP3 (Flash)   |
| 128 (0x80)            | IPP4 (Flash Override)  |
| 160 (0xA0)            | IPP5 (Critical)  |
| 192 (0xC0)            | IPP6 (Internetwork Control)  |
| 224 (0xE0)            | IPP7 (Network Control)   |

To discard all changes made to configuration, click '*Undo All Changes*' button. To set default parameters, click '*Defaults*' button (the figure below shows default values). To apply changes, click '*Submit Changes*' button.

### 5.1.2.4. Configuration of Subscriber Ports (Ports)

In 'Ports' menu, you may configure subscriber ports of the device.



You may use up to 8 subscriber profiles to configure the following port settings: *CallerID mode, Flash impulse duration, signal levels strengthening/weakening, priority between CFB and CW services, 'Music on hold' service, payphone mode.* In 'Subscriber profile' item of the 'Custom' tab, you may assign one of the configured subscriber profiles to each port. Profile 1 is assigned for all ports by default. To open the subscriber profile configuration window, click 'Subscriber profiles' in 'PBX/Ports' tab. If you have to configure a custom value for any of the parameters listed above, you have to configure it in 'PBX/Ports' menu by clicking 'Edit -Common' button. To use custom settings, it is absolutely necessary to select 'Custom' checkbox (in 'PBX/Ports' tab – 'Edit -Custom' or 'PBX/Ports') in the port configuration.



You don't have to reboot the gateway in order to apply port settings. Changing 'SIP port' parameter will lead to termination of current calls. Changing other parameters will not disrupt any of the established connections.

TAU-32M.IP\* WEB configurator

Network settings | **PBX** | Switch | Monitoring | System info | Service | Log out

Main | SIP/H323 Profiles | TCP/IP | **Ports** | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups | Distinctive Ring | Modifiers


**Attention! Changing of these parameters will lead to aborting of all calls!**

1-8 | 9-16 | 17-24 | 25-32 | Subscriber profiles

| Port | Type | Phone | User name | Custom settings                     | Category | Process flash          | Subscriber profile | SIP/H323 profile | Disabled                 | Edit |
|------|------|-------|-----------|-------------------------------------|----------|------------------------|--------------------|------------------|--------------------------|------|
| 1    | FXS  | 501   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |
| 2    | FXS  | 502   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |
| 3    | FXS  | 503   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |
| 4    | FXS  | 504   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |
| 5    | FXS  | 505   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |
| 6    | FXS  | 506   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |
| 7    | FXS  | 507   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |
| 8    | FXS  | 508   |           | <input checked="" type="checkbox"/> | off ▼    | Attended caltransfer ▼ | Profile 1 ▼        | Profile 1 ▼      | <input type="checkbox"/> |      |

## Configuration of ports

*Port settings table (tabs '1-8', '9-16', '17-24', '25-32'):*

- *Port*—port number.
- *Phone*—subscriber's number.
- *User name*—subscriber's name.
- *Custom*—when checked, use common settings for this port (configured by clicking 'Edit' button), otherwise use settings from the specified subscriber profile (configured in '*Subscriber profiles*' tab).
  - *Category*—select subscriber's category (cpc-rus), off—subscriber category will not be used. When this setting is enabled, the category will be sent in 'from' field, and 'tel uri' will be used instead of 'sip uri'.
  - *Process flash*—flash function operation mode (short clearback). For parameter description, see below.
  - *Subscriber profiles*—number of the subscriber profile, which parameters will be used for the current port (use '*PBX/Ports/Subscriber profiles*' tab to configure subscriber profile parameters.)
  - *SIP/H323 profile*—SIP/H323 profile number, that will be used for the current port.
  - *Disabled*—when checked, the port is disabled, otherwise it will be enabled. To disable the service for ports, select checkboxes against the desired ports and click '*Submit Changes*' button.
  - *Edit*  —the button which allows you to enter the port settings editing mode.
  - *Auto numeration*—automatic port enumeration.

## Settings of subscriber profiles

You may configure subscriber profiles in 'Subscriber profiles' tab:

The screenshot shows the 'Subscriber profiles' configuration page for Profile 1. The interface includes a navigation menu at the top with tabs for Network settings, PBX, Switch, Monitoring, System info, and Service. Below this is a sub-menu with tabs for Main, SIP/H323 Profiles, TCP/IP, Ports, Call limits, Suppl. Service Codes, Serial groups, FXO groups, Pickup groups, Distinctive Ring, and Modifiers. A red warning message states: "Attention! Changing of these parameters will lead to aborting of all calls!".

The 'Subscriber profiles' section is active, showing Profile 1 selected. The configuration parameters are as follows:

| Profile 1                                 |                                     |
|---|-------------------------------------|
| CallerID:                                 | off                                 |
| Hide date:                                | <input type="checkbox"/>            |
| Hide name:                                | <input type="checkbox"/>            |
| Min Flashtime (ms):                       | 200                                 |
| Max Flashtime (ms):                       | 600                                 |
| Gain receive (0.1 dB):                    | -70                                 |
| Gain transmit (0.1 dB):                   | 0                                   |
| SS7 category (SIP-T):                     | 10                                  |
| Category:                                 | off                                 |
| Modifier:                                 | off                                 |
| CFB has priority over CW:                 | <input type="checkbox"/>            |
| Play music on hold:                       | <input type="checkbox"/>            |
| Stop dial at #:                           | <input type="checkbox"/>            |
| Taxophone:                                | off                                 |
| CPC:                                      | <input type="checkbox"/>            |
| CPC time (ms):                            | 200                                 |
| FXO parameters                            |                                     |
| Outgoing direction parameters             |                                     |
| Flashtime:                                | 300                                 |
| Dialtone detection:                       | <input type="checkbox"/>            |
| Dialtone time detect (s):                 | 5                                   |
| Dialing delay (s):                        | 2                                   |
| Don't transmit prefix:                    | <input checked="" type="checkbox"/> |
| Transmit number:                          | <input checked="" type="checkbox"/> |
| 503 Service unavailable on busy (SIP):    | <input checked="" type="checkbox"/> |
| PSTN activity:                            | Off                                 |
| Dialing                                   |                                     |
| Method:                                   | DTMF                                |
| Interdigit delay:                         | 200                                 |
| Pulse time (ms):                          | 80                                  |
| Pause time (ms):                          | 80                                  |
| Incoming direction parameters             |                                     |
| Ring detection:                           | 2                                   |
| PSTN number prefix:                       |                                     |
| PSTN name prefix:                         |                                     |
| Use PSTN CallerID:                        | <input type="checkbox"/>            |
| Detect FXO line presence:                 | <input checked="" type="checkbox"/> |
| Block FXO line in outgoing direction:     | <input type="checkbox"/>            |
| Tone detect parameters                    |                                     |
| Minimum level of detectable signal (dBm): | -36                                 |
| Dialtone detection parameters:            | 425;0(2000/0/1)                     |
| Busytone detection parameters:            | 425;1(350/350/1)                    |
| Ringback tone detection parameters:       | 425;0(1000/4000/1)                  |
| Show tone detect parameters format        |                                     |

Buttons for 'Apply' and 'Defaults' are located at the bottom of the configuration area. A 'Save' button is visible in the bottom right corner of the interface.

- *CallerID*—select the Caller ID mode from the drop-down list. For Caller ID operation, subscriber's phone unit must support the selected method:
  - *Off*—Caller ID is disabled.
  - *Aon\_rus*—'Russian Caller ID' method. The number is served when subscriber's phone unit lifts the headset with its 500Hz frequency request.



- *Dtmf*—DTMF Caller ID method. The number is served between the first and second calls on the line by dual-frequency DTMF impulses.
- *Fsk\_bell202, Fsk\_v23*—FSK Caller ID method (using bell202 standard, or ITU-T V.23). The number is served between the first and second calls on the line by a stream of data with a frequency modulation.



**To enable Caller ID information reception, connected phone unit should support the configured Caller ID method.**



**In *Fsk\_bell202, Fsk\_v23* modes, Caller ID information is sent in MDMF format: time/date, subscriber's number and name.**

- *Hide date*—when checked, in *Fsk\_bell202, Fsk\_v23* modes, Caller ID information will be sent without time and date.
- *Hide name*—when checked, in *Fsk\_bell202, Fsk\_v23* modes, Caller ID information will be sent without subscriber's name.
- *Min Flashtime(ms)*—the lower limit of Flash impulse duration (ms).
- *Max Flashtime(ms)*—the upper limit of Flash impulse duration (ms).

For correct operation of Flash button on the subscriber's phone unit, its configured duration of flash dialling should fall within the following range: (Min Flashtime – Max Flashtime). Please note, that small values (70-20ms) of the lower limit may lead to situations, when dialling of digits in pulse phone unit operation mode will be interpreted as flash dialling. When the upper limit value is less than flash dialling duration configured for the subscriber's phone unit, pressing flash button will cause the clearback.



**If there is no effect (no 'PBX response' tone, indicating that the Hold service is performed) or the subscriber clearback occurs when you press the 'Flash' button, it means that configured 'Flash' settings for this port do not match the 'Flash' impulse generated by the phone unit, or 'Flash' is not processed by the gateway (Attendant CT, unattendant CT). If the 'Flash – Transmit flash' impulse transmission mode has been configured, the absence of the effect may also mean that the opposite gateway is not processing 'Flash' received from the IP network.**

- *Gain receive (0.1 dB)*—volume of voice reception (gain of the signal received from the communicating gateway and output to the speaker of the phone unit connected to TAU-32M.IP gateway.)



**For FXO ports, set the *Gain receive (0.1 dB)* value equal to 0.**

- *Gain transmit (0.1 dB)*—volume of voice transmission (gain of the signal received from the microphone of the phone unit connected to TAU-32M.IP gateway and transmitted to the communicating gateway.)

- *SS7 category (SIP-T)*—SS-7 category, sent in the SIP-T encapsulated message of SS-7 protocol. Corresponding Caller ID categories are listed in the table below.

| Caller ID category | SS-7 category |
|--------------------|---------------|
| 1                  | 10            |
| 2                  | 225           |
| 3                  | 228           |
| 4                  | 11            |
| 5                  | 226           |
| 6                  | 15            |
| 7                  | 227           |
| 8                  | 12            |
| 9                  | 229           |
| 10                 | 224           |

- *Category*—select subscriber category (cpc-rus):
  - *off*—subscriber category will not be used. When this setting is enabled, the category will be sent in 'from' field, and 'tel uri' will be used instead of 'sip uri'.
- *Modifier*—modifier table number, used for the current port.
- *CFB has priority over CW*—defines the priority between CFB (Forward on busy) and CW (Call wait) services. When checked, CFB service has a priority over CW, and vice versa.
- *Play music on hold*—use 'Play music on hold' service. When 'Hold' service is performed by this port, audio file stored in the gateway memory will be played to the opposite subscriber. When unchecked or the audio file is unavailable, 'hold' audio signal will be played to the opposite subscriber. To upload the audio file, use 'Service -> MOH' menu.
- *Stop dial at #*—when checked, use '#' button on the phone unit to end the dialling, otherwise '#' will be recognized as a DTMF symbol. When '#' is used to end the dialling, the call will be performed without the dialling timeout for the next digit.
- *Taxophone*—port operates in payphone mode:
  - *off*—port operates in normal mode.
  - *polarity*—payphone operation mode with polarity reversal. Perform line power polarity reversal on subscriber's response, and return it to original state on clearback.
  - *12kHz<sup>1</sup>*—payphone mode without polarity reversal. Generates 12 kHz meter pulse.
  - *16kHz<sup>1</sup>*—payphone mode without polarity reversal. Generates 16 kHz meter pulse.
- *CPC*—when checked, perform a short-time break of the subscriber loop on clearback from the opposite subscriber's side.

---

<sup>1</sup> Used only for TAU-32M.IP rev. B

- *CPC time(ms)*—duration of a short-time break of the subscriber loop.

FXO parameters:

Outgoing direction parameters:

- *Flashtime*—loop closure time for 'flash' impulse simulation. Default value is 300ms.
- *Dialtone detection*—when checked, detect the 'PBX response' tone before dialling in call direction from IP to FXO, otherwise detection will be disabled.
- *Dialtone time detect (s)*—duration of the 'PBX response' tone detection before dialling in call direction from IP to FXO.
- *Dialling delay, sec*—dialling delay, when the 'PBX response' tone detection is not used.
- *Don't transmit prefix*—when checked, transmit the complete number received from IP (from Request URI header of INVITE request) into the line, except for FXO unit subscriber number, otherwise it will not be transmitted.
- *Transmit number*—when checked, transmit the complete number received from IP (from Request URI header of INVITE request) into the line, including FXO unit subscriber number, otherwise it will not be transmitted.



**'Don't transmit prefix' and 'Transmit number' functions are used in SIP protocol operations only.**

- *503 Service unavailable on busy (SIP)*—if checked, reply 503 will be sent via SIP protocol when the subscriber's line (FXO) is busy, otherwise—486.
- *PSTN activity*—response to an event occurred in the subscriber line.
  - *Off*—disable responses to events.
  - *PSTN reversal polarity detection<sup>1</sup>*—when polarity reversal is detected, perform an action defined in 'PSTN reversal polarity detection' selector.
  - *PSTN answer detection*—detection of response on ringback tone.
- *PSTN reversal polarity detection*—action for polarity reversal detection:
  - *Release*—detection of polarity reversal is interpreted as a clearback signal (BYE request is transferred via SIP protocol).
  - *Answer*—detection of polarity reversal is interpreted as an answer signal (200 OK response is transferred via SIP protocol).

Dialling:

- *(Method)*—dialling type for the line:
  - *DTMF*—tone.
  - *pulse*—pulse.

- *Interdigit delay*—time interval between digits when dialling for the analogue phone line. Default value is 200ms.
- *Pulse time (ms)*—tone duration, in milliseconds.
- *Pause time (ms)*—pause duration, in milliseconds.

Incoming direction parameters:

- *Ring detection*—quantity of rings, that will be used by FXO for loop closure ('offhook') and transmitting the 'PBX response' tone into the line. Default value is 2.
- *PSTN number prefix*—prefix added to the number in CallerID sent to VoIP.
- *PSTN name prefix*—PBX prefix added to the name in CallerID sent to VoIP.
- *Use PSTN CallerID*—when checked, use CallerID received from the phone line for VoIP direction call, otherwise it will not be used.
- *Detect FXO line presence*—detection of subscriber's line connection to FXO for the line status view in monitoring.



**When the ringing arrives to FXO set, this setting may result in transmission of false data into monitoring on loss of subscriber's line connection.**

- *Block FXO line in outgoing direction*—block FXO set, if the subscriber's line is not connected to it.

Tone detect parameters:

- *Minimum level of detectable signal, dBm*—minimum level of detectable 'PBX response' and 'busy' tones, permitted values—from -20 to -40dBm.
- *Dialtone detection parameters*—'PBX response' tone detection parameters.
- *Busytone detection parameters*—'busy' tone detection parameters.
- *Ringback tone detection parameters*—ringback tone detection parameters.

Format of signal detection parameters:

*X;Z(A/B/1)*

*X,Y;Z(A/B/1)*

*X,Y;Z(A/B/1+2)*

*X,Y;Z(A/B/2)*

- X—first frequency component of a signal (Hz).
- X—second frequency component of a signal (Hz).
- Z—number of rings, required for signal detection. (Permitted values: 0 to 3; for 'PBX response' signal, zero value is used. Use zero value for ringback tone, if you have to detect the signal an unlimited number of times; used in order to avoid establishment of voice connection in IP networks prior to the answer of a subscriber.
- A—signal duration, ms.

- B—pause duration, ms (for continuous signals, value is 0).
- 1—apply time parameters A and B only to the 1st frequency component.
- 1+2—apply time parameters A and B to both frequency components.
- 2—apply time parameters A and B only to the 2nd frequency component.

Example: *480,620;1(500/300/1+2)*—dual-frequency signal with 480 and 620Hz frequency components will be detected after the first ring with signal and pause duration of 500 and 300ms respectively.

To apply settings, click '*Apply*' button. To exit the submenu, click '*Cancel*' button. To reset settings to default values, click '*Default*' button.

### Automatic enumeration

Click '*Auto numeration*' button in '*Ports conf.*' window to show the following menu:

| Auto numeration |                                     |  |  |
|-----------------|-------------------------------------|--|--|
| Prefix:         | <input type="text" value="a"/>      |  |  |
| First number:   | <input type="text" value="100000"/> |  |  |
| Postfix:        | <input type="text" value="b"/>      |  |  |

|         |          |         |          |         |          |         |          |
|---------|----------|---------|----------|---------|----------|---------|----------|
| Port 1  | a100000b | Port 2  | a100001b | Port 3  | a100002b | Port 4  | a100003b |
| Port 5  | a100004b | Port 6  | a100005b | Port 7  | a100006b | Port 8  | a100007b |
| Port 9  | a100008b | Port 10 | a100009b | Port 11 | a100010b | Port 12 | a100011b |
| Port 13 | a100012b | Port 14 | a100013b | Port 15 | a100014b | Port 16 | a100015b |
| Port 17 | a100016b | Port 18 | a100017b | Port 19 | a100018b | Port 20 | a100019b |
| Port 21 | a100020b | Port 22 | a100021b | Port 23 | a100022b | Port 24 | a100023b |
| Port 25 | a100024b | Port 26 | a100025b | Port 27 | a100026b | Port 28 | a100027b |
| Port 29 | a100028b | Port 30 | a100029b | Port 31 | a100030b | Port 32 | a100031b |

In the opened window, you may perform enumeration using a mask. In the '*First number*' field, enter *XXXX* number for the first port. All other ports will be enumerated by the following rule:

$XXXX + 1 \times N$ , where *N*—port number.

Prefix and postfix—constant parts, added in the beginning and in the end of a number.

To start enumeration, click '*Start*' button. To return to '*Ports*' menu, click '*Back*' button.


### Editing custom parameters of FXS type ports:

To edit parameters of a specific port, click  button in the corresponding row.

'*Custom*' tab—FXS type port custom settings:

| Port 1  |                                     |
|---|-------------------------------------|
| Phone:  | 501                                 |
| User name:  |                                     |
| Use alternative number:   | <input type="checkbox"/>            |
| Alternative number:   |                                     |
| Use alternative number as contact (only for serial groups members): | <input type="checkbox"/>            |
| Authentication name:  |                                     |
| Authentication password:  |                                     |
| Custom settings:  | <input checked="" type="checkbox"/> |
| Subscriber profile:   | Profile 1 ▼                         |
| SIP/H323 profile:   | Profile 1 ▼                         |
| Hot line:   | <input checked="" type="checkbox"/> |
| Hot timeout:  | 0                                   |
| Hot number:   | 400                                 |
| CLIR:   | <input type="checkbox"/>            |
| DND:  | <input type="checkbox"/>            |
| Disabled:   | <input type="checkbox"/>            |
| SIP port:   | 5060                                |
| Process flash:  | Attended calltransfer ▼             |
| Call waiting:   | <input type="checkbox"/>            |
| MWI:  | <input type="checkbox"/>            |

Apply Cancel Defaults

- *Phone*—subscriber's number.
- *User name*—subscriber's name.
- *Use alternative number*—when checked, use alternative number; otherwise it will not be used. May be used, when the gateway operates as a PABX, to assign a single subscriber's number to multiple phone lines.
- *'Alternative number'*—alternative subscriber's number. This number will be an alternative Caller ID of a subscriber and will be displayed on the subscriber's Caller ID display (transferred in the 'from' field URI in SIP protocol operations).
- *Use alternative number as contact (only for serial groups members)*—use an alternative number as a subscriber's contact (transferred in 'contact' header via SIP protocol). This setting is used only for ports located in the call group.
- *Authentication name*—username used for authentication. Used in SIP protocol operations, when in 'PBX/SIP-H323 Profiles/Profile n/SIP Custom' menu the independent authentication mode is selected (Authentication – user defined).
- *Authentication password*—password used for authentication. Used in SIP protocol operations, when in 'PBX/SIP-H323 Profiles/Profile n/SIP Custom' menu the independent authentication mode is selected (Authentication – user defined).
- *Custom settings*—when checked, use common settings for this port (configured by clicking 'Edit ' button), otherwise use settings from the specified subscriber profile (configured in 'Subscriber profiles' tab.) When checked, selection of the subscriber profile will be unavailable for this port.
- *Subscriber profiles*—number of the subscriber profile, which parameters will be used for the current port (use 'PBX/Ports/Subscriber profiles' tab to configure subscriber profile parameters).
- *SIP/H323 profile*—SIP/H323 profile number, that will be used for the current port.
- *Hot line*—when checked, 'Hotline/warmline' service is enabled. This service allows to establish an outgoing connection automatically without dialling the number right after the lifting of a headset – "hot line", or with a delay – "warm line". Direction of a service—from analogue phone line to

VoIP.



**This setting will not work, if 'IMS mode'—'Enable IMS' parameter in SIP profile settings—is enabled on the device.**

– *Hot timeout*—delay timeout in seconds for the start of the automatic dialling when the 'warmline' service is enabled.

– *Hot number*—number that will receive the call when 'Hotline/warmline' is enabled.

– *CLIR*—when checked, calling line identification restriction service—CLIR—is enabled.

– *DND*—when checked, 'do not disturb' service (temporary restriction for incoming calls) is enabled.

– *Disabled*—when checked, the port is disabled.

– *SIP port*—local UDP port used for port operations via SIP protocol.

– *Process flash*—flash function operation mode (short clearback). When 'flash' button is pressed on the subscriber's phone unit—if the duration of dialling falls within the range (Min Flashtime – Max Flashtime)—there are several gateway behaviours:

- *Transmit flash*—transmit flash into the channel using method described in 'Flash Transfer' item of the codec configuration (*Codecs conf.*) In this case, flash dialling will be processed by the communicating gateway.

- *Attended calltransfer*—'Call Transfer' service is enabled for the port with the wait for response of the subscriber, the call is being forwarded to. In this case, flash dialling will be processed locally by the gateway.

- *Unattended calltransfer*—'Call Transfer' service is enabled for the port without the wait for response of the subscriber, the call is being forwarded to. In this case, flash dialling will be processed locally by the gateway, and the call transfer will be performed when subscriber finished dialling a number.

- *No detect flash*—ignore (do not detect) short flash clearback, received from the subscriber.

- *Local CT*—transfer of the call to ports within the device is performed without REFER request transmission to the communicating gateway.



**This setting will not work, if 'IMS mode'—'Enable IMS' parameter in SIP profile settings—is enabled on the device.**



**For 'Calltransfer' service operation principles, see Section 7.1.**

– *Call waiting*—when checked, 'Call waiting service' will be enabled (service is available in flash function operation mode—call transfer).

– *MWI*—when checked, 'Message waiting indicator' service will be enabled. When the service is enabled, if the user has unread voice messages, intermittent 'PBX response' tone will be played when the phone is offhook; after that, the tone will become continuous. Voice message box operation depends on the Softswitch resources, TAU only plays the notification.



**This setting will not work, if 'IMS mode'—'Enable IMS' parameter in SIP profile settings—is enabled on the device.**

'Common' tab—FXS type port common settings:

| Port 1                    |                          |
|---------------------------|--------------------------|
| CallerID:                 | off                      |
| Hide date:                | <input type="checkbox"/> |
| Hide name:                | <input type="checkbox"/> |
| Min Flashtime (ms):       | 200                      |
| Max Flashtime (ms):       | 600                      |
| Gain receive (0.1 dB):    | 10                       |
| Gain transmit (0.1 dB):   | 60                       |
| SS7 category (SIP-T):     | 10                       |
| Category:                 | off                      |
| Modifier:                 | off                      |
| CFB has priority over CW: | <input type="checkbox"/> |
| Play music on hold:       | <input type="checkbox"/> |
| Stop dial at #:           | <input type="checkbox"/> |
| Taxophone:                | off                      |
| CPC:                      | <input type="checkbox"/> |
| CPC time (ms):            | 200                      |

Apply Cancel Defaults

Description of fields is equivalent to 'PBX/Ports/Subscriber profiles' tab fields shown above.



**Exclamation mark symbol means that the settings on this tab are taken from the subscriber profile.**

With 'Defaults' button, you may set the default values:

- Min Flashtime – 200 ms;
- Max Flashtime – 600 ms;
- Gain receive – -70 \*0.1 dB;
- Gain transmit – 0 \*0.1 dB.

'Call forward' tab—call forwarding service settings for FXS type port:

| Port 1             |                          |
|--------------------|--------------------------|
| CF Busy:           | <input type="checkbox"/> |
| CF No Reply:       | <input type="checkbox"/> |
| CF Unconditional:  | <input type="checkbox"/> |
| CF Out Of Service: | <input type="checkbox"/> |
| CFNR timeout:      | 0                        |

Apply Cancel Defaults

- *CF Busy*—when checked, CFB service is enabled—forward the call, when the subscriber is busy.
- *CF No reply*—when checked, CFNR service is enabled—forward the call, when there is no reply from the subscriber.
- *CF Unconditional*—when checked, CFU service is enabled—forward the call unconditionally.
- *CF Out Of Service*—when checked, OOS service is enabled—forward the call, when the subscriber is out of service.





For each service, the number that the call is forwarded to, is shown in the rightmost field of the row.

- *CFNR timeout*—subscriber response timeout (in seconds) for 'Call forward on no reply' service.

Against each service, there is a number that the call will be forwarded to.

«*Suppl. Service*» tab allows you to enable/disable supplementary services. For detailed description of supplementary service operations, see Section 5.1.2.6.

| Port 1                                 |                          |
|--|--------------------------|
| Call transfer                          |                          |
| Call transfer attended enable:         | <input type="checkbox"/> |
| Call transfer unattended enable:       | <input type="checkbox"/> |
| Call forward                           |                          |
| Call forward unconditional enable:     | <input type="checkbox"/> |
| Call forward on busy enable:           | <input type="checkbox"/> |
| Call forward on no answer enable:      | <input type="checkbox"/> |
| Call forward on out of service enable: | <input type="checkbox"/> |
| Others                                 |                          |
| Call waiting enable:                   | <input type="checkbox"/> |
| Do not disturb enable:                 | <input type="checkbox"/> |

Apply Cancel Defaults

'*Groups*' tab allows you to add/remove ports to/from serial groups. For detailed description of serial discovery group operations, see Section 5.1.2.7 Serial groups.

In '*Groups*' tab, you may see a the list of configured serial groups. To add port to the group, you should select the checkbox against the respective group; to remove port, deselect the checkbox:

| Port 1     |       |
|------------|-------|
| Group name | Enter |

Apply Cancel Defaults

'*PickUp*' tab — add/remove ports to/from the pickup groups.

For detailed description of pickup group operations, see Section 5.1.2.9 Pickup Group Configuration (Pickup Groups).

In '*PickUp*' tab, you may see a list of pickup groups. To add port to the group, you should select the checkbox against the respective group; to remove port, deselect the checkbox:

Custom Common Call forward Suppl. Service Groups **PickUp**

| Port 1                      |  | 1                        | 2                        | 3                        | 4                        | 5                        | 6                        | 7                        | 8                        |
|-----------------------------|--|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|
| Membership in PickUp groups |  | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
|                             |  | 9                        | 10                       | 11                       | 12                       | 13                       | 14                       | 15                       | 16                       |
| Membership in PickUp groups |  | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
|                             |  | 17                       | 18                       | 19                       | 20                       | 21                       | 22                       | 23                       | 24                       |
| Membership in PickUp groups |  | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
|                             |  | 25                       | 26                       | 27                       | 28                       | 29                       | 30                       | 31                       | 32                       |
| Membership in PickUp groups |  | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |

Apply Cancel Defaults

- *Membership in PickUp groups*—defines pickup groups that the port belongs to. Subscriber port that belongs to the group will be able to pickup the call received on any other port of this group.

To apply settings, click 'Apply' button. To reset settings to default values, click 'Defaults' button.

### Editing custom parameters of FXO type ports:

'Custom' tab—FXO type port custom settings:

Custom Common Call forward Groups

Port 25

|   |                                     |
|---|-------------------------------------|
| Phone:  | 8                                   |
| User name:  |                                     |
| Use alternative number:   | <input type="checkbox"/>            |
| Alternative number:   |                                     |
| Use alternative number as contact (only for serial groups members): | <input type="checkbox"/>            |
| Authentication name:  |                                     |
| Authentication password:  |                                     |
| Custom settings:  | <input type="checkbox"/>            |
| Subscriber profile:   | Profile 1 ▼                         |
| SIP/H323 profile:   | Profile 1 ▼                         |
| Hot line:   | <input checked="" type="checkbox"/> |
| Hot timeout:  | 0                                   |
| Hot number:   | 125                                 |
| No offhook at ringing:  | <input checked="" type="checkbox"/> |
| Use Hotline to PSTN:  | <input type="checkbox"/>            |
| Hotline Timeout to PSTN:  | 15                                  |
| Hotline Number to PSTN:   |                                     |
| CLIR:   | <input type="checkbox"/>            |
| Disabled:   | <input type="checkbox"/>            |
| SIP port:   | 5060                                |
| Process flash:  | Attended calltransfer ▼             |
| Call waiting:   | <input type="checkbox"/>            |
| MWI:  | <input type="checkbox"/>            |

Apply Cancel Defaults

- *No offhook at ringing*—this function allows to keep the loop open during ringing from TDM in IP until the voice frequency path is established to the communicating gateway via SIP protocol. This setting works with the 'Hotline' setting only.

Function operation:

- The gateway detects the ringing tone which is coming to FXO line.
- Next, TAU32M performs a call to a number configured for 'Hotline' service.
- Voice connection will be established between TAU32M and communicating gateway.

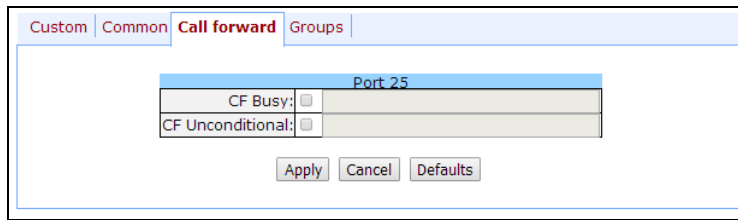
- Next, the subscriber loop closure will occur.
- *Use Hotline to PSTN*—when checked, enable 'warmline' service for calls directed from VoIP to analogue phone line, otherwise it will be disabled.
- *Hotline Timeout to PSTN*—delay timeout for the start of the automatic dialling when the 'warmline' service is enabled for calls directed from VoIP to analogue phone line. Default value is 15 seconds.
- *Hotline Number to PSTN*—number that will receive the call when the 'warmline' service is enabled for calls directed from VoIP to analogue phone line.

'Common' tab—FXO type port common settings:

| Port 25  |                                     |
|--|-------------------------------------|
| Gain receive (0.1 dB):   | 11                                  |
| Gain transmit (0.1 dB):  | 13                                  |
| SS7 category (SIP-T):  | 10                                  |
| Category:  | off                                 |
| Modifier:  | off                                 |
| Stop dial at #:  | <input type="checkbox"/>            |
| FXO parameters   |                                     |
| Outgoing direction parameters  |                                     |
| Flashtime:   | 300                                 |
| Dialtone detection:  | <input type="checkbox"/>            |
| Dialtone time detect (s):  | 5                                   |
| Dialing delay (s):   | 2                                   |
| Don't transmit prefix:   | <input checked="" type="checkbox"/> |
| Transmit number:   | <input checked="" type="checkbox"/> |
| 503 Service unavailable on busy (SIP):   | <input checked="" type="checkbox"/> |
| PSTN activity:   | Off                                 |
| Dialing  |                                     |
| Method:  | DTMF                                |
| Interdigit delay:  | 200                                 |
| Pulse time (ms):   | 80                                  |
| Pause time (ms):   | 80                                  |
| Incoming direction parameters  |                                     |
| Ring detection:  | 1                                   |
| PSTN number prefix:  |                                     |
| PSTN name prefix:  |                                     |
| Use PSTN CallerID:   | <input type="checkbox"/>            |
| Detect FXO line presence:  | <input checked="" type="checkbox"/> |
| Block FXO line in outgoing direction:  | <input type="checkbox"/>            |
| Tone detect parameters   |                                     |
| Minimum level of detectable signal (dBm):  | -36                                 |
| Dialtone detection parameters:   | 425;0(2000/0/1)                     |
| Busytone detection parameters:   | 425;1(350/350/1)                    |
| Ringback tone detection parameters:  | 425;0(1000/4000/1)                  |
| Show tone detect parameters format   |                                     |
| <input type="button" value="Apply"/> <input type="button" value="Cancel"/> <input type="button" value="Defaults"/> |                                     |

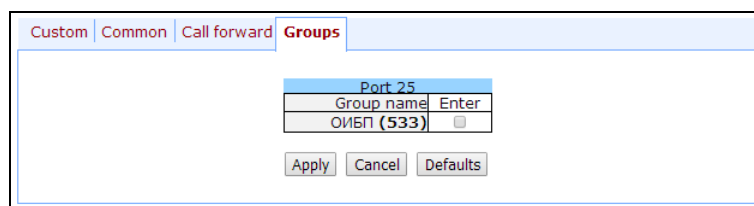
Description of 'Common' tab fields is equivalent to description of subscriber profile (Profile n) configuration menu fields shown above.

'Call forward' tab—call forwarding service settings for FXO type port:



Description of 'Call forward' tab fields is equivalent to description of similar tab fields for FXS type port shown above.

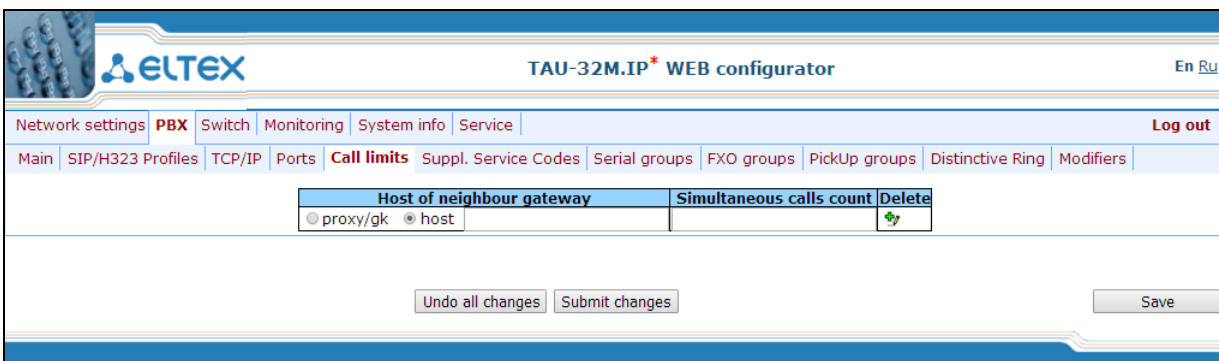
'Groups' tab allows you to add/remove ports to/from serial groups for FXO type port.



To apply settings, click 'Apply' button. To exit the submenu, click 'Cancel' button. To reset settings to default values, click 'Default' button.

### 5.1.2.5. Simultaneous Call Limits (Call limits)

In 'Call limits' submenu, you may configure simultaneous call limits for the communicating host.



- *Host of neighbour gateway*—hostname of a communicating gateway. To limit the calls via SIP-proxy or H323 Gatekeeper, select the '**proxy/gk**' checkbox (defines the total call limit through all proxies and from all profiles); to enter host address, select '**host**'.
- *Simultaneous calls count*—maximum number of simultaneous (incoming and outgoing) calls.

To discard all changes made to configuration, click 'Undo All Changes' button. To store changes to non-volatile memory of the device, click 'Save' button.

### 5.1.2.6. Configuration of Supplementary Service Codes (Suppl. Service Codes)

Supplementary services are provided to each subscriber, but in order to use a specific service, the subscriber must enable it first at the service provider. Service providers may create their own service plans containing several supplementary services. To do this, in 5.1.2.4 section, on **Suppl. Service** tab, select the checkboxes against the desired supplementary services.

Subscribers may manage state of services from their phone units. The following features are available:

- Service activation—activation and additional data input.
- Service verification.
- Service cancellation—deactivation of a service.

When the activation code is entered or the service is cancelled, subscribers may hear either a 'confirmation' tone (3 short tones), or a 'busy' tone (intermittent tone with tone/pause duration—0.35/0.35s.) 'Confirmation' tone means that the service has been activated or cancelled successfully, 'busy' tone—that this service is not enabled for this subscriber.

After service confirmation code entry, the subscriber may hear either 'PBX response' tone (continuous) or a 'busy' tone. 'PBX response' tone means that the service has been enabled and activated for the subscriber, 'busy' tone—that this service is not enabled for the subscriber.

The screenshot shows the 'Suppl. Service Codes' configuration page in the TAU-32M.IP WEB configurator. The page has a navigation menu with tabs for Network settings, PBX, Switch, Monitoring, System info, and Service. The 'Suppl. Service Codes' tab is active. Below the navigation, there is a table titled 'Supplementary Service Codes configuration:' with columns for Service, Code, Activate, Deactivate, Option, and Control. The table lists various services such as Call transfer, Call forward, and Call waiting with their respective codes and control codes. At the bottom of the table, there are buttons for 'Undo all changes', 'Defaults', 'Submit changes', and 'Save'.

| Service                         | Code | Activate | Deactivate | Option      | Control |
|---------------------------------|------|----------|------------|-------------|---------|
| Call transfer                   |      |          |            |             |         |
| Call transfer attended:         | 98   | *98#     | #98#       |             | *#98#   |
| Call transfer unattended:       | 97   | *97#     | #97#       |             | *#97#   |
| Call forward                    |      |          |            |             |         |
| Call forward unconditional:     | 21   | *21#     | #21#       | *21*option# | *#21#   |
| Call forward on busy:           | 22   | *22#     | #22#       | *22*option# | *#22#   |
| Call forward on no answer:      | 61   | *61#     | #61#       | *61*option# | *#61#   |
| Call forward on out of service: | 62   | *62#     | #62#       | *62*option# | *#62#   |
| Others                          |      |          |            |             |         |
| Call waiting:                   | 43   | *43#     | #43#       |             | *#43#   |
| Do not disturb:                 | 26   | *26#     | #26#       |             | *#26#   |

#### Supplementary Service Codes configuration:

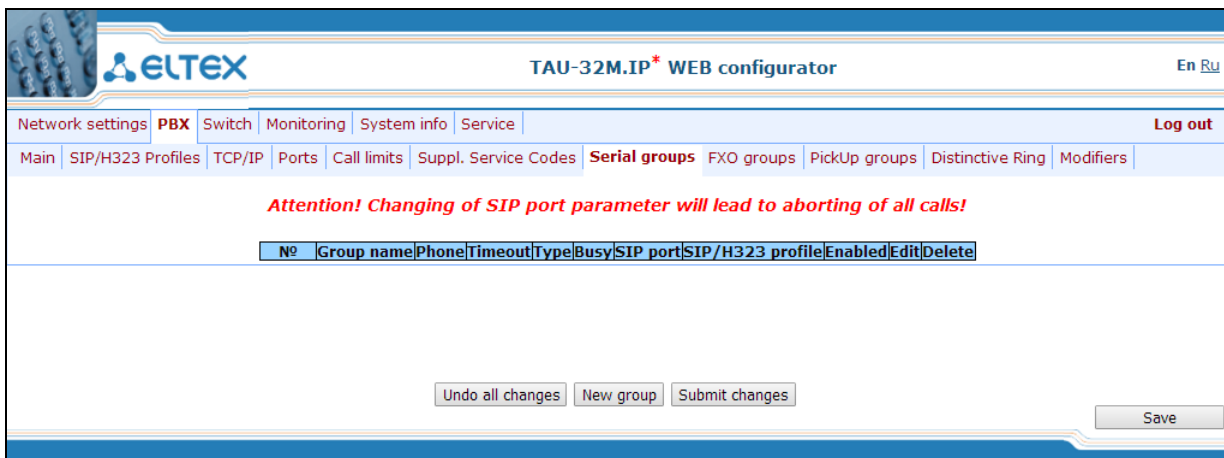
- Service—type of supplementary service:
  - *Call transfer attended*—'Call transfer' service with the wait for response of the subscriber, the call is being forwarded to.
  - *Call transfer unattended*—'Call transfer' service without the wait for response of the subscriber, the call is being forwarded to.
  - *Call forward unconditional*—'Call forward unconditional' service.
  - *Call forward on busy*—'Forward on busy' service.

- *Call forward on no answer*—'Forward on no answer' service.
- *Call forward on out of service*—'Forward on out of service' service.
- *Call waiting*—'Call waiting' service.
- *Do not disturb*—'Do not disturb' service.
- *Code*—supplementary service code.
- *Activate*—service activation.
- *Deactivate*—service cancellation.
- *Option*—access code, used for service parameters' configuration and forwarding services—a number that the call will be forwarded to.
- *Control*—service verification.

To discard all changes made to configuration, click '*Undo All Changes*' button. To set the default values, click '*Defaults*' button. To apply changes, click '*Submit Changes*' button. To store changes to non-volatile memory of the device, click '*Save*' button.

### 5.1.2.7. Serial groups

In '*Serial groups*' submenu, you may administer the call groups. You may configure up to 32 call groups in total.



**You don't have to reboot the gateway in order to apply call group settings. Changing SIP port parameter will lead to termination of current calls. Changing other parameters will disrupt the established connections for the current group only!!!**

Call groups allow to perform call center features. Gateway supports 3 call group modes: group, delayed group and search. In group mode, the call comes in to all free ports of the group simultaneously. When one of the group members answers, call transmission to other ports stops. In the delayed group mode, the call comes in to the first free port in the group list, and then, after the specific timeout, the next free port in the list will be added to the main one, and so on. When one of the group members answers, call transmission to other ports stops. In the search mode, the gateway continuously searches for a free group member, and the call is transferred to their number.

To add a new group, click 'New group' button:

- *Group name*—name of the group (used for SIP server authentication).
- *Password*—password (used for SIP server authentication).
- *Phone*—call group phone number.
- *Timeout*—group member call timeout (used for group types 'serial calling' and 'cycle'), in seconds.
- *Group type*—call group type:
  - *Group calling*—call comes in to all group ports simultaneously.
  - *Serial calling*—call comes in to all ports in turns depending on the selected group member call timeout (when zero value is defined for call timeout, the call will be transferred to the next port, only if higher ports in a queue are busy).
  - *Cycle*—search begins from the first port in the call group.
- *Busy mode*—incoming call processing mode for situations when all group ports are busy (clear—call clearback, wait—call queueing).
- *SIP/H323 profile*—SIP/H323 profile number, that will be used for the current group.
- *Enabled*—when checked, the call group is enabled.



**If the call group does not contain any ports, the group will not be used even with 'Enabled' flag checkbox selected.**

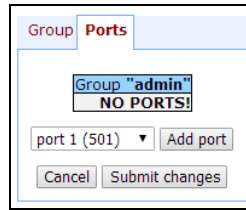
- *SIP port*—local UDP port used for group operations via SIP protocol.

To edit parameters of an existing group, click button in the corresponding row.

'Group'—group settings:

For description of menu fields, see above.

'Ports'—group ports:

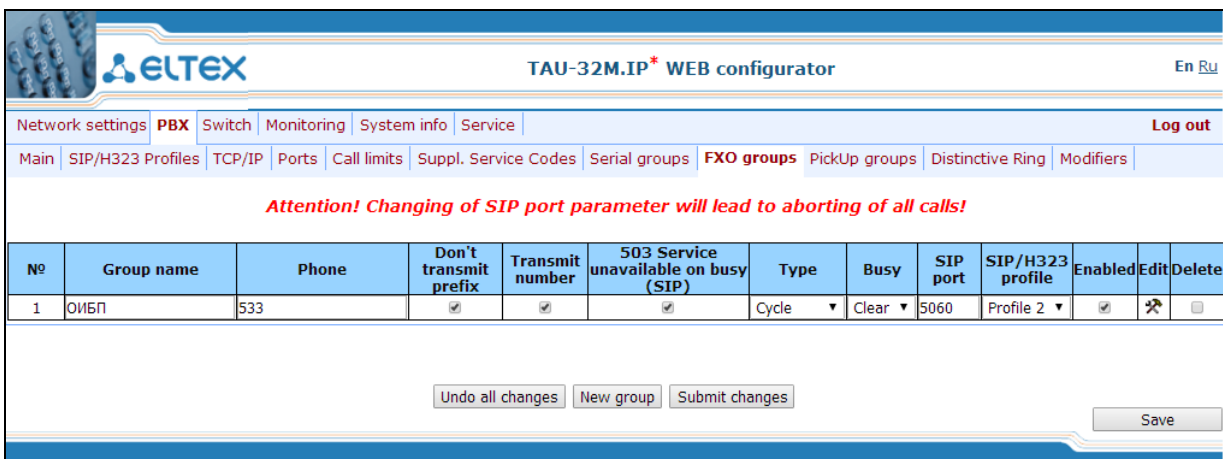


To add a port to a group, select the desired port from the drop-down list and click 'Add port' button.

To change the order of ports in a group, use arrow buttons (up, down); to delete a port from a group, click button.

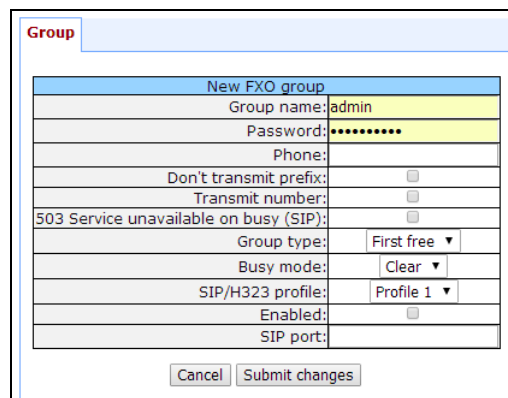
### 5.1.2.8. Configuration of FXO Groups (FXO groups)

In 'FXO groups' submenu, you may administer FXO groups. You may configure up to 32 call groups in total.



**You don't have to reboot the gateway in order to apply FXO group settings. Changing SIP port parameter will lead to termination of current calls. Changing other parameters will disrupt the established connections for the current group only!!!**


To add a new group, click 'New group' button:



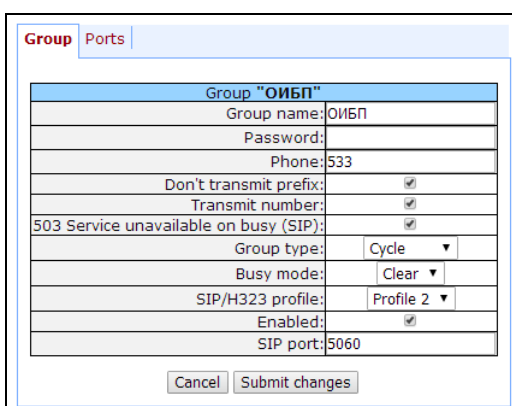
- *Group name*—name of the group (used for SIP server authentication).



- *Password*—password (used for SIP server authentication).
- *Phone*—FXO group phone number.
- *Don't transmit prefix*—when checked, transmit the complete number received from IP (from Request URI header of INVITE request) into the line, except for FXO unit subscriber number, otherwise it will not be transmitted.
- *Transmit number*—when checked, transmit the full number received from IP (from Request URI header of INVITE request) into the line, including FXO unit subscriber number, otherwise it will not be transmitted.
- *503 Service unavailable on busy (SIP)*—if checked, reply 503 will be sent via SIP protocol when all subscriber lines (FXO) are busy, otherwise—486.
- *Group type*—line selection mode:
  - *First free*—selection of a free line for a call is performed relatively for the first port in the call group.
  - *Cycle*—free line selection for a call is based on the port, which was busy last in a call group.
- *Busy mode*—*incoming call processing mode for situations when all group ports are busy* (clear—*call clearback*, wait—*call queueing*).
- *SIP/H323 profile*—SIP/H323 profile number, that will be used for the current group.
- *Enabled*—when checked, FXO group is enabled.
- *SIP port*—local UDP port used for group operations via SIP protocol.

To edit parameters of an existing group, click  button in the corresponding row.

'Group' tab—group settings:

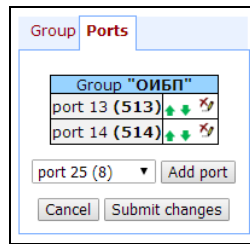


| Group "ОИБП"                           |                                     |
|--|-------------------------------------|
| Group name:                            | ОИБП                                |
| Password:                              |                                     |
| Phone:                                 | 533                                 |
| Don't transmit prefix:                 | <input checked="" type="checkbox"/> |
| Transmit number:                       | <input checked="" type="checkbox"/> |
| 503 Service unavailable on busy (SIP): | <input checked="" type="checkbox"/> |
| Group type:                            | Cycle ▼                             |
| Busy mode:                             | Clear ▼                             |
| SIP/H323 profile:                      | Profile 2 ▼                         |
| Enabled:                               | <input checked="" type="checkbox"/> |
| SIP port:                              | 5060                                |

Cancel Submit changes

For description of menu fields, see above.

'Ports' tab—group ports:



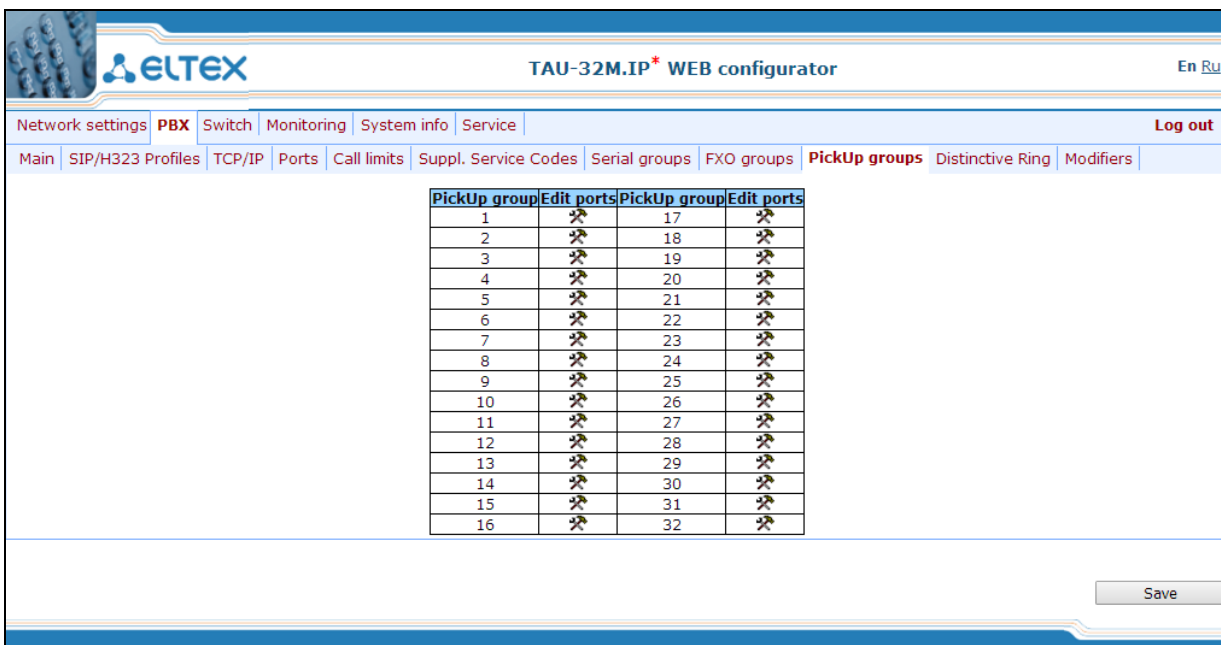
To add a port to a group, select the desired port from the drop-down list and click 'Add port' button. To change the order of ports in a group, use arrow buttons (up, down); to delete a port from a group, click button. To quit without saving settings, click 'Cancel' button.

To discard all changes made to configuration, click 'Undo All Changes' button. To apply changes, click 'Submit Changes' button. To store changes to non-volatile memory of the device, click 'Save' button.

### 5.1.2.9. Pickup Group Configuration (Pickup Groups)

In 'PickUp groups' submenu, you may configure pickup groups. You may configure up to 32 different pickup groups in total.

Pickup group—subscriber group, authorized to receive (or intercept) any calls directed at another subscriber of the group. I.e. each subscriber port that belongs to the group will be able to pickup the call received on any other port of this group by dialling a pickup code. To configure a pickup code, use 'PBX/SIP-H323 Profiles/Profile n/Dialplan' tab; for description, see Section 5.1.2.2.6.3 Configuration of pickup codes.



- *PickUp group*—pickup group sequential number [1 .. 32].
- *Edit ports*—edit pickup group parameters. To edit pickup group parameters, click icon in the corresponding row:

| PickUp group "6" |                          |                          |                          |                          |                          |                          |                          |                          |
|------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|
| Membership       |                          |                          |                          |                          |                          |                          |                          |                          |
| Port             | 1                        | 2                        | 3                        | 4                        | 5                        | 6                        | 7                        | 8                        |
| Enable           | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| Port             | 9                        | 10                       | 11                       | 12                       | 13                       | 14                       | 15                       | 16                       |
| Enable           | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| Port             | 17                       | 18                       | 19                       | 20                       | 21                       | 22                       | 23                       | 24                       |
| Enable           | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| Port             | 25                       | 26                       | 27                       | 28                       | 29                       | 30                       | 31                       | 32                       |
| Enable           | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
|                  |                          | Enable all               |                          | Disable all              |                          |                          |                          |                          |
| Cancel           |                          |                          |                          | Submit changes           |                          |                          |                          |                          |

- *Port*—subscriber port number.
- *Enable*—when checked, the port belongs to the pickup group; otherwise, it does not belong to this group.

When '*Enable*' checkbox is selected against the subscriber port, this port is included into the pickup group; otherwise, it is excluded from this group. To set permissions for all subscriber ports, click '*Enable all*' button. To deselect checkboxes for all subscriber ports, click '*Disable all*' button.



**If you need to add a port into multiple groups at once, use 'PBX/Ports/Edit port /PickUp' menu.**

To quit the pickup group configuration dialog without saving, click 'Cancel' button. To save changes, click '*Submit Changes*' button. To store changes to non-volatile memory of the device, click '*Save*' button.

### **Service usage:**

The call comes in to the phone unit of a subscriber that belongs to the pickup group. If the subscriber is unavailable or cannot answer the call for some reason, another subscriber that belongs to that group may answer the incoming call. To do this, they should pick up the phone and dial a pickup code, and the connection with the caller will be established after that.

Pickup group may be used in combination with a call group; in this case, all ports that belong to a call group should belong to the pickup group as well. Thus, each port that belong to a call group will be able to pickup an incoming call to a group number.

When subscriber dials the pickup code when there are no incoming calls to a group number, they will hear 'busy' tone.



**Pickup group operation will not be possible for calls coming in via SIP protocol with a ringback sent to the caller ('Remote ringback' setting) or via H.323 protocol (except for the calls that do not employ faststart and tunnelling.)**

### 5.1.2.10. 'Distinctive Ring' Service Configuration

This setting allows for the non-standard ringing to the callee, which allows to identify the number/group of numbers that the call is originated from. In total, 32 variations of the 'distinctive ring' may be used.

The screenshot shows the 'Distinctive Ring' configuration page in the ELTEX TAU-32M.IP WEB configurator. The page has a navigation menu at the top with options like 'Network settings', 'PBX', 'Switch', 'Monitoring', 'System info', 'Service', 'Main', 'SIP/H323 Profiles', 'TCP/IP', 'Ports', 'Call limits', 'Suppl. Service Codes', 'Serial groups', 'FXO groups', 'PickUp groups', 'Distinctive Ring', and 'Modifiers'. Below the menu are buttons for 'Undo all changes' and 'Submit changes'. The main content is a table with 32 rows, each representing a rule. The table has columns for '№', 'Rule', 'Ring, msec', 'Pause, msec', and 'Subscriber profiles' (with sub-columns 1-8). Each row contains a rule number, a mask, and ring/pause durations, with checkboxes for subscriber profiles. At the bottom, there are 'Undo all changes', 'Submit changes', and 'Save' buttons.

| №  | Rule | Ring, msec |      | Pause, msec |      | Subscriber profiles      |                          |                          |                          |                          |                          |                          |                          |                          |
|----|------|------------|------|-------------|------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|--------------------------|
|    |      |            |      |             |      | 1                        | 2                        | 3                        | 4                        | 5                        | 6                        | 7                        | 8                        |                          |
| 1  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 2  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 3  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 4  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 5  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 6  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 7  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 8  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 9  |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 10 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 11 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 12 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 13 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 14 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 15 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 16 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 17 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 18 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 19 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 20 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 21 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 22 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 23 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 24 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 25 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 26 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 27 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 28 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 29 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 30 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 31 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| 32 |      | 2          | x100 | 2           | x100 | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |

- *Rule*—mask of the number of the caller that will trigger the 'distinctive ring' with a call to the requested port.
- *Ring*—ringing duration.
- *Pause*—pause duration.
- *Subscriber profiles*—subscriber profiles which ports are affected by this rule.

#### Caller number mask record rule:

Rule1| Rule2|..| RuleN

### Caller number mask syntax:

- |—logical OR—used to separate rules.
- X or x—any number from 0 to 9, equal to a range [0-9].
- 0 - 9—numbers from 0 to 9.
- \*—\* character.
- #—# character.
- [ ]—define ranges (with a hyphen), or enumeration (w/o spaces, commas, and other characters between the digits), e.g.
  - Range: [1-5]—1,2,3,4, or 5.
  - Enumeration: [138]—1,3, or 8.
  - Range and enumeration: [0-9\*#]—0 to 9, and also \* and #.
- {min,max}—define the repetition count for a character located outside the parentheses, a range or \*# symbols.

*min*—minimum repetition count, *max*—maximum repetition count.

{,max}—equal to {0,max}.

{min,}—equal to {min,inf.}.

Example:

5{2,5}—caller's number may be equal to 55, 555, 5555, or 55555.

- . – 'dot' special symbol means that a preceding digit, range, or '\*', '#' characters may be repeated from one to infinity times. Equivalent to a record {0,}.

Example:

5x.\* — 'x' in this rule may be completely absent or may be present any number of times. Caller number may be equal to 5\*, 5x\*, 5xx\*, 5xxx\*, ...

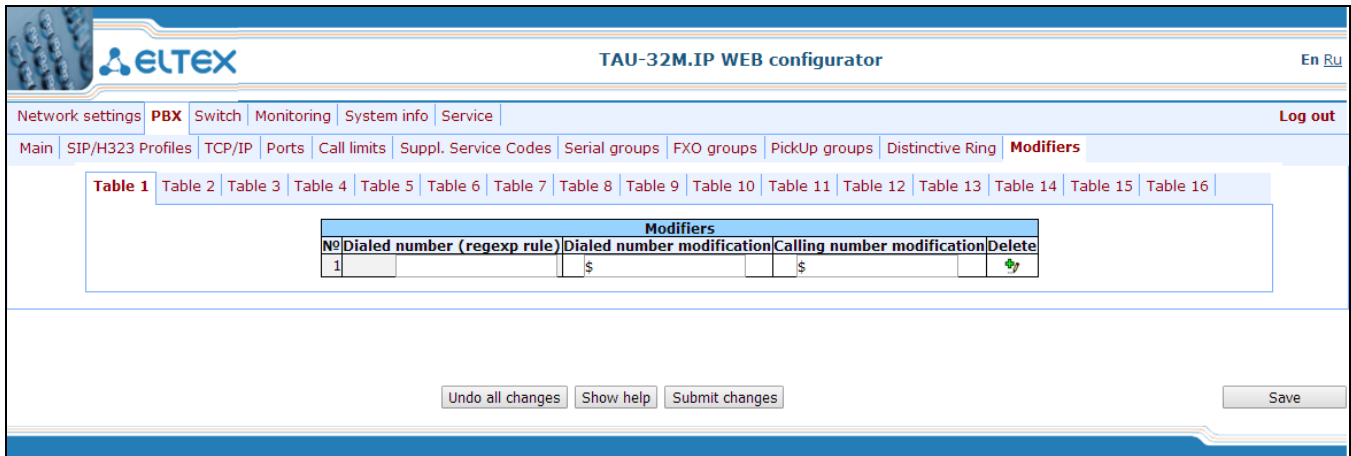
- +—digit, range, or '\*', '#' characters preceding the '+' symbol may be repeated from one to infinity times. Equivalent to a record {1,}.

#### 5.1.2.11. Modifiers

This setting allows for the modification of the associated and dialled numbers depending on the call direction. Modifiers are used in outgoing calls.



**Modifiers work only when routing rules are used, described with regular expressions (5.1.2.2.6.1 Configuration of Routing Rules); at that, in number modification routing rules, <:> characters should not be used.**



The gateway allows you to configure 16 modifier groups, each group contains one or several modification rules:

- *Dialed number (regexp rule)*—dialled number mask.
- *Dialed number modification*—dialled number modification rule.
- *Calling number modification*—modification rule for TAU subscriber's number (caller's number).

**Dialled number mask record rule:**

Rule1| Rule2|..| RuleN

**Caller number mask syntax:**

- |—logical **OR**—used to separate rules.
- **X** or **x**—any number from 0 to 9, equal to a range [0-9].
- **0 - 9**—numbers from 0 to 9.
- **\***—\* character.
- **#**—# character.
- **[ ]**—define ranges (with a hyphen), or enumeration (w/o spaces, commas, and other characters between the digits), e.g.

Range: **[1-5]**—1,2,3,4, or 5.

Enumeration: **[138]**—1,3, or 8.

Range and enumeration **[0-9\*#]**—0 to 9, and also \* and #.

- **{min,max}**—define the repetition count for a character located outside the parentheses, a range or \*# symbols.  
*min*—minimum repetition count, *max*—maximum repetition count.

**{,max}**—equal to {0,max}.

**{min,}**—equal to {min,inf.}.

Example:

**5{2,5}**—dialled number may be equal to 55, 555, 5555, or 55555

- . – 'dot' special symbol means that a preceding digit, range, or '\*', '#' characters may be repeated from one to infinity times. Equivalent to a record {0,}.

Example:

**5x.\*** —'x' in this rule may be completely absent or may be present any number of times.  
Dialled number may be equal to 5\*, 5x\*, 5xx\*, 5xxx\*, ...

- +—digit, range, or '\*', '#' characters preceding the '+' symbol may be repeated from one to infinity times. Equivalent to a record {1,}.

#### Modification rule syntax:

- – or . – digit deletion.
- **X** or **x**—digit/symbol or character in this position remains unchanged.
- **?** —digit/symbol in this position remains unchanged.
- **+**—addition of the succeeding digits/symbols (0-9, \*, #).
- **!** —breakdown finish, all other digits of a number are truncated.
- **\$**—breakdown finish, all other digits of a number remain unchanged.
- **0-9, #** and **\*** (without '+' sign)—substitution of a digit in this position.

Example:

When calling to six-digit numbers, beginning with 5 and 6, you need to transform the subscriber number in such manner as to add 383 prefix into the beginning of the subscriber number, and replace the first digit of the dialled number to 7.

Dialled number: [5-6]xxxxx

Dialled number modification: 7xxxxx

Calling number modification: +383\$

### 5.1.3. Switch

In 'Switch' menu, you may configure switch ports.

#### 5.1.3.1. Switch ports settings

In 'Switch ports settings' submenu, you may configure parameters of integrated Ethernet switch ports.

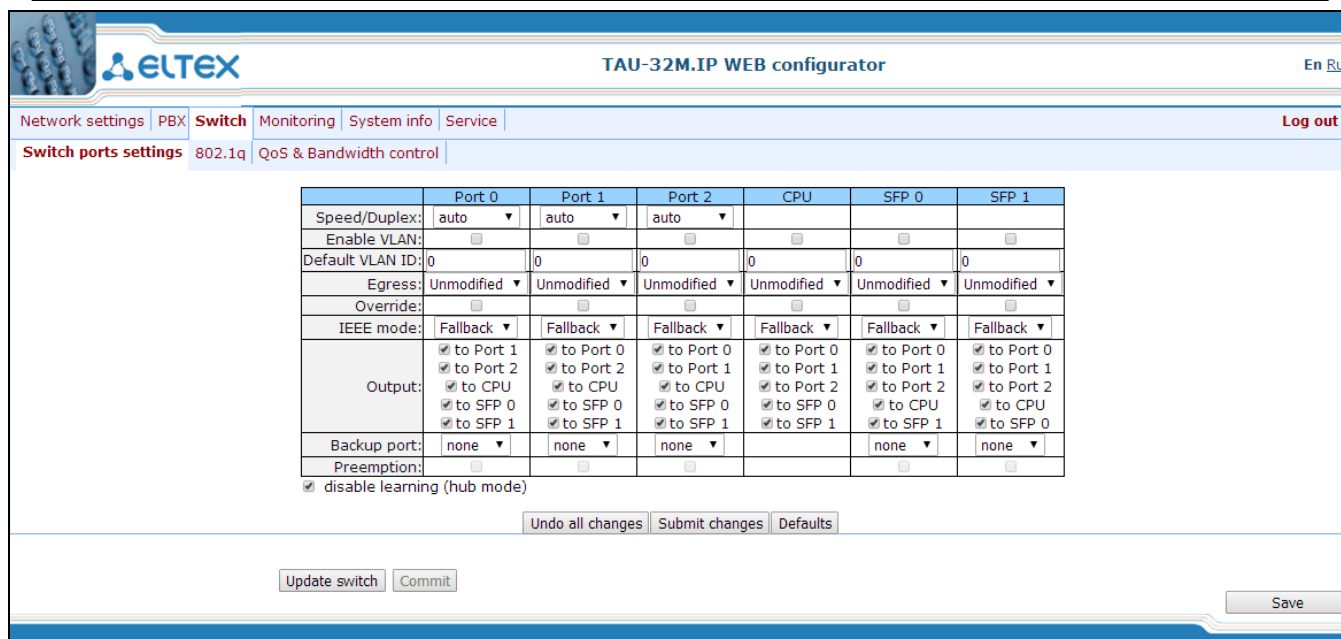
##### 5.1.3.1.1. Configuration

The switch can operate in four modes:

1. **Without VLAN settings**—to use this mode, *Enable VLAN* checkboxes should be deselected for all ports, *'IEEE Mode'* value should be set to *'Fallback'* for all ports, mutual availability of data ports should be set to *'Output'* with the respective checkboxes. '802.1q' routing table in '802.1q' tab should not contain any records.
2. **Port based VLAN**—to use this mode, *'IEEE Mode'* value should be set to *'Fallback'* for all ports, mutual availability of data ports should be set to *'Output'* with the respective checkboxes. For VLAN operation, use *'Enable VLAN'*, *'Default VLAN ID'*, *'Egress'*, and *'Override'* settings. '802.1q' routing table in '802.1q' tab should not contain any records.
3. **802.1q**—to use this mode, *'IEEE Mode'* value should be set to *'Check'* or *'Secure'* for all ports. For VLAN operation, use *'Enable VLAN'*, *'Default VLAN ID'*, and *'Override'* settings. Also, routing rules described in '802.1q' routing table in '802.1q' tab will apply.
4. **802.1q + Port based VLAN.** 802.1q mode may be used in combination with 'Port based VLAN'. In this case, *'IEEE Mode'* value should be set to *'Fallback'* for all ports, mutual availability of data ports should be set to *'Output'* with the respective checkboxes. For VLAN operation, use *'Enable VLAN'*, *'Default VLAN ID'*, *'Egress'*, and *'Override'* settings. Also, routing rules described in '802.1q' routing table in '802.1q' tab will apply.

For example of switch configuration using VLAN, see Appendix D.





Gateway switch is equipped with 3 x electrical Ethernet ports and 1 x port for CPU interactions:

- *port0, port1, port2*—electrical Ethernet ports of the device.
- *CPU*—internal port linked to the device CPU.
- *SFP0, SFP1*—optical (SFP) Ethernet ports of the device.

Switch settings:

- *Speed/Duplex*—speed and duplex settings of electrical Ethernet ports. Optical ports support only one mode: 1000 full duplex.
- *Enable VLAN*—when checked, enable '*Default VLAN ID*', '*Override*' and '*Egress*' settings for this port, otherwise they will be disabled.
- *Default VLAN ID*—when an untagged packet is received at the port, this will be its VID; when a tagged packet is received at that port, its VID is considered to be specified in its VLAN tag.
- *Egress*:
  - *unmodified*—packets will be sent by the port without any changes (i.e. as they came to another switch port).
  - *untagged*—packets will always be sent without VLAN tag by this port.
  - *tagged*—packets will always be sent with VLAN tag by this port.
  - *double tag*—each packet will be sent with two VLAN tags—if received packet was tagged and came with one VLAN tag—if the received packet was untagged.
- *Override*—when checked, it is considered that any received packet has a VID, defined in '*default VLAN ID*' row. True for both untagged and tagged packets.
- *IEEE mode*:
  - *disabled*—for a packet received by this port, routing rules described in the «*output*» section of the table will be applied.
  - *fallback*—if a packet with VLAN tag is received through this port, and there is a record in a '802.1q'

routing table for this packet, then it falls within a scope of routing rules, specified in the record of this table; otherwise, routing rules specified in 'egress' and 'output' will be applied to it.

- *check*—if a packet with VID is received through the port, and there is a record in a '802.1q' routing table for this packet, then it falls within a scope of routing rules, specified in the current record of this table, even if this port does not belong to the group of this VID. Routing rules specified in 'egress' and 'output' will not apply to this port.
- *secure*—if a packet with VID is received through the port, and there is a record in a '802.1q' routing table for this packet, then it falls within a scope of routing rules, specified in the current record of this table; otherwise, it is rejected. Routing rules specified in 'egress' and 'output' will not apply to this port.
- *Output*—mutual availability of data ports. Defines privileges that allow packets received by this port to be transferred to flagged ports.
- *Backup port*—select a port from the list as a backup port. Used in direction reservation mode.
- *Preemption*—returns to master port on its availability. Used in direction reservation mode.



**'Backup port' and 'Preemption' are used for direction reservation. In this case, main and backup ports are connected to a single switch with Ethernet cables. Backup port should be connected only when switch settings has been applied and saved.**

- *Hubmode*—Ethernet switch operation in hub mode. In hub mode, Ethernet switch will not learn MAC addresses of devices, that send packets, and all packets will be transferred to all switch ports. We recommend using this mode for network traffic mirroring from the switch ports to PC (tracing) only.

'Update Switch' and 'Commit' buttons allow to retain access to the switch when switch settings are applied. Click 'Commit' button in 30 seconds interval to confirm newly applied settings, or the previous settings will be restored.

- *Update Switch*—apply switch settings without restart.
- *Commit*—confirm applied settings.

Use 'Defaults' button to set default parameters (the figure below shows default values).

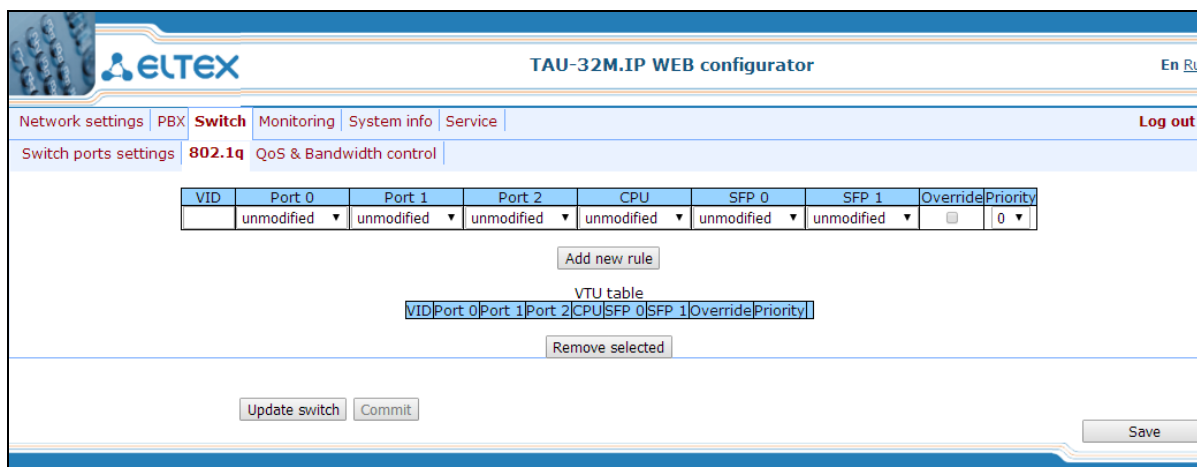
#### 5.1.3.1.2. Tracing, Network Traffic Mirroring

To perform tracing, you should do the following:

1. Configure hub mode—in 'Switch' tab, select 'Hubmode' checkbox, then click 'Update Switch' and 'Commit' buttons consequently.
2. Connect a PC to perform the tracing directly to TAU-32M.IP Ethernet port.
3. Run the application on the PC that captures network traffic. In the application, select Ethernet interface connected to TAU-32M.IP as a traffic capture interface.
4. After tracing, save captured traffic into a file.

### 5.1.3.2. 802.1q

In '802.1q' submenu, you may define the configuration of packet routing rules for switch operation in 802.1q mode.



Gateway switch is equipped with 3 x electrical Ethernet ports and 1 x port for CPU interactions:

- *port0*, *port1*, *port2*—electrical Ethernet ports of the device.
- *CPU*—internal port linked to the device CPU.
- *SFP0*, *SFP1*—optical (SFP) Ethernet ports of the device.

Adding records to the packet routing table (16 rules max.): in 'VID' field, enter an identifier of VLAN group, that the routing rule is created for, and assign actions for each port to be performed during transfer of packets with specified VID.

- *unmodified*—packets will be sent by the port without any changes (i.e. as they have been received).
- *not member*—packets with specified VID will not be sent by this port (i.e. the port is not the member of VLAN).
- *untagged*—packets will always be sent without VLAN tag by this port.
- *tagged*—packets will always be sent with VLAN tag by this port.
- *override*—when checked, override 802.1p priority for this VLAN; otherwise, leave the priority unchanged.
- *priority*—802.1p priority assigned to packets by VLAN, if 'override' checkbox is selected.

Then, click 'Add New Rule' button.

To remove records, select checkboxes for the rows to be removed and click 'Remove selected' button.



**'Update Switch' and 'Commit' buttons allow to retain access to the switch when switch settings are applied. Click 'Commit' button in 30 seconds interval to confirm newly applied settings, or the previous settings will be restored.**

### 5.1.3.3. QoS & Bandwidth control

In 'QoS & Bandwidth control' submenu, you may configure Quality of Service functions and bandwidth restrictions.

The screenshot shows the 'QoS & Bandwidth control' configuration page for the TAU-32M.IP device. The page is divided into several sections for different ports: Port 0, Port 1, Port 2, CPU, SFP 0, and SFP 1. Each port has its own set of configuration options.

|                              | Port 0                   | Port 1                   | Port 2                   | CPU                                 | SFP 0                    | SFP 1                    |
|------------------------------|--------------------------|--------------------------|--------------------------|-------------------------------------|--------------------------|--------------------------|
| Default VLAN priority:       | 0                        | 0                        | 0                        | 0                                   | 0                        | 0                        |
| QoS mode:                    | 802.1p preferred         | 802.1p preferred         | 802.1p preferred         | 802.1p preferred                    | 802.1p preferred         | DSCP only                |
| Remapping 802.1p priority 0: | 0                        | 0                        | 0                        | 0                                   | 0                        | 0                        |
| 1:                           | 1                        | 1                        | 1                        | 1                                   | 1                        | 1                        |
| 2:                           | 2                        | 2                        | 2                        | 2                                   | 2                        | 2                        |
| 3:                           | 3                        | 3                        | 3                        | 3                                   | 3                        | 3                        |
| 4:                           | 4                        | 4                        | 4                        | 4                                   | 4                        | 4                        |
| 5:                           | 5                        | 5                        | 5                        | 5                                   | 5                        | 5                        |
| 6:                           | 6                        | 6                        | 6                        | 6                                   | 6                        | 6                        |
| 7:                           | 7                        | 7                        | 7                        | 7                                   | 7                        | 7                        |
| Ingress limit mode:          | mult_broad               | mult_broad               | mult_broad               | mult_broad                          | mult_broad               | mult_broad               |
| Ingress rate prio 0 (kbps):  | 50000                    | 50000                    | 50000                    | 50000                               | 50000                    | 50000                    |
| Ingress rate prio 1:         | previous                 | previous                 | previous                 | previous                            | previous                 | previous                 |
| Ingress rate prio 2:         | previous                 | previous                 | previous                 | previous                            | previous                 | previous                 |
| Ingress rate prio 3:         | previous                 | previous                 | previous                 | previous                            | previous                 | previous                 |
| Egress limit on:             | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input checked="" type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| Egress rate limit (kbps):    | 0                        | 0                        | 0                        | 50000                               | 0                        | 0                        |

| 802.1p | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 |
|--------|---|---|---|---|---|---|---|---|
| Queue: | 1 | 0 | 0 | 1 | 2 | 2 | 3 | 3 |

| Diffserv | Queue | Diffserv | Queue | Diffserv | Queue | Diffserv | Queue |
|----------|-------|----------|-------|----------|-------|----------|-------|
| 0x00     | 0     | 0x40     | 1     | 0x80     | 2     | 0xC0     | 3     |
| 0x04     | 0     | 0x44     | 1     | 0x84     | 2     | 0xC4     | 3     |
| 0x08     | 0     | 0x48     | 1     | 0x88     | 2     | 0xC8     | 3     |
| 0x0C     | 0     | 0x4C     | 1     | 0x8C     | 2     | 0xCC     | 3     |
| 0x10     | 0     | 0x50     | 1     | 0x90     | 2     | 0xD0     | 3     |
| 0x14     | 0     | 0x54     | 1     | 0x94     | 2     | 0xD4     | 3     |
| 0x18     | 0     | 0x58     | 1     | 0x98     | 2     | 0xD8     | 3     |
| 0x1C     | 0     | 0x5C     | 1     | 0x9C     | 2     | 0xDC     | 3     |
| 0x20     | 0     | 0x60     | 1     | 0xA0     | 2     | 0xE0     | 3     |
| 0x24     | 0     | 0x64     | 1     | 0xA4     | 2     | 0xE4     | 3     |
| 0x28     | 0     | 0x68     | 1     | 0xA8     | 2     | 0xE8     | 3     |
| 0x2C     | 0     | 0x6C     | 1     | 0xAC     | 2     | 0xEC     | 3     |
| 0x30     | 0     | 0x70     | 1     | 0xB0     | 2     | 0xF0     | 3     |
| 0x34     | 0     | 0x74     | 1     | 0xB4     | 2     | 0xF4     | 3     |
| 0x38     | 0     | 0x78     | 1     | 0xB8     | 2     | 0xF8     | 3     |
| 0x3C     | 0     | 0x7C     | 1     | 0xBC     | 2     | 0xFC     | 3     |

Buttons: Undo all changes, Submit changes, Defaults, Update switch, Commit, Save

- *Default vlan priority*—802.1p priority assigned to untagged packets, received by this port. If *802.1p* or *IP diffserv* priority is already assigned to the packet, this setting will not be used ('default vlan priority' will not be applied to packets containing IP header, when one of the QoS modes is in use: *DSCP only*, *DSCP preferred*, *802.1p preferred*, and also to untagged packets.
- *QoS mode*—QoS operation mode:
  - *DSCP only*—distribute packets into queues based on IP diffserv priority only.
  - *802.1p only*—distribute packets into queues based on 802.1p priority only.

- *DSCP preferred*—distribute packets into queues based on IP diffserv and 802.1p priorities, if both priorities are present in the packet, IP diffserv priority is used for queuing purposes.
  - *802.1p preferred*—distribute packets into queues based on IP diffserv and 802.1p priorities, if both priorities are present in the packet, 802.1p priority is used for queuing purposes.
- *Remapping 802.1p priority*—remap 802.1p priorities for untagged packets. Thus, a new value may be assigned for each priority received in VLAN packet.
  - *ingress limit mode*—restriction mode for traffic coming to the port.
    - *off*—no restriction.
    - *all*—restrict all traffic.
    - *mult\_flood\_broad*—multicast, broadcast, and flooded unicast traffic will be restricted.
    - *mult\_broad*—multicast and broadcast traffic will be restricted.
    - *broad*—only broadcast traffic will be restricted.



**This mode is not suitable for restriction of TCP/IP traffic coming to the port. It was designed to prevent the broadcast storm. If you try to restrict TCP/IP traffic using this mode, the result will not match the configured value.**

- *ingress rate prio 0 (kpbs)*—bandwidth restriction for incoming port traffic, priority 0. Permitted values— from 70 to 250000kpbs.
- *ingress rate prio 1*—bandwidth restriction for incoming port traffic, priority 1. You can double the bandwidth (prev prio \*2) of priority 0, or leave it unchanged (same as prev prio).
- *ingress rate prio 2*—bandwidth restriction for incoming port traffic, priority 2. You can double the bandwidth (prev prio \*2) of priority 1, or leave it unchanged (same as prev prio).
- *ingress rate prio 3*—bandwidth restriction for incoming port traffic, priority 3. You can double the bandwidth (prev prio \*2) of priority 2, or leave it unchanged (same as prev prio).
- *Egress limit on*—enable the bandwidth restriction for outgoing port traffic.
- *egress rate limit*—bandwidth restriction for outgoing port traffic. Permitted values—from 70 to 250000kpbs.
- *802.1p priorities mapping*—allows to distribute packets into queues depending on the 802.1p priority:
  - *802.1p*—802.1p priority value.
  - *Queue*—outgoing queue number.
- *IP diffserv priorities mapping*—allows to distribute packets into queues depending on the IP diffserv priority (for basic diffserv values, see Table 7):
  - *diffserv*—IP diffserv priority value.
  - *Queue*—outgoing queue number.

For basic diffserv values, see Table 8.



Queue 3 has the highest priority, queue 0—the lowest priority. Weighted packet distribution to outgoing queues 3/2/1/0 is as follows: 8/4/2/1.

### 5.1.4. Monitoring

In 'Monitoring' menu, you may monitor the device status.

#### 5.1.4.1. Subscriber Port Monitoring (Port)

In 'Port' submenu, you may view the information on device subscriber port status.

| Features: |                |            |        |               |                    |                      |                         |               |          |                    |
|-----------|----------------|------------|--------|---------------|--------------------|----------------------|-------------------------|---------------|----------|--------------------|
| Port      | State          | Start time | Number | Dialed digits | Registration state | Last registration at | Next registration after | H.323 GK      | Test     | FXS/FXO statistics |
| Port 1:   | FXS 501 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |
| Port 2:   | FXS 502 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |
| Port 3:   | FXS 503 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |
| Port 4:   | FXS 504 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |
| Port 5:   | FXS 505 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |
| Port 6:   | FXS 506 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |
| Port 7:   | FXS 507 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |
| Port 8:   | FXS 508 onhook |            |        |               | off                | not connected        | not connected           | not connected | run test | get stat           |

#### Features:

- *Port*—subscriber port.
- *State*—number, configured on the port, port state, last known reason for port blocking:
  - onhook—phone is onhook.
  - offhook—phone is offhook.
  - dial—dialling number.
  - ringback—sending 'ringback' tone.
  - ringing—sending 'ringing' tone.
  - talking—call in progress.
  - conference—3-way conference.
  - busy—sending 'busy' tone.
  - hold—port is on hold.
  - blocked—port is blocked.

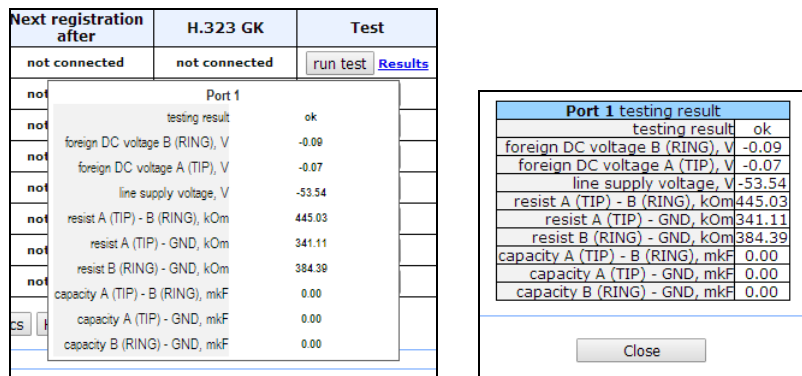
- testing—port is in testing mode.
- Number—number of the remote subscriber or two subscribers in conference mode.
- *Dialed digits*—digits dialed by the port before modification according to the routing plan.
- Registration state—SIP server registration status:
  - off—registration disabled.
  - ok—successful registration.
  - Failed—registration failed.
- *Last registration at*—last known successful registration on SIP server.
- *Next registration after*—remaining time for SIP server registration renewal.
- *H.323 GK*—H.323 gatekeeper registration time.
- *Test*—testing parameters of a subscriber line corresponding to this port.
- *FXS statistic*—request statistics of voice traffic transmission for this port.

If port was in 'blocked' state, then 'Last block cause' link will be active (reason and time of the last known port blocking):

- *leakage current has exceeded the permissible parameters*—leakage current block.
- *temperature current has exceeded the permissible parameters*—temperature block.
- *power dissipation has exceeded the permissible parameters*—power dissipation block.
- *reinitialization by changing the input voltage*—port reinitialization due to input voltage fluctuations.
- *hardware reset*—hardware reset.
- *low Vbat level*—low input voltage level.
- *FXS port out of order*—port is out of order/faulty.
- *Receiver offhook*—offhook block. If the subscriber's phone is offhook, and the 'busy' tone is played, after the expiry of two-minute interval the 'Receiver offhook' tone will be played to the subscriber's phone, and the port will switch into the blocked state.

If the port is already in 'blocked' state, and the 'Last block cause' link is inactive, it means that the port was blocked when the phone is offhook. This blocking will be performed after the 'busy' tone is played to the subscriber's phone for two minutes. Upon the expiry of two-minute interval, a loud triple-tone will be played to the subscriber's phone notifying them that the phone is offhook.

'Run test' button, located against each port, allows to test the subscriber line associated with this port. When the button is pressed, the test will be executed (it may take up to one minute.) To see the results when the test finishes, hover the mouse cursor over the 'result' link located against the respective port, or open the test results window by clicking the link:



Description of 'Port test results' informational window:

- *Common result*—test result status.
- Foreign DC voltage B (RING), V—foreign voltage in B wire (RING), V.
- Foreign DC voltage A (TIP), V—foreign voltage in A wire (TIP), V.
- *Line supply voltage*, V—line power supply voltage, V.
- *Ringing voltage*, V—call voltage, V.
- Resist A (TIP)–B (RING), kOm—resistance between A (TIP) and B (RING) wires, kΩ.
- *Resist A (TIP)-GND*, kOm—resistance between A (TIP) wire and ground GND, kΩ.
- *Resist B (RING)-GND*, kOm—resistance between B (RING) wire and ground GND, kΩ.
- Capacity A (TIP)–B (RING), mkF—capacity between A (TIP) and B (RING) wires, μF.
- *Capacity A (TIP)-GND*, mkF—capacity between A (TIP) wire and ground GND, μF.
- *Capacity B (RING)-GND*, mkF—capacity between B (RING) wire and ground GND, μF.



**Do not launch the test for multiple ports simultaneously. Port test cannot be interrupted!**

Test results description:

- *OK*—line test has been completed successfully.
- *TEST FAILURE*—invalid operand values were calculated during measurement. For example, division by zero has occurred. This error may appear in line resistance and capacity measurements upon the expiry of capacity measurement timeout.
- *STATE FAILURE*—occurs when the set detects leakage current, and during test, when the current line wire mismatches the required state.
- *RESISTANCE NOT MEASURED*—means that during the line resistance measurement one of the values was lower than the minimum allowed value (100Ω). As a rule, this error may be caused by a wire or ground short circuit.
- *CAPACITANCE NOT MEASURED*—means that during the line resistance measurement one of the values was lower than the minimum allowed value for line capacitance measurement (1800Ω). As a rule, this error may be caused by a phone offhook or a wire or ground short circuit.
- *EXTERNAL VOLTAGE FAILURE*—external voltage measured in line wires falls outside of allowable limits (-5V - +5V).



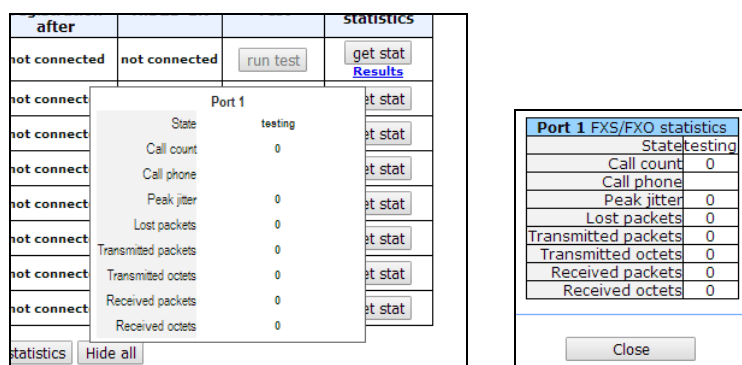
- TEST ERROR—test is interrupted by a processor command.

Click '*Hide test result*' button to remove test result information.

When you click '*Hide all*' button, all results for conducted tests of all types will be deleted.

### Performed Call Statistics

'**Get stat**' button located against each port allows to get the statistics on performed calls for the specific port. To see the statistics, hover the mouse cursor over the '*result*' link located against the respective port, or open the test results window by clicking the link:



Description of 'Port FXS statistics' informational window:

- *State*—current port status:
  - offhook*—phone is offhook.
  - onhook*—phone is onhook.
  - FXO offhook*—FXO port is busy.
  - FXO onhook*—FXO port is free.
  - dial*—dialling number.
  - ringback*—send 'ringback' tone.
  - ringing*—send 'ringing' tone.
  - talking*—call in progress.
  - conference*—3-way conference.
  - busy*—sending 'busy' tone.
  - hold*—port is on hold.
  - testing*—port is in testing mode.
- *Call count*—number of outgoing calls from the gateway startup.
- *Call phone*—last dialled number.
- *Peak jitter*—maximum jitter.
- *Lost packets*—quantity of lost packets.
- *Transmitted packets*—quantity of transferred voice packets.
- *Transmitted octets*—quantity of bytes in transferred voice packets.
- *Received packets*—quantity of received voice packets.

- *Received octets*—quantity of bytes in received voice packets.

'Hide test results', 'Hide blocking info', 'Hide FXS/FXO statistics', 'Hide all' buttons allow to hide line test data, blocking data, FXS/FXO statistics, and all listed data respectively.

When you click '*Hide FXS/FXO statistics*' button, generated statistics on performed calls on this port will be deleted.

When you click '*Hide all*' button, all results for conducted tests of all types will be deleted.

### 5.1.4.2. Board Parameter Status Monitoring (Status)

In '*Status*' submenu, you can monitor physical parameters of the board and SFP modules supporting DDM (digital diagnostics monitoring) function.




The screenshot shows the 'Status' monitoring page in the ELTEX WEB configurator. The page is titled 'TAU-32M.IP\* WEB configurator' and includes navigation tabs for 'Network settings', 'PBX', 'Switch', 'Monitoring', 'System info', and 'Service'. The 'Monitoring' tab is active, and the 'Status' sub-tab is selected. The main content area displays a table of hardware parameters and resources.

| Hardware:       |                 |                     |                  |                 |
|-----------------|-----------------|---------------------|------------------|-----------------|
| Parameter       | Value           |                     |                  |                 |
| Power, V        | Vinput<br>10.98 |                     |                  |                 |
| SM              | 0               | 1                   | 2                | 3               |
| Type            | 8FXS            | 8FXS                | 8FXS             | 8FXO            |
| Temperature, °C | 49              | 48                  | 47               | 39              |
| Fan state       | Fan 1           |                     | Fan 2            |                 |
|                 |                 |                     |                  |                 |
| SFP-0 Status    | Installed       |                     | LOS              |                 |
| Laser Fault     | No              |                     | Yes              |                 |
| Temperature, °C | Power, V        | Tx bias current, mA | Output power, mW | Input power, mW |
| N/A             | N/A             | N/A                 | N/A              | N/A             |
| SFP-1 Status    | Installed       |                     | LOS              |                 |
| Laser Fault     | No              |                     | Yes              |                 |
| Temperature, °C | Power, V        | Tx bias current, mA | Output power, mW | Input power, mW |
| N/A             | N/A             | N/A                 | N/A              | N/A             |
| Resources:      |                 |                     |                  |                 |
| CPU usage       | 6.0%            |                     |                  |                 |
| Disk space      | Size, kB        |                     | Available, kB    |                 |
|                 | 16384           |                     | 4708 (29%)       |                 |
| Memory          | Total, kB       |                     | Free, kB         |                 |
|                 | 44680           |                     | 15592            |                 |

Table '*Hardware*'—*platform sensor parameters*:

*Parameter*—controlled parameters and '*Value*'—controlled parameters' values:

- *Power (Vinput)*, V—board power supply voltage, V.
- *Type*—submodule type:
  - 8FXS—8 ports for connection of the subscriber to the analogue phone line.
  - 8 FXO—8 ports for the analogue phone line connection.
  - 4FXS, 4 FXO—4 for connection of the subscriber to the analogue phone line and 4 ports

- for the analogue phone line connection.
  - NONE—no module installed.
  - UNDEFINED—unable to identify the module type.
- *Temperature, °C*—temperature measured by sensors (each submodule has its own temperature sensor).
- *Fan state*—state of the fan:
  - —fan is on
  - —fan is off
  -  image blinks periodically—faulty fan



**Fans will turn on automatically when the temperature exceeds 55°C, and turned off when the temperature falls below 45°C.**

- *SFP-0 Status, SFP-1 Status*—status of SFP0/SFP1 optical module (works only for modules with DDM support):
  - *Installed*—indication of module installation ('Yes'—module is installed, 'No'—module is not installed).
  - *LOS*—indication of signal loss ('No'—no loss).
  - *Temperature, °C*—optical module temperature.
  - *Power, V*—optical module power supply voltage, V.
  - *Tx bias current, mA*—transmission bias current, mA.
  - *Output power, mW*—output power, mW.
  - *Input power, mW*—input power, mW.
- *Resources*—monitoring of system resources:
  - *CPU usage*—percentage of CPU utilization.
  - *Disk space*—information on disk space:
    - *Size*—disk space in kbytes.
    - *Available*—amount of free disk space in kbytes.
  - *Memory*—amount of RAM:
    - *Total*—total amount of RAM in kbytes.
    - *Free*—free amount of RAM in kbytes.

Click '*Advanced info*' button to open the window with advanced information on RAM utilization.

Permitted parameter values:

- Board power supply voltage should fall within the limits:  $8V < V_{input} < 16V$ .
- Temperature on a sensor should not exceed 90°C.

Fault indication:

- When the sensor malfunction occurs, the 'temperature detector failure' value will blink red in its window.
- Value falling outside of allowable limits will blink red.

- When the fan is out of order, a crossed out circle will blink.

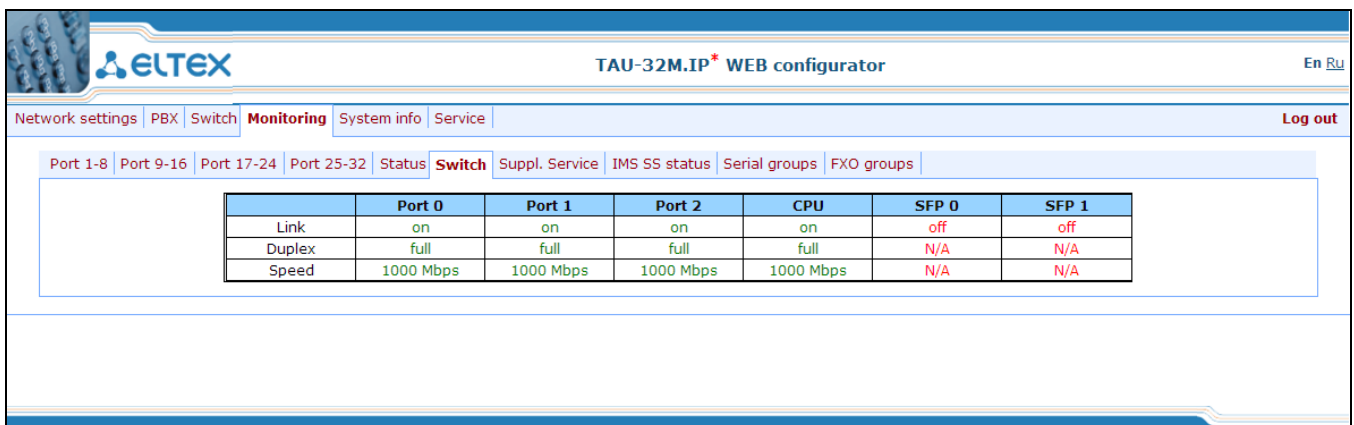
Description of Resources informational window (system resource monitoring):

- *CPU usage*—percentage of CPU utilization.
- *Disk space*—information on disk space.
- *Size*—disk space in kbytes.
- *Available*—amount of free disk space in kbytes.
- *Memory*—RAM.
- *Total*—total amount of RAM in kbytes.
- *Free*—free amount of RAM in kbytes.

### 5.1.4.3. Switch Port Status Monitoring (Switch)

In 'Switch' menu, you may view status of integrated Ethernet switch ports.

The switch is equipped with 3 x Gigabit Ethernet electrical ports (Port 0, Port 1, Port 2), 2 x optical ports (SFP 0, SFP 1), designed for connection to data networks and additional Ethernet devices, and 1 x internal CPU port for connection to TAU-32M.IP HOST processor.



|        | Port 0    | Port 1    | Port 2    | CPU       | SFP 0 | SFP 1 |
|--------|-----------|-----------|-----------|-----------|-------|-------|
| Link   | on        | on        | on        | on        | off   | off   |
| Duplex | full      | full      | full      | full      | N/A   | N/A   |
| Speed  | 1000 Mbps | 1000 Mbps | 1000 Mbps | 1000 Mbps | N/A   | N/A   |

Description of informational window:

- *Link*—port state:
  - off—port is inactive (no connection).
  - on—port is active (connection established).
- *Duplex*—transceiver operation mode:
  - N/A—value is not available, as the link is inactive.
  - Full—full duplex.
  - half—half-duplex.
- *Speed*—data transfer rate for a port:

- N/A—value is not available, as the link is inactive.
- 10 Mb, 100 Mb, 1000 Mb.

#### 5.1.4.4. Supplementary Service Status Monitoring (Suppl.Service)

In 'Suppl. Service' submenu, you can view the current status of supplementary services for subscriber ports of the device.

| Port    | Call transfer |          | Call forward unconditional |          | Call forward on busy |          | Call forward on no answer |          | Call forward on out of service |          | Call waiting |          | Do not disturb |          |
|---------|---------------|----------|----------------------------|----------|----------------------|----------|---------------------------|----------|--------------------------------|----------|--------------|----------|----------------|----------|
|         | Enable        | Status   | Enable                     | Status   | Enable               | Status   | Enable                    | Status   | Enable                         | Status   | Enable       | Status   | Enable         | Status   |
| Port 1: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |
| Port 2: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |
| Port 3: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |
| Port 4: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |
| Port 5: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |
| Port 6: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |
| Port 7: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |
| Port 8: | disable       | attended | disable                    | inactive | disable              | inactive | disable                   | inactive | disable                        | inactive | disable      | inactive | disable        | inactive |

Description of informational window:

- *Port*—subscriber port number.

Services:

- *Call transfer*—'Call transfer' service
- *Call forward unconditional*—'Call forward unconditional' service.
- *Call forward on busy*—'Forward on busy' service.
- *Call forward on no answer*—'Forward on no answer' service.
- *Call forward on out of service*—'Forward on out of service' service.
- *Call waiting*—'Call waiting' service.
- *Do not disturb*—'Do not disturb' service.
- *Enable*—service state ('enable'—enabled, 'disable'—disabled).
- *Status*—service status.

There are three status types for 'Call transfer' service:

- *Attended*—'Call transfer' service is enabled with the wait for response of the subscriber, the call is being forwarded to.

- *Unattended*—'Call transfer' service is enabled, without the wait for response of the subscriber, the call is being forwarded to.
- *Off*—'Call transfer' service is disabled.

*Status for other services:*

- *Active*—active.
- *Inactive*—inactive.

For 'Call forward' service, define the number configured for the call forwarding in the status field.

Use 'Refresh' button to refresh table data.

### 5.1.4.5. IMS SS Status Monitoring

In 'IMS SS status' menu, you may view the current state of services managed by the Softswitch with IMS support.

The screenshot shows the 'IMS SS status' monitoring page in the ELTEX TAU-32M.IP WEB configurator. The interface includes a navigation menu with options like 'Network settings', 'PBX', 'Switch', 'Monitoring', 'System info', and 'Service'. The 'Monitoring' section is active, and the 'IMS SS status' sub-menu is selected. Below the navigation, there are tabs for different port ranges: 'Port 1-8', 'Port 9-16', 'Port 17-24', and 'Port 25-32'. The 'Port 1-8' tab is currently selected, displaying a table with the following data:

| Port    | Call hold | Call transfer | Three-party conference | Call waiting | Hotline | Hot timeout | Hot number |
|---------|-----------|---------------|------------------------|--------------|---------|-------------|------------|
| Port 1: | off       | off           | off                    | off          | off     | off         | off        |
| Port 2: | off       | off           | off                    | off          | off     | off         | off        |
| Port 3: | off       | off           | off                    | off          | off     | off         | off        |
| Port 4: | off       | off           | off                    | off          | off     | off         | off        |
| Port 5: | off       | off           | off                    | off          | off     | off         | off        |
| Port 6: | off       | off           | off                    | off          | off     | off         | off        |
| Port 7: | off       | off           | off                    | off          | off     | off         | off        |
| Port 8: | off       | off           | off                    | off          | off     | off         | off        |

A 'Refresh' button is located below the table.

- *Port*—subscriber port number.

Services:

- *Call hold*—'Call hold' service status.
- *Call transfer*—'Call transfer' service status.
- *Three-party conference*—'3-way Conference' service status.
- *Call waiting*—'Call waiting' service status.
- *Hotline*—'Hotline/warmline' service status.
- *Hot timeout*—delay timeout in seconds for the start of the automatic dialling when the 'Hotline/warmline' service is enabled.

- *Hot number*—number that will receive the call when 'Hotline/warmline' is enabled.

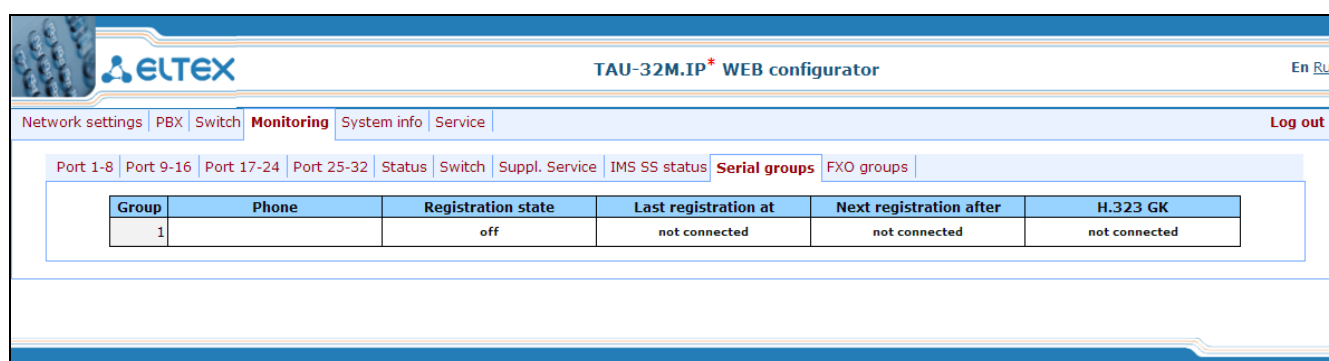
Service statuses:

- *Off*—IMS management is disabled.
- *Disable*—service is disabled.
- *Enable*—service is enabled.

Use '*Refresh*' button to refresh table data.

#### 5.1.4.6. Serial Group Registration Status Monitoring

In '*Serial groups*' menu, you may view the current state of serial group registration.



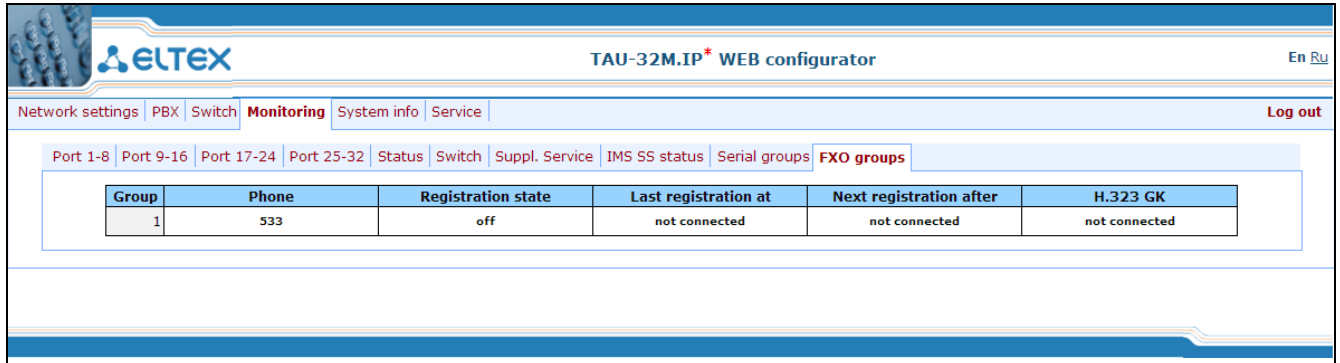
| Group | Phone | Registration state | Last registration at | Next registration after | H.323 GK      |
|-------|-------|--------------------|----------------------|-------------------------|---------------|
| 1     |       | off                | not connected        | not connected           | not connected |

Description of informational window:

- *Group*—group sequential number.
- *Phone*—call group subscriber number.
- *Registration state*—SIP server registration status:
  - *Off*—registration disabled.
  - *Ok*—successful registration.
  - *Failed*—registration failed.
- *Last registration at*—last known successful registration on SIP server.
- *Next Registration after*—remaining time for SIP server registration renewal.
- *H.323 GK*—H.323 gatekeeper registration time.

### 5.1.4.7. FXO Group Registration Status Monitoring

In 'FXO groups' menu, you may view the current state of FXO group registration.

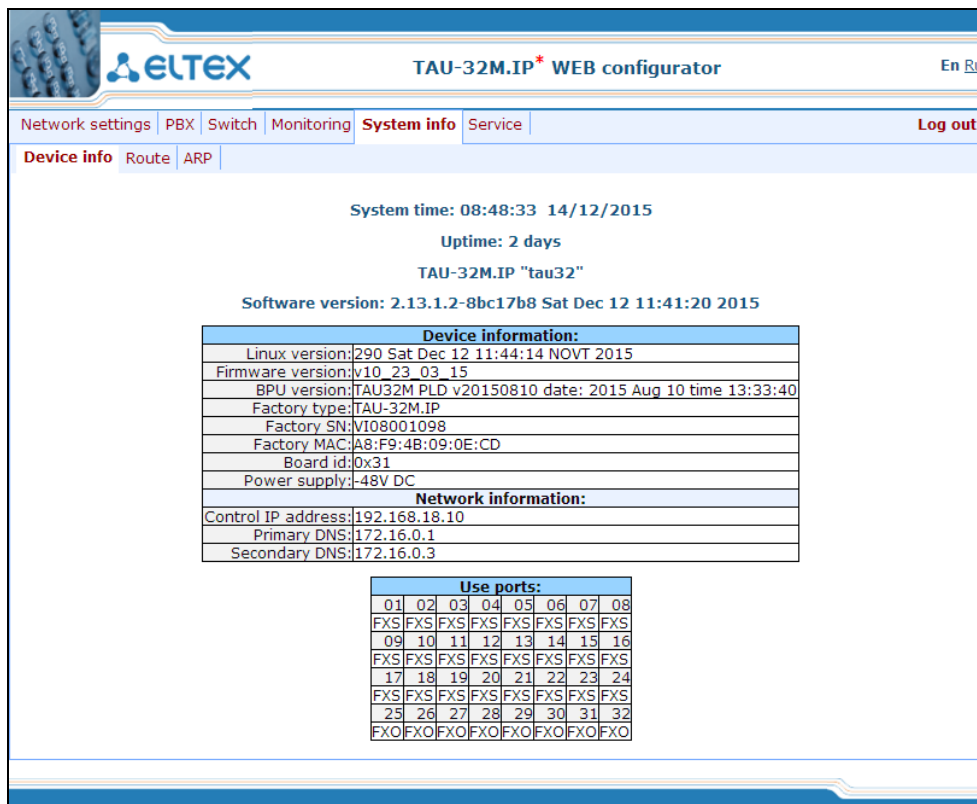


Description of parameters of this informational window is equivalent to Section 5.1.4.6.

### 5.1.5. System Information (System info)

#### 5.1.5.1. Service Status Monitoring (Device info)

In 'System info' menu, you can view the system information.



- *System time*—device system date and time in the following format: hours:minutes:seconds day/month/year.
- *Uptime*—time of the uninterrupted gateway operation.



- *TAU-32M.IP*—firmware version.
- *Software Version*—device firmware version.

### Device information

- *Linux version*—Linux OS version.
- *Firmware version*—media processor firmware version.
- *BPU version*—hardware version.
- *Factory type, SN, MAC*—factory settings.
- *User MAC*—MAC address, defined by user. In this case, factory MAC address will be ignored. You can specify MAC address from the CLI console only.
- *Board id*—hardware platform version.
- *Power supply*—type of power supply installed (AC or DC).

### Network information

- *Control IP-address*—IP address of the device used for management purposes.
- *Primary DNS*—primary DNS server address.
- *Secondary DNS*—secondary DNS server address.
- *Use ports*—subscriber unit type for each port (FXS—port for connection of the subscriber to the analogue phone line, FXO—ports for the analogue phone line connection, NONE—module, that the port belongs to, is not installed, UNDEFINED—unable to identify the type of module, that the port belongs to).

#### 5.1.5.2. Route

In '*Route*' menu, you can view the current routing table.

The screenshot shows the 'Route' menu in the TAU-32M.IP WEB configurator. The 'Kernel IP routing table' is displayed with the following data:

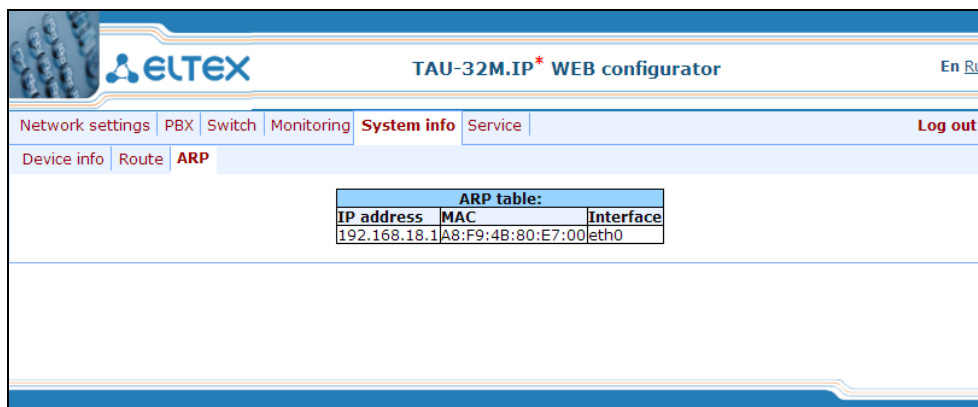
| Destination  | Gateway      | Genmask       | Flags | Metric | Ref | Use | Interface |
|--------------|--------------|---------------|-------|--------|-----|-----|-----------|
| 192.168.18.0 | 0.0.0.0      | 255.255.255.0 | U     | 0      | 0   | 0   | eth0      |
| 172.20.0.0   | 192.168.18.1 | 255.255.255.0 | UG    | 0      | 0   | 0   | eth0      |
| 10.100.101.0 | 192.168.18.1 | 255.255.255.0 | UG    | 0      | 0   | 0   | eth0      |
| 172.16.0.0   | 192.168.18.1 | 255.255.252.0 | UG    | 0      | 0   | 0   | eth0      |
| 192.168.0.0  | 192.168.18.1 | 255.255.0.0   | UG    | 0      | 0   | 0   | eth0      |

- *Destination*—destination network of host address.
- *Gateway*—gateway representing a router network address that should receive the packet transferred to the defined destination address.
- *Genmask*—destination network mask.

- *Flags*—describes route properties. For the specific route, may be defined the following flags:
- *U*—route is active.
- *G*—route is directed to the gateway.
- *H*—route is directed to the host, i.e. complete host address is defined as a destination. If this flag is missing, destination is a network address.
- *D*—route was created by forwarding.
- *M*—route was modified by forwarding.
- *Metric*—numeric index that defines the route preferability. The less the number, the higher the preferability of the route.
- *Ref*—number of references to the route for connection creation.
- *Use*—number of route discoveries performed by IP protocol.
- *Iface*—device network interface used for access through this route.

### 5.1.5.3. ARP

In *ARP* menu, you can view the device ARP table.



The screenshot shows the 'ARP table' section of the web configurator. It contains a table with the following data:

| ARP table:   |                   |           |
|--------------|-------------------|-----------|
| IP address   | MAC               | Interface |
| 192.168.18.1 | A8:F9:4B:80:E7:00 | eth0      |

- *IP address*—IP address of destination host.
- *MAC*—MAC address of destination host.
- *Interface*—network interface, that the destination host is available through.

## 5.1.6. Service

In 'Service' menu, you may update the firmware, work with configuration files and other service features.

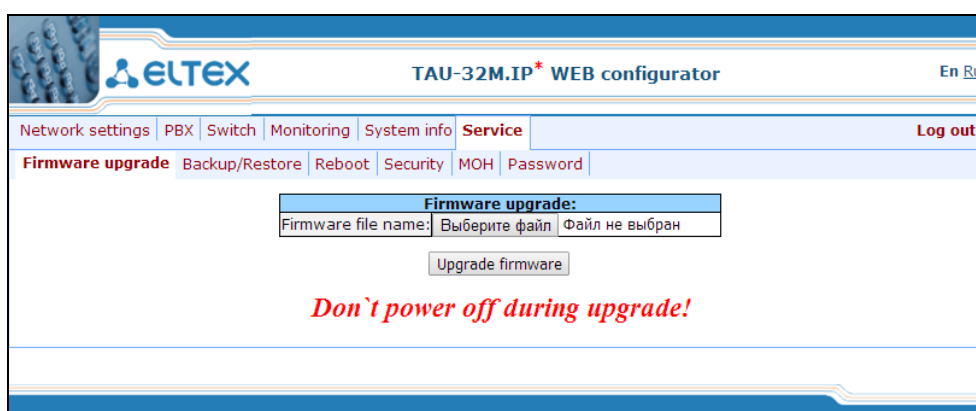
### 5.1.6.1. Firmware upgrade

In 'Firmware upgrade' submenu, you may update the firmware of the subscriber units.



**If the firmware version has been issued prior to September 2010, it is forbidden to update the file system and Linux kernel from a single archive!**

**Firmware versions prior to 1.11.x should be updated according to the instructions listed in the beginning of this operation manual.**



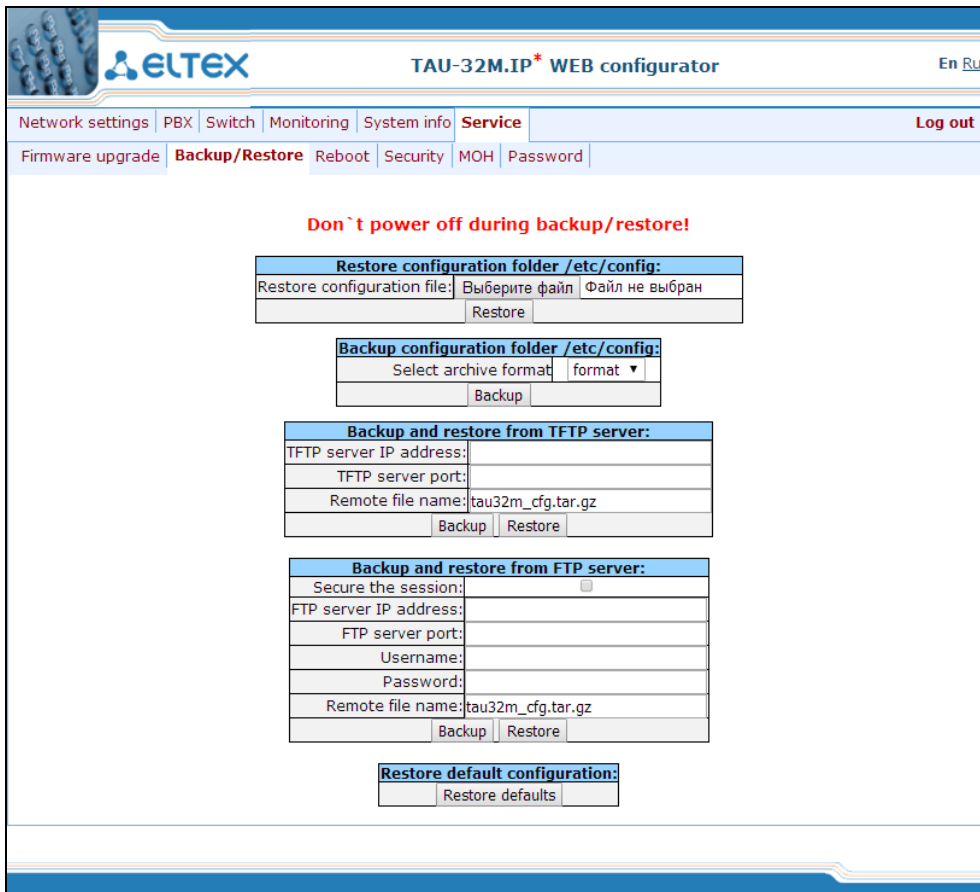
In 'Firmware upgrade' section, you can update the TAU-32M.IP firmware (firmware file is an image named *firmware.img*).

In the opened window, specify the path to the firmware file by clicking 'Select File' button and click 'Upgrade firmware' button.

### 5.1.6.2. Download/Upload Configuration (Backup/Restore)

In 'Backup/Restore' submenu, you may download/upload configuration files. We have implemented 3 ways to download/upload configuration files:

- 1 Using Web configurator.
- 2 Using TFTP server.
- 3 Using FTP server.



1. *Download/upload configuration files using web configurator*

Restore configuration folder /etc/config section description:

- *Restore configuration file*—configuration file that should be uploaded to device from PC.

To upload the configuration file: select the configuration file in '*Restore configuration file*' field using '*Select file*' button (file name should be as follows: tau32M\_cfg, with tar, or tar.gz extension) and click '*Restore*'.

Backup configuration folder /etc/config section description:

- *Backup configuration folder /etc/config*—download configuration to PC (configuration files will be saved on a PC in archive tau32M\_cfg.tar, or tau32M\_cfg.tar.gz depending on the selected format).

To download configuration files or other folders to a PC, click '*Backup*' button.

2. *Download/upload files using TFTP server*

*Backup/Restore from TFTP server:*

- *TFTP Server IP Address*—TFTP server IP address.
- *TFTP Server Port*—TFTP server port number.
- *Remote File Name*—uploaded or downloaded file name.

Click 'Restore' button, to upload configuration files from TFTP server to device. Click 'Backup' button to download files from device to TFTP server.

### 3. Download/upload files using FTP server

Backup/Restore from FTP server:

- *FTP Server IP Address*—FTP server IP address.
- *FTP Server Port*—FTP server port number.
- *User Name*—username.
- *Password*—password.
- *Remote File Name*—uploaded or downloaded file name.

Click 'Restore' button, to upload configuration files to device. Click 'Backup' button, to download configuration files from device.

Click 'Restore default' button to reset the configuration to factory defaults.

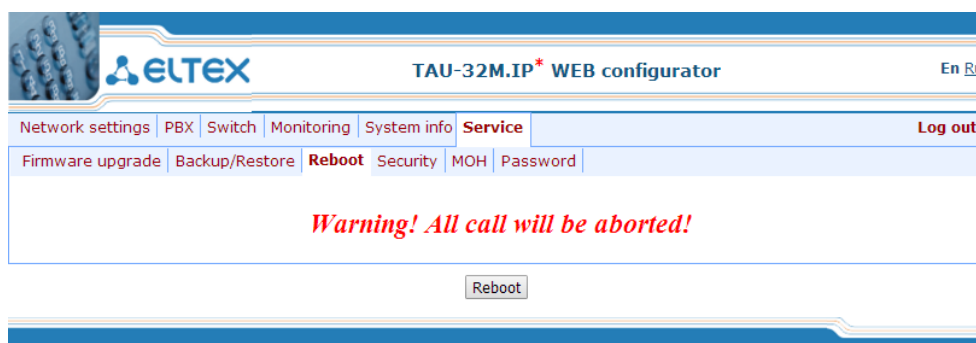


**When configuration resets to factory defaults, the device will be restarted automatically.**

After you upload a new configuration using any of these methods, restart the device by clicking 'Reboot' button in the 'Reboot' submenu.

#### 5.1.6.3. Reboot

In 'Reboot' submenu, you may reboot the device.



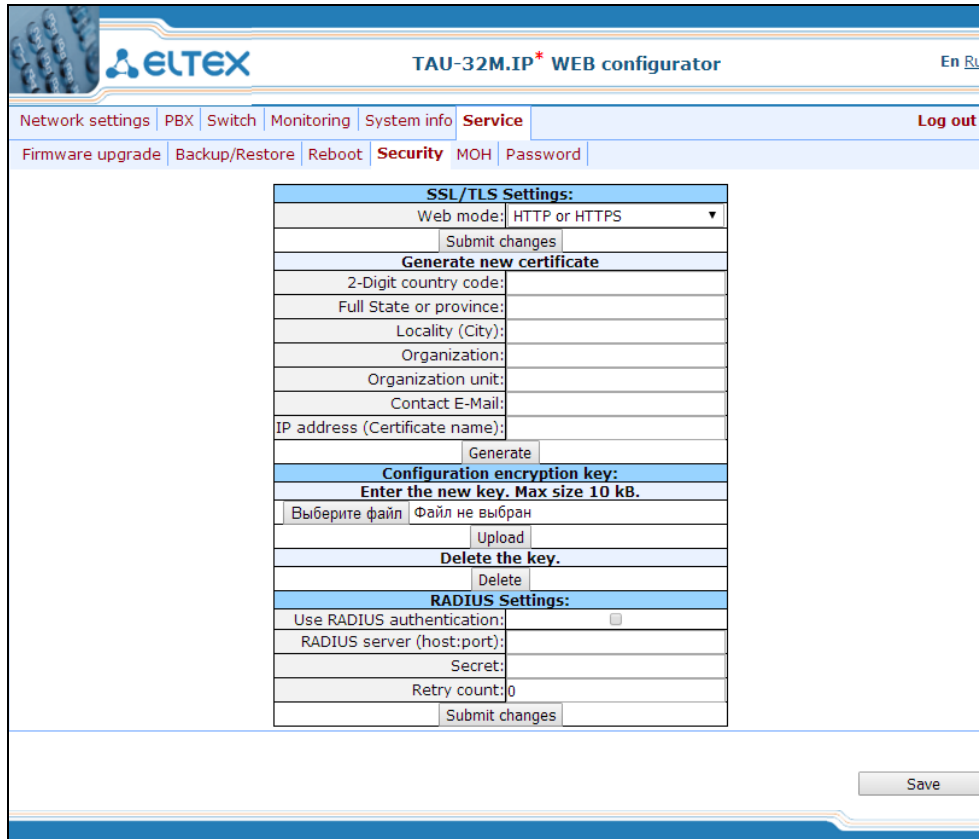
To reboot the device, click 'Reboot' button.



**Before performing a reboot, make sure that all changes are saved, otherwise they will be lost!**

#### 5.1.6.4. Encryption Features (Security)

In 'Security' submenu, you may obtain a self-signed certificate, which allows you to use an encrypted connection to the gateway via HTTP protocol and configuration file upload/download via FTPS protocol.



The screenshot shows the 'Security' configuration page in the ELTEX TAU-32M.IP WEB configurator. The page has a navigation menu at the top with options like 'Network settings', 'PBX', 'Switch', 'Monitoring', 'System info', 'Service', and 'Log out'. The 'Security' menu item is highlighted. Below the navigation, there are several configuration sections:

- SSL/TLS Settings:** Includes a dropdown for 'Web mode' (set to 'HTTP or HTTPS') and a 'Submit changes' button.
- Generate new certificate:** A form with fields for '2-Digit country code', 'Full State or province', 'Locality (City)', 'Organization', 'Organization unit', 'Contact E-Mail', and 'IP address (Certificate name)'. It includes a 'Generate' button.
- Configuration encryption key:** A section titled 'Enter the new key. Max size 10 kB.' with a file selection button ('Выберите файл'), a status indicator ('Файл не выбран'), and an 'Upload' button.
- Delete the key:** A section with a 'Delete' button.
- RADIUS Settings:** Includes a checkbox for 'Use RADIUS authentication', a text field for 'RADIUS server (host:port)', a text field for 'Secret', and a text field for 'Retry count' (set to 0). It includes a 'Submit changes' button.

A 'Save' button is located at the bottom right of the page.

- *WEB mode*—WEB configurator connection mode:
  - *HTTP or HTTPS*—unencrypted connection—via HTTP—as well as encrypted connection—via HTTPS—is enabled. At that, connection via HTTPS is possible only when generated certificate is present.
  - *HTTPS only*—only encrypted connection via HTTPS is enabled. Connection via HTTPS is possible only when generated certificate is present.

When you change connection mode in WEB configurator, click 'Submit Changes' button.

Generate new certificate:

- 2-Digit country code—2-digit code.
- Full State or province—location (region).
- Locality (City)—location (city).
- Organization—organization name.
- Organization unit—organization unit.
- Contact E-Mail—e-mail address.

— IP address (Certificate name)— gateway IP address.

When you enter all fields, click '*Generate*' button to generate self-signed certificate.

Configuration encryption key:

The key is used for configuration file encryption/decryption during its upload to/download from the device. When key is not defined, encryption will not work. Encryption uses AES-256 algorithm.



**For configuration file decryption on a PC, you may use *openssl* utility.**

**Usage: *openssl enc -aes-256-cbc -d -pass pass:'Password' -in 'encrypted file' -out 'decrypted file'***

To upload a new encryption key '*Enter the new key*', specify path to file to be uploaded to the device using '*Select file*' button and click '*Upload*'.

RADIUS Settings:

- *Use RADIUS authentication*—use RADIUS server for authentication of users administering the device via WEB, telnet, SSH.
- *RADIUS server (host:port)*—RADIUS server IP address.
- *Password (Secret)*—password used by client to access the RADIUS server.
- *Retry count*—number of retries during the access to RADIUS server. If the server authorization has failed, you will be able to manage the device via the local COM port only.



**On RADIUS server, you may configure passwords for any of the system users: admin, operator, supervisor, viewer. For detailed information on user privileges, see Section 5.1.6.6.**

To delete or change previously uploaded key, specify the path to the encryption key using 'Browse' button and then click '*Get access*'.

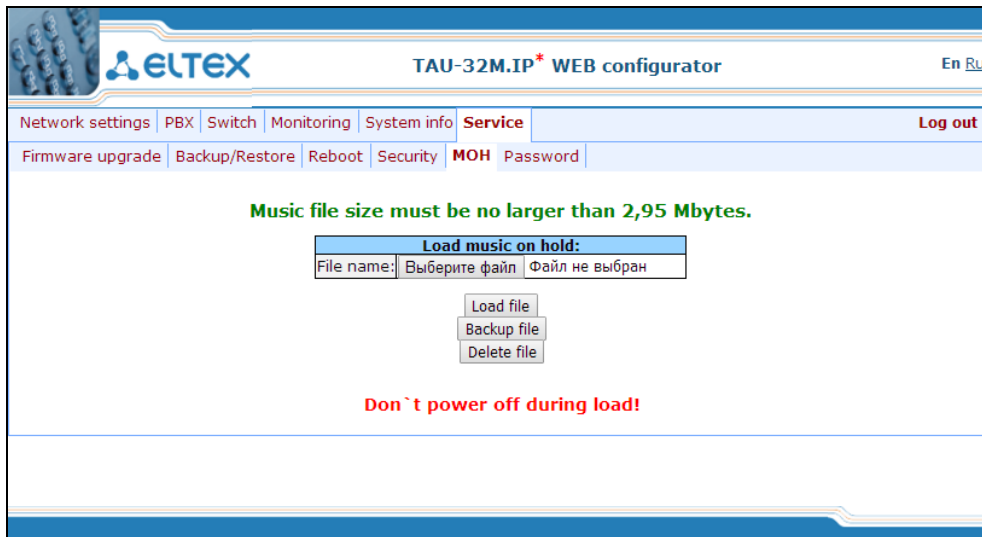
To save changes, click '*Save*' button.

#### 5.1.6.5. 'Music on Hold' service configuration (MOH)

In '*MOH*' submenu, you may upload/download audio file to/from the device in order to enable '*Music on Hold*' service. To activate '*Music on Hold*' service, select '*Play music on hold*' checkbox in subscriber port settings.



**Correct operation of the service is guaranteed only for connections that use G.711A and G.711U codecs!**



- *Select file*—specify a file to upload to the device.

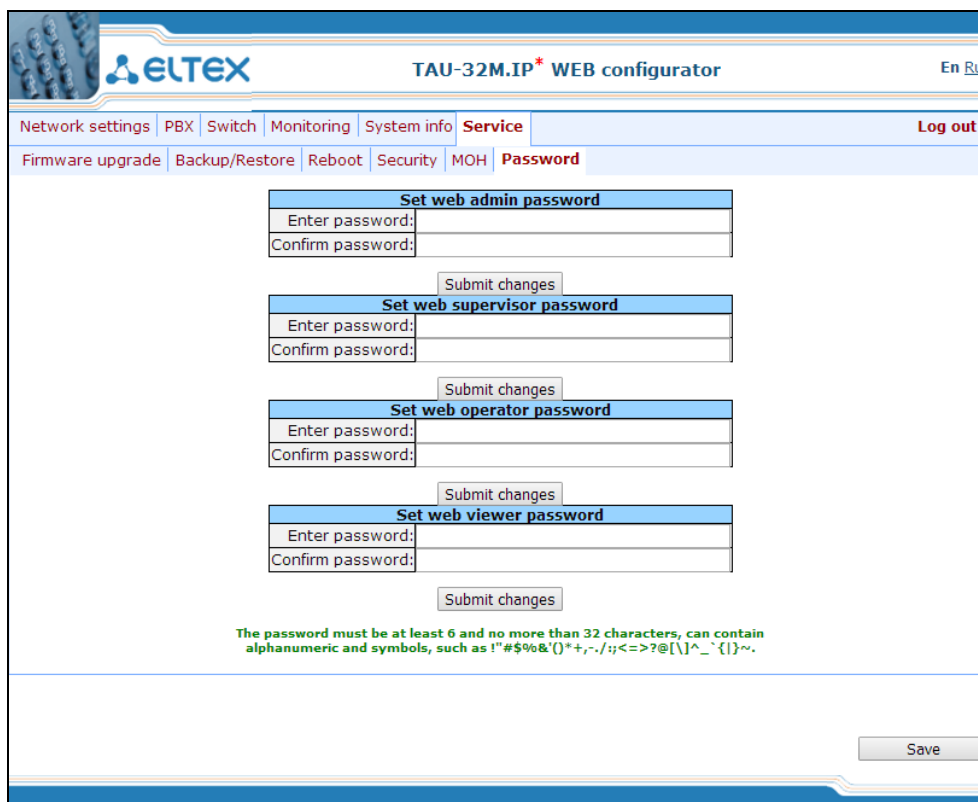
#### Audio file requirements:

- Format: CCITT A-law.
  - Attributes: 8000 kHz, 8 Bit, Mono.
  - File extension: wav.
  - To recode the file to the necessary format, you may use ffmpeg or any other conversion application. Example use of ffmpeg:
  - `ffmpeg -fs <X>M -i <inputfilename> -ar 8000 -acodec pcm_alaw -ac 1 <outputfilename>`.
  - where 'X'—file size limit, 'inputfilename'—input file name, 'outputfilename'—output file name.
- *Load file*—button that allows you to upload the file to the device.
  - *Backup file*—button that allows you to download the file to PC.
  - *Delete file*—button that allows you to delete the file from the device.



### 5.1.6.6. Changing Access Passwords using Web Configurator (Password)

In 'Passwords' submenu, you may work with passwords for device access via web interface.



Access passwords operations:

- *Set web admin password*—administrator password for device access via web interface (*admin* user).
- *Set supervisor password*—supervisor password for device access via web interface (*supervisor* user).
- *Set operator password*—operator password for device access via web interface (*operator* user).
- *Set viewer password*—viewer password for device access via web interface (*viewer* user).

User rights:

- *supervisor*—will be able to access all device parameters in read-only mode.
- *admin*—has full access to the device.
- *operator*—will be able to access the device for monitoring, viewing the system information, and also for configuration of protocols, routing settings, subscriber ports and groups.
- *viewer*—will be able to access the device for monitoring and viewing the system information.

To change the password, enter a new password into 'Enter password' field, and enter it again into 'Confirm password' field. To apply password, click 'Submit Changes' button. To save changes, click 'Save' button.

### 5.1.7. Change users

To change a user, click 'Log out' link.



To change the access, enter the corresponding user name (admin, operator, viewer), password (passwords for various access levels are defined by 'admin' user in '**Service/Password**' tab) and click *Log in*' button. To exit configuration program, click '*Cancel*' button.

## 5.2. TAU-32M.IP Configuration via web Interface Operator Access

To configure the device, establish connection in the *web browser*, e.g. Firefox, Internet Explorer, etc. Enter device IP address into address bar of web browser.

After entering IP address the device will request username and password.



**TAU-32M.IP factory default IP address—192.168.1.2, network mask—255.255.255.0**

**Initial startup username: *operator***

**password: specified by admin.**

The following menu will appear on the operator's terminal:

Web configurator supports indication of configuration changes, that is shown in the header bar of configuration interface (TAU-32M.IP WEB configurator.) Table 9 lists indicator states ('\*' character in the header bar of configuration interface).

Table 9—Indicator states \*

| Indicator State    | Description   |
|--------------------|---|
| * character is red | Changes has been made to the configuration, but it has not been saved to flash memory yet |

|                          |  |
|--------------------------|--|
| * character is blue      | Changes has been made to the configuration and applied with 'Apply config' button, but it has not been saved to flash memory yet |
| * character is not shown | No changes has been made to the configuration, or changes has been successfully saved to flash memory                            |



In all tabs, 'Save' button stores configuration into the non-volatile (flash) memory of the device.

Operator will be able to view and edit routing and subscriber port configuration.

Table 10 lists web configurator menu tabs available to the operator. For detailed web configurator description, see Section 5.1 of this document.

Table 10—Description of configuration menu, operator access

| Menu (en)                   | Description  |
|-----------------------------|--|
| <b>PBX</b>                  | VoIP (Voice over IP) configuration   |
| <i>Main</i>                 | Device basic settings  |
| <i>SIP/H323 Profiles</i>    | Configuration of SIP/H323 profiles   |
| <i>SIP Common</i>           | SIP common settings  |
| <i>H323</i>                 | H323 protocol settings (works in profile 1 only)   |
| <i>Profile 1..8</i>         | Profile configuration  |
| <i>SIP Custom</i>           | SIP custom settings for a profile  |
| <i>Codecs</i>               | Codec settings for a profile   |
| <i>Dialplan</i>             | Routing settings for a profile   |
| <i>Alert-Info</i>           | Configuration of a distinctive ring, formed by Alert-Info value                            |
| <i>TCP/IP</i>               | Configuration of network port range for various protocols                                  |
| <i>Ports</i>                | Configuration of Subscriber Ports  |
| <i>Call limits</i>          | Configuration of simultaneous call limits  |
| <i>Suppl. Service Codes</i> | Configuration of Supplementary Service Codes   |
| <i>Serial groups</i>        | Configuration of serial groups   |
| <i>FXO groups</i>           | Configuration of FXO groups  |
| <i>PickUp groups</i>        | Configuration of call pickup group   |
| <i>Distinctive ring</i>     | 'Distinctive ring' service administration  |
| <i>Modifiers</i>            | Configuration of number modifiers  |
| <b>Monitoring</b>           | Device monitoring  |
| <i>Port</i>                 | Device subscriber ports status information   |
| <i>Status</i>               | Gateway hardware platform status information—voltages, temperature sensors, fans, SFP data |
| <i>Switch</i>               | Switch port state monitoring   |

|                       |  |
|-----------------------|--|
| <i>Suppl. Service</i> | Information on the current status of supplementary services on subscriber port               |
| <i>IMS SS status</i>  | State monitoring of services managed by the software switch with IMS function support        |
| <i>Serial groups</i>  | Serial group registration status monitoring  |
| <b>System info</b>    | System information   |
| <b>Service</b>        | Firmware update, configuration file operations, rebooting device, setting/changing passwords |
| <i>Reboot</i>         | Rebooting device   |



**Before performing a reboot, make sure that all changes are saved, otherwise they will be lost!**

### 5.3. Non-privileged user access for device monitoring.

To monitor the device, establish connection in the *web browser* (hypertext document viewer), such as Firefox, Internet Explorer. Enter device IP address into address bar of web browser.



**TAU-32M.IP factory default IP address—192.168.1.2, network mask—255.255.255.0**

After entering IP address, the device will request username and password.



**Username: viewer, password: specified by admin.**

The following menu will appear on the operator's terminal:

The screenshot shows the 'System info' tab of the TAU-32M.IP WEB configurator. The interface includes a navigation menu with 'Monitoring', 'System info', and 'Log out'. Below the menu, the system time is 08:58:11 on 14/12/2015, with an uptime of 2 days. The device is identified as TAU-32M.IP 'tau32' with software version 2.13.1.2-8bc17b8. A detailed 'Device information' table follows, listing Linux, Firmware, BPU, and factory details. Below that is a 'Network information' table showing the control IP address (192.168.18.10) and DNS settings. Finally, a 'Use ports' table lists 32 ports (01-32) with their respective types (FXS or FXO).

| Device information: |  |
|---------------------|--|
| Linux version:      | 290 Sat Dec 12 11:44:14 NOV 2015                     |
| Firmware version:   | v10_23_03_15   |
| BPU version:        | TAU32M PLD v20150810 date: 2015 Aug 10 time 13:33:40 |
| Factory type:       | TAU-32M.IP   |
| Factory SN:         | VI08001098   |
| Factory MAC:        | A8:F9:4B:09:0E:CD                                    |
| Board id:           | 0x31   |
| Power supply:       | -48V DC  |

| Network information: |               |
|----------------------|---------------|
| Control IP address:  | 192.168.18.10 |
| Primary DNS:         | 172.16.0.1    |
| Secondary DNS:       | 172.16.0.3    |

| Use ports: |     |
|------------|-----|
| 01         | FXS |
| 02         | FXS |
| 03         | FXS |
| 04         | FXS |
| 05         | FXS |
| 06         | FXS |
| 07         | FXS |
| 08         | FXS |
| 09         | FXS |
| 10         | FXS |
| 11         | FXS |
| 12         | FXS |
| 13         | FXS |
| 14         | FXS |
| 15         | FXS |
| 16         | FXS |
| 17         | FXS |
| 18         | FXS |
| 19         | FXS |
| 20         | FXS |
| 21         | FXS |
| 22         | FXS |
| 23         | FXS |
| 24         | FXS |
| 25         | FXO |
| 26         | FXO |
| 27         | FXO |
| 28         | FXO |
| 29         | FXO |
| 30         | FXO |
| 31         | FXO |
| 32         | FXO |

Non-privileged users will only be able to view routing and subscriber port configuration.

#### 5.3.1. Monitoring

For detailed tabs description, see **Section 5.1.4** of this document.

#### 5.3.2. System info

For detailed menu description, see **Section 5.1.5** of this document.

## 5.4. SUPERVISOR Access

To login to the device, establish connection in the *web browser* (hypertext document viewer), such as Firefox, Internet Explorer. Enter device IP address into address bar of web browser (factory default address—192.168.1.2, network mask—255.255.255.0).

After entering IP address the device will request username and password. Username: *supervisor*, password: specified by admin.

The following menu will appear on the operator's terminal:

The screenshot shows the ELTEX TAU-32M.IP WEB configurator interface. The top navigation bar includes 'Network settings', 'PBX', 'Switch', 'Monitoring', 'System info', and 'Log out'. The 'System info' section is active, displaying the following information:

- System time: 08:59:14 14/12/2015
- Uptime: 2 days
- TAU-32M.IP "tau32"
- Software version: 2.13.1.2-8bc17b8 Sat Dec 12 11:41:20 2015

Below this, there are two tables:

| Device information: |  |
|---------------------|--|
| Linux version:      | 290 Sat Dec 12 11:44:14 NOV 2015                     |
| Firmware version:   | v10_23_03_15   |
| BPU version:        | TAU32M PLD v20150810 date: 2015 Aug 10 time 13:33:40 |
| Factory type:       | TAU-32M.IP   |
| Factory SN:         | VI08001098   |
| Factory MAC:        | A8:F9:4B:09:0E:CD                                    |
| Board id:           | 0x31   |
| Power supply:       | -48V DC  |

| Network information: |               |
|----------------------|---------------|
| Control IP address:  | 192.168.18.10 |
| Primary DNS:         | 172.16.0.1    |
| Secondary DNS:       | 172.16.0.3    |

| Use ports: |     |     |     |     |     |     |     |
|------------|-----|-----|-----|-----|-----|-----|-----|
| 01         | 02  | 03  | 04  | 05  | 06  | 07  | 08  |
| FXS        | FXS | FXS | FXS | FXS | FXS | FXS | FXS |
| 09         | 10  | 11  | 12  | 13  | 14  | 15  | 16  |
| FXS        | FXS | FXS | FXS | FXS | FXS | FXS | FXS |
| 17         | 18  | 19  | 20  | 21  | 22  | 23  | 24  |
| FXS        | FXS | FXS | FXS | FXS | FXS | FXS | FXS |
| 25         | 26  | 27  | 28  | 29  | 30  | 31  | 32  |
| FXO        | FXO | FXO | FXO | FXO | FXO | FXO | FXO |

Supervisor will be able to access all parameters of the device in read-only mode.

## 6. COMMAND LINE MODE AND TERMINAL MODE OPERATION

### 6.1. Basic Commands

CLI is available when the connection to the device is established via RS-232 (connection parameters: 115200, 8, n, 1, n; username: *admin*, w/o password), or Telnet/SSH.

Table 11—List of available commands

| Command  | Description  |
|--|--|
| <code>config</code>  | Enter the configuration mode   |
| <code>?</code>   | Show the list of available commands  |
| <code>help</code>  | Show help on CLI operation   |
| <code>quit, logout, exit</code>                              | Exit the command line mode   |
| <code>history</code>   | Show the list of previously entered commands   |
| <code>passwd</code>  | Change password for 'admin' user   |
| <code>ps</code>  | Show information on the current processes  |
| <code>reboot</code>  | Reboot gateway   |
| <code>route</code>   | Show/configure the routing table   |
| <code>save</code>  | Save configuration into non-volatile memory  |
| <code>shell</code>   | Go to Linux console  |
| <code>show hwaddr</code>                                     | Show MAC address   |
| <code>show ipaddr</code>                                     | Show IP address  |
| <code>show netmask</code>                                    | Show network mask  |
| <code>system</code>  | Show firmware version  |
| <code>traceroute</code>                                      | Trace the route to host  |
| <code>ping</code>  | Send echo (ping) request   |
| <b>Application operation commands</b>                        |  |
| <code>pbx restart</code>                                     | Command that allows to restart the main application  |
| <code>pbx registration &lt;n&gt;</code>                      | SIP server registration renewal for ports working in a single SIP profile, where <n> is a number of SIP profile  |
| <b>Statistics operation commands</b>                         |  |
| <code>pbx history</code>                                     | View the current call statistics   |
| <code>pbx statistic &lt;n&gt;</code>                         | View the port-specific statistics, where <n> is a port number  |
| <b>Automatic configuration commands</b>                      |  |
| <code>update cfg &lt;A.B.C.D&gt;<br/>&lt;filename&gt;</code> | Configuration update:<br>A.B.C.D. – IP address of a computer, that runs TFTP server, pointing to the folder with the file.<br>filename—configuration file name |



|  |  |
|--|--|
| <pre>update img &lt;A.B.C.D&gt; &lt;filename&gt;</pre>   | <p>Software update:</p> <p>A.B.C.D. – IP address of a computer, that runs TFTP server, pointing to the folder with the file.</p> <p>filename—firmware file name</p>  |
| <p><b>Configuration mode commands (use 'config' command to enter this mode). You may configure the device name in the configuration mode</b></p> |  |
| ?  | Show the list of available commands  |
| help   | Show help on CLI operation   |
| quit   | Exit the command line mode   |
| exit   | Exit the configuration mode  |
| history  | Show the list of previously entered commands   |
| <pre>mac set &lt;AA:BB:CC:DD:EE:FF&gt;</pre>   | Set the user MAC address   |
| mac clear  | Remove the user MAC address  |
| mac get  | Show the user MAC address  |
| reset <static   dhcp>  | Reset configuration to factory defaults (and set static or dynamic method to obtain network settings)  |
| save   | Save configuration into non-volatile memory  |
| <pre>set autoupdate &lt;par1&gt; &lt;par2&gt;</pre>  | <p>Automatic update configuration</p> <p>par1:</p> <ul style="list-style-type: none"> <li>cfg—configuration file name</li> <li>fw—firmware versions' file name</li> <li>interval_cfg—configuration update period</li> <li>interval_fw—firmware update period</li> <li>src—autoupdate mode</li> <li>tftp—autoupdate server address</li> <li>usage—autoupdate utilization</li> </ul> <p>par2: par1 parameter value</p> |
| <pre>set &lt;dhcp dhcp_gateway&gt; &lt;on off&gt;</pre>  | <p>Configure DHCP parameters for the main network:</p> <ul style="list-style-type: none"> <li>dhcp—use DHCP</li> <li>dhcp_gateway—use the gateway, received via DHCP</li> </ul>  |
| <pre>set ntp interval set ntp &lt;interval  ipaddr  timecorrect  usage&gt;</pre>   | <p>Configure NTP server operation parameters:</p> <ul style="list-style-type: none"> <li>interval—time resynchronization period</li> <li>ipaddr—NTP server IP address</li> <li>timecorrect—time zone</li> <li>usage—NTP protocol usage</li> </ul>  |
| <pre>set &lt;broadcast gateway  ipaddr  netmask  dns&gt; &lt;A.B.C.D&gt;</pre>   | Configure broadcast address, gateway, IP address, mask, DNS server address   |

|   |   |
|---|---|
| <pre>set pppoe &lt;par1&gt; &lt;par2&gt;</pre>  | <p>PPPoE settings</p> <p>par1:</p> <ul style="list-style-type: none"> <li>password—password</li> <li>usage—PPPoE usage</li> <li>user—user name</li> <li>vid—identifier of VLAN network, used by PPPoE operations</li> <li>vlan—VLAN subnet utilization</li> </ul> <p>par2: par1 parameter value</p>   |
| <pre>set &lt;control rtp signaling&gt; &lt;no_vlan vlan1 vlan2  vlan3 pppoe&gt;</pre> | <p>Define the interface for control, voice traffic (RTP), and signalling</p> <ul style="list-style-type: none"> <li>no_vlan—main network without VLAN</li> <li>vlan1,vlan2,vlan3—corresponding VLAN network</li> <li>pppoe—PPPoE interface</li> </ul>   |
| <pre>set &lt;snmp ssh telnet&gt; &lt;on off&gt;</pre>                                 | <p>Configuration of SNMP, SSH, TELNET protocols for gateway management</p>  |
| <pre>set &lt;vlan1 vlan2  vlan3&gt; &lt;par1&gt; &lt;par2&gt;</pre>                   | <p>VLAN subnet configuration</p> <ul style="list-style-type: none"> <li>vlan1,vlan2,vlan3—VLAN subnet number</li> </ul> <p>par1:</p> <ul style="list-style-type: none"> <li>broadcast—broadcast address</li> <li>cos—802.1p priority</li> <li>dhcp—use DHCP</li> <li>dhcp_gateway—use the gateway, received via DHCP</li> <li>id—VLAN network identifier</li> <li>ipaddr—IP address</li> <li>netmask—network mask</li> <li>usage—VLAN subnet usage</li> </ul> <p>par2: par1 parameter value</p> |
| <pre>show</pre>   | <p>Show network and autoconfiguration parameters</p>  |
| <pre>version</pre>  | <p>View configuration file version</p>  |

## 6.2. Call Statistics

### 6.2.1. Command Line Mode

CLI is available when the connection to the device is established via RS-232 (connection parameters: 115200, 8, n, 1, n; username: admin, w/o password), or Telnet/SSH. Use CLI command to enter this mode.

To view the current call statistics, use `pbx history` command.

Device RAM may store up to 2000 performed calls records. When the number of records exceeds 2000, the oldest records will be deleted, and the new ones will be added at the end of the file.

Table 12—Call statistics record format.

| Record          | Description                                  |
|-----------------|--|
| No              | Sequence number of the record                |
| Local           | TAU-32M.IP subscriber number                 |
| Remote          | Remote subscriber number                     |
| Remote host     | Remote host IP address                       |
| Start call time | Call received/performed time                 |
| Start talk time | Call start time                              |
| Duration        | Duration of call (seconds)                   |
| State           | Transient state, or reason for call clearing |
| Type            | Call type (outgoing, incoming)               |

Table 13—Transient states and reasons for call clearing output into statistics

| Transient states          | Description  |
|---------------------------|--|
| seize                     | Incoming or outgoing occupation  |
| talking                   | Subscriber in the call state   |
| holding                   | TAU-32M.IP subscriber put a remote subscriber on hold  |
| holded                    | TAU-32M.IP subscriber was put on hold by a remote subscriber                                     |
| conference                | Conference state, the subscriber is a 3-way conference initiator                                 |
| Reasons for call clearing | Description  |
| local                     | TAU-32M.IP subscriber put the phone offhook, didn't perform a call and put the phone back onhook |
| local busy                | TAU-32M.IP subscriber is busy  |
| remote busy               | Remote subscriber is busy  |
| invalid number            | Invalid number is dialled  |
| no answer                 | No response from subscriber  |
| no local user             | Incoming call to non-existent number   |
| no remote user            | Outgoing call to non-existent number   |
| no route                  | Call to unavailable direction  |

|                    |   |
|--------------------|---|
| local clear        | TAU-32M.IP subscriber clearback   |
| remote clear       | Remote subscriber clearback   |
| local fail         | Local or remote failure that has occurred during the connection establishment.  |
| remote fail        | Possible error reasons: codec mismatch, problems during TCP connection establishment (when H.323 is used), overload, resource bottlenecks (bandwidth), etc. |
| remote redirection | Redirection (before—CFB, CFNR, or after the call—CT) performed by the remote subscriber   |
| local redirection  | Redirection (before—CFB, CFNR, or after the call—CT) performed by TAU-32M.IP subscriber   |
| replaced           | This call is replaced by another one while performing 'Call Transfer' service   |
| pickuper           | Call is picked up   |
| pickuper succeed   | 'Call pickup' successfully performed by the subscriber  |
| Pickuper failed    | 'Call pickup' failed  |
| local limit        | Call clearback for the outgoing call concurrent connection limit  |
| remote limit       | Call clearback for the incoming call concurrent connection limit  |

### 6.2.2. Statistics File Operations

Call statistics file is located in /tmp folder on the device.

To transfer the statistics file to a local PC, you should do the following:

- 1 Connect using RS-232 serial port (connection parameters: 115200, 8, n, 1, n; username: admin, w/o password). Go to Linux console by executing **shell** command. Call statistics file is located in 'tmp' folder.
- 2 To perform statistics file readout, run TFTP server on a PC, and specify a directory for the file transfer.
- 3 Go to 'tmp' folder using `cd /tmp` command and transfer statistics file to a local PC: `tftp -pl voip_history <server ip address>`

```
[root@fxs32 /root]$ cd /tmp
```

```
[root@fxs32 /root]$ tftp -pl voip_history <server ip address>
```

### 6.2.3. Port-specific Statistics

CLI is available when the connection to the device is established via RS-232 (connection parameters: 115200, 8, n, 1, n; username: admin, w/o password), or Telnet.

To view the port-specific statistics, use the following command: `pbx statistic <n>`, where <n>—port number.

Table 14—Port statistics record format.

| Record                   | Description                           |
|--------------------------|---------------------------------------|
| Statistic of pbx port 1: | Port that statistics is gathered for  |
| pbx call count           | Number of calls performed by the port |
| pbx port state           | Current port status                   |
| pbx last number          | Last number dialled                   |
|                          |                                       |
| vapi statistic:          | Statistics for voice packets          |
| send packet              | Total amount of packets sent          |
| send octet               | Total amount of bytes sent            |
| receive packet           | Total amount of packets received      |
| receive octet            | Total amount of bytes received        |
| packet lost              | Total amount of packets lost          |
| peak jitter              | Peak jitter                           |

### 6.3. Configuration writing/readout

To configuration readout from the device, connect using RS-232 serial port (connection parameters: 115200, 8, n, 1, n; username: admin, w/o password). Go to Linux console by executing `shell` command. Device configuration is located in 'etc' folder.

To perform the configuration readout, run TFTP server on a PC, and specify a directory for storing the configuration.

Configuration download commands:

```
[admin@fxs32 /admin]$cd /
[admin@fxs32 /]$tar -cf conf.tar /etc/
[admin@fxs32 /]$tftp -pl conf.tar server ip address
```

To upload the configuration, run TFTP server on a PC, and specify a directory with 'conf.tar' configuration file. The archive should contain 'etc' folder.

Configuration record commands:

```
[admin@fxs32 /admin]$cd /
[admin@fxs32 /]$tftp -gl conf.tar server ip address
[admin@fxs32 /]$tar -xf conf.tar
```

Save settings using 'save' command.

Restart the gateway using 'reboot -f' command.

## 6.4. Setting Password for 'admin' User

Given that it is possible to remotely connect to TAU-32M.IP gateway via Telnet, in order to prevent an unauthorized access, we recommend to set password for *admin* user (admin user is not protected by password in factory settings). To set the password, connect to the gateway via COM port or telnet (factory settings address: 192.168.1.2, mask: 255.255.255.0) using terminal application, e.g. TERATERM.

Configuration procedure as follows:

- 1 Connect the null modem cable to COM port of a PC and TAU-32M.IP module 'Console' port (if configuration is performed via COM port), or connect the computer to the module Ethernet port using Ethernet cable (if configuration is performed via telnet).
- 2 Run the terminal application.
- 3 Configure COM port connection: data rate: 115200, data format: 8bit w/o parity, 1 stop bit, w/o flow control; or telnet connection: Factory default IP address: 192.168.1.2, port: 23.
- 4 Press <ENTER>. The following text will appear on screen:

```
*****
*   TAU-32M FXS Gateway   *
*****
```

```
Fxs32M login:
```

Enter *admin*; for factory settings, the password is not required.

- 5 Enter *passwd* command. The following text will appear on screen:

```
> passwd
Changing password for admin
New password:
```

- 6 Enter password, press <ENTER>, confirm password, press <ENTER>:

```
> passwd
Changing password for admin
New password:
Retype password:
Password for admin changed by admin
Oct 15 10:25:50 tmip auth.info passwd: Password for admin changed by
admin
```

- 7 If the password is not applied (it may occur, if the device has a legacy firmware version installed with the legacy file system), check the contents of the 'passwd' file. To do this, go to Linux console

by executing `shell` command, and edit the file using embedded editor 'joe' (use arrow buttons to move the cursor; exit the editor without saving: `<CTRL^C>`, exit and save changes: `<CTRL^(KX)>`): `joe /tmp/etc/passwd`. Add 'x' character into admin user string.

File contents before the edit: `admin::0:0: admin:/ admin:/bin/sh.`

File contents after the edit: `admin:x:0:0: admin:/ admin:/bin/sh.`

- 8 Save settings using 'save' command.
- 9 Restart the gateway using 'reboot -f' command.

## 6.5. Reset to Factory Defaults

Turn the device off. Press and hold the 'F' function button located on the front panel of the device. While holding the button, turn the power on. Hold the button pressed until 'Status' indicator flashes (flashed green and red rapidly), then release the button to avoid another reboot of the device. TAU-32M.IP will begin its operation in 'safemode'. In this mode, the device will be accessible by IP address 192.168.1.2 via WEB interface (user—**admin**, password—**rootpasswd**), or Telnet/SSH (username—**admin**, password is not defined). Access via RS-232 console in this mode, just as for Telnet, will be unprotected (username—**admin**, password is not defined).

To save factory configuration via WEB interface, click 'save' button in any tab.



**Warning!!! In legacy firmware versions, this function may lead to situations when device always starts up in 'safemode'. To restore the normal operation, you should reset device configuration to factory defaults via console or Telnet/SSH.**

Reset configuration to factory defaults, Telnet/SSH:

1. Connect the null modem cable to COM port of a PC and TAU-32M.IP module 'Console' port (if configuration is performed via COM port), or connect the computer to the module Ethernet port using Ethernet cable (if configuration is performed via Telnet/SSH).
2. Run the terminal application.
3. Configure COM port connection: data rate: 115200, data format: 8bit w/o parity, 1 stop bit, w/o flow control; or telnet connection: 192.168.1.2, port 23.
4. Press 'Enter'. The following text will appear on screen:

```
*****
*   TAU-32M FXS Gateway   *
*****
```

Fxs32M login:

Enter admin, password is not required.

5. To reset settings in the protected mode, execute the following commands:
  - a. To reset settings in CLI mode and retain the console password, execute the following commands:

```
> enable  
> config reset static
```

or, if you have to define the dynamic

obtaining of network settings in factory configuration (via DHCP protocol):

```
> enable  
> config reset dhcp
```

b.To reset settings in CLI mode and delete the console password, execute the following commands:

```
> shell  
reset2defaults static
```

or, if you have to define the dynamic obtaining of network settings in factory configuration (via DHCP protocol):

```
> shell  
reset2defaults dhcp
```



---

## 7. SUPPLEMENTARY SERVICES USAGE

### 7.1. Call Transfer

Call transfer service may be performed locally using gateway resources, or remotely using resources of a communicating device. If the service is performed using resources of a communicating device, the access to 'Call transfer' service is established via subscriber port settings menu—'PBX -> Ports'—by selecting '*Transmit Flash*' value in '*Flash transfer*' field, see Section 5.1.2.4. At that, you should specify the Flash impulse transfer method for utilized signalling protocol. Service process logics in this case will be defined by the communicating device.

When 'Call transfer' service is performed locally using gateway resources, the access to this service is established via subscriber port settings menu—'PBX -> Ports'—by selecting 'Attended calltransfer', 'Unattended calltransfer', or 'Local CT' in 'Flash transfer' field, see Section 5.1.2.4.

'*Attended calltransfer*' service allows you to temporarily disconnect an online subscriber (Subscriber A), establish connection with another subscriber (Subscriber C) and return to the previous connection without dialling or transfer the call while disconnecting Subscriber B (a subscriber that performs the service).

'*Attended calltransfer*' service usage:

While being in a call state with a Subscriber A, put him on hold with short clearback *flash (R)*, wait for 'PBX response' tone and dial a Subscriber C number. When Subscriber C answers, the following operations will be possible:

- R 0—disconnect a subscriber on hold, connect to online subscriber.
- R 1—disconnect an online subscriber, connect to subscriber on hold.
- R 2—switch to another subscriber (change a subscriber).
- R 3—conference.
- R 4—call transfer. Voice connection will be established between Subscribers A and C.
- clearback—call transfer. Voice connection will be established between Subscribers A and C.

Fig. 15 shows an algorithm of 'Attended calltransfer' service performed by Subscriber B via SIP protocol.

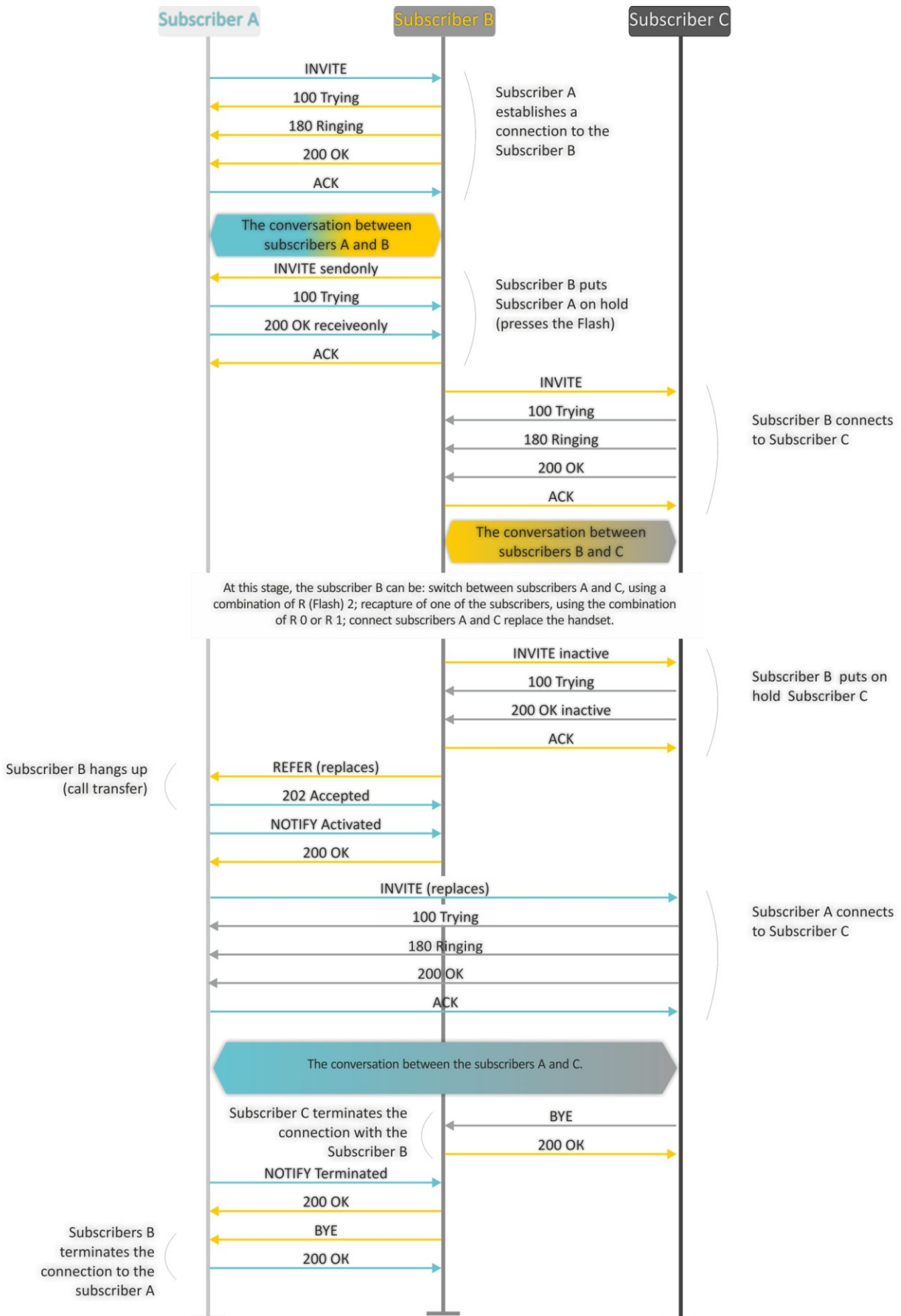


Fig. 15—Algorithm of 'Attended calltransfer' service performed by Subscriber B via SIP protocol

'Unattended calltransfer' service allows to put an online subscriber (Subscriber A) on hold with a short clearback *flash* and dial another subscriber's number (Subscriber C). Call will be transferred automatically when Subscriber A finishes dialling the number.

Fig. 16 shows an algorithm of 'Unattended calltransfer' service performed by Subscriber B via SIP protocol.

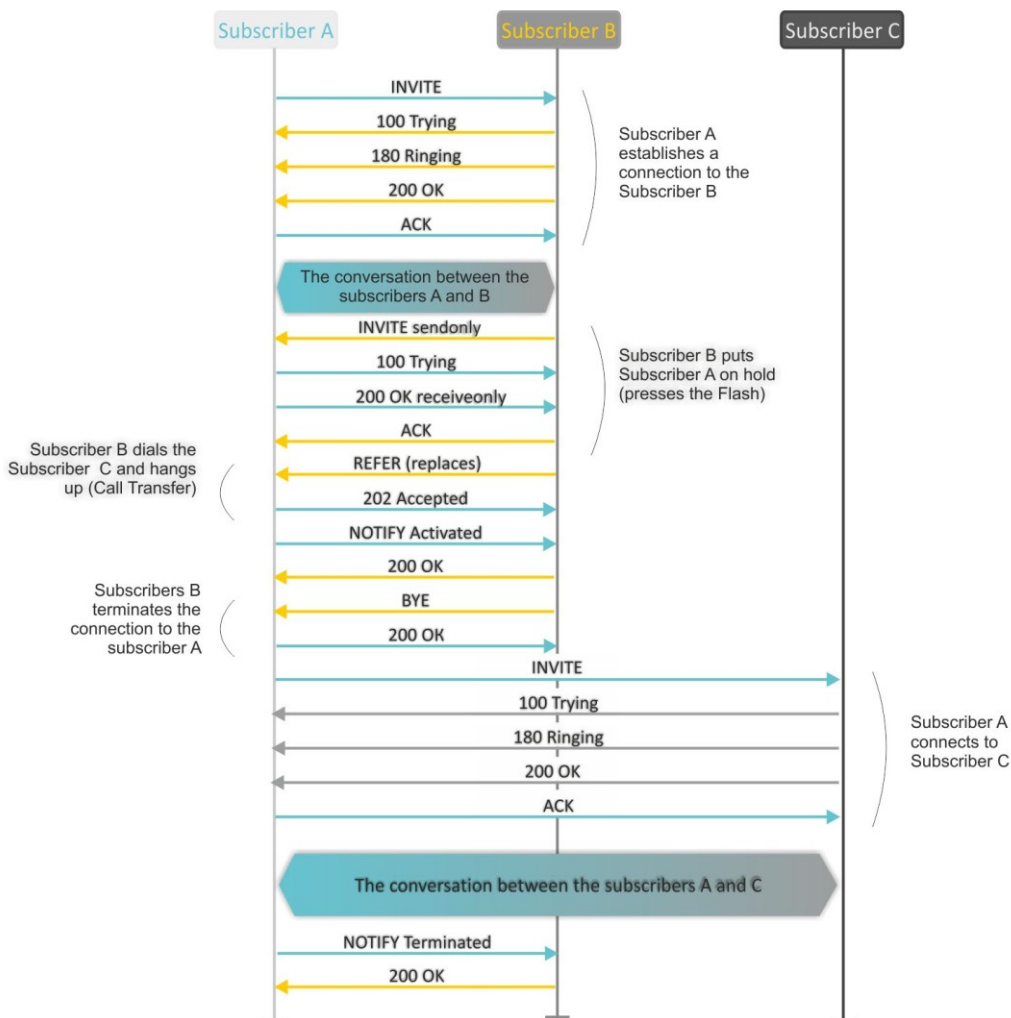


Fig. 16—Algorithm of 'Unattended calltransfer' service performed by Subscriber B via SIP protocol

## 7.2. Call Waiting

This service allows to inform "busy" users about new incoming calls with a special signal.

Upon receiving this notification, user can answer or reject a waiting call.

Access to this service is established via subscriber port settings menu—'PBX -> Ports'—by selecting 'Attended calltransfer', 'Unattended calltransfer', or 'Local CT' in 'Flash transfer' field and selecting '*Call waiting*' checkbox.

Service usage:

If you receive a new call while being in a call state, you may do the following:

- R 0—reject a new call.
- R 1—answer the waiting call and terminate the current call.
- R 2—answer the waiting call and put the current call on hold Further R 0/1/2/3/4 button actions are processed in accordance with the algorithm, described in Section 7.1 Call Transfer.
- R – short clearback (flash).

### 7.3. 3-way Conference

Three-way conference is a service, that enables simultaneous phone communication for 3 subscribers. For entering conference mode, see Section 7.1 Call Transfer.

Subscriber that started the conference is deemed to be its initiator, two other subscribers are the participants. In the conference mode, short clearback 'flash' pressed by the initiator is ignored. Signalling protocol messages, received from the participants and intended to put the initiator side into hold mode, force this participant to leave the conference. At that, the initiator and the second participant will switch into the ordinary two-party call mode.

The conference terminates, when initiator leaves; in this case, both participants will receive clearback message. If one of the participants leaves the conference, the initiator and the second participant will switch into a standard two-party call. Short flash clearback is processed as described in Sections 7.1 Call Transfer and 7.2 Call Waiting.

Fig. 17 shows an algorithm of '3-way conference' service performed locally on the device via SIP protocol.

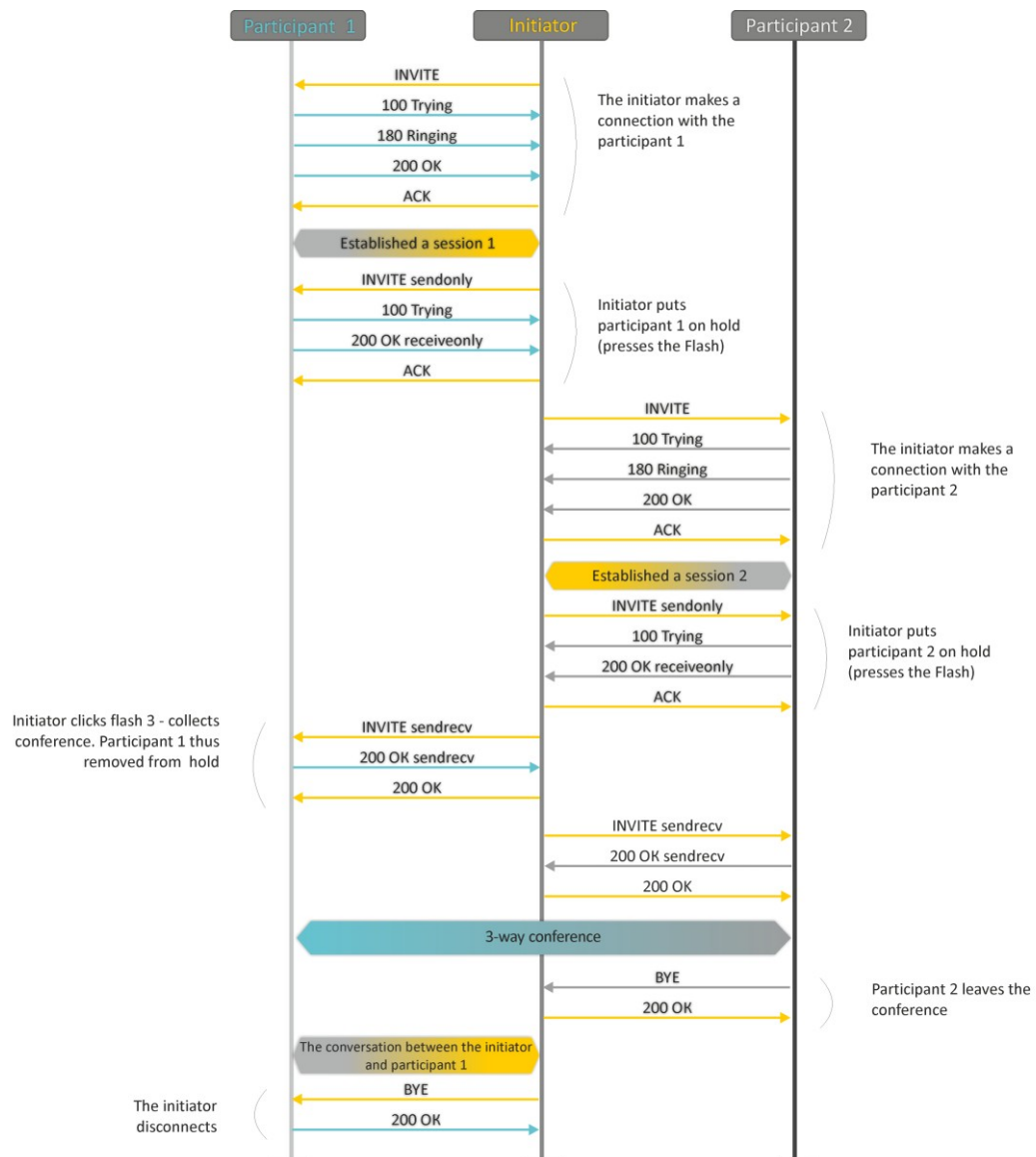


Fig. 17—Algorithm of '3-way conference' service performed locally on the device via SIP protocol

Fig. 18 shows an algorithm of '3-way conference' service performed at the conference server via SIP protocol ('REFER to focus' option).

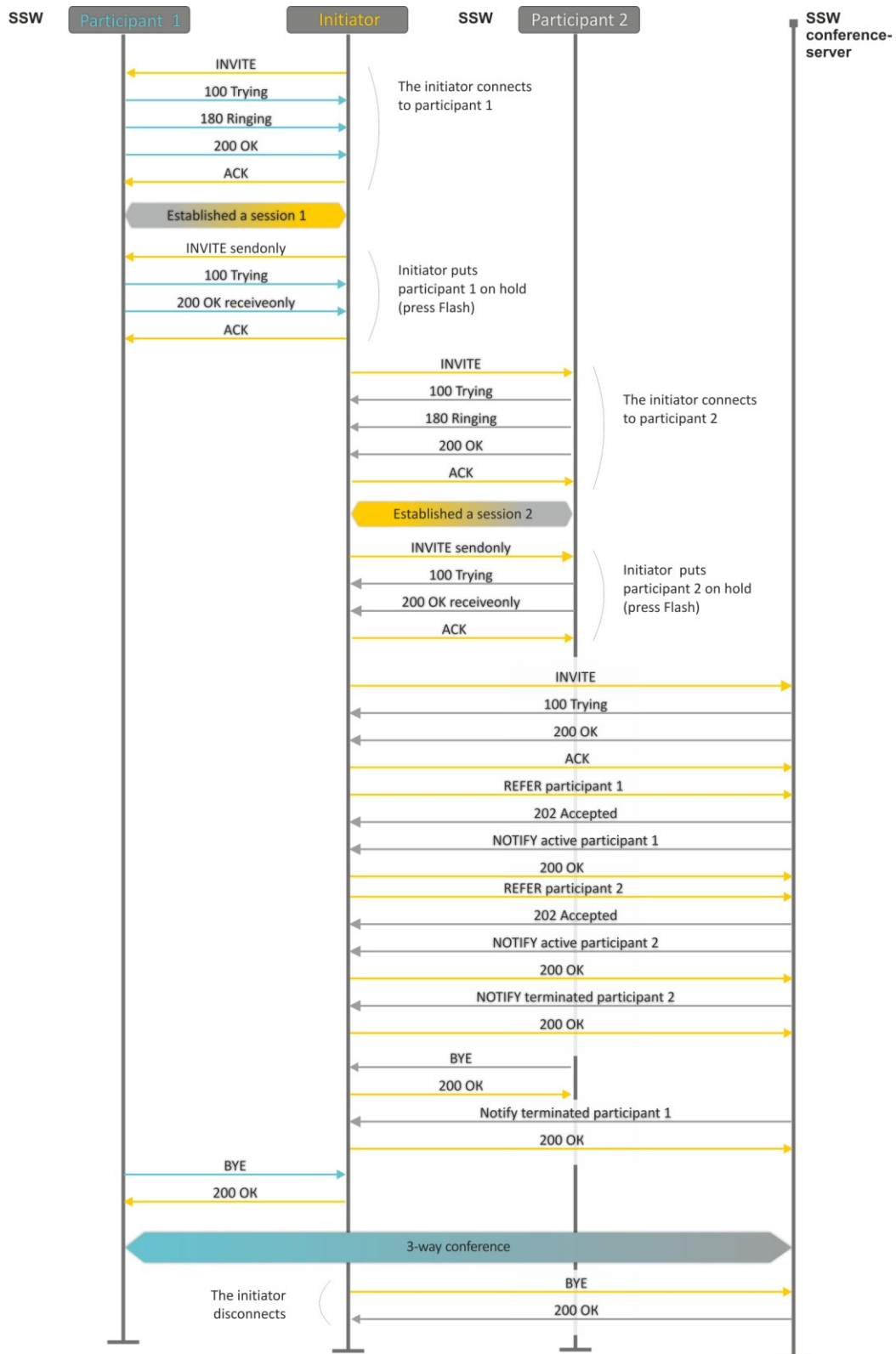


Fig. 18—Algorithm of '3-way conference' service performed at the conference server via SIP protocol (REFER to focus)

Fig. 19 shows an algorithm of '3-way conference' service performed at the conference server via SIP protocol ('REFER to user' option).

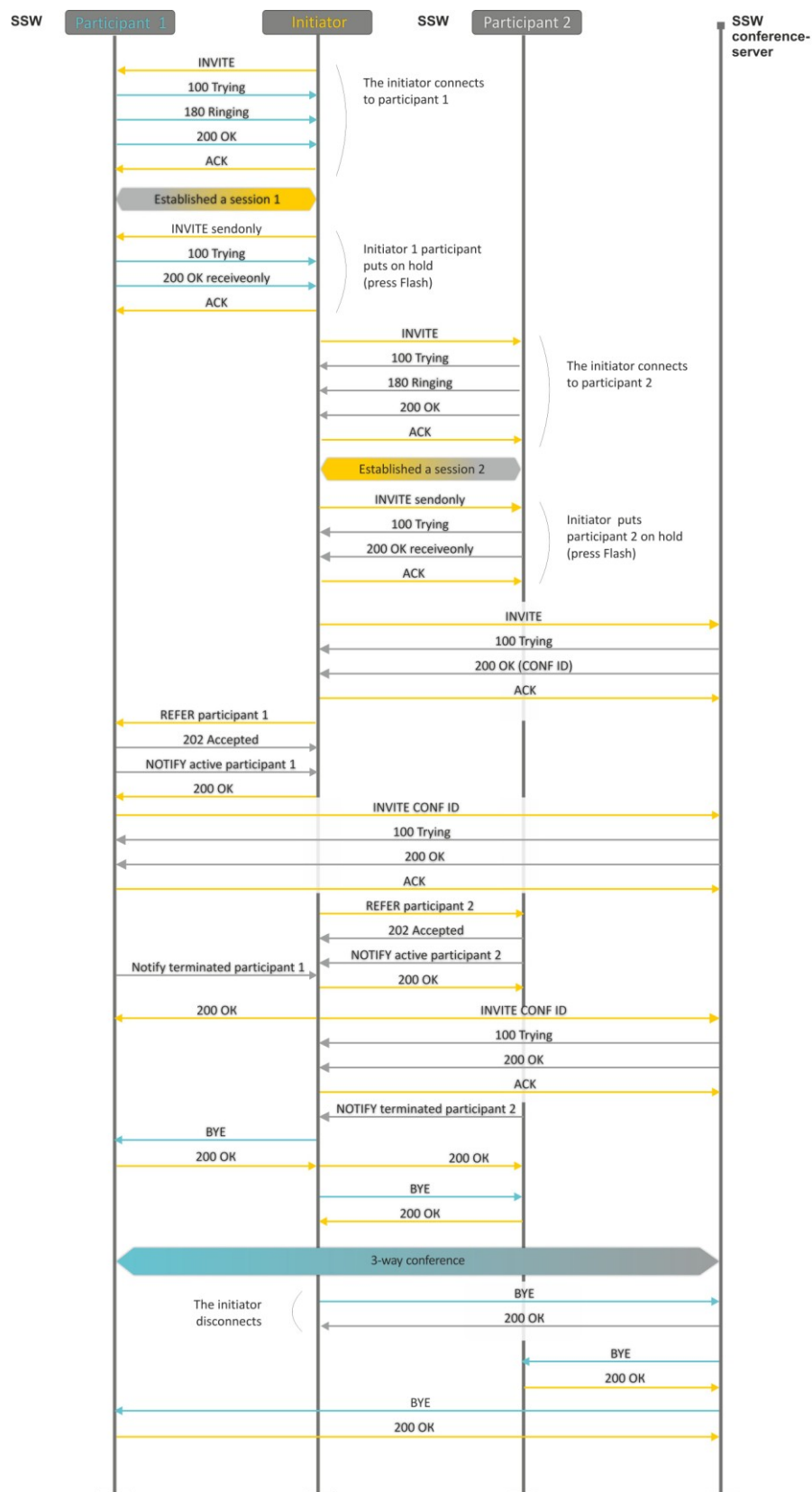


Fig. 19—Algorithm of '3-way conference' service performed at the conference server via SIP protocol (REFER to user)

## 8. CONNECTION ESTABLISHMENT ALGORITHMS

### 8.1. Algorithm of a Successful Call via SIP Protocol

SIP is a session initiation protocol, that performs basic call management tasks such as starting and finishing session.

SIP defines 3 basic connection initiation scenarios: between users, involving proxy server, involving forwarding server. Basic connection initiation algorithms are described in IETF RFC 3665. This section describes an example of a connection initiation scenario via SIP between two gateways, that know each other IP addresses in advance.

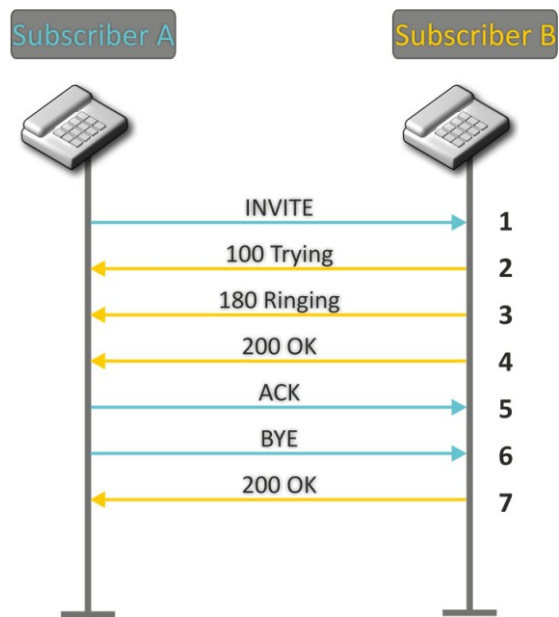


Fig. 19—SIP call algorithm

Algorithm description:

- 1 Subscriber A rings up Subscriber B.
- 2 Subscriber B gateway receives the command for processing.
- 3 Subscriber B is free. In this moment, 'ringing' tone is sent to the Subscriber B phone, and 'ringback' tone to Subscriber A phone.
- 4 Subscriber B answers the call.
- 5 Subscriber A gateway confirms session establishment.
- 6 Subscriber A clears back, 'busy' audio tone is sent to the Subscriber B.
- 7 Subscriber B gateway confirms received clearback command.



## 8.2. Call Algorithm Involving SIP Proxy Server

This section describes a connection initiation scenario between two gateways involving SIP proxy server. In this case, caller gateway (Subscriber A) should know subscriber's permanent address and proxy server IP address. SIP proxy server processes messages received from Subscriber A, discovers Subscriber B, prompts the communication session and performs router functions for two gateways.

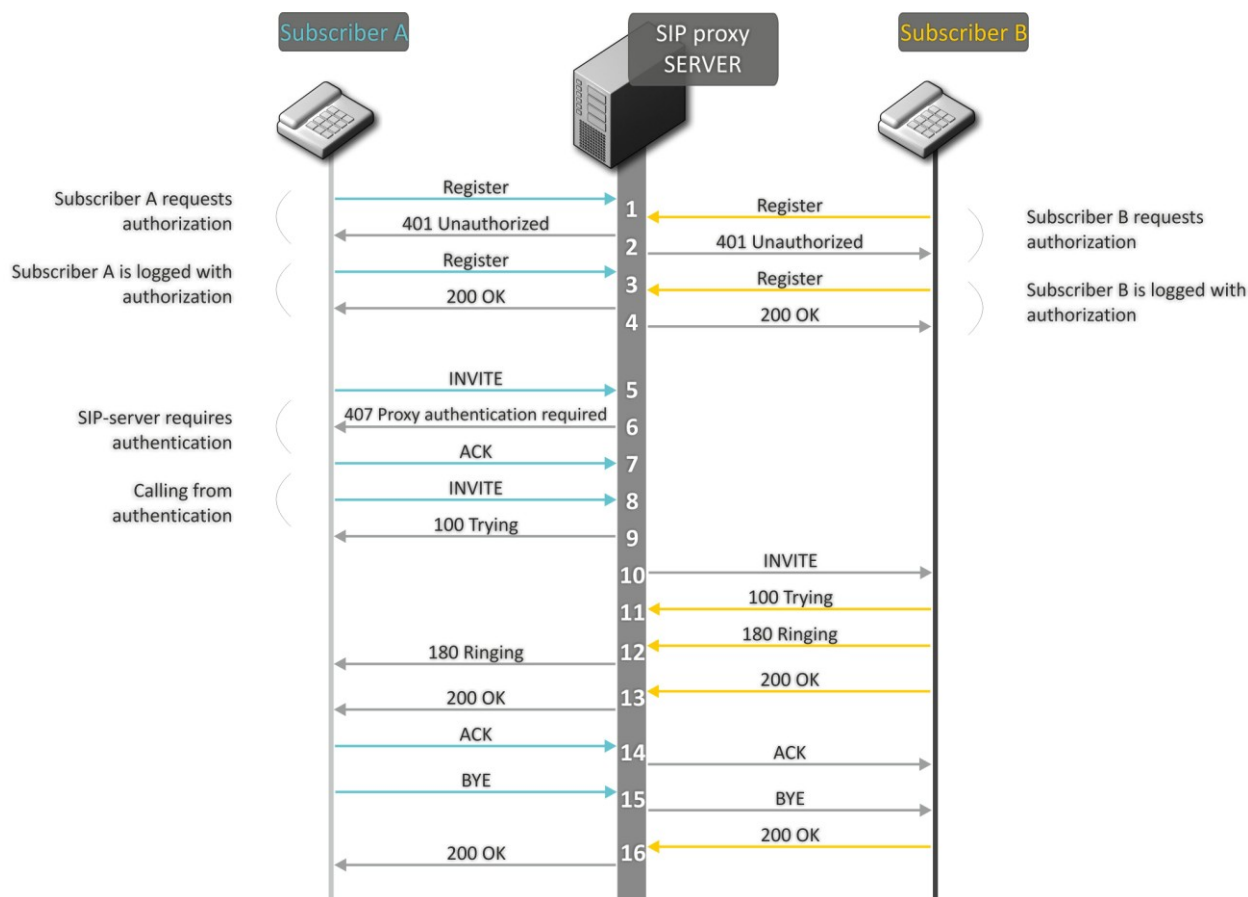


Fig. 20—Call algorithm involving SIP proxy server

Algorithm description:

- 1 Subscriber A and Subscriber B register at SIP server.
- 2 SIP server prompts for authorization.
- 3 Subscriber A and Subscriber B register at SIP server with authorization.
- 4 SIP server responses on successful registration.
- 5 Subscriber A rings up Subscriber B.
- 6 SIP server requests authentication.
- 7 Subscriber A gateway confirms received authorization request command.
- 8 Subscriber A rings up Subscriber B.
- 9 SIP server receives the command for processing.
- 10 SIP server translates Subscriber A call request directed at Subscriber B.
- 11 Subscriber B gateway receives the command for processing.

- 12 Subscriber B is free. In this moment, 'ringing' tone is sent to the Subscriber B phone, and 'ringback' tone to Subscriber A phone.
- 13 Subscriber B answers the call.
- 14 Subscriber A gateway confirms session establishment.
- 15 Subscriber A clears back, 'busy' audio tone is sent to the Subscriber B.
- 16 Subscriber B gateway confirms received clearback command.

### 8.3. Call Algorithm Involving Forwarding Server

This section describes a connection initiation scenario between two gateways involving forwarding server. In this case, caller gateway (Subscriber A) establishes connection unassisted, and the forwarding server only translates callee permanent address into its current address. Subscriber obtains forwarding server address from the network administrator.

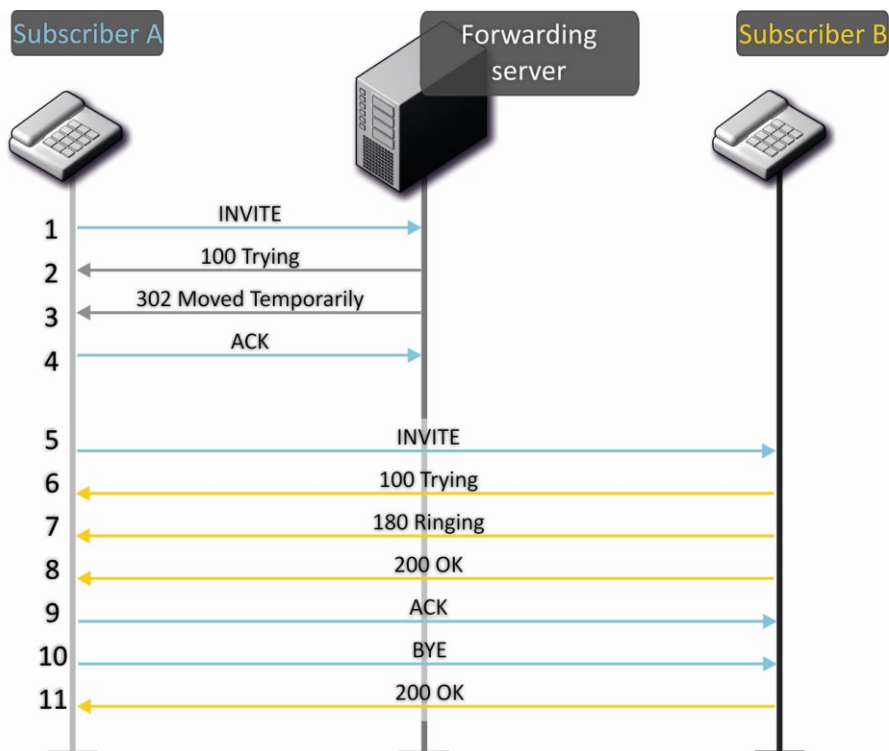


Fig.21—Call algorithm involving forwarding server

Algorithm description:

- 1 Subscriber A rings up Subscriber B. Call is sent to the forwarding server with the callee address information.
- 2 Forwarding server receives the command for processing.
- 3 Forwarding server requests the information on the Subscriber B current address from the location server. Received information (the callee current address and the list of callee registered addresses) is sent to Subscriber A in '302 moved temporarily' message.
- 4 Subscriber A gateway confirms the reception of reply from the forwarding server.

- 5 Subscriber A rings up Subscriber B directly.
- 6 Subscriber B gateway receives the command for processing.
- 7 Subscriber B is free. In this moment, 'ringing' tone is sent to the Subscriber B phone, and 'ringback' tone to Subscriber A phone.
- 8 Subscriber B answers the call.
- 9 Subscriber A gateway confirms session establishment.
- 10 Subscriber A clears back, 'busy' audio tone is sent to the Subscriber B.
- 11 Subscriber B gateway confirms received clearback command.

## 8.4. Algorithm of a Successful Call via H.323 Protocol

H.323 is ITU-T standard that describes specifications for audio and video data transmission via packet switching networks and includes standards for video and voice codecs, public domain applications, call and system management. H.323 protocol family includes three basic protocols: terminal equipment and zone controller interaction protocol—RAS, connection management protocol—H.225, and logic channel management protocol—H.245.

This section describes an example of a basic connection initiation scenario via H.323 protocol between two gateways without a gatekeeper.

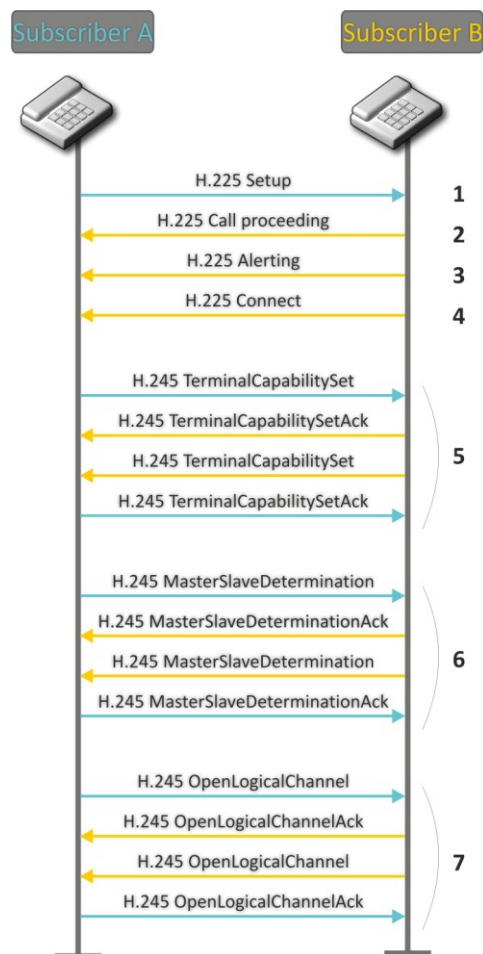


Fig. 22—H.323 call algorithm

Algorithm description:

Connection establishment (via ITU-Q.931/H.225 protocol)

- 1 Subscriber A gateway rings up Subscriber B (sends 'setup' message).
- 2 Subscriber B gateway sends a message, stating the possibility of process continuation.
- 3 Subscriber B gateway sends 'Alerting' notification message. Subscriber B is free. In this moment, 'ringing' tone is sent to the Subscriber B phone, and 'ringback' tone to Subscriber A phone.
- 4 Subscriber B gateway answers the call.

Logic channel establishment (via H.245 protocol)

- 5 Subscriber A gateway informs Subscriber B gateway on its supported capabilities (TerminalCapabilitySet). Subscriber B gateway confirms the request (TerminalCapabilitySetAck). The same procedure is repeated in reverse direction from Subscriber B to Subscriber A.
- 6 Operation mode is defined—which gateway will be the 'master', and which will be the 'slave'.
- 7 Each gateway sends a message for a logic channel opening (OpenLogicalChannel). If gateways are ready to receive the data, they send confirmation messages on logic channel opening (OpenLogicalChannelAck). Call RTP sessions opens.

## 8.5. Algorithm of a Successful Call via H.323 Protocol with Gatekeeper

Gatekeeper performs address translation and manages H.323 terminals' access to network resources.

This section describes an example of a basic connection initiation scenario via H.323 protocol with a gatekeeper.

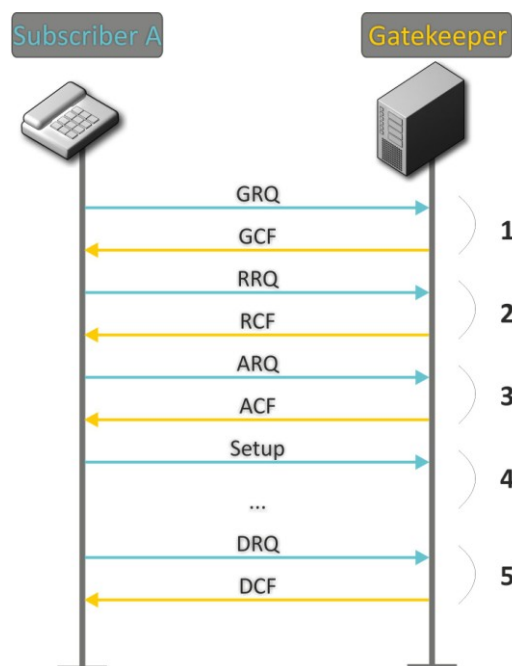


Fig. 23—Gatekeeper call algorithm

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Call establishment algorithm for a subscriber and a gatekeeper:

- 1 Gatekeeper discovery:
  - GRQ(gatekeeper request)—sending discovery request.
  - GCF(gatekeeper confirm)—successful discovery.
  
- 2 Subscriber registration on a gatekeeper:
  - RRQ (registration request)—registration request.
  - RCF (registration confirm)—successful registration.
  
- 3 Request to access GK resources (when performing outgoing call):
  - ARQ (admission request)—connection request.
  - ACF (admission confirm)—successful response to request by the gatekeeper.
  
- 4 Call (similar to Paragraph 8.3).
- 5 GK call resources deallocation.

## 9. DESCRIPTION OF CONFIGURATION FILES

This section lists description of a configuration file, used by the device.

For 'cfg.yaml' file description, see Tables 15 to 17.

To edit configuration files, you should:

- 1 Connect using RS-232 serial port (connection parameters: 115200, 8, n, 1, n; username: admin, w/o password). Go to Linux console by executing 'shell' command. Configuration file is located in 'etc/config' folder.
- 2 Edit the file using embedded editor 'joe' (use arrow buttons to move the cursor; exit the editor without saving: ctrl^c, exit and save changes: ctrl^(kx)): joe /etc/config/cfg.yaml.
- 3 When you finish editing and exit the editor, save settings with 'save' command.

### 9.1. Configuration file—CFG.YAML

Configuration file formation hierarchy:

```
#!version 1.0
```

```
Node1:
```

```
  Node2:
```

```
    Parameter1: Value1
```

```
    Parameter2: Value2
```

Configuration file version (#!version 1.0) is used for autoupdate.

When working with CFG.YAML, you should observe the following rules:

- Do not add/remove nodes.
- Do not use tab characters '/t'.
- Use spaces ' ' only.
- Add the same number a of spaces ' ' before each node with a specific nesting level.

Table 15—VOIP configuration

| <i>Field name</i>                        | <i>Description</i>  | <i>Values</i>         |
|--|---|-----------------------|
| <b>h323—H.323 protocol configuration</b> |   |                       |
| enableh323                               | H.323 protocol  | 0—disable<br>1—enable |
| timetolive                               | Time period in seconds, for which the device will keep its registration on a gatekeeper | 10-65535              |

|                  |   |   |
|------------------|---|---|
| keepalivetime    | Time period in seconds, after which the device will renew its registration on a gatekeeper  | 10-65535  |
| h235             | Authentication on the gatekeeper with H.235 protocol  | 0—disable<br>1—enable   |
| ignore_gcf       | Output authentication data in RRQ message via H.235 protocol  | 0—only in case of reception of supported hash method in GCF message<br>1—in any events  |
| disabletunneling | H.245 signal tunnelling through Q.931 signal channels   | 0—tunnelling enabled<br>1—tunnelling disabled   |
| disablefaststart | faststart feature   | 0—faststart enabled<br>1—faststart disabled   |
| usegatekeeper    | Registration on a gatekeeper  | 0—disable<br>1—enable   |
| gatekeeperip     | Gatekeeper IP address   | A.B.C.D   |
| h323aliase       | Gateway identifier  | String, 15 characters max.  |
| isgateway        | Method of device registration on gatekeeper   | 0—registered as a terminal device<br>1—registered as a gateway  |
| dtmftransfer     | Transfer method for flash and DTMF tones via H.323 protocol   | 1—H.245 Alphanumeric—basicstring compatibility is used for DTMF transmission, and hookflash compatibility for flash transmission (flash is transferred as '!' symbol).<br>2—H.245 Signal—dtmf compatibility is used for DTMF transmission, and hookflash compatibility for flash transmission (flash is transferred as '!' symbol).<br>3—Q931 Keypad IE—for DTMF and flash transmission (flash is transferred as '!' symbol), Keypad information element is used in INFORMATION Q931 message. |
| bearercapability | Select information transfer service (We recommend using value '3.1 kHz Audio'. All other values used only for compatibility with communicating gateways.) | 0 – Speech<br>8 – Unrestricted Digital<br>9 – Restricted Digital<br>16 – 3.1 kHz Audio<br>17 – Unrestricted Digital With Tones  |
| password         | Password used for H.235 protocol authentication   | String, 15 characters max.  |

| <b>range—TCP/IP protocol settings</b>                               |   |                       |
|---|---|-----------------------|
| tcpportmin  | The lower limit of a range of TCP ports used for H.323 - H.245/H.225 stack protocols' operation         | 1024-65535            |
| tcpportmax  | The upper limit of a range of TCP ports used for H.323 - H.245/H.225 stack protocols' operation         | tcpportmin-65535      |
| udpportmin  | The lower limit of a range of UDP ports used for H.323 stack RAS protocol operation                     | 1024-65535            |
| udpportmax  | The upper limit of a range of UDP ports used for H.323 stack RAS protocol operation                     | udpportmin-65535      |
| rtph323min  | The lower limit of a range of RTP ports used for H.323 protocol operation                               | 1024-65535            |
| rtph323max  | The upper limit of a range of RTP ports used for H.323 protocol operation                               | rtph323min-65535      |
| rtpsipmin   | The lower limit of a range of RTP ports used for SIP protocol operation                                 | 1024-65535            |
| rtpsipmax   | The upper limit of a range of RTP ports used for SIP protocol operation                                 | rtpsipmin-65535       |
| intrcpmin   | The lower limit of a range of ports used for pickup traffic transmission (SORM feature)                 | 1024-65535            |
| intrcpmax   | The upper limit of a range of ports used for pickup traffic transmission (SORM feature)                 | Intrcpmin-65535       |
| diffserv  | Type of service for RTP packets (for utilized values, see Table 8)                                      | 0-255                 |
| sip_diffserv  | Type of service for SIP packets (for utilized values, see Table 8)                                      | 0-255                 |
| verify_remote_media   | Control of parameters of media traffic received   | 0—disable<br>1—enable |
| <b>dvo—configuration of access codes for supplementary services</b> |   |                       |
| callwaiting   | 'Call waiting' service  | 00-99                 |
| ct_attended   | 'Call transfer' service with the wait for response of the subscriber, the call is being forwarded to    | 00-99                 |
| ct_unattended   | 'Call transfer' service without the wait for response of the subscriber, the call is being forwarded to | 00-99                 |
| cf_unconditional  | 'Call forward unconditional' service (CFU)  | 00-99                 |



|                                       |   |  |
|---------------------------------------|---|--|
| cf_busy                               | 'Forward on busy' service (CFB)   | 00-99  |
| cf_noanswer                           | 'Forward on no reply' service (CFNR)  | 00-99  |
| cf_outofservice                       | 'Forward on out of service' service (CFOOS)   | 00-99  |
| dnd                                   | Restrict all incoming calls, outgoing communication is possible   | 00-99  |
| <b>sip—SIP protocol configuration</b> |   |  |
| enablesip                             | SIP protocol  | 0—disable<br>1—enable  |
| invite_init_t                         | SIP timer—T1, ms  | 100-1000   |
| invite_total_t                        | Total timeout for message transmission, ms  | 1000-39000   |
| transport                             | Transport layer protocol, used for SIP message transmission   | 0—Use both UDP and TCP protocols, UDP priority will be higher<br>1—Use both UDP and TCP protocols, TCP priority will be higher<br>2—Use UDP protocol only<br>3—Use TCP protocol only |
| sip_mtu                               | Maximum SIP protocol data size in bytes, sent with UDP transport protocol   | 1350-1450  |
| publicip                              | IP address of a public NAT  | A.B.C.D  |
| shortmode                             | Use shortened field names in SIP protocol header  | 0—disable<br>1—enable  |
| port_reg_delay_t                      | Timeout between successive registrations of neighbouring ports (ms)   | 500..5000  |
| stun_enable                           | Use STUN server for public address discovery  | 0—disable<br>1—enable  |
| stun_server                           | STUN server IP address  | A.B.C.D  |
| stun_interval                         | STUN server polling period  | 10-1800  |
| <b>general—basic settings</b>         |   |  |
| device_name                           | device name   | String, 15 characters max.<br>or ""—parameter is not defined   |
| start_timer                           | Dialling timeout for the first digit of a number; when there is no dialling during the specified time, 'busy' tone will be sent to the subscriber, and the dialling will end. | 10-300   |
| duration_timer                        | Complete number dialling timeout  | 10-300   |
| wait_answer_timer                     | Subscriber's response timeout   | 40-300   |
| use_uni                               | Use prefix in SIP-T protocol operations   | 0—disable<br>1—enable  |
| unit_prefix                           | Prefix for SIP-T protocol operations  | 0—20 digits  |

|   |  |   |
|---|--|---|
| fans_force_enable                               | continuous fan operation   | 0—disable (turn on at threshold)<br>1—enable  |
| fans_threshold_temperature                      | Fans turn on threshold (°C)  | 35..55  |
| power_mode                                      | Extended range mode (in normal mode, voltage on subscriber units is 34V, in extended range mode—54V.<br>Parameter is used only for TAU-32M.IP rev. 2 | 0—disable<br>1—enable   |
| <b>trace—configuration of Syslog parameters</b> |  |   |
| sip_level                                       | SIP protocol log level   | -1..9   |
| h323_level                                      | H.323 protocol log level   | 0-6   |
| vapi_level                                      | VAPI library log level   | AB, where:<br>A=0..6 (Lib level), B=1..5 (APP level)  |
| vapi_enabled                                    | VAPI library logging   | 0—disable<br>1—enable   |
| app_info  | Send application info messages to Syslog server  | 0—disable<br>1—enable   |
| app_warn  | Send application warning messages to Syslog server   | 0—disable<br>1—enable   |
| app_err   | Send application failure messages to Syslog server   | 0—disable<br>1—enable   |
| app_dbg   | Send application debug messages to Syslog server   | 0—disable<br>1—enable   |
| trace_out                                       | Direction of Syslog information output   | off—do not store to syslog<br>syslog_server—store to SYSLOG server<br>stdout—store to STDOUT  |
| syslog_addr                                     | Syslog server IP address   | A.B.C.D   |
| syslog_port                                     | Syslog server port for message reception   | 1-65535   |
| run_syslog                                      | Run Syslog on device startup   | 0—disable<br>1—enable   |
| app_alarm                                       | Send alarm event messages to Syslog server   | 0—disable<br>1—enable   |
| <b>limits—call limits</b>                       |  |   |
| limit_0 to 19                                   | Call restriction rules<br>Examples:<br>limit_0: [proxy] 5<br>limit_1: 192.168.16.53 8  | A.B.C.D or FQDN or [proxy] N<br>where:<br>[proxy]—defines the restriction for calls through SIP-proxy or H.323 Gatekeeper<br>N—number of simultaneous calls |

| <b>groups—call groups</b>   |   |   |
|---|---|---|
| <i>group_0 to 31—call group configuration</i>   |   |   |
| phone   | Group number  | String, 20 characters max.<br>or ""—parameter is not defined  |
| name  | Group name used for authentication  | String, 20 characters max.<br>or ""—parameter is not defined  |
| password  | Authentication password   | String, 20 characters max.<br>or ""—parameter is not defined  |
| ports   | List of subscriber ports belonging to the group   | String, 30 characters max., ports are comma-separated, or ""—parameter is not defined<br>Warning!!! Enumeration of subscriber ports and pickup groups, used in a file, is less by 1 than enumeration, used in web interface and on the device housing!!!                      |
| type  | Group type  | 0—group call<br>1—serial discovery group<br>2—cyclic group  |
| timeout   | Call timeout for a single group member  | 0-99  |
| busy  | Call queueing, when all group members are busy  | 0—group without a queue<br>1—group with a queue   |
| enabled   | Group usage   | 0—disable<br>1—enable   |
| sip_port  | Local UDP port used for port operations via SIP protocol  | 0-65535   |
| profile_id  | SIP profile number  | 0-7   |
| <b>cadence—'Distinctive ring' service</b>   |   |   |
| <i>- cadence_0.. 31—you may use up to 32 'distinctive rings'</i>  |   |   |
| <b>Enumeration of 'distinctive rings', used in a file, is less by 1 than enumeration, used in web interface!!! Example: 'cadence 0' in a file corresponds to 'rule 1' in WEB interface.</b> |   |   |
| rule  | Mask of the number of the caller that will trigger the 'distinctive ring' with a call to the requested port | —logical OR—used to separate rules<br>X or x—any number from 0 to 9, equal to a range [0-9]<br>0 - 9—numbers from 0 to 9<br>*—* character<br>#—# character<br>[ ]—define ranges (with a hyphen), or enumeration (w/o spaces, commas, and other characters between the digits) |

|   |  |  |
|---|--|--|
| ring  | Ring duration  | 0-25500  |
| pause   | Pause duration   | 0-25500  |
| mask  | Subscriber profiles for ports using this rule  | Profile numbers from 0 to 7, comma-separated   |
| <b>modifiers</b>  | <b>Modifier configuration</b>  |  |
| <b>modifier_0 .. 15</b>   | <b>You can use up to 16 modifier groups</b>  |  |
| <p>– Enumeration of modifiers and their groups, used in a file, is less by 1 than enumeration, used in web interface!!!</p> <p><b>Example: 'modifier_0' in a file corresponds to 'modifier 1' in WEB interface.</b></p> |  |  |
| mod_rule_0..31  | Rule for modification in a group, specify 3 parameters, space-delimited: number dialling rule, modification for a dialled number, modification for a calling number. | For syntax, see Section <b>5.1.2.11 Modifiers</b>  |
| <i>fxo_group_0 to 15—configuration of FXO groups</i>  |  |  |
| transmit_number   | Transmit the complete number received from IP (from Request URI header of INVITE request) into the line, including FXO unit subscriber number, or not.               | 0—disable<br>1—enable  |
| dont_transmit_prefix  | Transmit the complete number received from IP (from Request URI header of INVITE request) into the line, except for FXO unit subscriber number, or not.              | 0—disable<br>1—enable  |
| sip_port  | Local UDP port used for port operations via SIP protocol   | 0-65535  |
| enabled   | Group usage  | 0—disable<br>1—enable  |
| busy  | Call queueing, when all group members are busy   | 0—group without a queue<br>1—group with a queue  |
| ports   | List of subscriber ports belonging to the group  | String, 30 characters max., ports are comma-separated, or ""—parameter is not defined<br>Warning!!! Enumeration of subscriber ports and pickup groups, used in a file, is less by 1 than enumeration, used in web interface and on the device housing!!! |

|   |   |  |
|---|---|--|
| password  | Authentication password   | String, 20 characters max.<br>or ""—parameter is not defined                           |
| name  | Group name used for authentication  | String, 20 characters max.<br>or ""—parameter is not defined                           |
| phone   | Group number  | String, 20 characters max.<br>or ""—parameter is not defined                           |
| profile_id  | SIP profile number  | 0-7  |
| type  | Line selection mode   | 3—First free—selection of the first free line<br>4—Cycle—cyclic line selection forward |
| fxo_call_busy   | Send reply 503 via SIP protocol when all subscriber lines (FXO) are busy, otherwise—486.                    | 0—disable (reply 486)<br>1—enable (reply 503)  |
| <b>profile—SIP profiles</b>   |   |  |
| <i>profile_0 .. 7—SIP profile configuration</i>   |   |  |
| <b><i>Enumeration of SIP profiles, used in a file, is less by 1 than enumeration, used in web interface!!!</i></b>    |   |  |
| <b>Example: 'profile_0' in a file corresponds to 'profile 1' in WEB interface.</b>                                    |   |  |
| <b><i>sip, codecs, regexprd, dialplan and sip_cadences parameters are configured separately for each profile.</i></b> |   |  |
| <b><i>sip—SIP protocol configuration</i></b>  |   |  |
| cw_ringback   | Send 180 or 182 message, when the second call is received on the port with an active 'Call waiting' service | 0—send 180<br>1—send 182   |
| ringback  | Parameter defines, whether the gateway should send a ringback tone upon receiving an incoming call          | 0—disable<br>1—enable  |


|                 |   |  |
|-----------------|---|--|
| ringback_sdp    | Transfer of 'ringback' tone upon receiving '183 Progress' message | <p>0—when an incoming call is received, the gateway will not generate a ringback tone.</p> <p>1—when an incoming call is received, the gateway will generate a ringback tone and send it to the communicating gateway in the voice frequency path. Voice frequency path forwarding will be performed along with '180 ringing' message transmission via SIP protocol.</p> <p>2—when an incoming call is received, the gateway will generate a ringback tone and send it to the communicating gateway in the voice frequency path. Voice frequency path forwarding will be performed along with '183progress' message transmission via SIP protocol.</p> |
| 100rel          | Utilization of reliable provisional responses (RFC3262)           | <p>0—reliable provisional responses are supported</p> <p>1—reliable provisional responses are mandatory</p> <p>2—reliable provisional responses are disabled</p>   |
| no_replaces     | Usage of 'replaces' tag during 'Call Transfer'                    | <p>0—enable</p> <p>1—disable</p>   |
| mode            | SIP server operation mode (SIP-proxy)                             | <p>0—disable</p> <p>1—SIP-proxy redundancy mode without main SIP-proxy management</p> <p>2—SIP-proxy redundancy mode with main SIP-proxy management</p>  |
| user_phone      | Usage of 'User=Phone' tag in SIP URI                              | <p>0—disable</p> <p>1—enable</p>   |
| uri_escape_hash | Transfer of hash symbol (#) in SIP URI                            | <p>0—as '#' symbol</p> <p>1—as escape sequence '%23'</p>   |

|                         |   |   |
|-------------------------|---|---|
| dtmfmime                | MIME extension type used for DTMF transmission in SIP protocol INFO messages                    | <p>dtmf—DTMF is sent in application/dtmf extension ('*' and '#' are sent as digits 10 and 11)</p> <p>dtmfr—DTMF is sent in application/dtmf-relay extension ('*' and '#' are sent as symbols '*' and '#')</p> <p>audio—DTMF is sent in audio/telephone-event extension ('*' and '#' are sent as digits 10 and 11)</p> |
| hfmime                  | MIME extension type used for Flash transmission in SIP protocol INFO messages                   | <p>dtmf—flash is sent as 'signal=hf'; if application/dtmf is used, then the flash is sent as the digit '16'</p> <p>hookf—flash is sent in Application/ Hook Flash extension (as 'signal=hf')</p> <p>broadsoft—flash is sent in Application/ Broadsoft extension (as 'event flashhook')</p>                            |
| register_retry_interval | Retry interval for SIP server registration attempts, when the previous attempt was unsuccessful | 10-3600   |
| inbound_proxy           | Rules for incoming calls  | <p>0—receive incoming calls from all hosts</p> <p>1—receive incoming call from SIP-proxy only</p>   |
| domain                  | SIP domain  | String, 20 characters max. or ""—parameter is not defined   |
| domain_to_reg           | Use domain for registration (REGISTER messages in request URI)                                  | <p>0—disable</p> <p>1—enable</p>  |
| options                 | Test the main proxy using OPTIONS, REGISTER, or INVITE messages in 'homing' redundancy mode     | <p>0 – INVITE</p> <p>1 – OPTIONS</p> <p>2 – REGISTER</p>  |
| keepalivet              | Period of time between OPTIONS or REGISTER management message transfers, ms                     | 10000-3600000   |

|                |   |  |
|----------------|---|--|
| outbound       | Use SIP-proxy as an outbound proxy for outgoing calls   | 0—disable<br>1—enable<br><br>2—enable and play busy tone if port is not registered   |
| obtimeout      | Dialling timeout for directions not specified in configuration, when 'outbound proxy' and 'dialplan' routing rules are used, in seconds   | 0-300  |
| expires        | Registration renewal time period  | 10-345600  |
| authentication | device authentication mode  | 1—enable SIP server authentication with common user name and password for all subscribers<br><br>2—enable SIP server authentication with different user names and passwords for each subscriber  |
| registration   | Usage of registration server<br><br>Used value is a decimal number, calculated from the binary representation of a string of registrars being used.<br><br>regrar: 4 3 2 1 0<br><br>I.e. usage of 3 and 4 registrars only will be equal to the following binary record: 11000, parameter value after conversion to a decimal system—24. | 0—disable<br><br>1—use regrab_0<br><br>2—use regrab_1<br><br>4—use regrab_2<br><br>8—use regrab_3<br><br>16—use regrab_4<br><br>3—use regrab_0 and 1<br><br>7—use regrab_0, 1, 2<br><br>15—use regrab_0, 1, 2, 3<br><br>31—use all regrabars |
| username       | User name for 'global' mode authentication  | String, 20 characters max.<br>or ""—parameter is not defined   |
| password       | Password for 'global' mode authentication   | String, 20 characters max.<br>or ""—parameter is not defined   |
| natsupport     | Parameter is not used   |  |
| publicip       | Parameter is not used   |  |
| stunserver     | Parameter is not used   |  |



|                        |   |   |
|------------------------|---|---|
| reduce_sdp_media_count | Remove inactive media streams during SDP session modification   | 0—disable<br>1—enable   |
| p_rtp_stat             | Use 'P-RTP-Stat' header in BYE request or in its reply to transfer RTP statistics                                       | 0—disable<br>1—enable   |
| timer                  | SIP session timer support (RFC 4028)  | 0—disable<br>1—enable   |
| min_se                 | Minimum time interval for connection health checks in seconds   | 90-1800   |
| session_expires        | Period of time in seconds that should pass before the forced session termination, if the session is not renewed in time | 90-80000  |
| proxy_0                | SIP proxy server address (0—main, 1—first redundant, ...)   | String, 40 characters max.<br>or ""—parameter is not defined  |
| proxy_1                |   |   |
| proxy_2                |   |   |
| proxy_3                |   |   |
| proxy_4                |   |   |
| regrar_0               | registration server address (0—main, 1—first redundant, ...)  | String, 40 characters max.<br>or ""—parameter is not defined  |
| regrar_1               |   |   |
| regrar_2               |   |   |
| regrar_3               |   |   |
| regrar_4               |   |   |
| keep_alive_mode        | Active session support mode for operations through NAT  | 0—off—disabled<br>1—options—use OPTIONS request as an active session support message<br>2—notify—use NOTIFY notification as an active session support message<br>3—CRLF—use CRLF special request as an active session support message |
| keep_alive_interval    | Active session support message transmission period  | 30-120  |
| conference_type        | Conference assembly mode  | 0—Local—conference assembly is performed locally at the gateway Voice packets are mixed at the gateway.<br>1—Remote—conference assembly is performed at the conference server Voice packets are mixed at the server                   |
| conference_serv_name   | Conference server name in Remote mode operation   | String, 50 characters max.  |

|                                     |  |  |
|-------------------------------------|--|--|
| ims_notify_on                       | Service (simulation service) management using IMS (3GPP TS 24.623)       | 0—disable<br>1—enable  |
| xcap_conference_name                | Name sent in XCAP attachment for '3-party conference' service management | String, 30 characters max.   |
| xcap_hotline_name                   | Name sent in XCAP attachment for 'Hotline' service management            | String, 30 characters max.   |
| xcap_cw_name                        | Name sent in XCAP attachment for 'Call waiting' service management       | String, 30 characters max.   |
| xcap_callhold_name                  | Name sent in XCAP attachment for 'Call hold' service management          | String, 30 characters max.   |
| use_alert_info                      | 'alert-info' header processing in INVITE request                         | 0—disable<br>1—enable  |
| changeover                          | Type of requests used for changeover to redundant proxy                  | 0 – INVITE, REGISTER<br>1 – REGISTER<br>2 – INVITE   |
| changeover_by_408                   | Redundant proxy changeover when response 408 is received                 | 0—no changeover when response 408 received<br>1—perform changeover when response 408 received  |
| only_register_changeover            | Type of requests used for changeover to redundant proxy                  | 0 – INVITE, REGISTER<br>1 – REGISTER<br>2 – INVITE   |
| ruri_full_compliance                | RURI control for incoming call   | 0—partial control (user)<br>1—full control (user, host, port)  |
| <b>codecs—device codec settings</b> |  |  |
| g711a                               | G.711A codec   | 0—disable<br>1, 2, 3, 4, 5—enable<br><br>The value represents the codec utilization priority:<br>1—the highest, 5—the lowest.<br><br> <b>Do not use two different g729 codecs simultaneously</b> |
| g711u                               | G.711U codec   |  |
| g726_32                             | G.726-32 codec   |  |


|              |   |  |
|--------------|---|--|
| g729a        | G.729 annexA codec (when defining codec compatibility, codec description is sent via SIP specifying that annexB is not used: a=rtpmap:18 G729/8000 a=fmtp:18 annexb=no) |  |
| g729b        | G.729 annexB codec  |  |
| g723         | G.723.1 codec   |  |
| g711pte      | Amount of voice data in milliseconds (ms), transmitted in a single RTP protocol voice packet for G711 codec   | 10, 20, 30, 40, 50, 60   |
| g729pte      | Amount of voice data in milliseconds (ms), transmitted in a single RTP protocol voice packet for G729 codec   | 10, 20, 30, 40, 50, 60, 70, 80   |
| g723pte      | Amount of voice data in milliseconds (ms), transmitted in a single RTP protocol voice packet for G723.1 codec   | 30, 60, 90   |
| g726_32_pte  | Amount of voice data in milliseconds (ms), transmitted in a single RTP protocol voice packet for G726-32 codec  | 10, 20, 30   |
| g726_32_pt   | G.726-32 codec payload type   | 96 – 127   |
| faxdirection | Transmission direction for fax tone detection and subsequent switching to fax codec   | <p>0—tones are detected during both fax transmission and receiving During fax transmission, CNG FAX signal is detected from the subscriber's line. During fax receiving, V.21 signal is detected from the subscriber's line (Caller and Callee)</p> <p>1—tones are detected only during fax transmission During fax transmission, CNG FAX signal is detected from the subscriber's line (Caller)</p> <p>2—tones are detected only during fax receiving During fax receiving, V.21 signal is detected from the subscriber's line (Callee)</p> <p>3—disables fax tone detection, but will not affect fax transmission (off fax transfer)</p> |

|                   |  |  |
|-------------------|--|--|
| dtmftransfer      | DTMF tone transmission method  | <p>0—inband, in RTP voice packets</p> <p>1—according to RFC2833 recommendation, as a dedicated payload in RTP voice packets</p> <p>2—outband, with SIP/H323 protocol methods</p>   |
| flashtransfer     | <p>Short clearback Flash transmission method</p> <p>(Flash transmission by the subscriber's port via IP network is possible only when 'Transmit flash' is configured on this port)</p> | <p>0—Flash transmission disabled</p> <p>1—Flash transmission is performed according to RFC2833 recommendation, as a dedicated payload in RTP voice packets</p> <p>2—Flash transmission is performed with SIP/H323 protocol methods</p> |
| faxtransfer       | Master protocol/codec used for fax transmissions   | <p>0—use G.711A codec for fax transmissions</p> <p>1—use G.711U codec for fax transmissions</p> <p>2—use T.38 protocol for fax transmissions</p>   |
| slave_faxtransfer | Slave protocol/codec used for fax transmissions  | <p>0—use G.711A codec for fax transmissions</p> <p>1—use G.711U codec for fax transmissions</p> <p>2—use T.38 protocol for fax transmissions</p> <p>3—do not use slave protocol/codec for fax transmissions</p>                        |

|                 |  |   |
|-----------------|--|---|
| modemtransfer   | Protocol used for data transfer (modem)                      | <p>0—use G.711A codec in VBD (V.152) mode to transfer data via modem connection</p> <p>1—use G.711U codec in VBD (V.152) mode to transfer data via modem connection</p> <p>2—use G.711A codec to transfer data via modem connection When entering modem data transfer mode via SIP protocol, echo cancellation and VAD are disabled with attributes described in RFC3108 recommendation:</p> <p>a=silenceSup:off - - - -<br/>a=ecan:fb off -;</p> <p>3—use G.711U codec to transfer data via modem connection When entering modem data transfer mode via SIP protocol, echo cancellation and VAD are disabled with attributes described in RFC3108 recommendation:</p> <p>a=silenceSup:off - - - -<br/>a=ecan:fb off -;</p> <p>4—disable modem signal detection</p> <p>5—use G.711A codec in CISCO NSE mode to transfer data via modem connection</p> <p>6—use G.711U codec in CISCO NSE mode to transfer data via modem connection</p> |
| payload         | Type of payload used to transfer RFC2833 packets             | 96-127  |
| nse_payload     | Type of payload used to transfer CISCO NSE packets           | 96-127  |
| silencedetector | Voice activity detector (VAD) and silence suppression (SSup) | <p>0—disable</p> <p>1—enable</p>  |

|                  |   |                                 |
|------------------|---|---------------------------------|
| echocanceller    | Echo cancellation   | 0—disable<br>1—enable           |
| ecan_nlp_disable | NLP disable   | 0—NLP enabled<br>1—NLP disabled |
| rtcp_period      | The voice frequency path status control function Defines the period of time, during which the opposite side will wait for RTCP protocol packets When there is no packets in the specified period of time, established connection will be terminated. Control period value is calculated using the following equation: RTCP timer* RTCP control period, seconds. | 2-65535                         |
| rtcp_timer       | Time period for control packet transfer via RTCP protocol, in seconds   | 5-65535                         |
| rtcp_xr          | Send RTCP Extended Reports packets  | 0—disable<br>1—enable           |
| comfortnoise     | Comfort noise generator   | 0—disable<br>1—enable           |
| jb_pt_delay      | Size of a fixed jitter buffer, used in fax or modem data transfer mode (ms)   | 0-200                           |
| jb_vo_delay_min  | Size of fixed jitter buffer or lower limit (minimum size) of adaptive jitter buffer (ms)  | 0-200                           |
| jb_vo_delay_max  | Upper limit (maximum size) of adaptive jitter buffer (ms)   | jb_vo_delay_min-200             |
| jb_vo_adaptive   | Use fixed or adaptive jitter buffer operation mode  | 0—fixed<br>1—adaptive           |

|  |  |   |
|--|--|---|
| jb_vo_del_threshold  | <p>Threshold for immediate packet deletion (ms):</p> <ul style="list-style-type: none"> <li>- If call quality is more important than delays, we recommend to set the maximum value for this setting—500ms.</li> <li>- And vice versa, if delays have a priority over the quality, we recommend to set the minimum value for this setting.</li> <li>- It is recommended, that 'Delay threshold' was greater than 'Delay max' for at least of 50ms.</li> </ul> | jb_vo_delay_max-500   |
| jb_vo_del_mode_soft  | Setting defines the method of packet deletion during buffer adjustment to lower limit.   | 0—Hard mode<br>1—Soft mode  |
| t38_bitrate  | Maximum fax transfer rate  | 9600, 14400   |
| t38_datagram   | Maximum datagram size  | 272-512   |
| rfc3264_pt_common  | <p>When performing outgoing call, receive DTMF tones in rfc2833 format with payload type proposed by a communicating gateway; otherwise, tones will be received with the payload type, configured on the gateway.</p> <p>Enables compatibility with gateways that incorrectly handle rfc3264 recommendation</p>  | 0—disable<br>1—enable   |
| <b><i>regexprd—configuration of gateway numbering schedule using regular expressions</i></b> |  |   |
| regex_on   | Configuration of a numbering scheme based on regular expressions   | 0—use dialplan, described in <b>dialplan</b> section<br>1—use numbering scheme based on regular expressions |
| proto  | Signalling protocol  | sip—SIP protocol<br>h323—H.323 protocol (for profile_0 only)  |

|  |   |   |
|--|---|---|
| <p>regex</p>   | <p>regular expression Example: regex: L15 S8 (5xxxx[x#*]@192.168.16.160:5062)</p> | <p>Syntax:<br/>LX SY (Rule), where X—L-timer value, Y—S-timer value.</p> <p>For timer and Rule description, see Section – Prefix—pickup code. Sequence of digits (for example, *8) that, when dialled, allows any subscriber of the group to pickup the call received by another subscriber of the group.</p> <ul style="list-style-type: none"> <li>– Protocol&amp;Target—it's necessary to select a pickup group—PickUp.</li> <li>– PickUp Group—defines the list of groups, that will use this code for the call pickup. Thus, a single code may be used for call pickups in different groups.</li> <li>– To enable this pickup code for all groups, click 'Enable all' button. To disable this pickup code for all groups, click 'Disable all' button.</li> </ul> <p><i>Configuration of Regular Expression Routing Rules</i></p> <p> Enumeration of pickup groups, used in a file, is less by 1 than enumeration, used in web interface!!!</p> |
| <p><i>dialplan—configuration of prefixes for routing and pickup groups</i></p> |   |   |



|                              |  |
|------------------------------|--|
| <p>dialplan_0 to 299</p>     | <p>Format: d1 d2 d3 d4 d5 d6 d7 d8 d9 d10 d11</p> <p>Example: 55 6 0 sip 192.168.16.92 "" 0 0 0 - 0</p> <p>d1—prefix Value: String, 20 characters max.</p> <p>d2—minimum length of a number dialled by the prefix Value: 1-20;</p> <p>d3—dialling timeout for the next digit of a number, in seconds Value: 0-20;</p> <p>d4—signalling protocol, used in prefix operations</p> <p>Value:</p> <p>h323—H.323 protocol operation (for profile_0 only); sip—SIP protocol operation; sip-t—SIP-T protocol operation; pickup—a pickup group.</p> <p>d5—address of a communicating gateway.</p> <p>Value:</p> <ul style="list-style-type: none"> <li>- A.B.C.D or FQDN— in point-to-point operation mode</li> <li>- 'gatekeeper'—when H.323 gatekeeper is used (for profile_0 only)</li> <li>- 'proxy'—when SIP proxy is used</li> </ul> <p>d6—dialling modifier, enables translation of a callee number Modifier is added at the beginning of a dialled number. Value: string, up to 8 digits, in quotation marks.</p> <p>d7—dialling modifier, enables translation of a callee number Defines the number of digits to be deleted from a dialled number for outgoing calls (the most significant digits of a number will be removed.</p> <p>Value: 0..20;</p> <p>d8—CdPN callee number type (for SIPT and H.323)</p> <p>Value:</p> <p>0 – unknown; 1 – subscriber; 2 – national; 3 – international.</p> <p>d9—play 'PBX response' tone when the first prefix digit is dialled. Value: 0—do not play, 1 – play</p> <p>d10—enable routing with a prefix for subscriber ports. Defines the prefix availability for subscriber ports.</p> <p>Value: String, 100 characters max.</p> <p>String formation rules: –portN,..portM or +portN,..portM, where '-' means that access with a prefix is denied for ports, "+" – allowed, portN,..portM—comma-separated list of ports.</p> <p>Example:</p> <p>+0,32—access is allowed for ports 1 and 33.</p> <p>Warning!!! Enumeration of subscriber ports and pickup groups, used in a file, is less by 1 than enumeration, used in web interface and on the device housing!!!</p> <p>d11—defines the preferred packetization time in SIP protocol operation.</p> <p>Value: 0—disable, 10, 20, 30, 40, 50, 60, 70, 80, 90—packetization time.</p> |
| <p>sip_cadences</p>          | <p><b>Non-standard ringing generated by 'alert-info' header processing</b></p>   |
| <p>- sip_cadence_0 .. 15</p> | <p><b>configuration of ringing generation rules</b></p>  |

| <b>Enumeration of rules, used in a file, is less by 1 than enumeration, used in web interface!!!</b>  |   |   |
|---|---|---|
| name  | Signal received in alert-Info header  | For description of these parameters, see Section <b>5.1.2.2.7 Alert-Info distinctive ring</b>   |
| name  | Signal received in alert-Info header  |   |
| <b>ports—configuration of device subscriber ports and subscriber profiles</b>   |   |   |
| <i>port_def 0..7—subscriber profile settings</i>  |   |   |
| Enumeration of subscriber profiles, used in a file, is less by 1 than enumeration, used in web interface!!!<br>Example: 'port_def_2' in a file corresponds to 'profile 3' in WEB interface. |   |   |
| aon   | Caller ID mode  | 0—Caller ID is disabled<br>1—'Russian Caller ID' method<br>2—DTMF Caller ID method<br>3—FSK Caller ID method using bell202 standard<br>4—FSK Caller ID method using ITU-T V.23 standard |
| taxophone   | Payphone mode   | 0—payphone mode is disabled<br>1—polarity reversal<br>2—16kHz meter pulse (for rev. B only)<br>3—12kHz meter pulse (for rev. B only)  |
| category  | SS category   | 0-255   |
| min_flashtime   | Lower limit of Flash impulse duration, ms   | 70-1000   |
| flashtime   | Upper limit of Flash impulse duration, ms   | min_flashtime (no less than 200)-1000   |
| fxoflashtime  | Loop closure time for 'flash' simulation, ms  | 70-1000   |
| fxoringtdm  | Quantity of rings, that will be used by FXO set for loop closure and transmitting the 'PBX response' tone | 2-10  |
| gainr   | Volume of voice reception, x0.1dB   | -230+20   |
| gaint   | Volume of voice transmission, x0.1dB  | -170+60   |
| cfb_pri_over_cw   | Priority between CFB (Forward on busy) and CW (Call wait) services  | 0—CW service has a priority over CFB<br>1—CFB service has a priority over CW  |

|                      |  |   |
|----------------------|--|---|
| aon_hide_name        | Transmission of Caller ID information in Fsk_bell202, Fsk_v23 modes  | 0—information will be sent with a subscriber name<br>1—information will be sent without a subscriber name             |
| aon_hide_date        | Transmission of Caller ID information in Fsk_bell202, Fsk_v23 modes  | 0—Caller ID information will be sent with time and date<br>1—Caller ID information will be sent without time and date |
| playmoh              | 'Music on hold' service  | 0—disable<br>1—enable   |
| usepstncid           | Use CallerID received from the phone line for VoIP direction call  | 0—disable<br>1—enable   |
| pstn_numberprefix    | Prefix added to the number in CallerID   | String, 20 characters max.<br>or ""—parameter is not defined  |
| pstn_nameprefix      | Prefix added to the name in CallerID   | String, 20 characters max.<br>or ""—parameter is not defined  |
| dont_detect_DT       | Detect the 'PBX response' tone before dialling in call direction from IP to FXO, or not  | 0—detect<br>1—do not detect   |
| fxo_delay_dialing    | Dialling delay, when the 'PBX response' tone detection is not used   |   |
| enable_cpc           | Use a short-time break of the subscriber loop on clearback from the opposite subscriber's side   | 0—disable<br>1—enable   |
| cpc_time             | Duration of a short-time break of the subscriber loop, ms  | 200-600   |
| transmit_number      | Transmit the complete number received from IP (from Request URI header of INVITE request) into the line, including FXO unit subscriber number  | 0—disable<br>1—enable   |
| dont_transmit_prefix | Transmit the complete number received from IP (from Request URI header of INVITE request) into the line, except for FXO unit subscriber number | 0—disable<br>1—enable   |
| fxo_call_busy        | Transfer reply 503 instead of reply 486 when the line is busy  | 0—send 486<br>1—send 503  |
| dialing              | Dialling type for the line   | Pulse<br>DTMF tone  |
| min_level_detect     | Minimum level of detectable signals, dBm   | 20-40   |

|                    |   |  |
|--------------------|---|--|
| dial_tone_detect   | 'PBX response' tone detection parameters  | X;Z(A/B/1)<br>X,Y;Z(A/B/1)<br>X,Y;Z(A/B/1+2)<br>X,Y;Z(A/B/2)<br>For detailed information, see Section 5.1.2.4  |
| busy_tone_detect   | 'Busy' tone detection parameters  | X;Z(A/B/1)<br>X,Y;Z(A/B/1)<br>X,Y;Z(A/B/1+2)<br>X,Y;Z(A/B/2)<br>For detailed information, see Section 5.1.2.4  |
| fxodeltdm          | Time interval between digits when dialling into the phone line, ms  | 100-1000   |
| pstn_200_at_answer | Action to be taken when line activity is detected   | 0—do nothing<br>1—Release—detection of polarity reversal is interpreted as a clearback signal (BYE request is transferred via SIP protocol)<br>2—Answer—detection of polarity reversal is interpreted as an answer signal (200 OK response is transferred via SIP protocol)<br>3—ringback tone detection Send a response upon the expiry of timeout pstn_rb_detect_timeout |
| rb_tone_detect     | Ringback tone detection parameters  | X;Z(A/B/1)<br>X,Y;Z(A/B/1)<br>X,Y;Z(A/B/1+2)<br>X,Y;Z(A/B/2)<br>For detailed information, see Section 5.1.2.4  |
| cpc_rus            | Subscriber category; when this setting is enabled, the category will be sent in 'from' field, and 'tel uri' will be used instead of 'sip uri' | 0—disable categories<br>1-10—subscriber category   |
| stop_dial          | '#' button operation  | 0—recognize '#' as DTMF tone<br>1—use '#' to end the dialling  |
| modifier           | Modifier group used by this profile   | 0-15   |

|  |   |  |
|--|---|--|
| pstn_rb_detect_timeout   | Ringback tone detection in the subscriber line<br>Allows to avoid establishment of voice connection in IP networks prior to the answer of a subscriber, or prior to detection of the specific number of rings. If there is no ringback tone within the specified value, it is considered that the callee has responded (reply '200 OK' is sent via SIP), in seconds | 1-60   |
| fxo_detect_line_presence   | Detection of subscriber's line connection to FXO for the line status view in monitoring   | 0—disable<br>1—enable  |
| fxo_block_line_presence  | Block FXO set, if the subscriber's line is not connected to it  | 0—disable<br>1—enable  |
| <i>port_0..31—port 0..31 custom settings</i>   |   |  |
| <p><b>Enumeration of subscriber ports, used in a file, is less by 1 than enumeration, used in web interface and on the device housing!!!</b></p> <p><b>! Example: 'port_0' in a file corresponds to 'port 1' in WEB interface and on the device housing.</b></p> |   |  |
| phone  | Subscriber number   | String, 50 characters max.<br>or ""—parameter is not defined   |
| user_name  | Subscriber name   | String, 50 characters max.<br>or ""—parameter is not defined   |
| auth_name  | Authentication username   | String, 50 characters max.<br>or ""—parameter is not defined   |
| auth_pass  | Authentication password   | String, 50 characters max.<br>or ""—parameter is not defined   |
| hotnumber  | Number that will receive the call when 'Hotline/warmline' service is enabled  | String, 20 digits max.<br>or ""—parameter is not defined   |
| tdmhotnumber   | Number that will receive the call when the 'warmline' service is enabled for calls directed from IP to the phone line   | String, 30 digits max.<br>or ""—parameter is not defined   |
| category   | SS category   | 0-255  |
| custom   | Use port custom settings  | 0—use common settings from common configuration for all ports<br>1—use custom settings specified for this port |

|                 |   |  |
|-----------------|---|--|
| aoon            | Caller ID mode  | <p>0—Caller ID is disabled</p> <p>1—'Russian Caller ID' method</p> <p>2—DTMF Caller ID method</p> <p>3—FSK Caller ID method using bell202 standard</p> <p>4—FSK Caller ID method using ITU-T V.23 standard</p> |
| min_flashtime   | Lower limit of Flash impulse duration (ms)  | 70-1000  |
| flashtime       | Upper limit of Flash impulse duration (ms)  | min_flashtime<br>(no less than 200)1000  |
| fxoflashtime    | Loop closure time for 'flash' simulation, ms  | 70-1000  |
| fxodeltdm       | Time interval between digits when dialling into the phone line, ms  | 100-1000   |
| fxoringtdm      | Quantity of rings, that will be used by FXO set for loop closure and transmitting the 'PBX response' tone | 2-10   |
| gainr           | Volume of voice reception, x0.1dB   | -230+20  |
| gaint           | Volume of voice transmission, x0.1dB  | -170+60  |
| category        | SS category   | 0-255  |
| calltransfer    | 'Call transfer' service   | <p>0—transmit flash to the line with SIP INFO/H.245/Q.931 methods</p> <p>1 – Attended CT</p> <p>2 – Unattended CT</p> <p>3—do not detect flash</p>   |
| callwaiting     | 'Call waiting' service  | <p>0—disable</p> <p>1—enable</p>   |
| cfb_pri_over_cw | Priority between CFB (Forward on busy) and CW (Call wait) services  | <p>0—CW service has a priority over CFB</p> <p>1—CFB service has a priority over CW</p>  |
| aoon_hide_name  | Transmission of Caller ID information in Fsk_bell202, Fsk_v23 modes                                       | <p>0—information will be sent with a subscriber name</p> <p>1—information will be sent without a subscriber name</p>   |
| aoon_hide_date  | Transmission of Caller ID information in Fsk_bell202, Fsk_v23 modes                                       | <p>0—Caller ID information will be sent with time and date</p> <p>1—Caller ID information will be sent without time and date</p>   |

|                      |  |   |
|----------------------|--|---|
| playmoh              | 'Music on hold' service  | 0—disable<br>1—enable   |
| usepstncid           | Use CallerID received from the phone line for VoIP direction call  | 0—disable<br>1—enable   |
| pstn_numberprefix    | Prefix added to the number in CallerID   | String, 20 characters max.<br>or ""—parameter is not defined                                    |
| pstn_nameprefix      | Prefix added to the name in CallerID   | String, 20 characters max.<br>or ""—parameter is not defined                                    |
| dont_detect_DT       | Detect the 'PBX response' tone before dialling in call direction from IP to FXO, or not  | 0—detect<br>1—do not detect   |
| fxo_delay_dialing    | Dialling delay, when the 'PBX response' tone detection is not used   |   |
| enable_cpc           | Use a short-time break of the subscriber loop on clearback from the opposite subscriber's side   | 0—disable<br>1—enable   |
| cpc_time             | Duration of a short-time break of the subscriber loop  | 200-600ms   |
| cpc_rus              | Subscriber Caller ID category  | 0—disable<br>1-10—category  |
| stop_dial            | '#' button operation   | 0—recognize '#' as DTMF tone<br>1—use '#' to end the dialling                                   |
| modifier             | Modifier group used by this profile  | 0-15  |
| port_profile_id      | Subscriber profile number  | 0-7   |
| profile_id           | SIP profile number   | 0-7   |
| transmit_number      | Transmit the complete number received from IP (from Request URI header of INVITE request) into the line, including FXO unit subscriber number  | 0—disable<br>1—enable   |
| dont_transmit_prefix | Transmit the complete number received from IP (from Request URI header of INVITE request) into the line, except for FXO unit subscriber number | 0—disable<br>1—enable   |
| dialing              | Dialling type for the line   | Pulse<br>DTMF tone  |
| min_level_detect     | Minimum level of detectable signals, dBm   | 20-40   |
| dial_tone_detect     | 'PBX response' tone detection parameters   | X;Z(A/B/1)  |
| busy_tone_detect     | 'Busy' tone detection parameters   | X,Y;Z(A/B/1)<br>X,Y;Z(A/B/1+2)<br>X,Y;Z(A/B/2)<br>For detailed information, see Section 5.1.2.4 |

|                       |   |  |
|-----------------------|---|--|
| hotline               | 'Hotline/warmline' service  | 0—disable<br>1—enable                                    |
| hottimeout            | Delay timeout in seconds for the start of the automatic dialling when the 'Warmline' service is enabled.  | 0-300  |
| tdmhotline            | 'Warmline' service that is enabled for calls directed from IP to the phone line   | 0—disable<br>1—enable                                    |
| tdmhottimeout         | Delay timeout in seconds for the start of the automatic dialling when the 'Warmline' service is enabled for calls directed from IP to the phone line. | 0-300  |
| no_offhook_at_ringing | Keep the loop open during ringing from TDM in IP until the voice frequency path is established  | 0—disable (close)<br>1—enable (keep open)                |
| ct_busy               | 'Forward on busy' service (CFB)   | 0—disable<br>1—enable                                    |
| ct_noanswer           | 'Forward on no reply' service (CFNR)  | 0—disable<br>1—enable                                    |
| ct_timeout            | Subscriber response timeout (for 'Call forward on no reply' service)  | 0-300  |
| ct_unconditional      | 'Call forward unconditional' service (CFU)  | 0—disable<br>1—enable                                    |
| ct_outofservice       | 'Forward on out of service' service (CFOOS)   | 0—disable<br>1—enable                                    |
| cfnr_number           | Number, that the call is forwarded to when there is no reply  | String, 20 digits max.<br>or ""—parameter is not defined |
| cfb_number            | Number, that the call is forwarded to when the subscriber is busy   | String, 20 digits max.<br>or ""—parameter is not defined |
| cfu_number            | Number for 'Call forward unconditional'   | String, 20 digits max.<br>or ""—parameter is not defined |
| cfoos_number          | Number, that the call is forwarded to when the subscriber is out of service   | String, 20 digits max.<br>or ""—parameter is not defined |



|                         |   |  |
|-------------------------|---|--|
| pickupgroup             | Include/exclude port to/from the pickup group   | String, 30 characters max., pickup groups that the port belongs to are comma-separated, or "" — parameter is not defined<br> <b>Enumeration of pickup groups, used in a file, is less by 1 than enumeration, used in web interface!!! Example: 'value 0' in a file corresponds to 'group 1' in WEB interface.</b> |
| dvo_dnd_en              | Permission to order supplementary services with the phone unit, DND service   | 0—disable<br>1—enable  |
| dvo_cf_outofservice_en  | Permission to order supplementary services with the phone unit, 'Forward on out of service' service (CFOOS)   | 0—disable<br>1—enable  |
| dvo_cf_noanswer_en      | Permission to order supplementary services with the phone unit, 'Forward on no reply' service (CFNR)  | 0—disable<br>1—enable  |
| dvo_cf_busy_en          | Permission to order supplementary services with the phone unit, 'Forward on busy' service (CFB)   | 0—disable<br>1—enable  |
| dvo_cf_unconditional_en | Permission to order supplementary services with the phone unit, 'Call forward unconditional' service (CFU)  | 0—disable<br>1—enable  |
| dvo_ct_unattended_en    | Permission to order supplementary services with the phone unit, 'Call transfer' service without the wait for response of the subscriber, the call is being forwarded to | 0—disable<br>1—enable  |
| dvo_ct_attended_en      | Permission to order supplementary services with the phone unit, 'Call transfer' service with the wait for response of the subscriber, the call is being forwarded to    | 0—disable<br>1—enable  |
| dvo_callwaiting_en      | Permission to order supplementary services with the phone unit, 'Call waiting' service  | 0—disable<br>1—enable  |
| dnd                     | Restrict all incoming calls, outgoing communication is possible   | 0—disable<br>1—enable  |
| usealtnumber            | Alternative number  | 0—disable<br>1—enable  |
| usealtnumber_as_private | Use an alternative number as a SIP contact  | 0—disable<br>1—enable  |

|                    |   |   |
|--------------------|---|---|
| altnumber          | Alternative subscriber number   | String, 20 digits max.<br>or ""—parameter is not defined  |
| sip_port           | Local UDP port used for port operations via SIP protocol  | 0-65535   |
| stop_dial          | '#' button operation  | 0—recognize '#' as DTMF tone<br>1—use '#' to end the dialling   |
| clir               | Service—calling line identification restriction<br>service—CLIR   | 0—disable<br>1—enable   |
| disabled           | Port status   | 0—port is enabled<br>1—port is disabled   |
| taxophone          | Port operation in payphone mode   | 0—off—port operates in normal mode<br>1—polarity—payphone operation mode with polarity reversal<br>Perform line power polarity reversal on subscriber's response, and return it to original state on clearback.   |
| pstn_200_at_answer | Action for polarity reversal detection  | Release—detection of polarity reversal is interpreted as a clearback signal (BYE request is transferred via SIP protocol)<br>Answer—detection of polarity reversal is interpreted as an answer signal (200 OK response is transferred via SIP protocol) |
| rb_tone_detect     | Ringback tone detection parameters  | X;Z(A/B/1)<br>X,Y;Z(A/B/1)<br>X,Y;Z(A/B/1+2)<br>X,Y;Z(A/B/2)<br>For detailed information, see Section 5.1.2.4   |
| cpc_rus            | Subscriber category; when this setting is enabled, the category will be sent in 'from' field, and 'tel uri' will be used instead of 'sip uri' | 0—disable categories<br>1-10—subscriber category  |
| modifier           | Modifier group used by this profile   | 0-15  |
| mwi_dialtone       | 'Message waiting indicator' service   | 0—disable<br>1—enable   |

|                          |   |                       |
|--------------------------|---|-----------------------|
| psn_rb_detect_timeout    | Ringback tone detection in the subscriber line<br>Allows to avoid establishment of voice connection in IP networks prior to the answer of a subscriber, or prior to detection of the specific number of rings. If there is no ringback tone within the specified value, it is considered that the callee has responded (reply '200 OK' is sent via SIP) | 1-60                  |
| fxo_detect_line_presence | Detection of subscriber's line connection to FXO for the line status view in monitoring   | 0—disable<br>1—enable |
| fxo_block_line_presence  | Block FXO set, if the subscriber's line is not connected to it  | 0—disable<br>1—enable |

Table 16—Device network settings (Network)

| <i>Field name</i>                      | <i>Description</i>  | <i>Values</i>                               |
|--|---|---|
| <b>network—device network settings</b> |   |   |
| ipaddr                                 | Device IP address in WAN network                              | A.B.C.D                                     |
| netmask                                | Net mask for the device location                              | A.B.C.D                                     |
| gateway                                | Default network gateway address                               | A.B.C.D                                     |
| broadcast                              | WAN network broadcasting address                              | A.B.C.D                                     |
| mtu                                    | Maximum transmission unit (WAN)                               | 86-1500                                     |
| autoupdate                             | Enable gateway software and configuration autoupdate          | 0—disable<br>1—enable                       |
| autoupdate_src                         | Autoupdate configuration source                               | no_dhcp<br>dhcp<br>dhcp_vlan1<br>dhcp_vlan2 |
| autoupdate_tftp                        | Autoupdate server address or domain name                      | String, 40 characters max.                  |
| autoupdate_cfg                         | Path to the configuration file                                | String, 40 characters max.                  |
| autoupdate_fw                          | Path to firmware versions file                                | String, 40 characters max.                  |
| autoupdate_proto                       | Autoupdate protocol   | tftp, ftp, http, https                      |
| autoupdate_auth                        | Authentication on autoupdate server                           | 0—disable<br>1—enable                       |
| autoupdate_user                        | Authentication login  | String, 20 characters max.                  |
| autoupdate_pass                        | Authentication password                                       | String, 20 characters max.                  |
| pppoe_vlan                             | Use separate VLAN for PPPoE access                            | 0—disable<br>1—enable                       |
| pppoe_vid                              | VLAN identifier, if there is a separate VLAN for PPPoE access | 1-4095                                      |

|               |  |   |
|---------------|--|---|
| pppoe_mtu     | Maximum transmission unit (PPP)  | 86-1400   |
| dhcpcd        | DHCP usage in WAN network  | 0–disable<br>1–enable   |
| dhcpcd1, 2, 3 | DHCP in VLAN1,2,3 networks   | 0–disable<br>1–enable   |
| vlan1, 2, 3   | VLAN1, 2, 3 usage  | 0–disable<br>1–enable   |
| v1ipaddr      | VLAN1,2,3 interface IP address   | A.B.C.D   |
| v2ipaddr      | Net mask, used for VLAN1,2,3 interface                                 | A.B.C.D   |
| v3ipaddr      | Broadcast address in VLAN1,2,3 interface subnet                        | A.B.C.D   |
| v1netmask     | VLAN 1, 2, 3 identifier  | 1-4095  |
| v2netmask     | 802.1p priority for VLAN1, 2, 3  | 0-7   |
| v3netmask     | VLAN assignment for voice traffic                                      | 0–disable<br>1 – VLAN1<br>2 – VLAN2<br>3 – VLAN3<br>4 – PPPoE |
| v1broadcast   | VLAN destination for SIP/H323 signalling traffic                       | 0–disable<br>1 – VLAN1<br>2 – VLAN2<br>3 – VLAN3<br>4 – PPPoE |
| v2broadcast   | VLAN destination for gateway management via WEB interface, telnet, ssh | 0–disable<br>1 – VLAN1<br>2 – VLAN2<br>3 – VLAN3<br>4 – PPPoE |
| v3broadcast   | DNS server IP address  | A.B.C.D   |
| vid 1,2,3     | Device time synchronization with an external server via NTP            | 0–disable<br>1–enable   |
| v1mtu         | Maximum transmission unit (VLAN 1)                                     | 86-1496   |
| v2mtu         | Maximum transmission unit (VLAN 2)                                     | 86-1496   |
| v3mtu         | Maximum transmission unit (VLAN 3)                                     | 86-1496   |
| cos 1,2,3     | NTP server address   | A.B.C.D   |

|                |   |   |
|----------------|---|---|
| rtp_vlan       | RTP transfer interface  | 0—use the main interface for RTP transfer<br>1-3—use VLAN interface for RTP transfer<br>4—use PPPOE interface for RTP transfer                      |
| sig_vlan       | Signalling transfer interface   | 0—use the main interface for signalling transfer<br>1-3—use VLAN interface for signalling transfer<br>4—use PPPOE interface for signalling transfer |
| ctl_vlan       | Management interface  | 0—use the main interface for management<br>1-3—use VLAN interface for management<br>4—use PPPOE interface for management                            |
| rtp_vlan       | Device access via Telnet protocol                                     | 0—disable<br>1—enable   |
| sig_vlan       | Device access via SSH protocol  | 0—disable<br>1—enable   |
| STP_EN         | STP protocol  | 0—disable<br>1—enable   |
| SNMP           | SNMP protocol   | 0—disable<br>1—enable   |
| dhcp_gw        | Obtain default gateway network address in WAN network via DHCP        | 0—disable<br>1—enable   |
| dhcp_gw1, 2, 3 | Obtain default gateway network address in VLAN1,2,3 networks via DHCP | 0—disable<br>1—enable   |
| ntpen          | NTP protocol  | 0—disable<br>1—enable   |
| ntpip          | NTP server IP address   | A.B.C.D   |

|   |   |   |
|---|---|---|
| ntp_interval  | NTP server synchronization period   | 0—disable<br>30–100000—use with the defined period in seconds |
| zoneinfo  | Timezone  | <b>for permitted values, see Appendix L</b>                   |
| dst_enable  | Daylight saving change  | 0—disable<br>1—enable   |
| dst_start   | Daylight saving change date and time  | String, 50 characters max.                                    |
| dst_end   | Daylight saving change set back date and time   | String, 50 characters max.                                    |
| dst_offset  | DST offset, in minutes  | 0-720   |
| cfg_interval  | Configuration file version check interval   | 1 - 99999   |
| fw_interval   | Software version check interval   | 1 - 99999   |
| dnsip   | Primary DNS server IP address   | A.B.C.D   |
| reserved_dnsip  | Secondary DNS server IP address   | A.B.C.D   |
| telnet_port   | TCP port for TELNET protocol operation  | 1-65535, 23 by default  |
| ssh_port  | TCP port for SSH protocol operation   | 1-65535, 22 by default  |
| web_port  | WEB server port number for HTTP protocol operation  | 80 by default   |
| https_port  | WEB server port number for HTTPS protocol operation   | 1-65535, 443 by default                                       |
| web_en  | Device access via web interface   | 0—disable<br>1—enable   |
| radius_enable   | Use RADIUS server for authentication of users administering the device via WEB, telnet, SSH | 0—disable<br>1—enable   |
| radius_server   | RADIUS server address   | A.B.C.D   |
| radius_secret   | Password to access the RADIUS server  | String, 50 characters max.                                    |
| radius_retry  | Number of retries during the access to RADIUS server  | 0-10  |
| use_vendor_info   | Use alternative value of DHCP Option 60   | 0—disable<br>1—enable   |
| vendor_info   | DHCP Option 60 alternative value  | String, 255 characters max.                                   |
| language  | Web configurator language   | en—English<br>ru—Russian                                      |
| <b>TR-069—configuration of TR-069 monitoring and management protocol settings</b> |   |   |
| Enable  | TR-069 device management process  | 0—disable<br>1—enable   |

|                            |   |   |
|----------------------------|---|---|
| URL                        | ACS server address  | http://<address>:<port>, where <address>—ACS server IP address or domain name, <port>—ACS server port, 10301 by default |
| Username                   | Username used by client to access the ACS server  | String, 50 characters max.  |
| Password                   | Password used by client to access the ACS server  | String, 50 characters max.  |
| PeriodicInformEnable       | ACS server periodical polling performed by the integrated TR-069 client at intervals equal to 'Periodic inform interval' value, in seconds. Goal of the polling is to identify possible changes in the device configuration | 0—disable<br>1—enable   |
| PeriodicInformInterval     | ACS server polling interval, in seconds   | 0-65535   |
| ConnectionRequestURL       | Parameter is not used, value should be blank  |   |
| ConnectionRequestUsername  | Username for ACS server access to TR-069 client Server sends ConnectionRequest notifications  | String, 50 characters max.  |
| ConnectionRequestPassword  | User password for ACS server access to TR-069 client Server sends ConnectionRequest notifications   | String, 50 characters max.  |
| NATMode                    | TR-069 client operation mode in the presence of NAT; identifies the method, that will be used by client for obtaining its public address information  | STUN<br>Manual<br>Off<br>See description on page 36.  |
| NATAddress                 | STUN server IP address or domain name   |   |
| STUNEnable                 | use STUN protocol for public address identification   | 0—disable<br>1—enable   |
| STUNServerAddress          | STUN server IP address or domain name   |   |
| STUNServerPort             | STUN server UDP port  | 1-65535, 3478 by default  |
| STUNMinimumKeepAlivePeriod | The time interval in seconds for periodic transmission of messages to STUN server for public address discovery and modification, in seconds   | 0-100000  |

|   |   |  |
|---|---|--|
| STUNMaximumKeepAlivePeriod              | The time interval in seconds for periodic transmission of messages to STUN server for public address discovery and modification, in seconds | 0-100000   |
| <b>snmp—snmp protocol configuration</b> |   |  |
| agentproto                              | Transport protocol  | udp  |
| agentport                               | Transport port used by agent  | 0-65535  |
| sys_object_id                           | Device OID  | String, 40 characters max.   |
| sys_name                                | Device system name  | String, 20 characters max.   |
| sys_location                            | Device location   | String, 20 characters max.   |
| sys_contact                             | Device manufacturer contact information   | String, 20 characters max.   |
| trap_sink                               | Trap receiver IP address  | (server manager or proxy agent)<br>A.B.C.D   |
| trap_type                               | SNMP protocol version   | v1<br>v2   |
| trap_community                          | Password contained in trap messages   | String, 20 characters max.   |
| rocommunity                             | Password for parameter reading (common: public)   | String, 20 characters max.   |
| rwcommunity                             | Password for parameter writing (common: private)  | String, 20 characters max.   |
| <b>snmp_users</b>                       | <b>SNMPv3 user configuration</b>  |  |
| user_0                                  | SNMPv3 user   | Login, password, access mode are written comma-separated in one string<br><br>Access mode:<br><br>- rw—read/write<br>- ro—read |

Table 17—Switch port settings (Switch)

| <i>Field name</i>                                      | <i>Description</i>                    | <i>Values</i>         |
|--|---------------------------------------|-----------------------|
| <b>vlan—example of switch configuration using VLAN</b> |                                       |                       |
| hubmode  | Ethernet switch operation in hub mode | 0—disable<br>1—enable |



|  |   |  |
|--|---|--|
| <p>Port mapping:</p> <p>0—GE0 (GE2) port</p> <p>1—GE1 (GE1) port</p> <p>2—GE2 (GE0) port</p> <p>3—CPU port (CPU)</p> <p>4—SFP0 port (SFP0)</p> <p>5—SFP1 port (SFP1)</p> |   |  |
| portmask0..5   | Mutual availability of data ports Defines the port that will receive the data from this port. | <p>A B C D E F, where</p> <p>A—port 0</p> <p>B—port 1</p> <p>C—port 2</p> <p>D—port 3</p> <p>E—port 4</p> <p>F—port 5</p> <p>A, B, C, D, E, and F may take the following values:</p> <p>0—data transmission to port is disabled</p> <p>1—data transmission to port is enabled</p>  |
| enable0..5   | Use 'Default VLAN ID', 'Override' and 'Egress' settings on ports 0..5                         | <p>0—disable</p> <p>1—enable</p>   |
| vid0..5  | Default VLAN ID   | 1-4095   |
| im0..5   | IEEE mode for ports 0-5   | <p>0 – fallback</p> <p>1 – check</p> <p>2 – secure</p>   |
| eg0..5   | Packet transfer rules for ports 0..5  | <p>0—unmodified—packets will be sent by the port without any changes</p> <p>1—untagged—packets will always be sent without VLAN tag by this port</p> <p>2—tagged—packets will always be sent with VLAN tag by this port</p> <p>3—double tag—each packet will be sent with two VLAN tags—if received packet was tagged and came with one VLAN tag—if the received packet was untagged</p> |

|  |   |  |
|--|---|--|
| ov0..5   | Override VLAN ID—when checked, it is considered that any received packet has a VID, defined in <i>'default VLAN ID'</i> row | 0—disable<br>1—enable  |
| portmode0..5   | Data transfer and port duplex mode Ports 3..5 values should always be set to 'auto'   | auto—automatic determination of speed and duplex<br>10f, 10h, 100f, 100h, 1000f—possible values for speed and duplex configuration |
| backup_port0..5  | Slave port for operation in direction reservation mode  | port0..5   |
| preemption0..5   | Return to the master port, if it is operational<br>Works in direction reservation mode                                      | on—enable return to the master port<br>off—stay on the slave port  |
| <b>vtu—configuration of packet routing rules for switch operation in 802.1q mode (VTU Table)</b> |   |  |
| vtu0 to vtu15  | VTU rules   |  |
| vtu0.vid   | VLAN ID   | 1-4095   |
| vtu0.port0   | Port operation mode 0   | 0 – unmodified<br>1 – untagged<br>2 – tagged<br>3 – not member   |
| vtu0.port1   | Port operation mode 1   |  |
| vtu0.port2   | Port operation mode 2   |  |
| vtu0.cpu   | Port operation mode 3   |  |
| vtu0.sfp0  | Port operation mode 4   |  |
| vtu0.sfp1  | Port operation mode 5   |  |
| vtu0.override  | VLAN priority override  | 0—disable<br>1—enable  |
| vtu0.priority  | VLAN priority   | 0-7  |
| <b>qos—Quality of Service functions and bandwidth restrictions</b>                               |   |  |

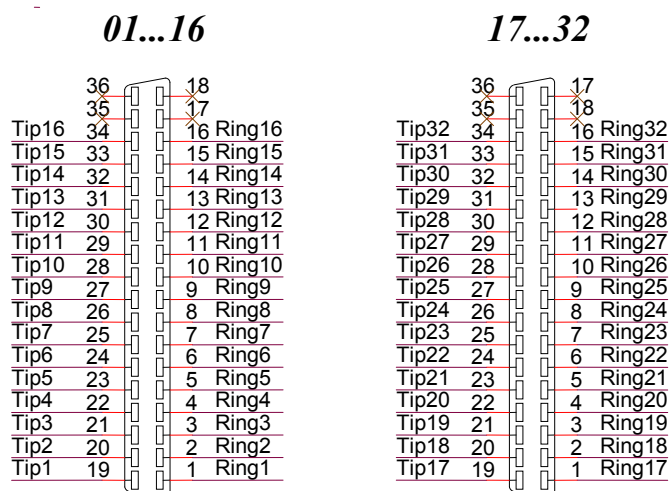
|  |  |   |
|--|--|---|
| <p>ieee_pri</p>  | <p>Distribution of packets into queues depending on the 802.1p priority</p> <p>Example: ieee_pri: 0xfa41 = 1111 1010 0100 0001.</p> <p>Packets with priorities 7 and 6 are placed into queue 3, with priorities 5 and 4—into queue 2, with priorities 1 and 2—into queue 0.</p>  | <p>0xDCBA</p> <p>A-D—hex numbers.</p> <p>D—2 high bits—queue for priority: 7, low for priority: 6</p> <p>C—2 high bits—queue for priority: 5, low for priority: 4</p> <p>B—2 high bits—queue for priority: 3, low for priority: 2</p> <p>A—2 high bits—queue for priority: 1, low for priority: 0</p> <p>00—queue 0</p> <p>01—queue 1</p> <p>10—queue 2</p> <p>11—queue 3</p> |
| <p><i>diffserv_remap—distribution of packets into queues depending on the IP diffserv priority</i></p> |  |   |
| <p>diffserv_remap003C_mask</p>   | <p>0xHGFEDCBA, where</p> <p>A—2 high bits—queue for priority: 0x3C, low for: 0x38</p> <p>G—2 high bits—queue for priority: 0x34, low for: 0x30</p> <p>F—2 high bits—queue for priority: 0x2C, low for: 0x28</p> <p>E—2 high bits—queue for priority: 0x24, low for: 0x20</p> <p>D—2 high bits—queue for priority: 0x1C, low for: 0x18</p> <p>C—2 high bits—queue for priority: 0x14, low for: 0x10</p> <p>B—2 high bits—queue for priority: 0x0C, low for: 0x08</p> <p>A—2 high bits—queue for priority: 0x04, low for: 0x00</p> <p>00—queue 0, 01—queue 1, 10—queue 2, 11—queue 3</p> |   |
| <p>diffserv_remap407C_mask</p>   | <p>0xHGFEDCBA, where</p> <p>H—2 high bits—queue for priority: 0x7C, low for: 0x78</p> <p>G—2 high bits—queue for priority: 0x74, low for: 0x70</p> <p>F—2 high bits—queue for priority: 0x6C, low for: 0x68</p> <p>E—2 high bits—queue for priority: 0x64, low for: 0x60</p> <p>D—2 high bits—queue for priority: 0x5C, low for: 0x58</p> <p>C—2 high bits—queue for priority: 0x54, low for: 0x50</p> <p>B—2 high bits—queue for priority: 0x4C, low for: 0x48</p> <p>A—2 high bits—queue for priority: 0x44, low for: 0x40</p> <p>00—queue 0, 01—queue 1, 10—queue 2, 11—queue 3</p> |   |

|                                |  |   |
|--------------------------------|--|---|
| <p>diffserv_remap80BC_mask</p> | <p>0xHGFEDCBA, where</p> <p>H—2 high bits—queue for priority: 0xBC, low for: 0xB8</p> <p>G—2 high bits—queue for priority: 0xB4, low for: 0xB0</p> <p>F—2 high bits—queue for priority: 0xAC, low for: 0xA8</p> <p>E—2 high bits—queue for priority: 0xA4, low for: 0xA0</p> <p>D—2 high bits—queue for priority: 0x9C, low for: 0x98</p> <p>C—2 high bits—queue for priority: 0x94, low for: 0x90</p> <p>B—2 high bits—queue for priority: 0x8C, low for: 0x88</p> <p>A—2 high bits—queue for priority: 0x84, low for: 0x80</p> <p>00—queue 0, 01—queue 1, 10—queue 2, 11—queue 3</p> |   |
| <p>diffserv_remapC0FC_mask</p> | <p>0xHGFEDCBA, where</p> <p>H—2 high bits—queue for priority: 0xFC, low for: 0xF8</p> <p>G—2 high bits—queue for priority: 0xF4, low for: 0xF0</p> <p>F—2 high bits—queue for priority: 0xEC, low for: 0xE8</p> <p>E—2 high bits—queue for priority: 0xE4, low for: 0xE0</p> <p>D—2 high bits—queue for priority: 0xDC, low for: 0xD8</p> <p>C—2 high bits—queue for priority: 0xD4, low for: 0xD0</p> <p>B—2 high bits—queue for priority: 0xCC, low for: 0xC8</p> <p>A—2 high bits—queue for priority: 0xC4, low for: 0xC0</p> <p>00—queue 0, 01—queue 1, 10—queue 2, 11—queue 3</p> |   |
| <p>tag_remap_mask0..5</p>      | <p>Remap 802.1p priorities for untagged packets</p>  | <p>0xHGFEDCBA, where</p> <p>H corresponds to packets with priority 7, A—with priority 0</p> <p>A-H—assigned priority, permitted value range 0-7</p> |
| <p>prio0..5</p>                | <p>802.1p priority assigned to untagged packets, received by this port and sent as tagged form the egress port.</p>  | <p>0-7</p>  |

|                        |   |   |
|------------------------|---|---|
| qos_mode0..5           | QoS operation modes   | <p>0—distribute packets into queues based on IP diffserv priority only</p> <p>1—distribute packets into queues based on 802.1p priority only</p> <p>2—distribute packets into queues based on IP diffserv and 802.1p priorities, if both priorities are present in the packet, IP diffserv priority is used for queuing purposes</p> <p>3—distribute packets into queues based on IP diffserv and 802.1p priorities, if both priorities are present in the packet, 802.1p priority is used for queuing purposes</p> |
| ingress_limit_mode0..5 | Restriction mode for traffic coming to the port                             | <p>0—no restriction</p> <p>1—restrict all traffic</p> <p>2—multicast, broadcast, and flooded unicast traffic will be restricted</p> <p>3—multicast and broadcast traffic will be restricted</p> <p>4—only broadcast traffic will be restricted</p>  |
| ingress_rate0..5       | Bandwidth restriction for traffic incoming to port<br>0-5 for queue 0, kbps | 70-250000   |

|  |  |  |
|--|--|--|
| <p>ingress_mask0..5</p>                        | <p>Bandwidth restriction for traffic incoming to port<br/>0-5 for queues 1-3, kbps<br/>rate0—band for queue 0<br/>rate1—band for queue 1<br/>rate2—band for queue 2<br/>rate3—band for queue 3</p> | <p>0x0 – rate3= rate2= rate1= rate0<br/>0x1 – rate3= rate2= rate1=2*rate0<br/>0x2 – rate1= rate0, rate3=<br/>rate2=2*rate1<br/>0x3 – rate1=2*rate0, rate3=<br/>rate2=2*rate1<br/>0x4 – rate2= rate1=rate0,<br/>rate3=2*rate2<br/>0x5 – rate2=rate1=2*rate0, rate3=<br/>=2*rate2<br/>0x6 – rate1= rate0, rate2=2*rate1,<br/>rate3=2*rate2<br/>0x7 – rate1=2*rate0,<br/>rate2=2*rate1, rate3=2*rate2</p> |
| <p>egress_rate0..5</p>                         | <p>Bandwidth restriction for traffic outgoing from<br/>the port, kbps</p>  | <p>70-250000</p>   |
| <p><b>lldp—LLDP protocol configuration</b></p> |  |  |
| <p>enable</p>                                  | <p>LLDP protocol</p>   | <p>0—disable<br/>1—enable</p>  |
| <p>tx_interval</p>                             | <p>LLDP message transmission period (seconds)</p>  | <p>0..65535</p>  |

## APPENDIX A. TAU-32M.IP NETWORK TERMINAL CONTACT PIN ASSIGNMENT



Ring[X] and Tip[X] contacts are designed for the phone unit connection.

*Wire colour and terminal contact correspondence table (NENSHI NSPC-7019-18 cable)*

| Wire colour         | Terminal contact | Wire colour          | Terminal contact |
|---------------------|------------------|----------------------|------------------|
| <b>White-blue</b>   | 1                | <b>Black-blue</b>    | 10               |
| Blue                | 19               | Blue                 | 28               |
| <b>White-orange</b> | 2                | <b>Black-orange</b>  | 11               |
| Orange              | 20               | Orange               | 29               |
| <b>White-green</b>  | 3                | <b>Black-green</b>   | 12               |
| Green               | 21               | Green                | 30               |
| <b>White-brown</b>  | 4                | <b>Black-brown</b>   | 13               |
| Brown               | 22               | Brown                | 31               |
| <b>Purple</b>       | 5                | <b>Yellow-blue</b>   | 14               |
| Grey                | 23               | Blue                 | 32               |
| <b>Red-blue</b>     | 6                | <b>Yellow-orange</b> | 15               |
| Blue                | 24               | Orange               | 33               |
| <b>Red-orange</b>   | 7                | <b>Yellow-green</b>  | 16               |
| Orange              | 25               | Green                | 34               |
| <b>Red-green</b>    | 8                | <b>Yellow-brown</b>  | 17               |
| Green               | 26               | Brown                | 35               |
| <b>Red-brown</b>    | 9                | <b>Yellow-grey</b>   | 18               |
| Brown               | 27               | Grey                 | 36               |

*Wire colour and E1 Line terminal contact correspondence table (HANDIAN UTP 18PR cable)*

| Wire colour         | Terminal contact | Wire colour          | Terminal contact |
|---------------------|------------------|----------------------|------------------|
| <b>White-blue</b>   | 1                | <b>Red-grey</b>      | 10               |
| Blue                | 19               | Grey                 | 28               |
| <b>White-orange</b> | 2                | <b>Black-blue</b>    | 11               |
| Orange              | 20               | Blue                 | 29               |
| <b>White-green</b>  | 3                | <b>Black-orange</b>  | 12               |
| Green               | 21               | Orange               | 30               |
| <b>White-brown</b>  | 4                | <b>Black-green</b>   | 13               |
| Brown               | 22               | Green                | 31               |
| <b>Purple-grey</b>  | 5                | <b>Black-brown</b>   | 14               |
| Grey                | 23               | Brown                | 32               |
| <b>Red-blue</b>     | 6                | <b>Black-grey</b>    | 15               |
| Blue                | 24               | Grey                 | 33               |
| <b>Red-orange</b>   | 7                | <b>Yellow-blue</b>   | 16               |
| Orange              | 25               | Blue                 | 34               |
| <b>Red-green</b>    | 8                | <b>Yellow-orange</b> | 17               |
| Green               | 26               | Orange               | 35               |
| <b>Red-brown</b>    | 9                | <b>Yellow-green</b>  | 18               |
| Brown               | 27               | Green                | 36               |



## APPENDIX B. ALTERNATIVE FIRMWARE UPDATE METHOD

When you cannot update the firmware via web interface or the console (telnet, RS-232), you may use an alternative firmware update method via RS-232.

To update the device firmware, you will need the following programs:

- Terminal program (for example: TERATERM);
- TFTP server program.

Firmware update procedure:

1. Connect to Ethernet port of the device.
2. Connect PC console port to the device console port using a crossed cable.
3. Run the terminal application.
4. Configure data rate: 115200, data format: 8bit w/o parity, 1 stop bit, w/o flow control.
5. Run TFTP server program and specify the path to 'chagall' folder. In this folder, create '300' subfolder, and place firmware.elf, initrd.300, zImage.300 in it (computer that runs TFTP server and the device should be located in a single network).
6. Turn the device on and stop the startup sequence by entering `stop` command in the terminal program window:

```
U-Boot 1.1.6 (Nov 13 2010 - 16:24:39) Mindspeed 0.06.2-candidate1

DRAM: 128 MB
Comcerto Flash Subsystem Initialization
found am29gl512 flash at B8000000
Flash: 64 MB
NAND: 64 MiB
In: serial
Out: serial
Err: serial
Reserve MSP memory
Net:  comcerto_gemac0: config phy 0, speed 1000, duplex full
      comcerto_gemac1: config phy 1, speed 1000, duplex full
      comcerto_gemac0, comcerto_gemac1
Write 'stop' to stop autoboot (3 sec)..
FXS-32>>
```

7. Enter `set ipaddr {device ip address} <ENTER>` (for example: `set ipaddr 192.168.16.112`).
8. Enter `set netmask {device network mask} <ENTER>` (for example: `set netmask 255.255.255.0`).
9. Enter `set serverip {IP address of a computer, that runs TFTP server} <ENTER>` (for example: `set serverip 192.168.16.44`).

10. To activate the network interface, execute `mii i` <ENTER> command:

```
=> mii i
Init switch 0: ..Ok!
Init switch 1: ..Ok!
Init phy 1: ..Ok!
Init phy 2: ..Ok!
=>
```

11. To update linux kernel, use `run updatecsp` command:

```
FXS-32>> run updatecsp
Using comcerto_gemac0 device
TFTP from server 192.168.16.44; our IP address is 192.168.16.112
Filename 'chagall/300/zImage.300'.
Load address: 0x1000000
Loading:
#####
#####
#####
#####
done
Bytes transferred = 1130944 (1141c0 hex)
Erase Flash Sectors 11-23 in Bank # 2
Erasing 13 sectors... .....ok
Copy to Flash... .....ok
done
FXS-32>>
```

12. To update the media processor firmware, use `run updatemsp` command:

```
FXS-32>> run updatemsp
Using comcerto_gemac0 device
TFTP from server 192.168.16.44; our IP address is 192.168.16.112
Filename 'chagall/300/firmware.elf'.
Load address: 0x1000000
Loading: #####
#####
#####
#####
#####
#####
#####
done
Bytes transferred = 1809497 (1b9c59 hex)
Erase Flash Sectors 24-55 in Bank # 2
```

```
Erasing 32 sectors... .....ok
Copy to Flash... .....ok
done
FXS-32>>
```

**13. To update the file system, use run updateefs command:**

```
FXS-32>> run updateefs
Using comcerto_gemac0 device
TFTP from server 192.168.16.44; our IP address is 192.168.16.112
Filename 'chagall/300/initrd.300'.
Load address: 0x1000000
Loading: #####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
#####
done
Bytes transferred = 3759224 (395c78 hex)
Erase Flash Sectors 56-183 in Bank # 2
Erasing 128 sectors... .....ok
Copy to Flash... .....ok
done
FXS-32>>
```

**14. Start up the device using 'run bootcmd' command.**

## APPENDIX C. GENERAL DEVICE SETUP / CONFIGURATION PROCEDURE

- 1 Using Ethernet cable, connect gateway Ethernet port to your local area network.
- 2 Device configuration is performed via WEB interface (see Paragraph 5.1 of this manual) using a web browser (e.g. Internet Explorer, Mozilla Firefox, Opera, Google Chrome). Initial connection to the gateway is performed by IP address, specified by the manufacturer (TAU-32M.IP factory default IP address—192.168.1.2, network mask—255.255.255.0).

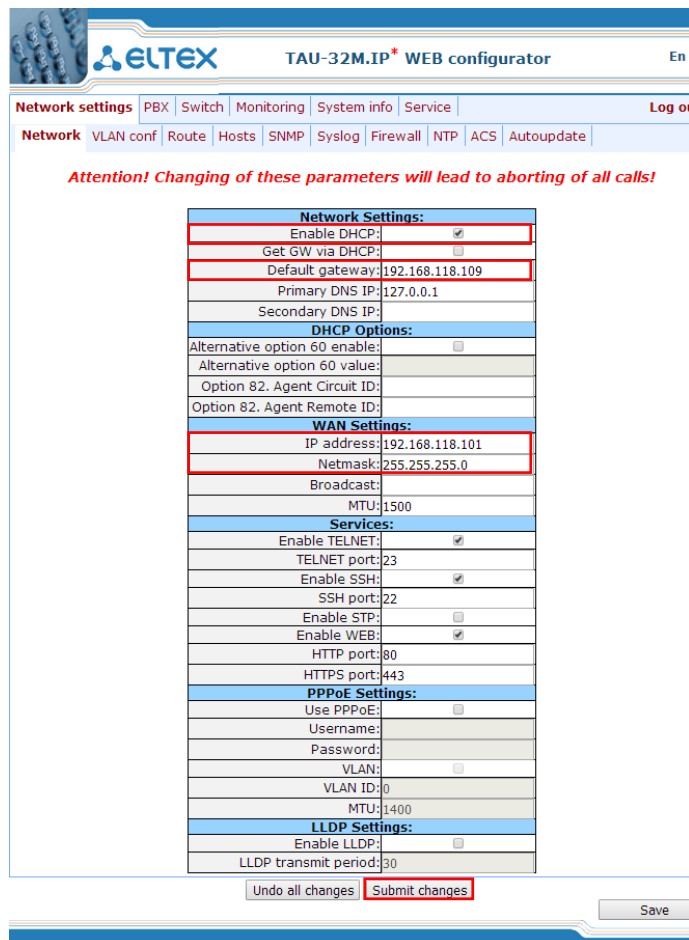
In WEB configurator, specify the following settings in 'Network settings -> Network' menu section:

- Device IP address corresponding to the established addressing in your network—'IP address' field.
- Subnet mask—'Netmask' field.
- Network gateway address—'Default gateway'.

Or you can use TAU-32M.IP as a DHCP server client in order to obtain IP address automatically: 'Network settings -> Network' menu section, select 'Use DHCP' checkbox.



**Make sure to apply changes with 'Submit Changes' button, located in the bottom of the page.**



The screenshot shows the 'TAU-32M.IP\* WEB configurator' interface. The 'Network settings' menu is selected, and the 'Network' sub-menu is active. A warning message states: 'Attention! Changing of these parameters will lead to aborting of all calls!'. The configuration table is as follows:

| Network Settings:             |                                     |
|-------------------------------|-------------------------------------|
| Enable DHCP:                  | <input checked="" type="checkbox"/> |
| Get GW via DHCP:              | <input type="checkbox"/>            |
| Default gateway:              | 192.168.118.109                     |
| Primary DNS IP:               | 127.0.0.1                           |
| Secondary DNS IP:             |                                     |
| DHCP Options:                 |                                     |
| Alternative option 60 enable: | <input type="checkbox"/>            |
| Alternative option 60 value:  |                                     |
| Option 82. Agent Circuit ID:  |                                     |
| Option 82. Agent Remote ID:   |                                     |
| WAN Settings:                 |                                     |
| IP address:                   | 192.168.118.101                     |
| Netmask:                      | 255.255.255.0                       |
| Broadcast:                    |                                     |
| MTU:                          | 1500                                |
| Services:                     |                                     |
| Enable TELNET:                | <input checked="" type="checkbox"/> |
| TELNET port:                  | 23                                  |
| Enable SSH:                   | <input checked="" type="checkbox"/> |
| SSH port:                     | 22                                  |
| Enable STP:                   | <input type="checkbox"/>            |
| Enable WEB:                   | <input checked="" type="checkbox"/> |
| HTTP port:                    | 80                                  |
| HTTPS port:                   | 443                                 |
| PPPoE Settings:               |                                     |
| Use PPPoE:                    | <input type="checkbox"/>            |
| Username:                     |                                     |
| Password:                     |                                     |
| VLAN:                         | <input type="checkbox"/>            |
| VLAN ID:                      | 0                                   |
| MTU:                          | 1400                                |
| LLDP Settings:                |                                     |
| Enable LLDP:                  | <input type="checkbox"/>            |
| LLDP transmit period:         | 30                                  |

At the bottom of the form, there are three buttons: 'Undo all changes', 'Submit changes', and 'Save'.

- 3 We highly recommend changing default password after device installation in 'Service -> Password'

menu section:

- When the respective protocol (SIP/H.323) is used in 'PBX -> SIP/H323 Profiles -> SIP Common' and 'PBX -> SIP/H323 Profiles -> H323' menu sections, you should activate operation via these protocols by selecting 'Enable SIP', 'Enable H323' checkboxes.

- During SIP protocol operations (PBX -> SIP/H323 Profiles -> Profile n), you have to configure SIP/H323 profile (by default, Profile 1 is defined for all subscriber ports). You may use up to 8 different profiles.

In 'PBX -> SIP/H323 Profiles -> Profile n -> SIP Custom' tab, perform the following settings:

- To be able to register device ports on the registration server, you should define a reservation, tcp in 'Proxy mode' menu item.

- Define the SIP proxy server address in 'Proxy' field, and registration server address in 'Registrar' field. As a rule, a single device is used as a SIP proxy and registration server; in this case, SIP-proxy server (Proxy) address is the same as for the registration server (Registrar).
  - To enable port authorization, you should set the following value for 'Authentication' parameter: 'global' or 'user defined'.
- a) When 'global' value is used, all ports will be authorized with the same name and password; in this case, authorization global name and password should be specified in 'Username' and 'Password' fields respectively.

The screenshot shows the 'SIP configuration' section of the TAU-32M.IP Rev2 WEB configurator. The 'Proxy mode' is set to 'parking'. The 'Proxy / Registrar / Use registration 1' fields are both set to '192.168.18.168'. The 'Authentication' dropdown is set to 'global', and the 'Username' field contains 'TAU-32.IP'. The 'Password' field is masked with asterisks. The 'Submit Changes' button at the bottom is highlighted with a red box.

- b) When 'user defined' value is used, each port will be authorized with its own name and password, in this case authorization name and password should be specified in 'PBX -> Ports -> Edit -> Custom' section, 'Authentication name' and 'Authentication password'

fields respectively.

ELTEX TAU-32M.IP Rev2 WEB configurator

Network settings: PBX | Switch | Monitoring | System info | Service | Log Out

Main | SIP/H323 Profiles | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups | Distinctive Ring

SIP Common | H323 | Profile 1 | Profile 2 | Profile 3 | Profile 4 | Profile 5 | Profile 6 | Profile 7 | Profile 8

SIP Custom | Codecs | Dialplan

**Attention!!! Changing of these parameters will lead to aborting of all calls!!!**

**SIP configuration:**

Proxy mode: parking

Proxy / Registrar / Use registration 1: 192.168.18.168 192.168.18.168

Proxy / Registrar / Use registration 2:

Proxy / Registrar / Use registration 3:

Proxy / Registrar / Use registration 4:

Proxy / Registrar / Use registration 5:

Home server test: invite

Keepalive time (s): 60

SIP-Domain:

Use domain to Register:

Registration Retry Interval (s): 30

Inbound:

Outbound: off

Dial timeout: 10

Expires: 1800

Authentication: user defined

Username: TAU-32.IP

Password: \*\*\*\*\*

Ringback at answer 183:

Ringback at callwaiting: 180 Ringing

Remote ringback: ringback with 183 Progress

DTMF MIME Type: application/dtmf-relay

Hook flash MIME Type: application/hook-flash

Escape hash uri:

User=Phone:

Remove inactive media:

P-RTP-Stat:

CT with replaces:

100rel: supported

Enable timer:

Min SE: 120

Session expires (0 - unlimited session): 0

**NAT settings:**

NAT Keep Alive Msg: off

NAT Keep Alive Interval (s): 30

**Conference settings:**

Conference mode: Local

Conference server: conf

**IMS settings:**

Enable IMS:

XCAP name for three-party conference: three-party-conference

XCAP name for hotline: hot-line-service

XCAP name for call waiting: call-waiting

XCAP name for call hold: call-hold

Undo All Changes | Re-registration | Defaults | **Submit Changes**

Save

Network settings: PBX | Switch | Monitoring | System info | Service | Log Out

Main | SIP/H323 Profiles | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups

**Attention!!! Changing of SIP port parameter will lead to aborting of all calls!!!**

1-8 | 9-16 | 17-24 | 25-32 | Subscriber profiles

Custom | Common | Call forward | Suppl. Service | Groups | Pickup

**Port 1**

Phone: 500000

User name:

Use alt.number:

Alt.number: 222

Authentication name: 520000

Authentication password: \*\*\*\*\*

Custom:

Subscriber profile: Profile 1

SIP/H323 profile: Profile 1

Hot line:

Hot timeout: 0

Hot number: 700001

CLIR:

DND:

Stop dial at #:

Disabled:

SIP port:

Process flash: Transmit flash

Call waiting:

**Apply** | Cancel | Default

Save

- When gateway operates through the Gatekeeper via H.323 protocol, in 'PBX -> SIP/H323 Profiles -> H.323' menu section, select the 'Gatekeeper used' checkbox and define IP address in 'GateKeeper address' field. H.323 protocol operation is possible only in Profile 1.

Network settings **PBX** Switch Monitoring System info Service Log Out

Main **SIP/H323 Profiles** TCP/IP Ports Call limits Suppl. Service Codes Serial groups FXO groups Pickup groups

SIP Common **H323** Profile 1 Profile 2 Profile 3 Profile 4 Profile 5 Profile 6 Profile 7 Profile 8

**Attention!!! Changing of these parameters will lead to aborting of all calls!!!**

| H323 settings:      |                                     |
|---------------------|-------------------------------------|
| Enable H323:        | <input checked="" type="checkbox"/> |
| Enable H.235:       | <input type="checkbox"/>            |
| Ignore GCF info:    | <input type="checkbox"/>            |
| Disable faststart:  | <input type="checkbox"/>            |
| Disable tunneling:  | <input checked="" type="checkbox"/> |
| Gatekeeper used:    | <input checked="" type="checkbox"/> |
| Is gateway:         | <input type="checkbox"/>            |
| Time To Live:       | 300                                 |
| Keep Alive Time:    | 60                                  |
| H323 alias:         | tau72ip3                            |
| GateKeeper address: | 192.168.18.27                       |
| H.235 Password:     | *****                               |
| DTMF Transfer:      | 3 - Q.931 Keypad IE                 |
| Bearer capability:  | 3.1 kHz Audio                       |

- To enable device authorization on the Gatekeeper via H.235 protocol, in 'PBX -> SIP/H323 Profiles -> H.323' menu section, select the 'Enable H.235' checkbox and specify the name and password in 'H.323 alias' and 'H.235 Password' fields respectively.

Network settings **PBX** Switch Monitoring System info Service Log Out

Main **SIP/H323 Profiles** TCP/IP Ports Call limits Suppl. Service Codes Serial groups FXO groups Pickup groups

SIP Common **H323** Profile 1 Profile 2 Profile 3 Profile 4 Profile 5 Profile 6 Profile 7 Profile 8

**Attention!!! Changing of these parameters will lead to aborting of all calls!!!**

| H323 settings:      |                                     |
|---------------------|-------------------------------------|
| Enable H323:        | <input checked="" type="checkbox"/> |
| Enable H.235:       | <input checked="" type="checkbox"/> |
| Ignore GCF info:    | <input type="checkbox"/>            |
| Disable faststart:  | <input type="checkbox"/>            |
| Disable tunneling:  | <input checked="" type="checkbox"/> |
| Gatekeeper used:    | <input checked="" type="checkbox"/> |
| Is gateway:         | <input type="checkbox"/>            |
| Time To Live:       | 300                                 |
| Keep Alive Time:    | 60                                  |
| H323 alias:         | tau32ip3                            |
| GateKeeper address: | 192.168.18.27                       |
| H.235 Password:     | *****                               |
| DTMF Transfer:      | 3 - Q.931 Keypad IE                 |
| Bearer capability:  | 3.1 kHz Audio                       |



- 8 In 'PBX -> SIP/H323 Profiles -> Profile n -> Codecs' section, select utilized codecs and define their selection priority. During H.323 protocol operation, all settings should be configured in Profile 1.

Network settings **PBX** | Switch | Monitoring | System info | Service | **Log Out**

Main | **SIP/H323 Profiles** | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups

SIP Common | H323 | **Profile 1** | Profile 2 | Profile 3 | Profile 4 | Profile 5 | Profile 6 | Profile 7 | Profile 8

SIP Custom | **Codecs** | Dialplan

*Attention!!! Changing of these parameters will lead to aborting of all calls!!!*

**Codecs configuration:**

List of codecs in preferred order:

|              |                                     |
|--------------|-------------------------------------|
| Use G.711A   | <input checked="" type="checkbox"/> |
| Use G.711U   | <input checked="" type="checkbox"/> |
| Use G.723    | <input checked="" type="checkbox"/> |
| Use G.729B   | <input checked="" type="checkbox"/> |
| Use G.726-32 | <input checked="" type="checkbox"/> |
| Use G.729A   | <input type="checkbox"/>            |

**Packet coder time:**

G.711 Ptime: 20 ms

G.729 Ptime: 20 ms

G.723 Ptime: 30 ms

G.726-32 Ptime: 20 ms

G.726-32 PT: 102

**Features:**

DTMF Transfer: INFO

Flash Transfer: INFO

Fax Detect Direction: Caller and Callee

Fax Transfer Codec: T.38 mode

Slave Fax Transfer Codec: G.711U

Modem Transfer: G.711A NSE

rfc2833 PT: 101

Silence suppression:

Echo canceller:

NLP disable:

Comfort noise:

**RTCP Configuration:**

RTCP timer:  5

RTCP control period:

**Cisco NSE Configuration:**

NSE PT: 100

**T.38 Configuration:**

Max Datagram Size: 512

Bitrate: 14400

**Jitter buffer Configuration:**

**Modem/Fax pass-thru:**

Delay: 200 ms

**Voice:**

Mode: Adaptive

Delay min: 0 ms

Delay max: 200 ms

Deletion threshold: 500 ms

Deletion mode: Hard

9 In 'PBX -> Ports' section, assign phone numbers to device ports.

**Attention!!! Changing of SIP port parameter will lead to aborting of all calls!!!**

| Port | Type | Phone  | User name | Custom                   | Process flash  | Subscriber profile | SIP/H323 profile | Disabled                 | Edit |
|------|------|--------|-----------|--------------------------|----------------|--------------------|------------------|--------------------------|------|
| 1    | FXO  | 500000 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |
| 2    | FXO  | 700001 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |
| 3    | FXO  | 700002 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |
| 4    | FXO  | 700003 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |
| 5    | FXO  | 700004 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |
| 6    | FXO  | 700005 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |
| 7    | FXO  | 700006 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |
| 8    | FXO  | 700007 |           | <input type="checkbox"/> | Transmit flash | Profile 1          | Profile 1        | <input type="checkbox"/> |      |

Buttons: Undo All Changes, Auto numeration, Submit Changes, Save

10 In subscriber port settings ('PBX -> Ports -> Edit -> Custom'), specify an active SIP profile number in 'SIP/H323 profile' (by default, Profile 1 is defined for all subscriber ports).

**Attention!!! Changing of SIP port parameter will lead to aborting of all calls!!!**

Port 1

Phone: 200301

User name:

Use alt.number:

Alt.number:

Authentication name: admin

Authentication password:

Custom:

Subscriber profile: Profile 1

**SIP/H323 profile: Profile 1**

Hot line:

Hot timeout: 0

Hot number:

CLIR:

DND:

Stop dial at #:

Disabled:

SIP port:

Process flash: Attended caltransfer

Call waiting:

Buttons: Apply, Cancel, Default, Save

- 11 Configure addressed dial peers ('PBX -> SIP/H323 Profiles -> Profile *n* -> Dialplan' menu section).  
During H.323 protocol operation, all settings should be configured in Profile 1.



Network settings **PBX** | Switch | Monitoring | System info | Service | **Log Out**

Main | **SIP/H323 Profiles** | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups

SIP Common | H323 | **Profile 1** | Profile 2 | Profile 3 | Profile 4 | Profile 5 | Profile 6 | Profile 7 | Profile 8

SIP Custom | Codecs | **Dialplan**

Regular Expression Dialplan

Protocol: SIP

Expressions:

```

I15 88
(2xxxxx@192.168.16.150|5xxxxx@192.168.18.97|7xxxxx@192.168.16.67|9xx@192.168.18.98
|*xx#|#xx#|#xx#|*xx*x+#)

```

Undo All Changes | Show help | Submit Changes

Save

- 12 When basic parameters are configured, click 'Save' button to save changes into the non-volatile memory of the device.

For additional information on configuration (parameters, fields), see device operation manual.

## APPENDIX D. EXAMPLE OF SWITCH CONFIGURATION USING VLAN

Objective: Tagged traffic comes to the switch port 0 with the following tags: 101, 102 and 103. Packets with VLAN ID=101 should be sent untagged to port 1, packets with VLAN ID=102 should be sent tagged to port 2. VLAN 103 is proposed to be used for telephony and device management, i.e. packets with VLAN ID=103 should be sent untagged to the switch CPU port.

- Using Ethernet cable, connect gateway Ethernet port to your local area network. Connect to the device using WEB configurator.
- Define the packet routing rules—'VTU table'—in 'Switch -> 802.1q.' submenu. For VLAN 101, port 0 is tagged, port 1 is untagged, other ports are not members of this VLAN. For VLAN 102, port 0 is tagged, port 2 is untagged, other ports are not members of this VLAN. For VLAN 103, port 0 is tagged, CPU port is untagged, other ports are not members of this VLAN (see Section 5.1.3.2).

| VID          | Port 0     | Port 1     | Port 2     | CPU        | SFP 0      | SFP 1      | Override                            | Priority |
|--------------|------------|------------|------------|------------|------------|------------|-------------------------------------|----------|
|              | unmodified | unmodified | unmodified | unmodified | unmodified | unmodified | <input type="checkbox"/>            | 0        |
| Add new rule |            |            |            |            |            |            |                                     |          |
| VTU table    |            |            |            |            |            |            |                                     |          |
| VID          | Port 0     | Port 1     | Port 2     | CPU        | SFP 0      | SFP 1      | Override                            | Priority |
| 101          | untagged   | untagged   | unmodified | unmodified | unmodified | unmodified | <input checked="" type="checkbox"/> | 0        |
| 102          | tagged     | unmodified | untagged   | unmodified | unmodified | unmodified | <input checked="" type="checkbox"/> | 0        |
| 103          | tagged     | unmodified | unmodified | untagged   | unmodified | unmodified | <input checked="" type="checkbox"/> | 0        |

Remove selected

- For switch ports, you should configure 'VTU table' operation mode in 'Switch -> Switch ports settings' submenu, i.e. 'IEEE Mode = Secure'. For untagged traffic coming to ports 1, 2, and CPU to be transferred to port 0 tagged, you should configure the respective Default VLAN ID tags—101, 102, and 103—for ports 1, 2, and CPU. Also, select 'Enable VLAN' checkboxes for these ports, that allow to use 'Default VLAN ID' settings (see Section 5.1.3.1).

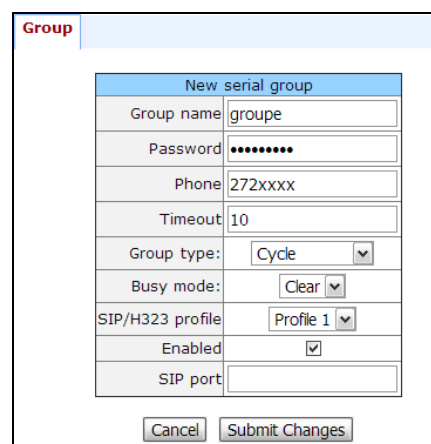
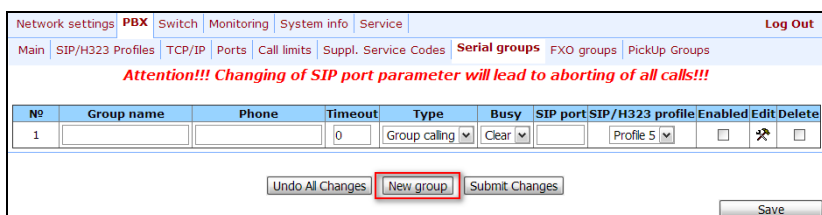
|   | Port 0   | Port 1   | Port 2   | CPU   | SFP 0   | SFP 1   |
|---|--|--|--|---|---|---|
| Speed/Duplex:   | auto   | auto   | auto   |   |   |   |
| Enable VLAN:  | <input checked="" type="checkbox"/>  | <input checked="" type="checkbox"/>  | <input checked="" type="checkbox"/>  | <input checked="" type="checkbox"/>   | <input type="checkbox"/>  | <input type="checkbox"/>  |
| Default VLAN ID:  | 0  | 101  | 102  | 103   | 0   | 0   |
| Egress:   | Unmodified   | Unmodified   | Unmodified   | Unmodified  | Unmodified  | Unmodified  |
| Override:   | <input type="checkbox"/>   | <input type="checkbox"/>   | <input type="checkbox"/>   | <input type="checkbox"/>  | <input type="checkbox"/>  | <input type="checkbox"/>  |
| IEEE mode:  | Secure   | Secure   | Secure   | Secure  | Secure  | Secure  |
| Output:   | <input checked="" type="checkbox"/> to Port 1<br><input checked="" type="checkbox"/> to Port 2<br><input checked="" type="checkbox"/> to CPU<br><input checked="" type="checkbox"/> to SFP 0<br><input checked="" type="checkbox"/> to SFP 1 | <input checked="" type="checkbox"/> to Port 0<br><input checked="" type="checkbox"/> to Port 2<br><input checked="" type="checkbox"/> to CPU<br><input checked="" type="checkbox"/> to SFP 0<br><input checked="" type="checkbox"/> to SFP 1 | <input checked="" type="checkbox"/> to Port 0<br><input checked="" type="checkbox"/> to Port 1<br><input checked="" type="checkbox"/> to CPU<br><input checked="" type="checkbox"/> to SFP 0<br><input checked="" type="checkbox"/> to SFP 1 | <input checked="" type="checkbox"/> to Port 0<br><input checked="" type="checkbox"/> to Port 1<br><input checked="" type="checkbox"/> to Port 2<br><input checked="" type="checkbox"/> to SFP 0<br><input checked="" type="checkbox"/> to SFP 1 | <input checked="" type="checkbox"/> to Port 0<br><input checked="" type="checkbox"/> to Port 1<br><input checked="" type="checkbox"/> to Port 2<br><input checked="" type="checkbox"/> to CPU<br><input checked="" type="checkbox"/> to SFP 1 | <input checked="" type="checkbox"/> to Port 0<br><input checked="" type="checkbox"/> to Port 1<br><input checked="" type="checkbox"/> to Port 2<br><input checked="" type="checkbox"/> to CPU<br><input checked="" type="checkbox"/> to SFP 0 |
| Backup port:  | none   | none   | none   |   | none  | none  |
| Preemption:   | <input type="checkbox"/>   | <input type="checkbox"/>   | <input type="checkbox"/>   | <input type="checkbox"/>  | <input type="checkbox"/>  | <input type="checkbox"/>  |
| <input checked="" type="checkbox"/> disable learning (hub mode)   |  |  |  |   |   |   |
| <input type="button" value="Undo all changes"/> <input type="button" value="Submit changes"/> <input type="button" value="Defaults"/> |  |  |  |   |   |   |

- Click 'Update switch' button to apply settings. Connect to the device using 103 VLAN and confirm applied settings with 'Commit' button. After that, modified switch settings could be saved in the non-volatile memory with 'Save' button.

## APPENDIX E. EXAMPLE OF PABX CONFIGURATION WITH TAU-32M.IP

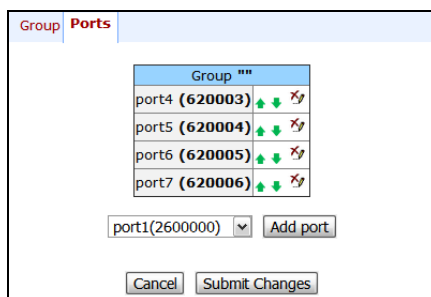
*Objective:* You have to build PABX with 4 subscriber numbers. A single number is allocated to PABX by a local exchange network—272xxxx. When a call comes to this number, it should be transferred to all 4 x PABX subscriber ports in turns. Ringing time for each number is 10 seconds.

- 1 Using Ethernet cable, connect gateway Ethernet port to your local area network. Connect to the device using WEB configurator.
- 2 As a rule, during the call group creation process at SIP server, only a single login/password is issued for multiple lines. At the gateway, you should create a cycle call group with 10 seconds timeout; to do this, click 'New group' button in 'PBX -> Serial groups' tab and fill in the required fields:



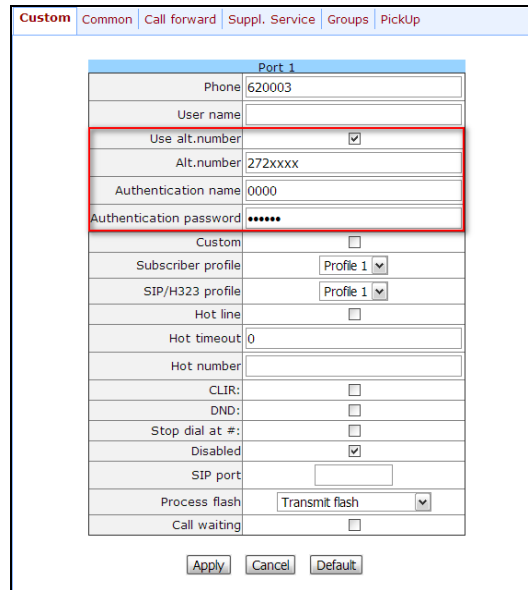
In group settings, specify login/password for registration on SIP server and assign the number allocated by a local exchange network (272xxxx) as a group number. Define SIP/H.323 profile for call group operation.

- 3 In group port settings ('PBX -> Serial groups -> Edit'), add ports into a call group (see Section 5.1.2.7 Serial groups).



- 4 In subscriber port settings—PBX -> PORTS -> Edit -> Custom tab, define the internal subscriber enumeration. Given that during outgoing calls a number 272xxxx should be transferred as a Caller ID, you should configure an alternative Caller ID. Enumeration is defined by the 'Phone' parameter in the

port settings, and an alternative Caller ID is configured by selecting 'Use alt.number' checkbox and specifying an external number in 'Alt.number' field. Also, in port settings, define login/password for authentication on SIP server.



| Port 1                  |                                     |
|-------------------------|-------------------------------------|
| Phone                   | 620003                              |
| User name               |                                     |
| Use alt.number          | <input checked="" type="checkbox"/> |
| Alt.number              | 272xxxx                             |
| Authentication name     | 0000                                |
| Authentication password | *****                               |
| Custom                  | <input type="checkbox"/>            |
| Subscriber profile      | Profile 1                           |
| SIP/H323 profile        | Profile 1                           |
| Hot line                | <input type="checkbox"/>            |
| Hot timeout             | 0                                   |
| Hot number              |                                     |
| CLIR:                   | <input type="checkbox"/>            |
| DND:                    | <input type="checkbox"/>            |
| Stop dial at #:         | <input type="checkbox"/>            |
| Disabled                | <input checked="" type="checkbox"/> |
| SIP port                |                                     |
| Process flash           | Transmit flash                      |
| Call waiting            | <input type="checkbox"/>            |

- Next, configure SIP/H.323 profile assigned to the call group (PBX -> SIP/H323 Profiles -> Profile n -> SIP Custom). Define SIP server address and allow registration and authentication on SIP server.

Network settings **PBX** | Switch | Monitoring | System info | Service | **Log Out**

Main | **SIP/H323 Profiles** | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups | Distinctive Ring

SIP Common | H323 | **Profile 1** | Profile 2 | Profile 3 | Profile 4 | Profile 5 | Profile 6 | Profile 7 | Profile 8

SIP Custom | Codecs | Dialplan

**Attention!!! Changing of these parameters will lead to aborting of all calls!!!**

| SIP configuration:                       |   |
|--|---|
| Proxy mode:                              | parking   |
| Proxy / Registrar / Use registration 1:  | 192.168.18.52   192.168.18.52   <input checked="" type="checkbox"/> |
| Proxy / Registrar / Use registration 2:  | <input type="checkbox"/>  |
| Proxy / Registrar / Use registration 3:  | <input type="checkbox"/>  |
| Proxy / Registrar / Use registration 4:  | <input type="checkbox"/>  |
| Proxy / Registrar / Use registration 5:  | <input type="checkbox"/>  |
| Home server test:                        | invite  |
| Keepalive time (s):                      | 60  |
| SIP-Domain:                              | radiususer  |
| Use domain to Register:                  | <input type="checkbox"/>  |
| Registration Retry Interval (s):         | 30  |
| Inbound:                                 | <input type="checkbox"/>  |
| Outbound:                                | off   |
| Dial timeout:                            | 10  |
| Expires:                                 | 1800  |
| Authentication:                          | user defined  |
| Username:                                | TAU-72.IP   |
| Password:                                | *****   |
| Ringback at answer 183:                  | <input type="checkbox"/>  |
| Ringback at callwaiting:                 | 180 Ringing   |
| Remote ringback:                         | ringback with 183 Progress  |
| DTMF MIME Type:                          | application/dtmf-relay  |
| Hook flash MIME Type:                    | application/hook-flash  |
| Escape hash uri:                         | <input type="checkbox"/>  |
| User=Phone:                              | <input type="checkbox"/>  |
| Remove inactive media:                   | <input type="checkbox"/>  |
| P-RTP-Stat:                              | <input type="checkbox"/>  |
| CT with replaces:                        | <input checked="" type="checkbox"/>                                 |
| 100rel:                                  | supported   |
| Enable timer:                            | <input checked="" type="checkbox"/>                                 |
| Min SE:                                  | 120   |
| Session expires (0 - unlimited session): | 0   |
| NAT settings:                            |   |
| NAT Keep Alive Msg:                      | off   |
| NAT Keep Alive Interval (s):             | 30  |
| Conference settings:                     |   |
| Conference mode:                         | Local   |
| Conference server:                       | conf  |
| IMS settings:                            |   |
| Enable IMS:                              | <input type="checkbox"/>  |
| XCAP name for three-party conference:    | three-party-conference  |
| XCAP name for hotline:                   | hot-line-service  |
| XCAP name for call waiting:              | call-waiting  |
| XCAP name for call hold:                 | call-hold   |

Undo All Changes | Re-registration | Defaults | Submit Changes

Save

- 6 For outgoing calls routing, configure addressed dial peers in the respective SIP/H.323 profile ('PBX -> SIP-H323 Profiles -> Profile n -> Dialplan' menu section).

Network settings **PBX** | Switch | Monitoring | System info | Service | **Log Out**

Main | **SIP/H323 Profiles** | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups

SIP Common | H323 | **Profile 1** | Profile 2 | Profile 3 | Profile 4 | Profile 5 | Profile 6 | Profile 7 | Profile 8

SIP Custom | Codecs | **Dialplan**

Regular Expression Dialplan

Protocol: SIP

Expressions:

```

115 88
(2xxxxx@192.168.16.150|5xxxxx@192.168.18.97|7xxxxx@192.168.16.67|9xx@192.168.18.98
|*xx#|#xx#|#xx#|*xx#|*xx*x+#)

```

Undo All Changes | Show help | Submit Changes

Save

Or you may use the *outbound* mode (configured in 'PBX -> SIP/H323 Profiles -> Profile n -> SIP Custom' section); in this case, all outgoing calls will be routed via SIP-proxy.

Network settings **PBX** | Switch | Monitoring | System info | Service Log Out

Main | **SIP/H323 Profiles** | TCP/IP | Ports | Call limits | Suppl. Service Codes | Serial groups | FXO groups | Pickup groups | Distinctive Ring

SIP Common | H323 | **Profile 1** | Profile 2 | Profile 3 | Profile 4 | Profile 5 | Profile 6 | Profile 7 | Profile 8

SIP Custom | Codecs | Dialplan

**Attention!!! Changing of these parameters will lead to aborting of all calls!!!**

| SIP configuration:                       |   |
|--|---|
| Proxy mode:                              | parking   |
| Proxy / Registrar / Use registration 1:  | 192.168.18.52 192.168.18.52 <input checked="" type="checkbox"/> |
| Proxy / Registrar / Use registration 2:  | <input type="checkbox"/>  |
| Proxy / Registrar / Use registration 3:  | <input type="checkbox"/>  |
| Proxy / Registrar / Use registration 4:  | <input type="checkbox"/>  |
| Proxy / Registrar / Use registration 5:  | <input type="checkbox"/>  |
| Home server test:                        | invite  |
| Keepalive time (s):                      | 60  |
| SIP-Domain:                              | radiususer  |
| Use domain to Register:                  | <input type="checkbox"/>  |
| Registration Retry Interval (s):         | 30  |
| Inbound:                                 | <input type="checkbox"/>  |
| <b>Outbound:</b>                         | <b>on</b>   |
| Dial timeout:                            | 10  |
| Expires:                                 | 1800  |
| Authentication:                          | user defined  |
| Username:                                | TAU-72.IP   |
| Password:                                | *****   |
| Ringback at answer 183:                  | <input type="checkbox"/>  |
| Ringback at callwaiting:                 | 180 Ringing   |
| Remote ringback:                         | ringback with 183 Progress                                      |
| DTMF MIME Type:                          | application/dtmf-relay  |
| Hook flash MIME Type:                    | application/hook-flash  |
| Escape hash uri:                         | <input type="checkbox"/>  |
| User=Phone:                              | <input type="checkbox"/>  |
| Remove inactive media:                   | <input type="checkbox"/>  |
| P-RTP-Stat:                              | <input type="checkbox"/>  |
| CT with replaces:                        | <input checked="" type="checkbox"/>                             |
| 100rel:                                  | supported   |
| Enable timer:                            | <input checked="" type="checkbox"/>                             |
| Min SE:                                  | 120   |
| Session expires (0 - unlimited session): | 0   |
| NAT settings:                            |   |
| NAT Keep Alive Msg:                      | off   |
| NAT Keep Alive Interval (s):             | 30  |
| Conference settings:                     |   |
| Conference mode:                         | Local   |
| Conference server:                       | conf  |
| IMS settings:                            |   |
| Enable IMS:                              | <input type="checkbox"/>  |
| XCAP name for three-party conference:    | three-party-conference  |
| XCAP name for hotline:                   | hot-line-service  |
| XCAP name for call waiting:              | cal-waiting   |
| XCAP name for call hold:                 | cal-hold  |



---

## APPENDIX F. CONFIGURATION EXAMPLE FOR PABX CONNECTED VIA FXO LINES

*Objective:* PABX is connected to a local exchange network using 4 subscriber lines (4 FXO sets are used on TAU-32M). You have to implement an access to a local exchange network: a) when dialling number of one of the lines, b) automatically, when the phone goes offhook; at that, the first free line will be taken.

Conditions: Subscriber sets that have TAU-323M FXO lines connected to them, may receive DTMF multi-frequency dialling. 'PBX response' and 'busy' tones transmitted by PABX to a local exchange network are non-standard single-frequency 600Hz signals.

- 1 Connect the gateway Ethernet port to a local area network using Ethernet cable. Connect to the device using WEB configurator.
- 2 Configure a subscriber profile (for example, Profile 2) for FXO lines, connected to a local exchange network (PBX -> PORTS -> Subscriber profiles -> Profile2).

Network settings **PBX** Switch Monitoring System info Service Log Out

Main SIP/H323 Profiles TCP/IP **Ports** Call limits Suppl. Service Codes Serial groups FXO groups PickUp groups Distinctive Ring

**Attention!!! Changing of SIP port parameter will lead to aborting of all calls!!!**

1-8 9-16 17-24 25-32 **Subscriber profiles**

Profile 1 **Profile 2** Profile 3 Profile 4 Profile 5 Profile 6 Profile 7 Profile 8

| Profile 2   |                                     |
|---|-------------------------------------|
| CallerID:   | off                                 |
| Hide date:  | <input type="checkbox"/>            |
| Hide name:  | <input type="checkbox"/>            |
| Min Flashtime (ms):   | 200                                 |
| Max Flashtime (ms):   | 600                                 |
| Gain receive (0.1 dB):  | -70                                 |
| Gain transmit (0.1 dB):   | 0                                   |
| Category:   | off                                 |
| CFB has priority over CW:   | <input type="checkbox"/>            |
| Play music on hold:   | <input type="checkbox"/>            |
| Taxophone:  | off                                 |
| CPC:  | <input type="checkbox"/>            |
| CPC time (ms):  | 200                                 |
| FXO parameters  |                                     |
| Outgoing direction parameters   |                                     |
| Flashtime:  | 300                                 |
| Dialtone detection:   | <input checked="" type="checkbox"/> |
| Dialtone time detect (s):   | 5                                   |
| Dialing delay (s):  | 2                                   |
| Don't transmit prefix:  | <input type="checkbox"/>            |
| Transmit number:  | <input type="checkbox"/>            |
| 503 Service unavailable on busy (SIP):                                      | <input checked="" type="checkbox"/> |
| PSTN activity:  | PSTN reversal polarity detection    |
| PSTN reversal polarity detection:   | Release                             |
| Dialing   |                                     |
| Method:   | DTMF                                |
| Interdigit delay:   | 800                                 |
| Pulse time (ms):  | 80                                  |
| Pause time (ms):  | 80                                  |
| Incoming direction parameters   |                                     |
| Ring detection:   | 2                                   |
| PSTN number prefix:   |                                     |
| PSTN name prefix:   |                                     |
| Use PSTN CallerID:  | <input checked="" type="checkbox"/> |
| Detect FXO line presence:   | <input type="checkbox"/>            |
| Block FXO line in outgoing direction:                                       | <input type="checkbox"/>            |
| Tone detect parameters  |                                     |
| Minimum level of detectable signal (dBm):                                   | -36                                 |
| Dialtone detection parameters:  | 600;0(2000/0/1)                     |
| Busytone detection parameters:  | 600;1(350/350/1)                    |
| Ringback tone detection parameters:   | 425;0(1000/4000/1)                  |
| Show tone detect parameters format  |                                     |
| <input type="button" value="Apply"/> <input type="button" value="Default"/> |                                     |

Required parameters:

*For incoming communication with a local exchange network*

For incoming communication, it's necessary to detect 'ringing' tone in the line in order to engage the set. As a rule, two rings are sufficient for tone recognition; at that, reception of a local exchange network subscriber Caller ID, that is sent after the first 'ringing', is guaranteed: 'Ring detection' parameter equals to 2. If PABX sends Caller ID after the second ring, this parameter should be increased by 1; at that, the connection establishment time will increase.

To send a local exchange network subscriber Caller ID to access point subscriber, select 'Use PSTN CallerID' checkbox; otherwise, FXO line number will be sent to the access point subscriber as a Caller ID.

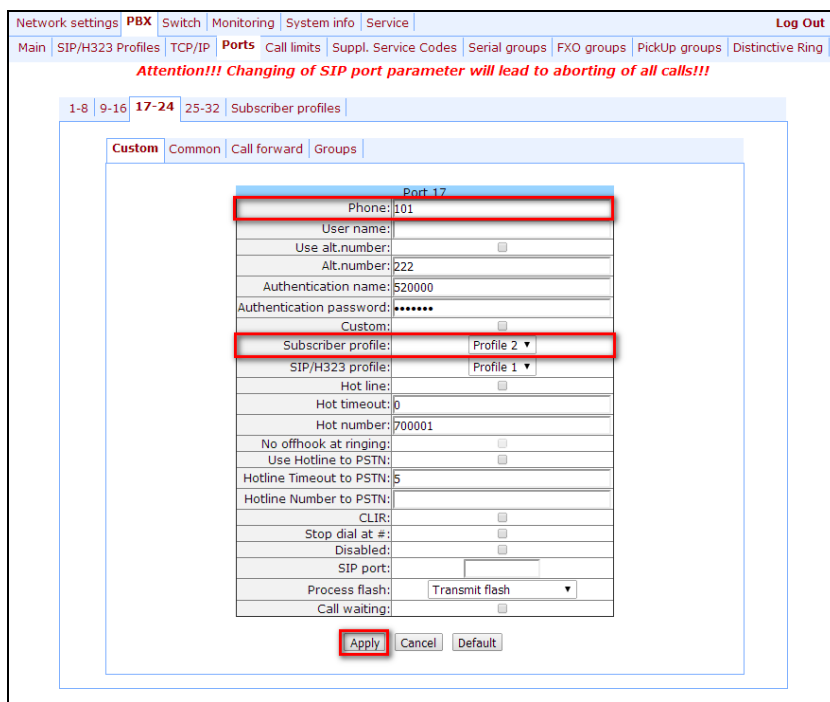
*For outgoing communication with a local exchange network*

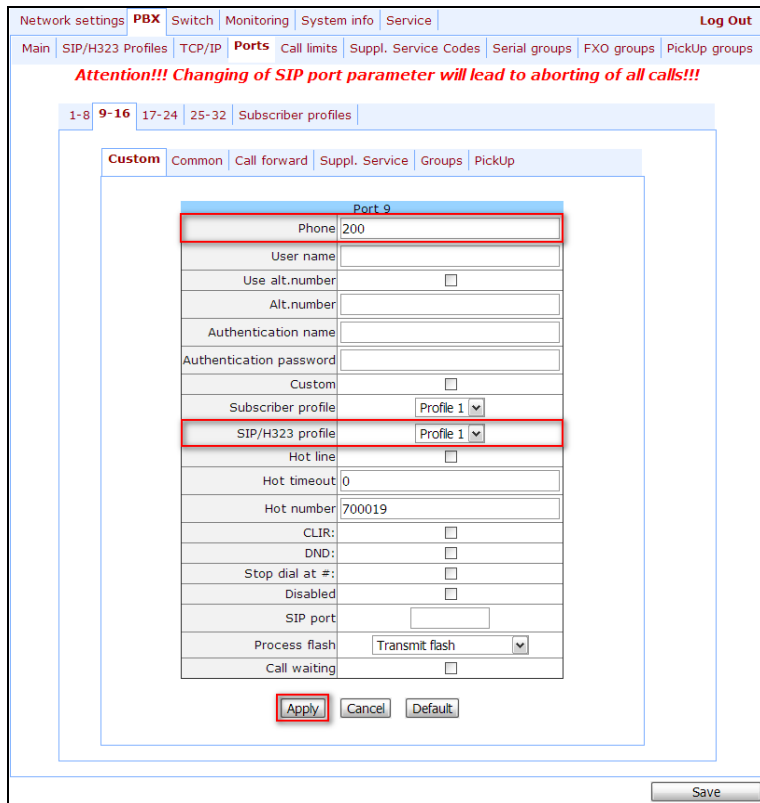
Upon FXO line engagement, 'PBX response' tone sent by a subscriber PABX set must be detected. If there is no tone, the set may be faulty. Given that PABX sends non-standard 600Hz signals according to the objectives, select '*Dialtone detection*' checkbox and configure '*Dial tone detection parameters*': 600;0(2000/0/1). Also, to have the ability to detect a 'busy' tone upon a local exchange network subscriber clearback, set the following value for '*Busy tone detection parameters*': 600;1(350/350/1). If a 'busy' tone is not detected, FXO line will be freed only after the TAU-32M subscriber clearback.

Click 'Apply' button to apply settings.

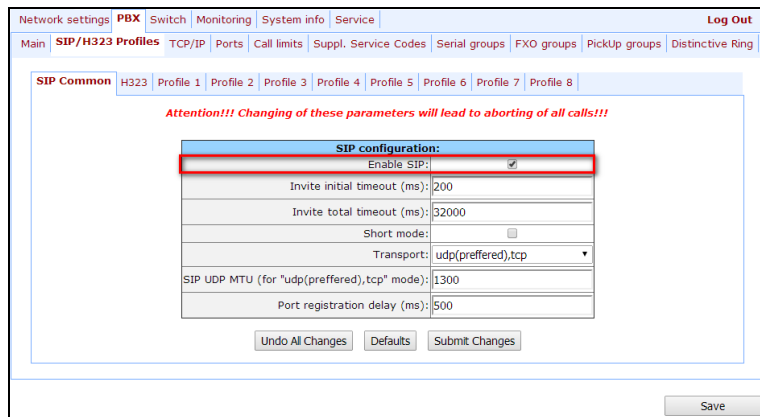
### 3 Configure subscriber ports and FXO lines (*PBX -> Ports -> Edit -> Custom*).

For 4 ports—FXO sets, connected to a local exchange network, assign subscriber line numbers—'*Phone*' parameter (for example, 101-104) and assign the configured Profile 2: '*Subscriber profile*' parameter takes '*profile 2*' value. For subscriber ports—FXS sets, configure subscriber numbers (for example, 200-2XX.) For all ports, configure '*SIP/H323 profile*' with the 1st number.



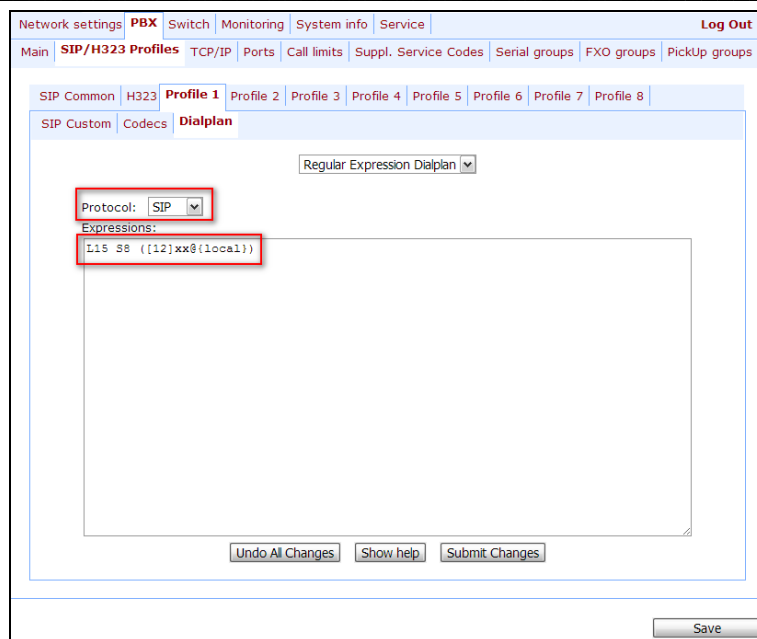


- 4 Make sure, that SIP is enabled in SIP settings (PBX -> SIP/H323 profiles -> SIP common):



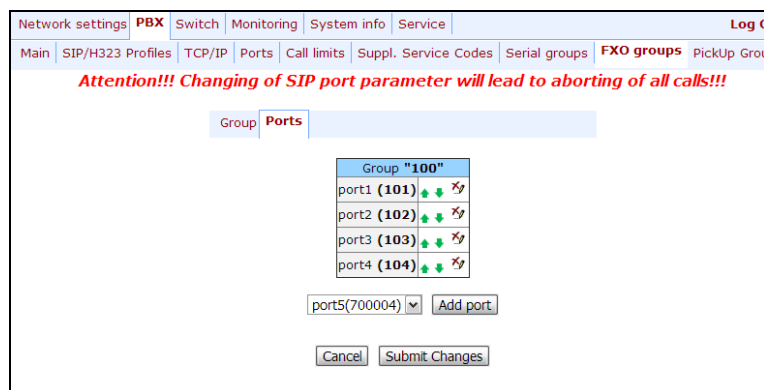
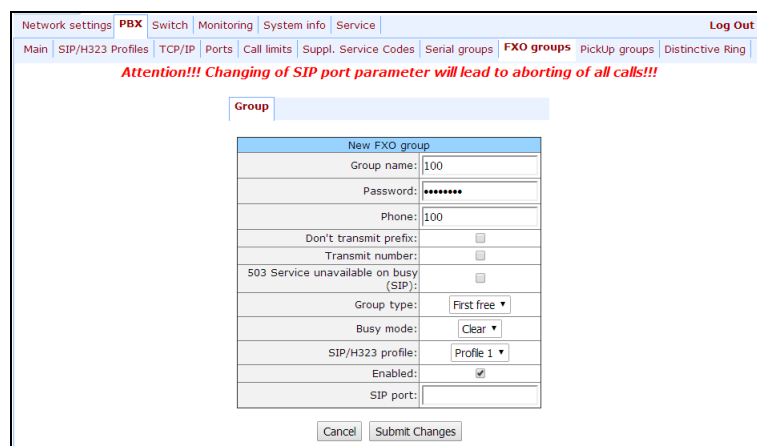
- 5 For outgoing calls routing, configure addressed dial peers. Settings should be modified for Profile 1, since subscriber ports have been configured for operation with this profile (PBX -> SIP/H323 profiles -> profile 1 -> dialplan):

To have the ability to access subscriber lines (numbers 101-104) and make intra-office calls (numbers 200-2XX), it will be sufficient to select SIP ('Protocol' parameter takes 'SIP' value) and configure the following routing rule: L15 S8 ([12]xx@{local}). Click 'Submit Changes' button to apply settings.



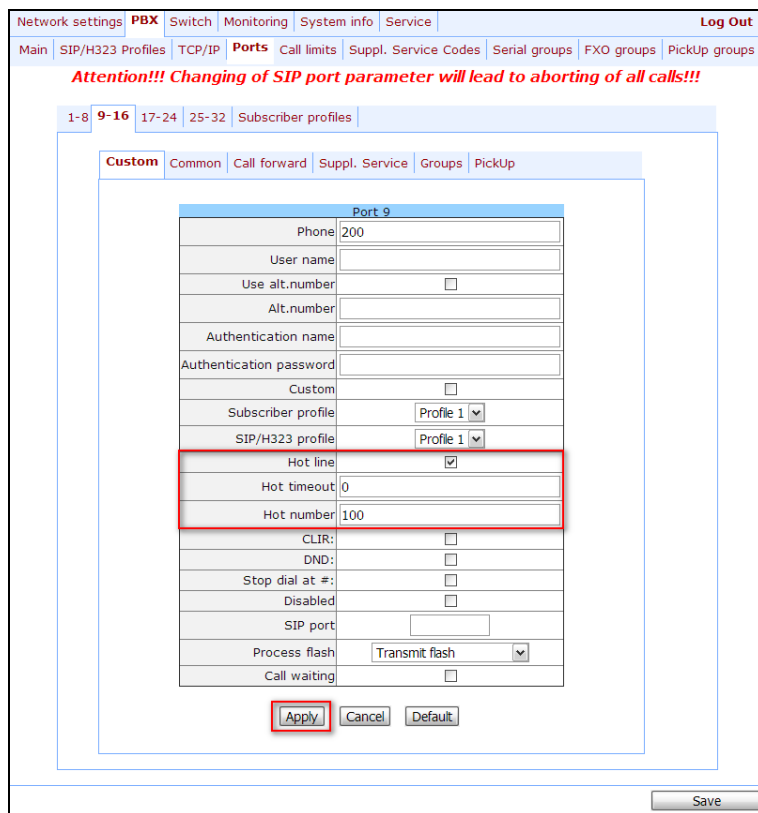
6 To implement the automatic line engagement when subscriber takes the phone offhook, you should:

6.1 Create FXO group (for example, with a number of '100'), configure it in Profile 1 (PBX -> FXO groups -> New group) and add the existing 4 x FXO lines (PBX -> FXO groups -> Edit -> Ports) in it:



6.2. For FXS subscriber ports, configure 'Hotline' service for FXO group number, so that the

free FXO line will become engaged when the phone goes offhook (PBX -> Ports -> Edit -> Custom). If zero timeout ('Hot timeout') is set, intra-office communication will be disabled; if timeout is non-zero, during this period of time you may begin to dial an internal number of a subscriber.



The screenshot shows the configuration interface for Port 9. The 'Hot line' checkbox is checked, and the 'Hot timeout' field is set to 0. The 'Apply' button is highlighted with a red box.

| Port 9                  |                                     |
|-------------------------|-------------------------------------|
| Phone                   | 200                                 |
| User name               |                                     |
| Use alt.number          | <input type="checkbox"/>            |
| Alt.number              |                                     |
| Authentication name     |                                     |
| Authentication password |                                     |
| Custom                  | <input type="checkbox"/>            |
| Subscriber profile      | Profile 1                           |
| SIP/H323 profile        | Profile 1                           |
| Hot line                | <input checked="" type="checkbox"/> |
| Hot timeout             | 0                                   |
| Hot number              | 100                                 |
| CLIR:                   | <input type="checkbox"/>            |
| DND:                    | <input type="checkbox"/>            |
| Stop dial at #:         | <input type="checkbox"/>            |
| Disabled                | <input type="checkbox"/>            |
| SIP port                |                                     |
| Process flash           | Transmit flash                      |
| Call waiting            | <input type="checkbox"/>            |

Buttons: **Apply** (highlighted), Cancel, Default

## APPENDIX G. AUTOMATIC CONFIGURATION PROCEDURE AND GATEWAY FIRMWARE VERSION CHECK

### 1. Using configuration settings

*Enable autoupdate*—this option allows to update gateway firmware and configuration automatically. When checked, firmware and configuration autoupdate is enabled based on Options 66 and 67 received via DHCP.

You may enable firmware and configuration version checks using the following options:

- *Configuration update interval*—device configuration check timer, in seconds.
- *Firmware update interval*—device firmware check timer, in seconds.

Software and configuration version checks utilize ***tau.versions*** file contents, and access either TFTP server defined in Option 66, or, if any of the Options 66 and 67 is not received via DHCP, ***update.local*** file contents.



**Default autoupdate settings are used only when software and configuration version checks are enabled.**

### 2. Software file types

FS—file system with working application.

CSP—gateway operating system.

MSP—media processor software.

ARM—platform software.

### 3. Default autoupdate settings

Gateway default settings as follow:

- Update mode: via **TFTP**;
- TFTP server: **update.local**;
- Path to file with firmware and configuration versions: **/tau.versions**.

### 4. Description of syntax for Option 66, 67 and firmware and configuration version file: tau.versions

#### 4.1. Option 66 syntax: TFTP server **FQDN** or **IP address**:

DHCP server configuration examples:

*Option tftp-server-name "update.local"*

*Option tftp-server-name "192.168.1.3"*

## 4.2. Option 67 syntax: '**Software file name and path; Configuration file name and path**'

4.2.1. Syntax **Software file path and name:** *firmware-pathFS/filenameFS; firmware-pathCSP/filenameCSP; firmware-pathMSP/filenameMSP; firmware-pathARM/filenameARM;*

- firmware-pathFS,CSP,MSP,ARM—path to the respective software file.
- filenameFS,CSP,MSP,ARM—name of the respective software file.

Software file name format:

- filenameFS—tau72.fs.{software version number}
- filenameCSP—tau72.csp.{software version number}
- filenameMSP—tau72.msp.{software version number}
- filenameARM—tau72.arm.{software version number}

4.2.2. Syntax **Configuration file path and name:** *conf-path/tau72\_<MAC>.dat*

- Conf-path—configuration file path.
- tau72\_<MAC>.dat—configuration file name When such name is received, gateway substitutes <MAC> with its own MAC address.

Example: Transferred name of a configuration file is tau72\_<MAC>.dat. When this name is received, the gateway generates availability request for tau72\_ A8F94B887D27.dat file on TFTP server.



**Configuration file is downloaded to PC via WEB interface in tau32M\_cfg.tar.gz format; to use it in autoconfiguration procedure, rename it to tau72\_<MAC>.dat.**

**To edit the file on a PC, unarchive the file, modify its data and create a new archive in the same format taking into account the path to file /etc/config; next, rename it to tau72\_<MAC>.dat.**

4.2.3. Example of Option 66 and 67 syntax, software file path and name, and gateway configuration for MAC address A8F94B887D27.

Transferred parameters:

- Option tftp-server-name “update.local”;
- Option bootfile-name  
“fs/tau72.fs.1.8.0;csp/tau72.csp.209;msp/tau72.msp.GA\_10\_23\_02\_03;  
arm/tau72.arm.20111117;cfg/tau72\_<MAC>.dat”



Gateway behaviour:

Download the software from *update.local* server, *fs/* path to file system with *tau72.fs.1.8.0* working application, *csp/* path to *tau72.csp.209* operating system, *misp/* path to *tau72.msp.GA\_10\_23\_02\_03* media processor software, *arm/* path to *tau72.arm.20111117* platform software.

Download configuration file from *update.local* server, *cfg/* path, filename—*tau72\_A8F94B887D27.dat*.

#### 4.3. *tau.versions* file format and syntax

##### 4.3.1. Format and syntax

- FS={FSversion} firmware-pathFS/filenameFS
  - CSP={CSPversion} firmware-pathCSP/filenameCSP
  - MSP={MSPversion} firmware-pathMSP/filenameMSP
  - ARM={ARMversion} firmware-pathARM/filenameARM
- FSversion/CSPversion/MSPversion/ARMversion—respective software version number.
  - firmware-pathFS,CSP,MSP,ARM—path to the respective software file.
  - filenameFS,CSP,MSP,ARM—name of the respective software file.

Software file name format:

- filenameFS—tau72.fs.{software version number}
- filenameCSP—tau72.csp.{software version number}
- filenameMSP—tau72.msp.{software version number}
- filenameARM—tau72.arm.{software version number}
- 

##### 4.3.2. *tau.versions* file contents example:

- FS=1.8.0 fs/tau72.fs.1.8.0
- CSP=209 csp/tau72.csp.209
- MSP=GA\_10\_23\_02\_03 msp/tau72.msp. GA\_10\_23\_02\_03
- ARM=20111117 arm/tau72.arm.20111117

## 5. Description of autoconfiguration and firmware update algorithm with Options 66 and 67

If the gateway receives Options 66 and 67 from DHCP server, these settings will have a priority in usage. Factory settings for automatic download of firmware and configuration files listed above will not work in this case.

If the gateway does not receive any of the Options 66 and 67 from DHCP server, settings listed above will be applied.

Firmware version check is performed using firmware file names received in Option 67.

When firmware and configuration update is required, subscriber port activity check is performed. If activity is detected, firmware download will be postponed, otherwise firmware download attempt will be made. In case of failure, the attempt will be repeated up to 5 times. When firmware update is complete, gateway will be restarted automatically.

## 6. Description of automatic software and configuration version checks

Firmware and configuration file version checks are performed in periods of time defined by '*Firmware update interval*' and '*Configuration update interval*' settings.

During device firmware version check, the gateway downloads ***tau.versions*** file and checks for versions contained in the file with the current device firmware versions. If the firmware version on TFTP server does not match the current device firmware version, the subscriber port activity check is performed. If activity is detected, firmware download will be postponed, otherwise firmware download attempt will be made. In case of failure, the attempt will be repeated up to 5 times. When firmware update is complete, gateway will be restarted automatically.

During device firmware version check, the gateway identifies differences with the current configuration, and if there were changes, they will be implemented. If the configuration download fails, the attempt will be repeated up to 5 times. Restart is not required to apply a new configuration.

### APPENDIX 3. CALCULATION OF PHONE LINE LENGTH

Table 18—Electrical resistance/cable type relationship for 1km of DC subscriber cable lines at 20°C ambient temperature.<sup>1</sup>

| Cable grade for subscriber lines of local exchange network   | Core diameter | Electrical resistance of 1km circuit, Ω, max. | Line length (other phone units), extended range mode enabled, km | Line length (other phone units), extended range mode disabled, km |
|--|---------------|---|--|---|
| TPP, TPPep, TPPZ, TPPepZ, TPPB, TPP epB, TPPZB, TPPBG, TPPepBG, TPPBbShp, TPPepBbShp, TPPZBbShp, TPPZepBbShp, TPpt | 0.32          | 458.0   | 1.638  | 0.983   |
|  | 0.40          | 296.0   | 2.534  | 1.520   |
|  | 0.50          | 192.0   | 3.906  | 2.344   |
|  | 0.64          | 116.0   | 6.466  | 3.879   |
|  | 0.70          | 96.0  | 7.813  | 4.688   |
| TPV, TPZBG   | 0.32          | 458.0   | 1.638  | 0.983   |
|  | 0.40          | 296.0   | 2.534  | 1.520   |
|  | 0.50          | 192.0   | 3.906  | 2.344   |
|  | 0.64          | 116.0   | 6.466  | 3.879   |
|  | 0.70          | 96.0  | 7.813  | 4.688   |
| TG, TB, TBG, TK  | 0.40          | 296.0   | 2.534  | 1.520   |
|  | 0.50          | 192.0   | 3.906  | 2.344   |
|  | 0.64          | 116.0   | 6.466  | 3.879   |
|  | 0.70          | 96.0  | 7.813  | 4.688   |
| TStShp, TASHp  | 0.50          | 192.0   | 3.906  | 2.344   |
|  | 0.70          | 96.0  | 7.813  | 4.688   |
| TSV  | 0.40          | 296.0   | 2.534  | 1.520   |
|  | 0.50          | 192.0   | 3.906  | 2.344   |
| KSPZP  | 0.64          | 116.0   | 6.466  | 3.879   |
| KSP, KSPZP, KSPPB, KSPZPB, KSPPt, KSPZPt, KSPZPK   | 0.90          | 56.8  | 13.204   | 7.923   |

Phone line length calculation for different types of cable<sup>2</sup>:

1. Cable resistance at 20°C

<sup>1</sup> Line length values for 'Rus' phone unit will be lower than specified in the table.

<sup>2</sup> Calculation is taken from <http://izmer-ls.ru/shle.html>

$$R_{cab} = L_{cab} * R_{sp20};$$

where

$R_{sp20}$  [ $\Omega$ /km] specific DC cable resistance at 20°C, table in Appendix 3.

2. Cable length

$$L_{cab} = R_{cab} / R_{sp20} \text{ [km]}$$

3. Loop resistance at 20°C

$$L_{loop} = 2 * L_{cab}$$

$$R_{loop} = L_{loop} * R_{sp20} = 2 * L_{cab} * R_{sp20};$$

$$L_{loop} = R_{loop} / R_{sp20}.$$

In case of phone lines, loop resistance includes phone unit resistance: 600 $\Omega$

If the extended range mode is enabled, ('Extended range loop' setting, see Section 5.1.2.1), equipment manufactured by Eltex provides maximum loop resistance of 2100 $\Omega$ . Subsequently, loop resistance excluding the phone unit equals to 1500 $\Omega$ . Thus, maximum loop length is calculated by the equation:

$$L_{loop} = 1500 / R_{sp20} \text{ [km]}.$$

4. Line length is calculated by the equation:

$$L_{line} = L_{cab} = L_{loop} / 2 = 1500 / (2 * R_{sp20}) = 750 / R_{sp20} \text{ [km]}.$$

5. If you have to consider the cable temperature, the cable line length will be calculated with an adjustment:

$$L_{line} = 750 / (R_{sp20} * (1 - a * (T - 20)))$$

where

a—is a temperature factor (table value)

T—cable temperature

## APPENDIX I. DEVICE FIREWALL CONFIGURATION—IPTABLES

Table—Device firewall configuration commands

| Command                       | Description  |
|-------------------------------|--|
| <code>iptables</code>         | Configuration of firewall rules                                    |
| <code>iptables-save</code>    | Save created firewall rules  |
| <code>iptables-restore</code> | Restore initial firewall rules, if the current rules are not saved |

To configure the firewall, connect to the gateway via COM port, SSH or Telnet (factory settings address: **192.168.1.2**, network mask: **255.255.255.0**) using terminal application, e.g. TERATERM, Putty, SecureCRT.

Firewall configuration procedure as follows:

1. Configuration via COM port: Connect the null modem cable to COM port of the PC and 'Console' port of the device.  
Configuration via SSH, Telnet: Connect the computer to the Ethernet port of the device using Ethernet cable.
2. Run the terminal application.
3. Configure COM port connection: data rate: 115200, data format: 8bit w/o parity, 1 stop bit, w/o flow control; or telnet, ssh connection: Factory default IP address: 192.168.1.2, port: 23 (telnet), port 22 (ssh).
4. Enter 'admin' as a login. Go to Linux shell by executing 'shell' command.
5. Create necessary tables according to `iptables` utility manual, use '`iptables -h`' command to view the manual.

### `iptables` utility usage examples:

a) accept TCP packets via port 25 from the host 212.164.54.162:

```
iptables -A INPUT -s 212.164.54.162 -p tcp -m tcp --dport 25 -j ACCEPT
```

b) reject all packets from the host 216.223.9.208:

```
iptables -A INPUT -s 216.223.9.208 -j DROP
```

c) reject all packets from the network 216.223.0.0/255.255.0.0:

```
iptables -A INPUT -s 216.223.0.0/255.255.0.0 -j DROP
```

r) view all tables:

```
iptables -L
```

6. Save created rules with '`iptables-save`' command.



**To restore previous rules, if changes have not been saved yet, use '`iptables-restore`' command.**

7. Enter '`save`' command to store the configuration into the non-volatile (flash) memory of the device.

## APPENDIX J. AUTOMATIC CONFIGURATION PROCEDURE AND GATEWAY FIRMWARE VERSION CHECK

'Enable autoupdate' is an option that allows to use automatic software and configuration updates, and perform their version checks in the defined periods of time.

### TAU-32M.IP automatic configuration and configuration file version check operation algorithm

For each TAU-32M.IP, a reference configuration file is created; in `/etc/config/cfg.yaml` configuration file, specify its current version `#ConfigFileVersion=YYYYMMDDHHMM`:

```
#!version 1.0
#tau-72 YAML config file
#Tree hierarchy:
#node1:
#   node2:
#       param1: value1
#       param2: value2
#NOTE: use spaces ' ' instead of tab '/t'
#NOTE: Don't del/add nodes
#NOTE: Use ':' after param names
#Remember, that quantity of spaces must be multiply to 8

#ConfigFileVersion=201302010905

Network:
    network:
        HOSTNAME: tau32m
```

During TAU-32M.IP startup, the gateway checks for the configuration file at the specified path on FTP/TFTP/HTTP/HTTPS server (and signs in to server, if necessary). If the configuration file is present, TAU-32M.IP will download it, store it in its file system and apply it as a current configuration file. Upon the expiry of '*Configuration update interval*' timeout, the gateway will re-download the configuration file from the server and compare versions of the current and downloaded configuration files (`ConfigFileVersion`). If the downloaded file version is higher than the current one, TAU-32M.IP saves and applies a new configuration; otherwise, the current configuration remains active.

When the operator wants to modify the gateway configuration, he should upload the modified configuration file with increased '*ConfigFileVersion*' value to the server, and the configuration will be updated automatically upon the expiry of '*Configuration update interval*' timeout. After restart, TAU-32M.IP will download configuration file from the server; this measure will protect the gateway from improper configuration. If you experience problems after configuring the gateway via Web configurator, restart the device to download the reference configuration.

Fig. 23 shows TAU-32M.IP automatic configuration and configuration file version check operation algorithm.

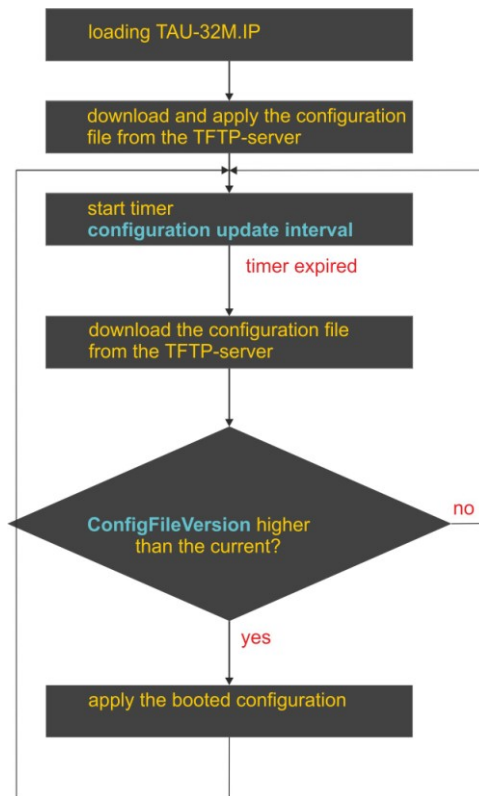


Fig. 23—TAU-32M.IP automatic configuration and configuration file version check operation algorithm



## Autoupdate and firmware version check operation algorithm

During TAU-32M.IP startup, and upon the expiry of 'Firmware update interval' timeout, the gateway checks for the version description file (tau.versions) at the specified path on TFTP server. If the configuration file is present, TAU-32M.IP will download it. This file contains information on versions of firmware files located at TFTP server as well as their paths and names. If versions of firmware located on server differ from the current ones (used by the gateway), the gateway checks for active call sessions. If there are no active call sessions, TAU-32M.IP will download firmware files with versions defined in *tau.versions* file. When download finishes, the gateway firmware will be updated; otherwise, 10 seconds timeout will be activated. When this timeout expires, the gateway checks again for active call sessions.

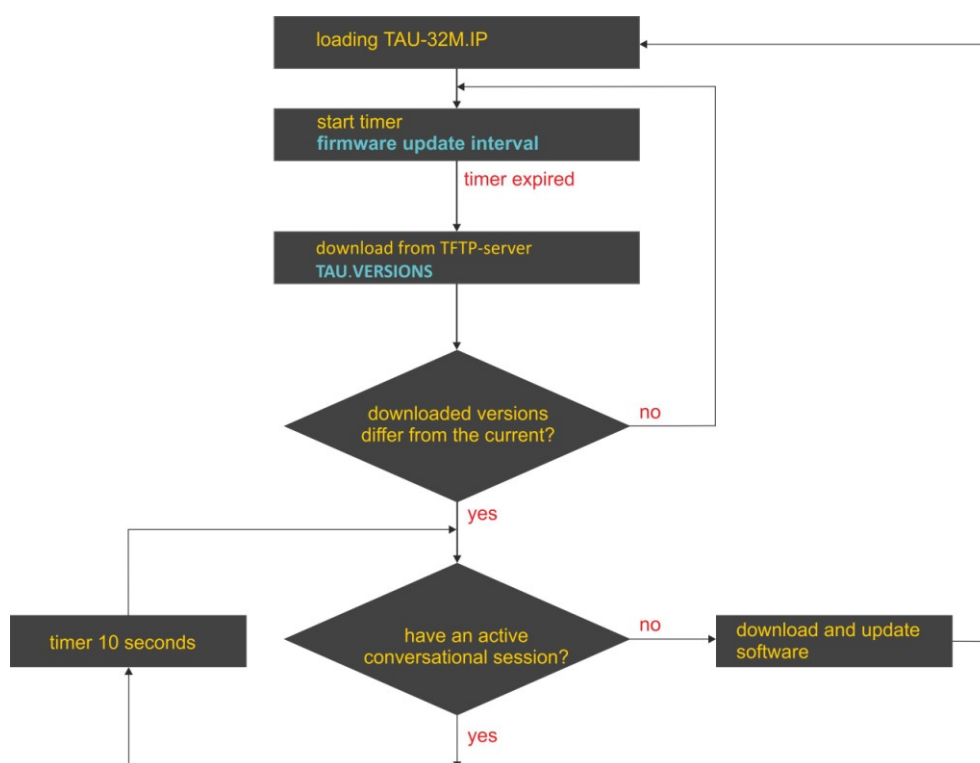


Fig. 24—Autoupdate and firmware version check operation algorithm

## Automatic configuration and firmware version check: parameter obtaining methods

**Method 1: Using DHCP Option 43 or Options 66 and 67 when DHCP is enabled in network settings or for one of VLANs.**

Gateway default settings as follow:

|   |                               |
|---|-------------------------------|
| Update mode   | via TFTP                      |
| TFTP server   | <b>update.local</b>           |
| Path to file with firmware and configuration versions | <b>tau.versions</b>           |
| Path to the configuration file                        | <b>tau32M_&lt;MAC&gt;.dat</b> |

**tau32M\_<MAC>.dat**—configuration file name When such name is received, gateway substitutes <MAC> with its own MAC address.

*Example:* Example: Transferred name of a configuration file is tau32\_<MAC>.dat. When this name is received, the gateway generates availability request for tau32\_A8F94B887D27.dat file on TFTP server.



**Configuration file is downloaded to PC via WEB interface in tau32M\_cfg.tar.gz format; to use it in autoconfiguration procedure, rename it to tau32\_<MAC>.dat.**

**To edit the file on a PC, unarchive the file, modify its data and create a new archive in the same format taking into account the path to file /etc/config; next, rename it to tau32\_<MAC>.dat.**

If autoupdate server requires authorization, configure the following parameters: Autoupdate auth, Username, Password.

If the gateway receives Options 43, 66, and 67 from DHCP server simultaneously, Option 43 will have a priority in usage. If Option 43 is missing, Options 66 and 67 will be processed. Factory settings for automatic download of firmware and configuration files listed above will not work in this case.

*Description of syntax for Option 43, 66, 67 and firmware and configuration version file: tau.versions*

*Option 43 syntax:*

**<suboption number><suboption length><suboption value>,**

where

- Suboption number and length are passed in a numeric (Hex) format
- Suboption value is passed as ASCII code

Suboptions necessary for autoupdate procedure:

- 5—autoupdate server address.  
Address should be received in the following format: **<proto>://<address>[:<port>]**,  
where  
<proto>—protocol (ftp, tftp, http, https)  
<address>—autoupdate server IP address or domain name  
<port>—autoupdate server port (optional parameter)
- 6—autoupdate configuration file name
- 7—autoupdate firmware file name

Example of the option record:

```
05:11:68:74:74:70:3A:2F:2F:61:75:74:6F:2E:72:75:3A:38:30:06:09:61:75:74:6F:2E:63:6F:6E:66:07
:08:61:75:74:6F:2E:6B:6D:67
```

where

- 05—autoupdate server address suboption number
- 11—length, 17bytes (0x11 = 17 dec)
- 68:74:74:70:3A:2F:2F:61:75:74:6F:2E:72:75:3A:38:30—suboption value (<http://auto.ru:80>)
- 06—configuration file name suboption number
- 09—length, 9bytes
- 61:75:74:6F:2E:63:6F:6E:66—suboption value (auto.conf)
- 07—software file name suboption number
- 08—length, 8bytes
- 61:75:74:6F:2E:6B:6D:67—suboption value (auto.img).

*Option 66 syntax: TFTP server **FQDN** or **IP address**:*

DHCP server configuration examples:

```
Option tftp-server-name "update.local"
Option tftp-server-name "192.168.1.3"
```

*Option 67 syntax: '**tau.versions file name and path; Configuration file name and path**'*

Syntax **tau.versions file path**: *conf-path/tau.versions*

Syntax **Configuration file path and name**: *conf-path/tau32M\_<MAC>.dat*

where **conf-path**—configuration file path

Example of Option 66 and 67 syntax, software file path and name, and gateway configuration for MAC address A8F94B887D27

Transferred parameters:

```
Option tftp-server-name "update.local";
```

```
Option bootfile-name "/tau32Mip/firmware/tau.versions;/tau32Mip/conf/tau32M_<MAC>.dat"
```

**Method 2: Using autoupdate parameter configuration, specified in 'Autoupdate Settings' section, when the static address is assigned in network settings, or when PPPoE is selected.**

In this case, 'Autoupdate protocol', 'Autoupdate server', 'Configuration file' and 'Firmware versions file' parameters are used, defined in 'Autoupdate Settings' section. If autoupdate server requires authorization, configure the following parameters: Autoupdate auth, Username, Password.

### tau.versions file format and syntax:

FS={FSversion} firmware-pathFS/fileNameFS

CSP={CSPversion} firmware-pathCSP/fileNameCSP

MSP={MSPversion} firmware-pathMSP/fileNameMSP

IMG={IMGversion} firmware-pathIMG/fileNameIMG

ARM={ARMversion} firmware-pathARM/fileNameARM

where FSversion/CSPversion/MSPversion/ARMversion—respective software version number:

- **firmware-pathFS,CSP,MSP,ARM**—path to the respective software file
- **fileNameFS,CSP,MSP,ARM**—name of the respective software file.

### Software file types:

- FS—file system with working application<sup>1</sup>.
- CSP—gateway operating system<sup>1</sup>.
- MSP—media processor software<sup>1</sup>.
- IMG—complete software image, includes FS, CSP, MSP, and ARM.
- ARM—platform software<sup>1</sup>.

Software file name format:

- **fileNameFS**—tau32.fs.{software version number}
- **fileNameCSP**—tau32.csp.{software version number}
- **fileNameMSP**—tau32.msp.{software version number}
- **fileNameIMG**—tau32.img.{software version number}
- **fileNameARM**—tau32.arm.{software version number}

---

<sup>1</sup> Not used in the current firmware version.

---

**tau.versions** file contents example:

```
FS=1.8.0 fs/tau32.fs.1.8.0
CSP=209 csp/tau32.csp.209
MSP=GA_10_23_02_03 msp/tau32.msp. GA_10_23_02_03
IMG=2.1.0 tau32Mip/firmware/img/tau32.img.2.1.0
ARM=20111117 arm/tau32.arm.20111117
```

---

## APPENDIX K. PROCESSING OF INFO REQUESTS CONTAINING APPLICATION/BROADSOFT AND APPLICATION/SSCC AND USED FOR SUPPLEMENTARY SERVICES

### 1) *Supplementary services, performed using BROADSOFT algorithm.*

Device supports 'Call waiting' service that uses algorithm performed by BROADSOFT softswitch. To perform the service, you should configure *flash* event transfer to application/broadsoft.

When the second call is received by the gateway, INFO request is received with contents: 'play tone CallWaitingToneN', where N may have a value from 1 to 4. Having received this request, the gateway will play 'notification' tone to the subscriber.

To release a notification tone, INFO request is received from the softswitch with contents: 'stop CallWaitingTone'.

To put the first call on hold and respond to the second call, the subscriber should press 'flash' button, gateway transfers INFO request with contents: 'event flashhook'.

### 2) *Supplementary services, performed using HUAWEI algorithm.*

Device supports 'Call waiting', 'Call transfer', and '3-way conference' services that use algorithm performed by HUAWEI softswitch. To perform these services, you should configure *flash* event transfer to application/sscc.

When the second call is received by the gateway, INFO request is received with contents: 'tone-type=beep; beep-duration=X; beep-gap=Y; beep-times=Z'. Having received this request, the gateway will play 'notification' tone to the subscriber with parameters: X—ring duration, Y—pause duration, Z—number of rings.

Other tones processed by the gateway are:

- tone-type=busy—'busy' tone playback
- tone-type=ringback—'ringback' tone playback
- tone-type=specialdial—'PBX response' tone playback Along with this tone, the softswitch sends 'dial-timer=N' parameter, that defines the dialling timeout from the gateway side. If N=0, the dialling timeout is unlimited. Used in order to dial the second subscriber number or code for the respective action execution (for example, 2—switch between subscribers, 3—conference.) If timeout is non-zero, when it passes, the gateway will transfer an additional INFO request containing all dialled digits during this timeout.

To put the first call on hold (to perform the second call or respond to the second call), the subscriber should press 'flash' button, gateway transfers INFO request with contents: 'event flashhook'.

## APPENDIX L. HELP ON TIMEZONES

Date line (UTC-12) Baker Island,Howland Island PST12 USA/Minor Outlying Islands

USA Canada (UTC-10) Hawaii Time HST10 Pacific/Honolulu

USA Canada (UTC-9) Alaska Time AKST9AKDT,M3.2.0,M11.1.0 America/Anchorage

USA Canada (UTC-8) Pacific Time PST8PDT,M3.2.0,M11.1.0 America/Los\_Angeles

USA Canada (UTC-7) Mountain Time MST7MDT,M3.2.0,M11.1.0 America/Denver

USA Canada (UTC-7) Mountain Time (Arizona, no DST) MST7 America/Phoenix

USA Canada (UTC-6) Central Time CST6CDT,M3.2.0,M11.1.0 America/Chicago

USA Canada (UTC-5) Eastern Time EST5EDT,M3.2.0,M11.1.0 America/New\_York

Atlantic (UTC-4) Bermuda AST4ADT,M3.2.0,M11.1.0 Atlantic/Bermuda

Central and South America (UTC-3) Argentina ART3 America/Argentina/Buenos\_Aires

Central and South America (UTC-3) Sao Paulo,Brazil BRT3BRST,M11.1.0/0,M2.5.0/0 America/Sao\_Paulo

Europe (UTC+0) GMT0 GMT0 GMT0

Europe (UTC+0) Dublin,Ireland GMT0IST,M3.5.0/1,M10.5.0 Europe/Dublin

Europe (UTC+0) Lisbon,Portugal WET0WEST,M3.5.0/1,M10.5.0 Europe/Lisbon

Europe (UTC+0) London,GreatBritain GMT0BST,M3.5.0/1,M10.5.0 Europe/London

Europe (UTC+1) Amsterdam,Netherlands CET-1CEST,M3.5.0,M10.5.0/3 Europe/Amsterdam

Europe (UTC+1) Berlin,Germany CET-1CEST,M3.5.0,M10.5.0/3 Europe/Berlin

Europe (UTC+1) Brussels,Belgium CET-1CEST,M3.5.0,M10.5.0/3 Europe/Brussels

Europe (UTC+1) Bratislava,Slovakia CET-1CEST,M3.5.0,M10.5.0/3 Europe/Bratislava

Europe (UTC+1) Budapest,Hungary CET-1CEST,M3.5.0,M10.5.0/3 Europe/Budapest

Europe (UTC+1) Copenhagen,Denmark CET-1CEST,M3.5.0,M10.5.0/3 Europe/Copenhagen

Europe (UTC+1) Madrid,Spain CET-1CEST,M3.5.0,M10.5.0/3 Europe/Madrid

Europe (UTC+1) Oslo,Norway CET-1CEST,M3.5.0,M10.5.0/3 Europe/Oslo

Europe (UTC+1) Paris,France CET-1CEST,M3.5.0,M10.5.0/3 Europe/Paris

Europe (UTC+1) Prague,CzechRepublic CET-1CEST,M3.5.0,M10.5.0/3 Europe/Prague

Europe (UTC+1) Roma,Italy CET-1CEST,M3.5.0,M10.5.0/3 Europe/Rome

Europe (UTC+1) Zurich,Switzerland CET-1CEST,M3.5.0,M10.5.0/3 Europe/Zurich

Europe (UTC+1) Stockholm,Sweden CET-1CEST,M3.5.0,M10.5.0/3 Europe/Stockholm

Europe (UTC+2) Helsinki,Finland EET-2EEST,M3.5.0/3,M10.5.0/4 Europe/Helsinki

Europe (UTC+2) Kyiv,Ukraine EET-2EEST,M3.5.0/3,M10.5.0/4 Europe/Kiev

Europe (UTC+2) Athens,Greece EET-2EEST,M3.5.0/3,M10.5.0/4 Europe/Athens

Asia (UTC+2) Amman EET-2EEST,M3.5.4/0,M10.5.5/1 Asia/Amman

Asia (UTC+2) Beirut EET-2EEST,M3.5.0/0,M10.5.0/0 Asia/Beirut

Asia (UTC+2) Damascus EET-2EEST,J91/0,J274/0 Asia/Damascus

Asia (UTC+2) Gaza EET-2EEST,J91/0,M10.3.5/0 Asia/Gaza

Asia (UTC+2) Jerusalem GMT-2 Asia/Jerusalem

Asia (UTC+2) Nicosia EET-2EEST,M3.5.0/3,M10.5.0/4 Asia/Nicosia

Asia (UTC+3) Aden AST-3 Asia/Aden

Asia (UTC+3) Baghdad AST-3ADT,J91/3,J274/4 Asia/Baghdad

Asia (UTC+3) Bahrain AST-3 Asia/Bahrain

Asia (UTC+3) Kuwait AST-3 Asia/Kuwait

Asia (UTC+3) Qatar AST-3 Asia/Qatar

Asia (UTC+3) Riyadh AST-3 Asia/Riyadh

Europe (UTC+3) Moscow,Russia MSK-3 Europe/Moscow

Asia (UTC+3:30) Tehran IRST-3:30 Asia/Tehran

Asia (UTC+4) Baku AZT-4AZST,M3.5.0/4,M10.5.0/5 Asia/Baku

Asia (UTC+4) Dubai GST-4 Asia/Dubai

Asia (UTC+4) Muscat GST-4 Asia/Muscat

Asia (UTC+4) Tbilisi GET-4 Asia/Tbilisi

Asia (UTC+4) Yerevan AMT-4AMST,M3.5.0,M10.5.0/3 Asia/Yerevan

Asia (UTC+4:30) Kabul AFT-4:30 Asia/Kabul

Asia (UTC+5) Aqtobe AQTT-5 Asia/Aqtobe

Asia (UTC+5) Ashgabat TMT-5 Asia/Ashgabat

Asia (UTC+5) Dushanbe TJT-5 Asia/Dushanbe

Asia (UTC+5) Karachi PKT-5 Asia/Karachi

Asia (UTC+5) Oral ORAT-5 Asia/Oral

Asia (UTC+5) Samarkand UZT-5 Asia/Samarkand

Asia (UTC+5) Tashkent UZT-5 Asia/Tashkent

Asia (UTC+5) Yekaterinburg YEKT-5 Asia/Yekaterinburg

Asia (UTC+5:30) Calcutta IST-5:30 Asia/Calcutta

Asia (UTC+5:30) Colombo IST-5:30 Asia/Colombo

Asia (UTC+6) Almaty ALMT-6 Asia/Almaty

Asia (UTC+6) Bishkek KGT-6 Asia/Bishkek

Asia (UTC+6) Dhaka BDT-6 Asia/Dhaka

Asia (UTC+6) Qyzylorda QYZT-6 Asia/Qyzylorda

Asia (UTC+6) Thimphu BTT-6 Asia/Thimphu

Asia (UTC+6) Novosibirsk NOV-6 Asia/Novosibirsk

Asia (UTC+6) Omsk OMST-6 Asia/Omsk



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Asia (UTC+7) Jakarta WIT-7 Asia/Jakarta  
Asia (UTC+7) Bangkok ICT-7 Asia/Bangkok  
Asia (UTC+7) Vientiane ICT-7 Asia/Vientiane  
Asia (UTC+7) Phnom Penh ICT-7 Asia/Phnom\_Penh  
Asia (UTC+7) Krasnoyarsk Asia/Krasnoyarsk

Asia (UTC+8) Chongqing CST-8 Asia/Chongqing  
Asia (UTC+8) Hong Kong HKT-8 Asia/Hong\_Kong  
Asia (UTC+8) Shanghai CST-8 Asia/Shanghai  
Asia (UTC+8) Singapore SGT-8 Asia/Singapore  
Asia (UTC+8) Urumqi CST-8 Asia/Urumqi  
Asia (UTC+8) Taiwan CST-8 Asia/Taipei  
Asia (UTC+8) Ulaanbaatar ULAT-8 Asia/Ulaanbaatar  
Asia (UTC+8) Irkutsk Asia/Irkutsk  
Australia (UTC+8) Perth WST-8 Australia/Perth Perth

Asia (UTC+9) Dili TLT-9 Asia/Dili  
Asia (UTC+9) Jayapura EIT-9 Asia/Jayapura  
Asia (UTC+9) Pyongyang KST-9 Asia/Pyongyang  
Asia (UTC+9) Seoul KST-9 Asia/Seoul  
Asia (UTC+9) Yakutsk YAKT-9 Asia/Yakutsk  
Asia (UTC+9) Tokyo JST-9 Asia/Tokyo

Australia (UTC+9:30) Adelaide CST-9:30CST,M10.5.0,M3.5.0/3 Australia/Adelaide  
Australia (UTC+9:30) Darwin CST-9:30 Australia/Darwin

Australia (UTC+10) Brisbane EST-10 Australia/Brisbane  
Australia (UTC+10) Melbourne,Canberra,Sydney EST-10EST,M10.5.0,M3.5.0/3 Australia/Melbourne  
Australia (UTC+10) Hobart EST-10EST,M10.1.0,M3.5.0/3 Australia/Hobart

Asia (UTC+10) Vladivostok VLAST-10 Asia/Vladivostok

Asia (UTC+12) Anadyr ANAT-12 Asia/Anadyr  
New Zealand (UTC+12) Auckland, Wellington NZST-12NZDT,M10.1.0,M3.3.0/3 Pacific/Auckland

Tonga (UTC+13) Nuku'alofa TOT-13 Tonga/Nuku'alofa

Kiribati (UTC+14) Caroline Island LINT-14 Kiribati/Caroline Island

## TECHNICAL SUPPORT

For technical assistance in issues related to handling of ELTEXALATAU Ltd. equipment please address to Service Centre of the company:

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Phone:

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+7(727) 220-76-07

E-mail: [post@eltexalatau.kz](mailto:post@eltexalatau.kz)

In official website of the ELTEXALATAU Ltd. you can find technical documentation and software for products, refer to knowledge base, consult with engineers of Service center in our technical forum:

<http://www.eltexalatau.kz/en/>