- — **Office PBX functionality**
- — **3G/4G channel redundancy**
- — **IPsec encryption**
- — **TR-069/DHCP-based autoprovisioning**
- — **Maximum cable line length — 6 km**[1]
- — **Measurement of subscriber's line parameters**

TAU-4.IP

TAU-8.IP

TAU-4.IP and TAU-8.IP VoIP gateways are an optimal solution to provide corporate clients with advanced VoIP services via analog phones.

Business solution

Due to the wide functional capabilities, TAU-4.IP and TAU-8.IP are used as an office PBX that provides internal and external routing, as well as a basic set of Value Added Services.

High-quality sound

A high-performance hardware platform provides high quality of voice data due to support for a standard set of audio codecs used in VoIP networks (G.711, G.723.1, G.726 (24 Kbps and 32 Kbps), G.729 (A/B)), voice features (echo cancellation, silence detector, comfort noise generator, DTMF dialing) and traffic prioritization rules (QoS).

Redundancy

The devices support automatic switching to a redundant 3G/4G channel in case of failure of the main Internet connection. If there are no redundant channels, local routing between voice ports is available.

Easy-to-use

Eltex.ACS software based on TR-069 specification provides centralized configuration downloading, intellectual firmware updating and collecting status data of VoIP gateways. Eltex.ACS reduces operating expenses (OPEX) and makes management of Eltex CPE easy.

Interface configuration

Name	WAN	FXS[1]	USB
TAU-4.IP	1 × 100M	4	1 × USB 2.0
TAU-8.IP	1 × 100M	8	1 × USB 2.0

[1] *It is not recommended to use on subscriber lines exposed to strong external electromagnetic fields or atmospheric electricity, as well as in cases where the telephone cable runs parallel to AC wires and computer cables.*

Features and capabilities

Interfaces
– 4 × FXS ports for TAU-4.IP[1]
– 8 x FXS ports for TAU-8.IP[1]
– 1 x 10/100BASE-T WAN port
– 1 x USB port

VoIP protocols
– SIP

Voice codecs
– G.711 (a-law, μ-law)
– G.723.1
– G. 726 (24 Kbps and 32 Kbps)
– G.729 (A/B)

Fax transmission
– T.38 UDP Real-Time Fax
– G.711 (a-law, μ-law) pass-through

Voice standards
– VAD (Voice Activity Detector)
– CNG (Comfort Noise Generation)
– AEC (Acoustic Echo Cancellation, G.168 recommendation)

Functional features
– Payphones connection
– Subscriber line's physical parameters measurement
– Local switching in case of a SIP-server connection failure
– VAS management via a phone
– Distinctive Ring
– Keep-alive during operation behind NAT
– Calling Party Control (CPC)

DTMF
– Signal detection and generation
– Transmission by INBAND, RFC 2833, SIP INFO methods

Value Added Services (VAS)
– Call Hold
– Call Transfer
– Call Waiting
– Call forwarding when busy (CFB)
– Call forwarding on no reply (CFNR)
– Call forwarding unconditional (CFU)
– Calling line identification (FSK Type I, FSK Type II, DTMF)
– Calling Line Identification Restriction (CLIR)
– Hotline/Warmline
– Call Group
– 3-Way Conference
– Pickup Group
– Hunt groups
– A dedicated server for 3-Way Conferencing (RFC 4579)

VoIP functions
– Local connection switching
– Operation without a SIP server
– Flexible dial plan
– Management profiles for FXS ports
– SIP profiles (support for up to 8 profiles)
– Geographical redundancy of a SIP server (up to 4 redundant SIP servers)
– Application of settings without reboot
– Voice transmission through a protected channel (encryption through IPsec)
– IMS (3GPP TS 24.623) for Call Hold, Call Waiting, 3-Way Conference, Hotline management
– Using SIP servers from DHCP-option 120
– Support for operation behind NAT (STUN and Public IP)
– Setting custom call-control signals

Quality of service (QoS)
– DSCP and 802.1p assignment for SIP and RTP packets
– Bandwidth redundancy

Network functions
– Different protocols for connection to a service provider network (Static, DHCP, PPPoE, PPTP, L2TP)
– Local DNS server
– Static and dynamic routing
– VLAN per service (VLAN for each service: Internet, VoIP, Management)
– Firewall
– Operation via 3G/4G USB modems with connection redundancy
– Print server
– IPsec (for voice transmission and remote control)

Management
– Web interface (Russian and English versions[2])
– SNMP (phone parameters configuration, monitoring and statistical data collection)
– CLI (Command Line Interface)
– Telnet
– Syslog
– Tcpdump
– SSH
– TR-069 (Eltex.ACS server is recommended)
– DHCP-based autoprovisioning (43, 66, 67 DHCP options)
– Management via IPsec encrypted channel

Security
– Username and password authentication
– Firewall
– Access rights differentiation for admin/user/viewer
– Password encryption
– Digest authorization
– Telnet access is disabled by default
– Access for the user/viewer by the standard password is disabled

[1] *It is not recommended to use on subscriber lines exposed to strong external electromagnetic fields or atmospheric electricity, as well as in cases where the telephone cable runs parallel to AC wires and computer cables.*
[2] *Supported from firmware version 1.2.1 and higher.*

Features and capabilities (continued)

USB port

- USB storage connection with FAT/FAT32/EXT2/EXT3/NTFS file systems — file exchange according to FTP protocol
- USB 3G/4G modems connection — 3G/4G channels redundancy
- Printer connection — setting up a print server

Technical features

- SDRAM 256 MB
- Flash 32 MB
- OS Linux

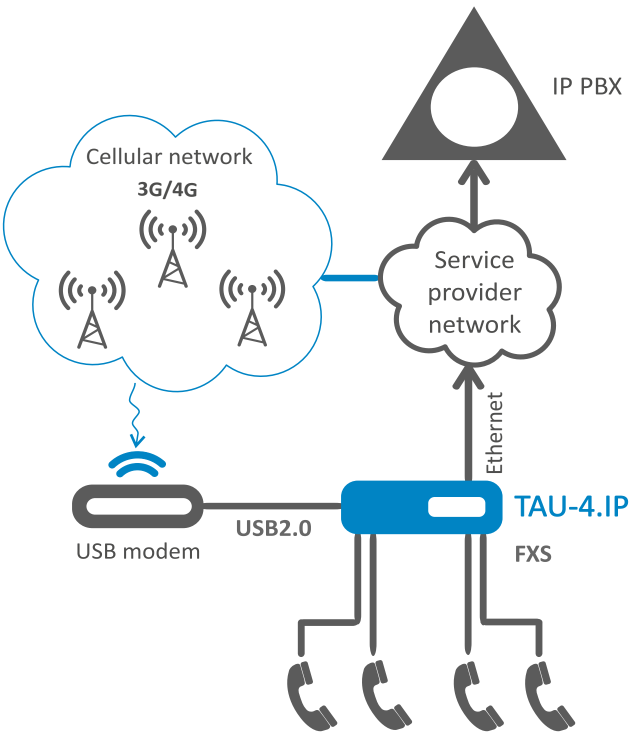
Physical features

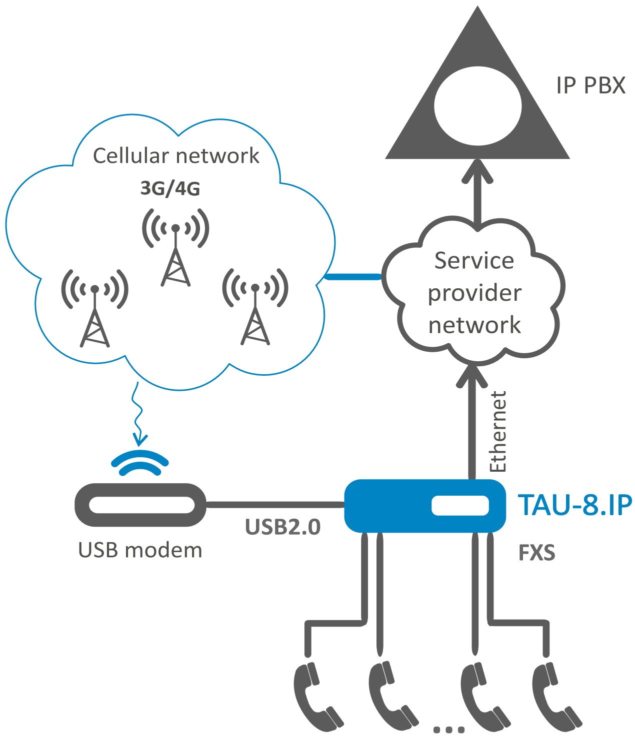
- Power adapter: 12V DC, 2A
- Power consumption:
  - up to 11W for TAU-4.IP
  - up to 16W for TAU-8.IP
- Operating temperature: from +5°C to +40°C
- Operating humidity: up to 80 %
- Dimensions: 218 × 46 × 116 mm, desktop case
- Weight: no more than 0.3 kg

Specifications

- 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 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 3361 DHCP Option 120
- RFC 3550 RTP A Transport Protocol for Real-Time Applications
- RFC 5806 Diversion Indication in SIP

Use case





Ordering information

Name	Description
TAU-4.IP	TAU-4.IP VoIP gateway: 4×FXS, 1×WAN, 1×USB, SIP
TAU-8.IP	TAU-8.IP VoIP gateway: 8×FXS, 1×WAN, 1×USB, SIP
Related software	
ACS-CPE-256	ACS-CPE Option of Eltex.ACS system for Eltex CPE auto configuration: 256 subscribers
ACS-CPE-512	ACS-CPE Option of Eltex.ACS system for Eltex CPE auto configuration: 512 subscribers
ACS-CPE-1024	ACS-CPE Option of Eltex.ACS system for Eltex CPE auto configuration: 1024 subscribers

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About Eltex

Eltex Enterprise is a leading Russian developer and manufacturer of communication equipment with 30 years of history. Complete solutions and their seamless integrability into the Customer's infrastructure are the priority growth areas of the company.