ELTEX
Integrated networking solutions

— **Office PBX functionality**
— **TR-069**
— **Measurement of subscriber's line parameters**
— **High quality of sound**
— **Fax transmission**

TAU-8N.IP VoIP gateway is an optimal solution to provide corporate clients with advanced VoIP services via analog phones.

Due to its wide functionality TAU-8N.IP can be used both as an isolated office PBX with internal switching and basic set of VHF, and in the mode of interaction with IP PBX.

High quality sound is provided by a high-performance hardware platform based on a modern chip, support for echo compensation functions (Simple, Speex and WebRTC), silence detector, comfort noise generator, reception and generation of DTMF signals, audio codecs used in VoIP-networks (G.711 (a-law, μ-law), G.723.1, G.729 (A/B), G.726 (16 Kbps, 24 Kbps, 32 Kbps, 40 Kbps), AAL2-G.726 (16 Kbps, 24 Kbps, 32 Kbps, 40 Kbps)), as well as mechanisms allowing to prioritize traffic from TAU 8N.IP on data transmission networks. Fax transmission is realized using G.711 pass-through audio codec and T.38 protocol.

Technical features

General parameters	
OS	Linux
Interfaces	
10/100BASE-T Ethernet port	1
10/100BASE-T MGMT port	1
FXS ports	8
USB port	1
VoIP	
VoIP protocols	SIP
Fax transmission	T.38, G.711 pass-through
Connection types	Static, DHCP, PPPoE, L2TP
Physical and environmental parameters	
Power	12 V DC, 2 A
Power consumption	no more than 19 W
Operating temperature	from +0 to +40 °C
Operating humidity	up to 80 %
Form factor	desktop version, wall mountable
Weight	0.3 kg
Dimensions (W×H×D), mm	208 × 38 × 115

Features and capabilities

Interfaces
– 1 × 10/100BASE-T Ethernet port
– 1 × 10/100BASE-T MGMT port
– 1 × USB port 2.0
– 8 × FXS RJ-11 ports

VoIP protocols
– SIP

VoIP functions
– Local connection switching
– Operation without a SIP server
– Flexible dialplan for FXS ports and SIP profiles
– Dialplan profiles
– Support for DHCP Option 120
– Application of settings without reboot
– Transmission of # symbol as %23
– IMS (3GPP TS 24.623) to manage Call Hold, Call Waiting, 3-Way Conference, Hotline, Call Transfer services

Voice codecs
– G.711 a-law, µ-law
– G.726 (16 kbps, 24 kbps, 32 kbps, 40 kbps)
– AAL2-G.726 (16 kbps, 24 kbps, 32 kbps, 40 kbps)
– G.723.1
– G.729 (A/B)

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

Voice standards
– VAD (Voice Activity Detector)
– CNG (Comfort Noise Generation)
– AEC (Acoustic Echo Cancellation, G.165, G.168 recommendations)

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

Supplementary services
– Call Hold
– Call Transfer
– Call Waiting
– Call Forwarding on Busy (CFB)
– Call Forwarding on No Reply (CFNR)
– Call Forwarding Uncoditional (CFU)
– Caller ID
– Calling Line Identification Restriction (CLIR) of Caller ID
– Hotline/Warmline
– Call Group
– Local 3-Way conference

Quality of Service (QoS)
– Diffserv and 802.1p assignment for SIP and RTP packets

Connection types
– Static IP address
– DHCP client
– PPPoE
– L2TP

Network functions
– Static routing
– Operation via 3G/4G USB modem with the possibility of automatic connection backup
– IPsec (for voice and remote control)
– VLAN
– Support for DHCP options
  – 1 – Subnet Mask
  – 3 – Default network gateway address
  – 6 – DNS Server Address
  – 12 – Host Name
  – 15 – Domain Name
  – 28 – Broadcast Address
  – 33 – Static Route
  – 42 – NTP Servers
  – 43 – Vendor Specific
  – 60 – Alternative Vendor ID
  – 66 – TFTP Server Address
  – 67 – Boot File Name (to download via TFTP from server option 66)
  – 120 – Outbound of SIP Server
  – 121 – Classless Static Route Options
  – 249 – Private/Classless Static Route (MS)

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

Management
– Web
– SNMP
– CLI
– Telnet
– SSH
– TR-069
– DHCP autoprovisioning
– IPsec
– TACACS+

Security
– Username and password authentication
– Firewall
– Access rights differentiation for admin, user and viewer

USB port
– Connection of USB drive with FAT/FAT32/NTFS file systems
– 3G/4G USB modem connection - 3G/4G communication backup

Technical features
– SD RAM 512 MB
– Flash 512 MB
– OS Linux

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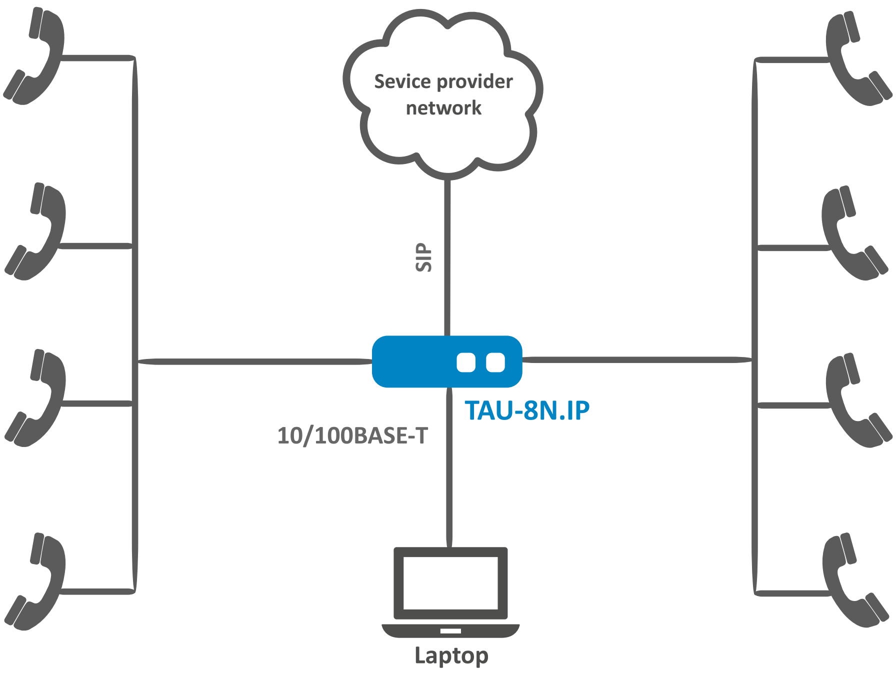
Use case



Ordering information

Name	Description
TAU-8N.IP	VoIP gateway TAU-8N.IP: 8×FXS, 1×WAN, 1×MGMT, 1×USB, SIP

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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.