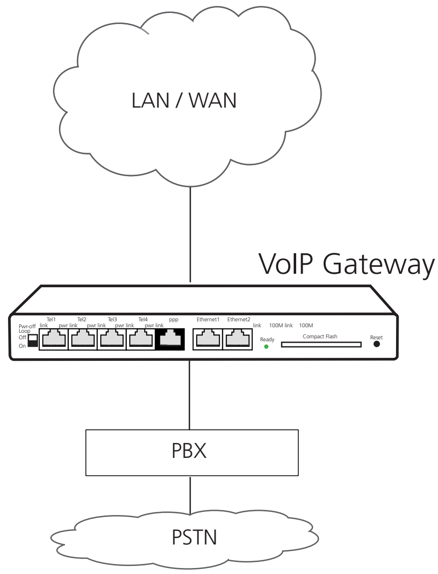
VoIP Gateway

Features

- Connects traditional telephony to IP telephony via Primary Rate Interfaces (PRI).
- Supports either 2 E1 per VoIP Gateway (up to 60 channels with EDSS1, QSIG, E1-CAS protocol), or 2 T1 per VoIP Gateway (up to 46 channels with QSIG protocol or up to 48 channels with T1-CAS protocol).
- Can be switched to ISDN, inserted into trunk lines or into cascade redundant systems.
- Work as Gatekeeper in the VoWiFi system
- Can be configured to hand on the PRI connections even with the power supply shut down.
- Scalable in both directions.
- Several gateways can be interconnected
- Internal memory can be increased by installing a "Compact Flash" Type 1 memory card.
- Two separate Ethernet interfaces
- Ethernet interfaces may take over routing functions between two networks.
- The second Ethernet interface may also be used for the connection with a second switch or be used as a management port.
- Supports up to 5000 Ascom endpoints.



Technical Specifications

Interfaces

4 × ISDN PRI:	2 × TE mode for trunk interface or 2 × TE and 2 × NT mode to insert in trunk lines
ISDN BRI (TEL):	ISDN interface configurable in TE/NT mode for routing, administration, synchronizing, backup or other
"Power-off" Loop	Interconnects two PRI interfaces in power off status
2 × Ethernet:	10/100-Base-TX auto negotiation Automatic recognition: Uplink / Downlink Power over Ethernet (PoE), IEEE 802.3af Both interfaces individually addressable LED for activity and 100Mbit Modus
Compact Flash:	Prepared for Compact Flash Cards Type 1

Physical

Size (l × w × d):	210 × 184 × 32 mm Designed to fit in 19`` rack using an additional frame (optional), one unit high (1U).
Weight:	1050 g
Material:	Stainless-steel

Power Supply:	Power over Ethernet (PoE), IEEE 802.3af or Internal mains adapter (European version only) 100 – 240V, 47 – 62 Hz, 15 W
Memory:	128 MB DRAM 16 MB Flash Remote firmware update
CPU:	RISC CPU for protocol processing Digital Signal Processor (DSP) for voice data processing for up to 60 channels

Environmental

Operating temperature:	0°C to +45°C
Humidity:	10% to 90% non-condensing
Storage temperature:	-10°C to +70°C

Voice over IP

Internet:	IP Internet Protocol – basis for TCP and UDP, DHCP Dynamic Host Configuration Protocol
H.323:	H.323 version 4 incl. H.225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450 RAS support for external Gatekeeper H.450.1 Defining Supplementary Services for H.323 H.450.2 Call Transfer H.450.3 Call Diversion H.450.4 Call Hold H.450.6 Call Waiting H.450.7 Message Waiting Indication H.450.8 Calling Party Name Presentation H.450.9 Completion of Calls to Busy Subscribers H.245 Fast Connect En-block dialling Overlapped sending
SIP	SIP version 2 (including HTTP digest authentication) conform RFC 3261 SIP over UDP, TCP, TLS (SIPS, V 7.0 or higher) RFC 2327 SDP: Session Description Protocol RFC 2396 URI generic syntax RFC 2617 Digest Authentication RFC 3261 SIP RFC 3264 An Offer/Answer Model with SDP RFC 3265 Session Initiation Protocol (SIP) - Specific Event Notification RFC 3326 The Reason Header Field for the Session Initiation Protocol RFC 3515 Sparks, The Session Initiation Protocol (SIP) Refer Method RFC 3891 SIP Replaces Header
Voice over IP:	RTP real time protocol – for speech transport SRTP secure speech transport – (V 7.0 or higher) RTCP real time control protocol – first level of “Quality of Service”
Fax over IP:	T.38 real time fax
DTMF:	H.245 Alphanumeric or Signal Type
Quality of Service:	TOS and DiffServ IEEE 802.1p / 802.1q

Specifications are subject to change without notice.

Voice Encoding:	G.711 A-law /µ-law (64 kbps), G.723.1 (5.3 and 6.3 kbps), G729A (16 kbps), G.726 (32 kbps), VAD (Voice Activity Detection), CNG (Comfort Noice Generation), Dynamic Jitter Buffering
Echo Compensation:	G.168

Administration

Access:	Via HTML/Web-Browser Password protected authentication
Troubleshooting:	Logs and Traces Status display of interfaces and connections PING SNMP Traps sending
Update:	Configuration recording / sending Bootcode and firmware update via HTML upload Automatic update via Update server

Data routing

DSL:	PPPoE Manual / automatic connection after boot
VPN:	PPTP Tunnelling Up to 32 tunnels simultaneously Encoding via MPPE
NAT:	For transformation of official IP addresses into private IP addresses and vice versa

Routing of Telephony connections

Connections:	VoIP-ISDN, ISDN-ISDN, VoIP-VoIP
Re-Routing:	Configuration of alternative routes
Billing:	Automatic Call Detail Records (CDR) generation
Calling Number Mapping:	Modifying of Calling and Called Party Number possible; delete, add or replace of dial prefixes
External Address Resolution:	Related to ENUM for H.323 and SIP protocols
Automatic Dial-tone generation:	European and US Standard
HTTP Interface:	Plays recorded messages deposed on a web server
ISDN protocols:	EDSS1, QSIG, T1-CAS, E1-CAS
QSIG-Interworking:	QSIG ECMA V.2

Additional Features

Time:	Exact time information via time server SNTP-Client & Server

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Compliance to European regulations and standards

Product marking:	CE
Safety:	EN/IEC 60950-1
EMC:	EN 55022 / EN 55024

Compliance to other regulations and standards

Product marking:	FCC ID: BXZIP6000 US: 9FVISNANIP6000 IC: 3724B-IP6000
EMC:	FCC 47 CFR Part 15 Subpart B Class B ICES-003
Hearing aid:	Hearing aid compatible according to relevant requirements in FCC Part 68 / Canadian CS-3 Part II

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