



QX-DFXS24

The QX-DFXS24 is a multi-functional analog VoIP gateway designed to provide seamless connectivity between IP telephony networks and legacy infrastructure, including PSTN/POTS telephones, fax machines, and traditional PBX systems. Utilizing the standard SIP protocol, it ensures full interoperability with SIP-based telephony systems.

Key Features

- **Capacity:** 24 FXS ports for analog device integration.
- **Advanced Capabilities:** Support for Fax over IP (FoIP) and flexible, customizable dial plans.
- **Reliability:** Delivers high-quality Voice-over-IP services suitable for call centers, SMEs, SOHO, and residential environments.
- **System Integration:** Fully compatible with Epygi QX and UC-line IP PBXs.
- **Simplified Deployment:** Supports auto-configuration for quick, scalable integration, allowing legacy analog phones to access most features and services typically reserved for Epygi-supported IP phones.

Capabilities

FXS ports	24
Ethernet LAN port	3
Ethernet WAN port	1
Fax over IP (T.38 and Pass-Through), IPv4 and IPv6 (Dual Stack), dial plan	

FEATURES

Telephony

- Call Hold
- Speed Dial
- Do-not-disturb
- 3-Way Conference
- Call Waiting
- Blind Transfer
- Attended Transfer
- Call Forward on Busy
- Call Forward on No Reply
- Unconditional Call Forward
- Message Waiting Indicator

Voice and FAX

- Hook Flash
- Modem/POS
- T.38/Pass-through
- Silence Suppression
- Comfort Noise Generation (CNG)
- Voice Activity Detection (VAD)
- Echo Cancellation (G.168), with up to 128ms
- Adaptive (Dynamic) Jitter Buffer
- Programmable Gain Control
- High-speed fax up to 14.4kbps
- VLAN 802.1P/802.1Q
- (Voice/data/management VLANs)
- Layer3 QoS and DiffServ
- Audio Codecs: G.711A/U law, G.729A/B, G.726, iLBC, Opus optional
- DTMF mode: Signal/RFC2833/INBAND

Connectivity

- Physical Interfaces:
 - 24 FXS ports (RJ-11 and RJ21), 1 LAN
 - Ethernet 10/100/1000 BASE-T port (RJ-45)
- Reversed Polarity
- Dial Mode: DTMF and Pulse
- Pulse: 10 and 20 PPS
- CallerID: DTMF/FSK
- Max Cable Length: 5KM
- Programmable Call Progress Tone

- Uplink connections:
 - 1 WAN Ethernet 10/100 BASE-T port (RJ-45)

- Phones:
 - 24 analog phones (or other analog devices) to connect via FXS ports

VoIP

- Protocol: SIP v2.0 (UDP/TCP), RFC3261, SDP(RFC2327), RTP(RFC2833), RFC3262, RFC3263, RFC3264, RFC3265, RFC3515, RFC2976, RFC3311, RFC3891, RFC3892
- SIP over TLS
- RTP/SRTP/RTCP, RFC2198, RFC1889
- RFC4028 Session Timer
- RFC3266 IPv6 in SDP
- RFC2806 TEL URI
- RFC3581 NAT, rport
- Primary/backup SIP server
- Outbound Proxy
- DNS SRV/ A Query/NATPR Query
- Early Media/Early Answer
- NAT: STUN, Static/Dynamic NAT

System

- CDR
- Web
- Auto Provisioning
- Ping/Tracert Test
- Network Capture
- Outward Test (GR909)

Environmental

- Power Supply: 100-240VAC, 50-60 Hz
- Power Consumption: 40W(Typical)
- Operating Temperature: 0°C ~ 45°C
- Storage Temperature: -20°C ~80°C
- Humidity: 10%-90% Non-Condensing
- Dimensions(W/D/H): 440*250*44mm
- Unit Weight: 3.2kg
- Compliance: CE, FCC