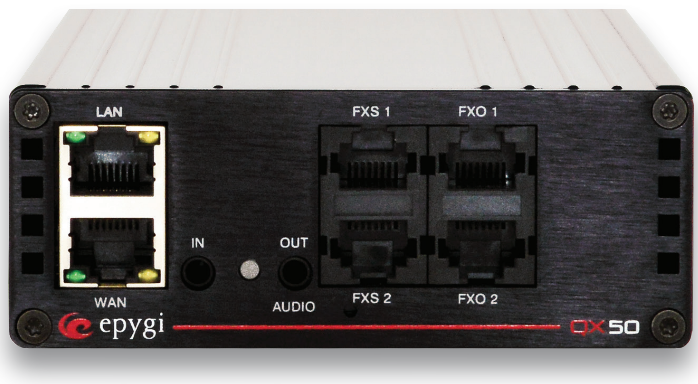

QX50

The QX50 IP PBX is designed for offices with as many as 202 users. The QX50 can support up to 200 IP devices and 16 concurrent calls. In addition, this system has two FXO analog PSTN connections and two FXS analog station ports. SIP trunking allows for the QX50 to connect directly to an ITSP with no additional equipment. The QX50 includes a firewall and SIP Intrusion Detection & Protection System for optimal security. Additional E1, T1, FXO and FXS ports can easily be provided using the Epygi QX Gateways. When rack-mounted and paired with an Epygi QX Gateway, power redundancy provides added protection.

Capabilities

Analog phones	2
IP phones	16
Additional IP phones with keys	184
Total phones	202
Concurrent calls	16
FXO PSTN ports	2
Ethernet LAN port	1
Ethernet WAN port	1
Audio In port	1
Audio Out port	1
SD slot	1

Interconnection *with* QX Gateways

GATEWAYS	Recommended Number (max)
QXFXO4	4
QXISDN4	2
QXE1T1	1
QXFXS24	2

FEATURES

Telephony

PBX Features
- Auto Attendant with standard and customizable scenarios and call history
- Call blocking, forwarding, hold, transfer, Call Relay and call waiting
- Caller ID detection and hiding caller ID
- Voicemail system
- Voicemail notification via SMS/email
- Caller ID-based voicemail profile
- Call park, call pickup, multicast paging, intercom
- Distinctive ringing
- Speed dial
- Many Extension Ringing, Call hunting
- Receptionist
- Call Park with Paging
- Call Park on Auto Attendant
- Call back from Auto Attendant
- Emergency Call Alert
- Hold music
- Call history with archiving
- Do Not Disturb
- Global speed dial
- Find Me / Follow Me
- Unified Messaging
- Three-way conferencing
- Hotline Service
- G3 fax support: T.38 and clear channel fax
- Universal Extension Recordings
- Busy auto redial
- Directory assistance, Dial by Name
- Phone Book
- Dial plans (call routing), time of day routing
- Scheduling, Day/Night Switching
- Alarm
- Dial & Announce (D&A)
- Class of Service
- Call queue
- Hot Desking
- Parent-Child extension configuration
- Local Authentication for making call
- PIN code Barring
- Calling Cost Control*
- Redundancy*
- Automatic Call Distribution (ACD)*
- Epygi ACD Console (EAC)*
- Epygi Automatic Outbound Calling (AOC)*
- Voicemail Transcription*
- CRM Integration*
- Call Recording (12 ports)*
- Barge-In*
- Conference Server* Audio (16 ports) / Video (8 ports)
- eQall Softphone*
- eQall SMS/WhatsApp Messaging*
- eQall Receptionist Console*

PC-Based Applications
- QX-Quadro Configuration Console (QCC)
- Epygi Media Streamer (EMS)
- Auto Dialer*

Voice and Video Features

Voice Coding:
G.711, G.726 (16, 24, 32, 40 Kbps), G.729A, iLBC (13,33 kbit/s, 15,2 kbit/s); VAD, CNG, G.168 echo cancellation G.722 and G.722.1 pass-through point-to-point HD call, OPUS

Video Coding:
H.263, H.263+ and H.264 pass-through point-to-point video call

VoIP Encryption:
SRTP

VoIP Signaling:
SIP v2, SIP/TLS

DTMF:
In band & out of band signaling support

VoIP Data and Signaling Protocols
- ITU-T G.711, G.726, G.729 Annex A, G.168-2000, 2002, Q.23, Q.24;
- IETF RFC 3951- iLBC;
- Telcordia (Bellcore) GR.506, GR.181; ETS_300 659_1,2,3;
- SIP, SIP/TLS (RFCs: 2246, 3261, 3263, 3265, 3311, 3323, 3428, 3515, 3578, 3581, 3842, 3856, 3863, 3891, 3892, 4028, 4235)
- SDP (RFC: 2327, 4568)
- RTP/SRTP (RFCs: 1889, 1890, 3389, 3550, 3551, 3555, 3711, 4733, 3952)
- Fax over IP (ITU-T: T4, T30, T38, V17, V21, V27 ter, V29)

POTS Signaling
- Loop start
- FSK and DTMF caller ID support
- FSK message waiting indicator support

Connectivity

Physical interfaces

Premise connections:
- 2 FXS short-loop FXS ports (RJ-11)
- 1 LAN Ethernet 10/100 BASE-T port (RJ-45)

Uplink connections:
- 2 FXO ports to the central office (RJ11)
- 1 WAN Ethernet 10/100 BASE-T (RJ45)

Audio port connections:
Line-in/line-out (line-in signal level - 0.5V RMS, Line-out R_{load} - 600Ohm to 10K Ohm)

Phones

IP phones:
- 16 IP phones by default
- 184 additional IP phones may be added with feature keys
- All IP phones can be connected both from LAN or WAN side or as remote extensions
- Auto provisioning support for all IP phones from selected manufacturers
- PnP configuration support for the most of IP phones from selected manufacturers
- Auto configuration using OpenVPN service for some of selected IP phones

Analog phones:
2 analog phones (or other analog devices) to connect via FXS ports

Auto Attendants and Virtual extensions

Auto Attendants:
Up to 250 Auto Attendants can be added**

Virtual extensions:
Up to 250 virtual extensions can be added**

System Capacity
- Up to 16 simultaneous VoIP calls with external parties
- Unlimited station-to-station calling for IP phones
- Two analog PSTN calls with external parties

Memory Storage
- SD card

Network

- STUN/Network Address Translation (NAT) traversal (RFC 3489)
- IPSec VPN with 3DES and AES encryption in tunnel mode (RFCs: 2402, 2406, 2409)
- Automatic Internet Key Exchange (IKE) keying support
- PPTP VPN, L2TP VPN
- Firewall security via:
  - Intrusion Detection & Protection System (IDS/IPS)
  - Network Address Translation (NAT)
  - Policy and service-based filtering
  - Stateful inspection firewall
- SIP Intrusion Detection System (SIP IDS)
- DHCP server on the LAN side
- DHCP client on the WAN side
- DNS server with forwarding functionality
- Simple Network Time Protocol (SNTP) server/client for computer clock synchronization
- PPPoE connection to the ISP with PAP/MS CHAP authentication
- IP DIFFSERV for QoS
- SIP tunneling
- Virtual LAN (VLAN/IEEE 802.1Q)
- Mail client to send voice and fax messages as email attachments (.wav and .tif/.pdf respectively) and system notifications
- DNS (DYNDNS) support with third party NAT/router with port forwarding and port translation

System

Management
- Multilingual web interface accessible from LAN and WAN (HTTP/HTTPS)
- Password control
- User rights management
- Remote diagnostics and software upgrade
- VoIP Carrier Wizard
- Monitoring via ecMON
- Download/restore configuration
- Legible and editable configuration files
- Auto configuration of IP phones via TFTP and HTTP
- SNMP monitoring and configuration
- Third Party Call Control (3PCC) XML RPC*
- Reset button with factory reset option
- Custom language pack
- System event notification via SMS/email
- Emergency recovery

Diagnostics/Testing
- System status LED
- Remote testing
- FXO and network diagnostics
- Security diagnostics
- System logs, SIP IDS logs
- Call capture

Billing and Statistics
- Radius Client (RFCS: 2865, 2866), Call Detail Records (CDR)

Environmental

Physical Dimensions

Rack-mountable devices:

Measurements:
8.0” x 4.0” x 1.6” (20.5 x 10.5 x 4.0 cm)

Weight:
1.26 lbs. (570 g)

Conditions

Operating temperature:
41°F - 104°F (5°C - 40°C)

Storage temperature:
41°F - 140°F (5°C - 60°C)

Non-condensing humidity:
5% - 90%

Powering Options
- Input: 85-264VAC, 47-63Hz, AC
- Auxiliary output power: 12.0VDC, 0.6A (max)

Power Consumption
- 2.7W (idle), 6.5W (max)

* Requires a software license key
** The total number of extensions used for IP phones, analog phones, Auto Attendants and virtual extensions can not exceed 250.

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