



# QX50

The QX50 IP PBX is designed for offices with as many as 50 users. The QX50 can support up to 48 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 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	32
Total phones	50
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

<b>GATEWAYS</b>	Recommended Number (max)
<b>QXFXO4</b>	4
<b>QXISDN4</b>	2
<b>QXE1T1</b>	1
<b>QXFXS24</b>	2

# FEATURES

## Telephony

### PBX Features

- Auto Attendant with standard and customizable scenarios
- 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, paging, intercom
- Distinctive ringing
- Speed dial
- Many Extension Ringing
- Receptionist
- Call hunting
- Call back from Auto Attendant
- Emergency Call Alert
- Hold music
- Call history
- 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
- Phone Book
- Dial plans (call routing), time of day routing
- Scheduling, Day/Night Switching
- Alarm
- Dial & Announce (D&A)
- Class of Service
- Call queue
- Calling Cost Control\*
- Redundancy\*
- Automatic Call Distribution (ACD)\*
- Epygi ACD Console (EAC)\*
- Epygi Automatic Outbound Calling (AOC)\*
- Call Recording (12 ports)\*
- Barge-In\*
- Conference Server\*
  - Audio (16 ports) / Video (8 ports)
- iQall Advanced Features\*

### PC-Based Applications

- Desktop Communication Console (DCC)\*
- QX-Quadro Configuration Console (QCC)
- Epygi Media Streamer (EMS)
- Epygi Hotel Console (EHC)\*
- 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:
  - SRTTP
- 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
  - 32 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 200 Auto Attendants can be added

### Virtual extensions:

- Up to 200 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 System (IDS)
  - 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
- 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
- Extension status watching (with DCC)
- 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 200.