QX50

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

<table>
<thead>
<tr>
<th><strong>Interconnection with QX Gateways</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>GATEWAYS</strong></td>
</tr>
<tr>
<td>QXFXO4</td>
</tr>
<tr>
<td>QXISDN4</td>
</tr>
<tr>
<td>QXE1T1</td>
</tr>
<tr>
<td>QXFXS24</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Count</th>
</tr>
</thead>
<tbody>
<tr>
<td>Analog phones</td>
<td>2</td>
</tr>
<tr>
<td>IP phones</td>
<td>16</td>
</tr>
<tr>
<td>Additional IP phones with keys</td>
<td>32</td>
</tr>
<tr>
<td>Total phones</td>
<td>50</td>
</tr>
<tr>
<td>Concurrent calls</td>
<td>16</td>
</tr>
<tr>
<td>FXO PSTN ports</td>
<td>2</td>
</tr>
<tr>
<td>Ethernet LAN ports</td>
<td>1</td>
</tr>
<tr>
<td>Ethernet WAN ports</td>
<td>1</td>
</tr>
<tr>
<td>Audio In port</td>
<td>1</td>
</tr>
<tr>
<td>Audio Out port</td>
<td>1</td>
</tr>
<tr>
<td>SD slot</td>
<td>1</td>
</tr>
</tbody>
</table>
**Telephony**

**PBX Features**
- Multi-level Auto Attendant with Interactive Voice Response (IVR) and VoiceXMLv2 support
- 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, Pacing, Intercom
- Distinctive ringing
- Speed dialing
- Many Extension Ringing
- Receptionist
- Call Hunting
- Automated Call Back from Auto Attendant
- Hold Hunting
- Call Detail Records
- Do Not Disturb
- Global speed dial
- Find Me/Find Me
- Unified Messaging
- Three-Way Conferencing
- Hotline service
- G3 fax support: T38 and clear channel fax
- Unified Fax Messaging
- Busy auto-redial
- Directory assistance
- Dial plans (call routing), time of day routing
- Class of Service
- Call Queue
- Redundancy
- Automatic Call Distribution*
- Call Recording (12 ports)* - It is advisable to use an SD memory card to expand the system memory.
- Barge-in*
- Audio (16 ports) / Video (8 ports)
- Conference Server*
- Mobile Tagging*

**Licensable PC-Based Applications**
- Desktop Communication Console (DCC)*
- Auto Dialer***

***Requires a software license key

**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 pass-through point-to-point HD call

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

**VoIP Encryption:**
- SRTP
- VoIP Signaling:
  - SIP, SIP/TLS
  - DTMF:
- In band & out of band signaling support

**VoIP Data and Signaling Protocols**
- IETF RFC 3951 - iLBC;
- Telcordia (Bellcore) GR.506, GR.181;
- ETSI, 300 659, 1.2.3;
- SIP, SIP/TLS (RFCs: 2246, 2361, 2363, 2665, 3131, 3323, 3324, 3325, 3428, 3515, 3578, 3581, 3725, 3842, 3856, 3863, 3891, 3892, 4028, 4235)
- SDP (RFC: 2237, 4568)
- RTP/SRTP (RFCs: 1889, 1890, 2833, 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 R0ad - 600Ohm to 10K Ohm)

**Phones**
- IP phones:
  - 16 SIP phones by default
  - 32 additional SIP phones may be added with feature keys
  - All SIP phones can be connected both from LAN or WAN side
  - Plug-and-Play (PnP) with select IP Phone manufacturers
- Analog phones:
  - 2 analog phones (or other analog devices) to connect via FXS ports

**Auto Attendants and Virtual Extensions**
- Auto Attendants:
  - Up to 20 standard and custom Auto Attendants can be registered
- Virtual Extensions:
  - Up to 200 Virtual Extensions can be registered**

**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 (SPCC) 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), CDRs

**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%
- Power Supply
  - Input: 85-264VAC, 47-63Hz, AC
  - 0.4A/115VAC, 0.2A/230VAC
  - Auxiliary output power: 12.0VDC, 0.6A (max)