With the ability to support 64 concurrent calls, the QX200 IP PBX is designed for offices with up to 200 employees. The system has four FXO ports in order to connect to the PSTN and two FXS ports for analog phones and fax machines. SIP trunking allows for the QX200 to connect directly to an ITSP with no additional equipment. The QX200 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.

### Interconnection with QX Gateways

<table>
<thead>
<tr>
<th>GATEWAYS</th>
<th>Recommended Number (max)</th>
</tr>
</thead>
<tbody>
<tr>
<td>QXFXO4</td>
<td>16</td>
</tr>
<tr>
<td>QXISDN4</td>
<td>8</td>
</tr>
<tr>
<td>QXE1T1</td>
<td>2 (E1 mode), 3 (T1 mode)</td>
</tr>
<tr>
<td>QXFXS24</td>
<td>8</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>24</td>
</tr>
<tr>
<td>Additional IP phones with keys</td>
<td>176</td>
</tr>
<tr>
<td>Total phones</td>
<td>202</td>
</tr>
<tr>
<td>Concurrent calls</td>
<td>64</td>
</tr>
<tr>
<td>FXO PSTN ports</td>
<td>4</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 Voice Mail system
Voicemail notification via SMS/email
Caller ID-based voicemail profile
Call Park, Call Pickup, Paging, Intercom
Distinctive ringing
Speed dialing
Many Extension Ringing Receptionist
Call Hunting
Automated Call Back from Auto Attendant Hold music
Call Detail Records
Do Not Disturb
Global speed dial
Find Me/Follow 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 (20 ports)* - It is advisable to use an SD memory card to expand the system memory.
Barge-In*
Audio (32 ports) / Video (10 ports)
Conference Server*
Mobile Toggling*
Licenseable 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
ITU-T G.711, G.726, G.729 Annex A,
IETF RFC 3951 - ILBC;
Telscript (Bellcore) GR.506, GR.181;
ETS_300 659 1.2.3;
SIP, SIPS/TLS (RFCs: 2246, 3261, 3263, 3265, 3311, 3322, 3324, 3325, 3428, 3515, 3578, 3581, 3725, 3842, 3856, 3863, 3891, 3892, 4028, 4235)
SDP (RFC: 2327, 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:
4 FXO ports to the central office (RJ11)
1 LAN Ethernet 10/100 BASE-T (RJ45)
Audio port connections:
Line-in/line-out (line-in signal level - 0.5V RMS, Line-out Roload - 600Ohm to 10K Ohm)
Phones:
IP phones:
24 SIP phones by default
176 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 400 and custom Auto Attendants can be registered
Virtual Extensions:
Up to 400 Virtual Extensions can be registered**
**The total number of extensions used for IP phones, analog phones, Auto Attendants and Virtual Extensions can not exceed 400.

System Capacity
Up to 64 simultaneous VoIP calls with external parties
Unlimited station-to-station calling for IP phones
Unlimited station-to-station calling for analog phones
Four analog PSTN calls with external parties
Memory Storage
SD card

Network
STUN/Network Address Translation (NAT) traversal (RFC 3489)
IPSec VPN with DES, 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 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/store configuration
Legible and editable configuration files
Auto configuration of IP phones via TFTP and HTTP
SNMP monitoring and configuration
Third Party Call Control (3PCC) XMLRPC*
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: 2665, 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:
1.28 lbs. (580 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)