



# QX200

With the ability to support 64 concurrent calls, the QX200 IP PBX is designed for offices with up to 258 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 & 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	24
Additional IP phones with keys	232
Total phones	258
Concurrent calls	64
FXO PSTN ports	4
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	16
QXE1T1	2 (E1 mode), 3 (T1 mode)
QXFXS24	8

# 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
- Authorized Phones
- 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\*
- Call Recording Transcription\*
- Text-to-Speech\*
- CRM Integration\*
- Call Recording (32 ports)\*
- Barge-In\*
- Conference Server\*
- Audio (32 ports) / Video (16 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, CNNG, 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:  
4 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:  
24 IP phones by default  
232 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  
IP phones Proxy Connection

### Analog phones:

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

### Auto Attendants and virtual extensions

Auto Attendants:  
Up to 400 Auto Attendants can be added\*\*  
Virtual extensions:  
Up to 400 virtual extensions can be added\*\*

### System Capacity

Up to 64 simultaneous VoIP calls with external parties  
Unlimited station-to-station calling for IP phones  
Four 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:

1.28 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%

### Powering Options

Input: 85-264VAC, 47-63Hz, AC

Auxiliary output power: 12.0VDC, 0.6A (max)

### Power Consumption

2.8W (idle), 6.7W (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 400.