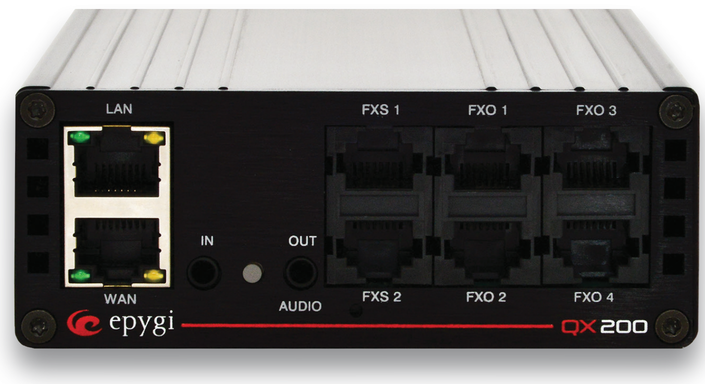


# QX200

The QX200 IP PBX is designed for small to mid-sized businesses, supporting up to 258 users and 64 concurrent calls. It provides integrated VoIP, PSTN connectivity, and advanced security features, making it a reliable and scalable communication solution. 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 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 by default | 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 (Default)

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
Media Streamer
Parent-Child extension configuration
Local Authentication for making call
PIN code Barring

## Telephony (Key Required)

Calling Cost Control
Redundancy
Automatic Call Distribution (ACD)
Epygi ACD Console (EAC)
Epygi Automatic Outbound Calling (AOC)
AI Transcription
- Voice Mail (VM)/Call Recording (CR) Transcription (with Diarization)
- Voice-Enabled Auto Attendant (IVR)
- Epygi Virtual Assistant (EVA)

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
Third Party Call Control (3PCC) XML RPC

PC-Based Applications
QX-Quadro Configuration Console (QCC)
Auto Dialer

## Voice/Video Protocols

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:
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
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)

* The total number of extensions used for IP phones, analog phones, Auto Attendants and virtual extensions can not exceed 400.

TD-QX200-Let-21

Epygi Technologies, LLC
561 Maitland Ave
Altamonte Springs, Florida 32701



Tel: (+1) 972.692.1166
Web: www.epygi.com
E-mail: sales@epygi.com