



QX20

The QX20 IP PBX is designed for offices with as many as 32 users. The QX20 can support up to 32 IP devices and 10 concurrent calls. SIP trunking allows for the QX20 to connect directly to an ITSP with no additional equipment. The QX20 includes a firewall and SIP Intrusion Detection for optimal security. E1, T1, FXO, ISDN BRI and FXS ports can easily be provided using the Epygi QX Gateways.



IP phones	12
Additional IP phones with keys	20
Total phones	32
Concurrent calls	6
Additional concurrent calls with keys	4
Ethernet LAN port (10/100/1000)	1
Ethernet WAN port (10/100/1000)	1
Audio In port	1
Audio Out port	1
SD slot	1



GATEWAYS	Recommended Number (max)
QXFXO4	4
QXISDN4	2
QXE1T1	1
QXFXS24	1

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
- Hold music
- Call history
- Do Not Disturb
- Global speed dial
- Find Me / Follow Me
- Unified Messaging
- Three-way conferencing
- G3 fax support: T.38 and clear channel fax
- Universal Extension Recordings
- Busy auto redial
- Directory assistance
- 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
- Redundancy*
- Call Recording (10 ports)*
- Conference Server*
- Audio (16 ports)/Video (8 ports)
- iQall Mobile Toggling*

PC-Based Application

- Desktop Communication Console (DCC)*
- 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, CNG, G.168 echo cancellation G.722 and G.722.1 pass-through point-to-point HD call

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)

Connectivity

Physical interfaces

Premise connections:

- 1 LAN Ethernet 10/100/1000 BASE-T port(RJ-45)
- 1 Ethernet 10/100 BASE-T port (RJ-45) for redundancy connection

Uplink connections:

- 1 WAN Ethernet 10/100/1000 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)

USB connections:

- USB host, type A
- USB OTG, mini type AB

IP Phones

- 12 IP phones by default
- 20 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

Auto Attendants and Virtual extensions

Auto Attendants:

Up to 100 Auto Attendants can be added

Virtual extensions:

Up to 100 virtual extensions can be added**

System Capacity

- Up to 10 simultaneous VoIP calls with external parties
- Unlimited station-to-station calling for IP phones

Memory Storage

microSD 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

OpenVPN

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 .tiff/.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

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

Input from type AB mini-USB,

Input from 5VDC/2A adapter

Power Consumption

2.7W (idle), 6.5W (max)

* Requires a software license key

** The total number of extensions used for IP phones, Auto Attendants and virtual extensions can not exceed 100.