



# QX3000

The QX3000 IP PBX supports offices with up to 3,000 users. Any number of required FXS, FXO, ISDN BRI, E1 and T1 analog or digital ports can be easily added by interconnecting with Epygi QXFXO4, QXISDN4, QXE1T1 and QXFXS24 Gateways. It can connect to the PSTN using Gateways or directly with SIP trunks through an ITSP. In addition, the QX3000 can support up to 500 concurrent calls.



IP phones	200
Additional IP phones with keys	2,800
Total IP phones	3,000
Concurrent calls	500
Ethernet LAN port	1
Ethernet backup	1



<b>GATEWAYS</b>	Recommended Number (max)
<b>QXFXO4</b>	32
<b>QXISDN4</b>	32
<b>QXE1T1</b>	16 (E1 mode) 20 (T1 mode)
<b>QXFXS24</b>	100

# 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
- Dial plans (call routing), time of day routing
- Scheduling, Day/Night Switching
- Alarm
- Dial & Announce (D&A)
- Class of Service
- Calling Cost Control\*
- Redundancy\*
- Call queue
- Automatic Call Distribution (ACD)\*
- Epygi ACD Console (EAC)\*
- Call Recording (240 ports)\*
- Barge-In\*
- Conference Server\*
  - Audio (288 ports)/Video (104 ports)
- Auto Dialer application support\*
- iQall Mobile Toggling\*

### PC-Based Applications

- Desktop Communication Console (DCC)\*
- QX-Quadro Configuration Console (QCC)
- Epygi Media Streamer (EMS)
- Epygi Hotel Console (EHC)\*
- 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.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; IETF RFC 3951- iLBC;

- 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

#### Network connections:

- 2 Ethernet 10/100/1000 BASE T (RJ45)

### IP Phones

- 200 IP phones by default
- Up to 2,800 additional IP phones may be added with feature keys
- All IP phones can be connected both from LAN 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 3,400 Auto Attendants can be added

#### Virtual extensions:

- Up to 3,400 virtual extensions can be added\*\*

### System Capacity

- Up to 500 simultaneous VoIP calls with external parties
- Unlimited station-to-station calling for IP phones

### Emergency Repair Boot-up Device

- DVD-ROM

## Network

- STUN/Network Address Translation (NAT) traversal (RFC 3489)

#### Firewall security via:

- Policy and service-based filtering

- Stateful inspection firewall

- SIP Intrusion Detection System (SIP IDS)

- DHCP server on the LAN side

- DNS server with forwarding functionality

- Simple Network Time Protocol (SNTP)

- server/client for computer clock

- synchronization

- 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

## System

### Management

- Multilingual web interface accessible from LAN (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\*
- Extension status watching (with DCC)
- Custom language pack
- System event notification via SMS/email
- Emergency recovery

### Diagnostics/Testing

- System logs
- 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:

- 16.8" x 14" x 1.7" (42.6 x 35.6 x 4.3cm)

#### Weight:

- 15 lbs (6.8 kg)

### Conditions

#### Operating temperature:

- 50°F - 95°F (10°C - 35°C)

#### Storage temperature:

- 31°F - 140°F (-35°C - 60°C)

#### Non-condensing humidity:

- 5% - 90%

### Power Supply

- 100 - 240V, 50-60Hz, 4A (max)

### Regulatory Compliance

- Power Supply Safety/EMC

- USA - UL listed, FCC

- Canada - CUL listed

- Germany - TUV Certified

- Europe/CE Mark

- EN 60950/IEC 60950-Compliant

\* Requires a software license key

\*\* The total number of extensions used for IP phones, Auto Attendants and virtual extensions can not exceed 3,400.