MICROTEL TECHNOLOGY

Simple solutions for your communication needs

QX2000

The QX2000 IP PBX supports offices with up to 2,000 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 QX2000 can support up to 300 concurrent calls.

IP phones	200
Additional IP phones with keys	1,800
Total IP phones	2,000
Concurrent calls	300
Ethernet LAN ports	1
Ethernet backup	1

Interconnection with QX Gateways

GATEWAYS	Recommended Number (max)
QXE1T1	16
QXISDN4	32
QXFXO4	32
QXFXS24	80

FEATURES

Telephony

PBX Features
- Multi-level Auto Attendant with Interactive Voice Response (IVR) and VoiceXML v2 support
- 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
- Automated 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
- Class of Service
- Call queue
- Redundancy
- Automatic Call Distribution (ACD)*
- Call Recording (152 ports, total max size is 108 days)*
- Barge-In*
- Audio (288 ports)/Video (104 ports) Conference Server*
- Auto Dialer application support*
- Mobile Toggling*

Licensable 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.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)

Phones
- IP phones:
  200 SIP phones by default
  Up to 1,800 additional SIP phones may be added with feature keys
  Plug-and-Play (PnP) with select IP Phone manufacturers

Auto Attendants and virtual extensions
- Auto Attendants:
  Up to 2,400 standard and custom Auto Attendants can be registered
- Virtual extensions:
  Up to 2,400 virtual extensions can be registered**

**The total number of extensions used for IP phones, analog phones, Auto Attendants and virtual extensions can not exceed 2,400.

System Capacity
- Up to 300 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
- 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

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*
- 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