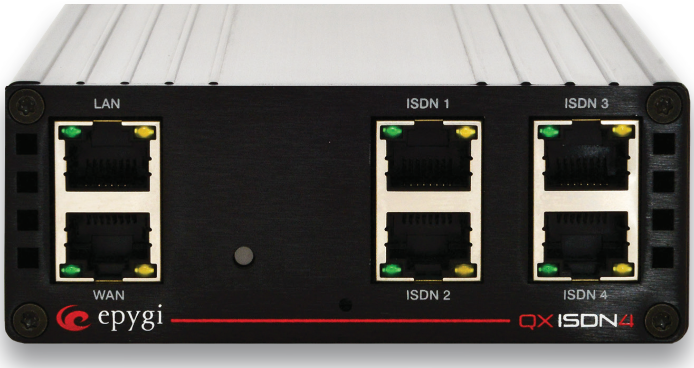

QXISDN4
QXISDN4+

The QXISDN4 Gateway includes four ISDN BRI ports to connect to an ISDN PBX. It can also connect to the ISDN lines from a central office or direct to an ISDN phone. It is designed to add inbound lines and balance outbound call volumes. It's a stand-alone SIP Gateway device that includes a VPN router, firewall, VPN capability and an Auto Attendant with standard and customizable scenarios. Integrating this product with any QX IP PBX allows the Gateway to then be managed through the IP PBX's GUI.

The QXISDN4+ activation license key is available to purchase for the QXISDN4. Once the QXISDN4+ activation license key has been installed and activated, the Gateway will function as an IP PBX and cannot be changed back. The QXISDN4+ can support 16 IP phones by default and almost all the applicable QX50 features that can be purchased and activated. The QXISDN4+ doesn't support Audio In, Audio Out and a SD card.

Capabilities

ISDN BRI ports	4
Ethernet LAN port	1
Ethernet WAN port	1
Call Routing capable of modifying caller ID or time of day routing	
Firewall, VPN Router, Auto Attendant, Stacking Options, Failover	

FEATURES

Telephony

PBX Features
- Auto Attendant with standard and customizable scenarios
- Call history with archiving
- Call blocking, unconditional call forwarding
- G3 fax support: T.38 and clear channel fax
- Dial plans (call routing), time of day routing
- Caller ID detection

Additional PBX Features*
- Call hold, transfer, Call Relay and call waiting
- 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, global speed dialing
- 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
- Do Not Disturb
- Find Me / Follow Me
- Unified Messaging
- Three-way conferencing
- Universal Extension Recordings
- Busy auto redial
- Directory assistance, Dial by Name
- Phone Book
- 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**
- Call Recording** (12 ports)
- Barge-In**
- Conference Server** Audio (16 ports) / Video (8 ports)
- eQall Softphone*
- eQall SMS Messaging*
- eQall Receptionist Console*

PC-Based Applications*
- QX-Quadro Configuration Console (QCC)
- Epygi Media Streamer (EMS)
- Epygi Hotel Console (EHC)**
- Auto Dialer**

Voice 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

VoIP Encryption:
SRTP

VoIP Signaling:
SIP v2, SIP/TLS

DTMF:
In band & out of band signaling support

Additional Voice Features*

Voice Coding:
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, OPUS

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)

ISDN BRI Signaling
- ITU-T: Q.921, Q.931 (DSS1), Q.951; ETSI ETS300 102 (NET5); NTT INS1500 for Japan

ISDN Features
- Supported modes: NT/TE BRI expansion for QX IP PBXs

Connectivity

Physical Interfaces

Premise connections:
1 LAN Ethernet 10/100 BASE-T port (RJ-45)

Uplink connections:
1 WAN Ethernet 10/100 BASE-T (RJ45)

ISDN Connection
4 ISDN BRI ports to the central office or local PBX (RJ45), NT and TE modes supported

Phones*

IP phones:
- 16 SIP phones by default
- 32 additional SIP phones may be added with feature keys
- All SIP phones can be connected both from LAN or WAN side
- Plug-and-Play (PnP) with selected IP Phone manufacturers

Auto Attendants and virtual extensions

Auto Attendants:
Up to 200 Auto Attendants can be registered***

Virtual extensions:
Up to 200 virtual extensions can be registered***

System Capacity*
- Up to 16 simultaneous VoIP calls with external parties
- Unlimited station-to-station calling for IP phones

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)
- DNS support with third party
- NAT with port forwarding and translation Mail client to send voice and fax messages as email attachments (.wav and .tif/.pdf respectively) and system notifications*

System

Management
- Operation modes: Master/Slave
- Easy interconnection with QX IP PBXs
- Web interface accessible from LAN and WAN (HTTP/HTTPS), the WAN management access can be switched off
- Password control
- User rights management
- Remote diagnostics and software upgrade
- Download/restore configuration
- Reset button with factory reset option
- Custom language pack
- System event notification via SMS/email
- Emergency recovery

Additional Management features*
- Third Party Call Control (3PCC) XML RPC**
- Auto configuration of IP phones via TFTP and HTTP

Diagnostics/Testing
- System Status LED
- Remote testing
- ISDN 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. (580g)

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
- 2.25W (idle), 4.6W (max)

* Available with QXISDN4+ key
** Requires a software license key
*** The total number of extensions used for IP phones, Auto Attendants and virtual extensions can’t exceed 200.

TD-QXISDN4-Let-11

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