QuadroM32x

2009 Internet Telephony Product of the Year

QuadroM32x: The Enterprise IP PBX Solution

The QuadroM32x is designed to deliver greater IP Line capacity, reaching 192 registered extensions and 64 concurrent calls. Epygi's IP PBX continues to deliver all the features currently found on our existing Quadro line with the addition of some vital enterprise-grade tools.

Audio and Video Conferencing and Automatic Call Distribution are a few of the unique features to this product. Third party integration with Unified Multimedia Communications platforms is another key solution for large businesses.

Expanding the digital and analog trunking capacity of the QuadroM32x is simple using Epygi Quadro Gateways. Epygi's FXO 4, ISDN, and E1/T1 Gateways with the Quadro and QuadroM IP PBX line. This flexibility will allow our customers to satisfy any configuration need.

Integrated Conference Server

Audio and Video Conferencing is a common feature used by today's large organizations. The QuadroM32x features an optional 64 person audio conference bridge with a vast array of built-in features. The presenter can mute participants, assign speakers and track overall activity. Regular participants can also view the active meeting and can indicate a request to speak during a muted session. This productivity enhancing tool is easily enabled using a software license key.

Redundancy Options

The QuadroM32x also offers a built-in E1/T1 digital trunk interface. This link can be used as the primary interface or as a redundant link while utilizing an ITSP as the primary service. Redundant Ethernet links are also included for secondary failover networks or a voice DMZ. A secondary QuadroM32x can be added as a hot stand-by option for increased redundancy.

What are Your VoIP BENEFITS?

- Large Capacity
- Increased Reliability & Redundancy
- True Enterprise Grade Solution

epygi®

Features

Telephony

PBX Features
- Call blocking, Forwarding, Hold, Transfer
- Call relay, Call waiting, Caller ID Detection
- Voice mail
- Call park, Pickup, Paging, Intercom
- Multi-level auto attendant with Interactive Voice Response (IVR) and VoiceXMLv2 support
- Voice mail with SMS notification
- Distinctive ringing
- Speed dialing
- Many extension ringing
- Receptionist
- Call hunting, Hiding Caller ID
- Automated Call back from Auto Attendant
- Hold music
- Call statistics
- Do Not Disturb
- Unified messaging
- 3-way conferencing
- Hotline service
- T.38 fax, fax relay and clear channel fax
- Unified Fax Messaging
- Busy auto-redial
- Directory assistance
- Dial plans (call routing)
- Time of day routing
- Call Queue
- Redundancy
- Voice Mail profile
- Automatic Call Distribution*
- Call Recording (32 ports) *
- Barge-in *
- Audio (64 ports)/Video (16 ports) Conference Server *

*Requires a software license key

Licensable PC-Based Applications
- Desktop Communication Console (DCC)
- 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 pass-through point-to-point HD call
- Video Coding:
  H.263 and H.264 pass-through point-to-point video call
- VoIP Encryption:
  SRTP
- VoIP Signaling
  SIP, SIP/TLS
- DTMF
  In band & out of band signaling support.

VoIP Data and Signaling Protocols
- ITU-T G711, G.726, G729 Annex A;
- IETF RFC 3951- iLBC; ITU-T Q.23, Q.24;
- Bellcore GR.506, GR.181; ITU-T G.168-2000, 2002; ETS_300659_1,2,3;
- SIP, SIPS/TLS (RFCs: 2246, 3261, 3263, 3265, 3311, 3323, 3324, 3325, 3428, 3515, 3578, 3581, 3725, 3842, 3856, 3863, 3891, 3892, 4028, 4235)
- SDP (RFC: 2327, 4568)
- RTP/SRTP (RFCs: 1889, 1890, 2833, 3389, 3550, 3551, 3555, 3711, 4733, 3952),
- Fax over IP (ITU-T: T4, T30, T38, V17, V21,V27 ter, V29)

Primary Rate ISDN (PRI) Signaling
- ITU-T: Q.921, Q.931 (DSS1), Q.951;
- ETSI ETS300 102 (NET5); ECMA-143-(QSIG); SR-NWT-002120 (NI2)
- NTT INS1500 for Japan
- PRI switch types: DSS1, NET5, QSIG, 5ESS, NTT INS1500, DMS 100

CAS Signaling
- CAS (MELCAS, ITU, ITU-T2, ITU-T: Q.400, Q.411, Q.421, Q.422, Q.440-Q.442, Q.450-Q.452, Q.454, Q.455, Q.457, Q.458, Q.460-Q.468, Q.470-Q.476
- Types: Loop Start, Ground Start; E&M Delay Dial, E&M Wink Start, E&M Immediate Start, E&M FGD
- R2 DTMF, R2 compelled, R2 non-compelled, R2 compelled with ANI, R2 non-compelled with ANI;
- R2 Parameters for Brazil and Mexico etc.)
- ANSI T1.403.02-199, T1.403.02a-2001

Connectivity

Physical interfaces
- Premise connections:
  2 short-loop FXS ports (RJ-11)
  1 LAN Ethernet 10/100 BASE T port (RJ45)
  1 Ethernet 10/100 BASE T port (RJ45)
- Uplink connections:
  1 E1/T1 port to the Central Office (RJ45)
  1 WAN Ethernet 10/100 BASE T (RJ45)

Phones
- IP phones:
  32 SIP phones by default
  Up to 160 additional SIP phones may be added with feature keys
  All SIP phones can be connected both from LAN or WAN side
  Plug-and-Play with select IP Phone manufactures
- Analog phones:
  2 Analog phones (or other analog devices) to connect via FXS ports

Auto Attendants and Virtual Extensions
- Auto Attendants:
  Up to 400 standard and custom AA can be registered
- Virtual Extensions:
  Up to 400 Virtual Extensions can be registered**

**The total number of extensions used for IP phones, Analog phones, AA and virtual extensions can not exceed 400 extensions.

System Capacity
- Up to 64 simultaneous VoIP calls with external parties
- Unlimited station to station calling for IP phones
- Unlimited station to station calling for analog phones
- 30/24 PSTN calls via E1/T1 with external parties

External Storage
- Compact Flash

Internet

- STUN/NAT traversal (RFC 3489)
- IPSec VPN with DES, 3DES amnd AES encryption in tunnel mode (RFCs: 2402, 2406, 2409). Manual and automatic IKE key support
- PPTP VPN
- L2TP VPN
- Firewall security via:
  - Intrusion Detection System
  - NAT (Network Address Translation)
  - Policy and service-based filtering
  - Stateful inspection firewall
- DHCP server on the LAN side
- DHCP client on the WAN side
- DNS server with forwarding functionality
- SNTP (Simple Network Time Protocol) server/client for computer clock synchronization PPPoE connection to the ISP with PAP/(MS)CHAP authentication
- IP DIFFSERV for QoS
- Virtual LAN (VLAN/IEEE 802.1Q)
- Mail client to send voice and fax messages as e-mail attachments (.wav and .tif) 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
- Remote diagnostics and software upgrade
- Auto-provisioning
- 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 (XML RPC and Windows ActiveX interface)
- Reset button with factory reset option
- Custom Language Pack

Diagnostics/Testing
- LEDs: Busy, Info, Fault, LAN, WAN, Loop settings
- Remote testing

Billing and Statistic
- Radius Client (RFCS: 2865, 2866), CDRs

Environmental

Physical Dimensions
- Rack-mountable devices:
- Measurements: 19” x 7.56” x 1.77” (48.0 x 19.2 x 4.5 cm)
- Weight: 2.47 lbs.(1090 g)

Conditions
- 41°F - 104°F (5°C - 40°C) operating temperature
- 41°F - 140°F (5°C - 60°C) storage temperature
- 5% - 90% non-condensing humidity

Power Supply
- Input 100 - 240 VAC; 50/60 Hz; 0.5 A

Regulatory Compliance
- Telecom: TBR12/TBR13; AS/ACIF

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