

MultiVOIP™

Digital Voice/Fax over IP Gateway



Benefits

- Toll bypass voice/fax communications
- PSTN voice quality
- Connects directly to the PBX or digital carrier service
- Turnkey solution

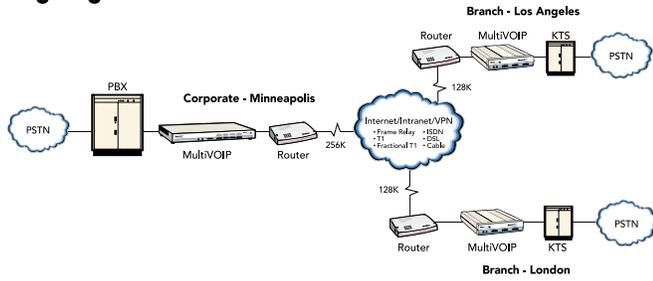
Multi-Tech's digital MultiVOIP provides toll-free voice and fax communications over the Internet or Intranet. By integrating voice and fax into your existing data network, you can realize substantial savings on inter-office long distance toll charges. The digital MultiVOIP family is available in 24/48-port (T1/PRI) and 30/60-port (E1/PRI) models. Both models connect directly to the PBX or digital carrier service to provide real-time, toll-quality voice connections to any office on your VOIP network.

Features

- 24/48-port (T1/PRI) or 30/60-port (E1/PRI) voice/fax ports for communication over an existing IP network or the Internet
- 10/100 Mb Ethernet connectivity and full IP compatibility with existing routers and WAN infrastructure
- Connects directly to a PBX or PSTN line via T1/E1 or PRI
- Voice compression to 5.3K bps per call with support for multiple algorithms, including ITU G.723 and G.729
- Single stage dialing eliminates user training
- Supports voice prioritization using the industry-standard Differentiated Services (DiffServ) protocol
- Supports H.323 and SIP for sending voice over the Internet
- Dial, busy, fast busy and ringback tones
- Configuration and management using bundled SNMP management software for central site logging and monitoring, live reporting, usage tracking, call history, and voice quality statistics
- Supports H.450 supplementary services to provide for call transfer, call forwarding, call hold, call waiting and name identification
- Two-year warranty

*Toll-free Digital
Voice/Fax
Communication over
the Internet or
Intranet*

Highlights

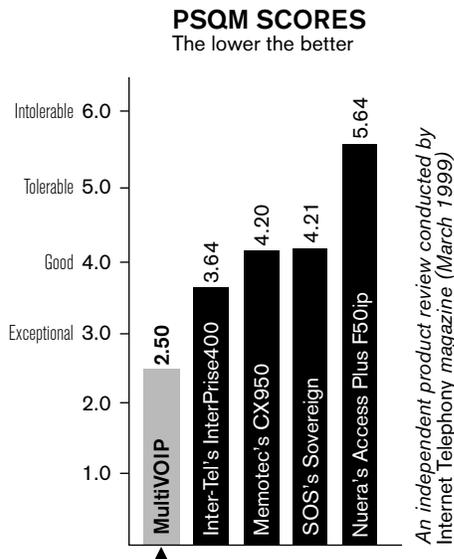


MultiVOIP Applications. MultiVOIP is ideal for businesses looking to reduce toll charges between frequently called sites. MultiVOIP is a voice over IP gateway that integrates seamlessly into your data network and operates alongside an existing PBX to simply extend voice capabilities to remote locations. It is designed to help maximize investments already made in the data and voice network infrastructure.

Save Thousands of Dollars Each Month. MultiVOIP can save substantial amounts in long distance charges. Even if a company uses one of the most inexpensive calling plans, a MultiVOIP network can quickly return the investment and begin paying the company back.

Easy Integration. With MultiVOIP, you avoid the hassle and expense of replacing your existing routers, WAN connections or phone system required by other VOIP solutions. MultiVOIP simply plugs into your Ethernet network. Neither your phone service or network is placed at risk. Minimum requirements: Ethernet network, WAN connection, IP addresses.

Award-winning Voice Quality. With MultiVOIP, you'll experience consistent toll-quality voice connections. Using the Perceptual Speech Quality Measurement (PSQM), Internet Telephony magazine found that MultiVOIP delivered exceptional voice quality. In fact, MultiVOIP outranked the competition.



Interoperability. MultiVOIP utilizes the H.323 and SIP protocols to provide complete interoperability with other Internet telephony solutions. The inbound IP call protocol is automatically detected and the voice channel is dynamically configured to match. The outbound IP call protocol is configured with the phone number, allowing you the flexibility to call H.323 or SIP devices from the same port.

Advanced Speech Technologies. MultiVOIP supports the Differentiated Services (DiffServ) Quality of Service (QoS) protocol which sets priorities for voice and fax traffic and allows transparent delivery. DiffServ helps move time-sensitive voice traffic across even low-bandwidth WAN connections, like 56K DDS and ISDN, with the priority and quality required by voice. Other features such as adaptive echo cancellation, forward error correction, bad frame interpolation, tunable latency and dynamic jitter buffers, further enhance voice quality.

Bandwidth Management. Bandwidth is used only when someone is speaking. The silence suppression/Voice Activity Detection (VAD) feature is an option that frees unused call bandwidth for data traffic. This is significant, since callers are usually silent for 60 percent of the call. When using silence suppression, MultiVOIP also offers Comfort Noise Generation (CNG) at the receiving end so the user knows the line has not dropped. In addition, MultiVOIP supports voice compression standards like G.729 (8:1) and G.723 (10:1). These standards help minimize the bandwidth required for voice. G.723, for instance, is the maximum compression rate and requires only 5.3K bps (plus minimal IP overhead). Even at maximum compression, your VOIP solution will still provide toll-quality voice.

No User Training. MultiVOIP provides single stage dialing by utilizing a Uniform Dialing Plan that is consistent with the E.164 (PSTN) standard numbering plan. This includes automatic appending and stripping of digits to dialed numbers to ensure that users will not require additional training to make VOIP calls. In fact, placing calls with MultiVOIP is like using your existing phone system.

Management. MultiVOIP is easily managed locally using a windows-based software application or remotely by the central office with a web browser or SNMP. Multi-Tech also includes its own SNMP management software called MultiVOIPManager which provides central site configuration, management and call monitoring for all MultiVOIP gateways on the network. It utilizes a Windows interface that makes it easy to view events like usage tracking, live use reporting, call history, and voice quality statistics. In addition, MultiVOIPManager eases administration by automatically e-mailing call logs based on volume or time.

Supplementary Services. MultiVOIP supports H.450 supplementary services to provide for call transfer, call forwarding, call hold, call waiting, and name identification. It also supports Q.SIG, an inter-PBX signaling protocol, for networking PBX supplementary services in a multi- or uni-vendor environment. And, with support for Dual Tone Multifrequency (DTMF), users can access their voice mail over the VOIP network.

Optional H.323 IP Telephony Gatekeeper. The MultiVOIP Gatekeeper is complementary product to the MultiVOIP Voice over IP gateway. The turnkey hardware and software solution provides corporate enterprise network managers and Intranet managers the power to define and control how H.323 voice traffic is managed over IP networks. With the MultiVOIP Gatekeeper, network managers have the ability to configure, monitor and manage the activity of registered network end points. In addition, managers can set policies and control network resources, such as bandwidth usage, to ensure optimal implementation.



You Be the Judge. Industry experts have recognized our VOIP gateways for their clarity. But don't take their word for it, or ours. You be the judge! Make a FREE VOIP call over the Internet by dialing 1-877-TRYVOIP. Hear for yourself how clear the connection can be.



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Specifications

Number of Trunks: 1 or 2 (T1/PRI - 24 or 48 Channels, E1/PRI - 30 or 60 Channels)
Signaling: T1-CAS/Robbed bit signaling; E1 - MFC/R2; PRI - National ISDN 2, 4ESS, 5ESS, DMS100, Austel ISDN, ETSI, France Telecom, HK_Telcom, NTT KDD Japan, Korean Operator
Line Code: T1-AMI or B8Zs; E1-AMI or HDB3
Frame Format: T1-ESF or D4 (SF); E1-I6 Frame plus CRC
Connectors: 1 or 2 RJ48

LAN Port

Interface: 10/100BaseT
Format: Ethernet/Ethernet II or SNAP

Command Port

Interface: RS-232C/D; RJ-45 (cable included)
Speed & Format: 115.2K bps asynchronous

Protocols

H.323 V4, SIP, H.450.2-H.450.4, H.450.6 & H.450.8*, RTP, RTCP, SMTP, Q.931, T.38* & Group 3 fax relays, Q.Sig*

Bandwidth Management

G.711, G.723, G.726, G.727, G.729 & proprietary voice compression, silence suppression, VAD, CNG

Voice Quality

DiffServ, G.165, G.168, adaptive echo cancellation, forward error correction, bad frame interpolation, tunable latency, dynamic jitter buffers

Management

Web browser*, Windows, SNMP agent, MultiVOIPManager, flash upgradeable

Power

Voltage & Frequency: 115v/240v AC, 47/60 Hz
Power Consumption: 27W

Dimensions

17.4" w x 1.75" h x 8.75" d: 7.5 lbs.
(44.2 cm x 4.5 cm x 22.2 cm; 3.4 kg.)

Certification

EMC: FCC Part 15 Class A, EN55022, EN55024, EN61000-3-2, EN61000-3-3
Safety: CE, UL 60950, EN60950, cUL, TS001
Telecom: FCC Part 68, CS-03, TBR21

* These features are expected to be released during Q4 as a free upgrade.

Ordering Information

Product	Description	Region
MVP2410	24/48-Port T1/PRI VOIP Gateway	US/Can
MVP24-48	24-Port T1/PRI Expansion Card	US/Can
MVP3010	30/60-Port E1/PRI VOIP Gateway	ROW
MVP30-60	30-Port E1/PRI Expansion Card	Euro/ROW

* Specify country when ordering.

Made in Mounds View, MN, U.S.A.

