MultiVOIP™

Voice/Fax over IP Gateways

Benefits

Toll bypass voice/fax communications

PSTN voice quality

Connects directly to phones, fax or PBX

Turnkey solution

The Multi-Tech 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 MultiVOIP family is available in analog and digital models ranging from one to 60 ports. All MultiVOIP products connect directly to phones, fax machines, key systems, PSTN lines, or a PBX to provide real-time, toll-quality voice connections to any office on your VOIP network. For Avaya™ Communication Manager environments, MultiVOIP provides distributed networking capabilities to small branch offices of Fortune 1000 corporations. MultiVOIP extends the call features of a centralized Avaya media server and provides local office survivability to small branch offices of up to 15 users using analog or IP phones. For more information and specific Avaya model numbers, visit the Avaya extranet at www.multitech.com/partners/avaya.

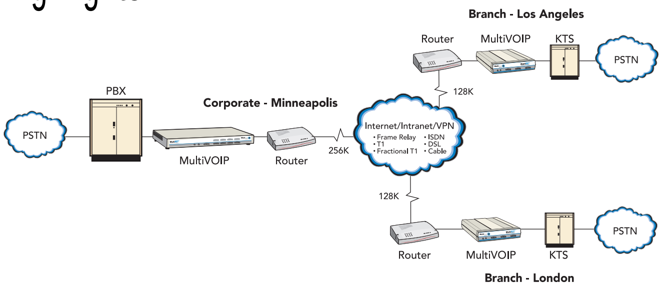
Toll-free Voice/Fax Communication over the Internet or Intranet

Features

- 1-, 2-, 4-, 8- analog ports or 24/30 (expandable to 48/60) digital ports for communication over an existing IP network or the Internet
- Ethernet connectivity and full IP compatibility with existing routers and WAN infrastructure
- FXS/FXO and E&M connectors on each channel for direct analog connection to phones, key telephones, PBX extensions, PSTN lines or PBX trunks (MVP130 supports FXS and FXO)
- Digital MultiVOIP connects directly to PBX or PSTN line via T1/E1 or PRI
- Supports H.323 or SIP for sending voice over the Internet
- Single Port Protocol (SPP) allows the use of dynamic IP addresses
- PSTN fail-over automatically routes calls over the PSTN network if the IP network is down
- Supports H.450 supplementary services to provide for call transfer, call forwarding, call hold, call waiting and name identification
- Voice compression to 5.3K bps per call with support for multiple algorithms, including ITU G.723 and G.729
- T.38 real-time fax relay for interoperability among other VOIP equipment
- Configuration and management using a Web browser or Windows
- Two-year warranty

MultiTech® Systems

Highlights



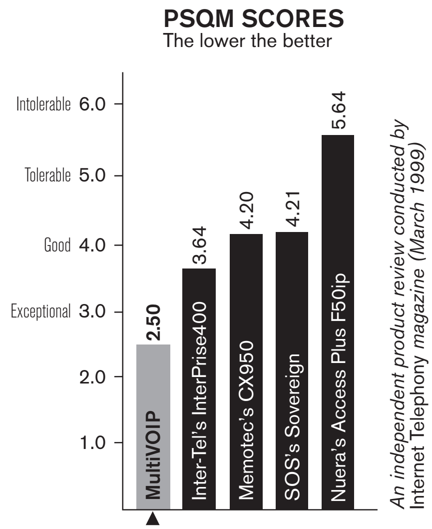
MultiVOIP Applications. MultiVOIP is specifically targeted at businesses looking to reduce toll charges between frequently called sites. MultiVOIP is a voice over IP gateway that integrates seamlessly into your data network and operates alongside existing PBXs, or other phone equipment to simply extend voice capabilities to remote locations. It is designed to help you maximize investments you've already made in your data and voice network infrastructure.

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.

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

Locations	MultiVOIP Cost	Long Distance Cost/Minute	Minutes/ Line/Day	MultiVOIP Payback
Corporate Site/ Minneapolis	$1,799 MVP410 (4 lines)	$0.04	90	125 days
Branch Site/ Los Angeles	$999 MVP210 (2 lines)	$0.06	60	139 days
Branch Site/ London	$999 MVP210 (2 lines)	$0.08	60	104 days

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. In addition, MultiVOIP also supports T.38 real-time fax relay for interoperability among other VOIP equipment.

PSTN Fail-over. PSTN fail-over allows MultiVOIP to automatically route calls over the PSTN network when the IP network is congested or completely down. This feature heighten reliability and augments QoS when conditions threaten to undermine voice quality. Utilizing user definable controls, MultiVOIP continually checks if the LAN/WAN is threatened by packet loss or latency, or to see if the network is completely down. If it detects a problem, MultiVOIP switches to "survivability mode" transparently routing all calls over PSTN lines connected to the MultiVOIP gateway. MultiVOIP continues to monitor the connection and automatically switches back to the LAN/WAN once the conditions improve.

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 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.

Complete Support for Multiple Telephony Interfaces. For maximum investment protection, the MultiVOIP two, four and eight-port models accommodate changing communication needs by providing a programmable FXS/FXO and an E&M interface for each port. This allows MultiVOIP to connect directly to a phone, fax machine, key phone system or PBX. It automatically detects whether the incoming call is a voice or fax call. The single port MultiVOIP supports FXS and FXO interfaces, while the digital MultiVOIP connects directly to a PBX or PSTN line via T1/E1 or PRI.

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 an added 7-8K bps for IP overhead). Even at maximum compression, your VOIP solution will still provide toll-quality voice.

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.

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.

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. In addition, MultiVOIP supports SIP extensions providing call forward and call transfer capabilities.

Avaya Small Office Media Gateway Solution. Avaya and Multi-Tech have partnered together to provide an affordable small office media gateway solution that delivers the features of Avaya's Communication Manager software to the branch offices of large corporations. The Multi-Tech MultiVOIP gateway, with integrated gatekeeper, cost-effectively extends the call features and networking benefits of a centralized Avaya Media Server to small branch offices, utilizing traditional analog devices, over an IP infrastructure. MultiVOIP also renders local office survivability, in the case of a LAN or WAN failure, by providing local, reliable PSTN trunking. For more information an specific models, go to www.multitech.com/partners/avaya.

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.

Specifications

Analog Models

Number of Ports: 1, 2, 4 or 8
Port Interface: FXO, FXS & E&M support on each port (MVP130 supports FXS and FXO)
FXS Interface: KTS, telephone set, or fax; ground and loop start
FXO Interface: PBX station; CO line, loop start, 2-wire
E&M Interface: PBX E&M trunk; 2- or 4-wire
E&M Signal Types: I through V
Dialing: DTMF or pulse
Connectors: 1 RJ-48 (E&M); 1 RJ-11 (programmable FXS or FXO) per port

Digital Models

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 Telcom, HK Telcom, NTT and KDD Japan, Korean Operator
Line Code: T1-AMI or B8Zs; E1-AMI or HDB3
Frame Format: T1-ESF or D4 (SF); E1-16 Frame plus CRC
Connectors: 1 or 2 RJ-48

LAN Port

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

Command Port

1-, 2-Port & Digital Interface: RS-232C/D; RJ-45 (RJ-45 to DB9 cable included)
4- & 8-port Interface: RS-232C/D; DB25
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, Q.Sig, T.38 & Group 3 fax relay, DTMF out-of-band (RFC Z833))

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: 1-Port - 4.5W, 2-Port - 19W; 4- & 8-Port -46W; Digital models - 27W

Dimensions

1-port model:
4.3" w x 1.0" h x 5.6" d; 8 oz.
(15.8 cm x 3.6 cm x 22.9 cm; 0.92 kg)
2-port model:
6.2" w x 1.4" h x 9.0" d; 2 lbs.
(15.8 cm x 3.6 cm x 22.9 cm; 0.92 kg)
4- & 8-port model:
17.4" w x 3.8" h x 8.0" d; 7.4 lbs.
(44.2 cm x 9.5 cm x 20.3 cm; 3.4 kg)
Digital Model:
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, ACA TS-001
Telecom: FCC Part 68, CS-03, TBR21

Ordering Information

Product	Description	Region
Analog		
MVP130*	1-Port FXS/FXO VOIP Gateway	Global
MVP210*	2-Port VOIP Gateway	Global
MVP410*	4-Port VOIP Gateway	Global
MVP810*	8-Port VOIP Gateway	Global
Digital		
MVP2410	24/48-Port T1/PRI VOIP Gateway	US/Can
MVP24-48	24-Port T1/PRI Expansion Card (expands a MVP2410 to 48 ports)	US/Can
MVP3010*	30/60-Port E1/PRI VOIP Gateway	Euro/ROW
MVP30-60	30-Port E1/PRI Expansion Card	Euro/ROW

* Specify country when ordering.

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

World Headquarters
Tel: (763) 785-3500
(800) 328-9717
www.multitech.com

EMEA Headquarters
Multi-Tech Systems (EMEA)
United Kingdom
Tel: +(44) 118-959 7774

Multi-Tech Systems (EMEA)
France
Tel: +(33) 1 64 61 09 81

MultiTech® Systems