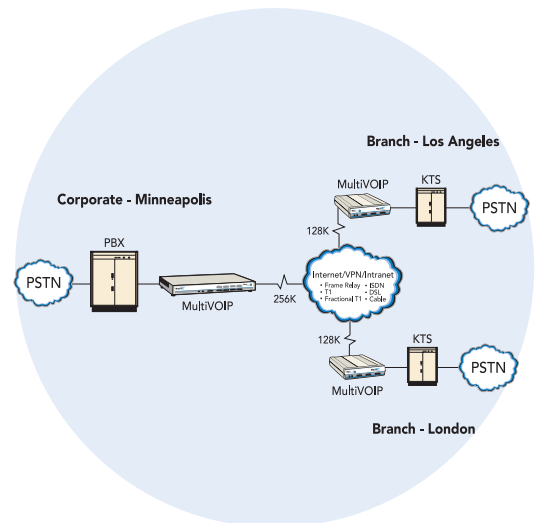# MultiVOIP®

Voice/Fax over IP Gateways

Benefits

- Toll bypass voice/fax communications
- PSTN voice quality
- Connects directly to phones, fax or PBX
- Turnkey solution

The MultiVOIP® gateway 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 gateway is available in analog and digital models ranging from one to 60 ports. All MultiVOIP gateways 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.

Features

- 1, 2, 4 or 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/DID and E&M connectors on each channel for direct analog connection to phones, key telephones, PBX extensions, PSTN lines or PBX trunks*

- Digital MultiVOIP gateways connect directly to PBX or PSTN line via BRI, T1/E1 or PRI

- Supports H.323, SIP or SPP 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

- Emergency transfer (power-out fail-over)

- Voice compression to 5.3K bps per call with support for multiple algorithms, including ITU G.723 and G.729

- QoS via DiffServ or 802.1p

- T.38 real-time fax relay for interoperability among other VOIP equipment

- VAD and CNG support

- Configuration and management using a Web browser or Windows

- Built-in modem for installation and maintenance on select models

- Two-year warranty

* MVP130 supports FXS, FXO and DID. MVP130-FXS supports FXS only.

MultiTech®
Systems

Highlights

MultiVOIP Gateway Applications. The MultiVOIP gateway is specifically targeted at businesses looking to reduce toll charges between frequently called sites. It integrates seamlessly into your data network and operates alongside existing PBXs, or other phone equipment to simply extend voice capabilities to remote locations. The MultiVOIP gateway is designed to help you maximize investments you've already made in your data and voice network infrastructure.
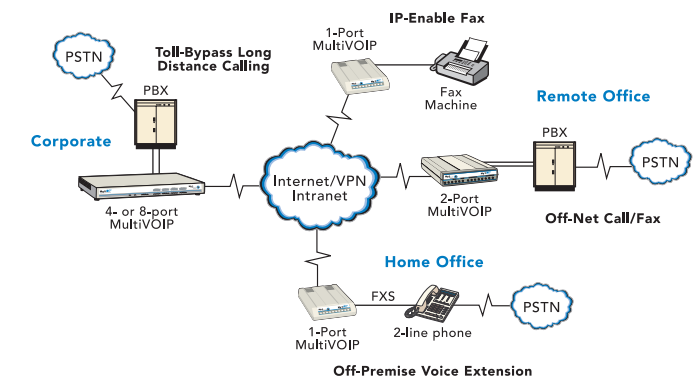
Toll Bypass Long Distance Calling. For multi-location businesses, inter-office calling typically represents 25-40% of a company's total long distance bill. To bypass these charges, each office installs and configures a MultiVOIP gateway on their network and connects it to their existing phone equipment to place calls, send faxes, or make modem connections to the other offices on the VOIP network.

Off-net Calling. Telecommuters or customers off the IP network can make free long distance calls by dialing into a local MultiVOIP gateway and placing toll-free calls to any location on the VOIP network. You can even have a MultiVOIP gateway at a remote site dial a local phone number for a free person-to-person long distance call.

Off-premise Voice Extensions. The MultiVOIP gateway extends the reach of a company's PBX into remote office/SOHO locations without installing another PBX. Now, anyone can place calls to the remote office by simply dialing an extension number. With support for H.450 supplementary services, the MultiVOIP gateway provides for call transfer, call forwarding, call hold, call waiting and name identification. And, with the low-cost, MultiVOIP MVP130-FXS single-port analog adapter, you have the ability to inexpensively connect a single analog phone in a SOHO location.

IP-enable Fax Machines. The MultiVOIP MVP130-FXS single-port analog adapter also inexpensively turns a fax machine into an IP network capable device. With T.38 fax compatibility it promises to keep your "legacy" fax equipment productive for as long as you need it. Ideal for IP-enabling fax machines in both large enterprise networks or small branch office/SOHO locations.

Wireless Voice Extensions. To extend a PBX to a building across the street, or to a location with difficult access, utilize a wireless or satellite bridge to connect the two networks. Now, you have voice and data connectivity without having to lay cables or pay monthly charges for dedicated lines.

Replace Expensive Tie Lines. A corporation that utilizes Tie lines to connect branch office PBXs to the corporate PBX can now use the company's IP-based Wide Area Network to complete the call.

Easy Integration. With the MultiVOIP gateway, you avoid the hassle and expense of replacing your existing routers, WAN connections or phone system required by other VOIP solutions. The MultiVOIP gateway 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. The MultiVOIP gateway can save your company substantial amounts in long distance charges. Even if your company uses one of the most inexpensive calling plans, a voice over IP 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	$1,499 MVP410 (4 lines)	$0.04	90	104 days
Branch Site/ Los Angeles	$899 MVP210 (2 lines)	$0.06	60	125 days
Branch Site/ London	$899 MVP210 (2 lines)	$0.08	60	94 days

Interoperability. The MultiVOIP gateway 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, Multi-Tech has developed it's own proprietary Single Port Protocol (SPP) to interoperate with other MultiVOIP gateways The advantage of using SPP is that it requires only one static IP address allowing all other IP addresses to be dynamic. It is also easier to install behind a firewall as it only requires one open port. The MultiVOIP gateway also supports T.38 real-time fax relay for interoperability among other VOIP equipment

PSTN Fail-over. PSTN fail-over allows the MultiVOIP gateway to automatically route calls over the PSTN network when the IP network is congested or completely down. This feature heightens reliability and augments QoS when conditions threaten to undermine voice quality. Utilizing user definable controls, the MultiVOIP gateway 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, the MultiVOIP gateway switches to "survivability mode" transparently routing all calls over PSTN lines connected to the MultiVOIP gateway. The MultiVOIP gateway continues to monitor the connection and automatically switches back to the LAN/WAN once the conditions improve.

Advanced Speech Technologies. The MultiVOIP gateway supports both DiffServ and 802.1p Quality of Service (QoS) protocols which set priorities for voice and fax traffic and allow for transparent delivery. The QoS protocols 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 and dynamic jitter buffers, further enhance voice quality.

Complete Support for Multiple Telephony Interfaces. For maximum investment protection, the MultiVOIP gateway two-, four- and eight-port models accommodate changing communication needs by providing a programmable FXS/FXO/DID and an E&M interface for each port. This allows the MultiVOIP gateway 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 gateway supports FXS, FXO and DID interfaces, while the digital and ISDN MultiVOIP gateways connect directly to a PBX or PSTN line via T1/E1, PRI or ISDN BRI.

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, the MultiVOIP gateway also offers Comfort Noise Generation (CNG) at the receiving end so the user knows the line has not dropped. In addition, the MultiVOIP gateway 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. The MultiVOIP gateway 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. The MultiVOIP gateway four- and eight-port models also include a modem for remote diagnostics, monitoring, control and configuration.
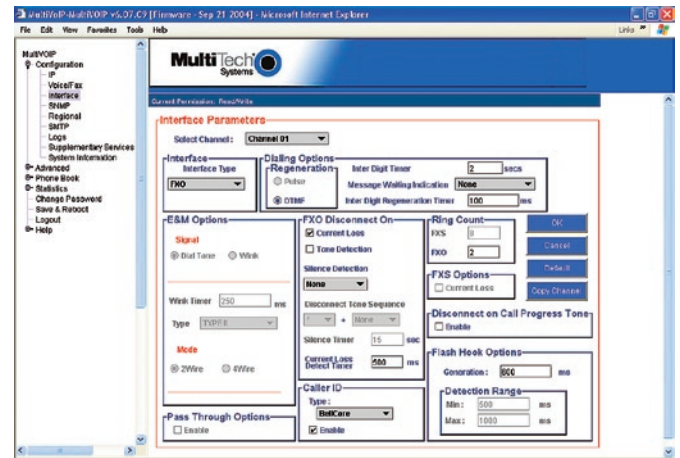
No User Training. The MultiVOIP gateway provides single stage dialing by utilizing a Uniform Dialing Plan that is consistent with the E.164 (PSTN) standard numbering plan. This includes automatic appending and stripping of digits to dialed numbers to ensure that users will not require additional training to make VOIP calls. In fact, placing calls with the MultiVOIP gateway is like using your existing phone system.

Supplementary Services. The MultiVOIP gateway 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, the MultiVOIP gateway 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 survivable media gateway solution that delivers the features of Avaya's Communication Manager software to the branch offices of large corporations. The MultiVOIP AV gateway, with integrated gatekeeper, cost-effectively extends the call features and networking benefits of an Avaya Media Server to small branch offices. The MultiVOIP AV gateway also renders local office survivability, in the case of a LAN or WAN failure, by providing local, reliable PSTN trunking. A cost-effective single-port gateway is available to turn your analog phones or fax machines into Avaya IP devices that can serve the needs of enterprise networks, small office environments and single user locations. For more information and specific models, go to www.multitech.com/partners/avaya

Comprehensive Service and Support. The Multi-Tech commitment to service means we provide a two-year product warranty and service that includes free telephone technical support, 24-hour web site and ftp support.

Specifications

Analog Models

Number of Ports: 1, 2, 4 or 8

Port Interface: FXS, FXO, DID, & E&M support on each port (MVP130 supports FXS, FXO, & DID; MVP130-FXS supports FXS only)

FXS Interface: KTS, telephone set, or fax; ground & loop start

FXO Interface: PBX station; CO line, loop start, 2-wire

DID Service Types: Wink-start; immediate-start; delay dial

DID Signaling Type: DTMF

DID Operational Mode: Dial Pulse Originating (DPO)

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, & 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

ISDN Models

Number of BRI S/T (2B+D) Interfaces: 2- or 4-ports

Signaling: BRI-National ISDN 2, 5ESS, National ISDN 1, DMS100, Austel ISDN, ETSI, France Telecom, HK Telecom, NTT & KDD Japan, Korean Operator

Line Code: Pseudoternary

Connectors: 2 or 4 RJ-45

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, SPP, 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 2833)

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, 802.1p, 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/240VAC, 47/60 Hz

Power Consumption:
  1-Port Model: 4.5W
  2-Port Model: 19W
  4- & 8-Port Models: 46W
  Digital Models: 27W

Dimensions

1-Port Model: 4.3" w x 1.0" h x 5.6" d; 8 oz. (10.9 cm x 2.5 cm x 14.2 cm; 0.23 kg)

2-Port Model: 6.2" w x 1.4" h x 9.0" d; 2 lbs. (15.7 cm x 3.6 cm x 22.9 cm; 0.91 kg)

4-Port & 8-Port Models: 17.4" w x 3.8" h x 8.0" d; 7.4 lbs. (44.2 cm x 9.6 cm x 20.3 cm; 3.4 kg)

Digital Models: 17.4" w x 1.8" h x 8.8" d; 7.5 lbs. (44.2 cm x 4.5 cm x 22.3 cm; 3.4 kg)

ISDN Models: 17.25"w x 1.75"h x 8.5"d: 6.75 lbs. (43.8 cm x 4.5 cm x 21.6 cm; 3.1 kg.)

Certification

EMC: FCC Part 15 Class A, EN 55022, EN 55024, EN 61000-3-2, EN 61000-3-3

Safety: CE, UL 60950, EN 60950, cUL, ACA TS-001

Telecom: FCC Part 68, CS-03, TBR21, ACA TS-031, TBR3

Ordering Information

Product	Description	Region
Analog		
MVP130-FXS*	1-Port Analog Adapter	Global
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	US/Can
MVP3010*	30/60-Port E1/PRI VOIP Gateway	Euro/ROW
MVP30-60	30-Port E1/PRI Expansion Card	Euro/ROW
ISDN		
MVP410ST-EU*	4-Channel BRI VOIP Gateway	Euro/ROW
MVP810ST-EU*	8-Channel BRI VOIP Gateway	Euro/ROW

* Specify country when ordering.

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

Features and specifications are subject to change without notice.

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Systems