MultiVOIP® SS

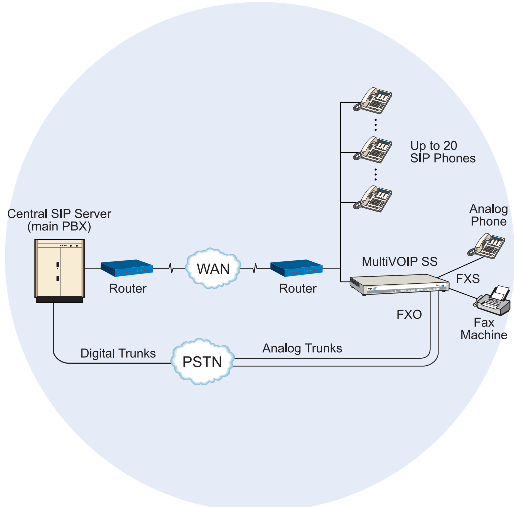
Survivable SIP Gateway and Server

Benefits

- Local office survivability
- Analog PSTN Trunking
- Multi-port ATA functionality

The MultiVOIP® SS survivable SIP gateway and server is ideal for small branch offices of large corporations that have deployed a distributed IP telephony network. It functions as the primary or secondary SIP server for IP phones used at the remote offices and renders local office survivability, in the case of a LAN or WAN failure, by providing local, reliable PSTN trunking. The MultiVOIP SS gateway also provides local PSTN access for emergency calls, such as 911, as well as normal inbound/outbound local calling. In addition, the gateway functions as an analog telephone adapter (ATA), adapting analog phones and fax machines to IP environments.

Features



- 2, 4 or 8 analog ports for communication over an existing IP network or the Internet
- Supports SIP for sending voice over the Internet
- 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
- Provides local office survivability in the event of a LAN/WAN failure
- Emergency transfer (power-out fail-over)
- PSTN trunking bridges the PSTN to the IP network for emergency calls as well as normal inbound/outbound calling*
- Multi-port ATA functionality for analog phones and fax machines
- Ethernet connectivity and full IP compatibility with existing routers and WAN infrastructure
- Voice compression to 5.3K bps per call with support for multiple algorithms, including ITU G.723 and G.729
- VAD and CNG support
- QoS via DiffServ or 802.1p
- T.38 real-time fax relay for interoperability among other VOIP equipment
- Supports SIP supplementary services including call forward, call transfer, and call hold
- Adaptive echo cancellation, forward error correction and dynamic jitter buffers
- Configuration and management using a Web browser or Windows
- Built-in modem for remote out-of-band management (4 and 8 port models)
- Two-year warranty

* PSTN trunking available if supported by the central SIP server

Highlights

Multi-port ATA Functionality. The MultiVOIP SS gateway is ideal for small branch offices (1-20 phones/employees) of large corporations that have migrated to a central SIP-based phone system. The gateway adapts analog phone equipment and fax machines to the IP network, affordably integrating the office into a pure voice-over-IP environment.

Local Office Survivability. The MultiVOIP SS gateway functions as the primary or secondary SIP server for IP phones used at the remote office. While in normal mode, the MultiVOIP SS gateway will transparently pass all IP phone registrations to the central PBX/server. This allows the remote IP phones to function with a full feature set. In the event of a LAN or WAN failure, the MultiVOIP SS gateway automatically redirects calls through local PSTN trunks connected to the gateway. Once in survivable mode, the SIP server built-into the MultiVOIP SS gateway takes full command providing station-to-station, station-to-trunk and trunk-to-station call support. This allows the office to maintain critical communications, with a limited feature set, until full functionality is restored.

In the event of a LAN failure, the MultiVOIP SS SIP gateway and server goes into LAN fail-over survivability mode. Since the LAN is down, IP phones will not function, however, inbound/outbound calls can still be made via analog phones or PSTN lines directly connected to the MultiVOIP SS gateway. The user would simply need to dial an extension to reach the PSTN trunk before making an outbound call.

In the event of a power failure, the MultiVOIP SS SIP gateway and server provides emergency transfer fail-over survivability. This is achieved by connecting an analog telephone or fax machine to the first port on the gateway, and a PSTN line to the second port. When in this mode, the gateway automatically draws power from the PSTN line and supplies it to the analog phone/fax handset to provide inbound/outbound calling.

PSTN Trunking. If the central PBX/server supports IP trunking, the MultiVOIP SS gateway can also provide local PSTN access for emergency calls such as 911, as well as normal inbound/outbound calling. This allows a branch office, such as a financial institution, to maintain a local phone number for its customers.

Ordering Information

Product	Description	Region
MVP210-SS	2-Port VOIP Gateway/SIP Server	Global
MVP410-SS	4-Port VOIP Gateway/SIP Server	Global
MVP810-SS	8-Port VOIP Gateway/SIP Server	Global

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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Specifications

Analog Ports

- Number of Ports: 2, 4 or 8
- Port Interface: FXS, FXO, DID, & E&M on each port
- 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, FXO or DID) per port

LAN Port

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

Command Port

- 2-Port Interface: RS-232C/D; RJ-45
- 4- & 8-port Interface: RS-232C/D; DB25
- Speed & Format: 115.2K bps asynchronous

Protocols

SIP, RTP, RTCP, SMTP, Q.931, 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, 802.1p, G.165, G.168, adaptive echo cancellation, forward error correction, bad frame interpolation, tunable latency, dynamic jitter buffers

Management

Web browser, Windows, flash upgradeable

Power

- Voltage & Frequency: 115V/240VAC, 47/60 Hz
- Power Consumption:
  - 2-Port Model: 19W
  - 4- & 8-Port Models: 46W

Dimensions

- 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)

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

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Freecall 1800 817 807
(within Melbourne call 03 8562 9000)

Melbourne Office: 4/585 Blackburn Road, Notting Hill, VIC 3168
Canberra Office: 2/42 Geils Court, deakin, ACT 2600

voip@alloy.com.au
www.voip.alloy.com.au