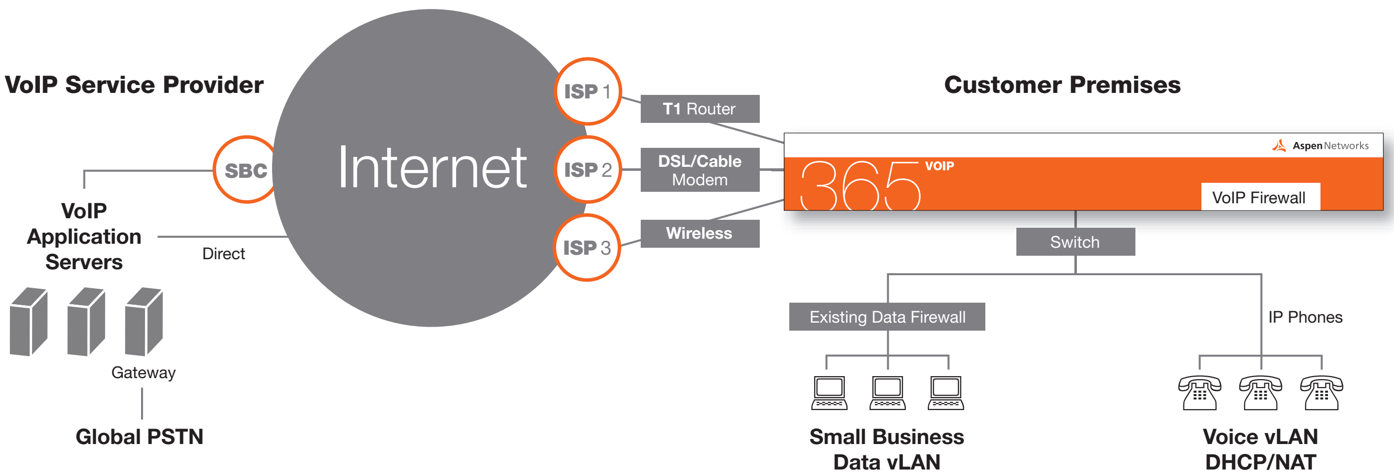
Aspen 365 VOIP

Aspen Networks

Aspen Networks helps hosted VoIP service providers and small-to-medium enterprises ensure non-stop IP performance for VoIP and other mission critical applications. The Aspen 365-VOIP™ is a SIP application layer gateway (ALG) that resolves NAT/Firewall traversal issues while allowing voice traffic to run on 2 or more WAN links for maximum reliability and quality of service. Aspen's advanced QoS features include: dynamic VoIP prioritization, sub-second fault detection/auto-failover, and SIP survivability. By protecting VoIP from WAN outages and link degradation, Aspen helps improve customer satisfaction and retention while dramatically lowering support costs.



Features	Benefits
Multi-link WAN Switch	
Dual CPU's (control processor separate from voice stream processor) with proprietary OS for fast switching and improved QoS	Dedicated stream processor minimizes jitter and loss, thus improving call quality for large numbers of users
Active, real-time call quality monitoring and analysis of individual voice streams	In a multi-link environment, the best path for a voice flow can be selected within milliseconds of a call-degrade event
Policy-based bandwidth allocation for VoIP and various types of data traffic, interoperable with all existing data firewalls	Complete control over voice and data performance as well as cost tradeoffs across multiple and diverse types of ISP links
Multi-link QoS: Built-in redundancy with auto-failover on up to 4 WAN links with separate policy controls for VoIP and data flows	Ensures 99.999% availability for voice and data. Optimizes both voice and data quality when multiple links are available.
Dynamic single link QoS and traffic shaping. Rate limiting on data only activates when voice is present; data–throttling is a function of voice load	Maximizes voice quality when only one link is available. Most of the time data gets full rate bandwidth (unlike static QoS schemes)
Transparent Ethernet switch "personality" with no public IP address termination of Internet-facing side	Minimizes vulnerability to Internet DDoS attacks; easy to install, no reconfiguration of existing LAN devices required
SIP Application Layer Gateway (ALG)	
SIP survivability – automatic recovery from failed hosted IP PBX servers in milliseconds, without SIP phone having to reboot or re-register	End users experience uninterrupted dial tone
Local call routing and remote SIP survivability–the Aspen SIP ALG handles local placement, which persists if hosted IP PBX becomes unavailable	Conserves WAN bandwidth, enabling more users per WAN link, and scalability in larger deployments
Parallel clustering of any number of Aspen 365-VOIP units while sharing the same set of 2-4 WAN links	Able to support arbitrarily large numbers of hosted VoIP users at a single geographic location
VoIP NAT/PAT/Firewall/ALG with state-based "pinhole" of RTP flows integrated with DHCP server	IP phones remain secure against VoIP specific attacks and intrusions
Interoperable with most VoIP application platforms (e.g. BroadSoft), softswitches and session border controllers (e.g. Acme Packets)	Adapts to a variety of customer configurations and VoIP service provider deployment models
Separate IP subnet VLANs supported for voice and data through Aspen's VLAN aware DHCP pooling	Supports the most commonly deployed LAN infrastructures for VoIP and data convergence
SIP proxy with registration pacing, softswitch redundancy	Conserves WAN bandwidth and enables a single hosted IP PBX to scale to large numbers of remotes

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Number of ISP Links Managed	4 independent ISP links (without requiring use of BGP)
Number of CPUs	2
Secure Management Port	1 RJ-45
LAN Ports	1 RJ-45
WAN/ISP Ports	2 RJ-45 physical ports Up to 4 active ISP links can be configured in software
DB-9 Serial Port	1

VoIP Traffic Management

Auto-correction in the presence of ISP or VoIP soft switch faults	Yes, auto-recovery is rapid without IP phone reboots
VoIP protocols supported in multiple ISP link handling and multi-link QoS	SIP/RTP, MGCP, H.323, Asterisk IAX
SIP ALG Certification	BroadSoft certified
SIP Phone Interoperability	Includes Polycom, Cisco and others
SIP ALG Features	Local call routing, remote SIP survivability, transparent SIP ALG, SIP registration rate pacing, integrated auto-recovery
Interoperable with other SIP ALG T1 routers on customer premises	Yes
QoS and Traffic Shaping	Yes

Centralized Operations and Management

Installation, Configuration, and Updates	Web-browser based GUI, command line interface, and remote updates
NOC Monitoring	Remote SSH, detailed syslog and event alerts for voice call related activity to industry-standard, centralized, network management platforms (e.g. Orion, HP OpenView)
SIP Call Trace/Analyzer	Built in, real time display
SIP Trace Collection at NOC	Yes, via embedded syslog client, multiple concurrent servers
Simultaneous, remote SSH over any available ISP WAN link	Over 4 Links simultaneously if desired. Remote management can be performed on ISP link not used by VoIP payload

VoIP Security/Admission Control

Stateful VoIP Firewall	Yes
PAT/NAT included	Yes
DHCP Server	Yes
Admission Control by MAC Address and IP Phone Manufacturer	Yes
Co-exist with existing LAN firewalls	Yes

Hardware

CTL CPU Memory	128 MB
FWD CPU Memory	64 MB
Hardware Watchdog Timer	Yes
Software Authentication Chip	Yes
Rack Mount Kit	Optional
Enclosure/Dimensions	1.75” x 12” x 10”
Weight	4.84 lbs

Environmental

Temperature	0°C to 50°C (32°F to 122°F)
Humidity	Less than 100% relative humidity, non-condensing. Up to 8000 ft. (2438 M)
Power	AC power 85-265V, 47-63Hz; 15 Watts; Internal power supply
Fans	1

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