IP to IP Mediation for SIP Trunking

Feature Focus

Applicable Product: Mediant™ 1000
Software version: 5.4

Customer Challenges

As Service Providers and Enterprises move towards all-IP services, they face additional challenges interoperating between different VoIP systems. As a result new requirements have recently been introduced to the market, including Operator-to-Business SIP Trunking, Business-to-Business connectivity and Business Inter-Branch connectivity. All these require IP to IP mediation, SIP translation and enhanced VoIP security, performed by enterprise-class Session Border Controllers (SBC).

An SBC is a B2BUA device (Back to Back User Agent), that acts as a server for the internal network, and as a client for external networks handling all aspects of a VoIP call, including: session set-up, session disconnection and the creation of a complete separation between networks. This capability also allows for the addition of features such as billing, network topology hiding, call admission control, authorization, and many more.

IP to IP mediation introduced in version 5.4 for the Mediant™ 1000 is the first of numerous SBC capabilities, which will be fully featured in the Mediant™ 1000-MSBG (Multi Service Business Gateway), planned for Ver. 5.6.

New features that are introduced as part of version 5.4 are:

- SIP to SIP normalization
- Network topology hiding
- Media transcoding and conversion
- Signaling Translation
- Multiple Service Provider connectivity
- Load balancing and redundancy between Servers/Softswitches
- Survivability

SIP to SIP Normalization

Allows two SIP systems to interconnect by means of normalizing different SIP implementations. The “Normalizer” acts as a server on one side, terminating incoming sessions of one SIP implementation, and as a client on the other side initiating new sessions of a different SIP implementation.

This capability can be used as follows:

1. Between an Enterprise IP-PBX and a service provider’s SIP trunk
2. Between an Enterprise headquarters’ IP-PBX and a branch office IP-PBX
3. Between an IP-PBX and an application server (e.g. Microsoft Office Communications Server)

Network Topology Hiding

By acting as a B2BUA device, external users have no visibility of the internal network topology (i.e. endpoints, signaling and traffic), thus providing an important degree of security. In addition, a further security level is achieved by encryption of SIP header fields exiting the network, which includes the following information:

1. **Via** - Used to route responses relating to a specific request
2. **Route** - Indicating the path subsequent signalling messages should take
3. **Record Route** - A list of SIP agents that have proxied the specific message
4. **Service Route** - A list of SIP proxies that may add service to the signalling message

Media Transcoding and Conversion

The gateway can translate voice traffic between two separate networks which use different coders or media payloads. An example is the transcoding between WAN and LAN on the edge of the Enterprise network (in the case of a SIP trunk being provided) or when peering between two Enterprise networks. Additional examples include:

1. Transcoding between G.711 and low bit rate coder (eg. G.729) or between two bit rate coders
2. Converting between T.38 and In-band FAX
3. Converting between DTMF In-band and RFC-2833 and to SIP INFO/NOTIFY

Signaling Translation

Converting between SIP <-> SIP-TLS over TCP or UDP (any to any).

Multiple Service Provider Connectivity

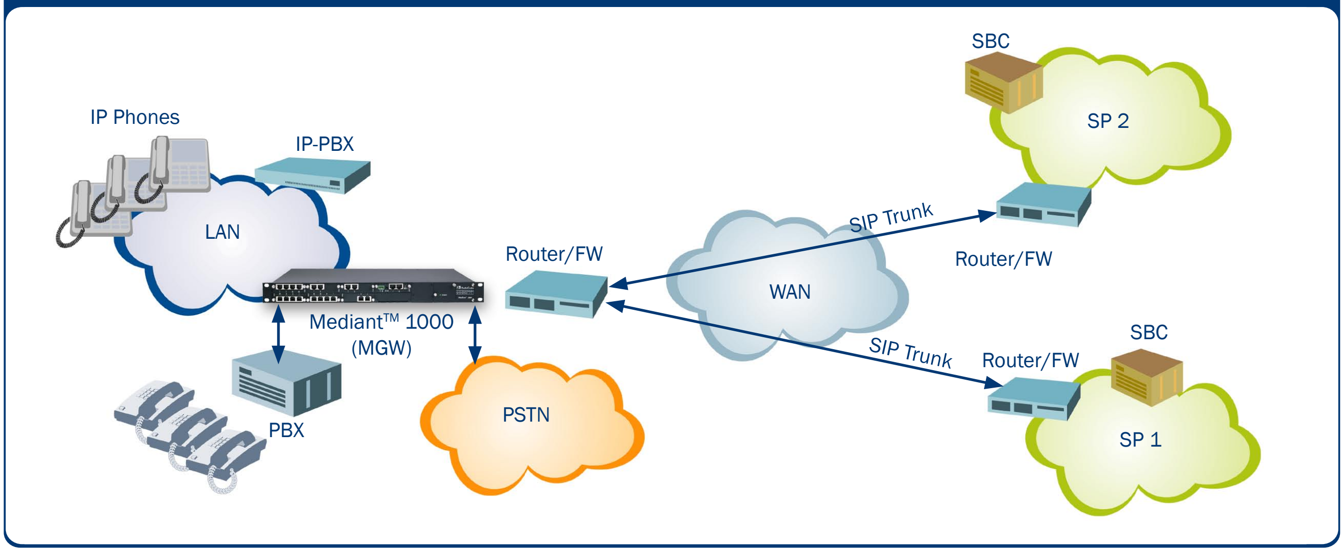
An Enterprise can use multiple SIP trunks (up to 2 in Version 5.4 and 8 in Version 5.6) provided by several service providers, allowing dynamic provider selection for least cost routing and service redundancy.

AudioCodes

IP to IP Mediation for SIP Trunking

Feature Focus

SIP Trunks to Multi ITSPs



Load-balancing and Redundancy between Servers/Softswitches

The Media Gateway allows concurrent connectivity to two servers (belonging to the same service provider) as well as enabling fallback and load-balancing between them.

Survivability

As a B2BUA device, the gateway provides Stand Alone Survivability and maintains internal connectivity when the central Softswitch is unavailable. This is a different method of survivability compared to the previous one as presented by the SAS feature, in which the gateway acts as a proxy (usually used in IP-Centrex environments).

What are the IP to IP session capacities of the Mediant 1000?

In version 5.4 Mediant 1000

- Each IP to IP session requires 2 DSP voice channels. The Mediant 1000 is limited to 120 DSP channels or 60 IP to IP Sessions.

In version 5.6 Mediant 1000 MSBG

- The maximum number of IP to IP sessions will be 300 sessions at General Availability release
- In the case of the same codec being used on both sides, no DSP resources are required
- In the case of different codecs being used, two DSP channels will be required per session, up to a maximum of 60 sessions
- The Mediant 1000 SBC function will support codec policing by determining the selected codecs according to call properties and scenarios

Additional AudioCodes Advantages

- Rich interoperability enabling compliance with a wide variety of Softswitches for the SIP Trunking Service*
- Rich interoperability enabling compliance with a broad range of IP-PBX vendors on the Enterprise side*
- Concurrent PSTN connection for Lifeline fallback and least cost routing
- Microsoft Certified Gateways providing an interoperability solution in many new Microsoft Office Communications Server (OCS) deployments

* Please visit AudioCodes web site for the complete interoperability list
http://www.audiocodes.com/MarcomContent aspx?voip=2240

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AudioCodes