

White Paper



# Voice over IP Networks: Managing the SIP Protocol With Link Balancers

Build **Smarter** Networks



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## 1. Executive Summary

SIP is the most widely used protocol for Voice over IP (VoIP) network communications, from retail service offerings to global enterprise deployments. SIP is supported by all the major VoIP and networking suppliers and manufacturers, and VoIP service providers as well.

The use of ISP redundancy and balancing products must be planned according to key elements defined in this document to ensure maximum return on investment when used for SIP-based applications.

## 2. How SIP operates

SIP, Session Initiation Protocol, is the Internet Engineering Task Force (IETF) protocol for VoIP and other text and multimedia sessions, like instant messaging, video, online games and other services. SIP messages are very much like HTTP (the World Wide Web protocol) or SMTP (email transmission protocol), messages consisting of headers and a message body.

SIP technical notes:

- ◆ SIP is a text-based protocol which uses UTF-8 encoding.
- ◆ SIP uses port 5060 for both UDP and TCP. SIP may use other transports.
- ◆ SIP message bodies for phone calls are defined in SDP - Session Description Protocol.

## 3. The SIP-NAT challenge

The SIP protocol suffers from NAT or firewall restrictions. SIP protocols only deal with the signaling of a telephone connection; voice data traffic is handled by another protocol (usually Real Time Protocol (RTP) over UDP) which randomizes the port used to transmit data.

In an environment where Network Address Translations (NAT) are used, the NAT device, such as a firewall or most link balancers, has no way of knowing which port is being used by the ~~RDP~~RTP protocol to transmit the voice packets, causing audio traffic to not be translated properly between the address spaces.

```

Content-Type: application/sdp
Content-Length: 248
v Message Body
v Session Description Protocol
  Session Description Protocol Version (v): 0
  Owner/Creator, Session Id (o): 3000 8001 8001 IN IP4 10.0.36.88
  Session Name (s): SIP Call
  Connection Information (c): IN IP4 10.0.36.88
  Time Description, active time (t): 0 0
  Media Description, name and address (m): audio 5006 RTP/AVP 0 8 4 18 3
    Media Type: audio
    Media Port: 5006
    Media Proto: RTP/AVP
    Media Format: ITU-T G.711 PCMU
    Media Format: ITU-T G.711 PCMA
    Media Format: ITU-T G.723
    Media Format: ITU-T G.729
    Media Format: GSM 06.10
    Media Attribute (a): sendrecv
  Media Attribute (a): rtpmap:0 PCMU/8000
  Media Attribute (a): rtpmap:8 PCMA/8000
  Media Attribute (a): rtpmap:4 G723/8000
  Media Attribute (a): rtpmap:18 G729/8000
  Media Attribute (a): rtpmap:3 GSM/8000
  Media Attribute (a):ptime:20

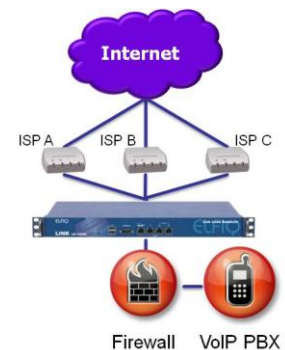
```

*Sample SDP protocol header with audio attributes and transport protocol description*

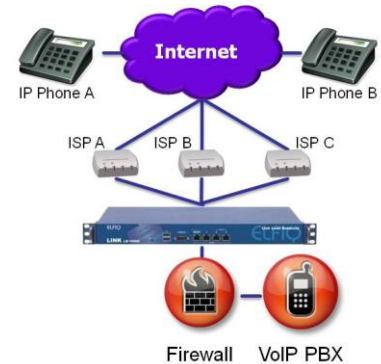
#### 4. Link balancer considerations

Link balancer products are deployed to provide bandwidth management across multiple ISPs and failover should an ISP be unavailable, and are based on NAT to handle traffic for Internet scenarios. In this context, handling SIP traffic and utilizing the benefits of link balancers can be problematic, and to address this concern, link balancing products such as those from Elfiq need to be SIP-aware to handle traffic for these scenarios:

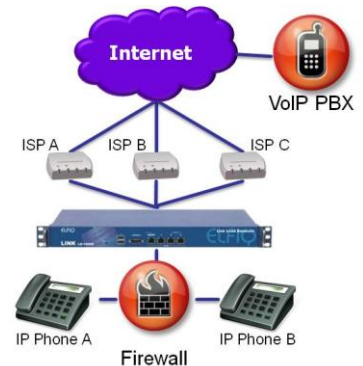
- ◆ **Managing multiple ISP links for SIP trunks:**
  - By nature, SIP trunks are not capable of being balanced since SIP requires the same source/destination address relationship or the session will break.
  - During normal operations, each SIP trunk may be used on separate ISP links.
  - For ISP link failover, should the selected SIP link for the SIP VoIP trunk fail, the link balancer must be able to supplement NAT by rewriting the addresses within the packets to re-establish the SIP trunk on an alternate link.



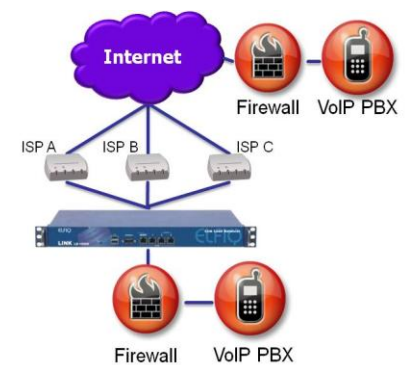
- ◆ **While managing IP phones outside the network, PBX is inside the network:**
  - Multiple phones can connect to the corporate PBX through multiple ISP links based on the balancer's set inbound balancing algorithm.
  - SIP telephones may be distributed across multiple ISP links to ensure an equal distribution of bandwidth and prevent massive loss of telephony should a link fail.
  - When an ISP link fails: the inbound balancing service provides the address of another ISP link, or the phone can directly register a second PBX address (which is the address of another ISP link) provided in its configuration. Despite the link failure, the load balancer ensures communication between the phones and the PBX.



- ◆ **Connecting to an outside IP PBX:**
  - Multiple SIP devices can connect to a PBX which is off-premises through any of the available links based on the balancer's algorithms and policies.
  - SIP clients should be balanced in a distributed fashion, having an equal number of devices assigned to each link to prevent massive loss of communications when an ISP link fails.



- ◆ **PBX to PBX**
  - In a configuration where two PBXs are connected via a SIP trunk, the SIP trunk may be assigned to any ISP link on the site with a link balancer.
  - Should the selected SIP link for the SIP VoIP trunk fail, the link balancer must be able to supplement NAT by rewriting the addresses within the packets to re-establish the SIP trunk on an alternate link.



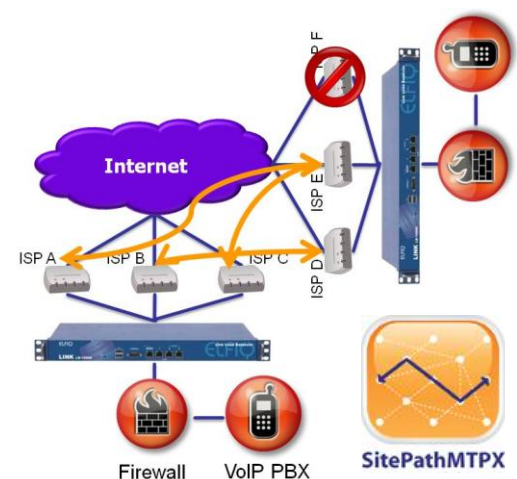
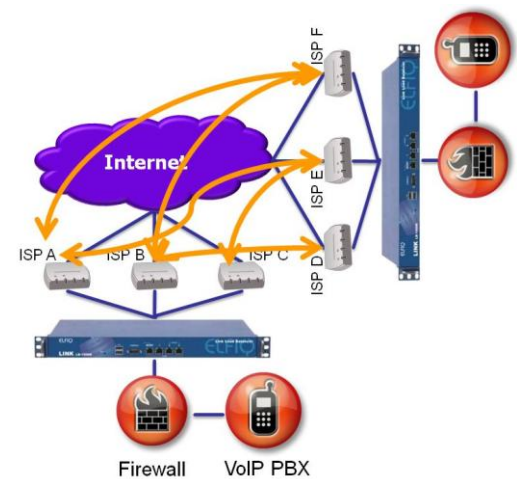
◆ **PBX to PBX with two link balancers**

- Point-to-point resilience is possible when linking two PBXs in two separate locations by employing a link balancer from Elfiq Networks at each endpoint. This method is by far the most efficient way of ensuring communications between two PBXs.

- With Elfiq Networks' SitePathMTPX technology, no SIP obstacles are present because SitePath MTPX creates "paths" between each site's ISP links and no longer relies on NAT. The PBX and the firewall are not aware of these paths and continue to operate normally. This is due to Elfiq's Layer-2 transparent integration where no modifications are required to the existing network components prior to installation of the link balancer.

- The link balancers in this scenario will exchange metrics and, depending on which algorithm is selected, the overall performance of data transfer will be improved.

- Should an ISP link fail, the SIP relationship between both PBXs is uninterrupted; the link balancers detect the failure in real time and start utilizing the remaining paths to maintain the flow of voice conversations. When the ISP link that was unavailable is able to process traffic again, it is reintegrated into the balancing process.



#### 4. Conclusion

Voice over IP communications utilizing the SIP protocol are becoming critical to most organizations with the migration to VoIP systems globally. When deploying such systems, the critical point of failure remains the ability to handle the required bandwidth. By utilizing a SIP-aware link balancer, the traditional obstacle of NAT no longer affects the opportunity to provide incremental bandwidth for increased capacity and failover circuits.

## Produced by Elfiq Networks

Elfiq Networks is a technology leader and innovator in the field of WAN link management and balancing. With successful installations in over 44 countries, Elfiq's Link Balancer products help organizations of any type and size perform more competitively every day with the ability to use multiple Internet and private links easily and securely.

For more information on Elfiq Networks' products and technologies, please contact:

Elfiq Networks  
1155 University, #712  
Montreal, Quebec, H3B 3A7  
Canada  
Telephone: 888-GO-ELFIQ / 514-667-0611  
Internet: [www.elfiq.com](http://www.elfiq.com)  
Email: [info@elfiq.com](mailto:info@elfiq.com)



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