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SIP and the Art of Converged Communications

SIP and the Art of Converged Communications

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Introduction

What is this hot topic you keep hearing so much about? To keep it simple, Session Initiation Protocol (SIP) is an internet signaling protocol, developed by the Internet Engineering Task Force (IETF), starting in 1996, for establishing, maintaining, and tearing down sessions between a variety of real-time media, including voice, video, and chat. SIP allows endpoints to locate other endpoints, whether stationary or mobile. SIP doesn't have to worry about transporting voice or video as Real Time Transport Protocol (RTP) takes care of that. It also relies on Session Description Protocol (SDP) to negotiate capabilities and codecs. SIP does not provide a directory service or authentication, but it does work with services such as LDAP or RADIUS. SIP is only concerned with signaling. This white paper looks at the way SIP is used in the converged Unified Communications environment.

SIP: What Is It Good For?

What can we use SIP for? The list is long and growing with the massive acceptance of the protocol. A few things currently using SIP are voice calls, web page click-to-dial, voice messaging, instant messaging, home automation, and interactive gaming. SIP is everywhere. In fact you may be using it without realizing it. SIP is the premier call control and signaling protocol for IP communication.

SIP plays a role with client applications such as SIP Users Agents in the SIP endpoints, network applications realized by SIP servers such as Registration Servers, Presence Servers, and IM servers to name a few, and distributed network applications in collaboration with enterprise and Service Providers. Evolving communication networks must meet business imperatives to reduce cost and improve the customer experience; user imperatives that enhance the user experience and guarantee their acceptance; and IT imperatives requiring flexible applications to accommodate changing business models and that allow for centralized configuration and administration while addressing security requirements. SIP can be implemented into an existing infrastructure or be the foundation of a new network infrastructure.

The key for enterprises will be to have a roadmap that guides them toward a fully SIP-based architecture while eliminating the need to rip and replace. While this "all SIP environment" for both wired and wireless endpoints and trunks is the eventual goal, it is essential that organizations be able to leverage existing investments and move toward that objective in a fashion that meets their budgetary requirements.

The Benefits of SIP

- SIP is indifferent to the media content it carries. This opens support for a variety of communication possibilities, e.g., video, voice, instant message (IM), etc., sessions.
- It uses an Address of Record (AOR), a single unifying public address for that user that links the user to all compatible devices:one phone number that can find you anywhere.
- It supports access to the PSTN as it ports the PSTN number to the SIP Media Gateway.
- SIP is based on Internet Protocols HTTP and SMTP, using the same messaging format and header structure, making it easily integrated with other Internet services and applications. It is text-based so it is easy to read

Sample SIP Packet:

```
INVITE sip:jones@abc.edu SIP/2.0
Via: SIP/2.0/UDP library.def.edu:5060;branch=z9hG4bKa329c
Max-Forwards: 70
To: C. Jones <sip:Jones@abc.edu>
From: Jack Smith<sip:sj.smith@def.edu>;tag=14937
Call-ID: 873625464@library.def.edu
CSeq: 1 INVITE
Subject: About 2013...
Contact: <sip:j.smithd@def.edu>
Content-Type: application/sdp
Content-Length: 167

v=0
o=Smith 6035554791 6035554791 IN IP4 library.def.edu
s=Phone Call
c=IN IP4 170.180.190.200
t=0 0
m=audio 52306 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

See how easy it is to read? The first word of the SIP packet, INVITE. is referred to as a method. Think of it as a verb or command used by SIP. Every message starts with a method. The original Request for Comment (RFC)) defined INVITE, BYE, REGISTER, CANCEL, ACK, and OPTIONS as SIP commands.

With the increased adoption of SIP, many more methods have been defined including one you are probably familiar with, MESSAGE. The ability for you to IM anyone, without concern for the device they are using or the network they are on, is due to the adoption of Session Initiation Protocol for Instant Messaging and Presence Leveraging Extensions (SIMPLE).

Sample IM SIP Message:

MESSAGE sip:owd@greenachers.com SIP/2.0

Via: SIP/2.0/UDP extensionservice.landgrantu.edu ;branch=z9hGbK4657

Max-Forwards: 70

To: Douglas <sip:owd@greenachers.com>;tag=54913

From: Henry Kimball <sip:h.kimball@landgrantu.edu>;tag=8705551212

Call-ID: ask88ask77@extensionservice.landgrant.edu

CSeq: 1 MESSAGE

Content-Type: text/plain

Content-Length: 36

About SIP and….

The Four (Voice) End-User Functions of SIP

SIP is limited to the setup, modification, and termination of sessions. It serves four major purposes:

- Determine the location of the recipient whether at work, home, mobile
- Deliver a description of the session (invitation) such as CODECs and calling-party information
- Deliver a response to the invitation as the call is accepted or rejected
- Tear down the session when one party hangs up

SIP Architecture Components

SIP architecture components are the basic building blocks of SIP-enabled communications that allow SIP-enabled solutions to work across networks and telecommunications infrastructures and products to enhance and revolutionize the world of real-time business communications.

- SIP User Agents (SIP Terminals) are the interfaces that initiate and terminate signaling
- SIP Registration Servers authorize and register user agents.
- SIP Location Servers contain the database that keeps track of users and their locations.
- SIP Proxy Servers process the SIP request on behalf of the user agents .
- SIP Redirect Servers map the SIP request to the device closest to the user.
- SIP Presence Servers accept, store, and distribute user information such as location, registration, and status.
- Session Border Controllers or Application Level Gateways sit at the edge of the SIP network and allow SIP session information to pass securely into and out of a customer network.

SIP architectures can be made up of separate servers for each role or with servers that can assume multiple roles.

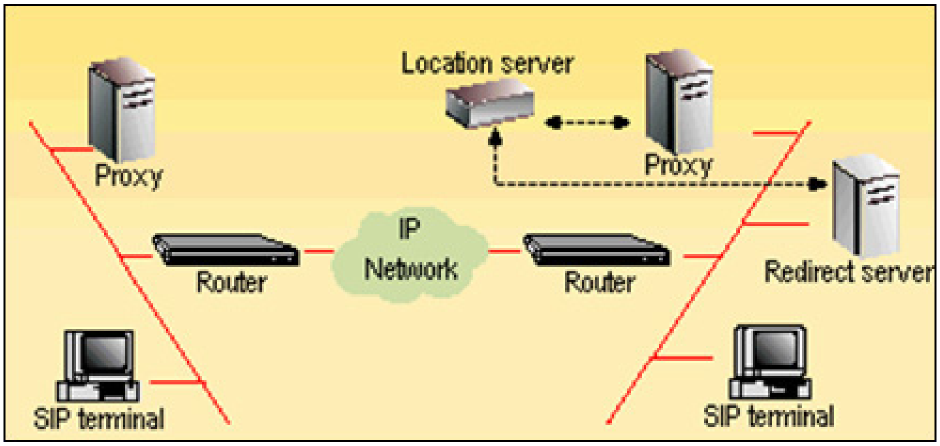

Figure 1. SIP Architecture Components

SIP Trunks

The deployment of SIP trunks is becoming very popular. SIP trunks reduce cost as TDM interfaces for CO (???) and PRI trunks are no longer required for local and long-distance access. In a corporate SIP network, SIP trunks can provide the ability to establish virtual numbers in other geographic areas worldwide, providing a saving on long-distance charges. These are just two of the benefits.

The components necessary to deploy SIP trunks are:

- A PBX or call server that supports external SIP trunks
- An enterprise edge/border element device capable of understanding SIP (i.e., session border controller)
- An Internet telephony (ITSP) or SIP trunk service provider for public SIP trunks
- A robust WAN for private SIP trunks

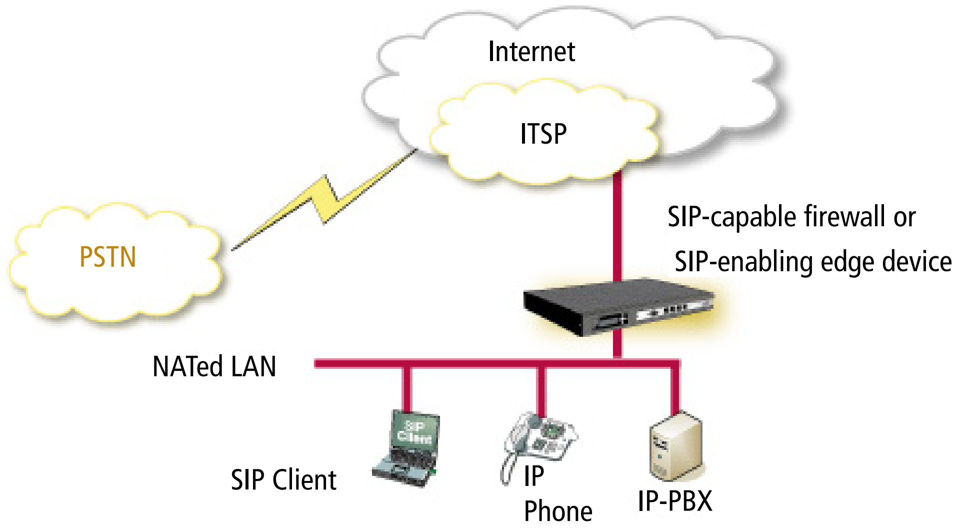

Figure 2. Components Necessary to Deploy SIP Trunks

Conclusion

Obviously, there is more to be considered when implementing a SIP solution. Why should you bother to learn about SIP? Because it is a pre-requisite for other courses, a technical curiosity, part of preparation for setting up SIP in your network, or a key technology for the evolution toward a converged communications structure.

Learn More

To learn more about how you can improve productivity, enhance efficiency, and sharpen your competitive edge, Global Knowledge suggests the following courses:

Introduction to Session Initiation Protocol (SIP) v2.0

Visit **www.globalknowledge.com** or call **1-800-COURSES (1-800-268-7737)** to speak with a Global Knowledge training advisor.

About the Author

Cheryl Nygaard has over 20 years experience training on customer premise equipment and has worked for Global Knowledge for the past 13 years. She currently holds APSS, ACIS and ACSS certifications in Avaya IP Office as well as ACIS certification in Avaya Aura Communication Manager. Cheryl is the Global Knowledge Avaya IP Ofice Course Director. She is also a Certified SonicWALL trainer and CompTIA Security+ certified.