ABOUT SIP TUTORIAL

For more information:

sales@indigosw.com
www.indigosw.com

USA / Americas

Indigo Software, Inc.
505 Montgomery Street
San Francisco, CA 94111
Sales 1 (253) 927 – 2590
Phone 1 (415) 887 – 3511
Fax 1 (415) 887 – 3001

EMEA

Indigo Software
50 Rue Wiertz
Brussels, 1050 – Belgium
Sales +32 2 235 09 60
Phone +32 2 230 35 94
Fax +32 2 280 26 76

INTRODUCTION

SIP originated in late 1996 as a component of the IETF Mbone (Multicast backbone), an experimental multicast network on top of the public Internet. It was used by the IETF for the distribution of multimedia content, including IETF meetings, seminars and conferences. Because of its simplicity, power and extensibility, SIP was rapidly adopted for other uses across the IETF, most notably as a Voice Over Internet Protocol (VoIP) standard. Today, several industry leading companies such as Cisco, Microsoft, Ericsson, Nortel have announced their support for the SIP standard.

SIP – as a part of the IETF standards process – is modeled upon other Internet protocols such as SMTP and HTTP and is used to establish, change and terminate sessions (or calls) between one or more users in an IP-based network. SIP enables the following:

- Register end points – such as IP phones and IM clients - for service access.
- Determine the location of the target end point. SIP therefore supports address resolution, name mapping, and call redirection.
- Determine the media capabilities of the target end point. Via the Session Description Protocol (SDP), SIP determines the highest level of common services between the end points. Conferences are established using only the media capabilities that can be supported by all end points.
- Register end user preferences and route calls to one or more end points based on end-user preferences.
- Register, filter and publish information about the presence - or the availability - of the target end point. If a call cannot be completed because the target end point is unavailable, SIP determines whether the called party is already on the phone or did not answer in the allotted number of rings. It then returns a message indicating why the target end point was unavailable.
- Establish a session between the originating and target end point – If the call can be completed, SIP establishes a session between the end points. SIP also supports mid-call changes, such as the addition of another end point to the conference or the changing of a media characteristic or codec.
- Handle the transfer and termination of calls — SIP supports the transfer of calls from one end point to another. During a call transfer, SIP simply establishes a session between the transferee and a new end point (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties.
- Communicate requests for Quality of Service to network elements along the path.

As SIP has been developed by the IETF, the creator of Internet technologies, SIP is built using the architectural principles of the Internet:

- Scalability: SIP uses the Internet model for scalability – fast and simple in the core, smart in the periphery.
- Simplicity: rather than inventing new protocols, SIP limits itself to what it does best – signaling – and re-uses other standard protocols for session description (SDP), media transfer (RTP), addressing (URLs), etc.
- Extensibility: history has taught the IETF that protocols are used in ways they never intended. Hence, SIP has numerous mechanisms to support extensions.
- Flexibility: SIP is not a complete system for interactive applications over the Internet. It does not determine architecture, deployment scenario or usage patterns. It does not dictate how much servers need to be used, where they reside and how they are connected. This way, carriers have unlimited flexibility in how the protocol is used and deployed.
- Easy service creation using common web-centric tools such as Java, XML, servlets, etc. This dramatically reduces service development time compared to the traditional world of telecommunications.

The SIP protocol has been formalized by the IETF in RFC 3261.

Several new protocols, of which SIMPLE is one of the most important, stemmed from the SIP standard. SIMPLE or SIP for Instant Messaging and Presence Leveraging Extensions

is a relatively new protocol that aims at standardizing the transfer of instant messages. The latest versions of the Microsoft MSN and Messenger clients support the SIMPLE protocol.

1. SIP COMPONENTS

SIP clients

SIP is a client-server protocol. The peers in a session are called User Agents (UAs). The main purpose of SIP is to enable session establishment between user agents.

A user agent can function in one of the following roles:

- User agent client (UAC)—A client application that initiates the SIP request.
- User agent server (UAS)—A server application that contacts the user when a SIP request is received and returns a response on behalf of the user.

Typically, most SIP end points are capable of functioning as both a UAC and a UAS. However, each end point functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiated the request.

SIP clients include:

- PC clients – SIP enabled client software that has instant messaging, voice and/or video capabilities.
- SIMPLE clients – instant messaging clients that support the SIMPLE standard for presence and instant messaging.
- IP Phones – can act as either a UAS or UAC and can initiate SIP requests and respond to requests.
- SIP Gateways – provide call control. Gateways provide many services, the most common being the translation between transmission formats and between communications procedures.

SIP servers

- SIP servers are applications that accept SIP

requests and respond to them. Actual SIP server implementations may contain a number of different server types, or may operate as a different type of server under different conditions.

SIP servers include:

- Proxy server —The proxy server is an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
- Redirect server—Provides the client with information about the next hop or hops that a message should take and requests the client contacts the next hop server or UAS directly.
- Registrar server—Processes requests from UACs for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.

2. HOW SIP ADDRESSES LIMITATIONS OF PREVIOUS TECHNOLOGIES

SIP is also powerful because it overcomes some of the limitations of other technologies such as H.323, MGCP and Megaco:

- SIP is future-oriented in that it performs signaling for session-based applications over an IP network. It performs signaling only, and can work with any transport protocol, existing or new.
- SIP makes use of universal addressing schemes, a SIP URL. Each person has such a universal SIP URL. The ENUM service for instance, translates the E164 phone numbers into SIP URLs.
- SIP-based routing is therefore much more flexible and powerful than PSTN routing. The latter is based on geographic information contained in the telephone number, whatever the actual user location.

- The SIP registration mechanism allows terminals used by a user to transparently register their current network locations into the network. This location is associated in the SIP network with the universal SIP address of the user, so that for any incoming call for that SIP address, the network finds the actual user location(s) and proxies the call there. Sequential and parallel searches are therefore possible and are very powerful.
- SIP therefore allows for personal and service mobility. SIP can redirect sessions to any device, on any network and in any geographical location. And SIP can provide users with their services in a mobile environment, e.g. buddy lists, voice mail, call-handling preferences or address book.
- The combination of the mobility provided by mobile networks and by the Internet allows for a whole range of new services. SIP will be a success in the convergence of mobile and fixed networks (e.g. mobile SIP to SMS).
- These features combined with the concept of people's presence for improving communication completion and the seamless integration with existing Internet applications offer a powerful framework for enhanced, real-time application development, unthinkable previously.

    It should be noted also that SIP works well together with other protocols, some of which are often viewed as overlapping in function, such as H.323, MGCP, and MEGACO.

3. SIP APPLICATIONS AND SERVICES

A number of applications that best illustrate the promises of SIP:

Enterprise applications
- VoIP within corporate private IP networks;
- Secure Instant Messaging, application sharing and whiteboarding within corporate IP networks;
- Internet voice and video conferencing;
- Click-to-talk call center buttons;
- Web-based call center ticketing with presence-based assignment to support agents;

- Automated timekeeping system;
- E-leaning solutions;
- Presence-based notification services, e.g. stock quotes.

Carrier centric services
- Instant communications: instant messaging, but also instant voice, instant video, instant conferencing and other presence-enabled applications;
- Combine presence information from web and GSTN to provide intelligent routing to PC, SIP phone, PSTN or mobile;
- VoIP, e.g. toll bypass via Internet;
- Web-initiated conferencing based on presence.

Consumer applications
- E-learning, e.g. help with homework in tutor cafés;
- Distributed gaming;
- Instant Messaging.

Wireless and 3G
- SIP registrars making use of mobile positioning information;
- Instant messaging and presence over GPRS;
- SMS-to-IM and IM-to-SMS;
- Multi-media support in 3G mobile networks.