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Session Initiation Protocol (SIP)

An Alcatel Executive Briefing

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1. What is SIP?

SIP (Session Initiation Protocol) is a signaling protocol used to establish the IP addresses and port numbers that end systems use to send and receive data. SIP is not a transport protocol (it doesn't transport data) – and the actual data packets do not follow the same path as the SIP packets. However, a mechanism does exist that allows a photo, business card information, or a web page to be transported with the signaling packets.

The SIP function involves signaling only, not the actual data being communicated. The philosophy behind SIP is to keep the communications needed to provide interoperability between Internet entities separate and simple, building additional services on top of it.

SIP is an end-to-end, client/server session signaling protocol that allows two or more Internet entities to find one another. It creates the signaling to establish paths and then tears them down when the session is over. Since SIP is not a transport protocol, RTP (Real time Transport Protocol) is needed.

SIP does not play any part in ensuring the required QoS (quality of service) is available for the actual data transfer that follows as there is no way to synchronize SIP signaling with QoS requirements.

SIP depends on the Session Description Protocol (SDP) to carry out negotiation and identification. SIP supports session descriptions that allow participants to agree on a set of compatible media types. It also supports user mobility by proxy and by redirecting requests to the user's current location.

The services that SIP provides include:

- **User location**: determination of the end system to be used for communications
- **Call set-up**: ringing and establishing call parameters for both called and calling parties
- **User availability**: determination of the willingness / availability of the called party to engage in communication
- **User capabilities**: determination of the media and media parameters to be used

Features of SIP include the following capabilities:

- **User mobility** (the use of the URL Internet address format)
  sip: john.smith@alcatel.com

SIP allows a session to be established without knowledge of an absolute IP or MAC address. This provides user mobility through the user himself declaring (programming) where he is physically located at one time, e.g., one day located at a phone in New York, the next day at a phone in Boston.

- **Manual / automatic call control** on top of the protocol. With simple, dynamic programming, SIP users can:
  - Redirect calls from unknown callers to an assistant
  - Reply with a web page
  - Send a JPEG image with the session invitation to allow the called party to see who is calling

SIP is Internet telephony, not VoIP.

2. SIP Services

SIP allows tailoring applications to the needs of individual users. Some of the basic services that SIP can provide are detailed below.

Presence services are a new form of communication. Presence is possible due to the datagram nature of the Internet. Presence can provide information about:

- Presence on the Internet (which parties are available)
- Location: office, home, traveling
- Call state: ready, on another call, etc.
- Willingness: available, in meeting, etc.
- Preferred medium: text, voice, video, email
- Instant messaging (text chat and voice chat)

SIP allows additional features to be used, for example, sending a JPEG image and / or business card with the signaling – so that a called party can see who is calling. Two major features that differentiate SIP are:

2.1 Splitting / forking a call

SIP provides the ability to split an incoming call so that several extensions ring at the same time from the same single originating call. The first extension to answer gets the call. This feature is useful if the called party is identified as being in one of several locations, each with a separate phone number, e.g., at the main office, branch office, factory, or at home. This feature can also be used to call one number and ring both a manager and his or her assistant.

2.2 Replying with alternative media

When a SIP server receives a connection request, it can return a web page to the calling SIP client. The page can be used to supply alternative numbers to call in the case of vacation time or absence due to illness, for example. The caller only has to click on the supplied Web icon to call the alternative number or party.

End users and / or third parties may program SIP. This adds tremendous flexibility in personalizing SIP communication services, allowing the user to dynamically change the response to calls. Examples include:

- Distinctive ringing: ring selected based on who is calling
- Discard all calls from a particular person / number outside business hours
- Redirect authenticated friends to my cell phone, everyone else to my assistant

3. SIP Issues

SIP packets use textual encoding, which may be considered inefficient, but it does allow much simpler debugging, and no special monitors are necessary to examine the SIP signaling packets.

Billing and accounting features are not yet defined since the current trend is to charge a flat rate for the SIP service, as with email. Issues about how to bill for calls that originate or end in a legacy telephony environment such as PSTN remain unresolved.

Emergency services (911 calls) are not included and are still under discussion.

Currently under consideration is for the transport of DTMF (dual tone multi frequency) to be in RTP and not SIP. This is because IVR (interactive voice response) is too difficult for SIP to handle, as would be the case, for example, in a flight reservation request over the phone: “Press 1 for…, press 2 for…, etc.”

The supporters of SIP are playing this issue down by declaring that in the near future, this operation will be performed using a browser or softphone to access a web site. However, IVR response using a normal telephone handset is not likely to disappear in the one- to five-year timeframe.

Since SIP is simple and effective, it will play a role in the near future for Internet communications.

4. SIP and QoS

SIP does not provide any form of QoS. In a practical sense, RSVP could be used to provide the resources necessary for real-time media transfer; however, RSVP is not being deployed everywhere. A more realistic approach is to over-provision bandwidth or provision MPLS tunneling.

5. SIP and H.323

H.323 was originally designed for video conferencing and LAN telephony, whereas SIP is designed for multimedia Internet communication. Both SIP and H.323 define mechanisms for call routing, call signaling, capabilities exchange, media control, and supplementary services.

The advantage of SIP is that it is backed by IETF, one of the most important standards bodies, while the advantage of H.323 is that it has a much larger piece of the current market.

The following is a comparison of the two protocols:

- H.323 defines hundreds of elements, while SIP has only 37 headers, each with a small number of values and parameters
- H.323 uses a binary representation for its messages, which is based on ASN.1, while SIP encodes its messages as text, similar to HTTP
- H.323 is not very scalable as it was designed for use on a single LAN and has some problems in scaling. Newer versions have suggested techniques to get around this problem
- H.323 is limited when performing loop detection in complex multi-domain searches. It can be done statefully by storing messages, but this technique is not very scalable. On the other hand, SIP uses a loop detection method by checking the history of the message in the header fields, which can be done in a stateless manner

6. SIP Conclusion

SIP is promising because it is a simple, open protocol that can be deployed in carrier and enterprise networks and because it enables new multimedia applications in voice, data, and video. For service providers, H.323 is a legacy technology that will eventually give way to SIP because of the inherent simplicity of SIP and potential for new media-blending services (Internet telephony).

However, in the enterprise market, H.323 has gained significant support because of its manageability, reliability, and interoperability with the PSTN, and will continue to exist for some time. Most serious implementers of SIP are considering a SIP-to-H.323 inter-working function / gateway as an essential element of any network.

There is a general consensus among standards organizations, companies, and technology experts that standardized procedures need to be specified to allow seamless inter-working between the two protocols. Bodies such as TIPHON (ETSI), IMTC, and IETF are working to address this topic.

Appendix A: Abbreviations and acronyms

SIP	Session Initiation Protocol
QoS	quality of service
VoIP	voice over IP
RTP	Real time Transport Protocol
IP	Internet Protocol
SDP	Session Description Protocol
URL	universal resource locator
MAC	media access control
PSTN	public switched telephone network
IETF	Internet Engineering Task Force
DTMF	dual tone multi frequency
IVR	interactive voice response
RSVP	Resource Reservation Protocol
MPLS	Multi Protocol Label Switching
TIPHON	Telecommunications and Internet Protocol Harmonization over Networks
ETSI	European Telecommunications Standards Institute
IMTC	International Multimedia Teleconferencing Consortium

Appendix B: Sources for further information on SIP

SIP Center (www.sipcenter.com)

SIP Forum (www.sipforum.org)

Session Initiation Protocol (SIP) Tutorial, 2000-2001, Jiri Kuthan and Dorgham Sisalem, GMD-Fokus (www.fokus.gmd.de/research/cc/mobis/siptutorial/)