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A Study of Session Initiation Protocol: To Manage Various Calls and Key Module of Telephone Directory

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ABSTRACT

Session Initiation Protocol (SIP) is a signaling protocol developed by SIP, which is used to setup, modify and tear down multimedia application. It is designed for networked multimedia application i.e. to establish the session in an IP network. It allows two end points to establish media session with other. The SIP is a VOIP protocol used for originating and terminating the media session. It is not only a voice protocol but a media protocol also. It doesn't have any control on the media. It conveys the media capabilities of an user agent but don't participate in the media transmission or any other. The RTP is used for media transmission. SIP specification was defined by IETF (Internet Engineering Task Force) RFC-3261(Request for Command). SIP is simple, flexible and versatile protocol. The objective of IETF is to make the internet work better by producing high quality relevant technical documents that influence the way people design, use and manage the internet. SIP has gained tremendous market acceptance for signaling communication services on the internet. SIP is designed to address the function of signaling and session's management with in a packet-based telephone network. In recent year SIP has been the choice for service related to VOIP. It is still growing and used all latest technology features.

Index Term: SIP, VOIP, IETF, Media Protocol, RFC

I. INTRODUCTION

Session Initiation Protocol (SIP) an application layer control (signaling) protocol for creating, modifying and terminating session with one or more participants in an IP network. As the name implies, the session initiation protocol is about initiation of interactive sessions between users.SIP actually doesn't define what a "session" is, this is described by content which is carried in SIP messages. Initiating a session means how the user to be contacted which is actually residing at a particular moment. A user might have a PC at work, at home and in the lab. The session may be text, voice, video or a combination of these. It a light weight signaling protocol which can be used in conjunction with other call setup and signally protocols. SIP doesn't have reserve resources but SIP can invite users to sessions with and without resource reservation. SIP doesn't reserve resources itself but can convey to the invited system, the information necessary to do this. Once the user has been called SIP perform another function- delivering a description of the session that the user is being invited to. SIP does this through the use of multipurpose internet mail extensions (MIME), which is widely used in web and email services to describe content. SIP works on UDP, TCP, RTP for call setup control and support different types of media like video conferencing and text messaging.

II. SIP CHARACTERISTICS

- User Location: It talks about the location of an end point.
- User Availability: To contacts an endpoint and determines the willingness of establishing a session.
- Session Setup: Exchange the media information to establish a session.
- Session Management: Modification of existing media session.
- Session Setup: Teardown of existing media session.

III. SIP KEY TERMS

Call Flow (Peer-To-Peer): it's a flow diagram of message between the parties. It exactly resembles how a media session is carried over.
User Agents: any endpoint which can participate in a communication session is called user agent. It usually operates in two modes-User Agent Client (UAC) and User Agent Server (UAS).
Eg. SIP phones, SIP soft clients, Web RTC etc.
UAC: who request something in a SIP session?
UAS: who responds after serving the request from UAC?
Example: in a school, student act as "UAC" and teacher act as "UAS" when student applying a leave request.

Request: is a message from UAC to UAS.
Response: is a message from UAS to UAC.
Caller: who invites the other party or who initiates a call.
Callee: who accepts that invitation or answer the calling parties.

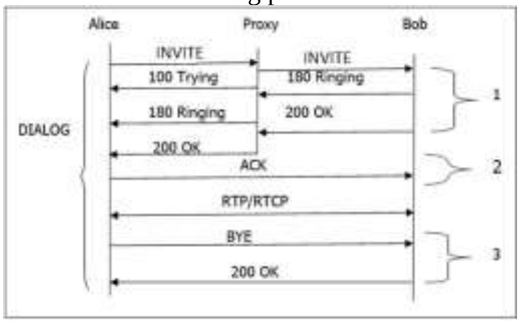


Fig: Basic SIP call flow (Peer-to-Peer)

- An INVITE request that is sent to a proxy server is responsible for initiating a session.
- The proxy server sends a **100 Trying** response immediately to the caller (Alice)
- The proxy server searches the address of Bob in the location server and forwards the INVITE request further.
- Thereafter, **180 Ringing** generated by Bob is returned back to Alice.
- A **200 OK** response is generated soon after Bob picks the phone up.
- Bob receives an **ACK** from the Alice, once it gets **200 OK**.
- At the same time, the session gets started and RTP packet or conversations start from both ends.
- After the conversation, any participant (Alice or Bob) can send a **BYE** request to terminate the session.
- Finally, Bob sends a **200 OK** response to confirm the BYE and the session is terminated.
- In the above basic call flow, three **transactions** are (marked as 1, 2, 3) available.

IV. SIP RELATED WORKING GROUP

1. SIP Ping: session initiation protocol investigation working group. It was formed to investigate the application present in SIP developing requirements for the SIP extensions and its publish "Best Current Practice" documents about the use of SIP.
2. SIP Core: responsible for core SIP standards.
3. SIMPLE: SIP for instant messaging and presence leveraging extensions working group.
4. PINT: PSTN and internetworking using SIP working group.
5. SPIRINTS: service in the PSTN/IN requesting internet services working group.

V. SIP PROTOCOLS

a) HTTP: It is used for web browsing.
b) SMTP: It is used for E-Mail. From HTTP , SIP borrowed a client/server design and use URL and URL from. From SMTP –SIP borrowed a text encoding scheme and header style.
c) SDP: is generally a part of SIP. It is defined in RFC 2327. It is used in conjunction with SIP. SDP doesn't deliver media itself but used between endpoints for negotiation of media type format and all associated properties.
d) **RTP:** It is used for delivering voice and video over IP network in SIP. It is defined by RFC 1890. RIP adds a bit-oriented header containing: Name of media source. Time setup, Codec type, Sequence number

VI. APPLICATIONS OF SIP

SIP provides multiple features like call forking with which a User can handle the incoming request from any location within IP network for getting flexible communication. There are various application of SIP.

- VOIP
- Unified communication
- Unified messaging
- IP-PBX functionality
- Web call counters
- Mobile phones and PDA's
- Desktop call management
- Voicemail messaging
- Instant messaging
- Internet call centers
- Soft switches
- Application server
- Media gateway

VII. SESSION CONTROL AND CALL CONTROL

Let we want to hold a video conference at a certain time and make it available to a wide no of participants. We have decided to encode the video stream using the MPEG-2 standard, to use the multicast IP address 224.1.1.1

for transmission of data and to send it using RTP over UDP port no 4000. So the question is how you would make call that information available to intended participants.
The IETF defined various protocols for session management.
i) SDP (Session Description Protocol)
ii) SAP (Session announcement Protocol)
iii) SIP (Session Initiation Protocol)
iv) SCCP (Simple conference control protocol)

There are 5 terms related to multimedia session.
1) Conference: it is a set of two or more communicating users along with the software they are using.
2) Session: it is a multimedia sender and receiver and flowing stream of data.
3) Session Announcement: it is a mechanism by which a session description is conveyed to user in a proactive fashion i.e. the session description was not explicitly requested by the users.
4) Session Advertisement: it is just same as session announcement.
5) Session Description: A well defined format for conveying sufficient information to discover and participate in a multimedia session.

VIII. SDP (SESSION DESCRIPTION PROTOCOL)

It is a format to agree on compatible media types and parameters for interactions. **It** is a general protocol for describing the session. It is used in conjunction with one or more other protocol. Eg. SIP
SDP is a text format for describing multimedia session. It is not really a protocol but similar to markup language like HTML. It can be carried of any protocol, Eg. RTSP or SIP. Also it describe unicast and multicast session. When describing a session the caller and callee indicate their respective “receive” capabilities, media formats and receive address/port.
The SDP convey the following information:

- The name and purpose of the session.
- Start and end time for the session.
- The media types(Eg. Audio, Video) that comprise the session.
- Detailed information needed to receive the (Eg. The multicast address to which data will be sent, the transport protocol to be used, the port no, the encoding scheme etc.

An SDP message contain 3 levels of information-
1. Session Level Description: it includes the session identifiers and other session level parameter such as IP address, subject, content information about the session or creator.
2. Timing Description: start and stop times, repeat times, one or more media level description.
3. Media type format.

IX. SDP MESSAGE FORMAT

It is a collection of SDP lines. The SDP syntax is very strict and all lines follows the same format. Like
<Character> = <value>
<Value>=parameter1, parameter2, parameter3 -----, parameter
Each SDP lines ends with a carriage return line feed and each line has a defined no of parameters. The 03 levels of information must be appear in the that order.

X. SDP MESSAGE CONTENTS

1. Session Description:
v- Protocol version
o- Originator and session identifier
s- Session name
i- Session Information
u- URL of description
e- Email-address
p- Phone No
c- Connection information (not required if included in all media)
b- Zero or more bandwidth information lines, one or more time description.
T & R- Lines
z- Time zone adjustment
a – zero or more session attribute

2. Connection information lines: The c-line must be either present at the session level or media level. It must be present at the media level. It is not present at the session level. If it is present at both levels then media level connection information overrides session level information.

C=<new type> <address type> <n/w address>

3. Time Description:
t- Time the session is active, r- Zero or more repeat times.

XI. SDP MESSAGE FORMAT

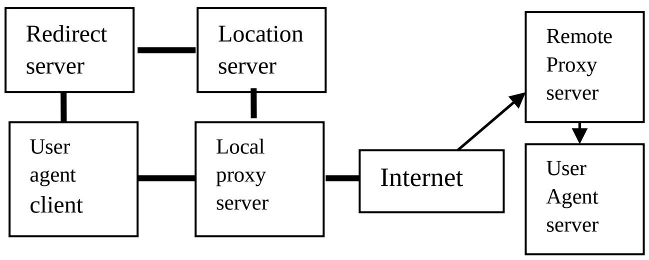
V = 0
O = ABCDE 2890844526 2890844526 IN IP4 10.120.42.3
S = IN IP4 10.120.42.3
T = 0 0
M = audio 49170 RTP/AVP 0 8 97
A = rtpmap:0 PCMU/8000
A = rtpmap:8 PCMU/8000 } Encoding Scheme
A = rtpmap:97 ILBC/8000
M = video 57372 RTP/AVP 3132
A = rtpmap: 31 H261/90000

The client use IP version 4 with an address of 10.120.42.3. It can support three audio codec's and one video codec. The audio codec's are G.711 uLaw (PCMU), G.711 aLaw (PCMA), and iLBC. The audio codec's will use port 49170 and all have a sample rate of 8000 Hz. The video codec is H.261 on port 51327.

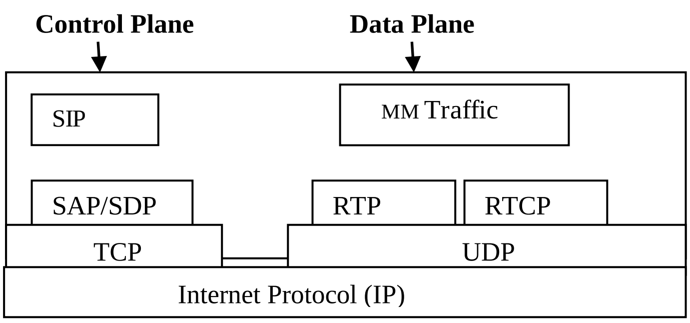
XII. SIP SERVICES

i) SIP provides mechanism for call setup.
   a. For caller to let callee know she wants to establish a call.
   b. So caller, callee can agree on media type encoding
   c. To end the call.
ii) Determine current IP address of callee
   a. Maps mnemonic identifiers to current IP address
iii) Call Management
   a. Add new media streams during call
   b. Change encoding during call
   c. Invite other
   d. Transfer and hold the calls.

Elements in a SIP	Environment



Protocol Stack of SIP



XIII. REPLY CODE IN SIP

1XX (Information) : Trying, Ringing and Queued

- 100-Trying
- 180-Ringing
- 181-Call is being forwarded
- 182-Queued
-

2XX (Successful) : The request was successful

- 200 OK-Request succeeded
- 202-Accepted

3XX (Redirection): Give information about the receiver new location.

- 300-Multilevel choices
- 301-Moved permanently/temporary
- 305-Use proxy

4XX (Request Failure): Failure response from a particular server.

- 401-Unauthorized
- 407-Proxy authentication required
- 408-Request timeout
- 480-Temporarily unavailable

5XX (Server Failure): Failure responses given when a server itself has error.

- 500-Server interval error
- 502-Bad gateway
- 504-Server timeout
- 505-Version not supported

6XX (Global Failure): Busy, decline, request not acceptable.

- 600-Busy everywhere
- 603-Decline
- 604-Does not exist anywhere
- 606-Not acceptable

XIV. SIP ARCHITECTURE OR ENTITIES

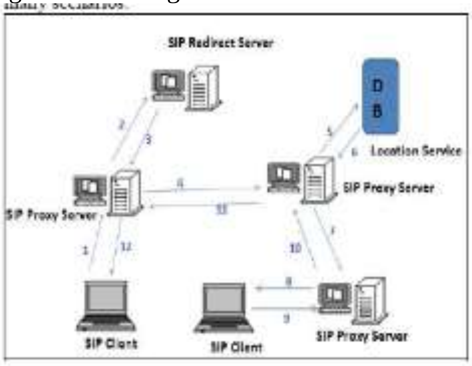
A SIP environment consists of number of entities.
User Agents:
Client: Make request
Server: Accept request
SIP server type:
Redirect Server
Proxy Server
Registrar Server
Location Server

Proxy server: These are most common type of server in a SIP environment. When a request is generated the exact address of the recipient is not known in advance. So client send the request to the proxy server. The server on behalf of the client forwards the request to proxy server or the recipient itself.

Redirect Server: It redirects the request back to the client indicating that the client needs to try a different route to get the recipient. It generally happens when a recipient has moved from its original position either temporarily or permanently.

Registrar Server: We know one of the prime jobs of the server is to detect the location of the user in a network. How do they know the location?. Basically the user has to register their location to a registrar server. A user from time to time refreshes their location by registering (sending a specific type of message to the registrar server).

Location Server: The addresses registered to a registrar are stored in a location server.



XV. SIP MESSAGES

There are two types of messages.

1. Request Message: Initiated by client to the server including INVITE, PRACK, BYE, CANCEL, UPDATE, OPTIONS etc.

2. Response Message: it is used to display the response of the server to the request of clients including 1xx, 2xx, 3xx, 4xx, 5xx, 6xx and ACK. Both the request and response message contain SIP header fields and SIP message fields.
   Invite: Request a session
   ACK: final response to the invite
   Options: ask for server capabilities.
   Cancels: cancels the pending request.
   BYE: terminate the sessions.
   Register: send user's address to the server
   Notify: to send event notice.
   Message: instant messaging
   Refer: call transfer
   Info: it is rarely used. It Provides mid-call session related information.

XVI. SIP HEADER RESPONSE

SIP Message=Request line | Response line
[HEADERS]CRLF
……….
CRLF
MESSAGE BODY (optional)
Request-line=
<METHOD>SP<REQUEST_URI>SP<VERSION
>CRLF
Response-line=
<VERSION>SP<Response-code>SP<Reason phrase>
CRLF

XVII. SIP REQUEST COMMAND FORMAT

Command Name	**Peer End User**	**Protocol Version**
Call-Id : Value		
Via : Value		
From : Value		
To : Value		
Contact : Value		
CSeq : Value		
Content-length: value		
Max-Forward: value		
Content-type:value		
SDP		

XVIII. SIP Response Format

Protocol Version	**Response Message Header**
Call-Id : Value	
Via : Value	
From : Value	
To : Value	
Contact : Value	
CSeq : Value	
Content-length: value	
Max-Forward: value	
Content-type: value	
SDP	

XIX. SIP Transactions

A transaction consists of a request, any non-final (1xx) response received and a final response (2xx, 3xx, 4xx, 5xx, 6xx) as well as the ACK of the responses. (ACK or PRACK) accepts for ACKs to 2xx responses.
Example:

a) SIP peer A sends an invite request to SIP peer B.
b) SIP peer B return a response of 100 trying, this is a non final response, so this transaction is not completed yet.
c) SIP peer B returns 200 ok (A final response) accepting the invitation, this complete the transaction.

XX. SIP DIALOGS

A dialog is just a series of transaction between two SIP peers. The purpose of dialog is to setup, modify and then tear down a session. There are many dialogs in progress between two SIP peers at any time. (Eg: there could be many simultaneous calls in progress between two SIP services.)
Dialogs are identified by the from, To and call ID fields in the header.
Example:

a) SIP peer A invites SIP peer B to a session and suggest a certain codec but doesn't include verification and so call is rejected.
b) SIP peer A again invites SIP peer B to a session, this time giving confirmation and invitation is accepted.
c) SIP peer B send an invitation to change the code used and it is accepted.
d) SIP peer A end the session.

XXI. CONCLUSION

SIP is based on request response phenomenon. It has gained tremendous market acceptance for signaling communication services on the internet. It is cost effective Communication based environment. It can run over fixed And wireless networks. It is much fastest increasing application layer protocol which provides interactive services like voice, video, text, invite users for session, modification and termination of session. SIP is based on KISS (keep it simple stupid) phenomena which can easier to implement and interoperate. It also allows standard based extension to perform specific functions. The most exciting application of SIP is a protocol platform which can be used for instant messaging. With SIP, everything that has been learned from web services can now used in third generation mobile networking also. The main challenge is the SIP message contains information that client or server keeps private, but in VOIP system and distributed architecture it is very difficult to keep confidential.

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