IJCSE International Journal of Computer Sciences and Engineering Open Access
Research Paper Volume-X, Issue-X E-ISSN: 2347-2693

Session Initiation Protocol (SIP)

Amit Suryawanshi[1*], Monika Bhati[2] & Rupesh Dubey[3]

[1*]ME(ECE), Rajiv Gandhi Technical University, INDIA, amit14surya@yahoo.com
[2]ECE, Rajiv Gandhi Technical University, INDIA, monikabhati87@gmail.com
[3] ECE, Rajiv Gandhi Technical University, INDIA, rupesh_dubey7@yahoo.co.in

www.ijcaonline.org

Received: 11 Nov 2013 Revised: 16 Nov 2013 Accepted: 22 Nov 2013 Published: 30 Nov 2013

Abstract**—** Session Initiation Protocol (SIP) is a signaling protocol developed by SIP working group specified by the Internet Engineering Task Force (IETF), SIP is a text-based protocol similar to SMTP and HTTP, used to create, manage and terminate sessions on IP data network at application layer. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. Such sessions include peer-to-peer voice, video, chat, interactive games, and virtual reality. This makes possible to implement services like voice-enriched e-commerce, web page click-to-dial or Instant Messaging with buddy lists on IP based data network resulting in reduced costs, improved customer service and easily manageable communications architecture. SIP is designed to address the functions of signaling and session management within a packet based telephony network. Signaling allows call information to be carried across the network boundaries. Session management provides the ability to control the attributes of an end-to-end call. In recent years, SIP has been the choice for services related to Voice over IP (VoIP). It is a standard (RFC 3261) proposed by Internet Engineering Task Force (IETF). SIP is still growing and being modified to take into an account all relevant features as the technology expands and evolves.

Index Term**—** SIP , IETF , SMTP, Text-Based Protocol

I. INTRODUCTION

Session Initiation Protocol [Ros01] is the most widely adopted protocol for VoIP and also for other real-time multimedia applications. SIP allows users to establish, modify, and terminate sessions between individual or multiple users. SIP requires the use of users data gram protocol (UDP), and TCP is optional. It works in performance with other protocols such as Session Description Protocol (SDP) [Han02] and Real-Time Transport Protocol (RTP) [Sch03] to offer a complete architecture for call set-up, control, and to support different types of media like video conferencing, text messaging. SDP is the encoded body of the SIP message which contains information about what media types the parties can and will use. RTP is an IP based protocol providing support for transport of real-time data such as video and audio streams in the form of data packets over IP. It adds a bit-oriented header containing name of media source, timestamp, codec types and sequence number. Destination in SIP is represented with Uniform Resources Indicator (URI), which have the same format as e-mail addresses. Two types of SIP URI's:

1. Address Of Record (AOR) :Identifies a user
2. Fully Qualified Domain Name (FQDN) : Identifies a device

More specifically, the telephony world was initially intended only for the voice services but with the advancement in the technologies the need for the data transfer comes into the picture. The old telephony networks were circuit switched but now they got transformed to packet switched. With the development of packet switched technologies, the users requirements had changed from circuit switched to high speed packet based technologies for data transfer services. The packet based data networks are faster and provide minimal delay.

Moreover, the voice is moving in the form of packets from one end to other end of the network. In this case, the voice signals are segmented into frames, encoded and encapsulated in RTP. The packets are then transported over an IP network. In addition, determine the media capabilities of the target end point Via Session Description Protocol (SDP). Integrating a packet switched network with a circuit switched network is necessary in order to provide significant cost saving, to achieve better performance, and improving interconnectivity with mobile terminals. A number of VoIP protocols exist today like H.323, SIP, Media Gateway Control Protocol (MGCP), and etc.

When the technology changes, power consumption has become a big issue now days. In [Lan04], the authors shown that the power consumed by the network elements like switches and equipment in homes, and building is 30% and 40% of the total power consumed for network equipment's. Moreover, the network equipment in a building almost consumes 15% of the electricity. Hence, we required some effective measures to reduce energy consumption as it will lead to reduce in heat emission by devices and saving money.

II. SIP ARCHITECTURE

A SIP environment consists of a number of connected entities.

Corresponding Author: *Amit Suryawanshi*

- User Agent Client/Server
- Proxy Server
- Redirect Server
- Registrar

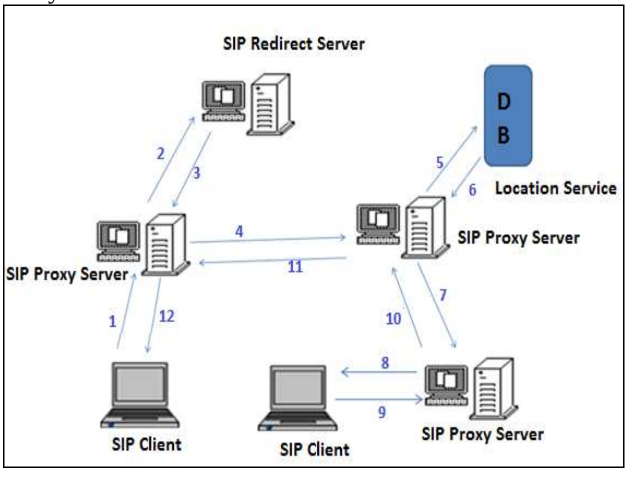
2.1 USER AGENT/CLIENT: - A User Agent (UA) is the entity which represents an end user in a client device. It usually operates in two modes: a User Agent Client (UAC) sends the initial request messages and processes responses; and a User Agent Server (UAS) accepts requests and sends responses

2.2 PROXY SERVER: - Proxy Servers [6] are involved in routing the SIP messages to the correct endpoint. Stateful proxies sometime make use of User Agents in a logical entity called a Back-To-Back-User-Agent

2.3 REDIRECT SERVERS: - Redirect Servers provides a new address or different route path to the recipient. The server may make use of a Location Server to persist location information

2.4 REGISTRAR: - A Registrar acts as current repository of a client's attachment to the network it is the User Agent that tends to reside on the end user's device. It receives registrations regarding current user locations.

The other Entities provide essential support services in many scenarios.



III. Message Types

A SIP message could be either a request or a response followed by status/request line, header field and optional part.

Request: sent from client to a server and define the operation sought by the client.

Response: sent from server to a client and provide the status of that request.

3.1 Request Message types:-

REGISTER – Login, registers the address listed in the TO header field with a SIP server INVITE – Start a call

ACK: - Confirms that the client has received a final response to an INVITE request

CANCEL: - abort a call setup

BYE: - End a call, Terminates a call and can be sent by either the calling or the called party

OPTIONS:-Queries the capabilities of servers

INFO:-Provides mid-call session related information. It is rarely used.

REFER - Call Transfer

MESSAGE - Instant messaging

SUBSCRIBE / NOTIFY – Used to request receipt of future NOTIFY or PUBLISH requests.

3.2 Response Message types:-

1xx:Provisional/Informational:-Request received,continuing to Process the request.

100-Trying

180-Ringing

181-Call is being forwarded

182-Queued

2xx: Success/Final: - The action was successfully received, understood, and accepted.

200 OK-Request succeeded

202-Accepted

3xx: Redirection: - Further action needs to be taken in order to complete the request.

300-Multilevel choices

301-Moved permanently/temporary

305-Use proxy

4xx: Client Error: - The request contains bad syntax or cannot be fulfilled at this server.

401-Unauthorized

407-Proxy authentication required

408-Request timeout

480-Temporarily unavailable

5xx: Server Error: - The server failed to fulfill an apparently valid request.

500-Server interval error

502-Bad gateway

504-Server timeout

505-Version not supported

6xx: Global Failure: - The request cannot be fulfilled at any server.

600-Busy everywhere

603-Decline

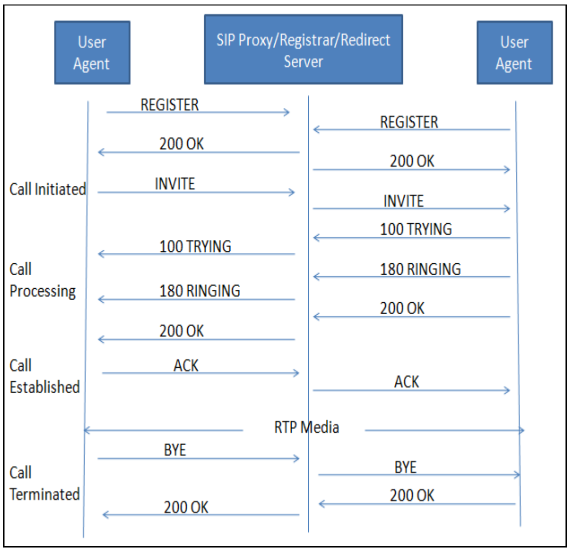
604-Does not exist anywhere

606-Not acceptable

IV. SIP HEADERS, RESPONSES

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

V. CALL SETUP PROCEDURE



VI. FEATURES OF SIP

SIP supports forking which means SIP server can send one incoming request to more than one UAS by collecting all the contact information's from redirect server for the particular destination. SIP message is using the MIME (Multipurpose Internet Mail Extension) [05] mechanism hence a message may contain information in text or binary forms. SIP allows a user to move between different locations by use of location server that register the current location (IP address) of the user and also provides some call features like

Call Transfer
Conferencing
Speed Dial
Call Forwarding/forking
Call Holding
Call Waiting
3-Way Calling

VII. SIP IDEOLOGIES

Creates, modifies and terminates sessions
Clients-Server model
HTTP based syntax
Text based protocol
Intelligent end points
Flexible headers
Only signaling
Makes use of RTP/RTCP for media transfer

VIII. APPLICATIONS OF SIP

SIP can be integrated into product such as:

- IP phones.
- Media Gateways.
- Web-enabled telephony portals.
- Internet call centers.
- Soft switches.
- Application servers

SIP provides multiple features like call forking with which a user can handle the incoming request from any location within IP network hence getting a flexible means of communication. It's easy to manage different sessions such as voice, video with different users at the same instant of time.

IX. RECOMMENDATIONS

SIP request message contains too many fields like INVITE, To, From, Via, Call-ID, CSeq, Contact, Content-Type, Content-Length, Max-Forwards, and etc. these fields need to be completely processed by the SIP entities though some of the fields are not important for them but in order to get their required details they have to process the whole message which is time consuming. If "Parsing and Framing" is applied to the message then processing time at each entity will be less as the Parsing will chop down the message into separate fields and then each entity will be getting only those fields which they actually wants and when required then framing the message back into its original combined form with Framing.

X. CONCLUSION

SIP is one of the fastest growing application layer protocol which is providing enhancement to IP based interactive services such as voice, video, text and much more with its attractive services like invite users to sessions, modification and termination of sessions all just with small set of messages and responses. SIP is cost effective means of communication in an IP based environment. It can run over most fixed And wireless networks. SIP uses the internet model and maps it onto telecom world .It uses email like addresses to identify users and accordingly forward the incoming request to one or multiple UAS, allowing users to move between different networks easily which makes SIP more powerful and reliable protocol. With the application of framing and parsing mechanism the speed of establishing a session could be faster, resulting in time saving.

REFERENCES

[1] J. Rosenberg, H. Schulzrinne, G. Camarillo, A.Johnston, J. Peterson, R. Sparks, M. Handley, and E.Schooler, \SIP: Session Initiation Protocol," RFC 3261(Proposed Standard), Jun. 2002, updated by RFCs 3265, 3853,4320, 4916. [Online]. Available: http://www.ietf.org/rfc/rfc3261.txt

[2] Handley, M., V. Jacobson, "SDP: Session Description Protocol," RFC 2327, April 1998.

[3] Schulzrinne, H., S. Casner, R. Frederick, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," RFC 3550, July 2003.

[4] S. Lanzisera, B. Nordman, and R. Brown, "Data network equipment energy use and savings potential in buildings," Energy Efficiency, vol. 5, pp. 149-162, 2012, 10.1007/s12053-011-9136-4.

[5] N. Freed and N. Borenstein, \Multipurpose internet mail extensions (mime) part one: Format of internet message bodies," RFC 2045 (Proposed Standard), United States, 1996. [Online]. Available: http://www.ietf.org/rfc/rfc2045.txt.

[6] J. Elwell, \Connected Identity in the Session Initiation Protocol (SIP)," United States, 2007. [Online]. Available: http://www.ietf.org/rfc/rfc4916.txt

[7] STREAM CONTROL TRANSMISSION PROTOCOL SUPPORT IN SESSION INITIATION PROTOCOL PROXY SERVER by Thomas Kwok Cheong Pang B.A.Sc. (Computer Engineering), Simon Fraser University, 1995.

[8] Session Initiation Protocol , KK Tan and HL Goh. P.13101314 Volume 2, IEEE International conference on Information Technology, 11:th14:th of December 2002:

[9] In-Stat - VoIP penetration forecast to reach 79% of US businesses by 2013

[10] S. Baset, J. Reich, J. Janak, P. Kasparek, V. Misra, D. Rubenstein, and H. Schulzrinnne, "How green is IP-telephony?" in Proceedings of the first ACM SIGCOMM workshop on Green networking, 2010, p. 7784.

[11] Walsh, Steffan Fries, National Institute of Standards and Technologies. NIST special publication 80058. January 2005, p. 3945.

[12] Design and Implementation of a SIPbased VoIP architecture , S. Zeadally and F. Siddiquini IEEE AINA 2004 SIP, Session Initiation Protocol used in VoIPKarin Tybring,Petra Wahlund,Zhu Yunyun

Authors Profile

1) Amit Suryawanshi.
Student ME IInd year
IPS EC Dept.

2) Prof. Monika Bhati
Associate Prof.
E&C Engg.
Institute of Engg. & Science
IPS Acodemy, Rajendra Nagar, Indore [M.P.]
monikabhati87@gmail.com

3) Prof. Rupesh Dubey
Associate Prof. and
Head Dept. of E&C Engg.
Institute of Engg. & Science
IPS Acodemy, Rajendra Nagar, Indore [M.P.]
rupesh_dubey7@yahoo.co.in

