SIP

session initiation protocol

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About this Tutorial

SIP is a signalling protocol designed to create, modify, and terminate a multimedia session over the Internet Protocol. It is an application layer protocol that incorporates many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP).

This tutorial covers most of the topics required for a basic understanding of SIP and to get a feel of how it works.

Audience

This tutorial has been prepared for professionals aspiring to learn the basics of SIP and make a career in telecom testing.

Prerequisites

Before proceeding with this tutorial, you should have a good grasp over preliminary networking concepts including some of the basic protocols such as TCP, UDP, HTTP, SMTP, and VoIP.

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1. SIP — Introduction

Session Initiation Protocol (SIP) is one of the most common protocols used in VoIP technology. It is an application layer protocol that works in conjunction with other application layer protocols to control multimedia communication sessions over the Internet.

VoIP Technology

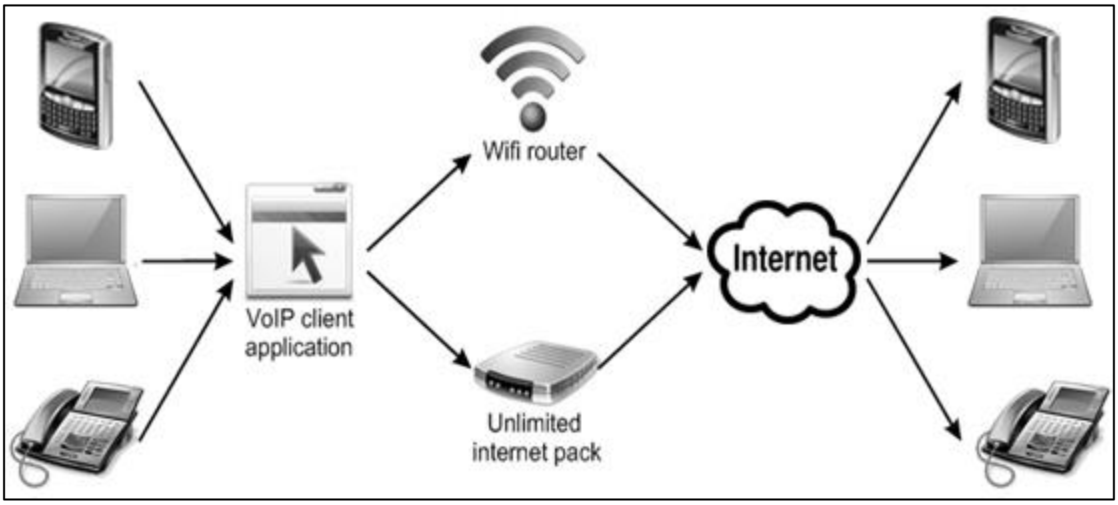
Before moving further, let us first understand a few points about VoIP.

- VOIP is a technology that allows you to deliver voice and multimedia (videos, pictures) content over the Internet. It is one of the cheapest way to communicate anytime, anywhere with the Internet's availability.

- Some advantages of VOIP include:

    - Low cost
    - Portability
    - No extra cables
    - Flexibility
    - Video conferencing

- For a VOIP call, all that you need is a computer/laptop/mobile with internet connectivity. The following figure depicts how a VoIP call takes place.



With this much fundamental, let us get back to SIP.

SIP – Overview

Given below are a few points to note about SIP:

- SIP is a signalling protocol used to create, modify, and terminate a multimedia session over the Internet Protocol. A session is nothing but a simple call between two endpoints. An endpoint can be a smartphone, a laptop, or any device that can receive and send multimedia content over the Internet.

- SIP is an application layer protocol defined by IETF (Internet Engineering Task Force) standard. It is defined in **RFC 3261**.

- SIP embodies client-server architecture and the use of URL and URI from **HTTP** and a text encoding scheme and a header style from **SMTP**.

- SIP takes the help of SDP (Session Description Protocol) which describes a session and RTP (Real Time Transport Protocol) used for delivering voice and video over IP network.

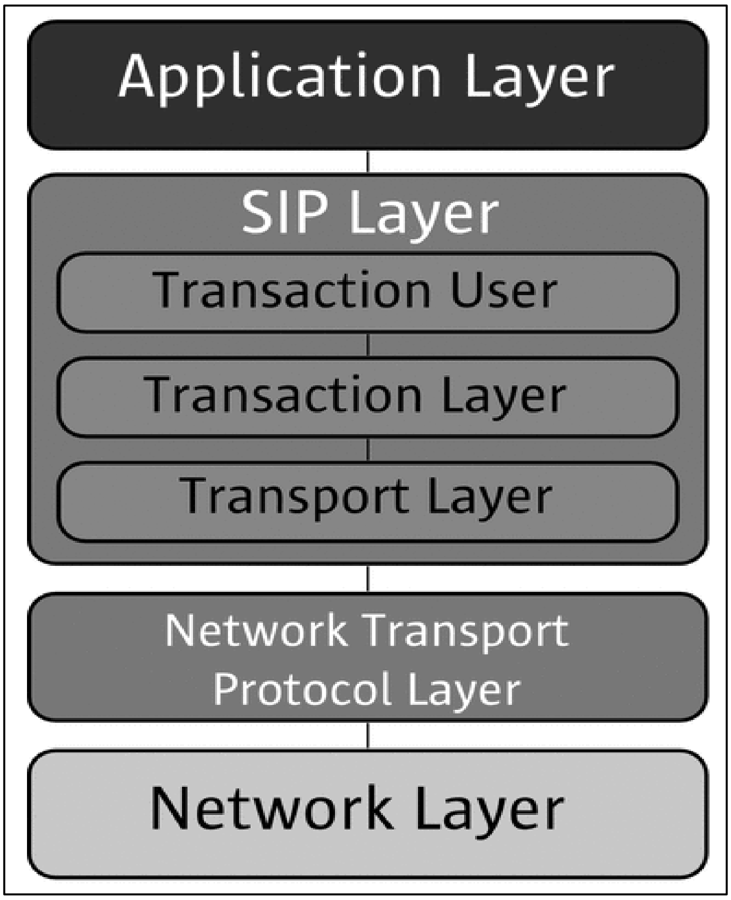
- SIP can be used for two-party (unicast) or multiparty (multicast) sessions.

- Other SIP applications include file transfer, instant messaging, video conferencing, online games, and steaming multimedia distribution.

Where Does SIP Fit In?

Basically SIP is an application layer protocol. It is a simple network signalling protocol for creating and terminating sessions with one or more participants. The SIP protocol is designed to be independent of the underlying transport protocol, so SIP applications can run on TCP, UDP, or other lower-layer networking protocols.

The following illustration depicts where SIP fits in in the general scheme of things:



Typically, the SIP protocol is used for internet telephony and multimedia distribution between two or more endpoints. For example, one person can initiate a telephone call to another person using SIP, or someone may create a conference call with many participants.

The SIP protocol was designed to be very simple, with a limited set of commands. It is also text-based, so anyone can read a SIP message passed between the endpoints in a SIP session.

2. SIP — Network Elements

There are some entities that help SIP in creating its network. In SIP, every network element is identified by a **SIP URI** (Uniform Resource Identifier) which is like an address. Following are the network elements:

- User Agent
- Proxy Server
- Registrar Server
- Redirect Server
- Location Server

User Agent

It is the endpoint and one of the most important network elements of a SIP network. An endpoint can initiate, modify, or terminate a session. User agents are the most intelligent device or network element of a SIP network. It could be a softphone, a mobile, or a laptop.

User agents are logically divided into two parts:

- **User Agent Client (UAC):** The entity that sends a request and receives a response.

- **User Agent Server (UAS):** The entity that receives a request and sends a response.

SIP is based on client-server architecture where the caller's phone acts as a client which initiates a call and the callee's phone acts as a server which responds the call.

Proxy Server

It is the network element that takes a request from a user agent and forwards it to another user.

- Basically the role of a proxy server is much like a router.

- It has some intelligence to understand a SIP request and send it ahead with the help of URI.

- A proxy server sits in between two user agents.

- There can be a maximum of 70 proxy servers in between a source and a destination.

There are two types of proxy servers:

- **Stateless Proxy Server:** It simply forwards the message received. This type of server does not store any information of a call or a transaction.
- **Stateful Proxy Server**: This type of proxy server keeps track of every request and response received and can use it in future if required. It can retransmit the request, if there is no response from the other side in time.

Registrar Server

The registrar server accepts registration requests from user agents. It helps users to authenticate themselves within the network. It stores the URI and the location of users in a database to help other SIP servers within the same domain.

Take a look at the following example that shows the process of a SIP Registration.

SIP Registration Example

Here the caller wants to register with the TMC domain. So it sends a REGISTER request to the TMC's Registrar server and the server returns a 200 OK response as it authorized the client.

Redirect Server

The redirect server receives requests and looks up the intended recipient of the request in the location database created by the registrar.

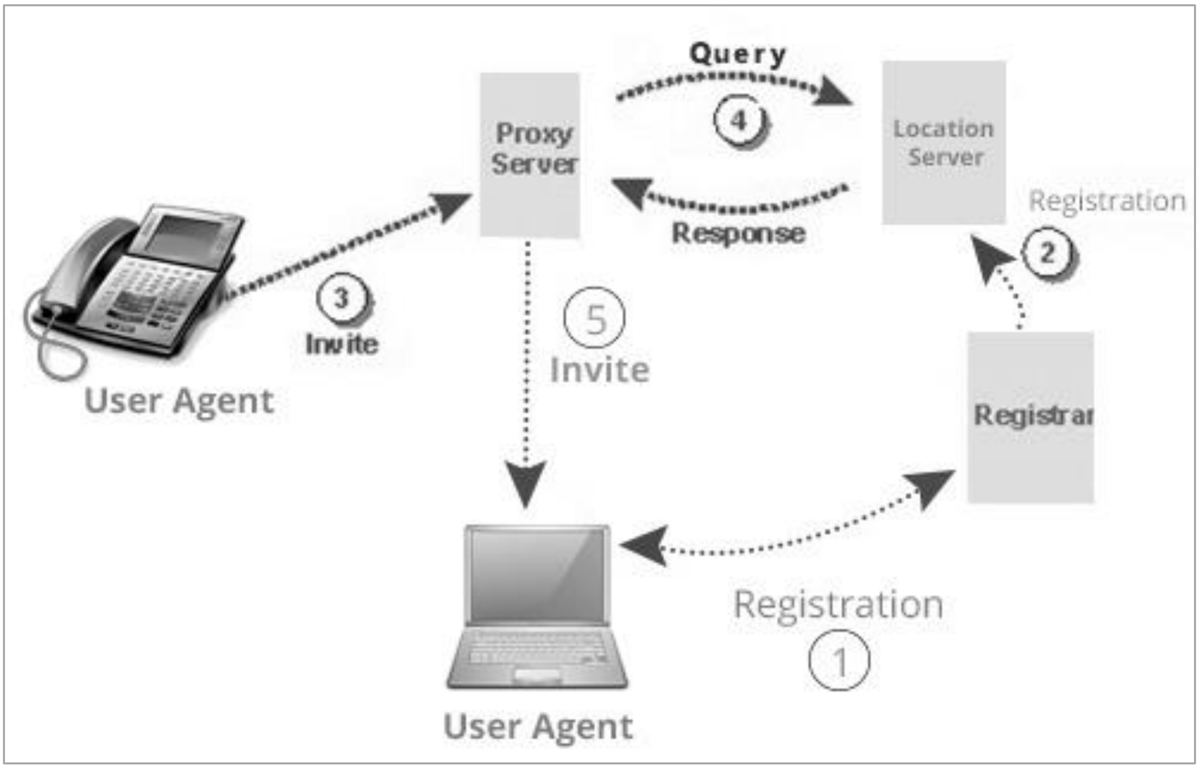
The redirect server uses the database for getting location information and responds with 3xx (Redirect response) to the user. We will discuss response codes later in this tutorial.

Location Server

The location server provides information about a caller's possible locations to the redirect and proxy servers.

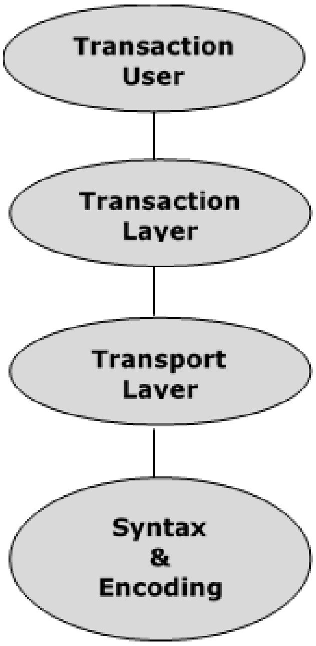
Only a proxy server or a redirect server can contact a location server.

The following figure depicts the roles played by each of the network elements in establishing a session.



SIP – System Architecture

SIP is structured as a layered protocol, which means its behavior is described in terms of a set of fairly independent processing stages with only a loose coupling between each stage.



- The lowest layer of SIP is its **syntax and encoding**. Its encoding is specified using an augmented **Backus-Naur Form grammar** (BNF).

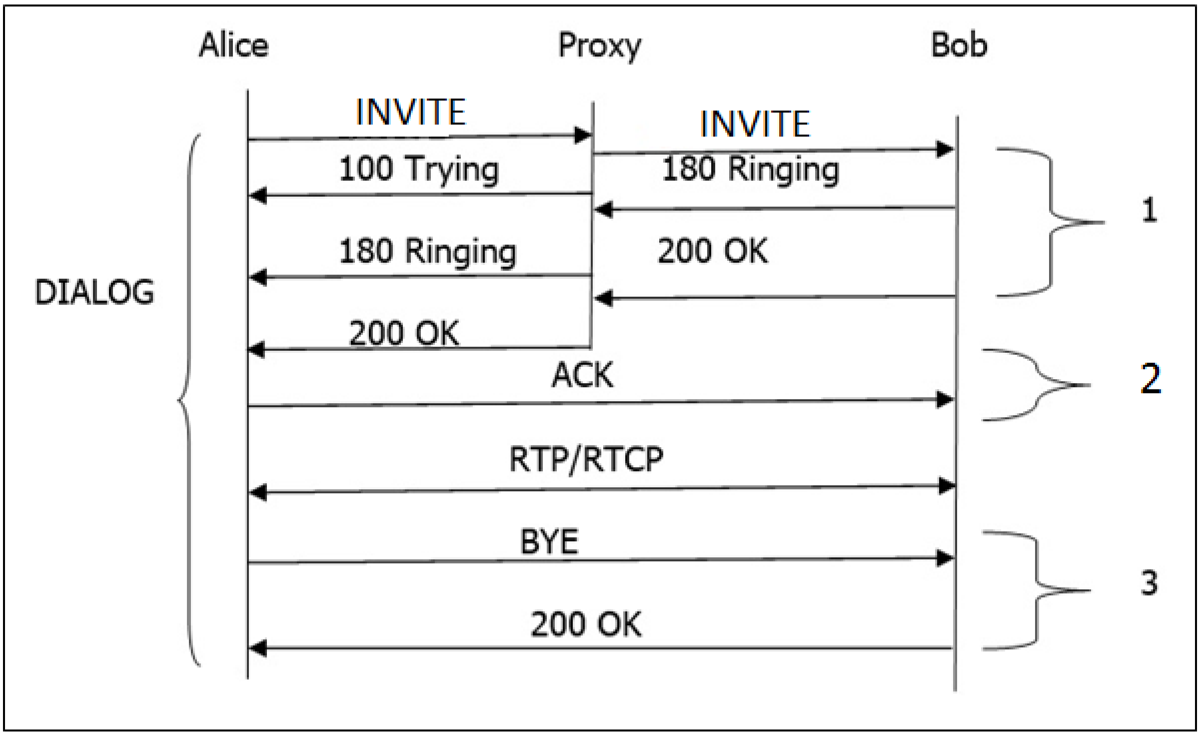
- At the second level is the **transport layer**. It defines how a Client sends requests and receives responses and how a Server receives requests and sends responses over the network. All SIP elements contain a transport layer.

- Next comes the **transaction layer**. A transaction is a request sent by a Client transaction (using the transport layer) to a Server transaction, along with all responses to that request sent from the server transaction back to the client. Any task that a user agent client (UAC) accomplishes takes place using a series of transactions. **Stateless proxies** do not contain a transaction layer.

- The layer above the transaction layer is called the **transaction user**. Each of the SIP entities, except the **stateless proxy**, is a transaction user.

3. SIP — Basic Call Flow

The following image shows the basic call flow of a SIP session.



Given below is a step-by-step explanation of the above call flow:

1. An INVITE request that is sent to a proxy server is responsible for initiating a session.

2. The proxy server sendsa**100 Trying** response immediately to the caller (Alice) to stop the re-transmissions of the INVITE request.

3. The proxy server searches the address of Bob in the location server. After getting the address, it forwards the INVITE request further.

4. Thereafter,**180 Ringing** (Provisional responses) generated by Bob is returned back to Alice.

5. A **200 OK** response is generated soon after Bob picks the phone up.

6. Bob receives an **ACK** from the Alice, once it gets **200 OK**.

7. At the same time, the session gets established and RTP packets (conversations) start flowing from both ends.

8. After the conversation, any participant (Alice or Bob) can send a **BYE** request to terminate the session.

9. **BYE** reaches directly from Alice to Bob bypassing the proxy server.

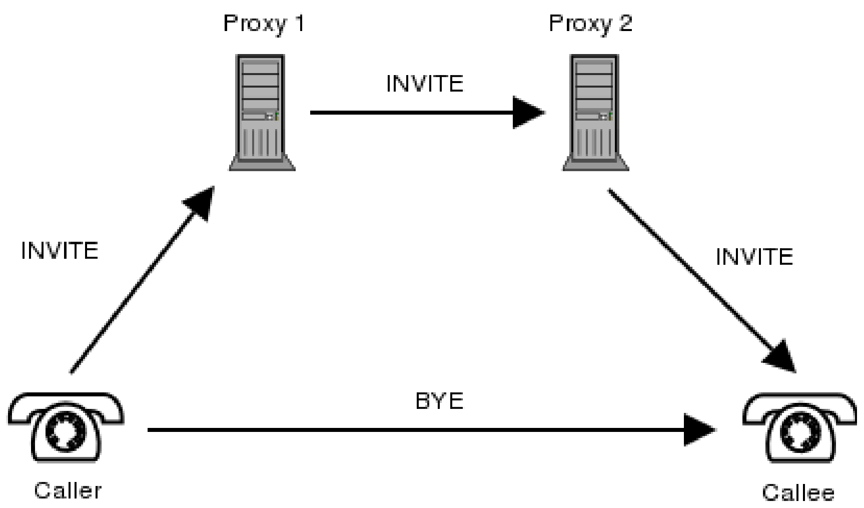
10. Finally, Bob sends a **200 OK** response to confirm the BYE and the session is terminated.

11. In the above basic call flow, three **transactions** are (marked as 1, 2, 3) available.

The complete call (from INVITE to 200 OK) is known as a **Dialog**.

SIP Trapezoid

How does a proxy help to connect one user with another? Let us find out with the help of the following diagram.



The topology shown in the diagram is known as a SIP trapezoid. The process takes place as follows:

1. When a caller initiates a call, an INVITE message is sent to the proxy server. Upon receiving the INVITE, the proxy server attempts to resolve the address of the callee with the help of the DNS server.

2. After getting the next route, caller's proxy server (Proxy 1, also known as outbound proxy server) forwards the INVITE request to the callee's proxy server which acts as an inbound proxy server (Proxy 2) for the callee.

3. The inbound proxy server contacts the location server to get information about the callee's address where the user registered.

4. After getting information from the location server, it forwards the call to its destination.

5. Once the user agents get to know their address, they can bypass the call, i.e., conversations pass directly.

End of ebook preview

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