Session Initiation Protocol
Basics and Working

*Shri S.M.Hafeez Ali, Lecturer Telecom/IRISET*

Voice over IP (VoIP) is an emerging application, as well as a rapidly growing market. Use of the corporate network or the Internet at large to carry telephone traffic has many advantages, not the least economic ones. A successful VoIP network must not only support IP-based telephones, but also provide a means of seamlessly integrating the IPbased network with traditional telephone networks. At the core of VoIP lies the Session Initiation Protocol (SIP) and a few related protocols.Further, as older Private Branch Exchanges (PBXs) and network switches are phased out, industry is moving toward a voice networking model that is SIP signaled, IP based, and packet switched, not only in the wide area but also on the customer premises

The Session Initiation Protocol (SIP), defined in RFC 3261, is an application level signaling protocol for setting up, modifying, and terminating real-time sessions between participants over an IP data network. SIP can support any type of single-media or multi-media session, including teleconferencing.

All in all, any given communication SIP will make use of following components:
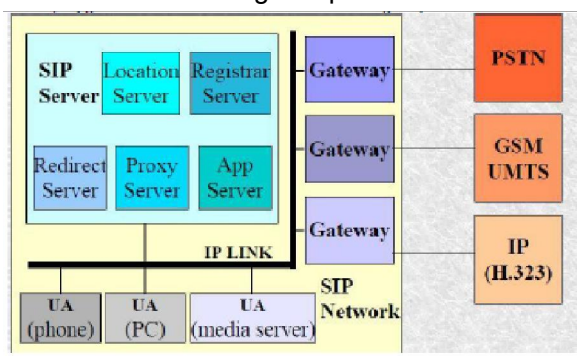
User Agent Client (UAC)

UAC is one side of double sided client components, the second side is known as User Agent Server (UAS). SIP is a text-based protocol with syntax similar to that of HTTP. UAC can be a Soft phone, hard phone or web phone. The UAC is capable of initiating up to six feasible SIP requests to a UAS which are INVITE, ACK, OPTIONS, BYE, CANCEL and REGISTER.

The sending UA must be able to encode audio/video so that the other end can decode it, and vice versa. "SDP" (session description protocol) is the protocol used by the UAs to tell each other what codecs they support.SDP is embedded into the SIP Messages.

SIP response types:

After a SIP request message, the receiver answers with a message. The code of the answer is made up of three digits that allow classify the different types. The first digit defines the answer class:

- Provisional (1xx): The request was received and is being processed.
- Success (2xx): The action was successfully received, understood, and accepted.
- Redirection (3xx): Further action needs to be taken in order to complete the request.
- Client Error (4xx): The request contains bad syntax or cannot be fulfilled at this server.
- Server Error (5xx): The server failed to fulfill an apparently valid request.
- Global Failure (6xx): The request cannot be fulfilled at any server.

User Agent Server (UAS)

UAS is the server that is responsible for hosting applications responsible for receiving one of six SIP requests from UAC. Once the request is received, it is also responsible for sending a responsethat accepts, rejects, or redirects the request back to UAC.

Proxy Server

In most cases proxy server acts as a mediator between UAC's and UAS's. It can also be used for name mapping. Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call).

Redirect Server

The redirect server is used during session initiation to determine the address of the called device. This server allows the redirection, which

in simple term means that enabled users will still be available for contact via same SIP identity, even though they have changed their geographical location.

RegistrarServer

The registrar server accepts REGISTER requests and places the information it receives (the SIP address and associated IP address of the registering device) in those requests into the location service for the domain it handles. It allows the users to change the address in which they were previously contactable.

Location Service

A location service is used by a SIP redirect or proxy server to obtain information about a callee's possible location(s). For this purpose, the location service maintains a database of SIP-address/ IP-address mappings.

The various servers are defined in RFC 3261 as logical devices. They may be implemented as separate servers configured on the Internet or they may be combined into a single application that resides in a physical server.

SIP Addressing

SIP URIs(Universal Resource Indicators) have a format based on e-mail address formats, namely user@domain. The URI may also include a password, port number, and related parameters. If secure transmission is required, "sip:" is replaced by "sips:." In the latter case, SIP messages are transported over TLS(Transport Layer Security) protocol.

SIP Forking

SIP differs from other signaling protocols in that it allows a call request to fork; that is, a server can send out two or more requests to different destinations (branches) based on one incoming request, either at once or in sequence if an earlier request failed. This feature supports a number of advanced telephony services, such as call forwarding to voice mail, automatic call distribution, and user location, where the same number can ring at home and at work, for example.

Phone Lines That Utilize Broadband Connection

SIP Trunks, also known as Business VoIP are virtual "cloud-based" phone lines that utilize a Broadband connection and IP phone system for access.SIP Trunking enables to eliminate the cost of maintaining two networks (POTS + Ethernet) by placing phone traffic on Ethernet network. Additionally, SIP Trunks allow two devices to have several different methods of media/communications streams running, allowing businesses to transmit several voice paths over one channel. This allows for everyone to receive a personal business phone number, but only one phone line is needed. World's best VoIP providers are now capable of providing SIP based VoIP.

Benefits of SIP trunking over E1 trunking

PRI (Primary Rate Interface) is a voice technology that has been widely-used since the 1980s. It is an interface standard used on an ISDN (Integrated Services Digital Network) to deliver multiple lines of voice and data into a business's exiting PBX via one physical line, called a circuit. PRI is a high-capacity service carried on E1 trunk lines between Exchanges. A E1 line carries voice via 32 digital channels.In addition, PRI uses a circuit-switched model for its voice connections between endpoints and has guaranteed quality of service (QoS).PRI is considered "old-school" telephony. It is physical hardware and also requires servicing from a telco for deploying, upgrading and troubleshooting.

SIP is a way to deliver voice via the Internet. SIP is a telephony networking protocol (much like other network protocols such as HTTP and SMTP), therefore it's a network technology rather than a telephone technology like PRI. SIP trunks are virtual; they don't require additional hardware to deploy.SIP trunking can eliminate the need to have a traditional PSTN gateway. This makes SIP trunking easier to install. SIP trunks use a packet-switched networking model that terminates to the service provider via IP and is typically a best-effort delivery with no QoS guarantees. While businesses opt for running SIP trunks directly over the Internet, telecommunications providers prefer to offer dedicated data lines directly between exchanges to ensure the quality and stability of their SIP trunks.

Hybrid Trunking

There are usage cases where Exchanges can mix SIP and PRI. One may be where a business uses PRI for local calling and then uses a SIP trunk from a hosted VoIP service for international calls as a way to save money. Although SIP trunks are increasingly replacing PRI, legacy PBX systems can use SIP trunking by implementing a VoIP gateway.

SIP Security

SIP often runs on top of the User Datagram Protocol (UDP) for performance reasons, and provides its own reliability mechanisms, but may also use TCP.Both signaling and media need to be secured against eavesdropping and alteration. In particular, authentication is important since there is no trusted third party (the phone company) to ensure the accuracy of the information contained in the session setup request. Also, proxy servers may only want to offer services to registered users, and registrations must be protected from malicious alteration. If a secure, encrypted transport mechanism is desired, SIP messages may alternatively be carried over the Transport Layer Security (TLS) protocol.

SIP based VoIP System Configuration

It is clear that SIP has emerged as the standard for VoIP call control and next-generation service creation, supporting interoperability with existing telephony systems and mobility.

FreePBX is a web-based open source GUI (graphical user interface) that controls and manages Asterisk (PBX), an open source communication server. FreePBX is licensed under the GNU General Public License (GPL), an open source license. FreePBX can be installed manually or as part of the pre-configured FreePBX Distro (https://www.freepbx.org) that includes the system OS, Asterisk, FreePBX GUI and assorted dependencies.