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1. Introduction to Session Internet Protocol

People have been inventing methods and techniques of communicating with each other since the beginning of time. The journey towards modern communication began some time ago, but significant progress took place in the late 1960s and early 1970s. Protocol, in general terms is defined as the medium – it is an official code of procedures. Internet Protocol is referred to a system of digital messages and rules set for their exchange. During the 60s and 70s, protocol services were initiated and research was conducted in packet switching (the transfer of communication data between applicants and communicators in the form of groups and packets), but official use of the Internet began in 1982 when Internet Protocol/Transmission Control Protocol (IP/TCP) was standardized, and Internet technology was introduced.

The advent of the Internet reshaped the world. High-level communication was made possible and the exchange of information between people was so convenient that even a person located at the South Pole could contact another person at the North Pole. This was made possible with the introduction of Internet Protocol, connecting two computers/networks through radio waves, without any wire or physical appliance. Communications like mailing, messaging and face to face interactions were boosted and awareness spread all around the world.

Session Internet Protocol is referred to as an IETF (Internet Engineering Task Force) signaling protocol, controlling the exchange and transfer of data and communications through signals. The IETF is classified into different parts to control various programs and technological activities related to its application around the web. Internet Protocol uses data packets, sometimes rendered as datagrams (grouped data that travel through protocols taking the data from one place to another), to send information and communication signals between networks. Video and voice communication among different networks and individual applications across the Internet is made possible by Session Internet Protocol.

2. History, Initiation & Implementation

Internet Protocol/Transmission Control protocol was standardized in 1982, enabling communications all around the world. Initial transmissions were not of voice and video; rather they were mostly text and picture-based. The first meeting of the Internet Engineering Task Force (IETF) was held in January 1986 between 21 U.S. research members who were responsible for its establishment. The first audio-cast of IETF took place in 1992, and after that, its sessions were conducted on Mbone (Multicast Backbone, backbone of IP multicast traffic across the network). After a number of significant achievements, a real milestone was achieved by Henning Schulzrinne and Mark Handley, when in 1996 they designed Session Internet Protocol, allowing video and voice transmissions over Internet Protocol.

On IETF's official website, it is stated that the group's primary objective is to make the Internet work efficiently by producing high-quality technical applications that could reshape the way people use, modify and design Internet and SIP technology. Introduction and integration of Session Internet Protocol technology revolutionized the multimedia universe. Modern techniques took over older communication methods and new trends were associated with Internet Protocol. SIP globally transformed use of the Internet.

Session Internet Protocol (SIP) is an application layer like other IP/TCP applications. It is a text-based protocol like HTTP (Hypertext Transfer Protocol) and SMTP (Simple Mail Transfer Protocol) and requires addresses to go from source to target.

The main purpose of developing Session Internet Protocol was to make the transmission of Voice over Internet Protocol (VoIP) fast and convenient, and to avoid dissipation of data during interactions. SIP technology enabled smooth execution of video chat, video conference, voice calls and voicemail between networks/people through Internet Protocol (Applications of VoIP).

3. Development & Applications

The Internet was initially developed for military operations and communications, but as it became popular among private organizations, people began to express significant interest in the technology. The advent of Session Internet Protocol provided people with a clear path to transmit multimedia across the Internet. SIP's implementation in ordinary networks not only made Voice over Internet Protocol (VoIP) efficient and reliable, but also standardized the concepts of online gaming, song listening, video watching and other multimedia transactions.

SIP technology also introduced IP calls and video conferencing to the market, bringing with it a variety of new applications and appliances to make SIP connections, namely Cisco's routers and switches. Online meetings, voice chat, video conferencing, and multimedia streaming are a hallmark of today's expanding technology, and the Internet has influenced organizations all around the world, compelling them to alter their communication infrastructure for the better.

4. Function & Capability

Internet telephone, Internet telephony, broadband telephony, Internet Protocol (IP) telephony, broadband phone and Voice over Broadband (VoBB) are terms that are collectively known as Voice over Internet Protocol (VoIP). With time and need, VoIP has become commonplace for people all around the planet.

The functioning of video calls and other multimedia transmissions were made possible by the introduction of Session Internet Protocol. SIP is solely responsible for creating and breaking the links of communication and interactions made through VoIP applications. SIP works in the same manner as HTTP, and in order to complete calls across the Internet, the technology requires a complete address to send and receive data packets which contain the communication codes.

The capability of SIP depends on the bandwidth of the technological appliances that are being used. Transmission of VoIP can also be made between networks by using other protocols like Media Gateway Control Protocol (MGCP), Real Time Transport Protocol (RTTP), Session Description Protocol (SDP) and several others; but for ordinary and private use of common sector, Session Internet protocol (SIP) is preferred above all others because it does not consume much storage and requires a very small amount of bandwidth to transmit huge amounts of data across the Internet. VoIP through Session Internet Protocol is also easy to afford which makes it more favorable among consumers.

5. SIP Clients & Servers

Session Internet Protocol relies on a User Agent (UA) to receive and send signals across networks and coordinate sessions. UA is a logical application, controlling the end points of SIP sessions. The main purpose of the technology is to act like both a User Agent Client (UAC), whose job is to send requests for establishing calls and communications paths, and like a User Agent Server (UAS), whose job is to accept/receive requests and then respond to them. UAC and UAS last only as long as it takes to connect the call. The moment the call is established, their function is terminated.

According to the infrastructure of Session Internet Protocol, its networks are defined under Uniform Source Identifier (URI), where URI is a string of characters that identify a name or resource in a network. The standard form of SIP URI is sip:username:password@host:port, where for secure transactions, "sips:" URI scheme is used instead of the standard "sip:". Although two end-points in SIP are capable of handling a call without causing an interruption in communication, servers are preferred for classified and high-level interactions.

5.1. Proxy Server

A proxy server is implemented in almost every SIP session. Its major characteristics include secure operation, reliability and strong connections. A proxy server acts like a User Agent Client (UAC), sending requests to the clients to make or break calls. It ensures that requests are sent accurately and appropriately to the right targeted server. The advantage of integrating a proxy server into your network is that it can supersede network policies and allow or deny a network connection. The policies controlled by the proxy server are set forth by the authorized users of the network.

5.2. Registrar Server

Session Internet Protocol also requires a registrar server for proper operation. These servers are frequently found alongside proxy servers. However, to ensure a more efficient network, they should be located with redirect servers. The purpose of a registrar server is to accept "REGISTER" requests. Among the types of requests that this server accepts are those for the registration of IP addresses. The information is linked to the domain name that registers IP addresses and this domain name is handled by registrar.

5.3. Redirect Server

Redirect servers are User Agent Servers that generate (redirection) responses for/to the requests received. They enable the establishment of a direct link with external Uniform Source Identifiers (URIs). Redirect servers also enable proxy servers to present and direct SIP session invitations to external domains.

6. Protocols

Voice over Internet Protocol allows cost-effective and reliable transmission of multimedia technology, and voice and video calls across the Internet. VoIP can be transmitted across networks through different protocols.

Following are the different protocols utilized for VoIP:

MGCP	Media Gateway Control Protocol
SGCP	Simple Gateway Control Protocol
RVP over IP	Remote Voice Protocol over IP Specification
SAPv2	Session Announcement Protocol
SDP	Session Description Protocol
SIP	Session Internet Protocol
Megaco H.248	Gateway Control Protocol

For ordinary communications and transmissions, SIP is preferred because it provides its own mechanism for setting up calls and tearing down links. It can create, modify, serve and terminate sessions between any number of participants.

The best feature of SIP is that it can also be implemented with other call service protocols. It supports five reasons to establish and terminate calls. Those are the user's capability, user's presence, user's location, call establishment (call handling), and call termination.

7. Reliability

Session Internet Protocol can create reliable connections:

1. Phone to phone.
2. Mobile to phone.
3. Mobile to Mobile.
4. Computer to Mobile/Phone.
5. Other handsets connectivity to each other.

These connections can be established through the modern technologies of Wi-Fi broadband and wireless Internet, allowing connectivity to other VoIP devices through SIP.

SIP provides a secure connection, and like HTTP, it requires a complete address to make the connection with the targeted device. It also ensures quality service and smooth communication among users.

8. Conclusions

SIP identifies callers and caller ID, and maintains the quality and reliability of Voice over Internet Protocol's communications. VoIP connectivity through SIP is favored for private and ordinary communication over other services. As SIP's messages can be transmitted over either Transmission Control Protocol (TCP) or User Datagram Protocol (UDP), it is mostly suitable for VoIP links through portable devices.

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