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Overview of VOIP communication over Satellite Network

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***Abstract*— Voice Over IP (VoIP) that involves the delivery of voice, data to the end user. The performance metrics of VoIP are signaling, bandwidth, delay, jitter and packet loss. Voice and data convergence is happening not just in terrestrial communication links, but also in satellite links. In satellite networks, VoIP performance is affected by mainly three factors - long delay, limited bandwidth and channel error. Both call setup time and quality of service (QoS) for voice calls are affected. The VoIP uses Signaling Protocol like Session Initiation Protocol (SIP) and H.323 and Transport Protocol like Real Time Protocol (RTP). The SIP is used for creating, modifying and terminating sessions with one or more participants. Now a day SIP is widely used for voice and video communication. In this paper different performance metrics of VoIP, which affects the performance of VoIP over the satellite network is defined.**

Index Terms— VoIP, SIP, Satellite Network

I. INTRODUCTION

Now a day multimedia applications are becoming a fundamental part of Internet. Different multimedia applications are VoIP, video streaming, multi-player games. VoIP is one of the most important applications from the traditional telecommunication networks to the Internet. VoIP services have been emerged as a low-cost alternative to Public Switched Telephone Network (PSTN) voice service. It provides an attractive solution for voice/data integration in public and private networks. Voice and data Convergence is happening in terrestrial communication links as well as in satellite links. The satellite links already have capacities to carry data packets. With global coverage and reach to remote areas, satellites are well positioned to enable growth of VoIP services. The satellite channels suffer from some shortcomings that affect end-to-end communications such as high Bit Error Rate (BER), long delays, channel error, limited bandwidth which present a challenge for providing a good quality IP telephony service over satellite systems.

Voice over IP has two architectures. One is **H.323** which has roots in the PSTN and the other is **SIP** which is Internet-based. Now a day the H.323 IP telephony has increasingly been replaced by SIP protocol because SIP is simpler than H.323 in developing and supporting software. H.323 defines a set of standards for the transmission of packet multimedia data over networks. **SIP** is an application layer protocol for OSI model, which describes a method for establishing and terminating user session, including multimedia content exchange like video and audio conferencing, instant messaging, online games. The paper is organized as follows. Section II provides the information about the VoIP and its protocols. In section III SIP is defined. Different performance metrics of VoIP is defined in section IV.

II. VOICE OVER INTERNET PROTOCOL

In data transmission and voice transmission: data are loss-sensitive and delay tolerant, while voice is loss-tolerant and delay sensitive. This is the major difference between data transmission and voice transmission over packet-based networks. Due to this reason, the transport layer in the VoIP protocol stack uses the User Datagram Protocol (UDP) to carry voice instead of the Transmission Control Protocol (TCP). TCP is used to carry signalling messages such as call setup and teardown. UDP is a connectionless protocol that offers nonguaranteed datagram delivery between end hosts. UDP gives applications direct access to the datagram service of the IP layer. An application running over UDP must provide their own mechanisms to deal with retransmission for reliable delivery, packetization and reassembly. Current multimedia applications use UDP as the underlying transport protocol. UDP is usually chosen in preference to TCP [2], because:

• **Start up Delay:** The three-way handshake before initiating data transfer induces a delay. UDP avoids this delay.

• **Statelessness:** TCP holds connection state. This increases the risks of potential state holding attacks. UDP avoids holding connection state.

• **Trading Reliability against Timing:** Multimedia data is timely, if it is not delivered by some deadline typically a small number of Round Trip Times (RTTs), the data will not be useful at the receiver side. TCP can introduce an arbitrary delay because of its reliability and in-order delivery requirements, making it unsuitable for real-time media.

Most VoIP applications are real-time; the Real-Time Transport Protocol (RTP) is run on top of UDP to provide end-to-end delivery services for data with real time characteristics. These services include payload type identification, sequence numbering, time stamping, and delivery monitoring.

The main VoIP standard is International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) recommendation H.323. It was designed for multimedia communications systems and is considered bulky for simpler voice applications. H.323 is by far the most widely supported protocol suit in VoIP platforms as it was the first VoIP standard that became available. H.323 specifies the technical requirements for multimedia communications systems in packet based networks such as local area networks, enterprise area networks, metropolitan area networks, intranets, and internets. The SIP was developed within the Internet Engineering Task Force (IETF) as an alternative protocol offering less complexity and more flexibility. Both H.323 and SIP are peer-to-peer protocols. Taking another approach to controlling telephony gateways, the Media Gateway Control Protocol (MGCP) assumes a call control architecture. The call control intelligence is outside the gateways and handled by external call control elements. Hence it is a master/slave protocol. The MGCP is a combination of two earlier protocols, the Simple Gateway Control Protocol (SGCP) and the Internet Protocol Device Control (IPDC), which were introduced by Telcordia and Level 3, respectively. All next-generation VoIP gateways support at least one of the newer protocols: SIP, SGCP, IPDC or MGCP [2]. Here we concentrate on SIP protocol of VoIP.

III. SESSIOM INITIATION PROTOCOL

The SIP (RFC 2543) is a signaling protocol for creating, modifying, and terminating sessions with one or more participants. The main objects of SIP are: User Agent, Proxy Server, Redirect Server, Location Server and Registrar Server. The SIP client called User Agent Client (UAC) and SIP server called User Agent Server (UAS) as shown in Fig.1. While device sends the SIP request which called SIP Client; while device receives the SIP request which called SIP Server. The exchange of request and response messages among the user agents is done through one or more SIP servers. These request messages are known as methods. There are six methods; INVITE, REGISTER, BYE, ACK, CANCEL and OPTIONS. The response messages are 100Trying, 180 Ringing, 200 OK and many others.

Fig.1 shows the typical SIP message exchange to establish a session. To set up a SIP session, the UAC sends an INVITE request to the UAS. Each server on the path confirms the reception of the request by returning a 100 Trying response to the previous hop. Instead of forwarding a request, a SIP server can reject it if it is unable to forward the request. Once the request is received by the UAS, it typically responds with a 180 Ringing response to indicate that the called user is being alerted and a 200 OK response when the user has accepted the session. After the 200 OK is received by the UAC, it sends an ACK request to complete the three way handshake of an INVITE transaction. The INVITE request is the only SIP request that uses a three way handshake. Sessions can be terminated at any time by sending a BYE request, which is confirmed with a 200 OK response[13].

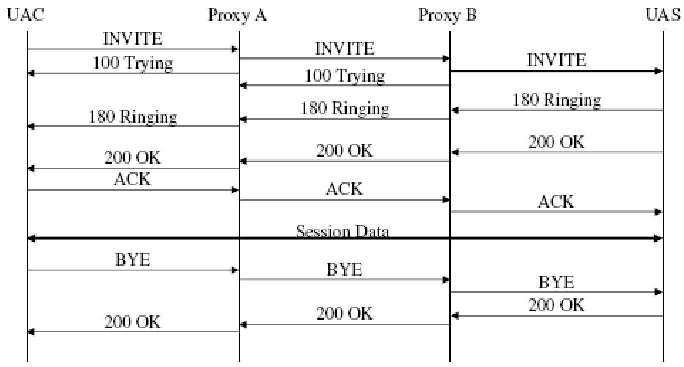

Fig.1 SIP call flow

The role of the **Proxy Server** is as same as the proxy of Web Service. Proxy Server is responsible to handle requests or forward requests to other servers. SIP messages will send to the final destination after be passed through one or more Proxy Server. **Registrar server** offers registration services and it updates its database as a new user arrives. Registrar Server is responsible for accepting the SIP request. It could indicate the user is at a particular address and support the personal mobility. The format presents as same as Email such as User@host. Users can send the SIP REGISTER request to tell the Registrar new position. Registrar combined with a proxy or redirect server to achieve user mobility [10]. **Location server** keeps track of the location of the users. It is updated by the registrar server on new registration. The main function of **Redirect Server** is mapping the destination address to zero or more new addresses. Redirect server forwards the request to possible proxy servers or user agent if the requested URI is not in its database. The most important difference between Proxy Server and Redirect Server is Redirect Server does not help to forward the request. SDP (Session Description Protocol, RFC 2327) was published by IETF in August 1998. SDP [9] is the protocol used to describe multimedia session announcement, multimedia session invitation and other forms of multimedia session initiation. A multimedia session is defined, for these purposes, as a set of media streams that exist for the duration of time. The main idea for developing SDP is to transmit the information about the multimedia session so that the receiver can join into this session.

IV. CHARACTERISTICS OF VOICE OVER INTRNET PROTOCOL

A. Speech Coding

Speech coding is used to compress voice signal for transmission over long distance. It involves the process of transforming the analog signal into digital signal. Sending the digital data to the far end and regenerates the voice at the far end. Various speech codecs are being used in PSTN and Internet. G.711 [10] is the most common codec of PSTN. It is also known as pulse code modulation (PCM). It operates at 64kbps. It uses two compression algorithms: μ-law in North America and Japan and a-law in rest of the world.

Table 1 Voice Codecs and bandwidth consumption

Voice codec	Codec bit rate (kbps)	Voice payload size (bytes)	Bandwidth (kbps)
PCMA	64	240	74.7
G.721	32	80	48
GSM	13.2	33	29.2
LPC	2.5	7	16.8

Table 1 contains different voice codecs and its bandwidth consumption. G.721 [11] produces a data rate of 32 kbps. GSM is the codec from mobile telecommunication domain. It operates at 13.2 kbps. It has excellent performance regarding the CPU demand. Linear predictive codec (LPC) [13] is an experimental codec that operates at 2.5 kbps. Different codecs are used to generate voice streams with different encoding schemes. Voice over IP uses RTP to carry voice packets. RTP uses sequence numbers and time stamps to identify out of order packets. RTP is encapsulated in UDP which is an unreliable transport layer protocol. So there is no guarantee of arrival of voice packets at the destination. If reliability has to be incorporated, it can be implemented in the application generating the voice packets. The voice payload with various headers is shown in Fig.2 there is an extra overhead of 40 bytes with each voice packet.

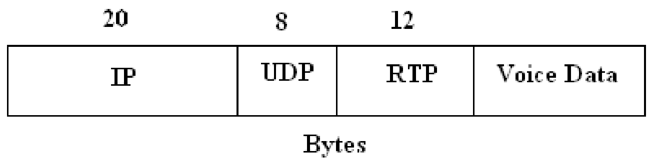

Fig.2 IPv4/IPv6 voice packet

B. Performance metrics

Following are the main performance metrics of voice over IP:

Bandwidth required depends on the voice codec and its algorithmic complexity. The IP bandwidth consumed by a voice call can be computed by the following formula [15].

Packet size = IP / UDP / RTP header + Voice payload size

PPS = Codec bit rate / Voice payload size (1)

Bandwidth = Packet size * PPS

Where PPS is the number of packets needed per second to deliver the codec rate. The IP / UDP / RTP header is fixed and its length is 40/60 bytes. The detail of the codecs and their bandwidth consumption is given in Table I. The bandwidth by each call is computed using (1).

Delay is the one way delay between the source and destination. In geostationary (GEO) satellite systems, this delay is dominated by the propagation delay which is approximately 250-270ms. VoIP is a real-time application, which cannot tolerate longer delays as the users will loose interactivity. According to ITU-T recommendations [16][17], one-way delay follows these constraints:

- Under 150 ms: acceptable
- 150 to 400 ms: acceptable with limitations
- Over 400 ms: unacceptable

Voice packets are transmitted by RTP. RTP identifies a voice stream by its unique Synchronization Source Identifier (SSRC). Additionally, individual packets can be identified by the port numbers, sequence numbers and timestamps.

Jitter is the variation in delay of the successive voice packets. Jitter occurs because different packets suffer different delays in the network. To rectify, this problem, jitter buffers are used at the receiver. First of all, enough packets are stored in the buffer. When sufficient amount of packets are accumulated, then they are played out. Jitter contributes in the overall delay of the voice packets. Jitter is an estimate of the inter arrival time of RTP packets and that’s why it’s referred as the inter arrival jitter [18]. If R represents the arrival time of a packet and S represents the RTP timestamp, then the inter arrival difference D (i,j) between two packets i, and j, can be calculated as,

$$D(i,j) = (R_j - R_i) - (S_j - S_i) = (R_j - S_j) - (R_i - S_i) \quad (2)$$

Packet loss is also a factor in degrading the voice quality. It is intolerable in time constrained applications like VoIP. Packet loss is devastating because voice packets are carried by UDP which is an unreliable and does not guarantee retransmission of lost packets. Packet loss is due to congestion, interference, noise and buffer overflow at the receiver. A packet arriving after a certain scheduled play out time is also discarded. Packet loss can be reduced using forward error correction (FEC) by transmitting redundant information and interleaving the packets. A packet loss up to 10% is acceptable in VoIP [17].

V. CONCLUSION

In this paper I surveyed the evaluation of the SIP signaling and QoS for VoIP over satellite networks. The performance metrics of VoIP are signaling, bandwidth, delay, jitter and packet loss are defined here. In satellite networks, VoIP performance is affected by mainly three factors - long delays , limited bandwidth & channel error. Originally VoIP standards are designed for terrestrial link which may not give optimum performance when we apply over satellite network as it is. So we need to customize or modify the existing protocols to suits the channel characteristics (long delay, limited bandwidth, channel error) of satellite network. This information will help to develop future system for satellite based disaster management project and will try to overcome the problems of VoIP over the satellite network.

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