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Primordial Abstraction of Voice over Internet Protocols (VoIP)

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Abstract— This paper include the study of voice over internet protocol, its architecture that how VoIP works , issues that occur in voice over internet protocol , VoIP codecs and many more .In the era of wireless communication technologies, the actual time services which includes data as voice (i.e Voice Over Internet Protocol) is growing stupendously. Different applications of VoIP which includes SKYPE, GOOGLE TALK, etc. are used to provide economical voice calls to the end users. VoIP is technology used for communication so as to transmit voice and data over internet protocols. VoIP is used for IP telephony services which includes voice messaging ,calling, video messaging as well as video conferencing etc.

Keywords— VoIP, VoIP Architecture, Quality of service and Measurement methods , Codecs

I. INTRODUCTION (VOICE OVER INTERNET PROTOCOL)

There is a huge increase in the use of VoIP and Internet within these recent years. A new way of communicating from one place to another place can be done through internet which is called as VoIP technology [1]. This technology allows to make call over an Internet Protocol (IP) network. VoIP is a form of communication over broadband Internet connection to make phone calls. VoIP allows to convert voice signal into digital one and to send these signals over the internet. Hence VoIP is cost efficient and provide flexibility, portable etc. and allow to provide voice and video conferencing services. The main reason why we use VoIP is because of its cost effectiveness. We can also say that VoIP is basically a digitized voice traffic that is to be transmitted over the network to make calls [2]. VoIP differs from circuit switch network because data gets splits up into packets and these packets are being sent over the network. The circuit with splits data into packets are known as packet switching circuits. With the packet switching, data is divided into packets and these packets of data are sent over the network where there is no need of reserving the entire duration of call between the sender and receiver. The main advantage of using VoIP is multiple routing traffic is cheaper and the between routers and switches the traffic flows free of cost. Digital data is form of packets which may includes voice, video, fax, music and telephony. There is increase in the usage of VoIP because of low cost and optimized functionality. As people used to pay lot of charges for calling but with the use of VoIP these charges can be reduced, as call can be done by using computer through speakers and microphones etc. And by using internet service call can be used anywhere and anytime, therefore the outsource companies can may their job easy by using VoIP. VoIP can be do its work under both single as well as multiple environment. The routing of VoIP can be analyzed into two ways [3] i.e. by performance and stability. VoIP provide various services from PC-to-PC, PC-to-Phone, Phone-to-Phone. Also the protocols that VoIP include is H.323 protocol, SIP protocol, Media Gateway Control Protocol. This paper includes how does VoIP Works in Section-II, Quality of service in VoIP system in Section-III, Quality of service measurement methods in Section-IV, VoIP codecs in Section-V and paper is concluded in Section VI.

II. HOW DOES VOIP WORK

Internet protocol is used for the transmission of voice as packet over IP network. The voice signal is converted into digital signal and the process of digitization of voice segregates the undesired noise signal and then by using compression codec or algorithm the compression of voice signal is done.

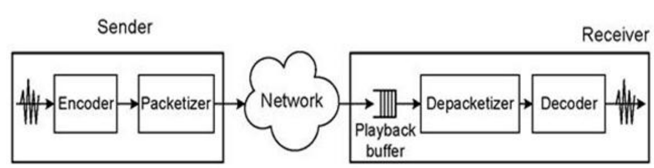

Figure 1: Basic Model of VoIP

The data as voice is divided into packets after compression and these packets of voice is sent over the IP network. Each and every packet requires the destination address, sequence number and data for error checking. To obtain these requirement along with other call management requirements, signalling protocol are added. At the destination, whenever the packets gets arrive, the sequence number tells that packet is reached at the destination and thus packet must be placed in order and hence to recover the data from these packet , decompression algorithm must be applied. For ensuring the proper spacing, here synchronisation and delays management need to be taken care. To store the packet arriving out of order through different routes and to wait when packet is arriving late, buffer is used .

III. VoIP QUALITY OF SERVICE

In order to determine the voice quality of VoIP service important parameters are used i.e. Quality of service. QoS is used to deliver the network service in proper manner at some desired level [9]. Hence the parameters involved in VoIP are– Bandwidth, Securing, Link abortion, Loss of Packet, Network design, Noise, Packet Delay, Echo, Throughput etc.

A. BANDWIDTH ASSIGNMENT

In VoIP system, assigning of bandwidth is an important factor. For voice communication, it is necessary for a network to assign a bandwidth when it cleave off between computer data and voice.

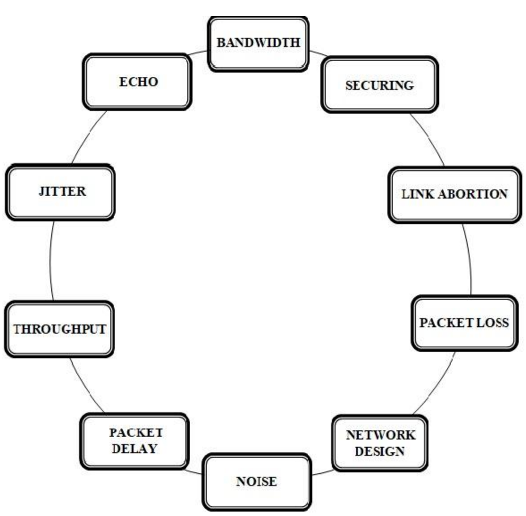

Figure 2: Quality of Service Parameters.

B. SECURING

As to avoid overhead and blockage, securing is also one of the important factor so as to secure the transmission of voice from one place to another.

C. LINK ABORTION

Link abortion can occur due to many reasons such as due to the problem in the equipment, detachment and unplugging of cable wire, some change in the network configuration. Basically link abortion leads to the loss of packet series, which may continue for sometime after restoring the link.

D. LOSS OF PACKET

The main reason of packet loss is the bandwidth limitation and way how packet move in the network i.e. from source to destination. As when IP packets are introduced in the network, some of the packet may get lost which results in the dumbness in the conversation which is not acceptable thing by the user.

E. DESIGNING OF NETWORK

The designing of the network should be done properly. If the network design is unable to maintain the voice transmission and data packet then reliability must get desolate. All the modernization of voice application must be supported by the designing of network .

F. NOISE

Noise is also one of the important factor in VoIP. The voice communication is sensitive to noise. The reason why signal reaches at destination with lead and lag is noise only, which also downgrade the voice quality. So as to measure the voice quality a score is used which is called as Mean Opinion Score (MOS).VoIP system has the voice quality range between 3.5 to 4.2.

Quality Scale	Score	Listening effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with reasonable effort

Figure 3: Mean Opinion Score (MOS) [4]

G. PACKET DELAY

Delay is the total time taken in communicating from one person to another person, spoken words and hearing the voice at the receiver end [11]. Delay occurs in VoIP when the voice data packet at the transmitter take more time then the expected time to reach at the destination. Delays are called as latency which may cause some disturbance in the voice quality.

H. ECHO

Echo is actually the replication of original sound. In VoIP, Echo means when at sender side the caller may listen his/her own voice after speaking bon the phone/microphone but callee dos not notice that .

I. THROUGHPUT

It is actually a measure of how much information a system can be processed in given amount of time .

J. JITTER

Jitter may occur when there is not guarantee of delivering the packet at the other end and which may leads to some variations in the transmission delay and leads to some negative effect.

IV. QOS MEASUREMENTS OF VOIP

There are two kinds of method by which we can measure the voice or speech quality[14] . These methods can be subjective methods and objective methods. Hence the two methods are discussed below:-

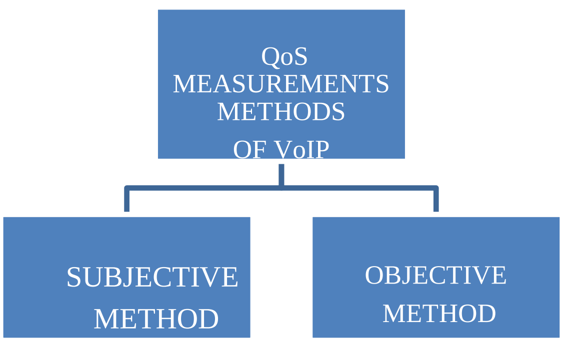

Figure 4: Quality of Service Measurement

A. SUBJECTIVE METHOD

The aim of this method is to provide the limited response choice and to find the average user's perception of system's speech quality, subjective method is used. For the subjective determination of voice quality MOS on user perception based which ranges from 1(poor) to 5(excellent) is introduced by International telecommunication union-Telecommunication in recommendation P.800.

B. OBJECTIVE METHOD

The method which leads to mathematical measurement of physical quantity of the system is known as objective method. This method is used to measure the delay and packet loss. The measurement can be done by two techniques which can be Intrusive and nonintrusive. Intrusive techniques can be applied on the developing technique and non intrusive technique can be applied on the real time traffic.

V. VARIOUS CODECS OF VOIP

Codecs are basically used for coding and decoding purposes, which may converts analog signal into digital signal form for transmission and then back to original signal i.e. uncompressed audio signal for replay. The underlying are the some VoIP codecs with their configuration.

CODECS	BITRATE (Kbps)	SAMPLE SIZE(bytes)	SAMPLE RATE (ms)	VOICE PAYLOAD (bytes)	VOICE PAYLOAD SIZE(ms)	PACKET PER SECOND(PPS)	BANDWIDTH	ALGORITHM	MOS	APPLICATION
G.711	64 kbps	80 bytes	10 ms	160 bytes	20ms	50	87.2 kbps	a-law & u-law	4.2	ISDN
G.722	64 kbps	80 bytes	10 ms	160 bytes	20 ms	50	31.2 kbps	SB-ADPCM	3.7	VoIP
G.723.1	5.3 to 6.3 kbps	20to 24 bytes	30 ms	20 to 24 bytes	3 ms	33.3	20.8 to 21.9 kbps	ACELP/ MP-MLQ	3.65 TO 3.9	VoIP
G.726	24 to 32 kbps	15 to 20 bytes	5 ms	60 to 80 bytes	20 ms	50	47.2 to 55.2 kbps	ADPCM	3.85	VoIP
G.728	16 kbps	10 bytes	5 ms	60 bytes	30 ms	33.3	31.5kbps	LD-CELP	3.61	PSTN
G.729	8 kbps	10 bytes	10 ms	20 bytes	20 ms	50	87.2 kbps	CS-ACELP	NA	VoIP

Figure 4: VoIP Codecs

VI. CONCLUSION

This paper discusses that how today the multimedia application based on internet such as voice plays an important role because of its reliability, low cost and flexibility etc . And, how in VoIP the signal gets converted into digital as the form of packets and sometime these packet may lost, there may occur some delays in receiving the packet at the destination which may leads to poor voice quality and that poor quality of voice may be improved by using some methods and by using Score known as Mean opinion score (MOS). Also this paper discusses some different VoIP codecs which consist of different applications and uses.

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