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Performance Progress in QoS Mechanism in Voice over Internet Protocol System

Harjit Pal Singh* & Sarabjeet Singh
Department of Physics,
Dr B R Ambedkar National Institute of Technology,
Jalandhar, India
harjit_nit@yahoo.co.in

Jasvir Singh
Department of Electronics Technology,
Guru Nanak Dev University,
Amritsar, India

Abstract: Voice over Internet Protocol (VoIP) is one of the most advancing technologies used to make calls through the internet. In Voice over Internet Protocol (VoIP) system, the speech signal is degraded when passed through the network layers. The speech signal is processed through the best effort policy based Internet Protocol (IP) network, which leads to the network degradations including delay, packet loss and jitter. The researchers have made enormous efforts to resolve the QoS issues to make VoIP best alternative to the traditional public switched telephone network. The work in this paper summarized the performance progress in quality of service (QoS) mechanism for enabling the VoIP system to provide the toll quality service to the users.

Keywords: VoIP, QoS, Signal Processing

I. INTRODUCTION

Voice over Internet Protocol (VoIP) is one of the most advancing technologies in the World of communication. VoIP is a way to make calls through the internet and VoIP system transmits the voice packet via internet protocol network. The traditional public switched telephone network (PSTN) is a circuit-switched network which requires a dedicated line for telecommunications activity and the most of the bandwidth is wasted [1]. For transmitting the voice packets over IP network is the best alternative to the traditional voice communication system. Since, Internet was initially designed for transmitting data traffic and it is performing this task really well. However, Internet is best-effort network and therefore it is not sufficient enough for the transmission of real-time traffic such as voice [2].VoIP has gained popularity due to the more advantages it can offer than PSTN systems especially that voice is transmitted in digital form which enables VoIP to provide more features and the most favorable benefit of the VoIP system is that it costs as much as half the traditional PSTN system in the field of voice transmission and this is because of the efficient use of the bandwidth requiring fewer long distance trucks between switches [3].

II. VoIP SYSTEM

VoIP involves digitization of voice streams and transmitting the digital voice as packets over the IP based packet networks like the internet. The Fig.1 shows the basic steps involved in the VoIP system. The analog speech signal is sampled and quantized to convert it into digital signal. Then echo cancellation and voice activity detection algorithms are implemented on the digitized signal. The signal is compressed using various ITU-T codecs such as G.711 [4], G.729 [5] and G.723.1 [6]. The codec is selected on the basis of the various factors such as bandwidth availability and quality of signal. The encoded speech is then packetized and each packet includes the headers at the various protocol layers such as RTP 12 bytes [7], UDP 8 bytes, IP 20 bytes and the payload composed of the encoded speech for a certain duration [8], depends on the codec used.

Then speech packets are routed to the destination through the internet. In addition, RTP and Real-Time Control Protocol (RTCP) were designed at the application layer to support real-time applications. The IP networks used the transmission control protocol (TCP) for transmission of the packet, since it guarantees the delivery of the packets at the receiver side. But for the delay sensitive voice communications, the TCP protocol is not preferred since it introduces the delay into the system by using acknowledge (ACK) scheme. For transmitting the voice packets, the user datagram protocol (UDP) is preferred in the VoIP communications. The voice packets are then sent over the IP network and the reverse process is carried out at the receiver side. The voice packets are depacketized and then decoded into the analog voice signal.

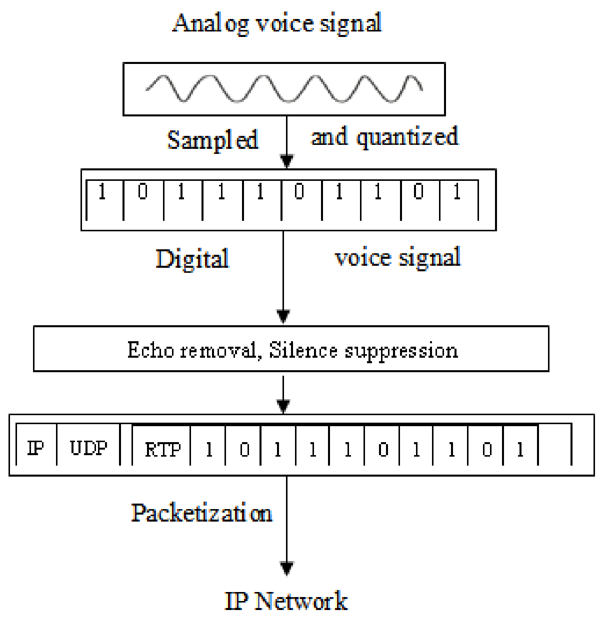

Figure 1 Voice Packet Preparation

III. QUALITY ISSUES IN VOIP

The speech signal when processed through the VoIP system, the signal is degraded with different impairment factors such as delay, jitter and packet loss. Delay can be defined as the total time it takes since a person, communicating another person, speaks words and hearing them at the other end. Unlike data applications, VoIP applications are very sensitive to delay although they can tolerate packet loss to some extent. End-to-end or mouth to ear delay is one of the main factors affecting QoS and should be less than 150 ms for good network connection as defined by ITU G.114 [9]. Delay is mainly caused by network congestion which leads to a slow delivery of packets [10].

IP network does not guarantee of packets delivery time which introduces variation in transmission delay. This variation is known as jitter [11] and it has more negative effects on voice quality. Since voice packets of the same flow are not received at the same time. Therefore, jitter buffer are introduced to diminish the jitter effect and make the conversation smoothly as it holds a number of packets in a queue before playout. The buffer queue size can be fixed or adaptive which varies based on network condition, voice character for better performance. Jitter buffer adaptive techniques perform better as it reduces the possibility of buffer overflow and underflow. Overflow issue is where number of packets received is getting larger than the buffer size as a result buffer discards packets that cannot hold. On the other hand, underflow buffer occurs when some packets are needed for playout but buffer is empty [12].

Sometimes due to the delay/jitter, packets are transmitted late and the load in queuing buffer increases, resulting in the discard of packets. Due to the loss of packets, conversation becomes irritating and discontinuous. In non-real time application, like data transfer, this packet loss problem can be solved by using retransmission mechanism of TCP. But with retransmission method, the lost packet will arrive very late at the far end which interrupts the voice communication. Burst packet loss affects adversely the communication. A Burst consists of consecutive lost packets. Some times to save bandwidth, multiple speech frames are transmitted in single packet. But if packet loss occurs, then this single packet lost leads to loss of multiple frames. The listening quality in this case is almost similar to that in the case of burst packet loss [13]. VoIP system can tolerate packet loss to some extend as 1% or less is acceptable for roll quality while for business quality 3% or less is acceptable [14]. Hence, more than 3% of packet loss degrades the speech quality.

IV. PERFORMANCE PROGRESS IN VOIP

The first successful VoIP connection was made by Informational Sciences Institute (University of Southern California) and Lincoln's Laboratory (Massachusetts Institute of Technology) in the late 1974 [15]. Tanaka et.al [16] designed and simulated packet voiced transmission system on 11/34 minicomputer along with a DR-11 digital input/output interface. The quality of speech was evaluated using subjective methods and suggested that the high speed of the network can solve delay problems in the system.

Jayant et.al [17] analyzed the effect of packet loss on the waveform coded signal. The improvement in signal quality had been proposed due to an odd-even sample-interpolation scheme but at the cost of increasing decoding delay and also proposed an optimal packet length in range of 16-32 ms. Mackie et.al [18] evaluated the effect of packet length and packet loss rate on the voice quality of packet voice system and the packet voice system was simulated through a SLAM network model to find queue size requirement as a function of packet size, delay distribution and expiring timeout. Goodman et.al [19] proposed the technique of reconstruction of missing voice packets in which the missing packets could be replaced with waveform segment from the correctly received packet in order to increase the maximum tolerable missing packet rate. Bolot [20,21,22] investigated effect of loss and delay of audio packets on the internet and the work was largely theoretical but supported by experimental evidence and critically proposed the use of techniques such as redundancy protection against packet loss.

The commercialization of VoIP grew up in later 1995, when Israeli company Vocaltec, developed and released first internet phone software called "Internet Phone". Internet phone used a sound card, microphone and speaker to send voice packets over a modem. Since Internet telephony seemed to be recognized as Commercial product, then researchers again tried to find new solutions to improve quality of service. Moon et.al [23] analyzed the problem of adaptively adjusting the playout delay in order to keep this delay as small as possible to avoid excessive losses and proposed the adaptive delay adjustment algorithm that tracked the network delay of recently received packets and efficiently maintained delay percentile information. Moreno et.al [24] proposed the packet error prone front end recognition scheme for VoIP system. The scheme found to be effective to packet loss since it was not constrained to error handling mechanism of the codec and was able to extract recognition feature vectors directly from ~~eh~~ encoded speech instead of decoding the signal. Boutremans et.al [25] analyzed the effect of adaptive joint playout buffer and forward error correction (FEC) scheme in the VoIP system and compared the play-first and play-best strategies.

The authors proposed that the delay aware play first strategy to be beneficial for VoIP application because of its simplicity and joint effect of the playout buffer. Agnihotri et.al [26] proposed the receiver based Global Local Search-Time Scale Modification (GLS-TSM) for speech quality improvement in Voice over IP system. The performance of this less complex scheme was limited to silence, noise substitution and packet repetition but from the evaluation results, it was observed that the GLS-TSM scheme could be used in the practical applications for improving signal quality in VoIP systems.

Ho et.al [27] implemented and verified that the system on chip (SOC) platform embedding a low-power SOC processor is a feasible and beneficial solution to achieve all fundamental goals of standalone IP phone with less power consumption. This low-power system used TMS320C54x DSP core and ARM7TDMI MCU core. DSP software provides PCM generation, tone generation, AEC, voice activity detection, voice playout and codec operations. DSP converts the voice processing into streams of packets and passes the packets to MCU for processing and transmission to packet network and reverse process on other side. MCU along with third-party software components capable of providing call channel features and networking with standard IP protocols. Mase [28] summarized end-to-end measurement based admission control (EMBAC) and analyzed for VoIP networks. The performance of the proposed admission control algorithm was close to the ideal method of virtual trunk based admission control.

The reliability issue was discussed in [29]. Casetti et.al [30] presented a framework that assumed variable rate speech coders at rates of 64 kbps, 13 kbps and 8 kbps and their rates were determined by an end-to-end control mechanism, based on measurements of packet delay and loss rates. Dong et.al [31] proposed selective error checking at MAC layer of 802.11 and used the fact that speech bits can tolerate errors, but should be protected for optimal quality reproduction and the simulation results showed that the speech quality can be substantially improved by modifying MAC layer with SEC to suit the narrow band AMR. Choi et.al [32] analyzed the scheduling algorithms for VoIP system according to the traffic behavior. The frame structure was divided in two parts: the first one got more priority and second part was distributed to normal data without any priority. The authors compared the hard and soft decisions with maximum rate algorithm and proportionally fair algorithm and observed that the soft proportionally fair algorithm worked well but with a limitation that as the traffic became high, the drop probability was increased. Sun et.al [33] proposed the non-linear regression models for perceived voice quality prediction. The authors derived non-linear regression models for various speech coders with aid of PESQ & E-model and proposed the use of minimum overall impairments as a criterion for buffer optimizations. The delay characteristics were derived with Weibull distribution and results predicted the optimum perceived voice quality in comparison to the other algorithms.

Narbutt et.al [34] performed an assessment for different audio codecs using adaptive playout buffer technique for voice over WLAN networks. The proposed algorithm adapted to varying network conditions and the estimation gain factor was updated with each incoming packet according to observed delay variations. The comparison was performed with different existing playout algorithms such as Ramjee [35] and Moon's [23] algorithms and the proposed algorithm resulted well. Filho et.al [36] proposed adaptive mechanism for FEC and analyzed the scheme for video-conferencing in VoIP system. The proposed scheme concealed more packets and controlled redundancy but increased the bandwidth requirements. Bai et.al [37] proposed a perceptual quality driven scheduling scheme which included grouping of voice packets and then mapping into subsets based upon requested voice quality and codec type.

This scheme was used along with weighted fair queuing scheme to control packet loss and network delay. Narbutt et.al [38] analyzed the effect of free bandwidth to improve the quality of service of VoIP on 802.11b WLAN network and performed an experiment using RTPtool along with G.711 codec to generate traffic and MEGN generator to generate background traffic. The authors monitored the bandwidth utilization by a WLAN probe application that passively sniffed packets at the L2/MAC layer of the wireless medium and provided information about three MAC bandwidth components (load, access and free bandwidth). The VoIP voice quality can be improved with wideband speech coding standard. To improve the signal quality & balance the load, Levy et.al [39] analyzed the performance of the VoIP system by dispersing the packets over multiple paths instead of single path in the network. The results were analyzed through the Bernoulli and Gilbert loss models with measured loss rate and observed improvement in the signal quality. Chan et.al [40] formulated 2-layer coloring problem to assign coarse time slots and frequency channels to VoIP sessions and proposed a clique analytical call admission scheme for multiple wireless LAN. The problem of packet collision and multi-cell mutual interference was solved and the VoIP capacity in multi-cell environment was increased but at the cost of header overhead and packet aggregation.

Han et.al [41] had raised the issue of noise reduction for VoIP speech codecs. A modified wiener filter based noise reduction scheme optimized to the estimated SNR at each frequency bin as a logistic function. The proposed system was used as preprocessing before speech encoding. G.711, G.723.1 and G.729A VoIP codecs were tested with proposed wiener filter system. Haghani et.al [42] had proposed and simulated a traffic aware multi-tap scheduling algorithm for VoIP applications in WiMAX networks and the simulated results presented the trade off between bandwidth efficiency and delay in each scheduling methods. The problem of the handover was analyzed in [43] by Becvar et.al for speech quality in VoIP over WiMAX networks. The handover could cause three type of degradation packet delay, jitter and packet loss in VoIP and the length of the handover interruption in VoIP packet stream depends linearly on the frame duration. The authors observed that as the number of security associations (SA) increases in WiMAX networks, the packet delays caused by handover increased and the significant increase in the speech quality could be achieved by using shorter frames. Epiphaniou et.al [44] analyzed the affects of jitter, delay & packet loss through various queuing mechanisms such as FIFO, RED, DiffServ for VoIP system and observed that the DiffServ architecture for VoIP system could efficiently improve the network performance. Kazemitabar et.al [45] reviewed various voice over wireless LAN quality and security issues.

Radhakrishnan et.al [46] analyzed the non-intrusive models for measuring & monitoring the voice quality using random neural networks and then proposed the feed-forward architecture for the better evaluation of the speech quality. The role of interpolation finite impulse filter for noise reduction and concealment of the lost packets was analyzed in [47,48,49]. The interpolated FIR (IFIR) filter was implemented on TMS320C6713 DSP processor for narrowband and wideband VoIP system and the implementation results indicated much improvement in the signal quality of the VoIP signal. The performance of IFIR filter had been investigated for narrowband and wideband VoIP system with G.729a [50] & AMR-NB [51] and AMR-WB [52] speech coders under various noisy conditions.

V. CONCLUSIONS

The work in this paper presented the review of the progress in the quality improvement mechanism in the VoIP system. The various quality degradation factors were discussed and at the same time, the solutions to reduce the effect of the degradations factors such as delay, packet loss and jitter had been presented. In future, some possible and useful evaluation results to improve VoIP quality over WLAN networks would be presented.

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