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ANABUT: A New Adaptive Buffering Technique to improve Speech Quality of VoIP users in Mobile Networks

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ABSTRACT

Voice over Internet Protocol (VoIP) is a system that allows the users to make and receive voice calls over internet. The main advantage of this system is to provide low cost and an efficient communication between countries. Instead of analog, the voice messages are transmitted in a digital format. It is also called IP Telephony, Audio Telephony, etc. The Real Time applications, (Multimedia Applications) are highly sensitive to delay, because voice packets cannot tolerate delay. A big problem of VoIP or IP packets (messages) is that it fails to reach to their intended destination in proper order. The reason for that is the data packets follow different routes at different times. It leads high delay variation between the packets. In order to compensate these variations, there need to have a strong mechanism to hold the packets for sometimes then, releasing gradually to the receiver. This adjustment of storage is called playout buffer. This could be either Fixed or Adapted. In existing technologies, it poses severe degradations in speech quality. A new adaptive buffering technique is introduced for enhancing the buffering method to improve the speech quality of VoIP users. It is achieved by estimating dynamic threshold value.

Keywords: Voice over Internet Protocol, Playout, Mean Opinion Score, E-Model, Quality of Service.

1. INTRODUCTION

Conventional way of maintaining voice service is a connection oriented service or circuit switching method [1]. Circuit switching offers reliable and consistent services but, it wastes the bandwidth while they are not in use or silence speech. An alternate to this, packet switching [2] is developed. Packet switched networks have a tremendous growth in today's environment because it provides digitized packets with less error, as an information instead of analogy because, an analog voice suffers from attenuation, noise and some other impairments but in digital it is eliminated. In packet switching, anything can happen for the packets while travel from source to their intended destination. Some packets may reach by late and some may arrive too quickly, so it is difficult to smooth the packets in a proper way especially for the real-time applications. In VoIP, packets are not reached to the destination in which originally how they are sent. In order to provide delay adjustment at the receiver side, it maintains one buffering mechanism called Playout buffer. It distributes the packets to the recipient evenly (equal gap). Even though for releasing the packets to receiver host is a challenging task. Because, it has to adapt the delay variations dynamically, as well as allot the packets in consistent form. If not in persistent form, most of the packets get lost. Those are labeled as lost packets and missed their playout time. As a result, developing the adaptive playout delay algorithm and playout time should be the most essential task for multimedia applications such VoIP, video conferencing and much more.

The objective of this proposed technique is to release the packets in smoothing manner for improving the speech quality of VoIP. The main purpose of the VoIP system is to provide voice with the same quality as it is

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generated by the source. The key factor of this system is an end-to-end delay, jitter, packet loss. In order to maintain the Quality of Service (QoS), real-time traffics should reach the receiver within an adequate time period with enough tolerance. This type of real-time traffic application is highly sensitive to delay. Some protocols like Real Time Protocol, Real Time Control Protocol, Stream Control Transmission Protocol and User Datagram Protocol are play a critical role for performing the transmission of real-time packets. However, in these cases there is no idea about the re-transmission of packets in any real time environment. So, the design of the real-time system must have capacity to forward the packets in between the routers and efficiently playout to the receiver. Generally, two categories of algorithms exits, packet based (individual) and talkspurt (collection of packets) based. Mean Opinion Score (MOS) evaluates the quality of the speech; it is measured by E-Model.

This paper is organized as follows. Section 2 explains past literatures related to voice services, Section 3 describes the proposed technique, Section 4 displays the results and discussions part and finally Section 5 concludes this paper.

2. RELATED WORKS

Luisa Repele et al., [3] proposed a new adaptive delay algorithm for reducing the delay, packet loss. To achieve this, they used cubic spline smoothing method which resizes the buffer dynamically, also smith predictor used to control behavior of the transmitter. *Atri Mukhopadhyay et al.,* [4] maintained the jitter in an acceptable level by proposing one new jitter based algorithm. It also mitigates the packet loss as well as the delay of VoIP traffic. They evaluated the results through the OpNET modeler. *J.P. Ouedraogo et al.,* [5] analyzed Gstreamer based jitter buffer algorithms and explained the importance of speech quality, jitter values from 10ms to 60ms. *Tibor Gyires et al.,* [6] discussed the significance of Quality of Experience in play out buffer. They concentrated both network delay and loss, then proposed QoE based algorithm which is statistically proved. *Liyun Pang et al.,* [7] extended the E-Model which is optimization based for improving the voice quality. It produces better results than existing adaptive buffer algorithms.

Chen-Chi Wu et al., [8] discussed different type of VoIP applications and their performances. They also tested the buffering adjustment of Skype, MSN and GTalk applications, how it adapts for the condition, also they developed one optimal play out buffering adjustment algorithm based on regression method. It frequently adapts the size according to the network condition. They proved MSN provides better results than other applications. *Yusuf perwaj et al.,* [9] proposed one modified play out buffering algorithm to improve the voice quality of the real-time applications. *Pavol Partila et al.,* [10] developed a new method to predict exactly the Mean Opinion Score (MOS) value with an extension of E-Model which ultimately reduces the jitter level and automatically adjusts the buffer size, also maintained the packet loss in VoIP based applications.

Hua-Ching Chen et al., [11] illustrated Novel adaptive play out buffer to minimize the packet loss and reduces the delay in buffer. For this, they set the constraints, end-to-end delay should be under 400ms, packet loss must be in below 10%, which absorbs high delay variations, but it works only for common cases. *Neha Tiwari et al.,* [12] examined the types of play out buffering algorithms, the importance of dynamic adaptation based buffering, overflow, underflow constraints are also discussed.

Better speech quality, choosing optimal playout buffering time, de jitter the packets, mitigating the waiting time in the buffer all these are essential for providing consistent delivery of audio packets. The related literatures show the important things to be carried out by the designer. Above mentioned factors are influenced to prepare this article and it mainly focusing the speech quality improvement.

3. ANABUT: A PROPOSED TECHNIQUE

A new adaptive buffering technique is explained as follows.

1. Observe the codec type and calculate its end-to-end delay (D)
2. Measure the Mean and Standard deviation values of received packets

3. Find the Codec's threshold delay which is computed by estimating the packetization delay and network delay.
4. Check and allow to increase the jitter buffer according to their codec types

   If (Delay (D) >= codec's Threshold delay)

      Increase the jitter buffer to quartile Q1 range

   Else

      Increase the jitter buffer to quartile Q3 range from the mean value

   End if
5. Repeat the steps 1 to 4 for all talkspurts

Here, internet delay should be above 100ms and the ANABUT did not consider the muted or silence packets. That is all these packets are defined as speech packets, because Voice Activity Detection (VAD) take care of those muted speeches. Threshold value varies from time to time, due to packetization delay. It allows the increment of buffer size from average level to end point. The benefit of quartile is the quickest prediction or the movement of buffer size.

4. RESULTS AND DISCUSSIONS

Threshold dynamically sets the delay constraints to receive all the incoming packets within its deadline. It creates the jitter buffer size with an acceptable level and also enhances the quality of speech. Large size of buffer causes biggest weighting time. If it is in smaller size, then it fails to adapt huge delay variation of several audible packets. In order to compensate this factor, this dynamic threshold is introduced here. To improve the voice quality, frequent updates are mandatory and, then only the receiver can hear the correct (exact) message. To identify the voice quality, R value is estimated through E-Model [13]. It is computed by considering the some major factors. R value only shows the essence of the speech quality. The formula to calculate R is,

$$R = R_0 - I_d - I_{eff} + A \tag{1}$$

Where, R_0 - Signal-to-Noise Interference Ratio, I_d - Network delay, I_{eff} - Equipment Impairment Factor and A is a Profit Coefficient factor. In existing technique (400 – 100)/*fr* formula is used. 400 denotes end-to-end delay, *fr* states no. on frames. Sample G.729 codec is taken here. For 10 frames, it provides 30 of I_d and it is fitted into the equation (1), the voice quality factor is 63.2. Obviously, it gives 3.59 MOS value, According to ANABUT, [(250 – 100) / 10] gives 15 of I_d which is almost equals to 3.9 of MOS value. Figure 1 shows the comparison results of existing and proposed ANABUT.

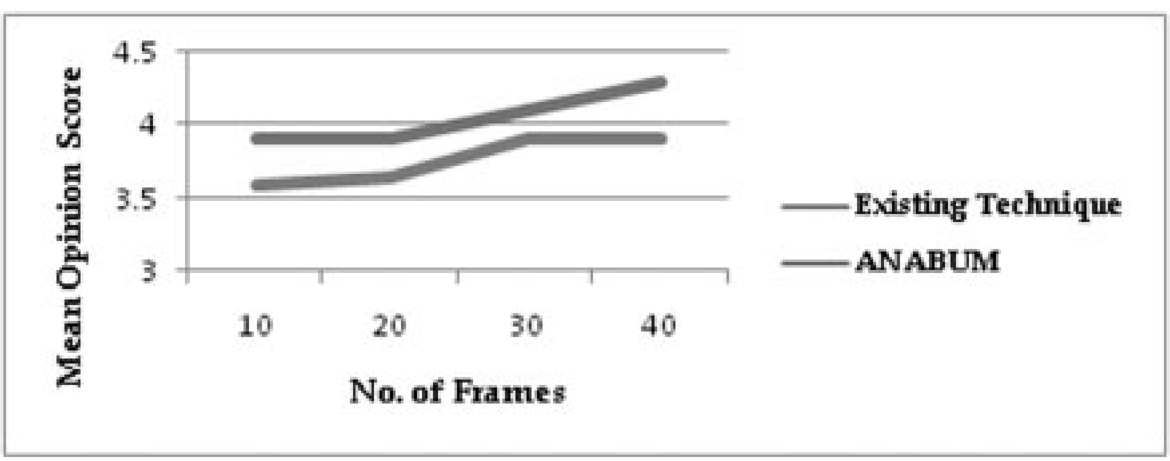

Figure 1: Comparison results of Existing and Proposed ANABUT

5. CONCLUSION

Most of the real-time application uses UDP for their transmission, because of connection less service. For UDP, smoothing buffer method is mainly used at the destination side for equating the delay variations. In VoIP, it is very much needed to improve the voice quality by a limited delay. This paper introduces ANABUT technique, which increases MOS value; it leads to get better speech quality results improvement, which tends to enhance the Quality of Service automatically. The ultimate aim of this ANABUT is to enhance the speech quality of VoIP users in heterogeneous networks.

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