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Performance Evaluation of Packet Loss Replacement using Repetititon Technique in VoIP Streams

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Abstract

. Voice over Internet Protocol is an upcoming technology. VoIP enables voice communication through the Internet. Internet provides lot of services to the people. VoIP differs from conventional networks. The online transmission of audio data over networks faces lot of problems. Real time voice transmission is not as easy as ordinary text data. Networks are not designed to support real time communications. The conversational quality of a VoIP communication is dependent on several factors such as networking conditions, coding process used, speech content, type of error correction, flow id. The factors which affects the Quality of Service (QoS) is due to delay, delay variation, packet loss, repeat - request, loss rate, QoS control, throughput, network security, network reliability, providing bandwidth, voice compression, echo suppression and jitter on the perceived conversational quality. Packet loss is a serious and critical issue for voice over internet protocol applications. It degrades the performance of VoIP. This paper provides Packet loss replacement mechanisms and discusses their suitability for use in IP-based networks by using a receiver based repetition technique. The comparison of static play out buffer with optimal play out buffer was carried out in this paper. It performs better quality monitoring of VoIP service at the receiver side. This work is implemented in ns-2. The performance results suggest that the overall quality of VoIP is better.

1. Introduction

Internet provides a large number of services to the people. The services are file sharing, exchanging and retrieving information, e-commerce, e-mail and advertisement. An interactive service like video conferencing is widely used in the internet community. The most popular service is voice over internet protocol (VoIP). VoIP is internet telephony. It replaces the traditional Public Switched Telephone Network (PSTN). VoIP offers wide range of benefits to both enterprise and communication network. VoIP allows people to talk with each other freely at low rates. Nowadays the industry pays more attention to voip applications.

The use of VoIP is growing rapidly. VoIP networks differ from conventional telephone networks in that voice quality is affected by a variety of network impairments such as packet loss, delay, jitter, echo, network security and throughput. These factors will affect the quality of voice over internet protocol. The existing internet service does not provide guaranteed service. It cannot satisfy the quality of service requirement of emerging technologies like VoIP. The packet loss is a serious problem which significantly decreases the quality of voice communication in the internet. This work focuses on packet loss which is a crucial problem to be addressed.

VoIP is an Internet telephony which simply means that the technology and techniques to make voice phone Calls, local, long distance, and international over the Internet using PC. The definition of Internet telephony is broadening day by day to include all forms of media (voice, video, image), and all forms of messaging and all variations of speed from real-time to time-delayed.

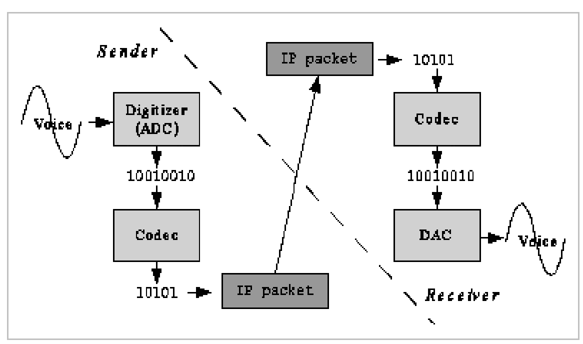

Figure 1. Transmission Technology of VoIP

The transmission technology of VoIP is shown in Figure1. The caller's voice is digitized. The digitized voice is compressed and then separated into packets using coder decoder algorithms. These packets are addressed and sent across the network which is to be reassembled in the proper order at the destination. Again, this reassembly can be done by a carrier, and Internet Service Provider, or by PC. During transmission on the Internet, packets may be lost or delayed, or errors may damage the packets. Conventional error correction techniques would request the retransmission of unusable or lost packets, but if the transmission is a real-time voice communication, this technique obviously would not work. So sophisticated error detection and correction systems are used to create sound to fill the gaps.

The streaming of audio or video content over the Internet is a challenging task. This is due to the fact that the Internet is a packet switched network with a little quality of service (QoS) guarantee. The major challenge of the VoIP network is maintaining quality. Packet loss is a serious and critical issue for voice over internet protocol applications. If a voice packet is not received when expected, it is assumed to be lost. Non real time applications like TCP have retransmission capability but real time applications like UDP does not have retransmission capability. Packet loss can occur for a number of reasons [10]. This paper attempts to address the problem of packet loss over the VoIP.

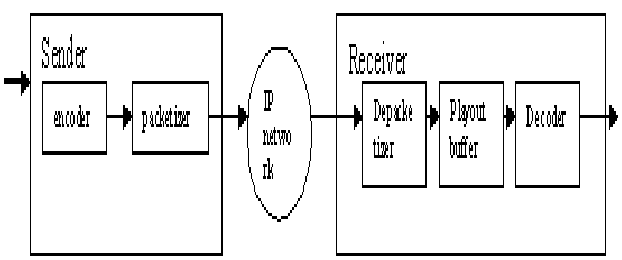

Figure 2. Components of VoIP

A VoIP application is composed of several building blocks shown in Figure 2. At the sender side, the encoder periodically samples the voice signal. A large variety of encoders are available. They are differentiated by the quality and bandwidth consumption. At the receiver side [1], speech frames are de-multiplexed and inserted into a play out buffer. A play out buffer enforces speech frames to be decoded at the same interval at which they were generated by the encoder. To do so, it might re-order, delay or even drop them if they arrive after their expected playback time. Play out buffers can be either fixed or adaptive. The network delay is constant in fixed play out buffer. In adaptive play out buffers, the delay depends on the networking conditions. The play out buffer delivers speech frames to the decoder, which actually playbacks them. Some decoders implement packet loss concealment (PLC) techniques which allow missing speech frames to be somewhat reconstructed by interpolating. PLC techniques can obviously mask a limited number of losses.

Factors Affecting QoS IN VoIP

The conversational quality of a VoIP communication is dependent on several factors. Packet loss degrades the performance of VoIP packet loss can occur due to the following reasons [12].

1. Delay in packet transmission from sender to receiver
2. Packets arriving too late at the receiver side
3. Heavy loading
4. Congestion of routers and gateways.
5. The variation in packet inter arrival time. The difference between when the packet is expected and when it is actually received is jitter.
6. The loss of voice packets from sender to receiver.

Benefits of VoIP

Telecommunications carriers around the world have already introduced IP into their networks because it provides economic benefits over traditional telecommunications networks.

Greater Efficiency: The conventional circuit-switched technology of the PSTN requires a circuit between the telephone company's switch and the customer's premise to be open and occupied for the entire duration of a call, regardless of the amount of information transmitted. In contrast, on IP networks, all content whether voice, text, video, computer programs, or numerous other forms of information travels through the network in packets that are directed to their destination by diverse routes, sharing the same facilities most efficiently.

Lower Cost: IP systems will offer a more economical means for providing communication connections. Also and this is one of the sources of concern on the part of incumbent voice long distance carriers -- Internet technology makes available to anyone with a personal computer and modem the ability to bypass the long distance PSTN.

Higher Reliability: In some respects, IP networks also offer the potential for higher reliability than the circuit-switched network because IP networks automatically re-route packets around problems such as malfunctioning routers or damaged lines. Also, IP networks do not rely on a separate signaling network, which is vulnerable to outages.

Supporting Innovation: IP is a nonproprietary standard agreed on by hardware and software developers, and is free to be used by anyone. This open architecture allows entrepreneurial firms to develop new hardware and software that can seamlessly fit into the network. In contrast, the circuit switched network operates as a closed system, thus making it more difficult for innovative developers to build and implement new applications.

The benefits of VoIP can be categorized as

1.Cost Reduction.

1) Reducing long distance telephone costs.
2) The sharing of equipment and operations costs across both data and voice users improves network efficiency.

2. Simplification. An integrated infrastructure that
1) Supports all forms of communication allows more standardization.
2) Reduces the total equipment complement.
3) Supports dynamic bandwidth optimization and a fault tolerant design.

3. Consolidation.
1) Universal use of the IP protocols for all applications holds out the promise of both reduced complexity and more flexibility.
2) Related facilities such as directory services and security services are easily shared.

4. Advanced Applications.
The longer-term benefits of VoIP are expected to be derived from multimedia and multi-service applications.

Loss of speech packet can cause a noticeable deterioration in the quality of speech. The rest of this paper is organized as follows. Section 2 reviews the reconstruction techniques of packet loss in VoIP. This is followed by the description of receiver based repetition method in section 3. In section 4, the experimental results are given. Finally section 5 summarizes the contribution.

2. Background Study

VoIP is a part of the group of technologies called voice over packet networks. Other network protocols like asynchronous transfer mode (ATM) can perform similar functions. Though the concept of VoIP is simple, the implementation and applications of it is a bit complicated. In order to send voice, the information has to be separated into packets just like data. Packets are chunks of information broken up into the most efficient size for routing. From there, the packets need to be sent and put back together in an efficient manner. For more efficient use, the voice data can be compressed so that it require less space and will certainly record only a limited frequency range. There are many ways to compress audio, the algorithm for which is referred to as a compressor/de-compressor (CODEC). Many number of Codec's existing depending on the application (e.g., conversations, music, movies and sound recordings). The Codec's are optimized for compressing voice, which significantly reduce the bandwidth used compared to an uncompressed audio stream. Speech Codec's are optimized to improve spoken words at the expense of sounds outside the frequency range of human speech. Recorded music and other sounds do not generally sound very good when passed through a speech CODEC.

The major challenge of VoIP network is maintaining quality. Several methods that were proposed by many authors can be categorized into two techniques [14] namely,

1. Sender based
2. Receiver based

Sender and Receiver based techniques are shown in Figure 3 and 4. It may be classified as "active" or "passive". Active techniques generally involve the receiver sending a message to the sender informing the sender which packets are lost.

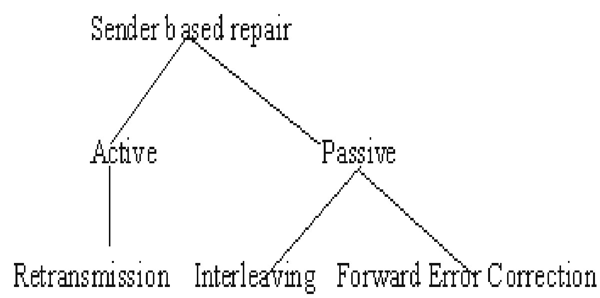

Figure 3. Sender based repair methods

There are generally considered to be two types of passive techniques: interleaving and forward error correction. In interleaving, In Interleaving [5] [13] [6] [7], the information of a speech is distributed in multiple packets. The data is regrouped before transmission in the sending side. They are rearranged in their original form at the receiver side. The loss of a single packet will not affect the conversation much. Thus instead of losing the whole packet small parts from distributed packets are lost. The advantage of interleaving is that it does not increase the bandwidth requirements of a stream. Forward error correction [16] comprises sending additional data with each packet, often referred to as redundancy data that is useable to reconstruct lost packets. Reed Solomon encoding/decoding is a well-known forward error correction technique. Passive methods usually require that all data in a given data stream be received prior to processing and reconstructing lost packets. As a result, these techniques may be time consuming and may require large buffering capacity in the receiver.

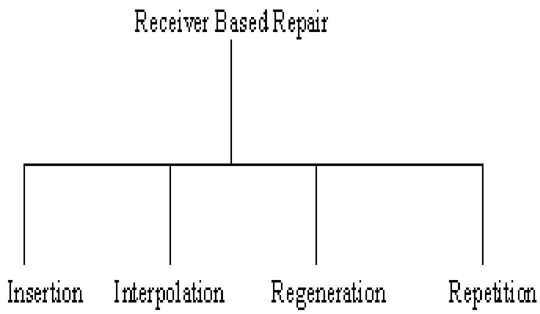

Figure 4. Receiver based repair methods

Silence Substitution - the method comprises replacing voice that is encoded in a lost packet with a period of silence. Packet Repetition is a receiver based repair method is shown in figure 4. This method comprises replacing a lost packet with a duplicate of a packet immediately preceding the lost packet. Pitch Estimation - the method comprises determining a fundamental frequency of voice encoded in packets preceding a lost packet and duplicating the fundamental frequency during a period in which voice encoded in the missing packet would be made audible. Linear Prediction - the method [10] comprises determining waveform parameters from a portion of an audio waveform preceding a segment of the waveform encoded in a lost packet. The lost segment is synthesized responsive to the predicted parameters using linear interpolation techniques. Optionally, a portion of the audio waveform following the lost segment may also be used to perform linear prediction.

A. GSM - AMR codec

GSM/AMR [4] product is a complete implementation of the Global System for Mobile (GSM) Adaptive Multi-Rate (AMR) voice coder, and is fully compliant with the ETSI GSM 06.90 specification. The Adaptive Multi Rate (AMR) speech codec is the default speech codec for GSM 2+ and WCDMA third generation wireless systems. GSM-AMR is based on Algebraic CELP (ACELP) and operates at eight bit rates ranging from 4.75 kbps to 12.2 kbps. It is developed to preserve high speech quality under a wide range of transmission conditions.

Technical Specification

- Performs compression and decompression of all 8 specified data rates, ranging from 4.75 kbps to 12.2 kbps
- Includes support for frame substitution and muting (GSM 06.91)
- Includes support for comfort noise generation (GSM 06.92)
- Includes support for DTX - Discontinuous Transmission (GSM 06.93)
- Includes support for VAD - Voice Activity Detection (GSM 06.94)
- Supports a C callable API for initialization / encoding and decoding Supports Multi channel Capability

Quality of AMR codec's

Generalizing the analysis of voice quality Table 2 was made. This table describes properties of AMR-n codec's in good conditions. It is necessary to notice that only two AMR codec's (AMR-7, AMR-6) are characterized as satisfying users which is tabulated in Figure 5. The other types of codec's falls to category Some dissatisfied or Many dissatisfied.

Codec mode	R rate kbit/s	MOS	User Satisfaction
AMR-7	12.2	4.06	satisfied
AMR-6	10.2	4.06	
AMR-5	7.95	3.91	Some Dissatisfied
AMR-4	7.4	3.83	
AMR-3	6.7	3.77	
AMR-2	5.9	3.72	
AMR-1	5.15	3.50	Many Dissatisfied
AMR-0	4.75	3.50	

Figure 5**.** Properties of AMR-*n* codecs

3. Receiver Based Packet Loss Replacement

In an audio communication system speech is encoded and packetized at the transmitter, sent over a network and then decoded at the receiver. Packet loss concealment algorithms are needed to conceal the packets of the speech signal that are lost during transmission. There are basically two types of techniques. These techniques are termed transmitter based and receiver based techniques for packet loss concealment. The techniques described in this work are receiver based and are applicable to the ITU recommendation G.711. G.711, unlike some CELP based coders, does not have built-in packet loss concealment algorithms. So a receiver-based algorithm is required to conceal a lost packet. One advantage of G.711 is that the signal returns to its original form immediately after a missing packet. With CELP based coders the signal takes time to recover after a missing packet.

The steps involved in VoIP transmissions are

Step 1. At the source, the voice is sampled.
Step 2. The voice data is encoded by encoder to save bandwidth.
Step 3. The encoded bit stream is converted into packets.
Step 4. The packets sent along the network.
Step 5. At the destination, these packets are reassembled.
Step 6. The decoder converts the voice data back into voice samples.
Step 7. These samples are put in a buffer.

The encoded audio samples are collected into packets and transmitted over the network. An artificial voice reconstruction delay is introduced at the receiver. This is achieved by buffering received packets for enough time, so that most of the late packets have a chance of being received before their playing time. In order for receivers to be able to reconstruct the timing of the received packets and calculate scheduled play out times, sequence numbers and timestamp information is included in RTP headers. The voice reconstruction delay needs to be adaptive to cope with changing network conditions. Receiver maintains network delay and readjusts the amount of buffering between talk spurts. The amount of buffering is enough to receive most of the packets, but some will always arrive too late to be played back and hence, can be considered lost. The lost packets are replaced by copies of last received packets. The necessary step towards this is, the buffer must have a copy of the last packet. The same speech is played twice in this method [9]. It is a receiver based packet loss concealment strategy.

If Packet2 contains b, which is lost due to some network problem, then the repetition method fills a as a second packet. The replacement of the above transmission is shown in figure 6.

Packet1	Packet2	Packet3	Packet4	Packet5
a	b	c	d	e
received	loss	received	received	received

Figure 6. The example data for transmission and its status

To replace the lost packet x_t, the copy of the previous packet x_{t-1} was considered.

$$X_t = X_{t-1}$$

The replacement using repetition is shown in figure 7.

Packet1	Packet2	Packet3	Packet4	Packet5
a	a	c	d	e

Figure 7. Replacement using Repetition

Packet repetition techniques assume that speech characteristics do not change very rapidly, and a preceding segment of speech is used to reconstruct the missing packet. The mechanism fails when the packet sizes are large and the loss rate is high, since the speech characteristics are more likely to have significantly changed. The addition of a piggybacked redundant encoding for the previous packet provides a substitution of fill-in audio for one lost packet. If two or more consecutive packets are dropped by the network, then both the primary and redundant information for some period of time will be lost. The receiver based method is proposed here. The quality of voice is estimated using E-model and MOS.

A. Voice Quality Measurements - E-Model And MOS

The subjective quality is predicted by E-model by an average listener combining the impairment caused by transmission parameters. The rating can be used to predict subjective user reactions, such as the Mean Opinion Score (MOS) [15]. According to ITU-T Recommendation, the E-model rating *R* is given by the following expression [8]

$$R = R0 - Is - Id - Ie + A \qquad (1)$$

Where

R -Transmission rating factor

R0 - signal to noise ratio

Is - the combination of all impairments which occur more or less simultaneously with the voice signal

Id - the mouth-to-ear delay impairment factor

Ie - represents impairments caused by low bit rate codecs – equipment impairment factor

A is the advantage factor or expectation factor

The resulting score is the transmission rating *R* factor, a scalar measure that ranges from 0 (poor) to 100 (excellent). *R* factor values below 60 are not recommended [3]. According to [8], the *R* factor is related to MOS as follows:

For $R < 0$ MOS = 1

For $0 < R < 100$ MOS = $1 + 0.035R + 7R(R-60)(100-R)\times10^{-6}$

For $R > 100$ MOS = 4.5

The E-Model not only takes in account the transmission statistics, but it also considers the voice application characteristics, like the codec quality, codec robustness against packet loss and the late packets discard. According to [2, 3, and 11], eq. (2) can be reduced to the following expression
where *Id* is a function of the absolute one-way delay *Ie* is, in short, a function of the used codec type and the packet loss rate $R = 93.4 - Id - Ie$ (2)

4. Experimental Results and Analysis

A simple networking environment is set up, which consists of only one VoIP flow established between two ns-2 nodes. The network set up is shown in figure 8. These nodes are connected through an 8 Mb/s wired link with a constant 10ms propagation latency. The VoIP sender employs a GSM AMR codec at 12.2 kb/s with one – to – one exponential model. A CBR traffic generator is used to introduce loss.

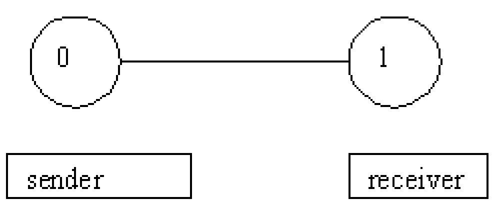

Figure 8. network structure

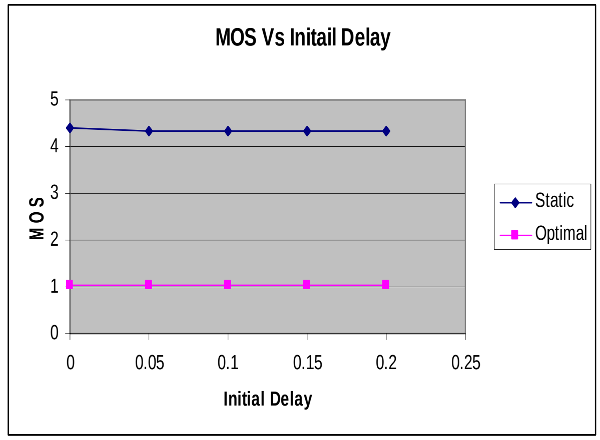

Figure 9. MOS versus Initial Delay

The run duration is 10 seconds. The probability of packet loss varies from 0.1 to 1.0. Figure 9 shows the MOS when the network delay increases from 1ms to 200 ms. The packets are never discarded. Therefore the Packet Loss Ratio (PLR) is 0.The results are shown when the packets are delayed according to exponential distribution. There are two play out algorithms

1. Static play out buffer
2. Optimal play out buffer

In VoIP systems, the voice packet must be played out at the receiver side in a timely manner and in the order they were emitted from the sending side. The basic function of a play out buffer is to collect packets and to store them, and then to send specified number of packets to the next mechanism.

Play out buffer is located at the end of VoIP system, and the main intention of it is to smooth speech. When variable delays take place, play out buffer can allow some later-arrived packets to be played out, which depends on the set-up of packet play out time, to keep the completeness of speech. However, any packet, arrived later than the play out time, will be simply discarded.

The set-up of play out time can be fixed or adaptive, but both need synchronization between sender and receiver due to the changing network delay. Fixed play out time set-up/scheduling is simply, but normally causes a constant delay and cannot follow the change of network delays. Adaptive play out scheduling was introduced to overcome these problems, and is controlled by corresponding play out buffer algorithm, which can utilize the silence time between two successive voice periods, referred to as talk spurt, to slow down or speed up play out time of each talk spurt.

The static play out algorithm shows the high quality performance than the optimal play out algorithm. The MOS values are 4.40 to 4.35 in static optimal algorithm whereas the values in optimal algorithm are 1.02 with 0.10 packet loss rate. Even though the delay varies from 0.0 sec to 0.2 sec, the static and optimal algorithm does not vary. The same quality is maintained for varying delays in both the algorithms.

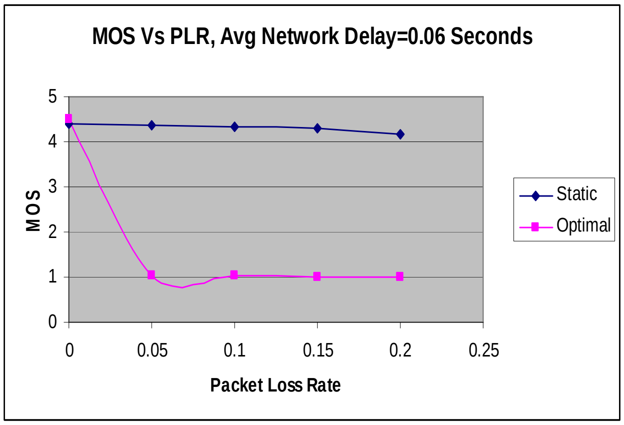

Figure10. MOS versus PLR

Figure 10 shows the results of MOS versus PLR. When the loss rate increases the MOS values are decreased in both the static and optimal play out algorithms with initial delay of 0.06 seconds. There is a slight variation in both the algorithm. From these results, the static algorithm maintains the high quality than the optimal algorithm. This work is implemented in ns-2.

The Mean Opinion Score (MOS) is the numerical method of expression in voice communication to determine whether the quality is good or bad. MOS is expressed in one number, from 1 to 5, 1 being the worst and 5 the best. The following Figure 11 represents MOS rating values. The introduction of VoIP has enabled many organizations to supplement or replace existing circuit switched telephone networks, providing a new value added services.

Mos Rating	**Range of Rating**	**User satisfaction**
5	$90 \leq R < 100$	Very much satisfied
4	$80 \leq R < 90$	good
3	$70 \leq R < 80$	Satisfied-some users dissatisfied
2	$60 \leq R < 70$	Poor-Many users Dissatisfied
1	$50 \leq R < 60$	Bad quality-Nearly all users dissatisfied

Figure 11 MOS Rating

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Quality of service is a long standing research topic for the internet. No QoS mechanism has ever been deployed across most of the internet. Internet applications are designed to adapt changing network conditions. The simplest QoS deployment consists of carefully engineering the deployed network so that congestion cannot occur because bandwidth has been over provisioned. The user satisfaction of voice quality is given in figure 12.

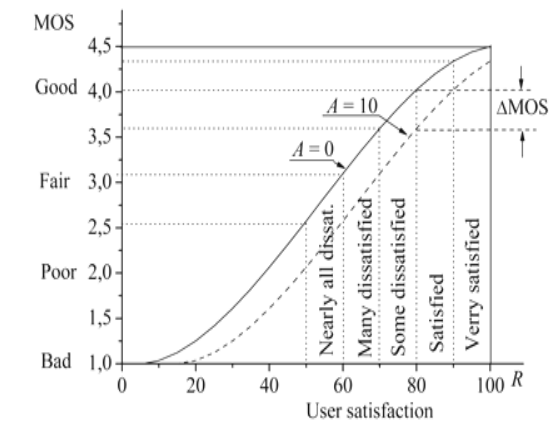

Figure 12. user satisfaction of voice Quality

Numerous technological problems are faced on the issue of Quality of Service. The packet loss is a major problem. This problem cannot be avoided but it can be minimized. The packet repetition with static play out algorithm performs moderately than other replacement techniques. Based on time delay, loss rate and MOS, the better results are obtained. Hence the receiver based repetition provides high quality VoIP signal in the streaming audio. This work can be extended by incorporating other statistical based methods.

5. Conclusions

Quality of service is a long standing research topic for the internet. No QoS mechanism has ever been deployed across most of the internet. Internet applications are designed to adapt changing network conditions. The simplest QoS deployment consists of carefully engineering the deployed network so that congestion cannot occur because bandwidth has been over provisioned. Voice over IP is the new and fancy development in the telecommunication industry. It promises to deliver cost savings to users and service providers and is driving the convergence of network and telecom. It offers improvements in quality, interoperability and applications in the near future.

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