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A Survey of Packet Loss in VoIP

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Abstract

This paper surveys packet loss in Voice over Internet Protocol (VOIP). The voice quality is affected by many network impairments. IP networks are on a steep slope of innovation that will make them the long term carrier of all types of traffic, including voice. Such networks are not designed to support real time voice communication because of their variable characteristics. 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, flowid. The factors which affects the quality of service 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 survey gives an overview of existing Packet loss concealment mechanisms and discusses their suitability for use in IP-based networks. Additionally, the impacts of IP over wireless networks on the requirements of error control mechanisms are discussed. Different network scenarios are used to assess the performance of retransmission-based error correction and forward error correction.

Keywords: Concealment, Packet Loss, QOS and VOIP.

Introduction
Voice Over Internet Protocol (VOIP) is one of the fastest growing applications for the internet today. Many users expect high quality telecommunication services. Voip is internet telephony. It is a category of hardware and software. It enables people to use the internet as the transmission medium for telephone calls by sending voice data in

packets using IP rather than by traditional circuit transmissions of the PSTN (Public Switched Telephone Network). Voice over IP networks differ from conventional telephone networks. VoIP technology has allowed phone calls to be routed over Internet infrastructure rather than the traditional Public Switched Telephone Network (PSTN) infrastructure. The technology, called Voice over Internet Protocol (VoIP), uses the Internet Protocol (IP) to route packets containing small portions of voice conversations between the callers.

The transmission technology of VOIP must be in digital. Hence the caller's voice is digitized. The digitized voice is compressed and then separated into packets using complex 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 in the gaps. After the packets are transmitted and arrive at the destination, the transmission is assembled and decompressed to restore the data to an approximation of the original form.

In Voice over IP (VoIP) applications, delay, jitter[13] and packet loss are the main network impairments that affect voice quality. Packet loss occurs when packets are lost during transmission or simply arrive too late to be used.

Packet loss can occur for a number of reasons

(1) Congestion of routers and gateways, which lead to packet being discarded
(2) Delays in packet transmission, with packet arriving too late at the receiver to be played back.
(3) Heavy loading of workstations, leading to scheduling difficulties in multi-tasking operating system

Transmission of data makes use of the TCP/IP protocol suite which allows for retransmission of missing packets, but VoIP, which uses UDP, does not allow retransmission and the missing packets are simply left out of the call. Such loss causes voice clipping and skips [3]. One of the frequently used methods was retransmission. Since retransmission mechanisms are often unacceptable for interactive real-time audio applications such as Internet phone, because of the increased end-to-end delay. Applications that run over UDP do not retransmit lost packets. There is a need for error recovery before transmitting the data. When using network services that do not guarantee the Quality of Service (QoS) required by audio-visual applications, the recovery from losses due to congestion in the network is a key problem that must be solved.

MOS (Mean Opinion Score) is the most well-known measure of voice quality. It is a subjective method of quality assessment. Upto 1% is usually undetectable, more than 3% is the maximum permitted within industry standards. Test subjects judge the quality of the voice transmission system either by carrying on a conversation or by listening to speech samples. They then rank the voice quality using the following scale: 5 – Excellent, 4 – Good, 3 – Fair, 2 – Poor, 1 – Bad [21].

MOS is then computed by averaging the scores of the test subjects. Using this scale, an average score of 4 and above is considered as toll-quality. MOS was originally designed to assess the quality of different coding standards.

History

The summary of survey was tabulated for a decade. The table 1shows the concepts used by researchers. The results obtained by them and the drawback they faced are also listed.

[17] In this paper, a new front end speech recognition over IP networks were proposed. They extracted the recognition feature vectors directly from the encoded speech instead of decoding it. They considered the ITU G.723.1 standard codec. The benefits quoted by authors are, this approach is very effective to packet loss since it is not constrained to the error handling mechanism of the codec. They compared new front end method with the conventional approach called Automatic Speech Recognition (ASR). There are two types of ASR. They are Speaker independent continuous speech and speaker independent isolated speech. The proposed method is compared with both ASR techniques. This scheme outperforms the conventional procedure. They concluded that the improvement is higher even though the network condition was worst.

[20] The author proposed Global Local Search – Time Scale Modification (GLS – TSM) receiver based scheme. This scheme is classified as sender-receiver based, network based, receiver based. This work focused only receiver based. This method provides flexible arrival delay cutoffs, reducing packet loss at the receiver, Low computational complexity, lost packets concealed effectively and no additional delay. But the performance is limited in silence, noise substitution and packet repetition. Rigorous objective and subjective tests for large number of input speech samples with varying network condition were conducted. These tests confirmed better performance. They concluded this fully receiver based scheme is suitable for any practical voip system.

[6] Describes an adaptive Joint Play out buffer and Forward Error Correction scheme. FEC techniques can be classified as media independent and media dependant. Media independent FEC uses block codes to provide redundant information. FEC send redundant information along with the original information. The benefits are, avoids delay, performs better than existing algorithms and recovered packet loss. The Drawback of this scheme was it introduces additional delay, uses block codes, provides redundant information. They compared the performance of play first and play best strategies. Play first is a delay aware FEC scheme. Play best is a non delay aware FEC scheme. Both lead similar results. Among these the author recommends delay aware play first strategy because of its simplicity. There is a real benefit using joint method.

[24] Discusses the maximal rate algorithm, proportionally fair algorithm and simple admission control scheme. The proportionally fair algorithm is suitable for elastic traffic when the channel condition is considered. The maximal rate algorithm shows twice of the loss rate for the same delay bound and load and it is a good choice

to improve the whole performance. Simple admission control scheme controls the average portion of slots occupied by voip packets. They compared hard and soft algorithm. The frame structure divides in to two parts. The first part of the frame gets more priority. The second part is distributed to normal data which do not need urgent delivery. There are two algorithms to schedule voip. One is maximal rate algorithm the other one is proportionally fair algorithm. Each scheduler is divided into 2 categories. They are maxhard, maxsoft, pfhard, pfsoft. They concluded that pfsoft showed the best result. But in this method if the traffic is high, the drop probability is also high.

Table1: Year wise findings.

YEAR	AUTHOR	CONCEPT	FINDINGS	DRAWBACK
1998 [18]	C.Perkins et al.	FEC	Reduces packet loss	Increases end to end delay
1999 [4]	Bolot et al.	Adaptive Delay aware error control	Reduces Delay	Not considered losses Not managed additional delay due to FEC
2001 [17]	Palaez-moreno et al	New front end approach speech recognition over IP	Very effective to packet loss	Performance was better without considering the network condition
2002 [20]	Samar Agnihotri et al	GLS – TSM receiver based scheme	Reduces packet loss at the receiver, Low complexity and Lost packets concealed effectively	Performance is limited with silence, noise substitution and packet repetition
2003 [6]	Catherine Boutremans et al	Adaptive joint playout buffer & FEC	Performs better than existing algorithm	Additional delay, blockcodes, redundancy
2004 [24]	Young-June Choi et al	Maximal rate algorithm Proportionally fair algorithm Simple admission control scheme	Good choice to improve the whole performance	Drop probability increases if the traffic increases
2004 [15]	Lingfen Sun et al	New method for predictive voice quality for buffer	Achieves the optimum perceived voice	Delay distribution model

YEAR	AUTHOR	CONCEPT	FINDINGS	DRAWBACK
		design / optimization	quality	
2004 [12]	K.Kondo et al	Linear prediction in both forward and backward direction	Reduces complexity and processing delay	Improved performance
2005 [21]	Shveni P metha	Comparative study of techniques to minimize packet loss	Reduced packet loss	Redundancy
2005 [8]	Fernando Silveira Filho et al	Adaptive forward error correction for interactive streaming over the internet	This method not only recovers more packets but also it performs efficiently	It increases bandwidth requirements
2005 [14]	Kiki Karadimou et al	Source-filter model for multichannel audio	Reconstructs the lost information exclusively at the receiver side without any overhead to the transmitter	Small overhead and delays for the total encoding / decoding process
2006 [9]	Hanoch et al	Interleaving – packet dispersion	Improves quality , balance the load	Concentrated on burst losses
2007 [1]	An chan et al	Clique analytical call admission in multiple WLAN	Prevent packet collision Solves multi cell mutual interference Increases voip capacity	Header overhead, packet aggregation
2007 [2]	Ashwin Kashyap et al	Zero stuffing and packet repetition scheme	works well with G.711 and G.722 codecs simultaneously	Artifacts are introduced
2007 [11]	Jes Thyssen et al	a candidate for the ITU-T G.722 packet loss concealment standard	alternative codec for packet loss concealment	Additional computational complexity and memory usage

[15] P**resent their analysis with an efficient new method for predicting voice quality for buffer design/optimization. In this method first, nonlinear regression models are derived for a variety of codecs (e.g.G.723.1/G.729/AMR/iLBC) with the aid of ITU PESQ and the E-model. Second, they propose the use of minimum overall impairment as a criterion for buffer optimization. This criterion is more efficient than using traditional maximum Mean Opinion Score (MOS). Third, they show that the delay characteristics of Voice over IP traffic are better characterized by a Weibull distribution than a Pareto or an Exponential distribution. Based on the new voice quality prediction model, the Weibull delay distribution model and the minimum impairment criterion, they propose a perceptual optimization buffer algorithm. Preliminary results show that the proposed algorithm can achieve the optimum perceived voice quality compared with other algorithms under all network conditions considered.** Preliminary results show that the proposed algorithm can achieve the optimum perceived voice quality compared with other algorithms under all network conditions considered.

[12] Move ahead on the idea of the linear prediction both in the forward and backward direction was proposed. Subjective quality is compared between the proposed method and the packet loss concealment algorithm. This method showed higher scores. There is a complexity and processing delay. They recommended adaptive LPC prediction to improve the quality. The adaptive LPC prediction order depends on the consecutive number of repetitive prediction. They planned to reduce the complexity using gradient LPC coefficient updates. The adaptive forward bidirectional prediction modes depending on the measured packet loss ratio is planned to reduce the processing delay.

[9] In their work, Delivery of real time streaming applications such as voice and video over IP in packet switched networks is based on dividing the stream into packets and shipping all the packets over a single path along the network. In contrast to traditional approach, the packets are dispersed over multiple paths. The reason is to improve quality, balance the load. The noticeable loss rate was used as a measure. They analyzed Bernoulli and Gilbert model for burst losses. The results suggested that the use of packet dispersion can be useful for voip applications.

In [1], a clique analytical call admission in the multiple wireless LAN scheme was proposed. Nowadays infrastructure WLAN is the most widely deployed network architecture. A 2–layer coloring problem was formulated to assign coarse time slots and frequency channels to voip sessions. The benefits are, prevents packet collision, solves multi cell mutual interference. The header overhead and packet aggregation is the big problem found by authors. The proposed scheme increases voip capacity in the multi cell environment. In the single cell scenario, all client stations are within the same cell and associated with the same AP. In a multi cell WLAN instead of one clique, multiple cliques can be formed.

Bolot et al. [4] proposed an adaptive rate/error control that optimizes a subjective measure of quality and incorporates a rate control. Their algorithm describes that the destination plays the best received copy of a given packet. They neither consider losses due to play out buffer overflow nor try to optimize the overall end-to-end delay. They do not manage the additional delay due to FEC. It is recognized that the end-to-end delay has a great impact on the perceived quality of interactive communications, with a threshold effect around 150ms. As a result the FEC scheme increases the delay. An adaptive delay aware error control was proposed in [16] to overcome this problem. This algorithm is based on the assumptions that if the source went to the trouble of adding some redundancy then the destination should wait for the redundant information to arrive.

[21] The author proposed many techniques to minimize the packet loss in voip. The first technique to replace lost packets**, Interleaving, Repetition**, **and Interleaving with Repetition** were used. In **Interleaving,** the information of a speech part is distributed in multiple packets. The data units are regrouped in a crossed form before transmission such that they are distributed, and at the receiver they are arranged in their original form. Thus instead of losing the whole packet small parts from distributed packets are lost. In **Repetition,** lost packets are replaced by copies of last received packets. In **Interleaving with Repetition,** the data are interleaved before sending and then any missing part is substituted using the repetition technique at the receiver. The second method was **Forward Error correction and Concealment (**FEC) adds redundancy to the transmission so that lost packets can be recovered, as long as the following packets are received successfully. Finally, **Optimized unequal error protection** method was used. In this method, certain packets are allocated more FEC protection than others depending on their perceived importance.

[18] FEC is used to mitigate the impact of packet losses. It increases the end to end delay since the destination has to wait for the redundant packets to be received in order to repair packet losses. It increases the bit rate requirement of an audio source. The sender driven mechanisms for error correction was proposed. They are **Retransmission, Insertion based error concealment, and Interpolation based repair.** Retransmission works well for small loss rates, In Retransmission **Interleaving, FEC** was discussed**.** Interleaving disperses the effect of packet loss whereas **FEC** is media dependant and media independent. In **Insertion based error concealment,** two schemes were used by author. One is **Silence substitution,** which fills the gap left by a lost packet with silence in order to maintain the timing relationship between the surrounding packets. It is only effective for short packet lengths and low loss rates. The other one is **Noise substitution,** instead of filling in the gap left by a lost packet with silence, background voice is inserted. In **Interpolation based repair, pitch waveform replication, time scale modification, and Regeneration based repair** were discussed. In **pitch waveform replication,** unvoiced speech segments are repaired using packet repetition and voiced losses repeat a waveform of appropriate pitch length. This performs better than wave form substitution. The **Time scale modification** performs better than both pitch waveform replication and waveform substitution. The interpolation of transmitted state and model based recovery are **Regeneration Based Repair**.

[8] Developed an adaptive mechanism for FEC selection using a predictive model. This method not only recovers more packets but also it performs efficiently. The computations required for the entire control mechanism must be fast. It increases bandwidth requirements, and controls redundancy. This methodology is applicable to video-conferencing.

[2] The author proposed zero stuffing and packet repetition schemes to reduce the packet loss. This scheme works well for G.711 and G.722 codecs simultaneously. This is useful in multirate system where both narrowband and wideband speech were supported. It introduces artifacts in both the methods.

[14] The source / filter model for multichannel audio was proposed by authors. This is useful for both stored recordings and streaming applications. This method reconstructs the lost packet only at the receiver side without redundancy. There is a small overhead for encoding decoding process.

[11] Suggested an alternative approach for G.722 packet loss concealment standard. The algorithm is based on waveform extrapolation in the speech domain. They compared many PLC codec algorithms. An additional complexity and memory usage are drawbacks.

Future Work

Repair methods for packet loss are known as voice reconstruction mechanisms [23]. Better performance was provided by adaptive FEC schemes [7, 16, 19, and 22]. The performance of these schemes is limited by potentially high buffering delays introduced and poor quality of speech delivered when schemes such as splicing, silence or noise substitution and packet repetition. The lost packets should be concealed as much as possible. For further research, the performance will be increased without introducing additional delay.

Conclusion

This survey provides an overview of existing approaches for packet loss techniques. Voice over IP is the new fancy development in the telecom 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. This paper surveyed packet loss techniques in voip. A variety of proposals for error recovery are reviewed. The proposed packet loss concealment algorithm gives a significant improvement in the quality of speech in voip. This mechanism is suitable for both unicast and multicast connections under all types of network conditions. The processing of damaged packets has been established as a suitable topic for further research. Packet loss tends to be a major cause of lost voice signals. It arises primarily from network congestion. Voice traffic can tolerate some packet loss. However, if the packet loss rate is greater than 5% it is considered harmful to the voice quality and a good concealment technique is required for reconstruction of the lost packets. In future an effort was directed to the development of a concealment algorithm that would maintain the quality of voice for lost packets.

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