# Speech Quality Requirements over DSL Networks

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***Abstract*— Quality of service (QoS) has been a feature of voice communication networks almost since their inception. The extension of traditional voice QoS methods to data communication networks and the Internet has been a longstanding research topic, although for many years it was not considered a critical issue due to the inherent differences between data and voice traffic and the relatively low cost of over provisioning bandwidth. In particular, over-provisioning of network bandwidth has been common practice since the earliest fiber-optic local-area networks (LANs) were deployed, and bandwidth on an optical network was found to be very economical. As VoIP moves from being an interesting and cheap application for enthusiasts to a public service for everybody, the speech quality requirements will be of increasing importance. There are a number of factors that contribute to the user perceived speech quality. Voice over a packet network may introduce new degradations such as packet loss, and increase other degradations such as delay. In this paper, we address Voice-over-IP (VoIP) as a trend of transferring voice over DSL access network and explaining the reasons of packet loss in speech signals and the solutions for preventing and recovering from packet loss. We concluded that retransmission of voice packets is not feasible for real-time applications like voice since they have very tight delay-bounds.**

***Keywords-com VoIP, QoS, DSL Networks, CODEC, FEC, Concealment, Jitter, DSLAM and ATM, Virtual Private Network (VPN), PPTP (VPN Protocol), IPSec (VPN Protocol), SIP (Signalling Protocol), H323 (Signalling Protocol), RTP (Real Time Transmission Protocol).***

## I. INTRODUCTION

VoIP has become very popular in recent years especially amongst corporate America and international callers because of its ease of use and low costs. VoIP is basically expressed by two computers calling each other and interpreting voice communications in the fastest and simplest way possible. The computer records a voice sampling and sends it at a much accelerated rate through the IP network to another computer where it is then played. The entire process is a bit more involved than explained here. To get the full gist of what makes VoIP possible, the technology behind it will be explained as follows: First, the computer must record the voice as sound samplings. If we try to download a music file from the internet, even over broadband, it can take a little while. To make the VoIP much faster than that, the computer uses CODECs to compress the sound samplings before sending them. Then, because only a fraction of the actual voice is sent and the rest is what the computer added, additional CODECs are used for better clarity of spoken words when played back. The first process is considered packetization. This is when the voice is recorded and compressed into several small packets. They are then collected and are prepared to be sent over the IP network. The time that it takes to send a single packet is somewhere between 10 and 30 milliseconds.

There is a possibility that some of the packets can be lost while the computer is trying to send all of them. The computer then does something called PLC, or packet-loss concealment. This is when the CODEC tries to replace the lost packets with acceptable audio. There are two other methods the computer may use to address the lost packets issue. One, it may send the packets more than once. This is called redundancy. The other method is to transfer information from the other packets to patch in the holes in the communication. This is done using mathematical operations and is called forward-error correction (FEC). VoIP is not without its glitches. Packets are sometimes delayed and/or aren't able to be played over the receiver's device. This results in choppy and unrecognized voice communications. Video is another form of communication that is able to be sent using VoIP. The transfer is basically the same as sending audio. The simple and powerful way to send video communication is the reason why VoIP is so attractive to so many people.

Quality of service is a very popular yet overloaded term in VoIP, which is very often looked from different perspectives by the networking and application developer's communities. In networking literature, QoS is quantified and measured by network-centric terms, such as throughput, end-to-end delay, bounds on delay and delay variation (jitter) or packet loss percentage and loss pattern. As a result, from a network engineering points of view, the design goal is to guarantee QoS by negotiating and assuring certain bounds of these metrics while at the same time trying to maximise network utilisation (which is usually translated to maximising revenue). In contrast, the view of QoS that application developers and application users have is more subjective: that of maximising the utility of the application. The term utility is an umbrella term which embraces perceived quality, that is, how pleasant or unpleasant is the presentation quality to the user of the application (i.e., visual quality of a displayed video sequence). Additionally, it may reflect the application's ability to perform its task (for example, in IP telephony if good conversation is achieved) or generate user interest (which in turn, may produce revenue - an important incentive). In certain occasions, QoS terms have been differently interpreted by different communities.

In networking, the term delay expresses the amount of time it takes for a data unit to propagate through the different paths of the network. For the application developer, i.e., a video codec designer, it is the time that is required for data to be encoded/decoded. It is very often the case that the two communities disregard the importance of this disparity in perspective. For example, until recently the image processing community considered that the underlying transmission infrastructure is providing a reliable transport medium, a circuit-switched equivalent, where the only delay was the propagation time and the losses were rare and corrected by the physical or data-link layer. Thus, they strived to maximise the quality of the encoded material by optimally selecting appropriate encoder/decoder parameters. In a non deterministic environment like the Internet, these assumptions do not hold. For example, packet loss may dramatically degrade the quality of the encoded stream and the perceptual distortion caused is usually far more significant to that introduced by encoding artifacts. It is imperative that these misconceptions are alleviated and that a mutual understanding of what quality stands for different communities is determined.

In this paper, we talk about VoIP quality of service over Digital Subscriber Line (DSL) where DSL is a collection of technologies used for the transmission of high-speed data over copper twisted-pair lines. It is used to connect the Network Service Providers (NSP) and the customers which are usually residences or small-to-medium sized businesses. At the customer’s home or office, the Customer Premises Equipment (CPE) provides access to the NSP’s network. The CPE connects to a DSL Access Multiplexer (DSLAM) located in the Central Office (CO) of the NSP. The DSLAM aggregates traffic from different customers and sends it over a high-speed uplink towards the core of the network as shown in Fig. 1. ADSL will play a crucial role over the next decade or more as telephone companies enter new markets for delivering information in video and multimedia formats. New broadband cabling will take decades to reach all prospective subscribers. Success of these new services will depend on reaching as many subscribers as possible during the first few years.

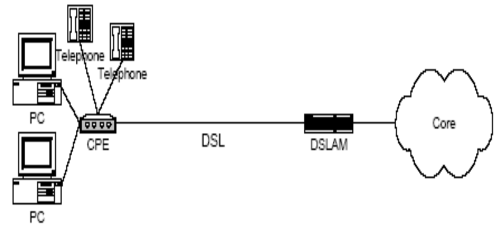


Figure 1. Topology of DSL Access Network

This paper is organized from eight sections described as follows: Section two discusses the DSL description with its all families. Section three presents IP Quality of service. In Section four, we will discuss VoIP QoS Issues and Compensation Process. Then, In Section five, we will describe the causes of Lost and Late Speech Packets .In Section six, we will provide packet loss recovery and error concealment. Section seven will presents numerical results and discussion. Finally, section eight covers summarization and conclusion.

## II. DSL Description

DSL is a telephone loop technology that uses existing copper phones lines, and provides a dedicated, high speed Internet connection. One of the big advantages of some DSLs (notably ADSL), are that they can co-exist on the same line with a traditional voice service such as "POTS" (Plain Old Telephone Service), and even ISDN. This is accomplished by utilizing different frequency ranges above the voice range (voice is up to 4 KHz). Essentially, this gives two lines in one: one for voice, and one for Internet connectivity. When all is working normally, there should be no interference between the two "lines". This gives DSL a potentially broad consumer base, and helps minimize costs for service providers. DSL is positioned for the Home and Small Office (SOHO) market that is looking for high speed Internet access at reasonable prices. Since it also typically provides dedicated, "always on" access, it can be used for interconnecting low to mid range bandwidth servers, and provides a great access solution for small LANs. The DSL provider (often, but not always, the phone company) will provide the DSL infrastructure. This would include your line, the DSLAM, and physical connection to the outside world. From there it is typically picked up by an ISP, who provides the traditional Internet services.

Consumer DSL plans are typically "best effort" services. While boasting speeds approaching T1, and even surpassing that in some cases, it is not necessarily as reliable as T1 however. Business class DSL offers more reliability at a higher cost than consumer plans, and is a good compromise where both reliability and bandwidth are at a premium. All in all, the cost of DSL compared to traditional telco services, such as T1, is attractive and substantially more affordable for home and small business users. DSL providers often do not have service contracts for home users, while business class DSL services typically do include similar SLAs (Service Level Agreements) to that offered for a T1 line.

### *A. The DSL Family*

- ADSL

Asymmetric Digital Subscriber Loop currently supports downstream rates up to 8 Mbps, and upstream of 1024 Kbps, hence the "asymmetric". ADSL is far and away the most widely deployed consumer DSL, and was specifically developed for the home and SOHO markets. The higher downstream rates lend it to those not running serious servers, at least anything more than a small, personal web site. ADSL is capable of sharing data with a POTS (or ISDN) voice line, so an additional line is not required. ADSL, like other DSLs, is limited by distance. 18,000 ft (5.5 km) is a typical cut-off point for telcos. ADSL does typically require either a splitter or filters to isolate the DSL signal from POTS. Sometimes referred to as "full rate" ADSL in order to differentiate it from G.Lite DSL. There are two line encodings for ADSL: DMT and CAP. DMT (a.k.a. Alcatel compatible) has won the standards battle and is now the standard and the most common. Also, note that modems must be compatible with the encoding. In other words, a CAP modem will not work with a DMT service, and vice versa. Also, ISDN requires "modems" (NTs), and related hardware such as filters, that are specific to that type of line since the signal on the line is very different for POTS and ISDN.

- G.Lite

G.Lite is sometimes referred to as "DSL Lite", "Universal DSL" or "splitterless ADSL", is a slower version of ADSL that requires no splitters or filters. G.lite uses a "fast retrain" technique to negate the various signal disturbances caused by normal POTS usage. Currently G.Lite supports speeds up to 1.5 Mbps/512 Kbps, and at one time was expected to become the dominant consumer DSL service.

- SDSL

Single-pair Digital Subscriber Loop, or also sometimes referred to as "Symmetric Digital Subscriber Loop" since it is indeed symmetric with a current maximum rate of 1.5 Mbps/1.5 Mbps. SDSL requires a dedicated line, and thus true SDSL is not as readily adaptable to the consumer market as ADSL. SDSL also uses a 2B1Q encoding (same as ISDN and some T1) which is considered more robust than the DMT or CAP encoding of ADSL. True SDSL is generally considered more of a server quality DSL, and is typically marketed as a business class service. It is worth noting that some providers may be promoting a "SDSL" service that is really ADSL pinched so that upstream/downstream are the same.

- IDSL

ISDN Digital Subscriber Loop, 144 Kbps/144 Kbps is really a new and improved ISDN from Lucent Technologies and uses the same 2B1Q line encoding as ISDN, SDSL and others. IDSL does require a dedicated line however. The benefits are that it is an "always on" technology, like other DSLs, and provides an additional 16 Kbps over traditional ISDN. It is being marketed by some DSL providers as a low end bit rate option, where line quality is not sufficient for higher speeds such as that of ADSL. Ironically, IDSL is generally priced significantly higher than ADSL.

- RADSL

Rate Adaptive Digital Subscriber Loop was developed by Westell and has a potential of 2.2 Mbps downstream and 1.0 Mbps upstream. What makes RADSL more flexible is that the sync rate can be dynamically adjusted up or down as line conditions change. This makes it more of a viable alternative where line conditions are marginal due to distance or other factors. In many respects, RADSL is an enhanced ADSL to meet a more diverse set of line conditions. Like ADSL, RADSL can piggyback on the POTS line. RADSL does not require any splitters or filters.

- HDSL

High bit-rate DSL was one of earliest versions of DSL. HDSL requires multiple, dedicated wire pairs, and is symmetric at 1.5 Mbps/1.5 Mbps (the speed actually depends on number of wire pairs used). Not a viable alternative for the consumer or SOHO markets.

- VDSL

Very high rate Digital Subscriber Loop is a DSL still in development with a current downstream capacity of 52.8 Mbps, and upstream of 2.3 Mbps. At this time, VDSL is limited to very short loop lengths, and is not yet a viable alternative. It may find application where there is fiber to the neighborhood, and thus the copper loop segment is relatively short.

- UDSL

Unidirectional Digital Subscriber Loop is a proposal from Europe that is not yet in use.

- G.SHDSL

The standards for G.SHDSL have just recently been finalized. SHDSL includes many enhancements, including better reach, better rate adaptation, and better upstream bandwidth. G.SHDSL is symmetric with speeds up to 2.3 Mbps, and will more than likely be marketed as an SDSL alternative.

## III. IP QUALITY OF SERVICE

Standard IP networks provide 'best effort' data delivery services by default. Time critical VoIP services require a significant change to the way IP networks are designed. The IP network design for the model shall have the following goals:

- **An end to end focus**; an understanding that different domains (LAN, Access & Core) have different characteristics and solutions.
- **A network that efficiently utilises the bandwidth available**; bandwidth is not free and will always have a cost associated with it.
- **A fully resilient network**; if toll quality voice services are to be delivered, the network will require full redundancy and availability.

- **Low end-to-end delay**; low delay variation and low packet loss ensuring these characteristics are met will ensure voice quality.
- **Delay**; End to end or 'Total Transmission Delay' is the sum of the compression, decompression, processing, buffering, queuing, transmission and the network delays. When this total delay exceeds a set amount (e.g. 150ms), the speakers experience problems on interactive dialogue.
- **Delay Variation** or '**Delay Jitter**'; is the variability in arrival time of a packet and when a packet does not arrive in time, it may have to be discarded. It cannot be re transmitted, as it would delay proceedings too much.
- **A jitter buffer**; in the egress MG exists to provide buffering of packets allowing time for late packets to arrive.
- **Packet Loss**; in some cases, packets may not reach their destination in IP networks; although the IP network can be well engineered it is difficult to totally eliminate this problem particularly in networks running multiple services (voice, video & data). Lost packets can also degrade the quality of the voice since parts of the original signal are lost. Lost packets can sometimes be 'concealed' in the MG by transmitting 'estimated' packets in their place. High packet loss (e.g. 1- 5%) can also adversely affect fax services. Traditional data services are normally unaffected by characteristics such as delay, delay variation and packet loss are not controlled by default. QoS mechanisms have to be employed to assure voice (& real time video services where implemented) will be delivered as a priority regardless of the current network conditions. There are two main methods to address these issues given by the following:-
- **Over provisioning;** over provisioning the bandwidth in the network to guarantee delivery.
- **QoS mechanisms;** the deployment of quality of service mechanism(s) within the IP network to guarantee delivery.

Purely over provisioning networks does not adequately address the voice quality issue on their own, for example TCP by nature is a bandwidth 'hungry' protocol will attempt to utilise all available bandwidth at a given moment. Over provisioning also requires very accurate network information and the ability to dynamically assign bandwidth instantly. Multiple QoS mechanisms are available today to address the demands of today's applications. The choice depends on factors such as the application requirements, network element functionality and domain. For instance, certain protocols are LAN specific, some focus on different characterises and some guarantee delivery (at the detriment to other possible business critical traffic). QoS protocols can be categorised mainly as:

- **Resource reservation**; network resources are apportioned according to an application's QoS request, and subject to bandwidth management policy. Some examples of resource reservation include RSVP and MPLS.
- **Prioritisation**; network traffic is classified and apportioned network resources according to bandwidth management policy criteria. To enable QoS, network elements give preferential treatment to classifications identified as having more demanding requirements. Some examples of prioritisation include Diffusive, LLQ (low latency queuing), CBWFQ (class-based weighted-fair queuing).

Critical to the success of the network is the correct choice and correct configuration of a QoS mechanism. Some QoS techniques are designed for less demanding applications than high quality real time voice so it is essential that each technique is examined in relation to the technical requirements of the network and for each application. In additional voice services require guaranteed delivery of call control and signalling packets as top priority (usually in order that network traffic can be removed gracefully in times of overload or operational difficulty).

## IV. VOIP QOS ISSUES AND COMPENSATION PROCESS

The advantages of reduced cost and bandwidth savings of carrying voice-over-packet networks are associated with some quality-of-service (QoS) issues unique to packet networks given by the following parameters:.

### *A. Delay*

Delay causes two problems: echo and talker overlap. Echo is caused by the signal reflections of the speaker's voice from the far-end telephone equipment back into the speaker's ear. Echo becomes a significant problem when the round-trip delay becomes greater than 50 milliseconds. As echo is perceived as a significant quality problem, voice-over-packet systems must address the need for echo control and implement some means of echo cancellation [1, 2]. Talker overlap (or the problem of one talker stepping on the other talker's speech) becomes significant if the one-way delay becomes greater than 250 milliseconds. The end-to-end delay budget is therefore the major constraint and driving requirement for reducing delay through a packet network.

- **Accumulation Delay (Algorithmic Delay)** This delay is caused by the need to collect a frame of voice samples to be processed by the voice coder. It is related to the type of voice coder used and varies from a single sample time (125 microseconds) to many milliseconds.
- **Processing Delay**: This delay is caused by the actual process of encoding and collecting the encoded samples into a packet for transmission over the packet network. The encoding delay is a function of both the processor execution time and the type of algorithm used. Often, multiple voice-coder frames will be collected in a single packet to reduce the packet network overhead.

- **Network Delay**: This delay is caused by the physical medium and protocols used to transmit the voice data and by the buffers used to remove packet jitter on the receive side. Network delay is a function of the capacity of the links in the network and the processing that occurs as the packets transit the network. The jitter buffers add delay, which is used to remove the packet-delay variation to which each packet is subjected as it transits the packet network. This delay can be a significant part of the overall delay, as packet-delay variations can be as high as 70 to 100 milliseconds in some frame-relay and IP networks.

- **Jitter**: Jitter buffer plays an important role in Voice over IP (VoIP) applications because it provides a key mechanism for achieving good speech quality to meet technical and commercial requirements [3]. The main objective of the work presented in is to propose a new, simple-to-use jitter buffer algorithm as a front-end to conventional static or adaptive jitter buffer algorithms to provide improved performance, in terms of enhanced user-perceived speech quality and reduced end-to-end delay. Supported by signal processing features, the new algorithm, the so-called Play Late Algorithm, alters the playout delay inside a speech talkspurt without introducing unnecessary extra end-to-end delay.

### *B. Lost-Packet Compensation*

Lost packets can be an even more severe problem, depending on the type of packet network that is being used. Because IP networks do not guarantee service, they will usually exhibit a much higher incidence of lost voice packets than ATM networks. In current IP networks, all voice frames are treated like data. Under peak loads and congestion, voice frames will be dropped equally with data frames. The data frames, however, are not time sensitive, and dropped packets can be appropriately corrected through the process of retransmission. Lost voice packets, however, cannot be dealt with in this manner. Loss occurs when networks drop voice packets. Lost-packet compensation means that since IP voice packets are UDP packets and that traditional IP networks have a tendency to lose packets when there is too much traffic for the amount of bandwidth available, the loss of a certain amount of voice packets will occur, to remedy this situation prioritization of the traffic has to be implemented in a way that data packets will be lost before any voice packets. Solutions like RSVP, Diffserv, Type Of Service (TOS) and Class Of Service (COS) bits and priority queuing are perfectly capable of effectively responding to the problem. Mayorga et al [4] first study the impact of packet loss in different transmissions with respect to different codecs and then propose reconstruction strategies to recover lost information.

#### *1) Interleaving*

This technique distributes the effect of the lost packets in order to reduce the impact on quality. 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 rearranged in their original form. Thus, instead of losing the whole packet small parts from distributed packets are lost.

#### *2) Repetition*

Lost packets are replaced by copies of last received packets.

#### *3) Simple Interpolation*

Consists of interpolating (averaging) by using the packets after and before the lost packet.

#### *4) Interleaving with Repetition*

The data are interleaved before sending and then any missing part is substituted using the repetition technique at the receiver.

#### *5) Interleaving with Interpolation Calculation*

The interleaving technique is used before sending and then the receiver interpolates to replace any missing parts in the jitter buffer.

### *C. Echo Compensation*

Echo in a telephone network is caused by signal reflections generated by the hybrid circuit that converts between a four-wire circuit (a separate transmit and receive pair) and a two-wire circuit (a single transmit and receive pair). These reflections of the speaker's voice are heard in the speaker's ear. Echo is present even in a conventional circuit-switched telephone network [4, 5]. However, it is acceptable because the round-trip delays through the network are smaller than 50 milliseconds and the echo is masked by the normal side tone every telephone generates. Echo becomes a problem in voice-over-packet networks because the round-trip delay through the network is almost always greater than 50 milliseconds.

## V. Causes of Lost and Late Speech Packets

Packet loss occurs as a result of buffer overflow(s) at network nodes due to heavy loads or bit errors incurred by packets during transit. There have been numerous studies on Internet packet loss statistics including. Packet loss is known to have some correlation with packet size, time of day (congestion), and network delay. According to , packet loss of 10% is unexceptional, and losses of up to 40% are possible on the Internet Internet packet loss is bursty and correlated meaning that if packet n is lost then there is high probability that packet n+1 will be lost . Packet loss is a normal phenomenon on packet networks. Loss can be caused by many different reasons: overloaded links, excessive collisions on a LAN, physical media errors and others. Transport layers such as TCP account for loss and allow packet recovery under reasonable loss conditions.

Audio CODECs also take into account the possibility of packet loss, especially since RTP data is transferred over the unreliable UDP layer. The typical CODEC performs one of several functions that make an occasional packet loss unnoticeable to the user. For example, a CODEC may choose to use the packet received just before the lost packet instead of the lost one, or perform more sophisticated interpolation to eliminate any clicks or interruptions in the audio stream. However, packet loss starts to be a real problem when the percentage of the lost packets exceeds a certain threshold (roughly 5% of the packets), or when packet losses are grouped together in large packet bursts. In those situations, even the best CODECs will be unable to hide the packet loss from the user, resulting in degraded voice quality. Thus, it is important to know both the percentage of lost packets, as well as whether these losses are grouped into packet bursts. When the RADCOM AudioPro analyzes audio streams, it provides both top-level statistics as well as drill-down analysis of individual packet loss

## VI. PACKET LOSS RECOVERY AND ERROR CONCEALMENT

A recovery process can be divided into two stages: loss recovery and error concealment [6, 8] Loss recovery is to recover the original content of a lost packet. Loss recovery can only recover a single lost packet, or work well under special network scenario. Recent researches show that packet loss can exhibit temporal dependency or bursts, which degrade its effectiveness [5]. Therefore, error concealment is needed to conceal the remaining loss in voice streams after loss recovery. Error concealment complements loss recovery.

### *A. Loss Recovery*

Loss recovery mechanisms may be split into two major classes: active retransmission and passive channel coding [9]. Retransmission increases the latency of packets and may not be suitable for VoIP. Passive schemes mainly use forward error correction (FEC). FEC adds redundancy information into voice streams for aiding the loss correction. FEC can be either media-independent or media-dependent. See Fig.2.

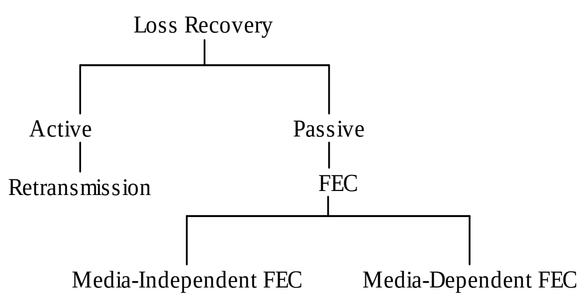


Figure 2. Loss Recovery Approaches

Media-independent FEC uses voice block, or algebraic, codes to produce additional repair packets. Parity coding and Reed-Solomon coding are two common schemes of block coding. They are relatively simple and easy to implement. The disadvantages are the additional delay imposed increased bandwidth, and difficulty in decoder implementation.

### *B. Error Concealment*

Error concealment schemes produce a replacement for a lost packet, which is similar to the original lost packet. This is possible because voice signals exhibit large amounts of short-term self-similarity. Depending on interactions with source encoding schemes, error concealment schemes can be divided into source-coder independent and source-coder dependent schemes, the techniques are listed in Fig. 3. The former does not exploit the knowledge of the underlying coding algorithms, and only creates a replacement by simple interpolation. The latter regenerates a replacement by exploiting features in individual coders [10, 11].

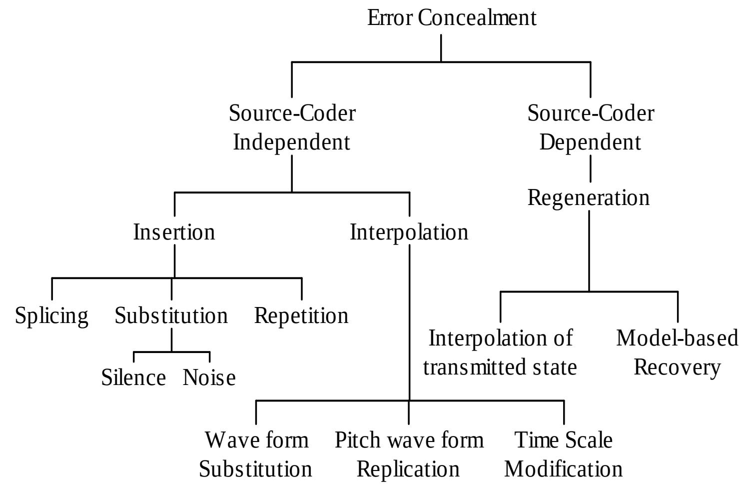


Figure 3. Error Concealment Schemes

Insertion-based repair schemes derive a replacement by inserting a simple fill-in, i.e. silence, background noise, comfort noise, or repeating packet of the last received packet. The simplest method is to splice together the voice on either side of the loss. Splicing disrupts the timing of the stream and is not an acceptable repair technique. The implementation of these schemes is simple. All of them but repetition generally result in poor performance.

### *C. Forward Error Correction and Concealment*

This approach [12] encompasses both loss correction and loss concealment algorithms. Loss Correction uses media-dependent and media-independent Forward Error Correction (FEC) techniques. FEC constitutes adding redundancy data to the normal voice stream to protect from packet loss. FEC introduces overhead in terms of the total amount of traffic on the network, but if the amount of redundancy is controlled then this approach can be used. In media-independent FEC, general protection codes like Reed Solomon or Viterbi are used to produce an extra protection packet that follows the protected set of voice packets. These codes do not depend on any particular underlying media characteristics, but introduce a higher delay which may not be tolerated by many applications, including VoIP. In media dependent FEC, the sender uses a high-quality codec to create the Voice samples and a lower quality codec to generate redundant bits that are added to every packet rather than being sent in their own separate packet. The receiving codec removes the redundancy. If the receiver must use that redundant data to substitute for a lost packet, the result is a lower-quality (but not missing) segment of voice. It introduces minimum delay but may introduce more computational processing delay as it is media dependent. Concealment techniques can be used to supplement FEC for even better lost-packet compensation. The most common concealment approaches include:

- Silence substitution is substitution of the lost frame by a silence frame of the same temporal frequency, but it can introduce noise if several of them are introduced.
- In noise substitution Gaussian noise frames are used to substitute for the missing frames. This produces better quality.
- In frame replication, missing frames are replaced by already present redundancy in the voice. This has low computational complexity and is efficient as more redundancy is expected to be present in the neighboring voice frames. It does not need large temporal size.
- Waveform substitution uses the frames prior to the lost frames and tries to use the most recent ones. It examines buffered frames and searches for the best match.

## VII. Numerical Results and Discussion

In overall perspective, usually the assessment of VoIP is carried out using the subjective quality measure called Mean Opinion Score (MOS) [13, 14]. MOS scales from 1 (lowest quality) to 5 (highest quality). However, this type of measurement focuses on the perceived quality provided by users. MOS is useful when considering the overall end-to-end quality of communications. In the research work [13], the assessment using MOS was not being adopted, since the focus is towards measuring the performance of VoIP communications with regards to the network environment. Based on their experiments, the delay and jitter values have been organized in the form of histogram charts, representing the number of packets versus the delay or jitter values. This shows the number of RTP packets experiencing the same jitter values or delay. It was noted that these values do not represent any real progression. Data from their study covers these parameters: delay, jitter values, and packet loss rate [13]. The delay values were obtained by calculating the difference between the RTP packet actual arrival time and the estimated arrival time. On the other hand, the jitter values [13] have been derived from the differences in the inter-arrival time of the RTP packets. The packet loss values are represented in the percentage form of the total RTP packets being transmitted.

### *A. Ideal Network Environment*

For the ideal Ethernet LAN Environment, the default queue algorithm used was the First-in-First- Out (FIFO). Based on the results obtained, it demonstrates that Both PPTP and IPSec incur a higher delay as compared to normal VoIP communication within the ideal network. This is true for both H.323 and SIP based VoIP communications. On the other hand, the jitter values for SIP and H.323 VoIP communications remained generally similar to each other. The packet loss was not considered as a factor within the ideal and ideal secure environments owing to a very low bit error rate of the lab network [13].

### *B. Non-Ideal Ethernet Network Environment*

The ideal network was then converted into a non-ideal one; with congestion and delays. In this regard an advanced queue discipline was implemented in the gateways. Token Bucket Filter (TBF) was used for the queue management. The advantage of using TBF queue discipline is such that it provides the condition where packets would be heavily queued at one end and will be burst out and some of the packets might be dropped due to the limitation of the bucket size. Table 1 shows the parameter values that have been implemented in the TBF queue [13].

TBF Parameters that have been adopted [13].

| Parameters | Description | Values selected |
|---|---|---|
| **Bucket/Burst** | This is the size of the bucket. Indirectly, it is also considered as the burst size, since the queue will be burst when the bucket is full | 1024 kbyte |
| **Latency** | The amount of time in which a particular packet is allowed to reside in the TBF bucket | 100 ms |
| **Rate** | Rate of the arrival for tokens. | 50 kbps |

According to [15], traffic shaping (as being conducted in this research) could also be used as a mean to reduce the impacts of interfering bursts on network performance. As was expect, both SIP and H.323 incurred higher delays in non-ideal secure network-to-network (more so under the IPSec) environment, with respect to the maximum RTP packets distribution. Figure 4 and 5 show the average delay values for both SIP and H.323 VoIP communications.

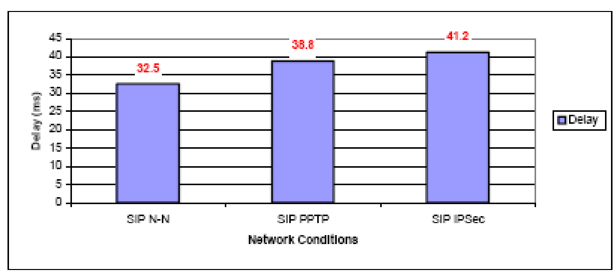


Figure 4. Comparison of Average Delays for RTP Packets transmission in SIP VoIP.Communications (Non-Ideal Network-to-Network) [13].

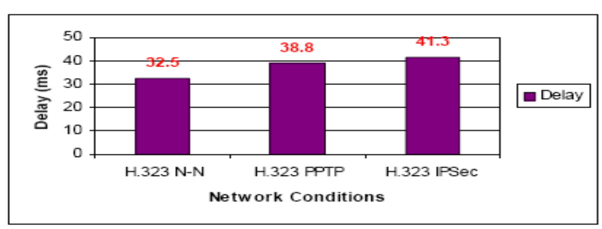


Figure 5. Comparison of Average Delays for RTP Packets transmission in H.323 VoIP. Communications (Non-Ideal Network-to-Network) [13].

It was noted that the results from the delay analysis shows lower values as compared to the assessment done by [16] over the real Internet backbones. This is considered as acceptable, since the analysis that has been carried out was done in a controlled lab environment, with only one single VoIP communication being conducted. According to [14], the delay values between 100 – 150 ms and above are detectable by humans and can impair the interactivity of the conversations. The results obtained are far less than those values. However, the analysis has been done in the controlled environment, unlike the real environment where the impact of delay and jitter are more severe. In analyzing the jitter values, the overall results obtained showed [13] that SIP exhibits higher jitter values as compared to H.323 VoIP communications in all the cases. Figure 6 illustrates the comparison made between H.323 and SIP based VoIP communications in a non-ideal networking environment. The introduction of IPSec and PPTP increased the jitter values for both H.323 and SIP based VoIP communications. In this study, IPSec produced the highest jitter values for both H.323 and SIP communications. Figure 7 shows the average jitter values for both H.323 and SIP.

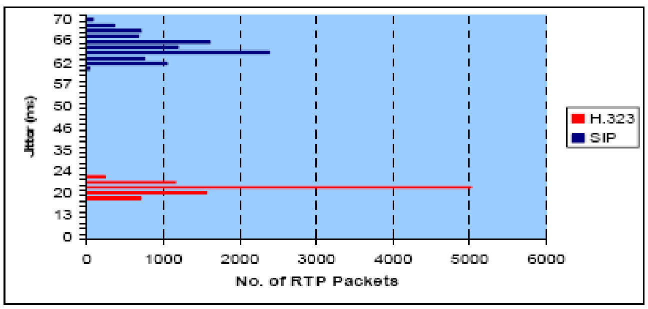


Figure 6. Comparison of Jitter Values for RTP Packets between H.323 and SIP VoIP.Communications (Non-Ideal Network-to-Network) [13].

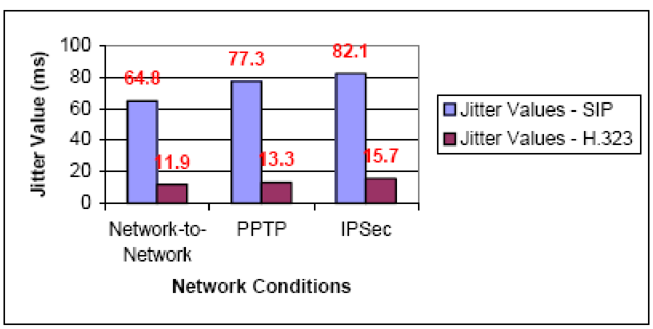


Figure 7. Comparison of Average Jitter Values for RTP Packets of SIP and H.323 VoIP. Communications for Non-Ideal Secure Ethernet Environment [13].

In relation to the packet loss analysis, the non-ideal and non-ideal secure network-to-network environments consistently produced relatively high packet loss rates. Packet loss rate is calculated by determining the number of RTP packets that are lost (unreachable to the destination) over the number of RTP packets being transmitted.

## VIII. Summery and Conclusion

In this paper, we have discussed QoS of VoIP across DSL networks. While much progress has been made in VoIP related QoS mechanisms, several issues remain to be addressed in order to fully deploy the technology: some mechanisms in data plane like loss packets and error concealment require further refinements for a scalable and efficiency implementation of QoS guaranteed VoIP system. Packet loss concealment can also be used at the receive end and this minimizes the impact of lost packets on the speech signal by mixing in synthesized speech derived from previous packets. Effective packet loss concealment has a large impact on user satisfaction and is necessary for achieving acceptable IP speech quality.

VoIP has established itself within the voice & data industries. Standards have matured to the point where robust, scalable and reliable products can be readily integrated within existing networks. VoIP brings together the best of the internet and voice worlds creating an environment that facilitates high value, cutting edge, mobility enabled, converged communications services. There is still however a significant amount of activity within standards organisations to improve and develop existing protocols particularly in the SIP arena. This activity is considered a refreshing change to the stagnation that existed with the development of voice services prior to VoIP. The deployment of VoIP and converged networks can deliver significant capital and operational cost savings in a scalable network, but care must be taken to properly deploy QoS to deliver sustained performance. A combination of bandwidth over-provisioning with QoS mechanisms such as Ethernet precedence or IP ToS should be considered, using networking equipment with advanced features such as rate-adaptive voice-encoding algorithms, access control lists enabled in dedicated hardware, and VLANs implemented in dedicated ASIC hardware. Either coarse- or fine-grained QoS queues may be employed depending on the application, although a full eight queues per port is the most flexible. Assignment of minimum and maximum bandwidth for each queue and providing access control lists on the network edge are useful practices to guarantee proper implementation of the QoS policy.

Only with a comprehensive knowledge of convergent technology and real implementation experience can a company generate accurate business cases and provide the highest probability of success for implementation. Careful design and integration can reduce the risk and delay with deploying IPT and VoIP solutions particularly as every user of the technology will have their own requirements with respect to services, legacy infrastructure, and operational support arrangements. Almost all businesses now agree that the benefits brought about by convergence are too good to ignore realising that it is not a question of 'if' but 'when' to invest in converged technology. Some businesses are delaying IT and telecoms investment and are preparing IP infrastructures in anticipation for a gradual migration process. Other more shrewd businesses recognise the importance of competitive advantage and that the benefits of deploying converged solutions today can open up exciting new opportunities, and deliver true value from their investments.

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