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Packet loss concealment in voice over the Internet

Rishikesh S. Gokhale
University of South Florida

Packet Loss Concealment in Voice Over Internet

by

Rishikesh S. Gokhale

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Department of Electrical Engineering
College of Engineering
University of South Florida

Major Professor: Wilfrido A. Moreno, Ph.D.
James T. Leffew, Ph.D.
Wei Qian, Ph.D.

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IN VOICE OVER INTERNET

Rishikesh S. Gokhale

ABSTRACT

Traditional telephony networks with their cumbersome and costly infrastructures are being replaced with voice being transmitted over the Internet. The Internet is a very commonly used technology that was traditionally used to transmit data. With the availability of large bandwidth and high data rates the transmission of data, voice and video over the Internet is gaining popularity. Voice is a real time application and the biggest problem it faces is the loss of packets due to network congestion. The Internet implements protocols to detect and retransmit the lost packets. However, for a real time application it is too late before a lost intermediate packet is retransmitted. This causes a need for reconstruction of the lost packet. Therefore, good reconstruction techniques are being researched. In this thesis a new concealment algorithm to reconstruct lost voice packets is reported. The algorithm is receiver based and its functionality is based on Time Scale Modifications of speech and autocorrelation of a speech signal. The new technique is named the Modified Waveform Similarity Overlap Add, (WSOLA) technique. All simulations were performed in MATLAB

CHAPTER 1

INTRODUCTION

The transmission of voice over packet switched networks such as an Internet Protocol (IP) network, like the Internet, is presently an area of active research. Much of the past work has focused on using packet switching for both voice and data in a single network. Renewed interest in packet voice and more generally packet audio applications has been fuelled by the availability of supporting hardware, increased bandwidth throughout the Internet and the desire to integrate data and voice services in the networks.

The motivation for transporting voice over IP networks is the potential cost saving, which are achieved by eliminating the circuit-switched telephony infrastructure. PC based programs such as Free Phone and MSN messenger services have demonstrated the feasibility of voice transport over the Internet. These successes have stirred a desire for wider deployment of VoIP.

1.1 Layered Model of the Internet

As is common with many communications systems, the protocols involved in Voice over IP, (VoIP), follow a layered hierarchy. The hierarchy follows from and can be compared with the theoretical model developed by the International

Standards Organization, (OSI seven layer model). Breaking a system into defined layers can make that system more manageable and flexible. Each layer has its separate functions and does not require detailed data or information from the layers around it. For example, IP datagrams can be transported across a variety of link layer systems including serial lines (using PPP), Ethernet and Token Ring. The link layer protocol, for the most part, is irrelevant to IP unless the protocol limits the size of its datagram's. Additionally, the link layer protocol is incompatible with IP since there is no need to be the same for the first link of a VOIP call and the final link of a VoIP call. As always there are exceptions such as IP over an ATM where the simple discreet layered model is considered.

The effect of each layer's contribution to the communication process is an additional header preceding the information being transmitted. The complete packet, which a layer creates, header and data, becomes the data passed to the next level for processing. Each layer adds an additional header portion as the message progresses. To illustrate, two basic layers, the Network layer and the transport layer along with the protocols used at these layers for voice transmission, are discussed.

1.1.1 Network Layer

The Internet Protocol is the lowest level protocol considered in this document. It is responsible for the delivery of packets between host computers. IP is a connectionless protocol. Therefore, it does not establish a virtual connection through a network prior to commencing transmission, which is a job for higher-level protocols. IP makes no guarantees concerning reliability, flow control, error detection or error correction. The result is that datagrams can arrive at the destination computer out of sequence, with errors or not even arrive at all. Nevertheless, IP succeeds in making the network transparent to the upper layers involved in voice transmission through an IP based network.

Any Voice over IP transmission must use the Internet Protocol (IP). However, IP is not well suited to voice transmission. Real time applications such as voice and video require a guaranteed connection with consistent delay characteristics. Higher layer protocols address these issues.

Figure 1.1 presents the structure of the datagram header that precedes the data to be transmitted. In its most basic form, the header is comprised of 20 octets. There are optional fields that can be appended to the basic header that offer additional capabilities. However, they are not relevant to the VoIP transmission studied in this research.

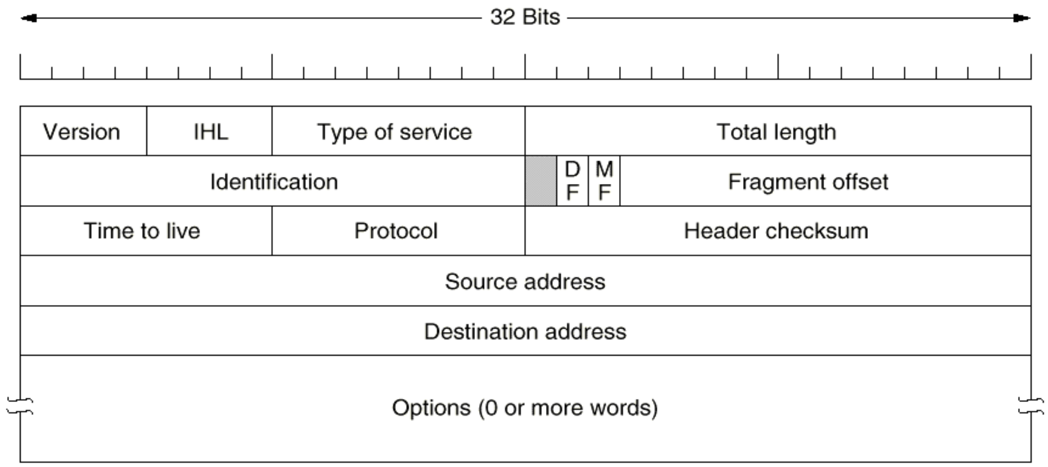


Figure 1.1: Network Layer Datagram

The various fields of the datagram serve specific purposes and provide specific information.

1. Version: The version of IP being used. For this format header the version would be IPv4.
2. IHL: The length of the IP header in units of four octets (32 bits). For the basic header shown in Figure 1.1, the value would be 5 since each line in the diagram represents four octets.
3. Type of service: Specifies the quality of service (QOS) requested by the host computer sending the datagram. Routers or Internet Service Providers do not always effectively support QOS.
4. Total length: The length of the datagram, measured in octets, including the header and payload.

5. Identification: As well as handling the addressing of datagrams between two computers or *hosts*, IP needs to handle the splitting of data payloads into smaller packages. This process, known as *fragmentation*, is required since lower link layer protocols such as Ethernet cannot always handle large packet sizes even though a single IP datagram can handle a theoretical maximum length of 65,515 octets. This field is a unique reference number assigned by the sending host to aid in the reassembly of a fragmented datagram.
6. Flags: Flags indicate whether the datagram may be fragmented and if it has been fragmented, whether further fragments follow the current fragment.
7. Fragment offset: This field indicates where this fragment belongs in the datagram. It is measured in units of 8 octets or 64 bits.
8. Time to live: This field indicates the maximum time the datagram is permitted to remain in the Internet system. This parameter ensures that a datagram that cannot reach its destination host is given a finite lifetime.
9. Protocol: This field indicates the higher-level protocol in use for this datagram. Numbers have been assigned for use with this field to represent such transport layer protocols as TCP and UDP.
10. Header checksum: This is a checksum covering the header only.

11. Source address: The IP address of the host that generated this datagram. IPv4 addresses are 32 bits in length. When written or spoken a *dotted decimal notation* is used (e.g.: 192.168.0.1).
12. Destination address: The IP address of the destination host. This is the last field of the datagram.

1.1.2 Transport Layer

Generally, there are two protocols available at the transport layer when transmitting information through an IP network. These are the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP). These protocols enable the transmission of information between the correct processes or applications on host computers. These processes are associated with unique port numbers. For example, the HTTP application is usually associated with port 80.

TCP is a connection-oriented protocol. Therefore, TCP establishes a communications path prior to transmitting data and handles sequencing and error detection, which ensures that a reliable stream of data is received by the destination application.

Voice is a real-time application and mechanisms must be in place to ensure that information is reliably received in the correct sequence and with predictable delay characteristics. Although TCP would address these requirements to a certain extent, there are some functions, which are reserved for the layer above TCP. Therefore, for the transport layer, TCP is not used and the alternative protocol, UDP, is commonly used.

In common with IP, UDP is a connectionless protocol. UDP routes data to its correct destination port but does not attempt to perform any sequencing or ensure data reliability. Figure 1.2 presents the structure of the transport layer datagram.

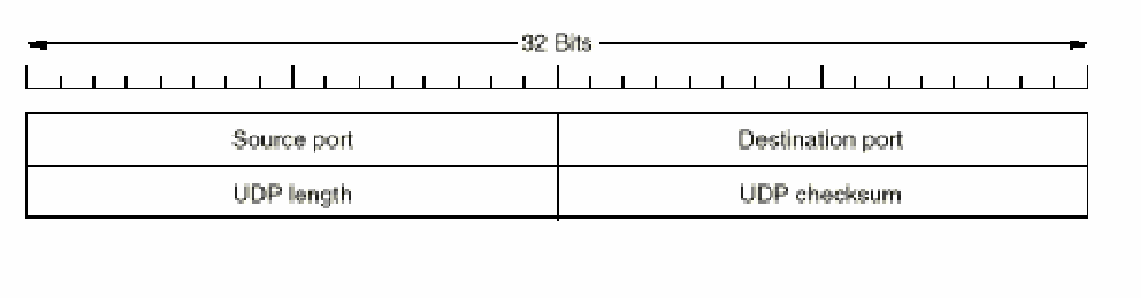


Figure 1.2: Transport Layer Datagram

The four fields of the transport layer datagram serve specific purposes.

1. Source port: Identifies the higher layer process, which originated the data.
2. Destination port: Identifies the higher layer process that will receive this data.
3. Length: The length in octets of the UDP data and payload. The minimum length is eight (8) octets.
4. Checksum: Optional field supporting error detection.

1.2 Motivation

In a Voice over IP, (VoIP), application, the voice is digitized and packetized at the sending facility at regular intervals, (e.g., every 10 ms), using an encoding algorithm. Then the voice packet is sent over the IP network to the receiver where it is decoded and played-out to the listener.

These voice packets are typically transported over an IP using the User Datagram Protocol (UDP). The UDP, unlike the Transmission Control Protocol, (TCP), does not have provisions for retransmission of lost packets. Lost packets are packets that do not arrive at the proper time or at any receiver. For this reason, UDP is characterized as a send and pray, (SNP), protocol. IP networks such as the Internet are inherently best effort networks with variable delay and loss.

The question is often asked. “Why not use TCP instead of UDP for the transmission on voice packets?” The simple reason is that, in the case of voice packets, “never” is significantly better than “late” for lost packets. By the time a lost packet is detected and retransmitted the delay is more than sufficient to render the voice packet useless. Therefore, a good concealment algorithm needs to be designed for lost packets.

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 this research an effort was directed to the development of a concealment algorithm that would maintain the quality of voice for lost packets.

Chapter 2 details the process of pitch detection, which is essential for a good concealment algorithm. Afterwards, some existing concealment techniques are explored. Then the new algorithm for packet loss concealment is introduced. Test results for the algorithm are given in the last chapter.

CHAPTER 2

PITCH DETECTION

The main work of this research consisted of developing an improved packet loss concealment algorithm based on time-scale modifications of speech. Existing time-scale modification algorithms did not take the pitch period of the speech signal waveform into consideration. As will be shown later, taking into consideration the pitch of the speech signal and modifying the existing time scale yields a much better quality for the reconstructed speech signal. Thus pitch detection for the signal is important. This chapter presents and explains an algorithm for pitch detection.

2.1 Introduction

A speech signal is passed through a low pass filter before pitch detection is performed. The low pass filtered speech signal is sampled at 8KHz and then quantized using a 16-bit quantizer. The digitized speech signal X(n) is processed as 20ms frames and a Linear Predictive Coding (LPC) error signal e(n) generated using a 10th order LPC analyzer. The signal, which is sampled at 8KHz and processed as 20ms frames, yields 160 sample points per frame.

2.1.1 Quantization

Sampling takes a snapshot of the input signal at an instant of time. When the snapshot is taken the sampled analog value must be converted to a binary number. The conversion from infinitely precise amplitude to a binary number is called quantization. During quantization the analog to digital converter uses a finite number of evenly spaced values to represent the analog signal. The number of bits used for the conversion determines the number of different values possible. Most modern converters use 12 or 16 bits. Typically, the converter selects the digital value that is closest to the actual sampled value. In Matlab a function exists for implementing quantization. The function is named "Quant" and digitizes values as multiples of a quantity. The syntax of "Quant" is quant(x, q), where x and q are the inputs to the function. The variable x is a, scalar or vector, matrix and the variable q is the minimum value. For example, if

$$x = [1.333 \quad 4.756 \quad 3.897] \tag{2.1}$$

and

$$y = \mathrm{quant}(x, 0.1) \tag{2.2}$$

then

$$y=[1.3 \quad 4.8 \quad 3.9]. \tag{2.3}$$

Thus x is rounded to the nearest multiple of q.

2.1.2 Linear Prediction Coefficients (LPC)

LPC is a method of separating out the effects of the source from a speech signal. LPC can be thought of as a way of encoding the information in a speech signal into a smaller space for transmission over a restricted channel. LPC encodes a signal by finding a set of weights for earlier signal values that can predict the next signal value. The next signal value is given by

$$y[n] = a[1]y[n - 1] + a[2]y[n - 2] + a[3]y[n - 3] + e[n]. \tag{2.4}$$

If values for a[1], a[2] and a[3] can be found such that the error signal e[n] is very small for a segment of speech, (for example, one frame), then only a[1], a[2], a[3] need to be transmitted instead of the signal values in the window. The speech frame can be reconstructed at the other end by using a default e[n] signal and predicting subsequent values from earlier ones.

A function exists in Matlab for implementing LPC. The function is named "lpc". The function and the syntax is given by

$$A = LPC\ (X,N), \tag{2.5}$$

where X is the signal whose linear prediction coefficients needs to be found and N is the order or the number of coefficients. A is represented by

$$A = [1, A(2), ..., A(N+1)]. \tag{2.6}$$

The pitch information is present in both the original digitized signal X(n) and the error signal e(n). Therefore, pitch detection is performed on both signals. Typically, any periodicity that appears in the original signal X(n) also appears in the error signal. However, as shown in later examples, several cases exist where pitch detection needs to be performed on both signals.

As shown in Figure 2.1, a particular formant structure in the waveform causes the periodicity of the waveform to be obscure. When a person is speaking the variations produced in the speech signal by acts such as opening the teeth or rounding the lips causes the frequency response of the speech signal to have several peaks. These peaks are known as formants. Since the LPC residual signal e(n) represents the speech waveform with the formant structure removed, pitch detection performed on e(n) provides a correct estimate of the pitch.

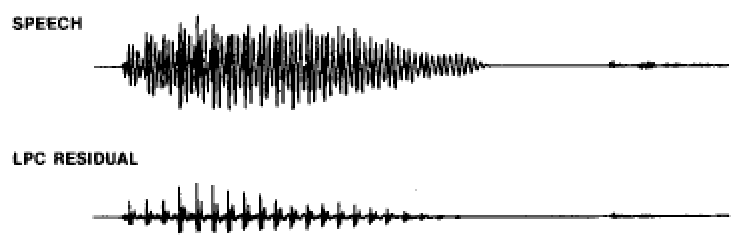


Figure 2.1: Speech Waveform Shows Lack of Clear Periodicity Due to Formants

Another case arises when the residual waveform fails to show clear periodicity in voiced frames. This condition is presented in Figure 2.2. Such a situation occurs when the fundamental frequency of the excitation information, which is found in the residual, is removed by LPC inverse filtering. The inverse filtering causes the residual to look noisy while the original speech signal appears to be clearly periodic.

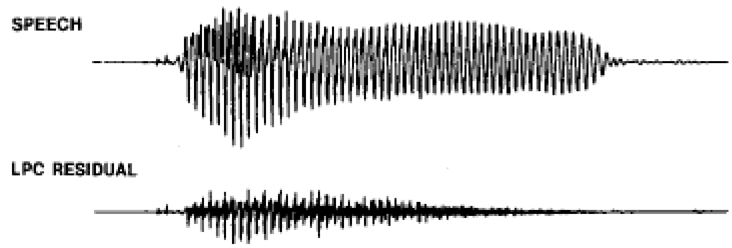


Figure 2.2: LPC Residual Shows Lack of Clear Periodicity

Once the LPC error signal is generated, the LPC error signal e(n) and the digitized speech signal X(n) is split into positive going and negative going signals. The resulting four signals are positive going X(n), negative going X(n), positive going e(n) and negative going e(n). These signals are named fa(n), fb(n), fc(n) and fd(n) respectively. Pitch detection analysis is performed on each of these signals individually by four pitch detectors that operate in parallel. The structure of the pitch detectors is identical. The pitch detector structure is described in the next section and differences occur only in the values of their control parameters. The pitch voter combines the four pitch detection estimates to produce a final pitch estimate. Figure 2.3 presents a block diagram of the entire process.

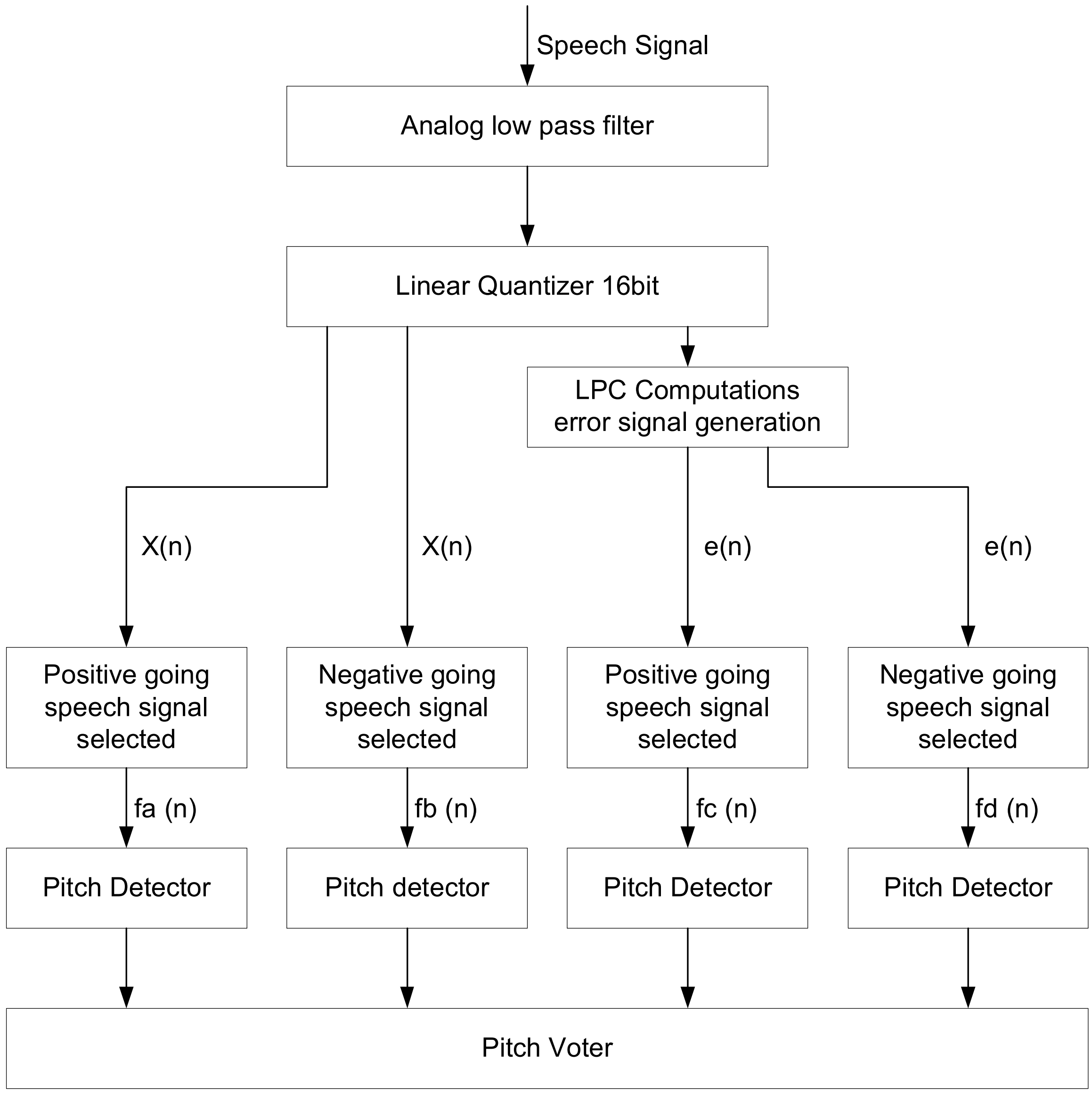


Figure 2.3: Block Diagram of the Pitch Detector

2.2 Pitch Detector

The pitch detector is responsible for detecting the pitch of the voice signal. In simple terms the pitch is nothing but the time period of the signal. The pitch detection process is divided into four steps.

1. Find a set of candidate pulses.
2. Find a subset of the set of candidate pulses such that a candidate distance (DC) separates all the selected pulses.
3. Perform linear interpolation on the selected pulses.
4. Perform a Pitch consistency test.

Each of these steps is described along with a flowchart and an algorithm of how this process is implemented in Matlab.

2.2.1 Finding a Set of Candidate Pulses

The operation starts by identifying a set of samples, called a Candidate Pulse Set, over a frame on which the pitch or periodicity is to be detected. In order to find these pulses the global maximum amplitude, M0, is found. M0 is the sample or pulse that has the highest amplitude among all the samples in the frame. Its location within the frame is D0. This global maximum is the first sample that enters the set of candidate pulses. All pulses selected after M0 must satisfy three conditions.

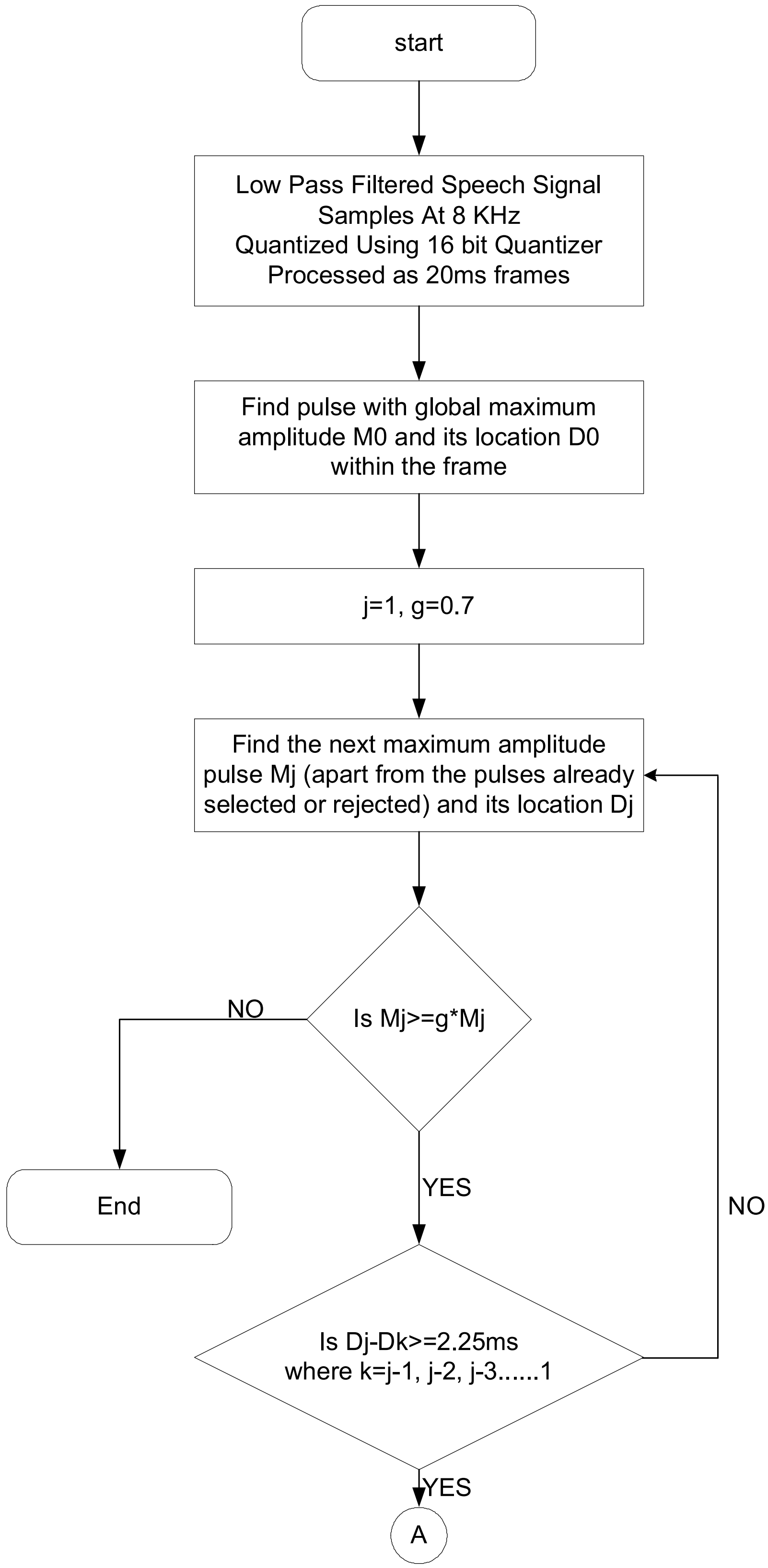
1. First: The next pulse selected must be a local maximum, which means it must have the maximum amplitude after excluding the pulses that have already been selected. This selection is reasonable since pitch pulses normally have amplitudes higher than any other pulses in the frame. Mj denotes the amplitude of this local maximum and its location within the frame is denoted by Dj.

2. Second: Any selected pulse must have amplitude at least equal to a fraction of the global maximum amplitude M0. That is

3. $Mj >= g*M0,$ (2.7)

   where g is called the threshold amplitude percentage. The value of g is normally set between 0.175 and 0.525 for a good pitch estimate.

4. Third: All the selected pulses must be separated by at least 2.25ms, which is 18 sample periods from all other selected pulses. The reason for including this condition is that the largest human speech frequency encountered is 400Hz. A frequency of 400Hz corresponds to a time period of 2.55ms. Therefore, the smallest human speech pitch is 2.55ms. If a small tolerance level, of approximately 10%, is allowed, it is only necessary for the selected pulses to be separated by 2.25 ms. Figure 2.4 presents a block diagram of the entire process.


start
Low Pass Filtered Speech Signal
Samples At 8 KHz
Quantized Using 16 bit Quantizer
Processed as 20ms frames
Find pulse with global maximum
amplitude M0 and its location D0
within the frame
j=1, g=0.7
Find the next maximum amplitude
pulse Mj (apart from the pulses already
selected or rejected) and its location Dj
Is Mj>=g*Mj
NO
End
YES
NO
Is Dj-Dk>=2.25ms
where k=j-1, j-2, j-3......1
YES
A

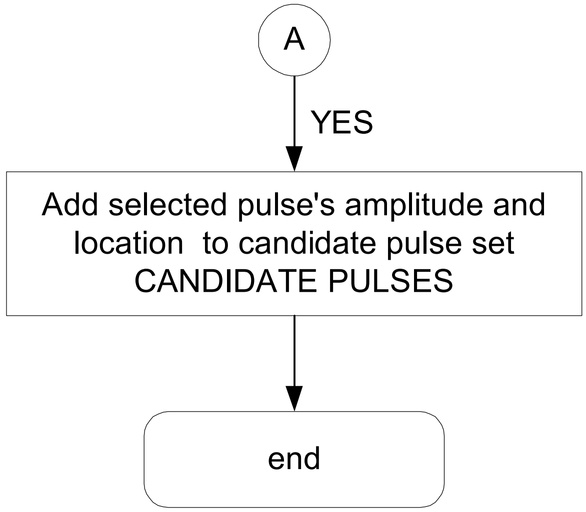


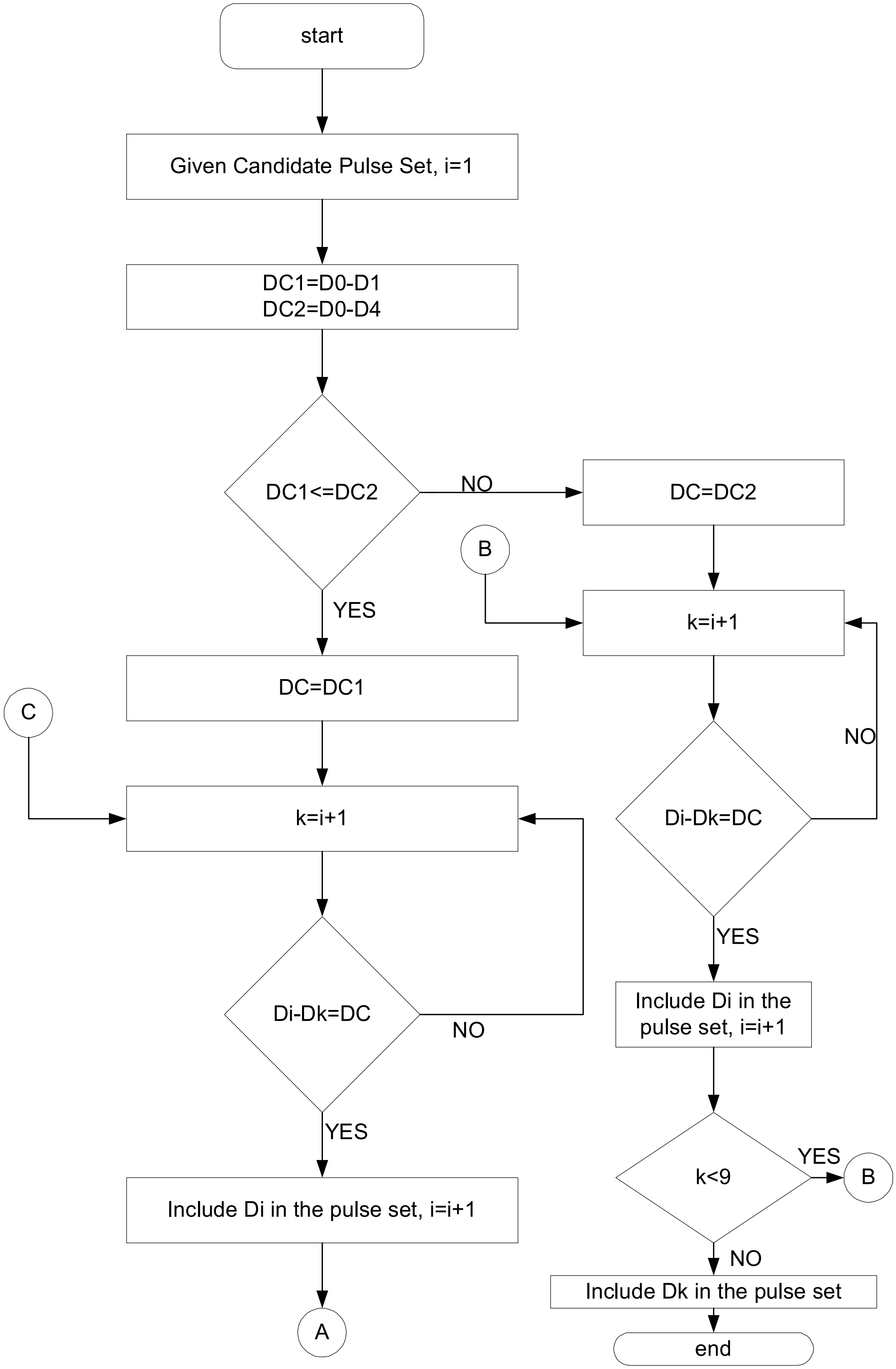
Figure 2.4: Flowchart For Finding a Set of Candidate Pulses

2.2.2 Finding a Subset of the Set of Candidate Pulses

The candidate pulse set consists of pulses with amplitudes Mj and locations Dj. These amplitudes and their locations are used to find a distance that is the smallest distance over which a subset of these pulses is periodic. The periodic distance is determined recursively by considering the distance from the global framing maximum M0 to the closest adjacent pulse. This distance is called the candidate distance (DC) and is given by

$$DC = |D0 - Dj| \tag{2.8}$$

If this distance does not separate a subset of maxima in the frame, plus or minus a breathing threshold B, then the candidate distance is discarded and the process begins again with the next closest adjacent candidate pulse. Figure 2.5 flowcharts the process of finding a subset of the candidate pulse set. Figure 2.6 presents an example set of candidate pulses.


start
Given Candidate Pulse Set, i=1
DC1=D0-D1
DC2=D0-D4
DC1<=DC2
NO
DC=DC2
YES
DC=DC1
B
k=i+1
C
k=i+1
NO
Di-Dk=DC
YES
Include Di in the pulse set, i=i+1
Di-Dk=DC
NO
YES
Include Di in the pulse set, i=i+1
k<9
YES
B
NO
Include Dk in the pulse set
end
A

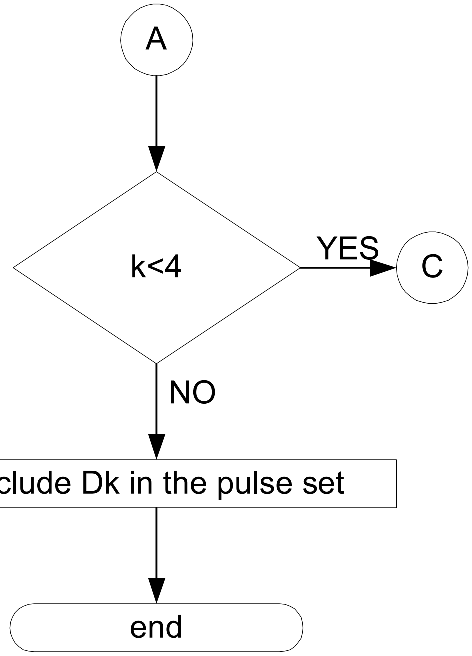


Figure 2.5: Flowchart For Finding a Subset of The Candidate Pulse Set

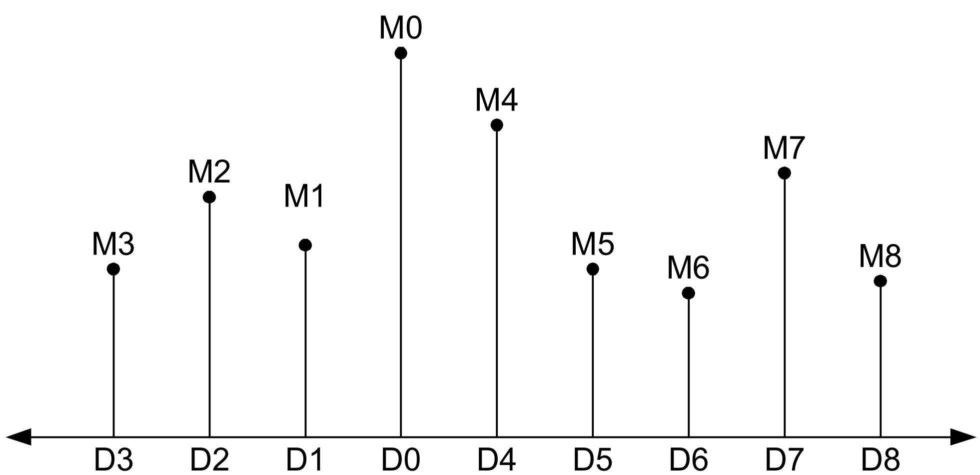


Figure 2.6: Example Showing a Candidate Pulse Set

2.2.3 Performing Linear Interpolation on The Selected Pulses

Once a subset of the set of candidate pulses has been found such that all the adjacent pulses in the subset are separated by a fixed distance DC, plus or minus a breathing threshold B, which is normally equal to 1.25 for a good estimate of pitch distance, the selected pulses must pass an interpolation test in order to ensure a smooth amplitude transition. In a voice signal segment no sudden jumps are observed. Therefore, there has to be a smooth amplitude transition. The amplitude test performs linear interpolation between the global maximum M0 and each of the other pulses Mj, i > 0, within the chosen subset of candidate pulses. The amplitude of each of the Mj pulses must be greater than q times these interpolated values. Figure 2.6 shows the subset of candidate pulses before the interpolation test is applied. For a good estimate of the pitch distance, it has been found empirically, that q should lie between 0.72 and 0.78. Equations for M1, M2, M3 and M4 show mathematically how the interpolation is performed.

$$M1 > q*[M2 + (M0 - M1) / (| D0 - D2 |) * (| D1 – D2 |)] \quad (2.9)$$

$$M3 > q*[M4 + (M0 – M4) / (| D0 – D4 |) * (| D3 – D4 |)] \quad (2.10)$$

$$M3 > q*[M5 + (M0 – M5) / (| D0 – D5 |) * (| D3 – D5 |)] \quad (2.11)$$

$$M4 > q*[M5 + (M0 – M5) / (| D0 – D5 |) * (| D4 – D5 |)] \quad (2.12)$$

Figure 2.7 presents a representation of the result of Linear Interpolation.

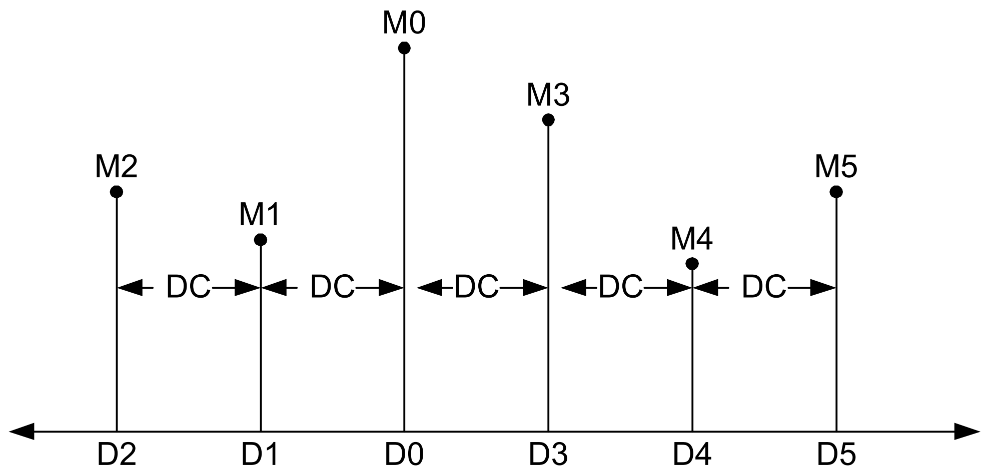


Figure 2.7: Linear Interpolation

The interpolation is performed with respect to all the pulses following a particular pulse in a particular direction. If the subset of the candidate pulse set passes the interpolation test, then it contains a valid set of pulses and DC is a valid pitch distance. If any of the above equations fails to provide a valid result then the DC is not valid and must be computed again from the previous process of finding a subset of the set of candidate pulses.

2.2.4 Pitch Consistency Test

If a pitch DC estimate is found over two consecutive frames T(i) and T(i - 1) then the two estimates must be consistent with each other such that

$$|T(i - 1) - T(i)| <= A, \tag{2.13}$$

where A is the pitch threshold. If the pitch threshold is valid then the DC is a good estimate of the pitch distance. If the calculation, for pitch threshold, in

Equation (2.13) is not valid then and a new pitch threshold is calculated in accordance with Equation (2.14), which is given by

$$|T(i - 1) - 2*T(i)| <= A. \tag{2.14}$$

Equation (2.14) corrects any pitch doubling error that might have occurred. If neither Equation (2.13) nor Equation (2.14) is valid then a new candidate distance must be calculated. The best value for pitch threshold A is 1.25 ms.

The algorithm presented in this chapter for pitch detection proves to be a very effective and accurate algorithm. The pitch value detected was used in the packet loss concealment algorithm that was developed for this research. The packet loss concealment algorithm is discussed in Chapter 3.

CHAPTER 3

PACKET LOSS CONCEALMENT

In this research, Time Scale Modification, (TSM), of speech was used to conceal lost packets in a voice packet stream. TSM is traditionally used to alter the rate of a signal in order to either expand or compress the signal.

3.1 Time Scale Modification of Speech

TSM is the process of changing the perceived rate of articulation of speech. It is a process of compressing, hastening, or expanding, slowing down, the time scale of an audio segment. A signal, which is time scale compressed has shorter duration while a signal, which is time scale expanded has a longer duration. Uses of time scale compression are fast listening of messages on answering machines, voice mail systems or synchronizing speech with the typing speed for dictation. Similarly a simple use of time scale expansion or slowing down speech is that it helps in the comprehension of rapidly spoken speech segments.

As stated earlier, Time Scale Modification of the speech signal is required in order to conceal packet loss in a voice stream. Thus Time Scale Modification should keep the principal characteristics of speech such as timbre, pitch and frequency unaltered. However, the problem with time scaling a speech signal X(t) is the corresponding frequency distortion. The duality between time scaling and frequency scaling becomes clear by considering the signal $y_a(t)$ that corresponds to an original signal $x_a(t)$ played at a speed that is 0.5 times higher than the recording speed. Thus, an original time span Δt is played in $\Delta t / \alpha$ to produce $y_a(t)=x_a(\alpha t)$. From the definition of the Fourier transform for analog signals, uniform scaling in one domain corresponds to reverse scaling in the transformed domain. This phenomenon is presented in Figure 3.1.

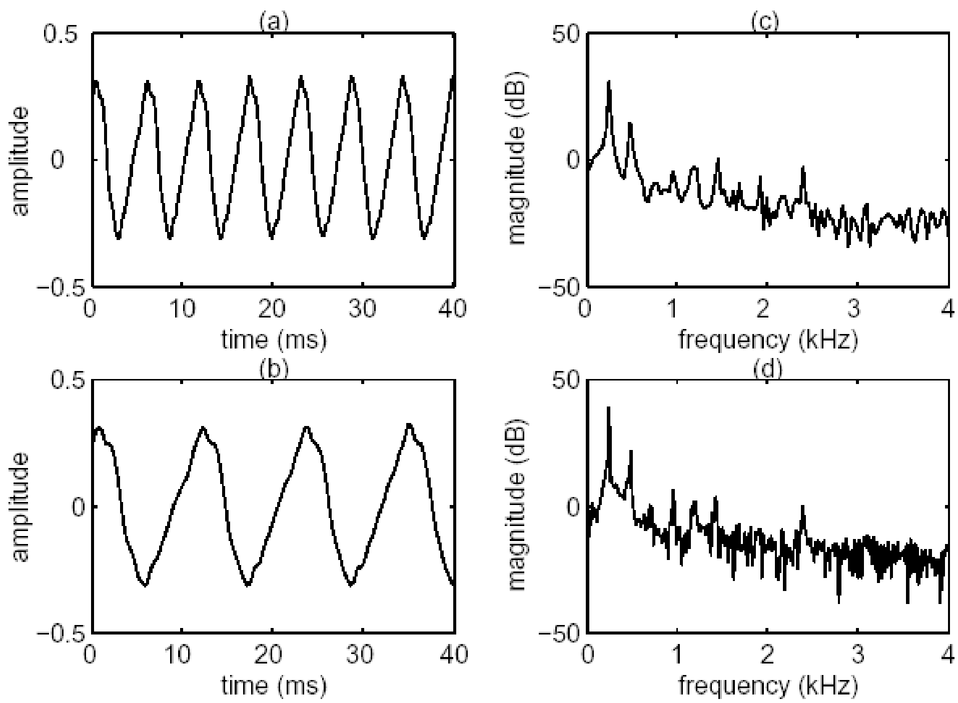


Figure 3.1: Time Scaled Waveform With Reverse Frequency Scaling

In view of this mathematical duality, it was the intent of this research to devise a method for performing time scaling without affecting other speech attributes such as pitch. A method, which is a modified form of the Waveform Similarity Overlap

Add, (WSOLA), algorithm was devised to achieve the objective. The next sections discuss existing methods, including conventional WSOLA, for Time Scale Modification.

3.2 Short Time Fourier Transform (STFT)

The Fourier Transform is the most commonly used frequency domain representation of signals in signal processing. The Discrete Fourier Transform is defined as follows

$$X(e^{+j\omega}) = \sum_{n=-\infty}^{n=+\infty} x(n)e^{-j\omega n} \tag{3.1}$$

Speech evolves slowly. Therefore, if a short time analysis strategy is used along with the Fourier Transform a Short Time Fourier Transform is obtained. A short time strategy implies segmenting the signal and applying the Fourier Transform to the segments. Segmenting is achieved by windowing. A common window function that is used is the Hamming window. A mathematical definition of the STFT is developed as follows. Signal x(n) is segmented using windowing function w(n)

$$X_{\omega}(n, m)=\omega(n)\, x(n+m) \tag{3.2}$$

Next the Fourier transform is applied to obtain

$$X(\omega,m) = \sum_{n=-\infty}^{n=+\infty} x(n+m)\omega(e^{-j\omega n}), \tag{3.3}$$

which is the Short Time Fourier Transform representation. Since windowing is used, the precision of the Fourier Transform is limited. However, the STFT works well for consecutive overlapping signal segments.

The Short Time Fourier Transform is the basic mathematical tool that is applied for implementation of packet loss concealment. Two techniques that are presently used are first discussed. They also form the basis for the technique implemented in this research. The two methods are termed Overlap Add, (OLA), and Synchronization Overlap Add, (SOLA).

3.3 Packet Loss Concealment

In an audio communication system speech in 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. The basic function of these algorithms is to generate a synthetic speech signal to cover the missing speech packets. 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 chapter 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. 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.

Time Scale Modification techniques for speech signals are used to cover up the missing packets at the receiver end. In simple terms the packets that precede the lost packets are stretched in time to cover up the length of the missing packets. This action is presented in figure 3.2. In Figure3.3 three preceding packets are stretched to make up for the loss of one packet. As shown packet 2 is lost during transmission and Time Scale Modification is performed on packets 3, 4 and 5 in order to cover the missing packet.
The next sections deal with some of the existing packet loss concealment techniques and section 3.7 introduces the modified WSOLA technique of packet loss concealment by Time Scale Modification.

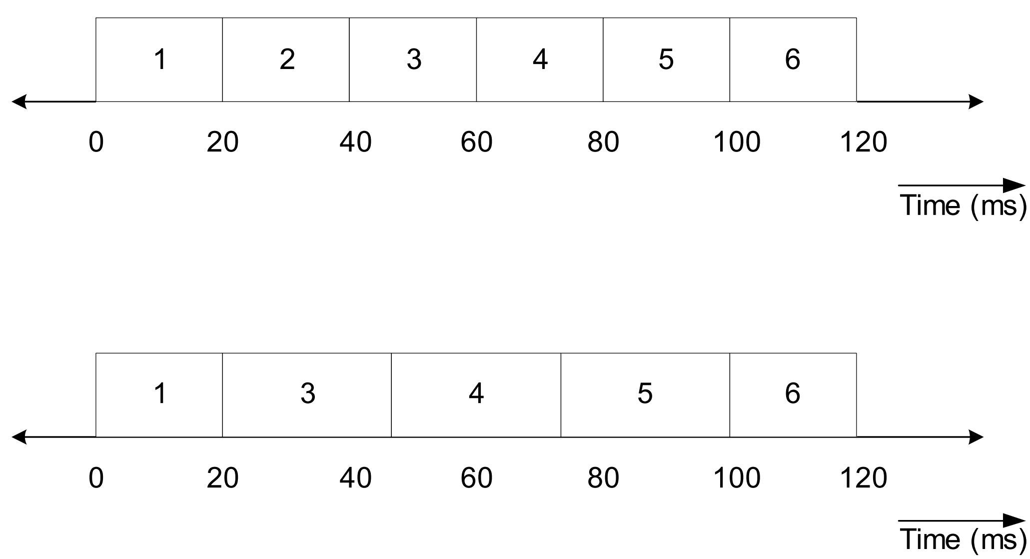


Figure 3.2: Lost Packet Reconstructed Using Two Previously Received Packets

3.4 Overlap-Add Synthesis Method

Considering a signal x(n) and performing a STFT on it produces a transformed signal X(n,m) as discussed in section 3.2. If this signal is modified to achieve time scaling, another signal Ŷ(w,n) is produced that is different from x(n) when the inverse STFT performed. In fact, Ŷ(w,n) may not even have an inverse STFT. However, this time scaled signal will contain information that best characterizes the signal modification. A synthesis formula that provides a correct value of Ŷ(w,n) such that it's inverse STFT is valid was derived by using the least mean squared error technique. In this method y(n), the inverse STFT of Y(w,n), is constructed such that Ŷ(w,n) is maximally close to Ŷ in the mean square error sense. The mean square error

$$E = \sum_k \frac{1}{2\Pi} \int_{-\Pi}^{+\Pi} |\hat{Y}(\omega,k) - Y(\omega,k)|^2 \, d\omega \qquad (3.4)$$

is minimized over all signals y(n). Parseval's theorem allows equation 3.4 to be written as

$$E = \sum_k \sum_{m=-\infty}^{m=+\infty} (\hat{y}_\omega(m,k) - y(m+k)\omega(m))^2 \qquad (3.5)$$

The signal y(n) which minimizes E is obtained by solving

$$\partial E / \partial y(n) = -2\sum_k (\hat{y}_\omega(n-k,k) - y(n)\omega(n-k))\omega(n-k) = 0, \qquad (3.6)$$

which yields

$$y(n) = \frac{\sum_k \omega(n-k)\hat{y}_\omega(n-k,k)}{\sum_k \omega^2(n-k)}. \qquad (3.7)$$

The OLA synthesis formula reconstructs the original signal if $X(\omega,m)$ is a valid STFT or a signal whose STFT is maximally close to $X(\omega,m)$ in the least squares sense is constructed. Furthermore, the denominator in equation 3.7 is required only to compensate for a possible non-uniform weighting of samples in the windowing procedure. The synthesis operation can be simplified if the windowing function and the synthesis time instants *k* can be chosen such that

$$\sum_{k} \omega^2(n-k) = 1 \tag{3.8}$$

A common choice in speech processing, that satisfies this simplifying condition, is the choice of a Hanning window with 50% overlap between successive segments.

The OLA synthesis yields a close realization of the time-scale modification in the time domain. By adopting a short-time analysis strategy for constructing $X(\omega,m)$ and by using the OLA criteria for synthesizing a signal $y(n)$ from the modified representation

$$\hat{Y}(\omega,m) = M_{xy}[X(\omega,m)] \tag{3.9}$$

will always provide modification algorithms that can be operated in the time domain if the modification operator M_{xy} [.] works only on the time index *m* such that

$$\hat{Y}(\omega,m) = X(\omega,M_{xy}[m]). \tag{3.10}$$

Taking the inverse Fourier transform yields

$$\hat{Y}(\omega,m) = X_{\omega}(n,M_{xy}[m]). \tag{3.11}$$

Combining equation (3.7) and equation (3.11) yields

$$y(n) = \frac{\sum_m \omega(n-m)x_\omega(n-m, M_{xy}[m])}{\sum_m \omega^2(n-m)}. \quad (3.12)$$

It is clear from the equation (3.12) that modification is obtained by excising segments $x_\omega(n,M_{xy}[m])$ from the input signal by using the window and repositions them along the time axis before constructing the output signal by the weighted overlap-addition of the segments. However, the periodicity of the time-scale modified signal, presented in Figure 3.3(b), is changed from the original signal, presented in Figure 3.3(a), if the above formula is applied to the time warping, τ (m), of a signal. In general, poor results are obtained when using

$$\hat{Y}(\omega,m) = X(\omega, \tau^{-1}(m)). \quad (3.13)$$

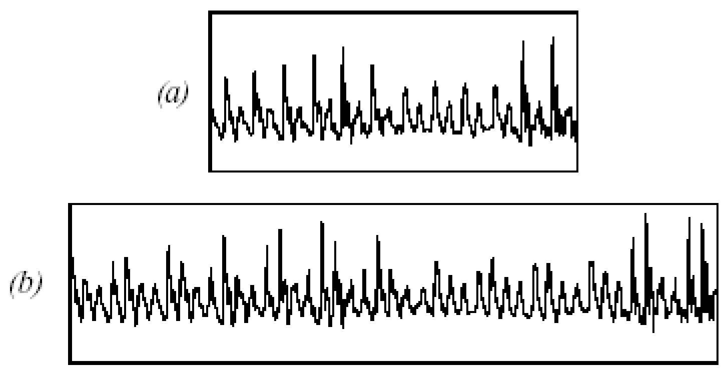


Figure 3.3: Periodicity Change

3.5 The Synchronized Overlap Add Method

Roucos and Wilgus developed the Synchronized Overlap-Add (SOLA) algorithm. They sought to accomplish Time Scale Modification by providing the algorithm with an initial guess that was closer to the desired signal. The SOLA algorithm modifies the time-scale of a signal, through analysis and synthesis, in two steps. The analysis step consists of windowing the input signal for every $S_{a,}$ (Shift analysis), samples as depicted in Figure 3.4. The synthesis step consists of overlap-adding the windows. L_{ω} is a window length, which is fixed and a multiple of the pitch period. From the analysis step, for every $S_{s,}$ (Shift synthesis), samples a rate-modified unshifted signal is produced as depicted in Figure 3.4. Each new window is aligned to maximize the correlation with the sum of previous windows before being added. This reduces discontinuities arising from the different interframe intervals used during analysis and synthesis. The resulting timescale modified signal is free of clicks, and pops. Figure 3.4 presents an example of the time-scale expansion of a signal using the SOLA algorithm.

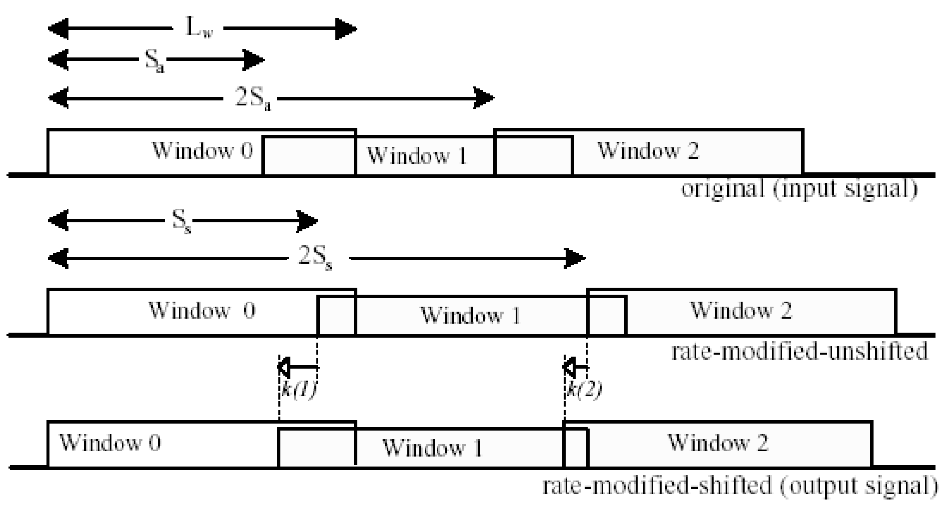


Figure 3.4: Time Scale Modification Using The SOLA Method

In the “Synchronized Overlap-Add” algorithm, windows are added synchronously with the local period. The time-scale modified signal, y(n), which is obtained from the “Synchronized Overlap-Add” of windowed segments is given by $x_\omega(n) = \omega(n)x(n)$, where x(n) is the input signal and ω(n) is the window function, is given by:

1. Initializing the signals $y_\omega(n)$ and r(n):

$$y_\omega(n) = x_\omega(n); \text{ for } n = 0,\cdots,L_\omega - 1 \tag{3.14}$$

$$r(n) = \omega(n); \quad \text{for } n = 0,\cdots,L_\omega - 1 \tag{3.15}$$

2. Updating $y_\omega(n)$ and r(n) by each new frame of the input signal, $x_\omega(n)$, is effected as·follows

$$y_\omega(mS_s - k(m) + j) = y_\omega(mS_s - k(m) + j) + x_\omega(mS_a + j) \quad \text{for } 0 <= j <= L_m - 1$$

$$y_\omega(mS_s - k(m) + j) = x_\omega(mS_a + j) \quad \text{for } L_m <= j <= L_\omega - 1 \tag{3.16}$$

where L_m is the number of overlapping points between the new window $x_\omega(mS_a+j)$ and the existing sequence $y_\omega(mS_s - k(m) + j)$ for the current frame m.

$$r(mS_s - k(m) + j) = r(mS_s - k(m) + j) + \omega(mS_a + j) \quad \text{for } 0 <= j <= L_m - 1$$

$$r(mS_s - k(m) + j) = \omega(mS_a + j) \quad \text{for } L_m <= j <= L_\omega - 1 \tag{3.17}$$

$$k(m) = \max R_{xy}^m(k) \tag{3.18}$$

$$R_{xy}^m(k) = \frac{\sum_{j=0}^{L_m-1} y_\omega(mS_s - k + j)x_\omega(mS_a + j)}{\sqrt{\left[\sum_{j=0}^{Lm-1} y_\omega^2(mS_s - k + j)\right]\left[\sum_{j=0}^{Lm-1} x_\omega^2(mS_a + j)\right]}} \tag{3.19}$$

3. Normalizing yw(n) by the buffer of appropriately shifted windowing functions r(n) to obtain the final output y(n) yields

$$y(j) = \frac{y_{\omega}(j)}{r(j)} \quad \text{for all j.} \tag{3.20}$$

As outlined in the above equations, k(m) > 0 corresponds to a shift backwards along the time-axis of the m^{th} frame that maximizes the normalized cross correlation $R_{xy}^{m}(k)$ between the m^{th} window and the rate-modified shifted signal composed of windows 0 to window (m-1). L_{ω} is the number of data points in each window frame $x_{\omega}(mS_a + j)$. Maximizing the cross-correlation insures the current window is added and averaged with the most similar region of the reconstructed signal as it exists at that point. The shifting operation insures that the largest amplitude periodicity of the signal will be preserved in the rate-modified signal. The resulting signal is called the rate-modified shifted signal to distinguish it from the rate-modified unshifted signal, which is obtained simply by overlap adding (see Figure 3.4).

It is known that the straightforward OLA synthesis from the time-scaled and down sampled STFT

$$\hat{Y}(\omega, kS) = X(\omega, \tau^{-1}(kS)) \tag{3.21}$$

results in a signal

$$y_1(n) = \frac{\sum_{k} \omega^2(n-kS)x(n-kS+\tau^{-1}(kS))}{\sum_{k} \omega^2(n-kS)} \tag{3.22}$$

that is heavily distorted, as illustrated in Fig 3.3. In equation (3.22), 'S' is a down sampling factor introduced to reduce the amount of information that needs to be processed. In order to avoid pitch period discontinuities or phase jumps at waveform-segment joins, each input segment needs to be realigned to the already formed portion of the output signal before performing the OLA operation. Thus, the synchronized OLA algorithm produces the time-scale modified signal

$$y(n) = \frac{\sum_k v(n - kS + \Delta k)x(n - kS + \tau^{-1}(kS) + \Delta k)}{\sum_k v(n - kS + \Delta k)} \tag{3.23}$$

in a left-to-right fashion with a windowing function v(n) and a shift factor Δk belonging to the set [-Δmax ··· +Δmax] that is chosen to maximize the cross-correlation coefficient between the current segment

$$v(n\text{-}kS\text{+}\Delta k)\ x(n\text{+}\ \tau^{-1}(kS)\text{-}kS\text{+}\Delta k) \tag{3.24}$$

and the already formed portion of the output signal

$$y(n;k-1) = \frac{\sum_{l=-\infty}^{k-1} v(n - lS + \Delta t)x(n + \tau^{-1}(lS) - lS + \Delta t)}{\sum_{l=-\infty}^{k-1} v(n - lS + \Delta t)}. \tag{3.25}$$

SOLA is computationally efficient since it requires no iterations and can be operated in the time domain. The time domain operation implies that the corresponding STFT modification affects the time axis only. Application of SOLA, yields

$$\hat{Y}(\omega,kS\text{-}\Delta k) = X(\omega,\ \tau^{-1}(kS)). \tag{3.26}$$

The shift parameter Δk implies a tolerance on the time warp function. However, in order to ensure a synchronized overlap-addition of segments, the desired time warp function, $\tau(n)$, is not realized exactly. A deviation on the order of a pitch period is allowed.

3.6 Waveform Similarity Overlap Add (WSOLA)

Further enhancement of the SOLA algorithm is the (WSOLA) technique. It considers that a time-scaled version of an original signal should be perceived to consist of the same acoustic events as the original signal but with these events being produced according to a modified timing stricture. In WSOLA, this can be achieved by constructing a synthetic waveform *y*(n) that maintains maximal local similarity to the original waveform *x*(m) in all neighborhoods of related sample indices m= τ^{-1}(n). Using the symbol '$\Leftrightarrow$' to denote "the maximal similarity" and using the window ω(n) to select such neighborhoods

$$y(n+m)\,\omega(n) \Leftrightarrow x(n+\tau^{-1}(m) + \Delta_m)\omega(n) \tag{3.27}$$

$$\hat{Y}(\omega,m) \Leftrightarrow X(\omega,\tau^{-1}(m) + \Delta_m) \tag{3.28}$$

Comparing equations (3.27) and 3.28 yields an alternative interpretation for the timing tolerance parameters Δ_k since the waveform similarity criterion and the synchronization problem are closely related. As shown in the above equations, the Δ_m was introduced in order to obtain a meaningful formulation of the waveform similarity criterion since two signals need to be considered as identical if they only differ by a small time offset.

3.6.1 Practical Implementation of WSOLA

Analysis segment size, (Ss), is fixed irrespective of the input speech characteristics. Time scale factor alpha is set to less that 1 depending on the desired expansion. Overlap segment size, (S0), is computed as 0.5 times Ss and is fixed. Once these parameters are fixed the output signal is formed from the input speech signal. The first two iterations for the procedure are depicted in Figure 3.5.

During the first iteration the first Ss samples of the input are directly copied to the output. If1 denotes the index of the last sample of the output and overlap index O1 is determined as S0 samples from the end of the last available samples of the output. The samples of the output between If1 and O1 are the ones that are overlap added. The first search index, (S1), is determined as alpha times O1. This search index is marked on the input signal and a search window is determined. The search window consists of samples around S1. Once within the window the best cross correlating samples are determined using the cross correlation equation

$$R(K) = \frac{\sum_{j=0}^{j=S0} X(Si + j + k)Y(Oi + j)}{\left| \sum_{j=0}^{j=S0} X^2(Si + j + k)(\sum_{j=0}^{j=S0} Y^2(Oi + j)) \right|^{1/2}} \tag{3.29}$$

where K=Si – Loff to Si + Hoff. Loff and Hoff are both 10 samples each. The maximum m is k=m where normalized R(k) is maximum. The best index B1 is determined as (S1+m).

Using equation (3.29) the beginning of the best correlating samples is determined as index B1 and is marked in the input as shown in Figure 3.5. Next the S0 samples beginning at B1 are multiplied by an increasing ramp function, whereas the S0 samples marked in the output beginning at O1 are multiplied by a decreasing ramp function. The two sets of samples generated by multiplying ramp functions are added and replace the So output samples beginning at O1 to form the output for the first iteration.

The second iteration is now similar to the first. S0 samples beginning at the end of the best correlating samples are copied to the output at the end. If2 is the index of the last sample of the output. O2 is S0 samples left of If2. New search index S2 is found to be alpha times O2. A new value for B2 is found using the cross correlation equation and the same process repeats. The number of iterations and the values to be chosen for alpha depends on the number of packets lost and the amount of expansion needed. Figures (3.5) to (3.8) present a graphical sketch of the technique.

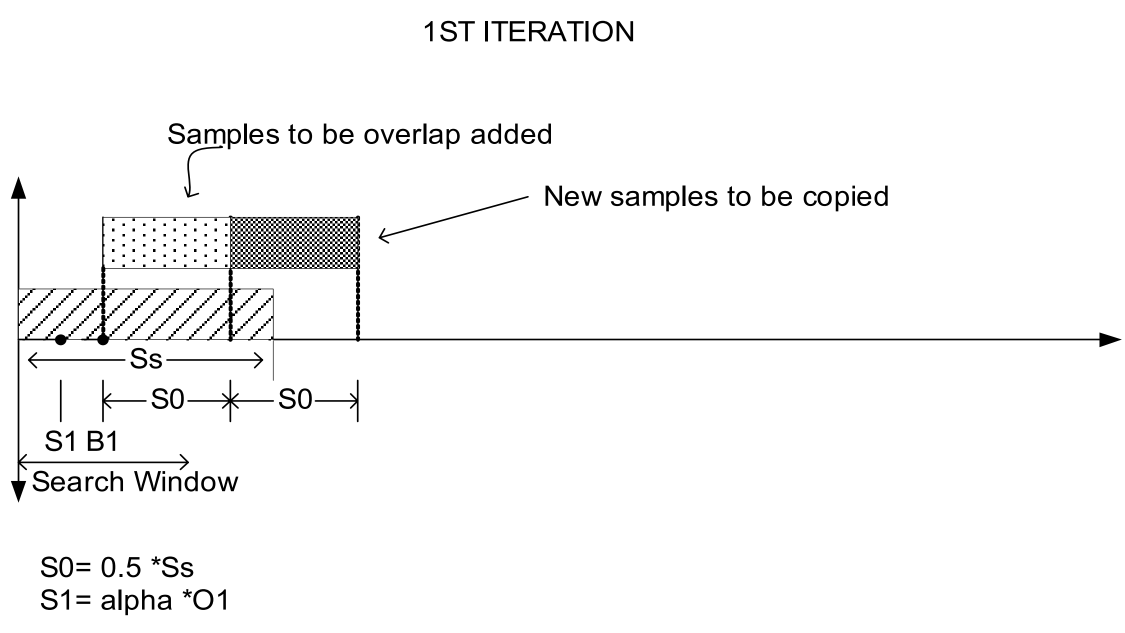


Figure 3.5: Input For The 1st Iteration of The WSOLA Method

1ST ITERATION

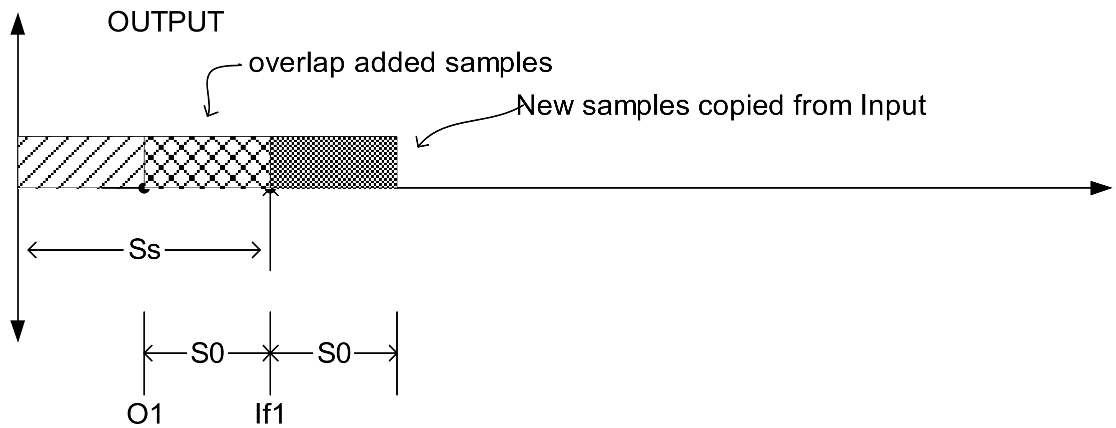


Figure3.6: Output For The 1st Iteration of The WSOLA Method

2ND ITERATION

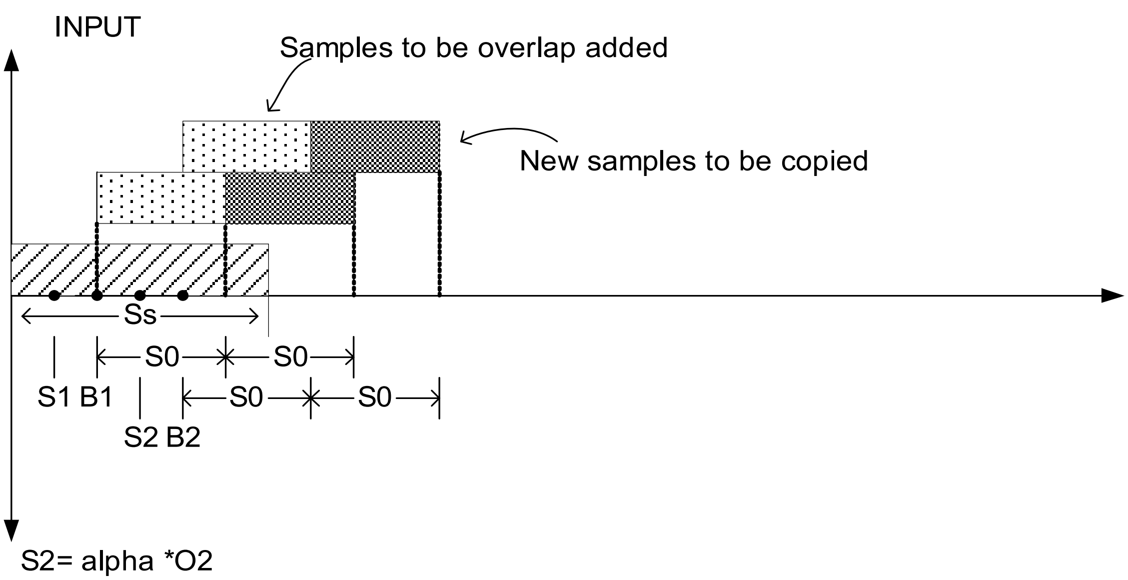


Figure3.7: Input For The 2nd Iteration of The WSOLA Method

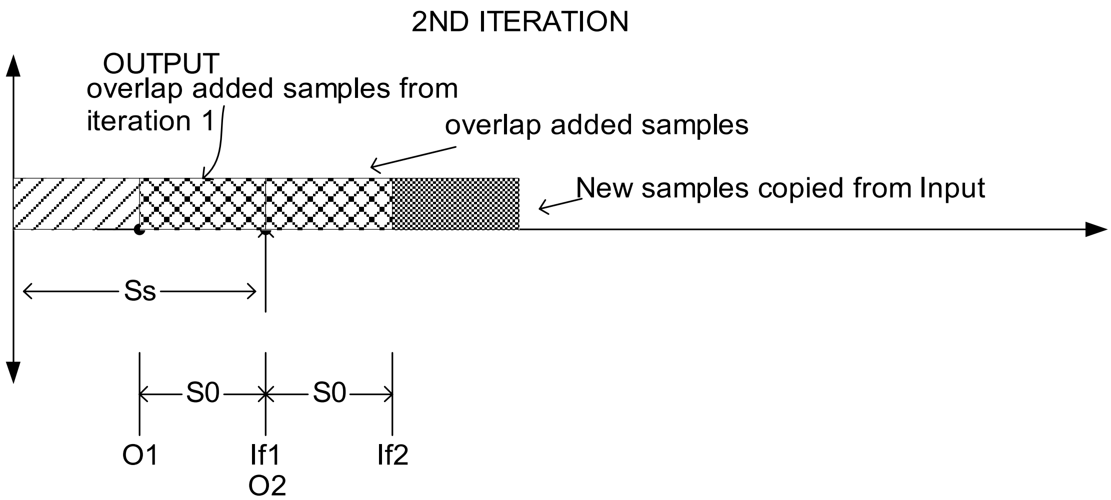


Figure3.8: Output For The 2nd Iteration of The WSOLA Method

3.6.2 Drawbacks of WSOLA

The Waveform Similarity Overlap Add technique has been discussed in the above sections. It is now summarized with respect to the constraints involved and the drawbacks of the method are discussed. Following sections introduce a new modified technique that overcomes these drawbacks.

WSOLA and its constraints:

1. Analysis segment size (Ss) is fixed irrespective of the input speech signal characteristics.
2. Time scale factor (alpha) is set to less that 1 to achieve the required expansion.
3. Overlap segment size (S0) is 0.5 times Ss.
4. If1 is the index of the last sample of the output.
5. Overlap index (O1) is S0 samples to the left of If1.

Two major drawbacks exist for WSOLA that greatly affect the quality of speech produced upon expansion of a speech signal. First, the analysis segment size (Ss) is fixed irrespective of the input speech signal characteristics. Therefore, the optimum quality of the time scale expanded signal is not obtained. If Ss is too large for the input speech signal, the resultant speech upon expansion includes echoes and reverberations. Second, the overlap segment size if 0.5 times Ss. Therefore, the user does not have the flexibility, for a given time scale factor, of design with respect to quality of speech and complexity of computations for a given system that has restraints. If a particular system has limitations with respect to processing power and memory a complicated algorithm will not be processed efficiently and the quality produced by the processing algorithm (speech quality) cannot be enhanced. Vice versa, a system with good processing power and memory will handle a complex algorithm and speech quality can be enhanced. With these issues in mind the WSOLA algorithm was modified to overcome the drawbacks and provide a better quality output signal for speech.

3.7 Modified WSOLA

The new algorithm was modified with respect to the analysis segment size (Ss) and the degree of overlap (f) in order to overcome the drawbacks of WSOLA mentioned previously.

The segment size (Ss) is computed as a function of the pitch period of the input speech signal. If P is the pitch period of the input speech signal then, depending on P, Ss is defined as follows:

For $P > 60$,

$$Ss = 2 * p. \tag{3.30}$$

For $40 < P < 60$,

$$Ss = 120. \tag{3.31}$$

For $P < 40$,

$$Ss = 100. \tag{3.32}$$

The overlap segment size (S0) is f times Ss where f is the degree of overlap. The degree of overlap is chosen as a function of quality and complexity. An $f > 0.5$ provides higher quality at the expense of more complexity while an $f < 0.5$ provides reduced complexity at the cost of quality.

The other constraints remain the same as in the WSOLA method. However, introducing these changes produced a higher quality speech signal. A discussion of this effect is presented in the results chapter. The practical implementation of the modified WSOLA technique is described next.

3.7.1 Practical Implementation of Modified WSOLA

As discussed earlier, the degree of overlap, (f), is chosen according to the requirements of the system and user flexibility. Analysis segment size, (Ss), is optimized to input speech characteristics, in particular the pitch, of the input signal. Time scale factor alpha is set to less that 1 depending on the desired expansion. Overlap segment size is computed as f times Ss and is fixed for a given pitch period and degree of overlap (f). Once these parameters are fixed the output signal is formed from the input speech signal. The first two iterations for the procedure are explained in Figure 3.9.

During the first iteration the first Ss samples of the input are directly copied to the output. If1 denotes the index of the last sample of the output and overlap index O1 is determined as S0 samples from the end of the last available samples of the output. The samples of the output between If1 and O1 are the ones that are overlap-added. The first search index, (S1), is determined as alpha times O1. This search index is marked on the input signal and a search window is determined. The search window consists of samples around S1. Once within the window the best cross correlating samples are determined using the cross correlation equation

$$R(K) = \frac{\sum_{j=0}^{j=S0} X(Si+j+k)Y(Oi+j)}{\left| \sum_{j=0}^{j=S0} X^2(Si+j+k)(\sum_{j=0}^{j=S0} Y^2(Oi+j)) \right|^{1/2}}, \tag{3.33}$$

where K=Si – Loff to Si + Hoff. Loff and Hoff are both 10 samples each. The maximum m is k=m where normalized R(k) is maximum. The best index B1 is determined as (S1+m).

Using equation (3.33) the beginning of the best correlating samples is determined as index B1 and is marked in the input as shown in Figure 3.9. Next the S0 samples beginning at B1 are multiplied by an increasing ramp function, whereas the S0 samples marked in the output beginning at O1 are multiplied by a decreasing ramp function. The two sets of samples generated by multiplying by ramp functions are added and replace the S0 output samples beginning at O1 in order to form the output for the first iteration.

The second iteration is similar to the first iteration. S0 samples beginning at the end of the best correlating samples are copied to the output at the end. If2 is the index of the last sample of the output. O2 is th S0 samples left of If2. New search index S2 is found as alpha times O2. A new value for B2 is found using the cross correlation equation and the same process repeats. The number of iterations and the values to be chosen for alpha depends on the number of packets lost and the amount of expansion needed. Figure 3.9 through Figure 3.12 present a graphical sketch of the technique.

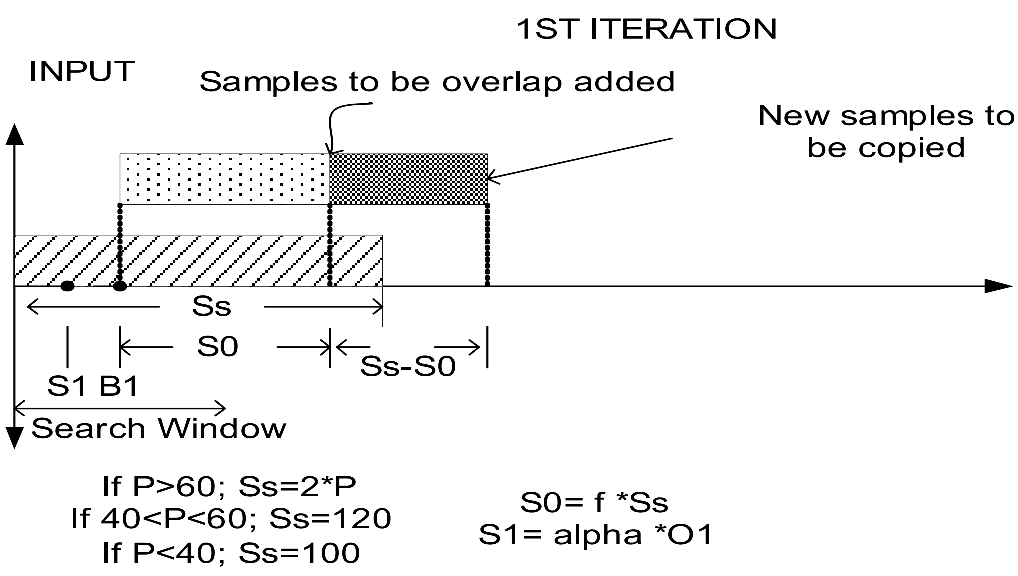


Figure3.9: Input For The 1st Iteration of The Modified WSOLA Method

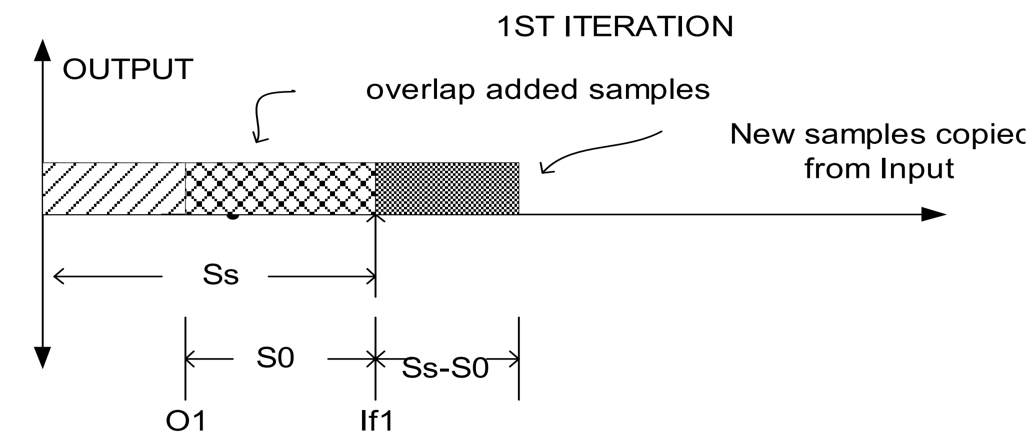


Figure3.10: Output For The 1st Iteration of The Modified WSOLA Method

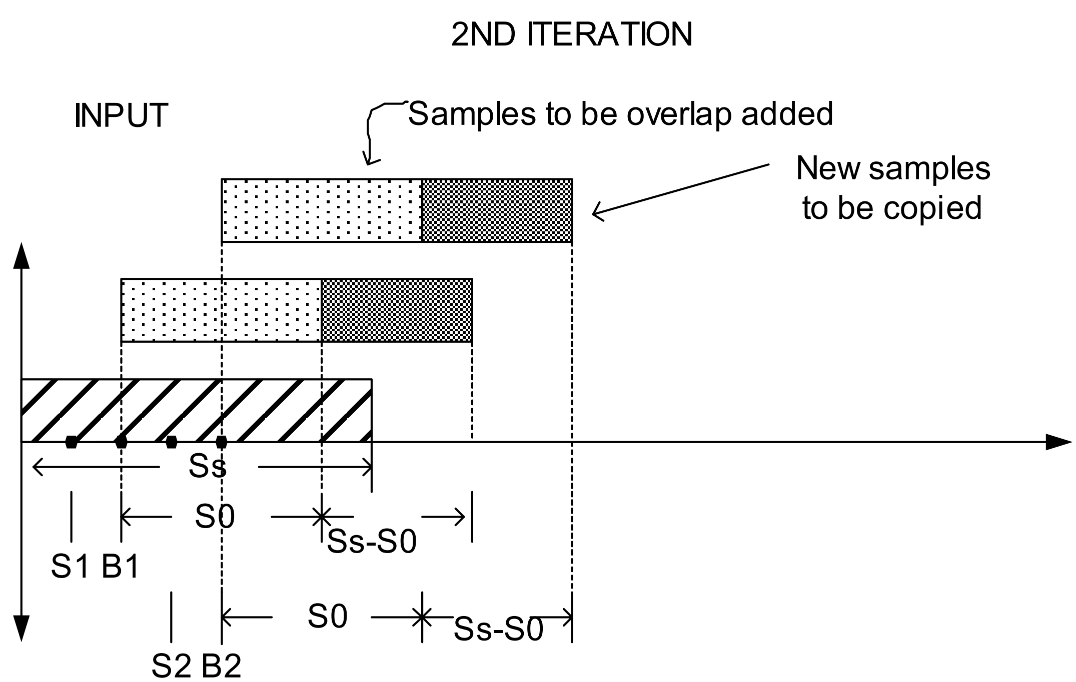


Figure3.11: Input For The 2nd Iteration of The Modified WSOLA Method

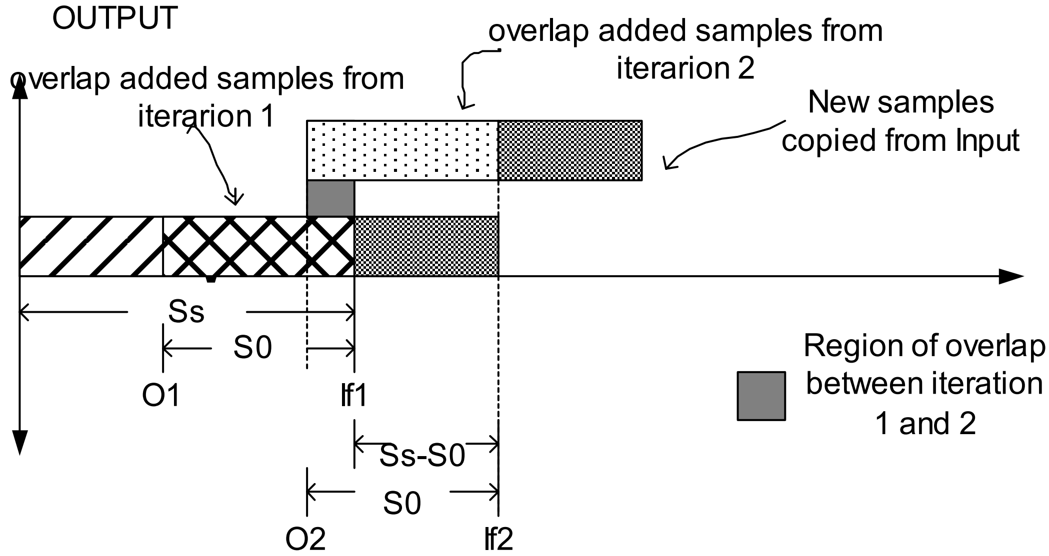
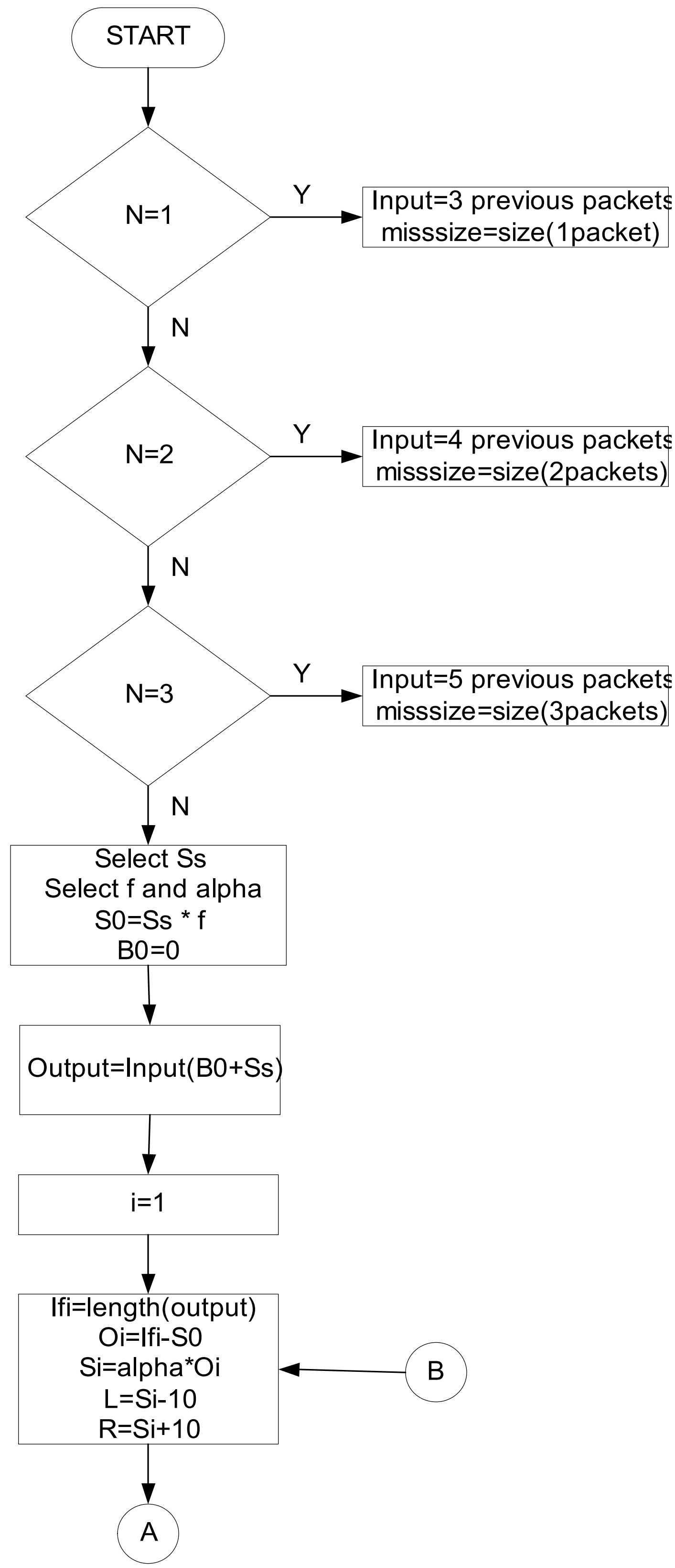
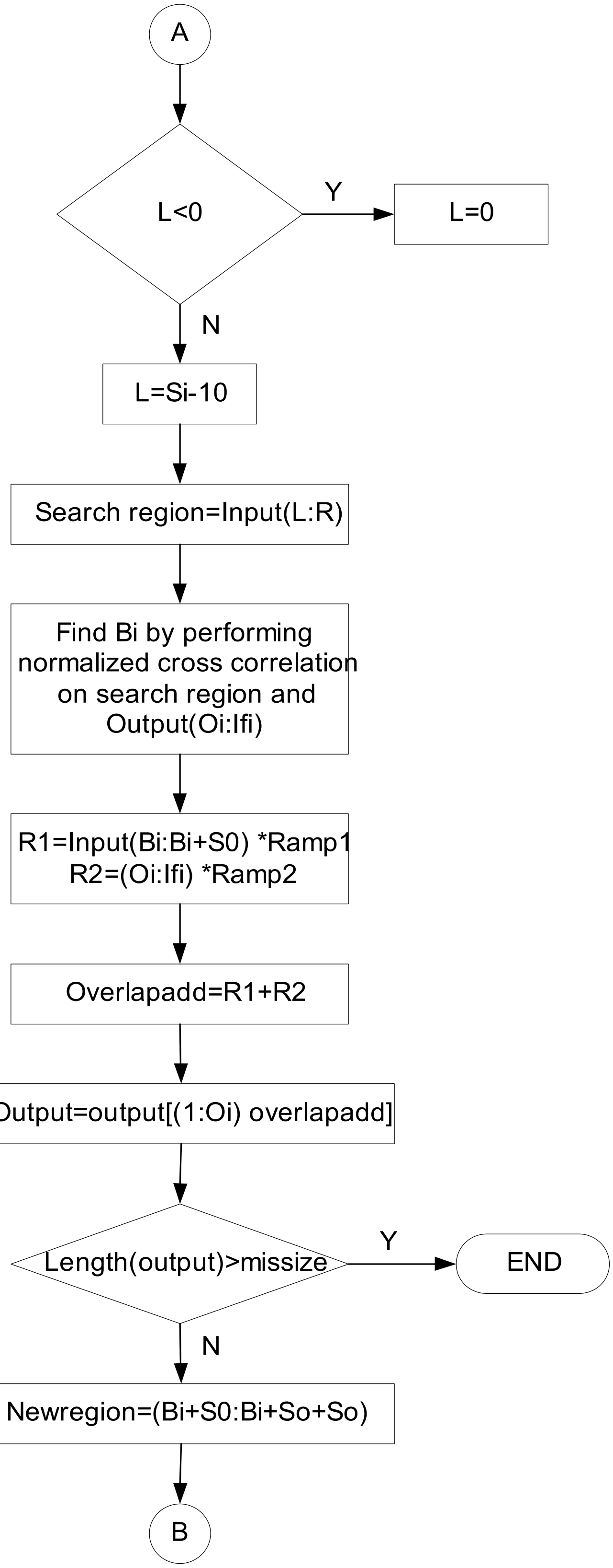


Figure3.12: Output For The 2nd Iteration of The Modified WSOLA Method

The modified WSOLA technique was simulated in Matlab. The flowchart for the simulation is as presented in Figure 3.13.


START
N=1
Y
Input=3 previous packets
misssize=size(1packet)
N
N=2
Y
Input=4 previous packets
misssize=size(2packets)
N
N=3
Y
Input=5 previous packets
misssize=size(3packets)
N
Select Ss
Select f and alpha
S0=Ss * f
B0=0
Output=Input(B0+Ss)
i=1
Ifi=length(output)
Oi=Ifi-S0
Si=alpha*Oi
L=Si-10
R=Si+10
B
A


A
L<0
Y
L=0
N
L=Si-10
Search region=Input(L:R)
Find Bi by performing normalized cross correlation on search region and Output(Oi:Ifi)
R1=Input(Bi:Bi+S0) *Ramp1
R2=(Oi:Ifi) *Ramp2
Overlapadd=R1+R2
Output=output[(1:Oi) overlapadd]
Length(output)>missize
Y
END
N
Newregion=(Bi+S0:Bi+So+So)
B

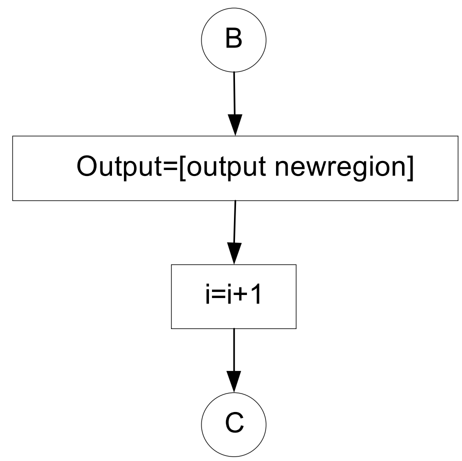


Figure 3.13: Flowchart For The Matlab Simulation of The Modified WSOLA Method

CHAPTER 4

RESULTS AND CONCLUSIONS

The pitch detection module and the modified WSOLA module work together to form the entire packet loss reconstruction process. In this research receiver based packet loss reconstruction was investigated. At the receiver, as the first packet of the voice signal arrived it was stored in a buffer before it was played out to the listener. The next five consecutive packets were also stored in the buffer for a total of six packets. The six most recent packets were always stored in the buffer at any given time. As the next packets arrived the most recent packets were stored and the earlier packets erased. Packet loss concealment was performed as follows:

1. The 3 most recently arrived packets concealed a lost packet.
2. The 4 most recently arrived packets concealed Consecutive lost packets.
3. The 3 most recently arrived packets concealed consecutive packets.

If more than three consecutive packets were lost the quality of the recovered speech was not good and the speech signal severely affected. In the case that packets among the first six packets were lost, the preceding packets were used for reconstruction.

At the receiver as the speech signal arrived it was first sampled at the rate of 20 KHz, then packetized to include 160 samples per packet and send through the pitch detection module to calculate its pitch. The samples went through the buffer where the six most recent samples were stored before they were played out to the listener. Whenever the receiver detected lost packets the modified WSOLA module activated to perform packet reconstruction and then the voice signal was played out.

4.1 Measuring the Quality of Speech

In voice communications, the mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit. MOS is a widely accepted scheme used to test the quality of coders and many other signal processing devices. The scheme uses subjective tests in the form of opinionated scores that are mathematically averaged to obtain a quantitative indicator of system performance. To determine MOS, a number of listeners rate the quality of the speech spoken by male and female speakers. A listener gives each sentence a rating from 1 to 5 where (1) is bad, (2) is poor, (3) is fair, (4) is good and (5) is excellent. The MOS is the arithmetic mean of all the individual scores and can range from 1, which is worst to 5, which is best.

4.2 Tests and Results

In order to test the new algorithm two tests were conducted. Eight voice samples, four male and four female, with typical voice conversation were recorded. Each sample had a length of 5 seconds. A sampling rate of 8 KHz was used and the packet length was 160 samples per packet. Each speech signal consisted of 250 packets, which were sampled at an 8 KHZ rate to yield 40,000 samples.

4.2.1 Test 1

Each of the eight voice samples was distorted in order to produce 1 lost packet, 2 consecutive lost packets and 3 consecutive lost packets. Reconstruction was performed using WSOLA and the modified WSOLA using the following criteria:

1. The 3 most recently arrived packets concealed 1 lost packet.
2. The 4 most recently arrived packets concealed 2 consecutive lost packets.
3. The 3 most recently arrived packets concealed 3 consecutive packets.

4.2.2 PESQ Score

PESQ stands for Perceptual Evaluation of Sound Quality and is an enhanced perceptual quality measurement for voice quality in communication networks. It was specifically developed to be applicable to end-to-end voice quality testing under real network conditions. It is specified by the International Telecommunications Union recommendation ITU-T P.861 [10]. This test rates the quality of speech on a scale of 1 to 5. The worst score is 1 and the best score is 5. The voice samples reconstructed by the modified WSOLA method were subjected to the PESQ test.

A comparison of the PESQ test results for 1 random packet loss for the WSOLA AND modified WSOLA methods are presented in Figure 4.1.

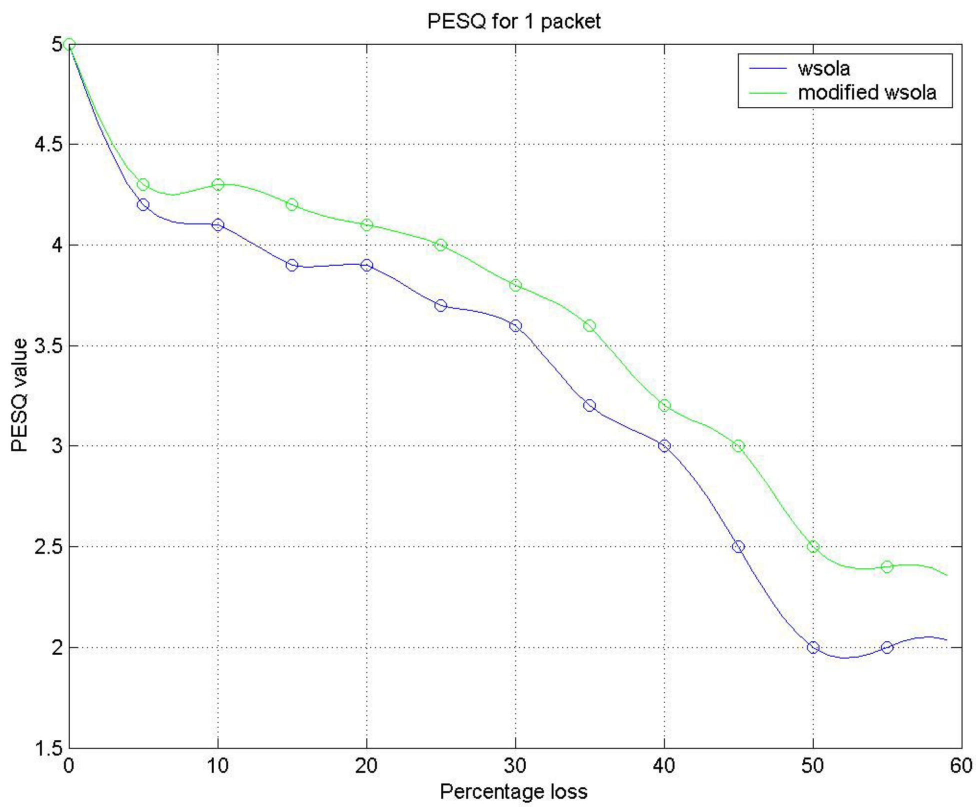


Figure 4.1: PESQ Scores For 1 Random Packet Loss

A comparison of the PESQ test scores for, 2 consecutive random packet losses for the WSOLA AND modified WSOLA methods are presented in Figure 4.2.

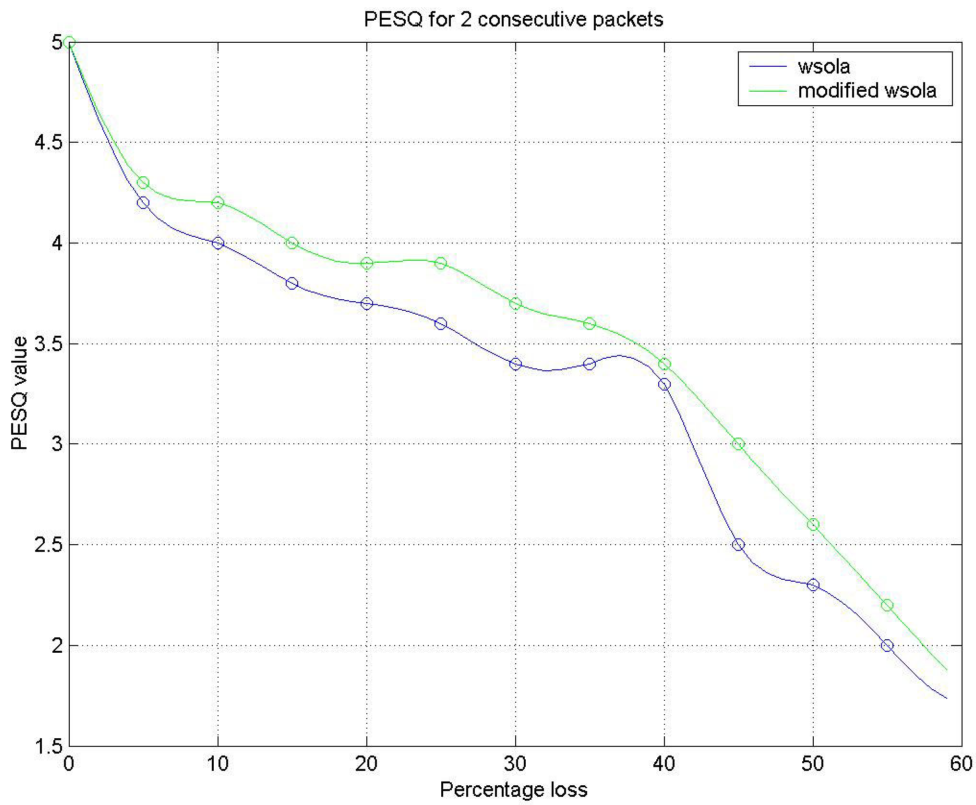


Figure 4.2: PESQ Scores For 2 Consecutive Random Packet Losses

A comparison of the PESQ test scores for three consecutive random packet losses for the WSOLA AND modified WSOLA methods are presented in Figure 4.3.

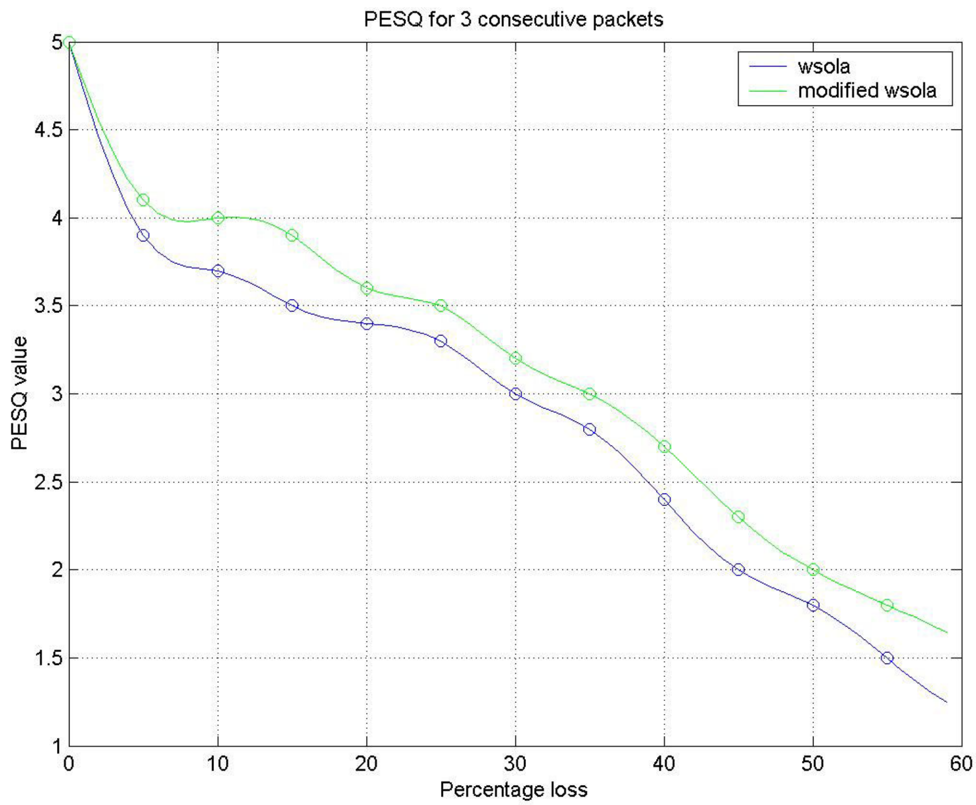


Figure 4.3: PESQ Scores For 3 Consecutive Random Packet Losses

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