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Examine the Accuracy of VOIP Signal Quality over Wireless Channels

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ABSTRACT

The Aim of this paper is to give a basic introduction on VOIP and describe QOS problems. To get the best quality in real time. To investigate the performance of speech over VoIP using commercial 3G wireless network.More specifically, an investigation of IP-based voice communication with emphasis on the effects of a wireless channel and its conditions on the quality of the received speech will be investigated. Simulation will be implemented using Matlab software, and the experiments will be conducted on commercial VoIP networks. The attention is focused on the quality of service in wireless network.

Keywords: MATLAB, VOIP, 3G, VOICE QUALITY.

I. INTRODUCTION

A change from analog to digital telephony has well been established in most countries worldwide. Second, there is an increasing trend towards the use of internet telephony, also known as Voice over Internet Protocol (VoIP). VoIP is the transmission of voice using a packet switched network, which is usually based on the transmission control protocol/internet protocol (TCP/IP) suite. First of all, voice transmission over the Internet can be cheaper than that over traditional telephone networks. Second, it provides a handful of new opportunities and applications to its users; these new features are almost impossible to implement without the use of a packet switched network. There are many different ways in which two or more users can be connected to a VoIP network, but the main concept of inter connecting both PSTN and VoIP are used.

II. METHODS AND MATERIAL

1. Key Challenges for VOIP over 3G Wireless Networks

Packet delay: Packet delay is of two types: handling and propagation. Handling (packetization) delay is the amount of time it takes for a speech signal to be processed by the computer's hardware before it is transmitted to the medium. Propagation delay is the amount of time it takes a signal to travel from transmitter to receiver and is dependent upon the medium used.

Packet delay variation: Packet delay variation or jitter is the variation in delay between consecutive packet inter-arrival times. It is an effect of packet switched networks and it can be more annoying than the packet delay itself since its effects vary over time.

Packet loss: Another problem of most packet switched networks is packet loss. It can happen at any point in the network especially in media that are prone to errors like a wireless medium.

Attenuation and noise: In order to fully recover the signal at the receiver's end, two conditions must be met. The signal strength must be sufficient to be detected by the receiver's circuitry and must be higher than noise.

Clarity of received speech: This is another most effective measure on any telephone call is to how the user perceives what he or she hears which means how clear and undistorted the sound from the other user is.it obvious that such a measurements is subjective. It is subjective to the user's background, mood and attitude.

Multipath: In the case of mobile wireless networks, a loss model that accounts for multiple copies of the same signal due to multipath effects must be considered. The main sources of multipath are diffraction, reflection and scattering. These multiple copies have varying delays and, in some cases, they might be the only signals received, e.g., in the case of non-line of sight (NLOS) reception.

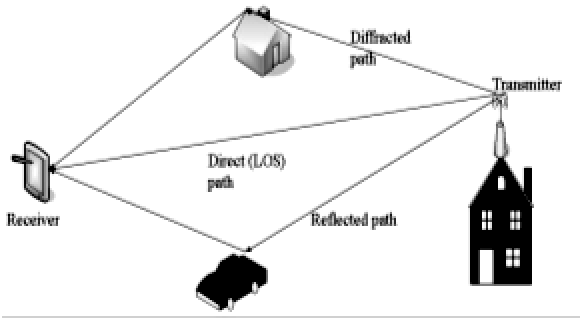


Figure 1: Schematic of multipath channel where the signal follows a direct, a reflected and a diffracted path.

VoIP availability: For many years the PSTN companies have been advertising their four 'nines' of availability as their main advantage over VoIP and cellular telephony. What they really mean is that their network (from PBX to PBX) availability is close to 99.99%; if the availability of home appliances has to be accounted for, the total availability would be much lower. In order for VoIP to fully replace the traditional telephony, it has to get closer to its competitor’s level of availability. Availability, is given by

$$\text{Availability} = \frac{\text{MTBF}}{\text{MTBF} + \text{MTTR}}$$

2. Channel

A) Linear predictive coding: In linear predictive coding of speech, the source is represented by the voiced or unvoiced excitations. In this approach, a linear filter is used to model the vocal tract. The input to this filter is a random noise or a periodic pulse, depending on whether the excitation is voiced or unvoiced. The filter can be analytically represented as:

$$y_n = \sum_{i=1}^{M} a_i y_{n-i} + Ge_n \tag{2}$$

Where, $e_n$is the excitation (also known as prediction error), Mis the filter order, Gis the filter gain and a_i are the filter coefficients. The linear predictive filter coefficients are obtained by minimizing the prediction error power with respect to the filter coefficients (a_i), which is given by

$$E[e_n^2] = E\left[\left(y_n - \sum_{i=1}^{M} a_i y_{n-i}\right)^2\right] \tag{2}$$

$$= \sum_{i=1}^{M} a_i y_{n-i} + Ge_n$$

Where,$E[\cdot]$ is the expectation operator.In practice, in order to compress a segment of speech, it is first divided into smaller segments. This segment is then classified as voiced or unvoiced based on its energy and frequency contents. These parameters are transmitted after obtaining the pitch and vocal tract filter coefficients. At receiver, the received parameters are then used to reproduce the voice. LPC is preferably used in applications where compression ratios are of most importance.

B) Code Excited Linear Prediction (CELP): CELP methods improve the voice quality by using better excitation techniques compared to LPC. The output of the filter is given as

$$y_n = \sum_{i=1}^{M} a_i y_{n-i} + \beta y_{n-P} + Ge_n \tag{3}$$

Where, G is the filter gain and a_i are the filter coefficients. P is the fundamental harmonic period, also known as pitchand $\beta$is a scaling factor. The pitch periodicity contribution is βy_{n-P} and is calculated every subframe. Equation 3.3 can be treated as a cascade of two filters. The first filter extracts the pitch, and the second is a long term formant filter. The excitation is created using the codebook approach so that it is not necessary to extract voicing patterns or pitch. The codebook is generated offline. In this approach, the synthesized outputs are compared every time with the predetermined codebook to find the best match. An example of CELP is shown in Figure 3.2. In this example, the first codebook is called stochastic codebook. This codebook is fixed and predetermined; however, the second is adaptive. The excitation of each segment is the sum of adaptive and stochastic codebook outputs. After excitation e[n]is produced, a copy of it is fed back to the adaptive codebook which then adapts to the current segment. In order for the scheme to provide minimum error between the input and synthesized speech, the codebook indices are scaled using gains.

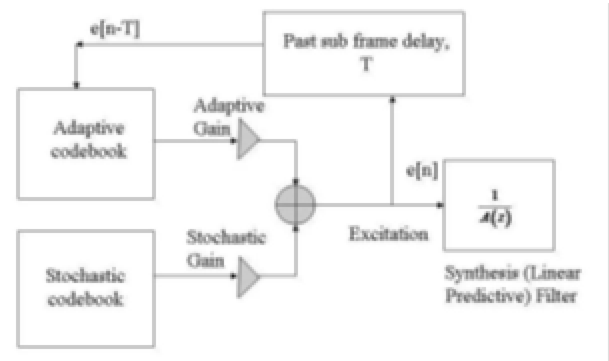


Figure 2: Block diagram of CELP featuring two codebooks.

3. Fadings

Fading channel has used two types of channel, first is Rayleigh channel and second is rician channels. Rayleigh channels no paths are consider dominant, in Rayleigh distribute signals have to be different path have different phases and similar signal strength have receive into produced Rayleigh distribute channel, Rayleigh channel have to be probability density function are shown as

$$f_R(r) = \frac{r}{\sigma^2}\exp\left(-\frac{r}{2\sigma^2}\right),\ r \geq 0_r$$

Where σ^2 are average receive power and rare signals magnitude, in the case of racian channel distributions have to be dominants line of sight, signals are presents, resulting amplitudes have be modulated with the rician probability density function, they are given below

$$f_R(r) = \frac{r}{\sigma^2}\exp\left(-\frac{r^2+K^2}{2\sigma^2}\right)I_0\left(\frac{Kr}{\sigma^2}\right), \qquad r \geq 0, \kappa \geq 0$$

I_0 have two zero orders magnitudes Bessel functions, K are ratio of dominant paths power into remaining paths power, dominant path power are used to the line of sight, when in addition case if we put k=∞ then the channel has becomes to an additive white Gaussian noise. In other hand if we put k=0 then the channel has becomes to Rayleigh channels. Measured of fading channel can be used for frequency and time dispersions are obtain into coherence time and coherence bandwidth, where Tc are the coherence time the decor relate have two time domains sample are given by

$$T_c \propto \frac{1}{f_d}$$

Where F_d are Doppler shifts, F_c are the frequency carrier, F_d as shown in

$$f_d = \pm\frac{f_c}{c}v\cos(\psi)$$

Where ψ is the angle of radiations and relative motions, c has speed of light. V has the relative velocity of transmitters and receivers .:

$$B_c \propto \frac{1}{\sigma_\tau} \tag{4}$$

Where, σ_τ is the rms delay spread of the channel.Depending on the amount of time delay spread, channels can be classified as flat or frequency selective; and depending on the amount of Doppler spread, they can be classified as fast or slow fading. A discussion on wireless standards is presented below.

III. RESULTS AND DISCUSSION

In Figure 3, the speech is first recorded through microphones. It is recorded for two sentences by author for three different intervals and is shown in Figures 4 and 5. The samples contain two times repetition of the sentences, so in total are 12 samples. The two sentences are:

Sentence 1: *VoIP are transmission of voice used to a packet switched network.*

Sentence 2: *There are based on to the transmission control protocols and internet protocols.*

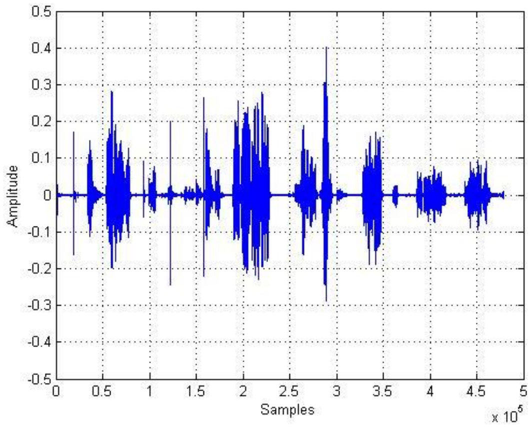


Figure 3 : Speech samples of author corresponding to sentence 1.

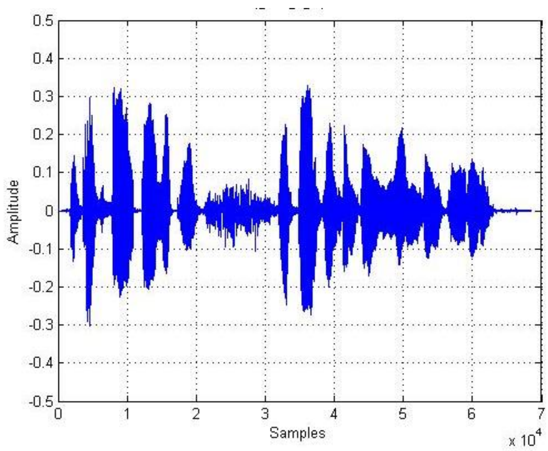


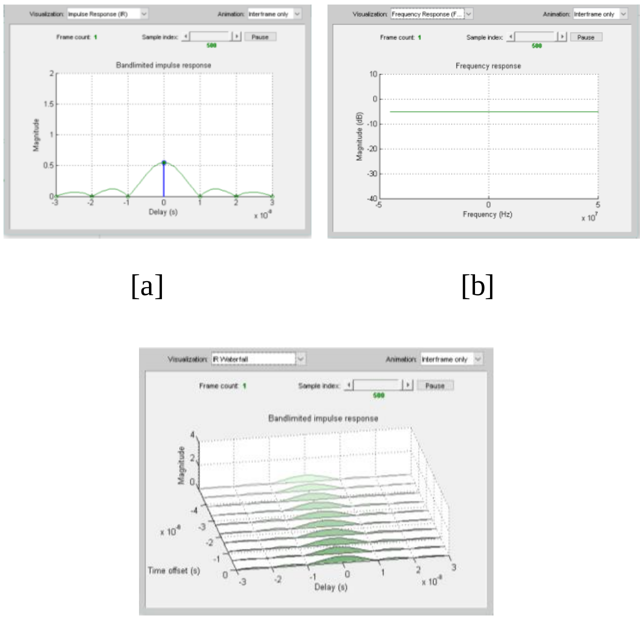
Figure 4 : Speech samples of author corresponding to sentence 2.

The recorded speech is transmitted through a network which includes at least one wireless link along the end to end path. This causes distortion to the speech sample due to packet errors, which occurs due to fading caused by wireless channels. The speech is received at the other end and is distorted and can be completely faded. To estimate the quality of the speech, the received speech is compared with the original speech through bit error rate, as follows:

$$BER = \frac{Number\ of\ bit\ errors\ in\ the\ received\ signal}{Number\ of\ bits\ in\ the\ original\ transmitted\ signal} \quad (5)$$

VOIP network was simulated in Matlab digitize, compress, packetize and transmit into receiver, it can be follow to the reverse procedure. The Rician and Rayleigh fading channels are simulated in Matlab, and their characteristics is shown in

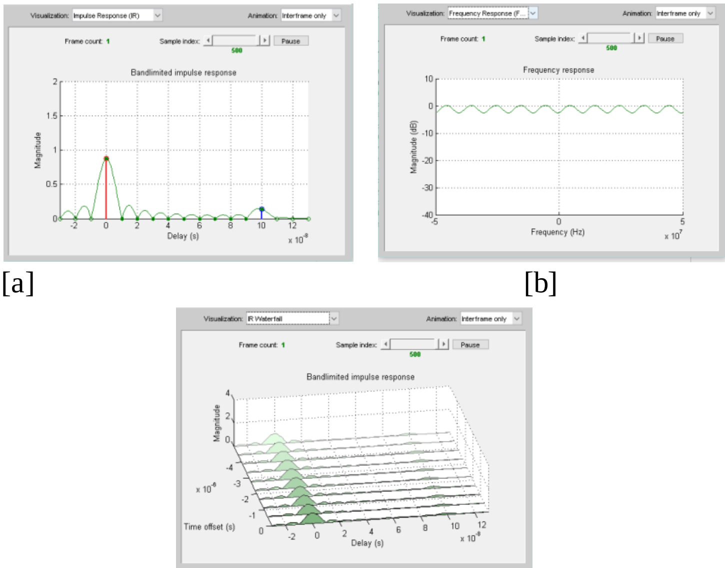
Figure 5 and 6 respectively.



[a] [b]

[c]

Figure 5. The characteristics of Rician channel



[a] [b]

[c]

Figure 6 : The characteristics of Rayleigh channel.

The Rayleigh fadings channels and Rician channel fadings channel were implemented into Matlab with used to AWGN, addive white Gaussian noise AWGN with convolutional coding. The recorded signals in Figures 4.2 and 4.3 are passed through these channels and the received signal is compared with the input signal using BER. The BER performance of AWGN, Rician and Rayleigh fading channels is shown in Figure 4.6. For a particular value of EbNo, the Rayleigh fading channel gives highest BER and the AWGN channel is the lowest.

The effect of K factor of Rician channel is shown in Figure 4.7.if K factor are increasing then the BER are also decrease. At the value close to 0.06.If we put K=0 and down to 10^{-5} as the value of K factor reaches at 25.In addition case if we put k=0 then the channel becomes to relay channel, If we put K=∞ then the channel becomes to additive white Gaussian noise channels, we use the BER high impractical can be get for the transmissions of information. If K factor increase then order of magnitude improve, if SNR value is 6dB, then BER value are 0.5 and channel has make into in appropriates, if SNR value increase from 6 degree to 23 degree, then the BER has decrease and the value closed to 10^{-6}, if SNR value between 6 dB to 16dB, then they have no significant improvement in BER, if SNR value more than 16dB then the channel has improve in BER, if SNR value increases from 22dB to 23dB then BER of transmission on decreases.

4. Parameters

Parameters used for simulation BER in AWGN ,Rayleigh & Rician fading channel.

Table

Eb/No	AWGN of BER	Rayleigh of BER	Rician of BER
5	0.007512	0.5	0.00999
10	0.00001	0.0512	0.007512
15	-	0.0008122	0.0855
20	-	0.0001	0.565
25	-	0.00005	0.0259
30	-	0.00001	0.0092

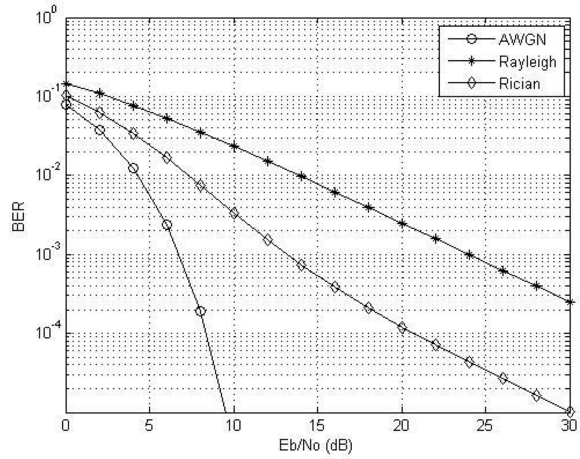


Figure 7 : BER characteristics of AWGN, Rician and Rayleigh channels

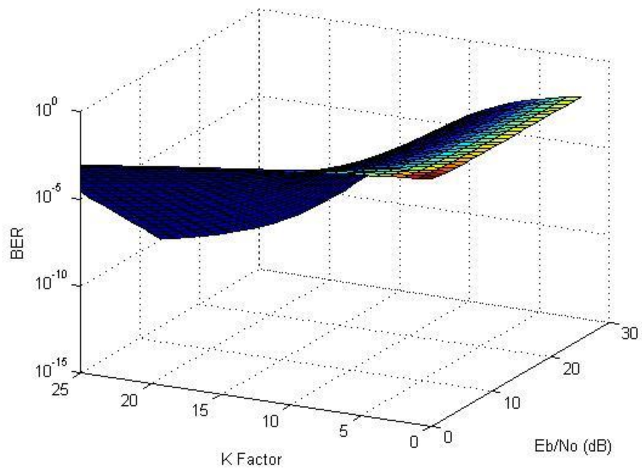


Figure 8. K-factor Vs Eb/No of the Rician channel.

4.1 Implementation

The system is implemented two fold. Firstly, the effect of channels is described and then it is evaluated for a commercial SKYPE VOIP service. This is mentioned below.

4.2 In AWGN Speech through Rician Fadings channels.

In these implementation, In AWGN the speech sample have to be modulate aare modulated (baseband)and transmitted through a Rician fading channel in additive white Gaussian noise. After demodulation and decompression in the receiver, the BER have to determine then we have decidedness in to the after to the de modulations and de compressions. If K factor of the Rician channels have to be adjust to 1 ,then we have to get noise level should be fixed or variables If we have to be added a convolutional coder with (n, k ,K) then get (2,1,3) before the signals have to be modulate after the signals have to be demodulate, then the code bit has decode into extract these receiver speech bit in the reverse end. If the value of K factors of the rician channels have to be adjust again 1 then we get noise level can be into fix or variables. The same channels have to be used into determine the effect on to the BER. Dominant path of SLR have to measure in the rician channels feedings. In this case channels have to be also examine when the effect transmitting speech via same signals. Then result has to be shows in fig. 9 to 13.

At different values of K factor in Rician channel, different values of BER is obtained. It is because the characteristics of the channel changes with K factor. If the uncompressed signals have to with BER 10^{-3} presented. In general compress signals have do not even and decidable, then they have to slightly under stables, they have to easy realise compress the speech signals then we have to get the result in gains or biot rate. But signals have to be became very more sensitivity error. In this case, all results shows in fig. 4.8 to 4.12, if they have to be increase in SNR then we get to decrease of BER, in this case, these effect have due to into multipath, dominant path have to be became strangers then uncertainties can be about ISI, and the pulse decimations has decrease, but strengths of the main paths have been becoming stronger, then the receivers have to be discriminated between pulses and delay copy of previous pulses.

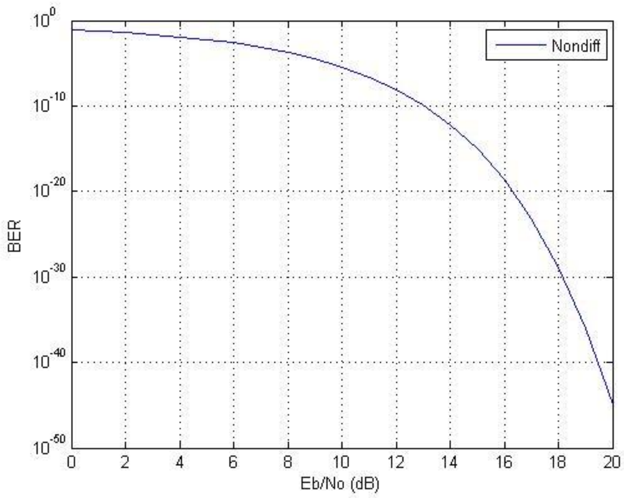


Figure 9 : Theoretical effects of EbNo and BER .

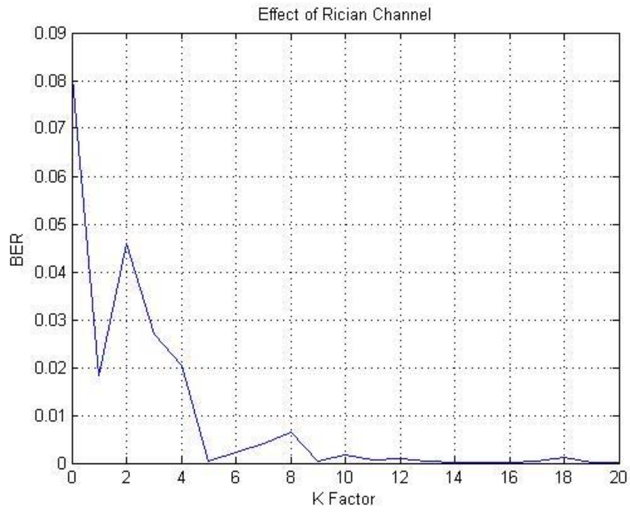


Figure 10 : K-factor Vs BER for the Rician channel.

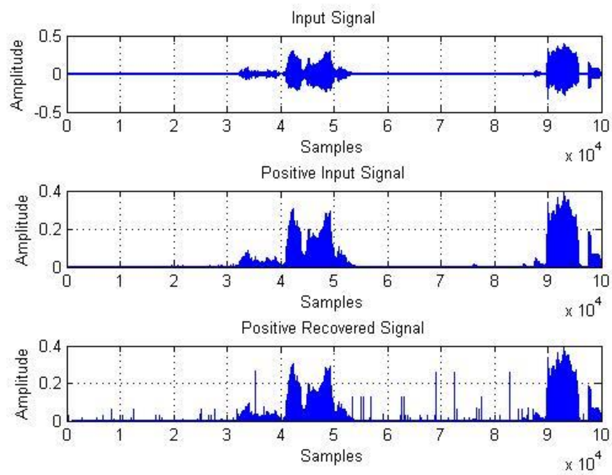


Figure 11: Positive recovered signal for Rician channel with K=20.

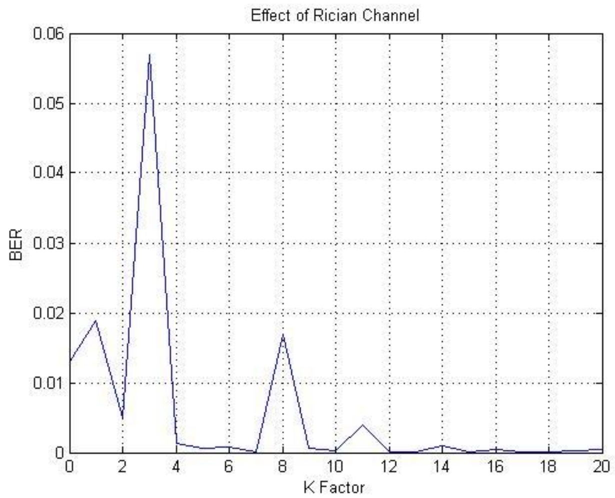


Figure 12 : K-factor Vs Eb/No of the Rician channel for the second sample.

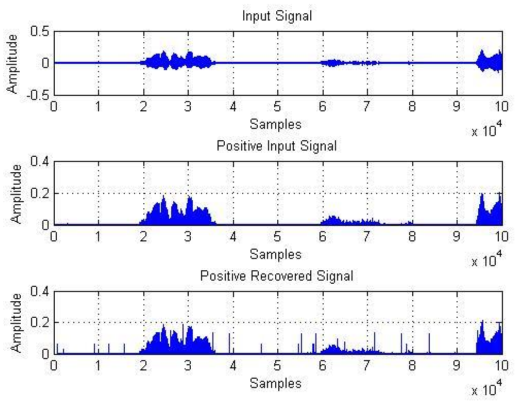


Figure 13: Positive recovered signal for Rician channel with K=20 for the second sample.

Now next test the Rician fadings channels, they have to be used for secondary paths delays, then the results have to be obtained BER of the signal increases. The result have to be get from effects of multipath in this path, the signal strength have to be large the first restoration increase, when the delays variation of secondary path have to be get became into larger the receiver have to be unable. In Rician feedings channels, the effects of secondary path have to be also examine, then we get to the BER increases, if result are obtained logically 1 then the secondary signals have to be strangers, in additional case, when channels coding have to be use then improve the remaining speech quality. They have to be preferable d into used a high compressions ratio 10:1 then lower compression ratio 2:1 without into convolution coding. If we have to get 10:1 compression ratio then coding channel have to be 12 trans mit have to be 6 kbps. It is receive to 100 % speech. In general way they have to be used 2:1 compression ration then we have get to results are totally transmit signal of 40 kbps. But when the receive speech have to be 30% then drawback of channel coding have to be circuits, complexity. We have to use to low cost in delays. In which that, they have to be importance in real time application.

4.3 Skype VOIP

In commercial VOIP network they have to examine the voice quality packets lost and delays, the skype service is used. The VOIP skype experiment can be seen in fig 4.13. The two users have to be connecting to each other into a LAN. One is connected to wireless excess network and other is connected to 3G wireless network. They have two users communicate through skype for signalling. After the connection there are two users are connected, first user are transmit and record signal and then second user record the signal, therefore signal have travel from user first to user second via router. Then user has to get relative excess point. The results have to be obtained an attenuation of fadings into the paths and they are studied, therefore we have to compare with original speech sample recorded at user first, two sets of measurements were obtained, first is indoor environment and 2 is outdoor environment, the average receiver signals have to be measure for remaining speechs.ther fore, the measurements have to there repetition for each sentences making a total of 12 reading of two sentences, the result are shown in fig 4.14,secondary out door environments have to receive 100% speech of signal ,if signal strength is -80 db , if the signal strength is below -80 db ,then we have to get reaming speech. If the signal strength is -85 db ,then twelve time into out of seven, therefore the skype have to log off ,then the internet connection has to lost and the devices have to be reconnect in to the wireless network. If the signal strength is -90 db then the system has no connect between wireless lanes and 3 G LTE networks.

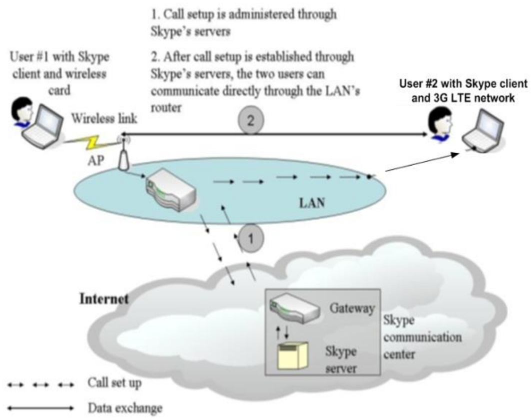


Figure 14. Two user interconnection with skype.

Connection of indoor environments , if the signal strength is -75 db m , then the totally speech signal has to be recived.it is because the signal strength has below -75 dbm , then they have has to be obtained , if the signal strength of -78 dbm , therefore , skype has to log off,it is lost and users device has to be reconnects, if the signal strength is -80 dbm , then they have to skypee users has log off the networks ,. In other hand, signal strength is -90 dbm then the users have to be no connections the between the wireless LANs and laptop. In general case, compare results between indoor environment and outdoor environment, it can say that, the outdoor environment are very performs results as compare to the indoor environment. For outdoor environment , it has not significations observation , they are transmits signal have to attenuations , in this case , the have to be one directs path and multipath effects are limited , in the 2nd case are observation, for indoor environments , they have observe significant ex wall , furniture, if the trans mit signal has be suffer from an attenuation feedings . The turn leads have been degradations as compare into outdoor case,

Examine The voice quality

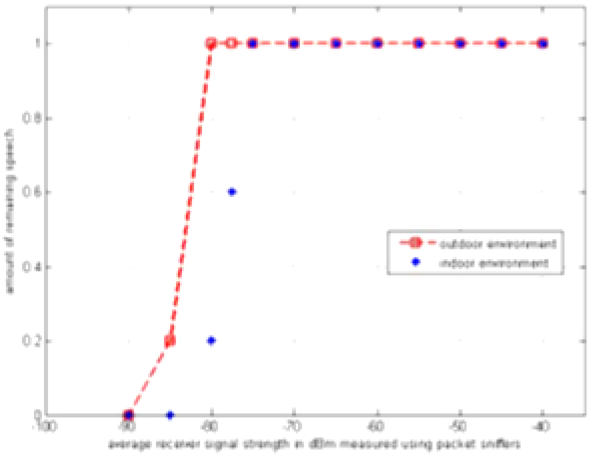


Figure 15. skypee experiment measurements of signal strength and remaining speech.

IV. CONCLUSION

The first was Matlab based and examined the effects of wireless channel, compression ratio and recognition quality of the received speech. The second consisted of experiments on commercial Skype VoIP networks in order to measure speech recognition and comprehension. The results of these simulations and experiments were reported The increase in the secondary path delay variation causes an increase in the BER of the signal. As the signal strength of the secondary paths increases, the BER increases as well .Performance of an outdoor wireless network was better than that of an indoor network due to the effect of multipath occurring indoors.

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