International Journal of Scientific Engineering and Technology (ISSN : 2277-1581)
Volume No.3 Issue No.10, pp : 1275-1279 1 Oct 2014

Effects of the Wireless Channel in VOIP (Voice Over Internet Protocol) Networks

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Abstract*: *In the present paper quality of voice has been analyzed in VoIP communications model. An investigation of IP-based voice communication with emphasis on the effects of a wireless channel on the quality of the received speech is attempted and the effects of voice signal compression and wireless channel conditions as well as channel coding on the voice quality and recognition are investigated through simulation. Simulation is implemented using Matlab and Speex software.

Key words: VoIP networks, RTCP, Speech compression, CELP, Packetizing & Depacketizing.

Introduction

IP Telephony is defined as the use of IP networks to transmit both voice and data packets. VON (or Internet Telephony) is used to describe the usage of the Internet to transmit both voice and data packets. VoIP is used to describe the usage of managed IP networks to transmit both voice and data packets (usually associated with Carrier-Class networks). In the course of History VON was the predecessor of VoIP, and its success led to the interest and development of IP Telephony and VoIP [1]. The effects of passive interruptions and communication delay on a phone conversation quality have been subject of investigation.[2]. The results indicate that there is a strong relationship between the number of passive interruptions on the conversation and the quality of the received speech. The factors analyzed are delay, jitter, packet loss, link errors, echo and Voice Activity Detection (VAD). [3-6].

Ways to smooth the negative effects of these factors are presented. An evaluation of real time control protocol's (RTCP) effectiveness is attempted in, and the results show that even though RTCP is effective for low delay networks, it can be inaccurate for networks with large, volatile delays [7]. Furthermore, the effects of different VoIP network architectures are investigated. The work of is further expanded with an experimental study on the RTCP effectiveness on networks with large propagation delays. [7].

Theoretical considerations

There are many different ways in which two or more users can be connected to a VoIP network, but the main concept of interconnection remains pretty much the same. First, a call control protocol is used to initiate the connection between the two users. After the connection has been established, the users can talk. As shown in Fig 1, the voice of one user is digitized, compressed and then packetized before being sent through a wired or wireless communication channel to the other user. At the other end, the opposite procedure is followed: the received packet is depacketized, decompressed, converted to analog form and then played back to the user. In order for the conversation to be natural, the same procedure must be followed in both directions so a full duplex communication is established [8].

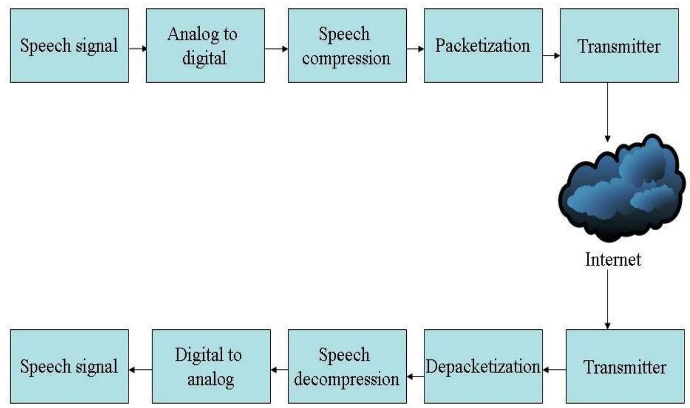

Fig 1: Basic VoIP Communication System

The simplest implementation that one can have includes two devices running a VoIP application separated by the internet. In order for the two users to communicate (voice), a logical connection must be initiated by a call control protocol. Then they have to be connected to local area networks, which in turn are connected with a gateway router to the internet. A simplified sketch of the above mentioned configuration can be seen in Fig.2 [9-10].

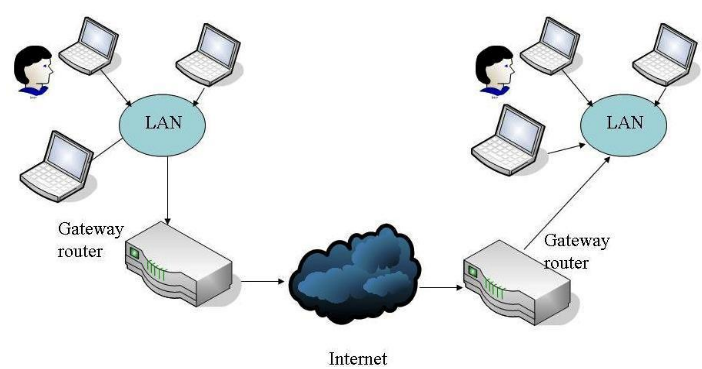

Fig 2: IP to IP VoIP Implementation Interconnecting two Different LANs with Internet

In order for VoIP to fully replace the traditional telephony, it has to get closer to its competitor's level of availability. Availability according to is given by Matlab was used in order to simulate a wireless VoIP network. The speech is digitized, compressed, packetized, and transmitted; the receiver then follows the reverse procedure. A speech recording is first input to Speex. [11] After the speech is compressed, it is exported to Matlab, which simulates the various fading channels. After passing through the simulated channel, the received signal is decompressed using Speex. The decompressed signal is input to the speech recognition software, which typically recognizes only a part of the speech sample depending on the distortion applied to the speech signal in the fading channel. The amount of recognized words in the distorted speech sample is compared to the amount of recognized words in the original speech sample.

Results

Four wireless channels were implemented for the needs of this simulation: Rician and Rayleigh fading channels in additive white Gaussian noise (AWGN) and Rician and Rayleigh channels in AWGN with convolutional coding. Additionally, a Matlab simulation was implemented to simulate a fading channel for an audio file without compression for the purpose of comparing results of the simulation.

The first simulation examined the effect of varying the *K* factor of a Rician channel on bit error rate. The results of the simulation are plotted in Figure 3 and were obtained based on averaging results from 50 Monte Carlo runs.

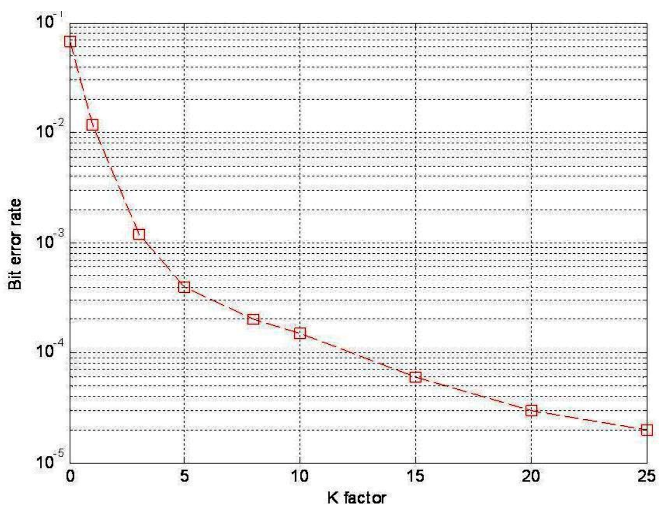

Fig. 3:The Bit Error Rate as a function of the Ratio of Dominant over Secondary Path (*K* factor) for the Rician Fading Channel based on 50 Monte Carlo Simulation Runs

By increasing the *K* factor, the BER decreases from a value close to 0.06 for $K = 0$ down to 10-5 for a *K* factor of 25. For K = 0, the channel becomes a Rayleigh channel, and, for $K = \infty$, it is an additive white Gaussian noise channel. The case of $K = 0$ represents the worst case scenario, which yields a high BER that renders it impractical for transmission of information. For an increase of the *K* factor of an order of magnitude, the improvement in BER is between one and two orders of magnitude.

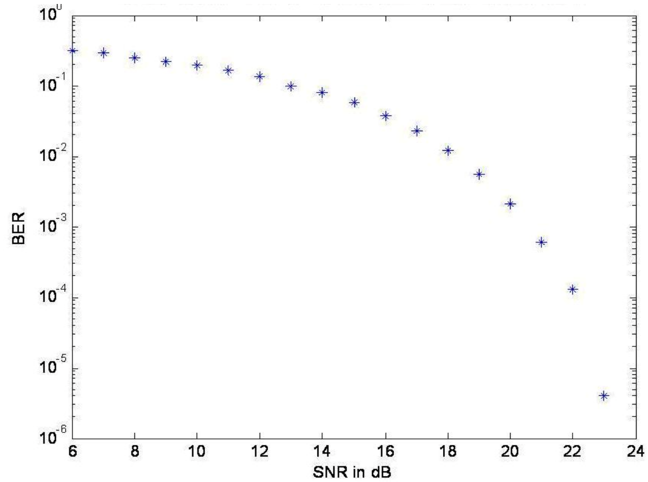

Fig 4: Effect of SNR on Bit Error Rate for a Rician Fading Channel based on 50 Monte Carlo Simulation Runs

The next is the simulation of a Rician fading channel without convolution coding to study the effects of SNR on the bit error rate. The results are plotted in Fig 4. At SNR = 6 dB, the BER = 0.5, which makes the channel inappropriate for transmission of information. As the SNR increases from 6 dB to 23 dB, the BER decreases and reaches values close to 10-6. For values of SNR between 6 dB and 16 dB, there is no significant improvement in the BER. For values of SNR of more than 16 dB, there is a rapid improvement in the BER. An improvement of about one order of magnitude is obtained for an increase in SNR from 22 dB to 23 dB. The general remark for this simulation is that, by increasing the SNR of the dominant path, the BER of the transmission decreases.

Comparison the BER plots of the compressed and uncompressed signal through the same channel, there is no significant difference between the two cases. What makes a difference is the amount of audible distortion caused in each case for the same amount of errors. Specifically, the uncompressed signal with a BER of 10-3 presents a noticeable amount of distortion but it is still understandable. On the other hand, the compressed signal is not even decodable. It is easy to realize that compressing a speech signal results in a gain in bit rate, but the signal becomes more sensitive to errors.

For all the cases shown in Fig 5, an increase in SNR leads to a decrease of BER regardless of channel coding or speech compression. This effect is due to multipath. As the dominant path becomes stronger, the uncertainty about ISI and consecutive pulse discrimination decreases. When the strength of the main path becomes strong enough, it is easier for the receiver to discriminate between a pulse and a delayed copy of a previous pulse.

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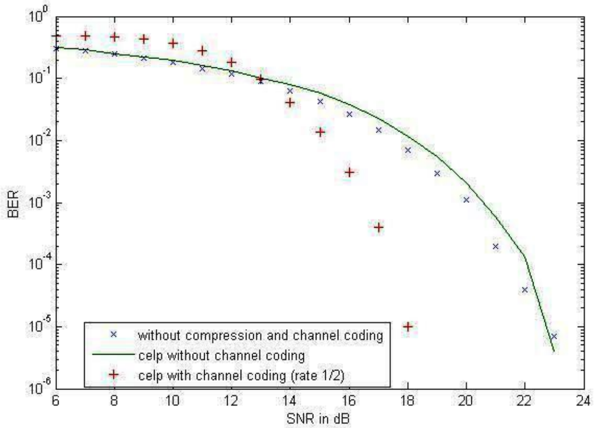

Fig 5: Effects of SNR on the BER for a Rician Fading Channel with Convolutional Coding, and Speech Coding based on 50 Monte Carlo Simulation Runs

Comparing the uncompressed and compressed speech, as the SNR of the dominant path increases, it is seen that, after a threshold value of SNR, there is a coding gain increase as the SNR increases. More specifically, after a SNR of 13 dB where there is no coding gain, an increase in coding gain is obtained. A maximum coding gain of 5 dB is obtained for BER = 10-5. Furthermore, one may notice that, before the threshold value of SNR = 14 dB, there is a negative coding gain, meaning that the results are better without channel coding rather than with it in Fig. 5.

In Fig 6 an increase in the secondary path delay variation causes an increase in the BER of the signal. After a 30 ns delay is inserted, it is noticed that the BER reaches a value close to 0.5. The reason for this result comes immediately from the effect of multipath. As the signal strength in the paths that the secondary signals follow becomes larger, the ISI distortion increases. When the delay variation of the secondary path becomes too large, the receiver is unable to discriminate between a pulse and the delayed copy of a previous pulse Fig 6.

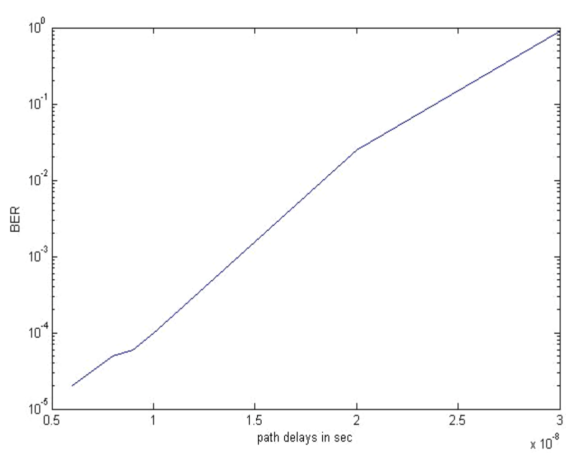

Fig 6: Effect of Secondary Path Delays on the BER of a Rician Fading Channel based on 50 Monte Carlo Simulation Runs

In Fig 7 as the signal strength of the secondary paths increases, the BER increases as well. This result is logical if one considers that as the secondary signals get stronger, they make the discrimination of a pulse and a delayed copy of a previous pulse a harder task for the receiver. For the specific channel, when the secondary paths are -7 dB weaker than the main path signal (which is 0 dB), it is impossible for the receiver to correctly detect the pulses (BER reaches 0.5). On the other hand, when the secondary paths are -15 dB weaker than the main path signals (which is 0 dB), the BER is 10-6.

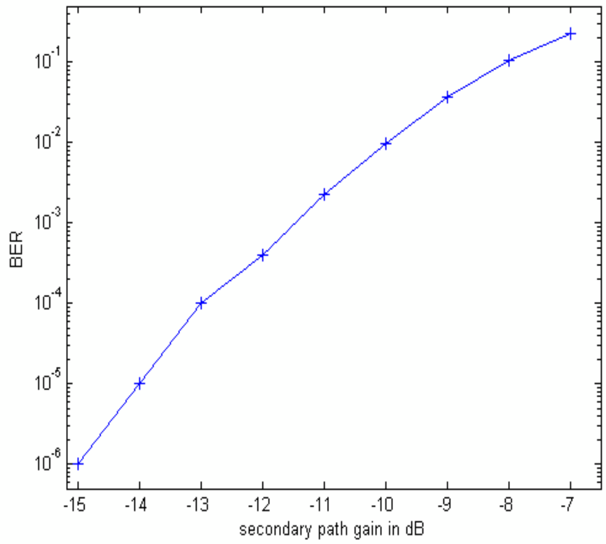

Fig 7: BER as a Function of Secondary Path Gain for a Rician Fading Channel based on 50 Monte Carlo Simulation runs.

Next, the effect of compression ratio on the speech quality is examined. Five different compression ratios were used, and the results of the simulation can be seen in Figure 8. Sixty Monte Carlo runs were used to calculate the average amount of remaining speech for each compression ratio. As the signal is compressed at higher rates, the amount of remaining speech becomes

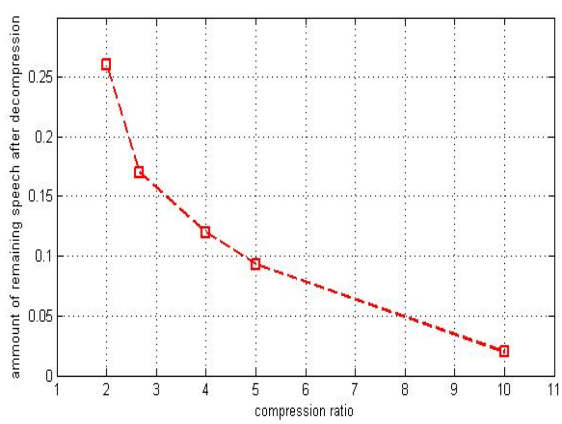

Fig 8: Effect of Compression Ratio on the Remaining Speech after Decompression based on 50 Monte Carlo Simulation Runs

smaller. It is expected that the more compressed a signal is, the more "sensitive" it is to the effects of errors. Every bit in a compressed signal represents a larger amount of data than in an uncompressed signal. Thus, when losing a bit that represents compressed data, the amount of information lost is much more than in the uncompressed case. For the simulation under discussion, by decreasing the compression ratio from 10:1 to 2:1, there is an increase in the amount of remaining speech from 0.01 to 0.26 of the original speech sample.

After determining the effects of compression on the amount of received speech, channel coding was introduced to determine its effects on the amount of remaining speech. The same setup as in the previous subsection was used, and a simulation of 50 Monte Carlo runs was executed. Two different rates were used for the convolution coding, ½ and ¾, and both gave the same results as in figure 9.

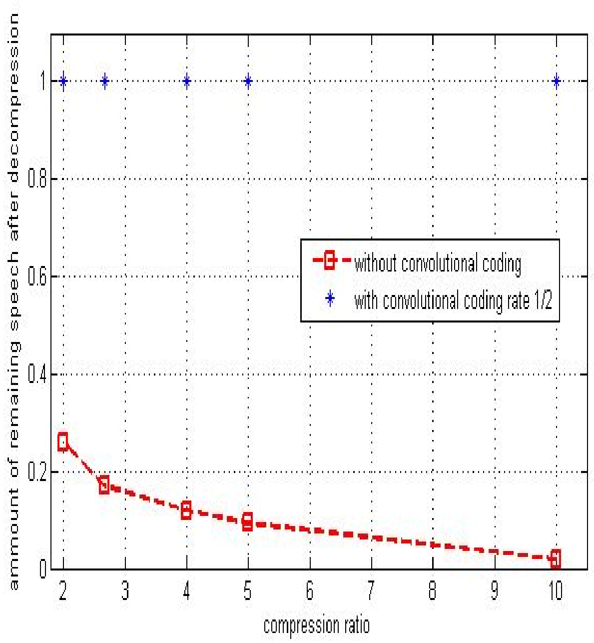

Fig 9: Effect of Compression Ratio on the Remaining Speech after Decompression with Channel Coding based on 50 Monte Carlo Simulation Runs

Another important conclusion comes from comparing the amounts of compression and their results with and without coding. It is preferable to use a high compression ratio (10:1) with convolutional coding rather than use a lower compression ratio (2:1) without convolutional coding. If a 10:1 compression ratio with a channel coding rate of ½ is used, a total of 6 kbps is transmitted, but 100% the speech is received. On the other hand, by using a 2:1 compression ratio without channel coding, the result is a total transmitted signal of 40 kbps, but the received speech is only 30% of the original speech. The drawback of channel coding is circuit complexity, cost, and delay, which are important in real-time applications.

Conclusions

This paper investigated the quality of received voice with emphasis on the effects of wireless channel, speech compression and channel coding. Matlab, Speex, and Dragon Naturally Speaking software were used to simulate VoIP communication. Matlab was used to simulate various wireless channels and Speex was used to compress speech signals using CELP. The simulation quantified the effects of wireless channel, compression ratio and channel coding on the received speech quality. The metrics used were the BER and the amount of speech that remained at the receiver‘s end. The wireless channels simulated were Rician fading channels with additive white Gaussian noise and Convolutional coding.

Simulations showed that for the Rician fading channel, an increase in the SNR causes a decrease in BER. There is no significant difference between the BER of the signal when transmitting compressed and uncompressed speech. What makes a difference is the amount of audible distortion caused in each case for the same amount of errors. 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 [12-17].

This study was based on simulation in Matlab. In this case, improvements as well as additions can be made. In this work, simulation was focused on a specific kind of baseband modulation, without investigating the effects of different modulation schemes on the quality of the received speech. It was observed though that different modulation schemes used in a wireless network can affect the network performance and thus the VoIP communication quality. We suggest an investigation, through simulation, of the effects of modulation on the received speech quality of VoIP over wireless communications. In the future by passing into the more channels we can obtain more and more significant results drastically.

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