International Journal of Engineering Research-Online
A Peer Reviewed International Journal
Articles available online http://www.ijoer.in

Vol.3., Issue.5., 2015
(Sept.-Oct.)

RESEARCH ARTICLE

ISSN: 2321-7758

PERFORMANCE EVALUATION OF DIGITAL AUDIO BASED COMPRESSED SPEECH TRANSMISSION OVER NOISY CHANNELS USING VARIOUS CODING ALGORITHMS

IQRA FAROOQ[1],AMARDEEP SINGH VIRK[1], [2*]MEHBOOB UL AMIN

[1]AIET Faridkot, ECE Department

[2*]University of Kashmir, Electronics and Communication Engineering

International Journal of Engineering Research-online (IJOER)
ISSN:2321-7758
www.ijoer.in

ABSTRACT

In this paper we evaluate the Bit-Error-Rate performance of the secure speech signal over various Digital Modulation Techniques and the results are plotted in terms of graphs. A basic speech synthesis model is developed using various Digital Signal Processing Operations and the performance is evaluated using various Modulation techniques. The various coding techniques are proposed to obtain the better performance of the proposed simulation model

Keywords—BER, FFT,BPSK, QPSK,DSP,MFCC

1. INTRODUCTION

Speech coders, whose goal is to represent the analog speech signal in as few binary digits as possible, can be described as belonging to one of three fundamentally different coding classes: waveform coders, Vocoders, and hybrid coders. A waveform coder attempts to mimic the waveform as closely as possible by transmitting actual time- or frequency-domain magnitudes. For example, in Pulse Coded Modulation (PCM), the input speech itself is quantized. In differential PCM (DPCM) and adaptive DPCM (ADPCM), the prediction residual is quantized. In addition, Subband Coders (SBC) are also waveform coders. Speech quality produced by waveform coders is generally high, although at high bit rates. Vocoders, or parametric coders, analyze the waveform to extract parameters that in some cases represent a speech production model. The waveform is synthetically reproduced at the receiver based on these quantized parameters. Vocoder types include formant, homomorphic Vocoders, as well as Linear Prediction Coding (LPC) and Sinusoidal Transform Coding (STC). Vocoders can generally achieve higher compression ratios than waveform coders; however, they provide more artificial speech quality. In hybrid coders, the high compression efficiency of Vocoders and high-quality speech reproduction capability of waveforms are combined to produce good quality speech at medium-to-low bit rates. The so-called analysis-by-synthesis coders, such as the Coded Excited Linear Prediction (CELP), GSM, and Mixed Excitation Linear Prediction (MELP) are all hybrid coders. Speech and audio coders can be evaluated in terms of five attributes: bit rate, speech quality, delay, complexity, and robustness to acoustic noise and channel errors. Speech quality can be measured subjectively and objectively. Subjective measurements are obtained from listening tests, whereas objective measurements are computed from the original and decoded speech signals. A popular measure is the signal-to-noise ratio (SNR), which is a long-term measure of the accuracy of reconstructed speech. Temporal variations can be better detected and evaluated using a short-time SNR for each segment of speech. The segmental SNR (SEGSNR) is then defined as the average of the short-time SNRs. Modified versions of SEGSNR will be used to evaluate the quality of coders. In this thesis we are going to develop efficient coding algorithms for the performance evaluation of speech transmission over fading channels in terms of Bit Error Rate (BER), SNR e.tc

Mel Frequency Cepstral Coefficients (MFCC)

The human ear resolves frequencies non-linearly across the audio spectrum. Empirical evidence suggests that designing a front-end to operate in a similar nonlinear manner improves recognition performance. A simple filter bank designed to provide non-linear resolution on the Mel frequency scale can be used. With MFCCs, the filters used are triangular, and they are equally spaced along the Mel scale. To implement this filter bank, the window of speech data is analyzed using a Fourier transform, and the magnitude coefficients are correlated with each triangular filter and accumulated. Each bin holds a weighted sum representing the spectral magnitude in that filter bank channel. The magnitudes of the power estimates from each channel are finally compressed using a logarithmic function. The resulting spectral estimates reflect two of the most studied aspects of auditory signal processing: frequency selectivity and magnitude compression. Because the spectral estimates are somewhat smooth across filter number and highly correlated, each frame is roughly decorrelated using the Discrete Cosine Transform (DCT) to obtain the Mel-Frequency Cepstral Coefficients (MFCCs)

II.RELEATED WORK

In [1] Cyclic Channel Coding algorithm for Original and Received Voice Signal at 8 KHz using BER performance through Additive White Gaussian Noise Channel is proposed and it is seen that Cyclic Code reduces the effect of error on transmitted signal caused by AWGN channel.

In [2] Rashed et al have investigated the impact of Forward Error Correction (FEC) codes namely Cyclic Redundancy Code and Convolution Code on the performance of OFDM wireless communication system for speech signal transmission over both AWGN and fading (Rayleigh and Rician) channels in terms of Bit Error Probability.. From the simulations it is observed that Convolution coding out-performs the conventional Cyclic Redundancy Code

Nasr M.E et al [3] proposed Performance of coded speech modulated signals through Rician-fading mobile channel. A modified approach for estimating the bit-error-rate of modulated transmitted speech signal through an independent Rican Fading channel is described. Diversity technique is proposed to improve SNR. The amount of the minimum value of channel SNR required for obtaining good speech quality is proposed.

In [4], authors consider the performance of coded OFDM using turbo-codes, for application in digital broadcasting systems. Authors showed that turbo codes can give performance improvements of some order of dB on a Rayleigh fading channel, over the conventional convolutional codes in the existing standards

M. K. Gupta, et.al. Illustrates in [5] the way to increase the system throughput while maintaining system performance under desired bit error rate. From their study by simulation it is concluded that it is possible to improve the performance of uncoded OFDM can be improved by convolution coding scheme.

In [6], performance of Interleaved CRC encoded QPSK based wireless communication system are analyzed. Author found by simulation that interleaved CRC encoded QPSK based system provides unique performance in proper identification and retrieval of transmitted color image.

III. METHODOLOGY

First of all we will acquire audio voice.wav format and plot its Welch power spectral density (PSD) The Sampling Frequency used is 8000 Hz. To get pitch of the recorded voice autocorrelation is used. Then we draw the frequency Spectrum of the wave using Fast Fourier Transform (FFT).To get the samples in time domain IFFT of the recorded waveform is drawn .We obtain the random data of these samples in 0's and 1's The signal is modulated using QPSK Modulator and passed through various noisy channels and then we will calculate the bit-error-rate for each channel.

Convolutional coding

Convolutional codes are generated by the convolution of the input sequence with the impulse response of the encoder. The encoder accepts blocks of *k-bit* input samples and, by operating on the current block of data and the *m* previous input blocks, produces an *n-bit* block of output samples.

The coding rate of the encoder is given by the ratio *Rc=k/n* and the convolutional encoder is specified by these three parameters (*n,k,m*).

Viterbi decoder

We improve BER performance results by using a soft-decision decoding algorithm. In soft-decision decoding, the demodulator maps the received signal to log-likelihood ratios. These probability measures are based on the logarithm of the likelihood that the correct data are received instead of corrupted data. When log-likelihood ratios are provided as the input to the Viterbi decoder, the BER performance of the decoder is improved. An algorithm can be made to perform soft-decision Viterbi decoding by changing a few demodulator and Viterbi-decoder System-object parameters.

Space-time Coding

The space-time coding techniques are used to exploit the diversity from channels. Many traditional space-time codes were used to extract spatial diversity from flat-fading MIMO channel. Whereas, they are not effective at extracting the additional frequency (multipath) diversity of a frequency-selective fading channel. In general, the maximum achievable diversity order may reach the product of the number of transmit antennas, the number of receive antennas, and the number of resolvable propagation paths (the channel impulse response length). In order to achieve such high orders, the transmitted symbols must be properly spread over the carriers and the transmit antennas. An important strategy used to map the information symbols on the tones and antennas is the space-time-frequency code. It is used to extract both spatial and frequency diversity. The design of space-frequency and space-time-frequency codes is currently an active area of research

IV. ALGORITHM FOR PROPOSED WORK

Step 1 Generate the waveform of recorded speech signal " Iqra" in Matlab

Load handel.mat

Iqra= 'handel.wav';

[y, Fs] = wavread (Iqra)

Step 2. Plot the FFT of the Waveform at Mel frequency

z=fft (y.*hamming (length(y))).

c=dct (Z).

Step 3 .Generate the Discreet samples of the Waveform

x=Fs.*ifft (c, Fs).

Stem(x)

Step4. Generate Random bits

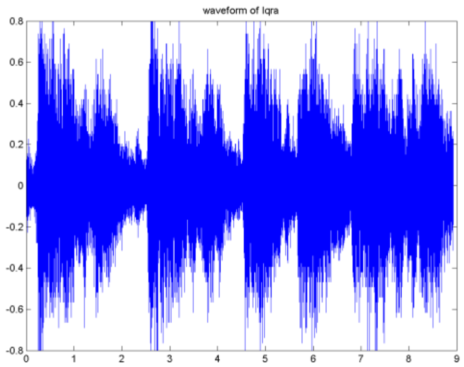
Step5. Modulator using QPSK Modulator

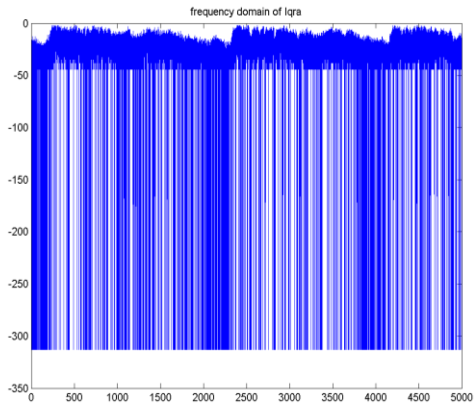
Step.6 Pass the data through various noisy channels

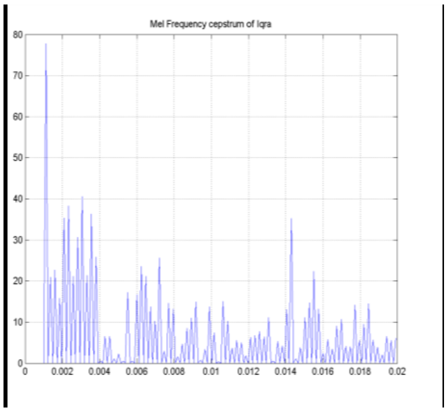
Step7. Calculate the Bit-Error-Rate

Step 8.Apply various error detecting and correcting techniques like convolutional coding and virtebri decoding algorithm for the Performance Evaluation.

V. RESULTS AND DISCUSSIONS







The simulation result for BER vs. SNR for BPSK modulation is shown in figure 2 .From the fig it is seen that theory and simulation curves almost coincide with each other. At higher SNR the simulation slope for BER increases linearly and then gradually falls.

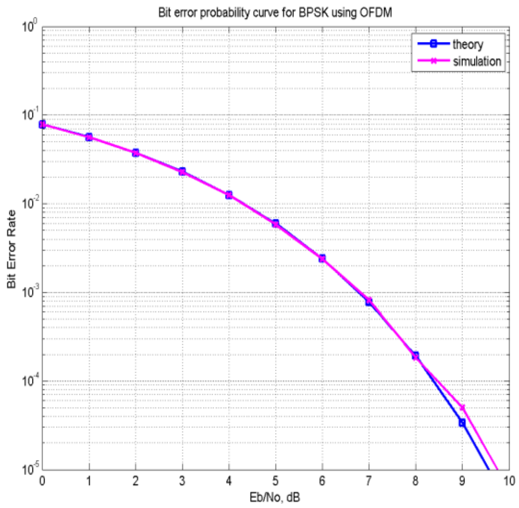


Figure2 BER vs. SNR for BPSK modulation.

Figure 3 shows the simulation result for BER vs. SNR for QPSK modulation. From the figure it is obvious that the slope of simulation curve falls gradually compared to the theoretical value resulting in the decrease in bit error rate. Also at higher SNR the simulation curve shows a greater deviation from theory values.

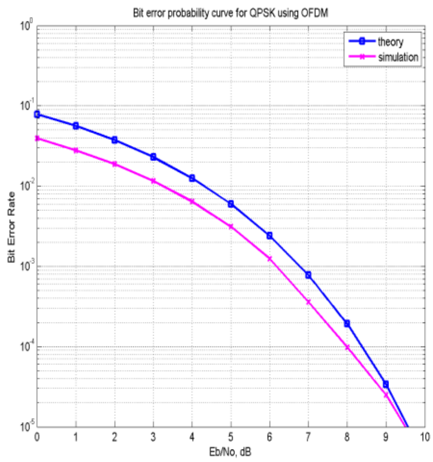


Figure 3 BER vs. SNR for QPSK modulation.

Figure 4 shows the BER vs. SNR results for 16 PSK modulation scheme. From the figure it is seen that simulation curve falls as compared to theory curve. .At SNR of 9db the curve gradually rises almost touching the theoretical value and then falls linearly.

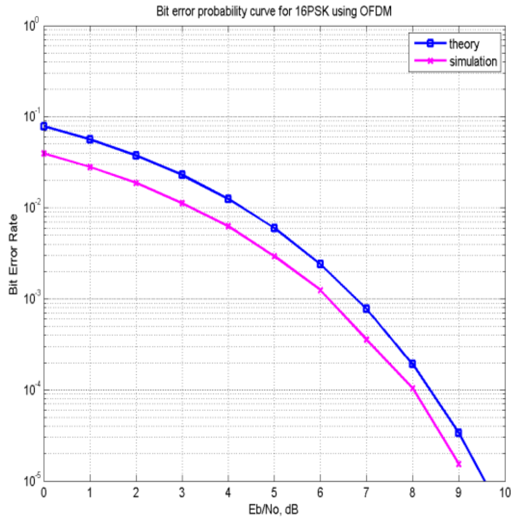


Figure 4. BER vs. SNR for 16 PSK.

Figure 5 shows the results for 32 QAM modulation. From the figure it is seen that the simulation curve falls as compared to theory curve resulting in decrease in Bit error rate of the system. At a SNR of 9db.the curve is almost parallel to x axis and then the curve falls with a linearly reaching a minimum value of bit error rate and remains constant there at higher signal –to-noise values.

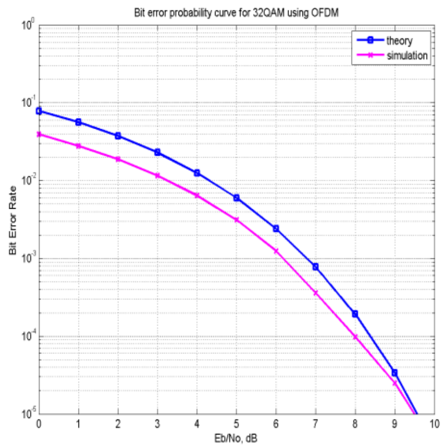


Figure 5. BER vs. SNR for 32 QAM modulation`

Figure 6 shows the BER vs. SNR result for 64 QAM modulations. It is seen from the simulation result that the simulation curve falls as compared to theory curve showing a deviation and decrease in bit error rate of system. At SNR of 8 db the slope tends to fall and at 9 db the slope falls linearly deviating more from theoretical value. At 9.5 db the slope is almost zero with bit error rate remaining almost constant and then curve falls again linearly and bit error rate reaches a minimum value. At 10db the curve again starts to rise and then again falls at higher signal to noise ratio.

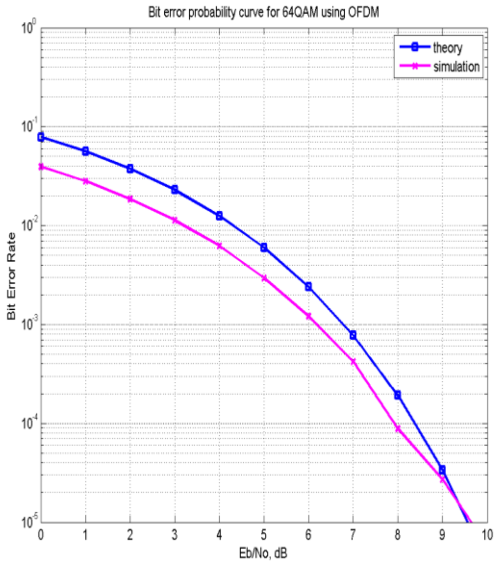


Figure 6. BER vs. SNR for 64 QAM modulation.

VI. CONCLUSION

We have presented BER VS SNR of the Secure Speech Signal and evaluated the performance of the proposed model. Performance is evaluated over various digital modulation techniques –BPSK, QPSK, 16-PSK,32QAM,64QAM.64-QAM shows the better performance with the significant decrement of BER at higher SNR.

VII. REFERENCES

[1]. B. Strope, Modeling auditory perception for robust speech recognition, Ph.D. thesis, University of California, Los Angeles, 1999.

[2]. A.M. Kondoz, Digital speech coding for low bit rate communication systems, Wiley, England, 1995.

[3]. S. Young, D. Kershaw, J. Odell, D. Ollason, V. Valtchev, and P. Woodland, The HTK Book (for HTK Version 3.0), July 2000.

[4]. G. Fant, Acoustic theory of speech production, Mounton and Co., Gravenhage, The Netherlands, 1960.

[5]. J. Flanagan, Speech analysis, synthesis and perception, Springer-Verlag, New York, 1972.

[6]. L. Rabiner and B.H. Juang, Fundamentals of speech recognition, Prentice Hall, Englewood, New Jersey, 1993.

[7]. B.W. Kleijn and K.K. Paliwal, Speech coding and synthesis, Elsevier, Amsterdam, Netherlands, 1995.

[8]. T. Painter and A. Spanias, "A review of algorithms for perceptual coding of digital audio signals," in International Conference on Digital Signal Processing Proceedings, July 1997, vol. 1, pp. 179–208.

[9]. A. Alwan, S. Narayanan, B. Strope, and A. Shen, Speech production and perception models and their applications to synthesis, recognition, and coding, chapter in book "Speech processing, recognition, and artificial neural networks", pp. 138–161, Springer-Verlag, UK, 1999.

[10]. E. Zwicker and H. Fastl, Psychoacoustics, Springer-Verlag, Berlin, Germany, 1990.

[11]. B. Moore, An introduction to the psychology of hearing, Academic Press,

[12]. London, UK, 1989.

[13]. C.D. Geisler, From sound to synapse: Physiology of the mammalian ear, Oxford University Press, New York, 1998.

[14]. H. Fletcher, "Loudness, masking and their relation to the hearing process and the problem of noise measurement," Journal of the Acoustical Society of America, vol. 42, pp. 275–293, 1938.

[15]. J.B. Allen, "Fletcher's role in the creation of communication acoustics,"Journal of the Acoustical Society of America, vol. 99, pp. 1825–1839, 1996.

[16].] S.S. Stevens and H.W. Davis, Hearing, John Wiley & Sons, New York, 1938.

[17]. H. Fletcher, "Auditory patterns," Review of Modern Physics, pp. 47–65, 1940.

[18]. K. Brandenburg and G. Stoll, "ISO MPEG-1 audio: A generic standard for coding of high quality digital audio," Journal of Audio Engineering Society, vol. 42, no. 10, pp. 780–792, Oct. 1994.

[19]. S. Shlieng, "Guide to MPEG-1 audio standard," IEEE Transactions on

Broadcasting, vol. 40, no. 4, pp. 206–218, Dec. 1994.

[20]. M.R. Schroeder, “A brief history of speech coding,” Proceedings International Conference on Communications, pp. 26.01.1–4, Sept. 1992.

[21]. A. Gersho, “Advances in speech and audio compression,” IEEE Transactions on Speech and Audio Processing, vol. 82, no. 6, pp. 900–918, Jun1994.

[22]. N. Jayant, “Signal compression: Technology targets and research directions,” IEEE Journal on Selected Areas in Communications, vol. 10, no. 5, pp. 796–818, June 1992.

[23]. T. Kailath, Lectures on Wiener and Kalman filtering, Springer Verlag, New York, 1981.

[24]. L.R. Rabiner and R.W. Schafer, Digital signal processing

[25]. G.D. Forney, “The Viterbi algorithm,” IEEE Transactions on Communications, vol. 61, no. 3, pp. 268–278, Apr. 1973.

[26]. C.-E. Shannon, “A mathematical theory of communication,” Bell Syst. Technical Journal, vol. 27, pp. 379–423, 1948.

[27]. W.W. Peterson, Error-correction codes, MIT Press and Wiley & Sons, 1961.

[28]. C.-E. Shannon, “A mathematical theory of communication,” Bell Syst. Technical Journal, vol. 27, pp. 623–656, 1948.

[29]. J. Proakis, Digital communications, McGraw-Hill, 1995.

[30]. S. Wilson, Digital modulation and coding, Prentice Hall, 1996.

[31]. S. Lin and D. Costello, Error control coding: Fundamentals and applications, Prentice Hall, New Jersey, 1983.

[32]. G. Ungerboeck, “Channel encoding with multilevel/phase signals,” IEEE Transactions on Information Theory, vol. 28, no. 1, pp. 55–67, Jan. 1982.