Chapter 2
Studies and Implementation of Subband Coder and Decoder of Speech Signal Using Rayleigh Distribution

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Abstract In the last 40 years a number of coding techniques for analog sources (speech and images) has been employed. Subband coding, a kind of transform coding, splits analog speech signal into a number of different smaller frequency bands. By subbanding data rate has been reduced to 12.13804 Kbps [Sangita et al. Studies and implementation of subband coder and decoder of speech signal, Proceedings of national conference on electronics, communication and signal processing, 8–16, 1] on 64 Kbps telephone line. In this paper a method has been proposed by which data rate has been reduced to 9.4875 Kbps using Rayleigh distribution where data rate can be reduced to 9.6 Kbps [Crochiere et al. Digital coding of speech in subbands, The BELL System Technical Journal, 2]. Proposed method can save data rate which in turn saves bandwidth as well as spectrum. Moreover this proposed method provides acceptable probability of error and quantization noise i.e. SNR.

Keywords DM · PCM · SNR · Subband · Probability of bit error

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2.1 Introduction

Sub-band coding (SBC) is a kind of transform coding [3, 4]. A signal is divided into a number of different frequency bands and encodes each one independently. It enables a data compression by discarding information about frequencies which are masked. The result differs from the original signal, but if the discarded information is chosen carefully, the difference will not be noticeable, or more importantly, objectionable [5–7]. A paper—"**A low-complexity audio data compression technique using subband coding (SBC) and a recursively indexed quantizer (RIQ)**" compared SBC and RIQ to conventional coding techniques with SNR 2–5 dB higher than that of other coders of similar computational complexity of wideband audio signals [8]. The basic concept of "**Frequency Domain Coding of Speech**" methods is to divide the speech into frequency components by a filter bank (sub-band coding), or by a suitable transform (transform coding), and then encode them using adaptive PCM. Recent developments and examples of the "Vocoder-driven" adaptive transform coder for low bit-rate applications is also discussed [9]. In "**Subband Coding of Speech Signals Using Decimation and Interpolation**"—**a** structure of a two-channel quadrature mirror filter with low pass filter, high pass filter, decimators and interpolators, is proposed to perform subband coding of speech signals in the digital domain. The results show that the proposed structure significantly reduces the error and achieves considerable performance improvement compared to delta-modulation encoding systems [10]. Subband coder reduces and controls quantization noise. Here bit allocation on each subband is done on perceptual criterion. So that quality of the coded signal is improved over the full spectrum coding. Computer simulated data provides 16 and 9.6 Kbps over 64 Kbps data rate [2].

2.2 Basic Idea of the System

Figure 2.1a is an example of Power Spectral Density (PSD) of an Voice Signal. Here, Voice signal has been considered to be restricted to 3.5 kHz only. Power Spectral Density to be in watt/HZ or dB. In this figure frequency axis is divided into number of subbands (say 0–f1, f1–f2, f2–f3, f3–f4, etc.). The frequency band (0–f_1) is base band signal whereas (f_1–f_2), (f_2–f_3), (f_3–f_4), etc. are band pass signals. Each band will translated to baseband by multiplying with lowest frequency component of the said subband. Here seven subbands have been considered (Fig. 2.1b).

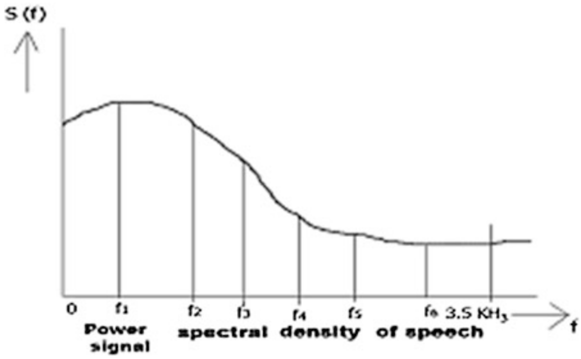

Fig. 2.1a Power spectral densities versus frequency of speech signal using subband

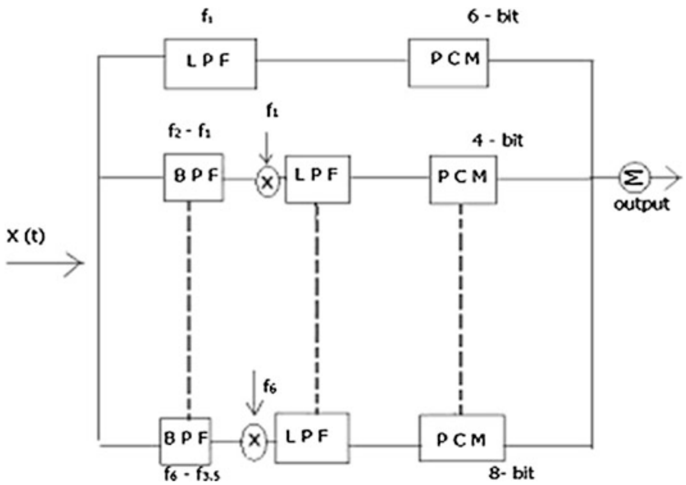

Fig. 2.1b Block diagram of subband coding transmitter

Transmitter consists of one LPF and six BPFs. All BPFs outputs are multiplied by the lowest frequency component of those bands at the multiplier block. Then outputs are PCM and then added by summer. Finally the summed output is put into channel.

At the receiver signals are decoded by seven decoders. Then each signal is passed through LPF of cut-off frequency f_1, f_2–f_1, f_3–f_2 etc. From second to

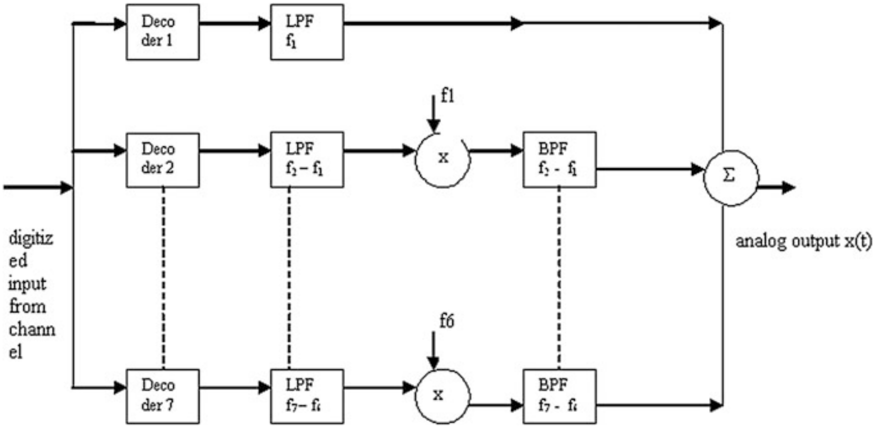

Fig. 2.1c Block diagram of receiver

seventh signal outputs are multiplied by their respective lowest frequency components and then passed through BPFs of f_2–f_1, f_3–f_2 etc. Then the outputs are summed up to get the replica of the original signal. Data rate from the above signal is reduced from 64 to 19.5 Kbps [1]. Data rate can be further reduced, if Power spectral density of voice signal is multiplied by the probabilities of occurrences which is 12.13804 Kbps lower than 19.5 Kbps by using MATLAB Simulation [1]. Figure 2.1c shows SNR, quantization noise produced out of subbanding.

2.3 Proposed Scheme

2.3.1 Case I: Subbanding Data Rate 12.0128 Kbps

Matlab simulation of subband data rate [1] has been reduced to 12.0128 Kbps where existing telephone line data rate 64 Kbps. Figures 2.2a, 2.2b, 2.2c, 2.2d have been shown as MATLAB simulation of data rate, cumulative data rate, SNR and probability of bit error of subbands respectively [11].

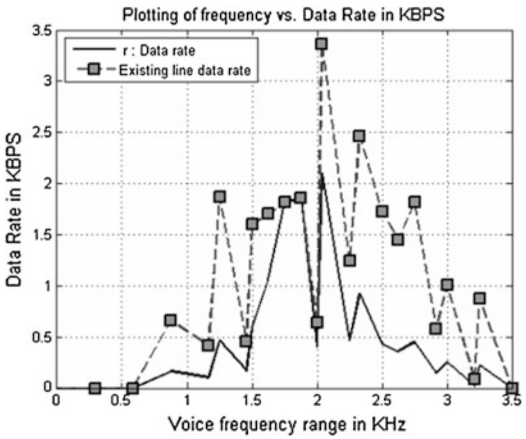

Fig. 2.2a Matlab simulation output of frequency versus data rate

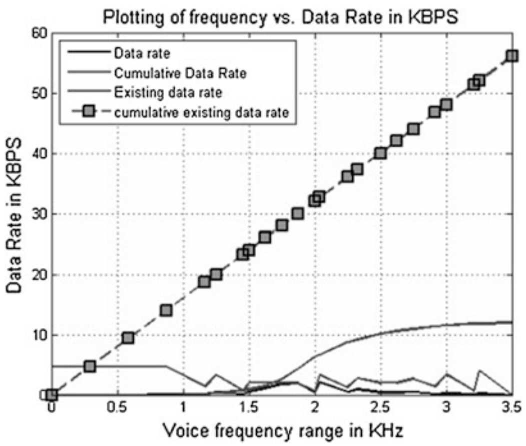

Fig. 2.2b Matlab simulation output of frequency versus cumulative data rate

2.3.2 Case II: Subbanding Data Rate 11.0959 Kbps

By changing the bit allocation by perceptual criterion data rate has been reduced to 11.0959 Kbps from 12.0128 Kbps. Figure 2.3a, 2.3b, 2.3c, 2.3d have been shown as MATLAB simulation of data rate, cumulative data rate, SNR and probability of bit error of subbands respectively.

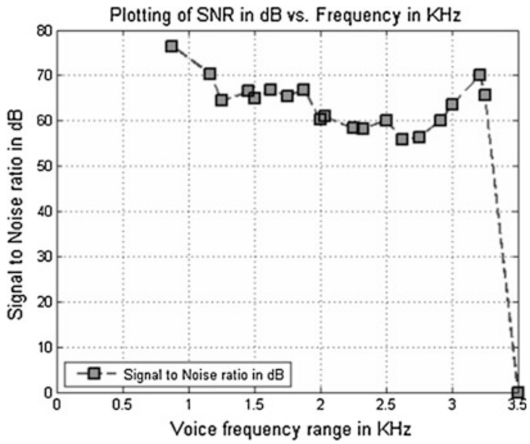

Fig. 2.2c Matlab simulation output of frequency versus SNR

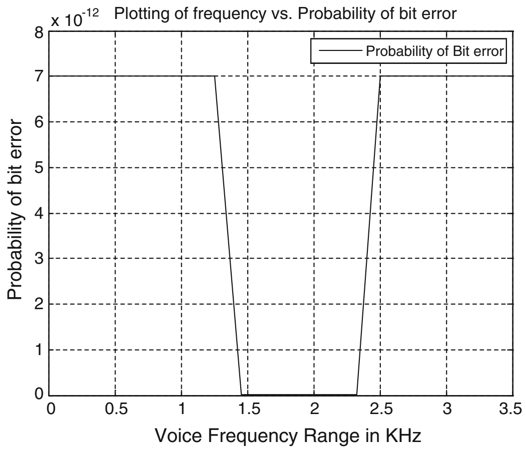

Fig. 2.2d Probability of bit error

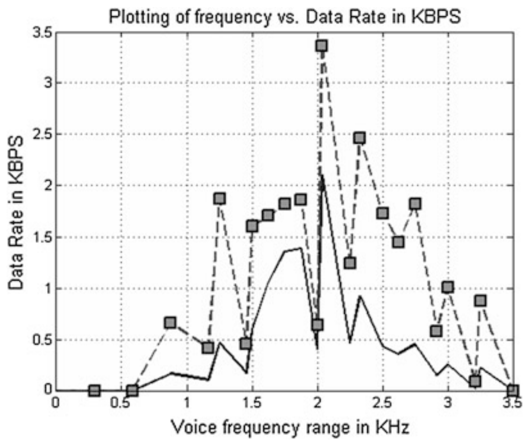

Fig. 2.3a Matlab simulation output of frequency versus data rate

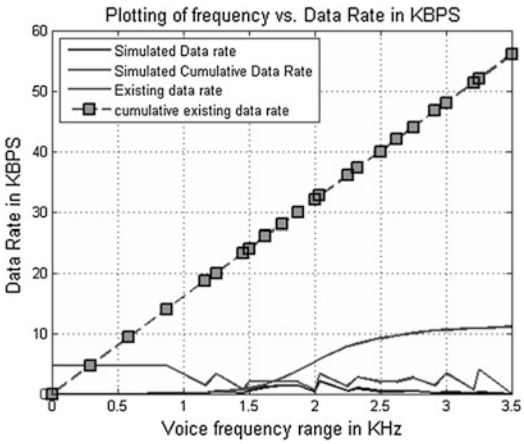

Fig. 2.3b Matlab simulation output of frequency versus cumulative data rate

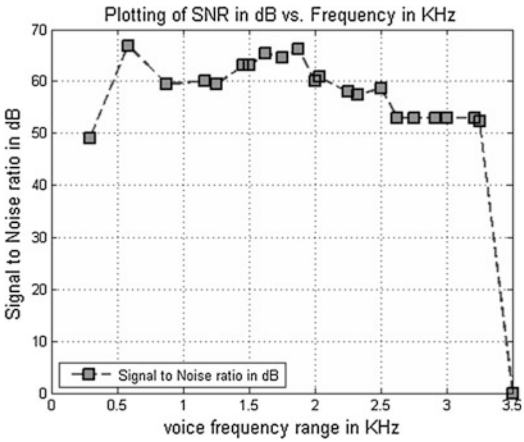

Fig. 2.3c Matlab simulation output of frequency versus SNR

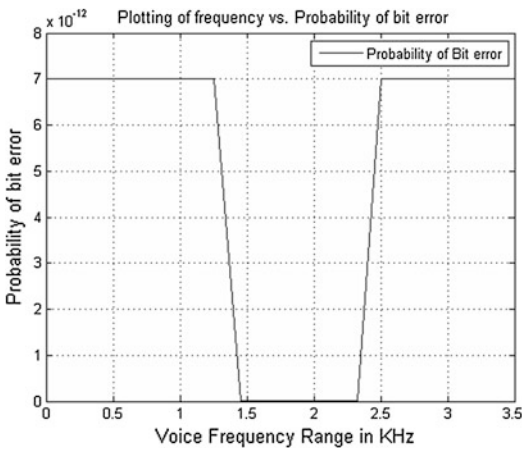

Fig. 2.3d Probability of bit error

2.3.3 Case III: Subbanding Data Rate 10.0494 Kbps

By changing the bit allocation by perceptual criterion data rate has been reduced further to 10.9494 from 11.0959 Kbps. Figures 2.4a, 2.4b, 2.4c, 2.4d have been shown as MATLAB simulation of data rate, cumulative data rate, SNR and probability of bit error of subbands respectively.

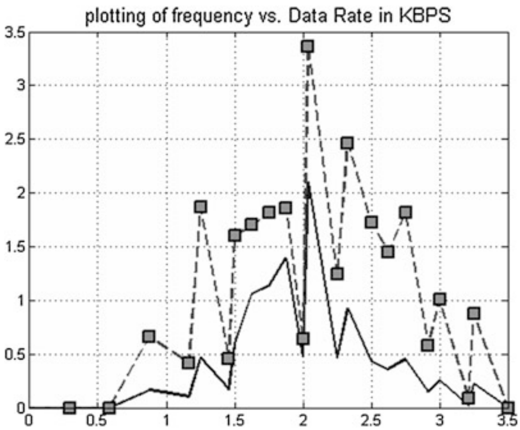

Fig. 2.4a Matlab simulation output of frequency versus data rate

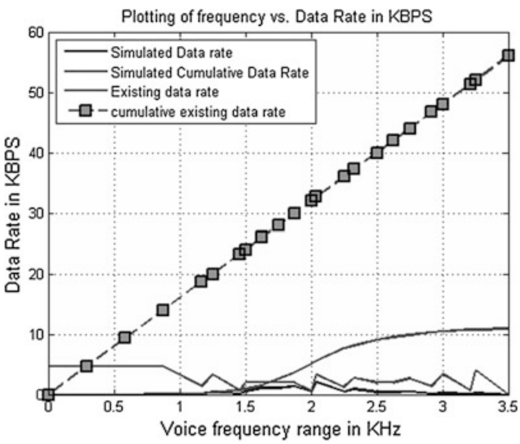

Fig. 2.4b Matlab simulation output of frequency versus cumulative data rate

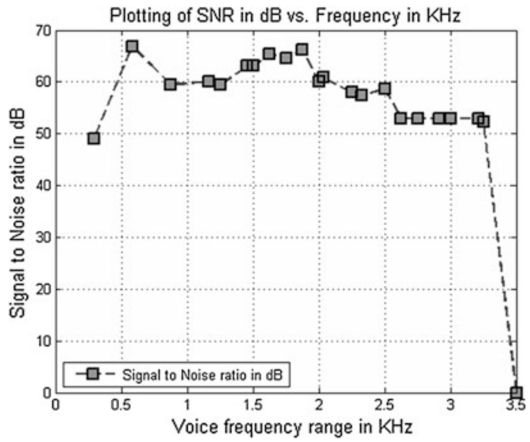

Fig. 2.4c Matlab simulation output of frequency versus SNR

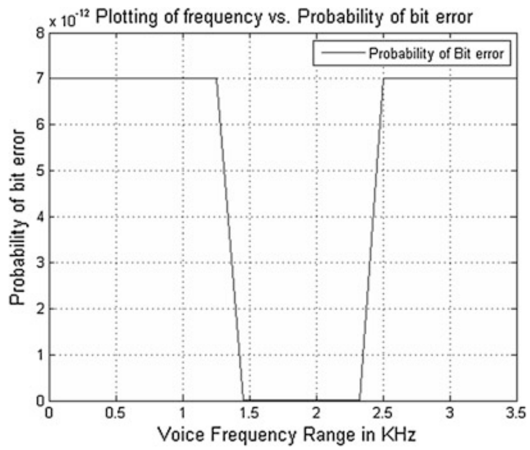

Fig. 2.4d Probability of bit error

2.3.4 Case IV: Subbanding Data Rate 10.7175 Kbps

By changing the bit allocation by perceptual criterion data rate has been reduced to 10.7175 Kbps from 10.9494 Kbps. Figures 2.5a, 2.5b, 2.5c, 2.5d have been shown as MATLAB simulation of data rate, cumulative data rate, SNR and probability of bit error respectively of subbands.

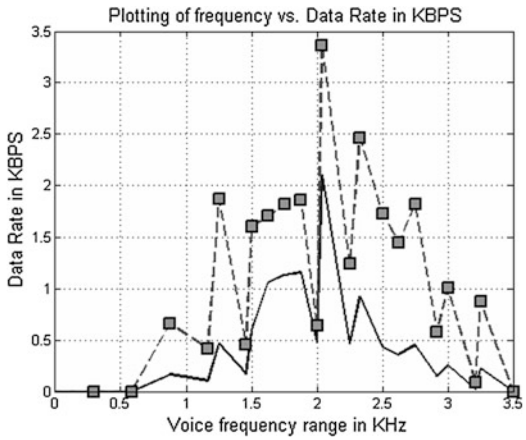

Fig. 2.5a Matlab simulation output of frequency versus data rate

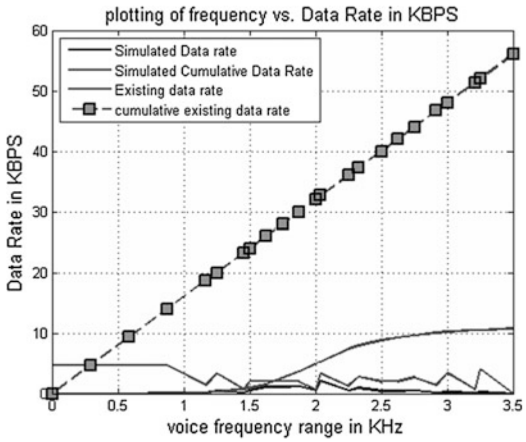

Fig. 2.5b Matlab simulation output of frequency versus cumulative data rate

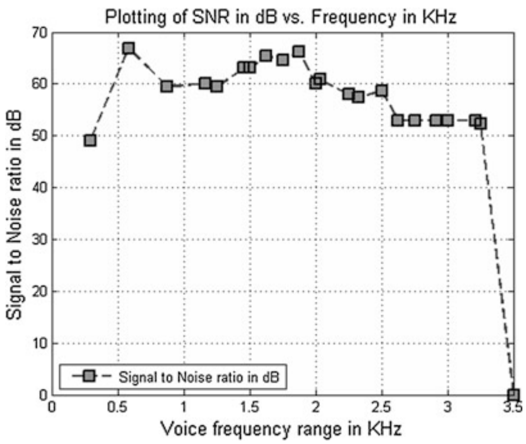

Fig. 2.5c Matlab simulation output of frequency versus SNR

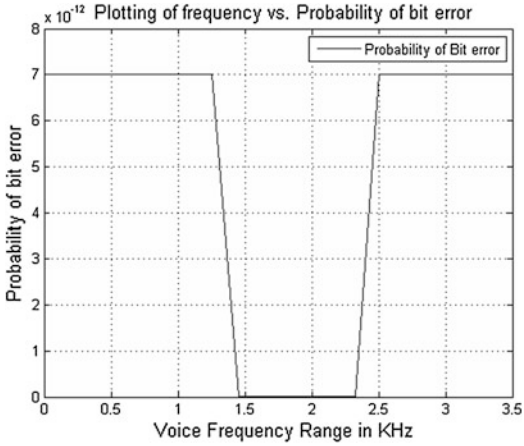

Fig. 2.5d Probability of bit error

2.3.5 Case V: Subbanding Data Rate 10.4867 Kbps

By changing the bit allocation by perceptual criterion data rate has been reduced to 10.4867 Kbps from 10.7175 Kbps. Figures 2.6a, 2.6b, 2.6c, 2.6d have been shown as MATLAB simulation of data rate, cumulative data rate, SNR and probability of bit error respectively of subbands.

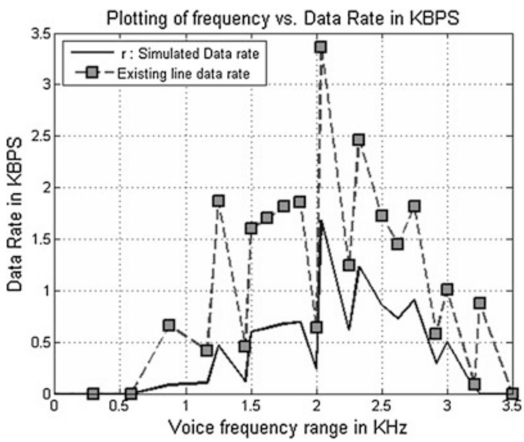

Fig. 2.6a Matlab simulation output of data rate versus frequency

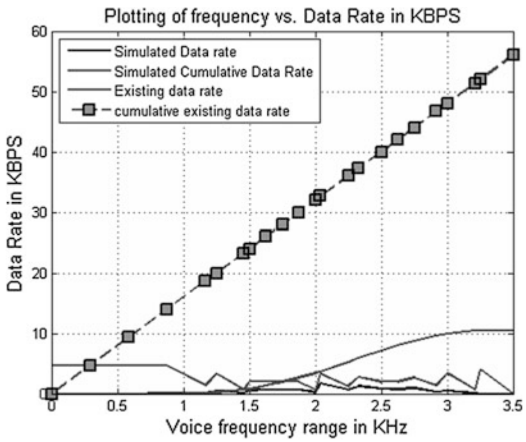

Fig. 2.6b Matlab simulation output of frequency versus cumulative data rate

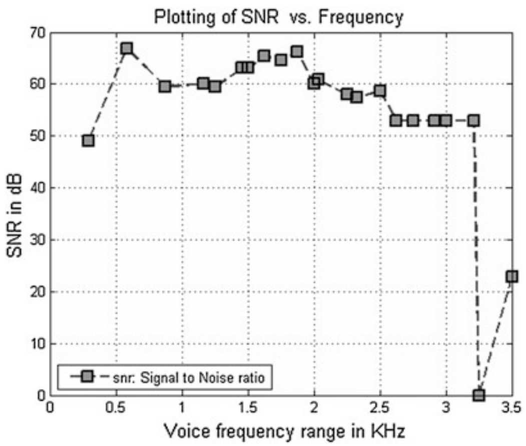

Fig. 2.6c Matlab simulation output of SNR versus frequency

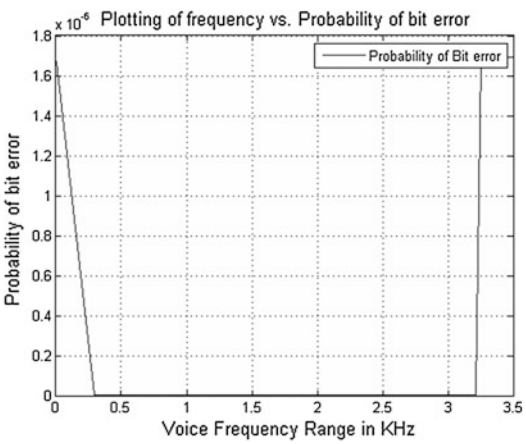

Fig. 2.6d Probability of bit error

2.3.6 Case VI: Subbanding Data Rate 9.4875 Kbps Using Rayleigh Distribution

Changing the bit allocation by perceptual criterion and Rayleigh distribution Power Spectral Density, data rate has been reduced to 9.4875 Kbps from 10.4867 Kbps which is lowest in this study [12]. Figures 2.7a, 2.7b, 2.7c, 2.7d have been shown as MATLAB simulation of data rate, cumulative data rate, SNR and probability of bit error respectively of subbands.

The above studied cases are tabulated in Table 2.1.

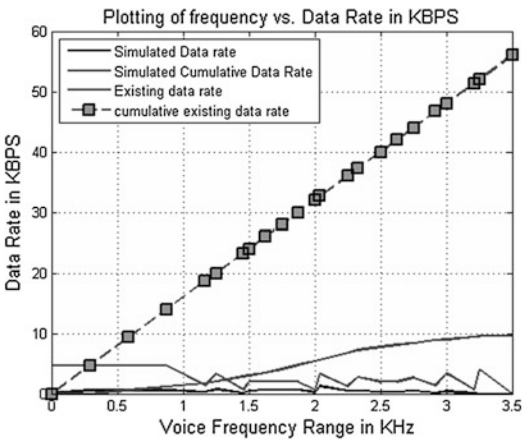

Fig. 2.7a Matlab simulation output of frequency versus cumulative data rate

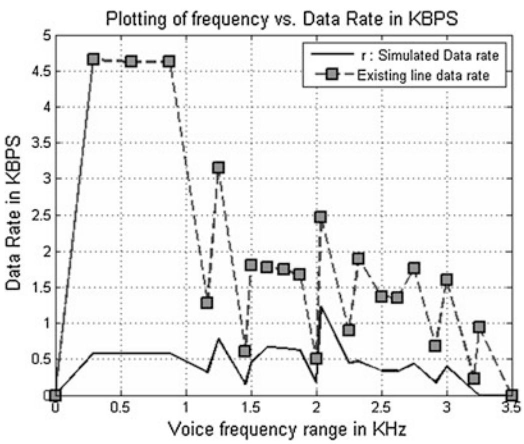

Fig. 2.7b Matlab simulation output of frequency versus data rate

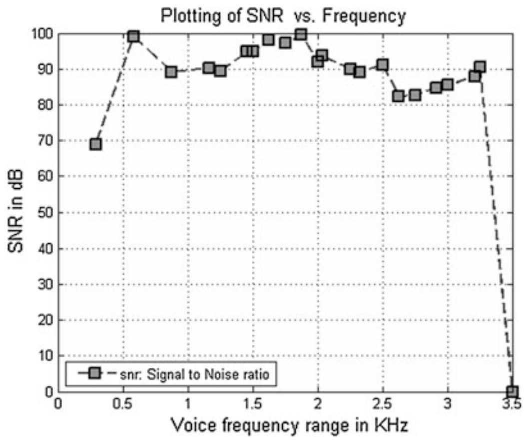

Fig. 2.7c Matlab simulation output of frequency versus SNR

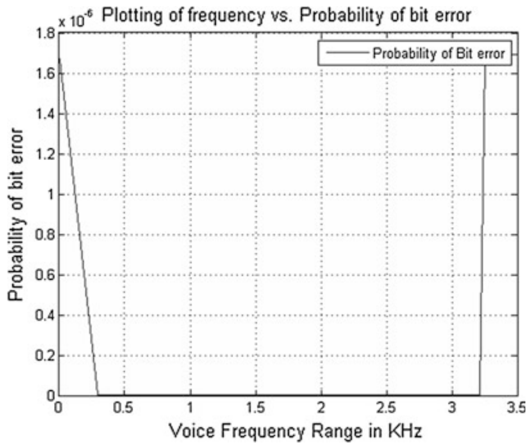

Fig. 2.7d Probability of bit error

2.4 Conclusion and Future Work

The new proposed method reduces data rate by using Rayleigh distribution. Several bit combinations by perceptual criterion have been taken. Result of those combinations have been tabulated in Table 2.1. It is clear that all the cases reduces the data rate drastically without losing signal to noise ratio criterion as well as probability of bit error. Rayleigh distribution outperforms all the other cases. In future authors would like to optimize those schemes by different algorithms.

Table 2.1 Subband coding data rate comparative table

Exixting data rate in Kbps	Simulated data rate in Kbps [1]	Simulated data rate in Kbps	Simulated data rate in Kbps	Simulated data rate in Kbps	Simulated data rate in Kbps	Simulated data rate in Kbps	Result in Kbps [2]
	Case I	Case II	Case III	Case IV	Case V	Case IV	
64	12.0128	11.0959	10.9494	10.7175	10.4867	9.4875	9.6

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