A Lower Capacity Bound of Secure End to End Data Transmission via GSM Network

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Abstract—**Global System for Mobile communications (GSM) is a widely spread, reliable and P2P channel through all over the world. These characteristics make GSM a channel suitable for a variety of applications in different domains especially security applications such as secure voice communication. Performance and usage of GSM applications extremely depends on the transmission data rate. Hence, transmitting data over GSM is still an attractive topic for research. This paper considers the problem of digital data transmission through the GSM voice channel. A lower capacity bound for data transmission through the GSM Adaptive Multi Rate (AMR) voice codec is presented. The GSM channel is modeled in a simple manner to overcome it's memory and non-linearity effects. A new statistic based on received samples is extracted and a novel method to transmit data over that channel which asymptotically tends to the achieved lower bound is offered.**

Keywords-component: Secure Data Transmission, GSM voice channel, lower bound on Capacity , Adaptive Multi Rate

I. INTRODUCTION

Nowadays, with the spread of various voice communication systems such as PSTN, VOIP and Mobile Networks, voice channels are widely available. Previous research [1], [2], [3] has shown that it is possible to use such channels for data communication in low-rate applications. Although high speed data dedicated channels are currently available, using compressed voice dedicated channels for data communication is still interesting due to the following reasons.

First, the wide coverage of common voice channels makes the proposed low-rate data channel available everywhere, especially in areas where data dedicated networks have not yet become widespread.

Second, in order to maintain the required speech quality, voice channels are designed for better QoS in comparison with data services. In general voice traffic is prioritized over data connections on networks where both exist. This makes the proposed low-rate data channel a reliable and more consistent real-time channel with better QoS especially in network congestion periods.

Finally, the proposed data channel will be more secure by applying point to point encryption and avoiding the common network gateways.

A data modem of the type described here has been suggested for transmission of encrypted end-to-end secure voice or data over the GSM voice channel in [1], [4]. This channel can also be used for real-time and secure transmission of Point of Sale (PoS) transaction information from the POS terminal to the financial host through the GSM voice channel [5].

Providing data connection over voice dedicated channels is a challenging problem because of several limitations.

First, due to the low bandwidth of speech signal, voice dedicated channels are extremely narrow-band with maximum bandwidth of 4 KHz which inherently limits the achievable data rates.

Second, as common voice channels use several techniques like Discontinuous Transmission (DTX), Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) in order to reduce bandwidth usage during voice silence periods, the modulated signal should pass through the voice channel without raising any alarms such as VAD.

Finally, the voice codecs used in such channels cause several unpredictable distortions to the modulated signal during the compression/decompression process.

The previously proposed methods for the problem of data communication through compressed voice channels can be classified to three different groups due to the conceptual idea they are based on [6]. These three groups are referred to as "Parameter Mapping", "Codebook Optimization" and "Modulation Optimization" and are described below, respectively:

The "Parameter Mapping "method involves mapping of the input bit stream onto speech parameters of some speech production model. In this approach the modulator is a signal synthesizer which uses the input data to synthesize a speech-like signal based on the speech production model. Katugampala et al. [1], [4] mapped the input data on pitch frequency, Line Spectral Frequencies (LSF) and energy of speech frames. This approach achieved a raw data rate of 3kbps with 2.9% Bit Error Rate (BER) on the GSM Enhanced Full Rate (EFR) voice channel. A similar method is utilized by [7] and [8] for data communication through codecs that apply Algebraic Code Excited Linear Prediction (ACELP) speech coding technique. This approach modulates the input data stream on algebraic codebook pulse positions making a PPM signal. This method achieved a raw data rate of 3kbps with 1.2% BER on the GSM Enhanced Full Rate (EFR) voice channel.

The "Codebook Optimization" method is to encode the input bit stream onto a codebook of predefined waveforms (symbols) that are optimized for efficient communication through the desired voice codec. Sapozhnykov et al. [6] numerically optimized the symbol waveforms by a pattern search algorithm over all symbols of a fixed length that meet the voice codec limited bandwidth constraint. This modem

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was tested on GSM HR (Half rate), FR (Full Rate), AMR (Adaptive Multi Rate) and EFR (Enhanced Full Rate) voice codecs and achieved a throughput of 4kbps with 2.5% Symbol Error Rate (SER) on EFR voice codec. Boloursaz et al. [9] modeled the AMR voice codec by a discrete memoryless channel and generated the codebook by searching for the symbol set that maximizes this channel's capacity. The symbols where chosen from a database of waveforms derived from observed human speech. This modem was tested on different compression rates of GSM AMR voice codec and achieved a raw data rate of 1kbps with SER of 6.8×10^{-2} on AMR 4.75 voice codec.

Finally, the "Modulation Optimization" uses common digital modulations without or with little modifications and optimizes their parameters for efficient data communication through compressed voice channels. Chmayssani et al. [10], [11] used the conventional Frequency Shift Keying (FSK) and QAM to modulate the input bit stream into audio signals and achieved a data rate of 3kbps with 3×10^{-3}BER on GSM EFR voice channel.

As seen above, previous studies have suggested different approaches to modulate data into audio signals and reported different rate and BER performances. Though, it seems more appropriate to calculate and derive the capacity of such channels, this research derives an inner bound on the capacity of GSM Adaptive Multi Rate (AMR) Voice channel [12]. In addition, it demonstrates a novel communication scheme which approaches this limit using a sub optimum detector. The derived bound is then compared with previously achieved rates in prior works. It is shown that there exists a large gap between previously reported rates in the literature and the achieved bound. Therefore, the derived bound encourages additional research to introduce communication schemes that achieve this limit.

The rest of this paper is organized as follows. Section II outlines the proposed method to calculate lower bound on capacity of GSM voice channel. Section III presents simulation results of proposed method for GSM channel and compares this bound to other rates that have been achieved yet. Section IV tries to design a sub optimum simple detector which provides a rate that is slightly lower than the derived bound.

II. PROPOSED METHOD

In this section, a lower bound on GSM voice channel capacity is introduced. Generally, bit stream data is transmitted through the channel via appropriate symbols. Suppose that each symbol has K samples, therefore by transmitting$N$ bits,$K \times N$ samples of transmitted waveforms will be available at the receiver side. Furthermore, to calculate a lower bound based on transmitted data for the mentioned channel, a semi-rectangular subspace in $\mathbb{R}^K$ is defined, each received symbol corresponds to a point in that subspace. $\overrightarrow{S_n}$ represents the n^{th} received symbol and $s_{i,n}$ stands for the amplitude of i^{th} sample of the n^{th} received symbol.

$$\overrightarrow{S_n} = [s_{1,n}, \ldots, s_{k,n}]$$

Furthermore, the total subspace is quantized into isometric l^K cubes. In this new discrete subspace, l represents the number of quantization levels, which means that each axis is divided into l equal parts. The infimum and supremum point of each axis is defined as below:

$$A_{k\,sup} \triangleq \max_n (S_{k,n}) \tag{1}$$

$$B_{k\,inf} \triangleq \min_n (S_{k,n}) \tag{2}$$

$$SZ_k \triangleq \frac{A_k - B_k}{l} \tag{3}$$

Where$S_{k,n}$ represents the amplitude of k^{th}received sampleand$l$ represents number of slots in each axis which is supposed to be the same for all axes for simplicity.

By defining a matrix of $M = [m_{i_1,\ldots,i_k}]_{\underbrace{l\times l\times\ldots\times l}_{k}}$. Each index represents the total number of received samples which are located in $m_{i_1,\ldots,i_k}$.It can be formulated as below:

$$m_{i_1,\ldots,i_k} = \underset{i=1,\ldots,N}{\text{count}} \tag{4}$$

$$\begin{cases} B_k + (i_k - 1) \times SZ_k \le s_{i,n} < B_k + i_k \times SZ_k, & L-1 \ge k \ge 1 \\ B_k + (i_k - 1) \times SZ_k \le S_{i,n} \le A_k & k = L \end{cases} \tag{5}$$

In addition, by dividing the value of each cube into N (total number of points), probability of locating an arbitrary point in the specific cube is derived as following:

$$N = [n_{i_1,\ldots,i_K}]_{\underbrace{L\times L\times\ldots\times L}_{K}} \tag{6}$$

$$n_{i_1,\ldots,i_K} = \frac{m_{i_1,\ldots,i_K}}{N} \tag{7}$$

$$H_Y = \sum_{i_1, i_2, \ldots, i_K} -N_{i_1,i_2,\ldots,i_K} \times \log_2 N_{i_1,i_2,\ldots,i_K} \tag{8}$$

Fig. 1 illustrates an example of received symbols which leads to create a rectangular subspace in $\mathbb{R}^3$. Considering shown points, corresponding for specified square cube can be written as below:

$$n_{2,2,2} = \frac{10}{51} \tag{9}$$

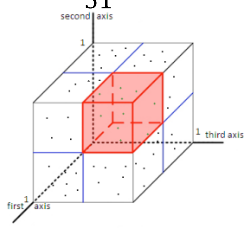

Fig.1 Cubic example, $l = 2$, $N = 51$, $n_{2,2,2} = 10$

Clearly, by increasing l, the quantity of H_Y and $H_{(Y|X=j)}$increases asymptotically to its final value. It can be justified by comparing the definition of discrete entropy and the continuous one. Thus by increasing the number of sections in each axis, length of each subsection tends to zero and computation tends to continuous format. Therefore, the mentioned entropies converge by increasing l.

In this research, speech like symbols are used for data communication through GSM voice channel because of their acceptable performance as reported by [3], [9].The following section chooses these samples to attain a lower bound for the GSM voice channel.

III. SIMULATION

Since GSM is a voice channel, the transmission rate is fixed and equals to 8000 samples per second. Thus, in this approach, symbol rate is$\frac{8000}{K}$, where$K$ represents the number of samples devoted to each symbol. For each symbol rate, appropriate statistically efficient speech like symbols are transmitted through the channel [3], [8].

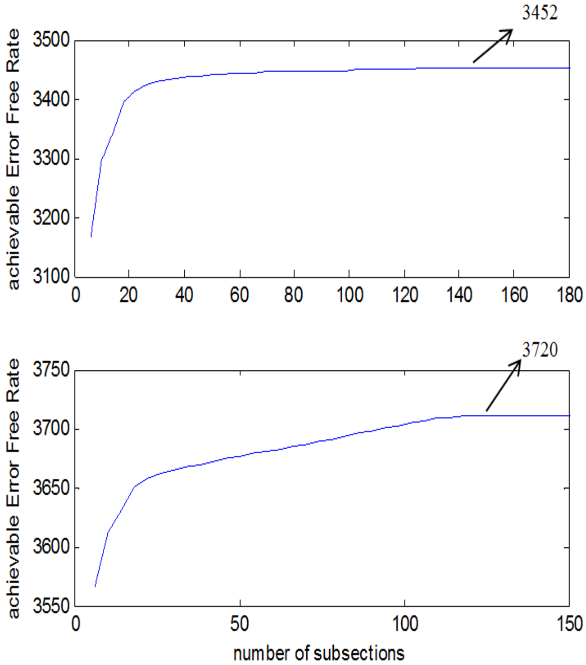

Fig. 2 GSM 12.2 channel capacity, a) Rate=8000 b) Rate= 4000

Simulations of GSM FR and HR speech codec are performed with the C library from [12]. By Transmitting sufficient symbols over the channels and using the approach introduced in previous section, total capacity which is the product of capacity per channel use and channel use per second, is derived

In Fig. 2-a , K= 2, hence channel use per second is $\frac{8000}{2} = 4000$ and the calculated capacity bound on the GSM FR channel is 3708 bps. In Fig.2-b, K is set to 1, therefore the final rate is 8000bpsand the achieved capacity bound is 3400bps. Other rates of $\frac{8000}{K}$; $K \geq 3$ are not simulated, owing to the calculated capacity in these rate are evidently lower than transmission bit rate, i.e. $\frac{8000}{3} = 2666$, which are lower than the above achieved transmission rates.

Fig.3-d, shows the 1650 bps as a lower bound capacity for GSM HR at transmission rate of 2000 bps.

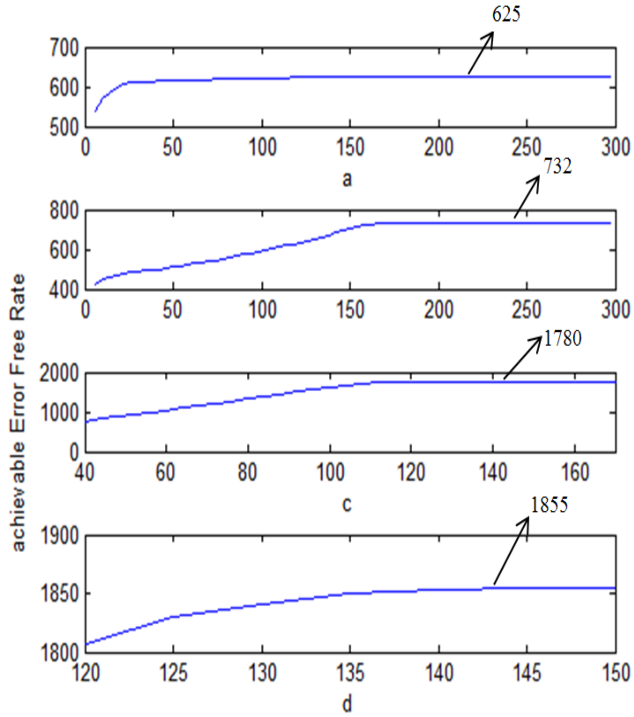

Fig. 3 GSM 4.75 channel capacity a) Rate=8000 b) Rate=4000 c) Rate=2666 d) Rate=2000

As it is seen from the Fig.2 and Fig.3, depending on transmission rate, maximum achievable data rate changes. This occurred because of GSM channel essence which has both memory and nonlinear blocks. Considering the discussed results, the lower bound for the GSM FR & HR are 3708 bps and 1855 bps respectively. These bounds are achieved for data transmission rates of 4000 and 2666 bps.

Another point which is necessary to check is channel symmetric properties. Since the input data stream is usually $i.i.d$, accordingly it is more appropriate if the mutual information has its maximum value on the symmetric point. The following figure demonstrates maximized mutual information values versus probability distribution for 12.2k and 4.75kGSM voice channel, which satisfies mentioned consideration. Comparing mentioned mutual information and capacity for a Binary Symmetric Channel (BSC), it is obvious that both GSM channels are approximately binary symmetric channels. This means that maximum achievable rate for these channels are derived when using equal probability input symbols.

The gap between the achieved rate in the literatures and the lower bound conveys that this field of study is still an open problem for researchers and it is needed to propose approaches to achieve higher bit rates. In the next section one new method is proposed.

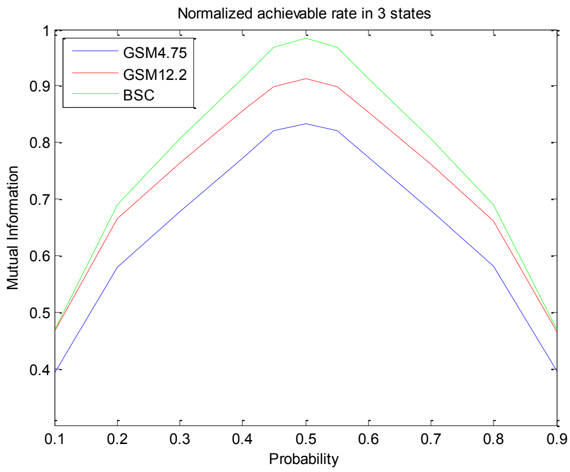

Fig. 4 Normalized mutual information over Probability of zero transmission

IV. SUBOPTIMUM DETECTOR

The purpose of this section is designing a sub optimum simple detector based on channel properties. GSM voice channel characteristics are more complicated and have a variety of nonlinear blocks [12], [13]. It is clear that working with this channel and precisely designing a detector is more complicated and nearly impossible. Accordingly, to design a sub optimum detector, the above channel, is simplified and conceptually presented in Fig.5.

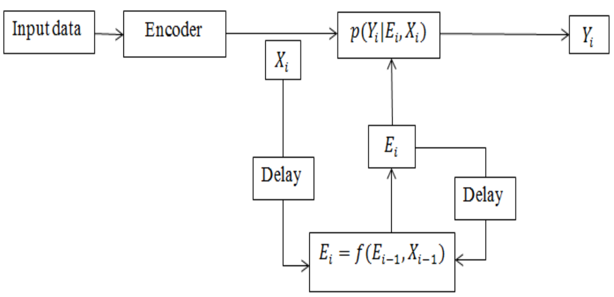

Fig. 5 GSM Simplified Model

In this Model, the output of the GSM voice channel (Y_i), depends on the input data (encoded into X_i), state of the system (E_i), and the transmission probability $p(Y_i|E_i,X_i)$,representing statistical behavior of the channel. Also the function f shows that how the state of the system is built.At the next stage, an appropriate detector should be designed to decide what symbols are transmitted, based on posterior probability. It can be derived as follows:

$$\max_{m,m=1,..,M} P(\vec{S}^m|\vec{S}), \tag{11}$$

Where$\vec{S}^m$ and $\vec{S}$ represent m^{th} transmitted symbol and received symbol, respectively. As the above probability is a function of all samples of previous symbols, it's so intricate to solve this optimization directly. The main two complication reasons are:

1. The relations between previous symbols are not clear. In other word, channel memory does not have a well-defined manner, hence it would be a big problem to obtain the mathematical presentation for $P(S_m|r)$.
2. The joint distributions of samples are not known.

Accordingly, two steps should be taken to overcome the mentioned problem. Firstly, the channel memory should be removed as much as possible. Secondly, a statistically discriminate feature should be extracted from the samples of each symbol which have a well-known distribution.

A. Channel Memory Problem

To conquer this challenge, according to AMR standard [12], [13] choosing farther samples from previous frame (not sub frame) causes less dependency between output samples. Considering this fact, interleaving may be a good solution to decrease dependencies between samples. On the other hand, the interleaving can be utilized as an appropriate source coding in this type of channel with memory. The previous state can be justified as follows:

Dividing the input signal into N_1 rows of samples, in which each row contains N_2samples.The below $N_1 \times N_2$ components are achieved:

$$\begin{gathered} X_1 X_{N_1+1} X_{2N_1+1} X_{3N_1+1} \dots X_{(N_2-1)\times N_1+1} \\ X_2 X_{N_1+2} X_{2N_1+2} X_{3N_1+2} \dots X_{(N_2-1)\times N_1+2} \\ \\ X_{N_1} X_{2N_1} X_{3N_1} X_{4N_1} \dots X_{N_2\times N_1} \end{gathered} \tag{12}$$

$$\begin{aligned} C &= \frac{1}{N_2 \times N_1} \max_{p(x_1,x_2,\dots,x_{N_2\times N_1})\le 1} I(X_1, X_2, \dots, X_{N_2\times N_1}; Y_1, Y_2, \dots, Y_{N_2\times N_1}) \\ &= \frac{1}{N_2 \times N_1} \max_{p(x_1),p(x_2)\dots p(x_{N_2\times N_1})} [H(X_1, X_2, \dots, X_{N_2\times N_1}) \\ &- H(X_1, X_2, \dots, X_{N_2\times N_1}|Y_1, Y_2, \dots, Y_{N_2\times N_1})] \end{aligned} \tag{13}$$

By extending the mutual information terms:

$$\begin{aligned} &H(X_1, X_2, \dots, X_{N_2\times N_1}) \\ &- H(X_1, X_2, \dots, X_{N_2\times N_1}|Y_1, Y_2, \dots, Y_{N_2\times N_1}) = \\ &\qquad H(X_1, X_{N_1+1}, \dots, X_{(N_2-1)\times N_1+1}) \\ &-H(X_1, X_{N_1+1}, \dots, X_{(N_2-1)\times N_1+1}|Y_1, Y_2, \dots, Y_{N_2\times N_1}) \\ &\qquad +H(X_2, X_{N_1+2}, \dots, X_{(N_2-1)\times N_1+2}) \end{aligned} \tag{14}$$

$$-H\left(\begin{matrix} X_2, X_{N_1+2}, \dots, X_{(N_2-1)\times N_1+2} \\ |Y_1, Y_2, \dots, Y_{N_2\times N_1}, X_1, X_{N_1+1}, \dots, X_{(N_2-1)\times N_1+1} \end{matrix}\right) \tag{15}$$

$$\pm \cdots$$

$$= I\left(X_1, X_{N_1+1}, \dots, X_{(N_2-1)\times N_1+1}; Y_1, Y_2, \dots, Y_{N_2\times N_1}\right)$$

$$+ I\begin{pmatrix} X_2, X_{N_1+2}, \dots, X_{(N_2-1)\times N_1+2}; \\ Y_1, Y_2, \dots, Y_{N_2\times N_1}, X_1, X_{N_1+1}, \dots, X_{(N_2-1)\times N_1+1} \end{pmatrix}$$

$$+ \dots$$

$$\geq \frac{1}{N_2 \times N_1}[I\left(X_1, X_{N_1+1}, \dots, X_{(N_2-1)\times N_1+1}; Y_1, Y_{N_1+1}, \dots, Y_{(N_2-1)\times N_1}\right) \tag{16}$$

$$+ I\left(X_2, X_{N_1+2}, \dots, X_{(N_2-1)\times N_1+2}; Y_2, Y_{N_1+2}, \dots, Y_{(N_2-1)\times N_1+2}\right)$$

$$+ \dots$$

$$+ I\left(X_{N_1}, X_{2N_1}, \dots, X_{N_2\times N_1}; Y_{N_1}, Y_{2N_1}, \dots, Y_{N_2\times N_1}\right)]$$

$$= \frac{1}{N_2} \times I\left(X_{N_1}, X_{2N_1}, \dots, X_{N_2\times N_1}; Y_{N_1}, Y_{2N_1}, \dots, Y_{N_2\times N_1}\right)$$

$$= I\left(X_{N_1}; Y_{N_1}\right)$$

B. Joint Distribution of Samples

As noted before, the joint distribution of output samples for each symbol, is neither known nor discriminate enough to properly decide which symbol is transmitted. Hence the other attempt is to derive a feature which has known behavior for mathematical modeling and contains the received information as much as possible. To propose this feature, Euclidean distance from the mean of received symbol is targeted as follows:

$$r_{0,n}^0 = \left\| \vec{S_n^0} - < \vec{S^0} > \right\| \tag{17}$$

$$r_{0,n}^1 = \left\| \vec{S_n^1} - < \vec{S^0} > \right\| \tag{18}$$

$$r_{1,n}^0 = \left\| \vec{S_n^0} - < \vec{S^1} > \right\| \tag{19}$$

$$r_{1,n}^0 = \left\| \vec{S_n^1} - < \vec{S^1} > \right\| \tag{20}$$

Where $< \vec{S^0} >$ and $< \vec{S^1} >$, are the mean of received $\vec{S}^0$, $\vec{S}^1$ and $\vec{S_n^0}$ and $\vec{S_n^1}$ represent the n[th] received symbol of $\vec{S}^0$, $\vec{S}^1$.

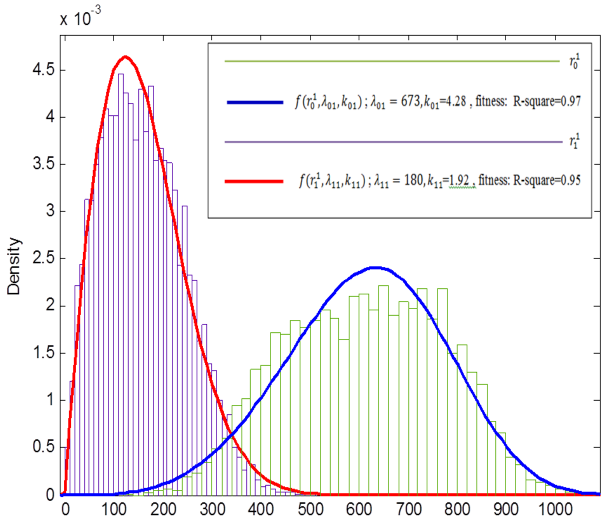

Fig. 6 two feature in Weibull distribution for GSM 12.2 at Bit rate =4000

Density profiles of mentioned random variables ($r_0^0, r_0^1, r_1^0, r_1^1$) are obtained by simulation analysis. In both mentioned figures, Weibull distribution is fitted to the extracted density function. Weibull probability distribution is known as below:

$$f(x; \lambda, k) = \begin{cases} \frac{k}{\lambda}\left(\frac{x}{\lambda}\right)^{k-1} e^{-\left(\frac{x}{\lambda}\right)^k} & x \geq 0 \\ 0 & x < 0 \end{cases} \tag{21}$$

Where $k > 0$ is the shape parameter and $\lambda > 0$ is the scale parameter of the PDF [14].

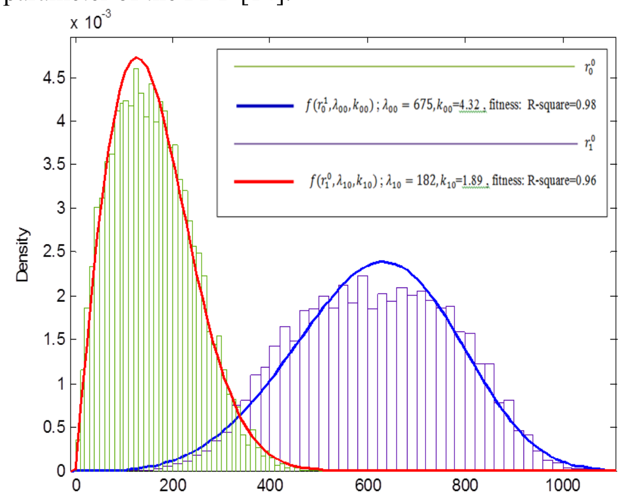

Fig.7last two Weibull features for GSM 12.2 at Bit rate =4000

According to previous statement, suboptimum receiver structure is designed and depicted in Fig. 8.

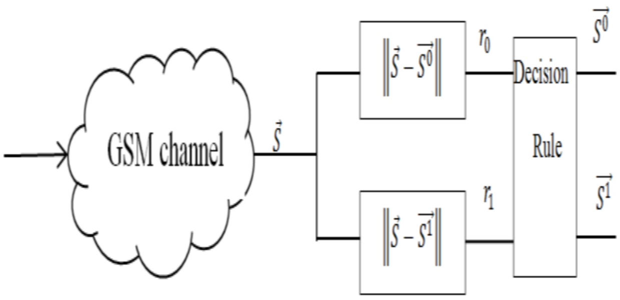

Fig. 8 Structure of Suboptimum Detector

Using Probability PDF of these features and applying posterior probability rule to this structure, transmission rate of 4[k] in GSM FR with less than 1 percent error is achieved. The decision making in receiver side is based on Maximum Likelihood Detector (MLD) as follows:

$$f\left(r_0, r_1 | \vec{S}^1\right) \underset{S^0}{\overset{S^1}{\gtrless}} f\left(r_0, r_1 | \vec{S}^0\right) \tag{22}$$

As r_0 and r_1 are statistically independent, then :

$$f(r_0|\vec{S}^1) \times P(r_1|\vec{S}^1) \underset{S^0}{\overset{S^1}{\gtrless}} f(r_0|\vec{S}^0) \times P(r_1|\vec{S}^0) \tag{23}$$

So the decision rule is defined as below:

$$f(r_0;\lambda_{01},k_{01})f(r_1;\lambda_{11},k_{11}) \underset{S^0}{\overset{S^1}{\gtrless}} f(r_0;\lambda_{00},k_{00})f(r_0;\lambda_{10},k_{10}) \tag{24}$$

By Supposing γ_1 and γ_0 as detection region for $\vec{S}^0$, $\vec{S}^1$ then the probability of error derived as follows:

$$P_e = \frac{1}{2}\iint\limits_{\gamma_1} f(r_0,r_1|S^0)dr_0\,dr_1 + \frac{1}{2}\iint\limits_{\gamma_0} f(r_0,r_1|S^1)dr_0\,dr_1 \tag{25}$$

To sum up, memory and nonlinearity of GSM channels are two major barrier to transmit data. In this section by using combination of interleaving and new proposed feature, a novel method has been offered which it's performance tends to proposed lower bound on GSM capacity.

V. Conclusion

In this paper by introducing a method of calculating capacity, a lower capacity bound for GSM voice channel is presented. Then by modeling AMR voice channel as a simple channel with memory, interleaving is used to conquer its memory effects. A novel sub optimum detector based on Weibull PDF is offered for data extraction. Finally, the performance of the proposed sub optimum detector is compared with both existing methods and achieved lower bound of capacity.

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