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Performance Deviation Measurement Using Conventional & Practically Derived Threshold in MF Receiver for Binarised Sound Signals

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Abstract – *The need and necessities of communication systems are expanding day by day. Today system reliability is an important user requirement. Matched filter based receiver (MF receiver) work on threshold (λ) which is logarithmic function of input signal parameters, r_0 and r_1. Under certain assumptions, the threshold (λ) at the MF receiver is considered as '0' i.e λ=0 since probability of input signal r_0 and r_1 are assumed as 0.5. But many input signals of practical importance and under other circumstances threshold (λ_{pract}) at the MF receiver may not be equal to zero i.e $\lambda_{pract} \neq 0$. This leads to the deviation in the performance which is evaluated by using λ = 0, and considering λ_{pract} at the MF receiver. Here λ_{pract} is the practically derived threshold at the MF receiver using a well defined methodology. In this work for different considered digitized sound signal as input, various results are plotted in the form of percentage BER deviation vs. SNR curves. From the results it is found that substantial performance deviations do exist for sound categories especially at low SNR values and smaller window length. The simulation analysis study is carried out using MATLAB 7.6.0.324 (R2008a) version.*

Key Words: MF Receiver, BER, MATLAB, SNR.

1. INTRODUCTION

Future mobile radio systems will have to meet exacting requirements. Data rate per user is expected to increase significantly, but could also vary substantially between the peak & typical values. With data traffic dominating over voice transmissions, the demands in data rate between downlink and uplink are becoming asymmetric. Quality of service – a complex parameter which can be defined in several ways – is of particular interest to mobile users. And with many future services likely to be location based, mechanisms will be necessary to derive the user's location or other context. Although hidden from the user, one of the most important issues is the integration of packet-switched [10] and IP-based traffic [11].

Network operators have made significant investments in building IP core networks based on internet system architectures. Further efforts are needed to optimize these and ease the integration of fixed and wireless networks. Frequency spectrum and bandwidth allocation will be important considerations. Radio spectrum is scarce, and therefore expensive and hence future systems will have to be very efficient in how they use the limited spectrum available. Alternative methods of spectral allocation and use could also be considered. The system must be able to dynamically change the allocated resources as users' requirements and available capacities change [11].

In mobile based Communication System [10], there were many problems in network like low bandwidth, slow data rate, data are not secure (because signals being available in open). The subject communication runs into cyber security problems. Our main objective is to provide a highly secure environment that is simple to use and deploy. So we will use agent base communication which is more effective in mobile communication and improve security for obtaining the goal. Personalization and seamless access will be key features in making future mobile services and devices easier to use and offering greater added value. To make the most of the opportunities offered by new technologies, future mobile services and devices will need to be much easier to use. The user can be anywhere, at home, on the move and still able to access his customized services. Continuity of service from fixed to mobile access and seamless roaming of services across operators, heterogeneous networks and terminals, country and cultural boarders should become a reality [11].

1.1 Communication System Block Diagram

Communication systems convey information from one point to another via physical channels that propagate electromagnetic, acoustic, particle density, or other waves. The three basic elements of every communication systems are Transmitter, Receiver and Channel.

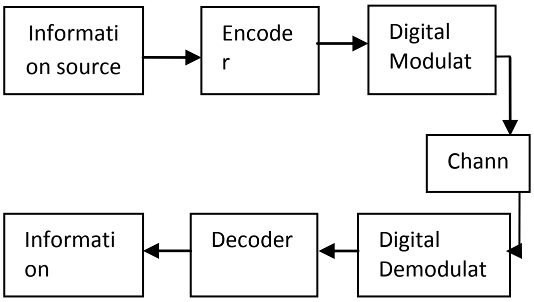

Fig -1: Block Diagram of Digital Communication

The first block is source of information in which we can use two types of information sources analog information sources and digital information sources [5]. The message produced by a source, normally, is not electrical. Hence an input transducer is used for converting the message to a time – varying electrical quantity called message signal. Similarly, at the destination point, another transducer converts the electrical waveform to the appropriate message. The transmitter is located at one point in space, the receiver is located at some other point separate from the transmitter, and the channel is the medium that provides the electrical connection between them [5].

The purpose of the transmitter is to transform the message signal produced by the source of information into a form suitable for transmission over the channel. The received signal is normally corrupted version of the transmitted signal, which is due to channel imperfections, noise and interference from other sources. The receiver has the task of operating on the received signal so as to reconstruct a recognizable form of the original message signal and to deliver it to the user destination [5].

The encoder converts the input i.e. symbol sequence into a binary sequence of 0's and 1's by assigning code words to the symbols in the input sequence. At the receiver, the decoder converts the binary output of the channel decoder into a symbol sequence [4].Error control is accomplished by the channel coding operation that consists of systematically adding extra bits to the output of the source coder. These extra bits do not convey any information but helps the receiver to detect and/or correct some of the errors in the information bearing bits. The channel decoder recovers the information bearing bits from the coded binary stream [4].

In digital modulation, an analog carrier signal is modulated by a discrete signal. Digital modulation methods can be considered as digital-to-analog conversion, and the corresponding demodulation or detection as analog-to-digital conversion. The changes in the carrier signal are chosen from a finite number of M alternative symbols (the modulation alphabet) [5].

The modulator transforms the signal or symbol to be transmitted into the signal that is propagated across the channel; the channel may add noise and distortion. The channel is the physical transmission medium over which the communication is sent. It may be wires, radio airwaves, fiber optics, etc. All channels have physical limitations which will distort and attenuate the transmitted signal and which will add noise to the transmitted signal. Thus, the received signal will not be an exact duplicate of the transmitted signal [5].

1.2 Optimum Reception in Additive White Gaussian Noise (AWGN)

The performance of the receiver is usually measured in terms of the probability of error and the receiver is said to be optimum if it yields the minimum probability of error [6]. The measure of performance used for comparing digital modulation schemes is the probability of error .The receiver makes errors in the decoding process due to the noise present at its input. The receiver parameters as H(f) and threshold setting are chosen to minimize the probability of error [6].

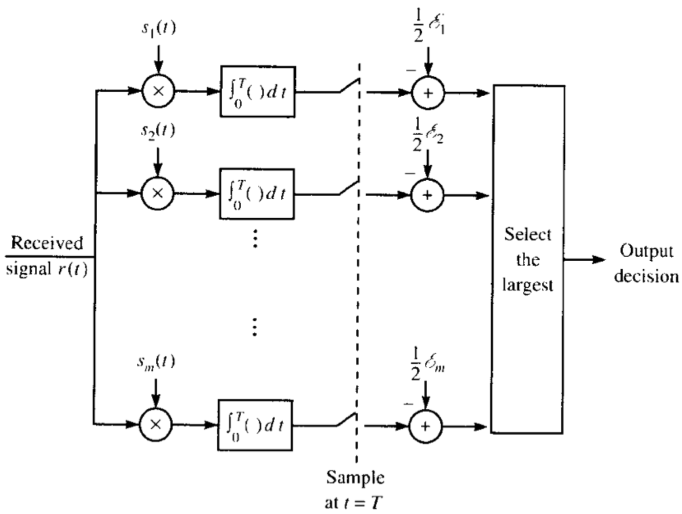

Fig -2: Block diagram of optimum receiver [6]

Signal detection in MF receiver is based on the use of threshold which is logarithmic function of input signal parameters, r_0 and r_1 [6].

$$\lambda_{opt} = \frac{N_o}{4AT_b}\ln\frac{r_0}{r_1} \qquad (1)$$

The subscript *opt* in Eq. (1) indicates that this threshold is optimal provided parameters N_o, A, T_b, r_0, r_1 are known at the receiver. Usually, at the receiver, N_o, A, T_b are known and r_0 and r_1 are assumed as 0.5. This leads to λ = 0 [6].

2. SIGNAL CONSIDERED AND METHODOLOGY ADOPTED

2.1 Sound Signals

Sound is a vibration of a medium (such as air), associates a pressure value to every value of time and three space coordinates. It can be sampled at a discrete set of time points. A sound signal is converted to an electrical signal by a microphone, generating a voltage signal as an analog of the sound signal, making the sound signal available for further signal processing.

A sound signals is generally characterized by following parameters: The first one is frequency and second one is pitch. The shorter the wavelength, higher would be the frequency, and the higher the pitch, of the sound. Since the sounds are travelling at about the same speed, the one with shorter wavelength will go by more frequently, it has higher frequency and pitch. In other words, it sounds higher.

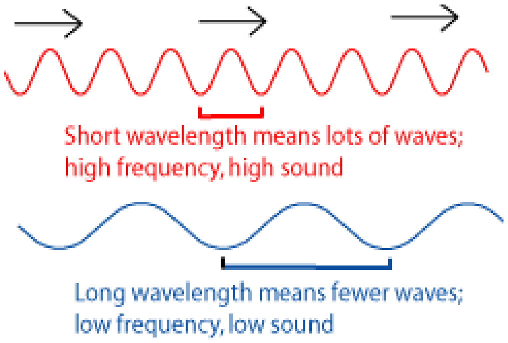

Fig -3: Relationship between wavelength & frequency of sound [7]

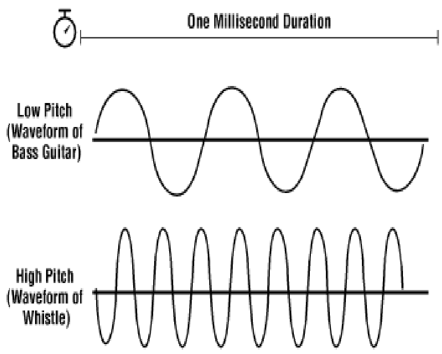

Fig -4: Relationship between pitch and frequency of sound [8]

As discussed, within the sound category, four sound samples have been considered. A brief explanation of each has been presented here:

1. **Speech signals:**
Speech signals mostly consist of male voice and female voice. Adult men and women have different pitch and frequency of sound. The female voice has high pitch as compared to male voice. We have considered here single voice and no instrument in background and no echo is present.

2. Instrumental:
This category consists of different sound sample such that each sound sample represents a single instrument sound i.e monophonic. They do not have back vocals.

3. Music:
This category consists of different sound samples of songs which includes instrument in background. This is polyphonic type sound.

2.2 Methodology Adopted

The next step is to discuss the methodology that is adopted for the evaluation of the performance deviation of matched filter based receiver. The step-wise methodology for the evaluation of performance deviation is as follows:

Step 1: Initially take a sample of sound signal.
Step 2: Select a particular *SNR.*
Step 3:Convert the test signal into binary form using appropriate A/D converter. Let L' is the length (in number of bits) of this binarized sound.
Step 4: Consider a window-length N_w, and define $\lfloor L = L' / N_w \rfloor$.
Step 5: If l=1, take first N_w bits of the binarized sequence. Else, take next N_w bits of this binarized sequence. Increment counter; $l = l + 1$.
Step 6: Modulate using polar NRZ line coding scheme.
Step 7: Transmit through AWGN channel.
Step 8: Receive the noisy signal at the receiver.
Step 9: Demodulate using conventional Matched filter based receiver with assumption $r_0 = r_1 = 0.5$ ($\lambda^{0.5,0.5} = 0$) and find *BER* at the receiver output.. Name this as $BER^{0.5,0.5}$ and store it at some memory location.
Step 10: Using r_0 & r_1, find out λ^{r_0,r_1} .Use this threshold to evaluate BER^{r_0,r_1} at the receiver output and store it at some memory location.
Step 11: Check if $l < L$? If 'YES', go to Step 5 & repeat the procedure. If 'NO' then find out $BER_{avg}^{0.5,0.5}$ & $BER_{avg}^{r_0 r_1}$ by averaging all the *BER* that were stored previously. Further, find *BER* deviation as per & go to the next step.
Step 12: Repeat the procedure for different *SNR* values. Else, go to the next step.
Step 13: Repeat the procedure for other window sizes. Else, go to the next step.
Step 14: Repeat procedure for different test samples. Else stop the simulation & go to next step.

Step 15: Plot percentage *BER* deviation vs. *SNR* curve for different window sizes (N_w) & different test signals.

3. RESULTS AND DISCUSSION

3.1 Evaluation of Performance Deviation for Image Category

For the evaluation of the performance deviation various sounds signals have been sub-divided into four distinct categories. These are: Male sounds, Female sounds, Instrumental sounds and Music sounds. Male sounds are related to the high wavelength, low pitch and low frequency as compared to Female sounds. Likewise Instrumental and music sounds are pertaining to sounds covering different instruments and songs respectively. Instrumental sounds are monophonic type while music sounds are polyphonic type. Within each sound category, five different sound samples have been considered for the evaluation of performance deviation. The results are plotted in the form of percentage *BER* deviation (%η) vs. *SNR* curves for all the considered N_w values.

Different parameters that are considered for the evaluation of performance deviation are as follows:

- MATLAB 7.6.0.324 (R2008a) version is used for the simulation purpose.
- Channel is assumed to be AWGN channel. Further the considered range of *SNR* is $-30\ dB \le SNR \le +10\ dB$.
- Following three window sizes (N_w) has been considered for the analysis, N_w = *1000, 5000 and 10,000.*
- Standard A/D converter without A-Law and µ-law is used for the sound binarization.
- The size of each considered sound sample is *5000.*

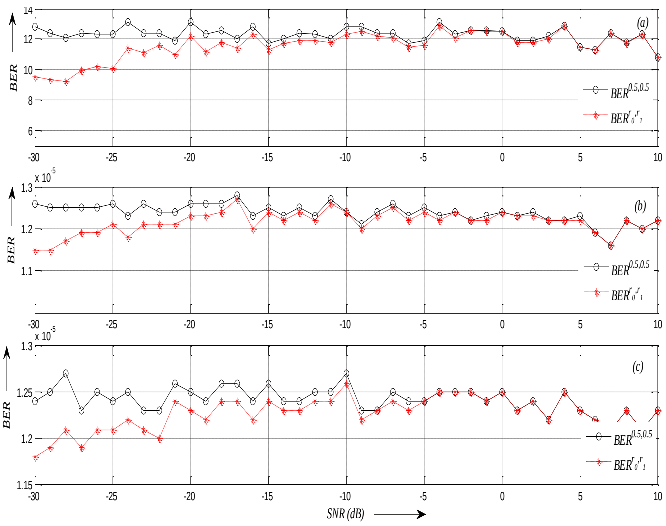

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Fig -5: *BER* vs. *SNR* curve corresponding to sound samples, for a) N_w = *1000* b) N_w = *5000* c) N_w = *10,000*

From the result shown in Fig. 5, following conclusions can be drawn:

- It can be seen in Fig. 5 (a) that, for all considered *SNR* values and at N_w=*1000*, *BER* vs. *SNR* curve corresponding to $\lambda^{0.5,0.5}$ is much higher than *BER* vs. *SNR* curve corresponding to λ^{r_0,r_1}. Further, this shows that the conventional matched filter based receiver become suboptimal when threshold $\lambda^{0.5,0.5}$ is used for the detection of practical digital signals, which is sound here.

- This deviation in *BER* is seen for other considered N_w values such as *5000* and *10,000* as shown in Fig 5(b) & Fig. 5(c). But, this *BER* deviation decreases with increase in N_w.

- For all considered N_w values, at *SNR = -30 dB,* the maximum deviation in *BER* is observed. This deviation appears to decrease with increase in *SNR*.

In order to compare the results corresponding to all the sounds categories that are considered, percentage BER deviation (average) vs. SNR (dB) curves are plotted for a given window sizes i.e. N_w = *1000, 5000 & 10,000* , as shown in Fig. 6, Fig. 7 & Fig. 8 respectively. Further, all the results of Fig. 6 to Fig. 8 are averaged and plotted in Fig. 9.

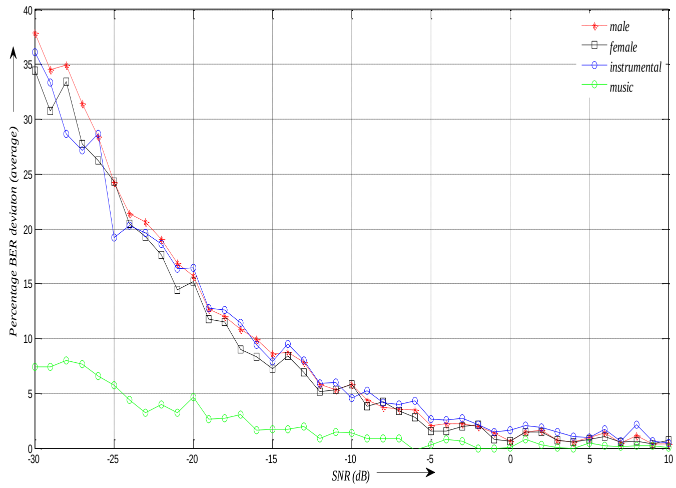

Fig -6: Comparison of average percentage *BER* deviation for all considered sound categories, corresponding to N_w *=1000.*

- From Fig. 6 following conclusions can be drawn, corresponding to N_w *=1000*, the percentage average *BER* deviation vs. *SNR* curve corresponding to male sound category has maximum deviation at *SNR = -30 dB* i.e. approx. *37%.* After *SNR = -30 dB* the male, female and instrumental sound categories are almost similar. The percentage average BER deviation of music sound category is less than male, female and instrumental sound categories. Thus for given *SNR*, more deviation in average *BER* is observed corresponding to male, female and instrumental sound categories than for the case of music sound category.

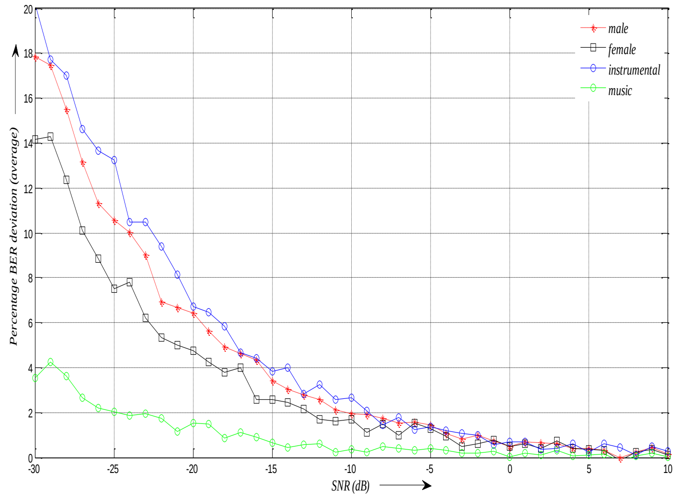

Fig -7: Comparison of average percentage *BER* deviation for all considered sound categories, corresponding to N_w *=5000*.

- As shown in Fig. 7, corresponding to $N_w = 5000$, at *SNR = -30 dB*, maximum percentage *BER* deviation of approx. *20 %* is observed for the case of instrumental sound category, and, minimum of approx. *3.5%* is observed for the case of music sound category. As *SNR* increases, this deviation decreases further.

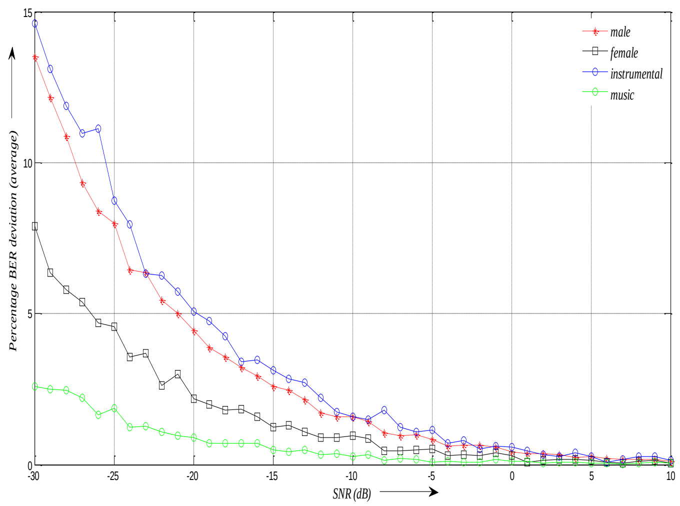

Fig- 8: Comparison of average percentage *BER* deviation for all considered sound categories, corresponding to N_w *=10,000*.

- As shown in Fig. 8, corresponding to $N_w = 10{,}000$, at *SNR = -30 dB,* maximum percentage *BER* deviation of approx. *15%* is observed for the case of instrumental sound category, and, minimum of approx. *2.5%* is observed for the case of music sound category. As *SNR* increases, this deviation decreases further.

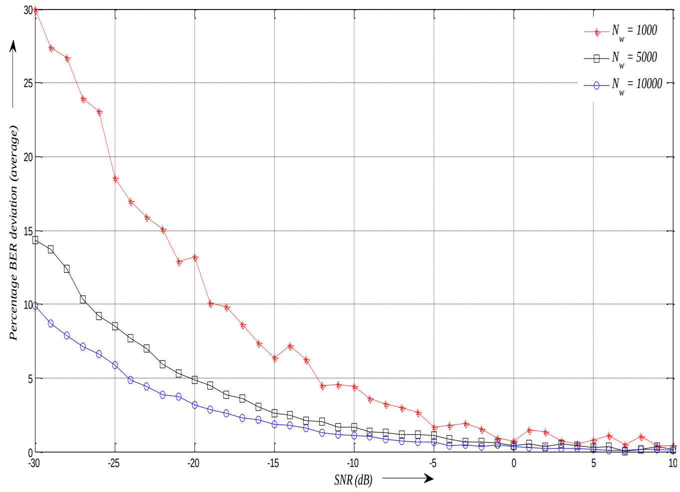

Fig -9: Final average percentage *BER* deviation curve for all sounds

- From Fig. 9, it is clear that, for a given *SNR*, maximum percentage average *BER* deviation of approx. *30%* is observed for the case of $N_w = 1000$ and minimum deviation of approx. *10%* is observed for the case of $N_w = 10{,}000$.

4. CONCLUSION

In this work, the performance deviation at the receiver output is evaluated for the case when i) $\lambda^{0.5,0.5}$ and ii) λ^{r_0,r_1} are used as thresholds for the signal detection, at the receiver. For this, practical signal such as different sound are transmitted, window-wise. As practical signals, four categories, of sound, have been considered. Sound signals have been sub categorized as male sounds, female sounds, instrumental sounds, and, music sounds. Five samples from within these sub categories have been taken up for the evaluation of performance deviation, at the receiver output. Various results obtained are plotted in terms of percentage performance deviation vs. *SNR* curves, for a given window size *(N_w).* Here percentage performance deviation is used as the performance parameter and $BER^{0.5,0.5}$ and BER^{r_0,r_1} are the bit error rates at the output of MF receiver using $\lambda^{0.5,0.5}$ and λ^{r_0,r_1} respectively. Further, in order to generalize these results for a given category, results corresponding to each category are averaged to obtain average percentage *BER* deviation and is plotted as % performance deviation vs. *SNR* curve, for a given N_w. Using extensive simulation results that is carried out using MATLAB 7.6.0.324 (R2008a), it is found that a substantial performance deviation do exist, when digitized sound signals are transmitted window-wise. Further, corresponding to different sound categories, higher performance deviation is observed at lower *SNR* values, and, corresponding to smaller window size (N_w). The maximum *80%* performance deviation in *BER* is observed, for the case of female sound sample, at *SNR = -30 dB*. Also, for a given *SNR* and N_w, the maximum average performance deviation is seen for the instrumental sound samples, and, minimum average performance deviation is observed for the music sound samples. Since a considerable performance deviation is seen when λ^{r_0,r_1} is used over $\lambda^{0.5,0.5}$, for bits detection at the receiver, this study and analysis will certainly be useful in optimizing the performance of MF based receivers.

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