Evaluation of Diversity Gain in Digital Audio Broadcasting

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ABSTRACT
Digital Audio Broadcasting (DAB) is a digital method of delivering radio services from base station to the receiver. The main aim of this paper is to get better reliability of DAB reception in mobile (moving) environment. DAB is very forceful in mobile environment, but in some situations, reception fails in certain spots due to undesired audio interruptions. Antenna diversity systems are considered to improve the diversity reception in DAB. The diversity gain depends on the two characters a) correlation of the signals from the different antennas which is influenced by antenna separation, b) parameters of transmitted signals. In this paper, theoretical evaluations of DAB are to be considered. Different parameters of DAB transmitted signals are evaluated by using suitable simulation package.

General Terms
Signal Processing & Digital Radio Channels.

Keywords
Digital Audio Broadcasting, diversity, mobile reception, audio quality, bit error rate

1. INTRODUCTION
At century later, radio broadcasting became one of the most widespread mass media. At this age, all the broadcast transmission standard was based on the analog AM and FM. These radio services provide good quality of services for fixed receptions. But, for mobile reception system was badly affected by multipath propagation. This causes fading and occasional loss of signals. For this reason, Digital Audio broadcasting (DAB) was developed.

DAB is an efficient and reliable radio broadcast system, which provides very high robustness in mobile environment. Even though, it shows some undesired interruptions in spots with insufficient field strength like other digital transmission systems. Particularly, in vehicles on the moving environment, an undesired audio interruption occurs regularly [1]

Diversity is a technique used to get better receiver performance. When multiple antennas are used, different diversity systems can be applied. For DAB, a transmitting diversity is built up as single frequency network (SFN). In SFNs, all transmitters covering a particular area broadcast the same information and operate in the same frequency with continuous coverage zones[9].

For DAB, the antenna diversity is realized at the transmitter. In this paper, the space diversity is considered at receiver site (i.e.) the receiving antennas are erected at different locations [1-4] Hence; the distance between the antennas in terms of the wavelength is an important design parameter. This paper gives a complete overview of DAB transmission theory, diversity systems and combiner techniques in section II. Parameters of DAB signals are analyzed in section III. Simulation results are presented in section IV.

2. ANALYSIS OF DAB THEORY AND DIVERSITY TECHNIQUES
In DAB, Coded Orthogonal Frequency Division Multiplexing (COFDM) modulation and DQPSK modulation techniques has been implemented. The OFDM modulation consists of 1,536 subcarriers that are transmitted in parallel. The useful part of the OFDM symbol period is 1 millisecond, which results in the OFDM subcarriers each having a bandwidth of 1 kHz due to the inverse relationship between these parameters, and overall OFDM channel bandwidth is 1,536 kHz [10] & [11]

Thus, the information is transmitted by the phase difference between two subsequent complex transmit symbols. The OFDM guard interval for Transmission Mode I (TM I) is 246 microseconds, which means that overall OFDM symbol duration is 1.246 milliseconds. The old version of DAB uses convolution coding for error correction coding. The new DAB+ (DAB upgrade) standard incorporated with Reed- Solomon error correction coding.

Diversity is a way to protect against deep fades, a choice to combat fading. The most common application of diversity is ‘Space Diversity”, (i.e.) multiple antennas are used, instead of one. It is easy to realize for mobile reception by second antenna on car’s roof. For DAB, a diversity system that uses directional antennas cannot be used. Since the mobile reception in car requires omni directional antennas. The signals from the multiple antennas have to be combined by the combiner technique. A simple combiner technique is the “Selection Combiner”. It has least complexity and selects path with the highest SNR [3]

3. PARAMETERS OF DAB SIGNALS
In car, DAB receiver experiences a constantly varying coverage area. The receiver must be able to identify the reception quality of the signal from one transmitter and intelligently switch to another transmitter. In general, even though the two transmitters are identical, they might have different transmission parameters (like signal delay etc) or they might use different broadcast systems. That is, same

station can be aired simultaneously over DAB, DAB+ (DAB upgrade) or DMB (Digital Multimedia Broadcasting). In many cases, some of the DAB services are the simulcast of existing FM services with Software Defined Radio (SDR). Sometimes, same DAB service can be simulcast over two or three DAB ensembles emanating from different regions.

Since any DAB system introduces a significant processing delay in the receiver due to digital processing (e.g. de-interleaver, decoder, etc), the overall delay perceived by the listener will be increased. The signal is delayed by 2-4 seconds depending upon the decoding circuitry. FM signals are available without delay. Even though, broadcasters compensate for the propagation delay between the DAB service and FM service, there will still be a significant delay between the DAB decoded output and FM decoded output. It is also observed that the levels of these decoded outputs might not be same. The differences between the signals are minimized by using space diversity and to achieve noise free linking between them[8].

It is necessary, to find the differences of the received signals from the multiple antennas for perfect synchronization.

3.1 Time Delay Measurement

The time delay measurement between the two signal is determined by a cross correlation. The determined cross correlation peak describes the time delay between both signals. Most of the signal processing literatures gives the details on correlation between the two signals either in time domain or frequency domain. The choice of time domain or frequency domain is based on length of the signal. If length of the signal is small, time domain method will give faster result over frequency domain method and vice versa.

Cross correlation of x(n) and y(n) is sequence, $r_{xy}(l)$

$$r_{xy}(l) = \sum_{-\infty}^{\infty} x(n)y(n-l) \qquad l = 0, \pm1, \pm2 \ldots \quad \ldots. (1)$$

Since, Correlation is similar to convolution except for time reversal process, it can be deducted that correlation is convolution of two signals without the folding process. So the above cross correlation equation can be expressed as

$$\boldsymbol{r_{xy}(l) = x(l) * y(-l)} \quad \ldots.. (2)$$

According to properties of convolution, convolution in time domain is equal to multiplication of these signals in frequency domain.

Based on this, the cross correlation of two discrete signals x and y in time domain is same as multiplication of Fourier Transform of X with conjugate of Fourier transform of Y.

Cross correlation output =
IFFT (FFT(X) * Conj FFT(Y))

3.2 Audio Level Measurement

The audio level of the signal not only depends on the amplitude of the signal, but also depends on the amplitude of various frequency components present in the signal. As the human perception of sound varies with respect to the frequencies in it, the loudness of the signal must be measured [8].

The human ear acts like a filter. The outer ear performs direction dependent filtering and the middle ear gives good response only for the mid- range (500-5000Hz) of audible frequencies. Inner ear normally gives same response for all the frequencies. These characteristics of the human ear must be considered while measuring the loudness of the signal.

After applying the filter, the signal contains only the most influenced frequency components. The low frequency components and some of high frequency components, which are not much effective for the human ear, are filtered. The difference between RMS values of the filtered signals gives the actual loudness difference between the two signals.

3.3 Bit Error Rate Calculation

In DAB, OFDM is used as a modulation technique. It can provide large data rates with sufficient robustness to radio channel impairments. It uses many sub carriers, up to 1536 at 1 KHz, where each carrier is independently modulated using Phase Shift Keying (PSK). Hence, in this project, BPSK digital modulation technique for OFDM system over AWGN and Rayleigh fading channels is considered to obtain the BER performance [7].

3.3.1 Relationship between E_s/N_o (or) E_b/N_o for OFDM system

To obtain the BER performance, it is must to find out the relation between E_s/N_o (or) E_b/N_o for OFDM system. Normally for BPSK system, bit energy and symbol energy are the same. But, for an OFDM –BPSK system, they are not the same.

The relation between bit energy and symbol energy is as follows

$$\left(\frac{Es}{N0}\right) = \frac{Eb}{N0}\left(\frac{n.DSC}{n.FFT}\right)\left(\frac{Td}{Td+Tcp}\right) \qquad ----- (3)$$

If we express the above equation in dB, then we can write

$$\left(\frac{\boldsymbol{Es}}{\boldsymbol{N0}}\right)\boldsymbol{dB} = \left(\frac{\boldsymbol{Eb}}{\boldsymbol{N0}}\right)\boldsymbol{dB} + \boldsymbol{10log_{10}}\left(\frac{\boldsymbol{n.DSC}}{\boldsymbol{n.FFT}}\right) + \boldsymbol{10log_{10}}\left(\frac{\boldsymbol{Td}}{\boldsymbol{Td+Tcp}}\right)$$

(4)

where, T_d is the data symbol duration, T_{cp} is the cyclic prefix, n.FFT is size of FFT and n.DSC is the number of subscribers used in the OFDM system.

3.3.2 Rayleigh Multipath Channel model

The Rayleigh fading channel model is reasonable for an environment, where there are large numbers of reflectors. The channel is modeled as n-tap channels with real and imaginary part of each tap being an independent Gaussian random variable.

The circularly symmetric complex Gaussian random variable is in the form

$$Z = X + jY \qquad (5)$$

where, real X and imaginary Y parts are zero mean independent and identically distributed Gaussian random variable. The impulse response is in form

$$h(t) = \frac{1}{\sqrt{n}}(h1(t - t1) = h2(t - t2) + \ldots + hn(t - tn) \quad \text{----- (6)}$$

where, h_1 (t - t_1) is the channel coefficient of the first tap, h_2 (t - t_2) is the channel coefficient of the second tap and so on. The real and imaginary part of each tap is an independent Gaussian variable with mean 0 and variance ½. The term $\frac{1}{\sqrt{n}}$ is for normalizing the average channel power over multiple channel realizations to 1.

3.3.3 BPSK Modulation

BPSK is the simplest form of PSK [5] & [6]. It uses two phases which are separated by 180^0 and so can also be termed 2-PSK. For BPSK modulation the channel can be modeled as

$$y = Ax + n \quad \text{----- (7)}$$

where, y is the received signal at the input of the BPSK receiver, x is the modulated signal transmitted through the channel, 'a' is the channel amplitude scaling factor for the transmitted signal and it is usually 1. 'n' is the Additive White Gaussian Noise random variable, with zero mean and variance σ^2. For AWGN, the noise variance in terms of noise power spectral density (N_0) is given by

$$\sigma^2 = \frac{N0}{2} \quad (8)$$

The theoretical BER for BPSK modulation scheme over an AWGN channel is given by

$$Pb = \frac{1}{2} erfc\left(\sqrt{\frac{Eb}{N0}}\right) \quad \text{----- (9)}$$

For BPSK modulation schemes the symbol energy is given by

$$Es = RcEb \quad \text{----- (10)}$$

where E_s=symbol energy per modulated bit (x), R_c is the code rate of the system if a code scheme is used. Consider the case, no coding scheme is used, therefore, R_c=1. E_b is the Energy per information bit.

Assuming E_s=1 (Symbol energy normalized to 1).
E_b/N_0 can be represented as

$$\frac{Eb}{N0} = \frac{Es}{RcN0} \quad \text{----- (11)}$$

From the equation 8, the noise variance for the given E_b/N_0 can be calculated as

$$\sigma^2 = \left(2R0\frac{Eb}{N0}\right)^{-1} \quad \text{----- (12)}$$

BER for BPSK in a Rayleigh fading channel is defined as

$$Pb = \frac{1}{2}\left(1 - \sqrt{\frac{\frac{Eb}{No}}{\frac{Eb}{No}+1}}\right) \quad \text{----- (13)}$$

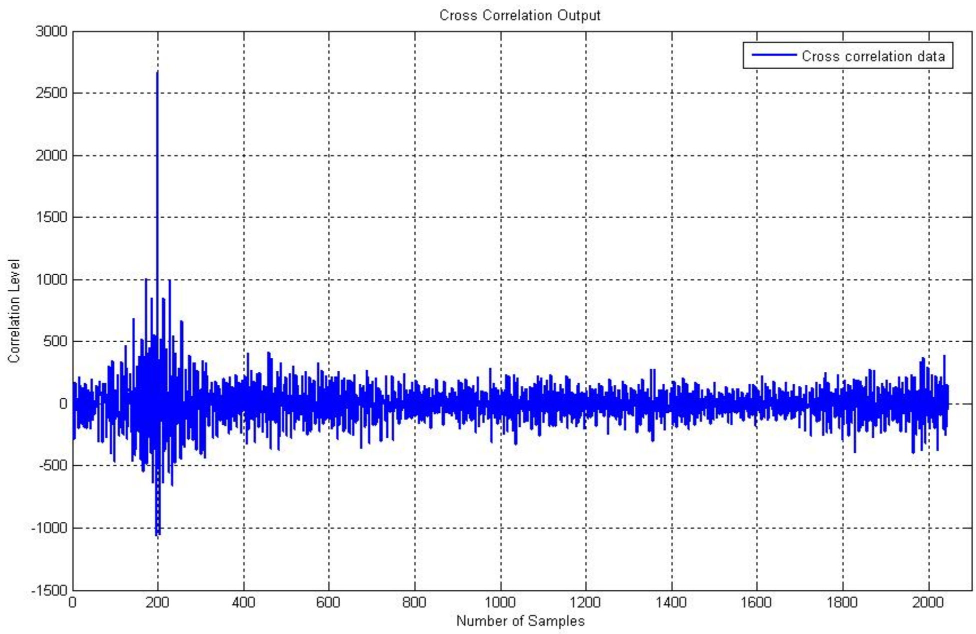

Fig.1: Cross correlation output

4. RESULTS AND DISCUSSION

In this project, various parameters of transmitted signals are measured for moving DAB receiver. The time delay measurement and audio level measurement between the two transmitters are measured and it must be used to ensure the smooth transient between the transmitters without any audible or loss of audio in case of bad reception situations.

4.1 Time Delay Measurement

Fig.1 shows the cross correlation output of audio signals from two different antennas. Maximum peak in the fig.1 represents the maximum time delay between the sample signals. The future work can be extended by designing a switching control algorithm for receiver side to decide the optimal switching or switching back, based on the various transmission parameters and on reception conditions.

4.2 Audio Level Measurement

Fig.2 shows dB difference of audio signals from two different antennas. The difference between the RMS values of the sample signals gives the actual loudness difference between that signals. The future work can be extended by setting an alternate audio linking source module to perform the time delay compensation. The audio level of alternate audio source is adjusted and ensures that the adjusted level is equivalent to main audio level, using measured dB difference value.

4.3 Bit Error Rate Calculation

From the simulation results of BER, we can observe that the theoretical and simulated result of BPSK modulation over AWGN channel is the same. From the fig.3, the numbers of taps do not introduce much deviation to the real performance given by simulation results. And also found that as the energy per bit to noise ratio increases in any systems, a document in bit error rate is encountered. The purpose of this measurement is to implement and find the efficient combination of modulation system that performs better in the radio channels that are mostly multipath. The future work can be extended to measure the performance of D-QPSK modulation for DAB reception.

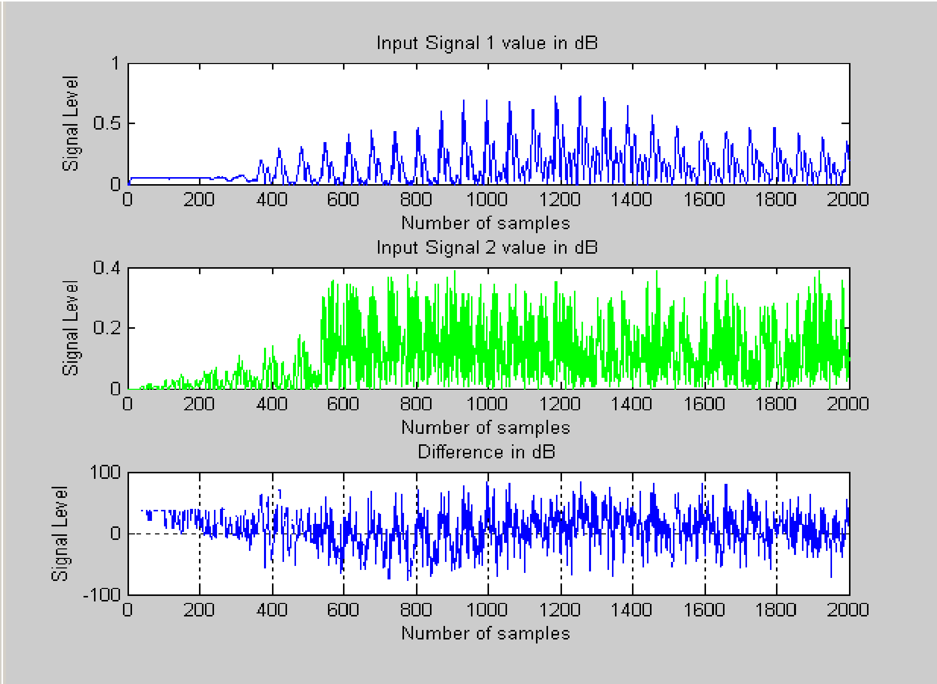

Fig.2: dB difference

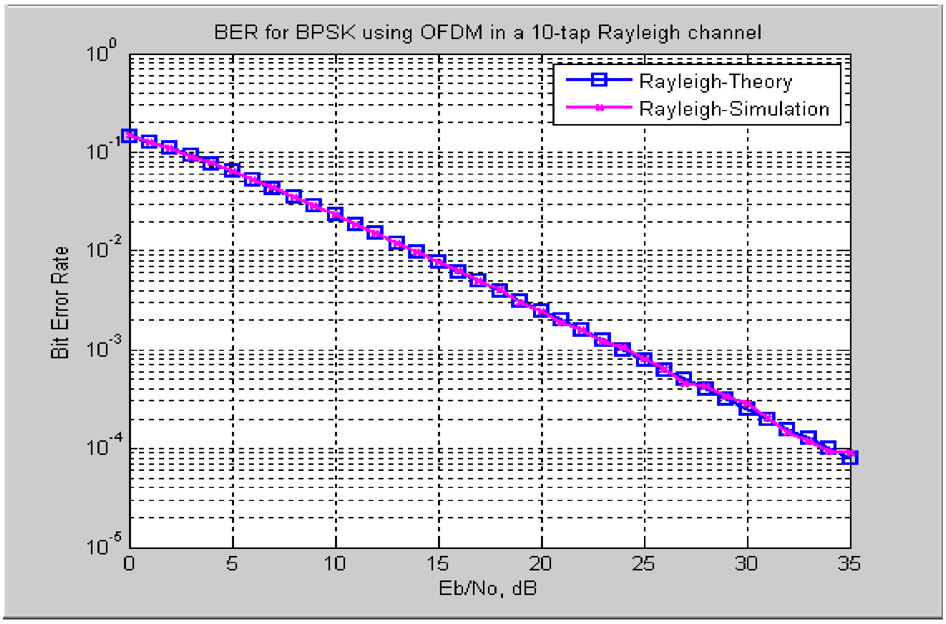

Fig.3: Bit Error Rate performance

5. CONCLUSION

In this paper, the theoretical evaluations of DAB transmitter and various types of diversity and combiner techniques are discussed. The details about audio level and time delay measurement of DAB transmitted signals are presented and simulated using sample audio signals. The performance of the OFDM system using binary phase shift keying is performed.

In future work, switching control algorithm for time delay and adjustment audio system for audio level can be carried out with space diversity to improve the reception of DAB signals in mobile (moving) environment.

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