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Performance Analysis of Digital Audio Broadcasting System through AWGN and Rayleigh Channels

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Abstract-This paper presents the performance analysis of Eureka-147 DAB system. DAB transmission mode-II is implemented. A frame-based processing is used in this study. Performance studies for AWGN and Rayleigh channels have been conducted. For all studies SER has been used as performance criteria.

Index Terms- Radio, Digital Audio Broadcasting (DAB), Eureka 147, European Telecommunications Standards Institute (ETSI), Symbol Error Rate (SER), coded Orthogonal Frequency Division Multiplex (COFDM), and Frequency Response.

I. Introduction

Radio broadcasting is one of the most widespread electronic mass media comprising of hundreds of programme providers, thousands of HF transmitters and billions of radio receivers worldwide. Since the broadcasting began in the early 1920s, the market was widely covered by the AM services. Then came the FM and now we live in a world of digital communication systems and services. Digital telecommunication has advantages over analog systems such as storage capacity, reliability, and quality of service, miniaturization and many more.

The new digital radio system, Digital Audio Broadcasting (DAB) has the capability to replace the existing AM and FM audio broadcast services in many parts of the World in near future. This was developed in the 1990s by the Eureka 147 DAB project. DAB is very well suited for mobile receivers and provides very high tolerance against multipath reception and inter symbol interference (ISI). It allows use of single frequency networks (SFNs) for high frequency efficiency. In several countries in Europe and overseas, broadcasting organizations, network providers and receiver manufacturers are already implementing digital broadcasting services using the DAB system. Perceptual audio coding (MPEG-2), Coded Orthogonal Frequency Division Multiplexing (COFDM), provision for the multiplex of several programmes and data transmission protocols, are the new concepts of digital radio broadcasting.

II. RELATED WORK

During its development DAB system has been publicly demonstrated many times. It has been subject to extensive computer simulations and field tests in Europe and elsewhere. It is now in regular service in many European countries and throughout the world. In 1995, the European DAB Forum (Euro Dab) was established to pursue the introduction of DAB services in a concerted manner world-wide and it became the World DAB Forum in 1997 [1]. As a result of developments within the Eureka 147 project, the DAB Standard or DAB Specification in the form of EN 300401 was approved by the European Telecommunications Standards Institute (ETSI), which defines the characteristics of the DAB transmission signal, including audio coding, data services, signal and service multiplexing, channel coding and modulation [2].

The first digital sound broadcasting systems providing CD-like audio quality was developed in early 1980s using Satellite technology. The system employed very low data compression and was not suitable for mobile reception. It used frequency in the range 10-12 GHz. Therefore it was not possible to provide service to large number of listeners. It was realized terrestrial digital sound broadcasting would do the job and to develop this new digital solution an international research project was necessary. So, in 1986 few organizations from France, Germany, United Kingdom and The Netherlands signed an agreement to cooperate in the development of a new standard and with this Eureka-147 project was born [2] [6]. Members of European Broadcasting Union (EBU), who were the part of work on the satellite delivery of digital sound broadcasting to mobiles in the frequency range between 1 and 3 GHz, also joined the Eureka-147 project. Later International Telecommunications Union (ITU-R) and the European Telecommunications Standards Institute (ETSI) started the standardization process.
Following goals were set up for DAB from the beginning with the sole aim of quality audio for mobile reception:

- High quality digital audio services (near CD quality).
- Well suited for mobile reception in vehicles, even at higher speeds.
- Efficient frequency spectrum usage
- Transmission capacity for ancillary data.
- Low transmitting power.
- Terrestrial, cable and satellite delivery options.
- Easy tuning of receivers.
- Large coverage area than current AM and FM systems.

Eureka 147 consortium alone started choosing the most appropriate transmission method based on thorough simulation and field test. Results showed that broadband solutions performed better than the narrow-band proposal, while the frequency-hopping solution was considered too demanding with respect to network organization. Since the spread-spectrum

was not developed as hardware therefore coded Orthogonal Frequency Division Multiplex (COFDM) system was chosen finally.
The next issue was audio coding standard selection for the Eureka-147 project. By that time the MPEG (Moving Pictures Expert Group) had already been standardized for data compression both video and audio coding. The solutions proposed by the Eureka-147 were sent to the MPEG Audio group to be evaluated with other several options from other countries. The performance offered by the methods submitted by the Eureka consortium was clearly superior so they were standardized by the MPEG as MPEG Audio Layers I, II and III. It took a long time until the final decision to which standard should be used for DAB was taken [4]. Finally, Layer II, also known as MUSICAM was chosen.

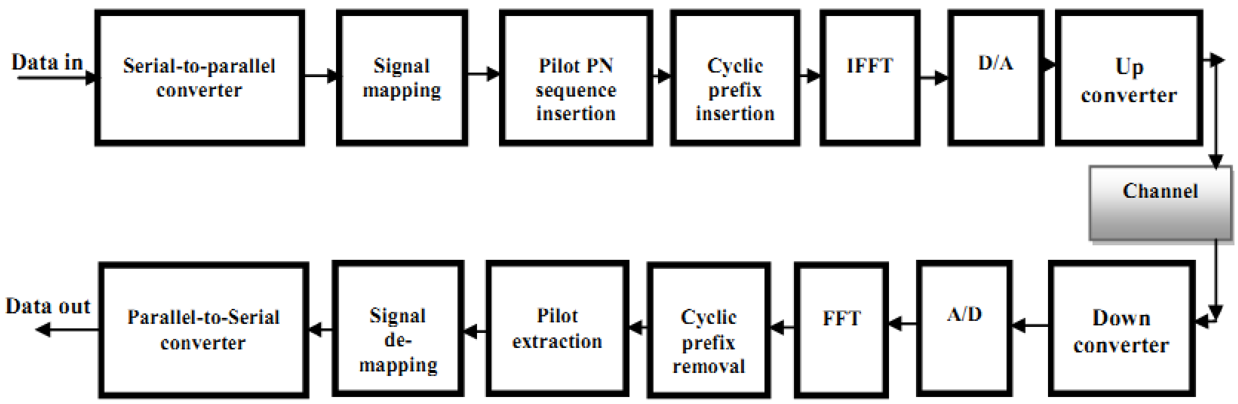

Figure 1: Basic block diagram of OFDM system

Another important specification for the DAB was the bandwidth consideration. From a network and service area planning point of view, one transmitter with the 7 MHz bandwidth of a TV channel was too much inflexible, but showed very good performance in a multipath environment [2]. Therefore considerable reduction in transmission bandwidth was necessary. In Canada experiments with the COFDM system also revealed that performance degradation begins around 1.3MHz and lower. Therefore appropriate bandwidth for a DAB channel was fixed at 1.5MHz.

With this one 7 MHz TV channel can be divided into four DAB blocks, each carrying ensembles of five to seven programs. The first DAB standard was achieved in 1993 and then in 1995 the ETSI adopted DAB as the only European standard for digital radio. The Eureka 147 DAB standard as digital radio is accepted Worldwide except, USA and Japan.

III. ARCHITECTURE

This section gives an overview of the conceptual architecture of Digital audio Broadcasting and describes the basic building blocks. The design decisions draw from several concepts and approaches. Since COFDM is the heart of Digital Audio Broadcasting (DAB), to be more presice OFDM, therefore the basic block diagram of the same is given below.

IV. SIMULATION AND PERFORMANCE EVALUATION

The above system was simulated in MATLAB's Simulink. The following are the results obtained from the software imitating the DAB quite successfully.

Figure 2 shows the real and imaginary parts of the Transmitted signal. This Composite signal is fed to the channel after the Parallel to serial converter block.

Figure 3 Shows the Real and Imaginary parts of the Received Composite signal at the output of the OFDM Receiver. As compared with the input signal the Wave shape, Frequency and Phase is the same as that of the input signal.

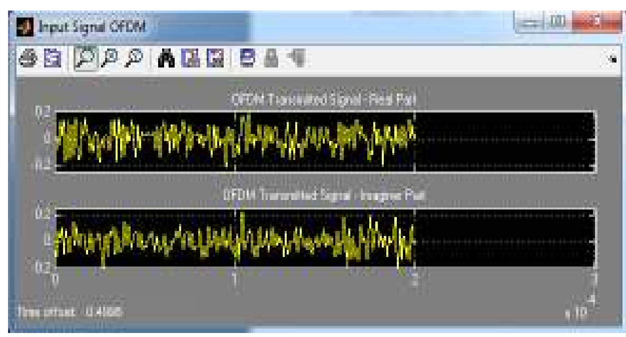

Figure2: Transmitted signal in the simulator.

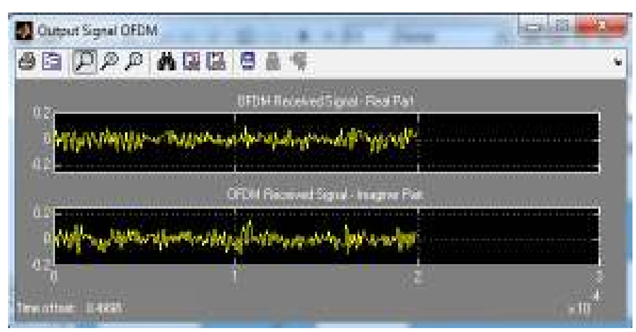

Figure 3: Received signal in the simulator.

The only difference between the two signals is the Amplitude level or the power level. The output signal has a lower power as compared with the input signal.

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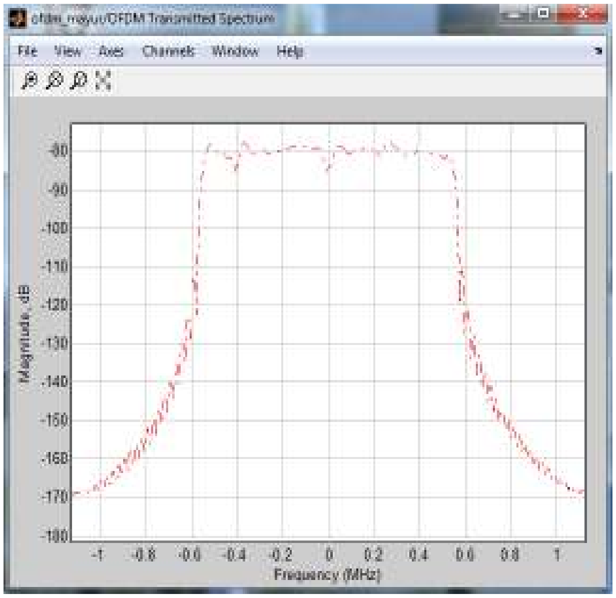

Figure 4: Spectrum of Transmitted Signal in the simulator.

Figure 4 Shows the Frequency Response of the Transmitted Composite signal. This can be seen on a spectrum analyzer places at the output of the parallel to serial converter block in the simulation

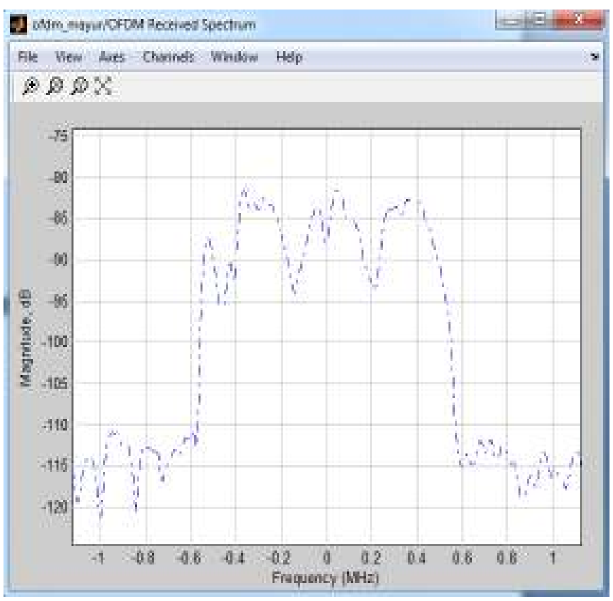

Figure 5: Spectrum of Received signal in the simulator.

Fig 5 shows the Frequency Response of the output signal, which can be seen at the Spectrum Analyzer placed at the output of the Decoder in the Receiver. As compared with figure 4.7, the spectrum is still in good condition. The magnitude has not decreased so much; it is not much affected by noise, as can be seen in the frequency axis from -0.6 to +0.6.

There is noise and distortion in the spectrum especially at higher frequencies outside the bandwidth of 1.5 Mhz, i.e. from -0.6 to -1 and from +0.6 to +1.

Table 4.1 Parameters of the main Composite signal.

Main Composite Analysis Signal			
Second	**Total Symbol**	**Error Symbol**	**SER**
20	1.683$_e$+004	0	0
40	3.425$_e$+004	0	0
1.00	5.168$_e$+004	18	0.00035
1.20	6.928$_e$+004	22	0.00032
1.40	8.661$_e$+004	36	0.00042
2.00	1.033$_e$+005	44	0.00043
2.20	1.197$_e$+005	44	0.00037
2.40	1.375$_e$+005	55	0.0004

Table 1 analysis the Main Composite signal, it shows that as the Total symbol bits increases with the passage of time, the Error symbols also increase. Initially the Error Symbols are Zero but at time = 1 minute it starts to increase, and goes up to 55 at time = 2 minute and 40 second. Also the Symbol Error Rate (SER) increases with Total symbols transmitted. It is zero initially but starts to increase at time = 1 minute. It goes upto 0.0004 at time = 2 minute and 40 second.

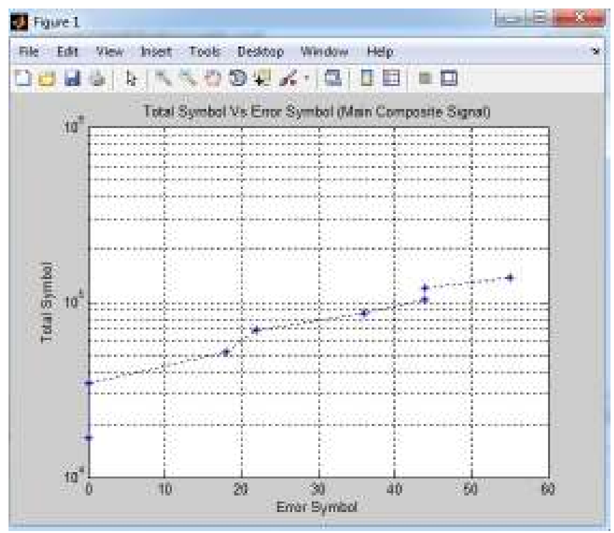

Figure 6: Total symbol Vs Error Symbol (Main composite Signal)

The graph shown in Figure 4.8 is a relationship between Total Symbols sent from Transmitter to Receiver and the Error symbols encountered at the receiver. The Total symbols are taken on Y – axis, and the values starts from 10^4 Symbols to 10^6.

Error symbols are taken on X –axis. The value varies from 0 to 60 symbols. The error symbol is 0 when the Total Symbol varies from 10^4 up to $10^{4.3}$. Then it increases up to 60 when the Total Symbols vary from $10^{4.3}$ to $10^{5.05}$. Therefore the Error Symbols occur between $10^{4.3}$ to $10^{5.05}$.

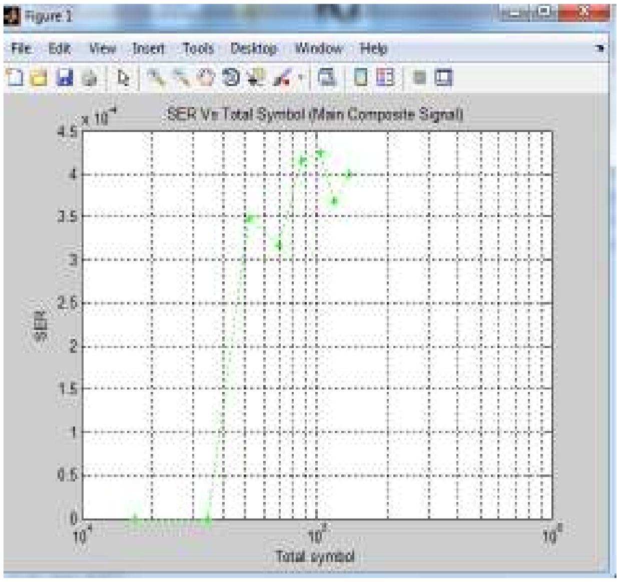

Figure 4.9 SER Vs Total symbol (Main Composite signal)

Figure 4.9 shows how Symbol Error Rate (SER) changes with respect to the Total symbols sent. On X – Axis Total Symbol are taken from 10^4 to 10^6, but the SER varies only from $10^{4.08}$ to $10^{5.05}$.

On Y – Axis SER is taken. SER is 0 until when the Total symbol is at $10^{4.25}$. Then it rises up to $4.3\text{x}10^{-4}$, which is in line with ETSI (EN 300 401).

Table 4.2 Parameters of the Encoded signal.

Encoded Signal Analysis			
Second	**Total Symbol**	**Error Symbol**	**SER**
20	2.295e+004	29	0.001264
40	4.671e+004	86	0.001841
1.00	7.047e+004	131	0.001859
1.20	9.447e+004	177	0.001874
1.40	1.181e+005	220	0.001863
2.00	1.409e+005	247	0.001753
2.20	1.632e+005	294	0.001801
2.40	1.875e+005	347	0.001851

Table 4.2 shows the analysis of the encoded signal. With the passage of time the Total Symbols increases. The Error symbols also increases from 29 to 347. The Symbol Error Rate (SER) also increases with respect to the increase in the Total symbol.

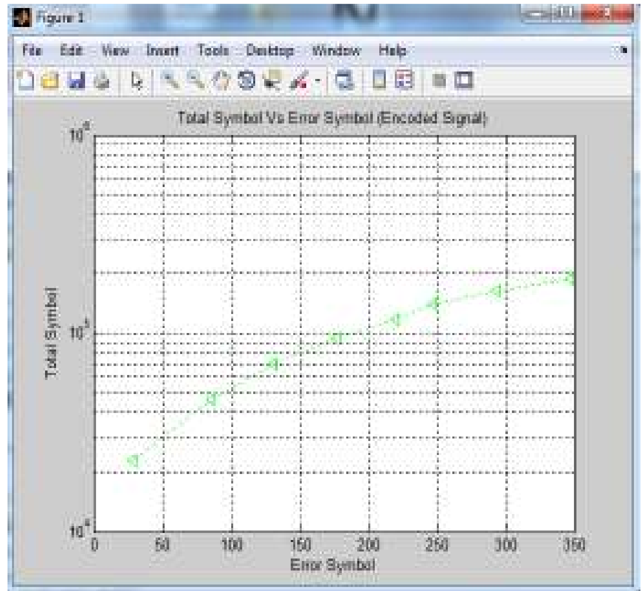

Figure 4.10 Total Symbol Vs Error Symbol in Encoded Signal.

On Y – axis Total Symbol has been taken from 10^4 to 10^6. On X – axis Error Symbol is taken from 0 to 350. The Error symbol varies between Total Symbol $10^{4.12}$ to $10^{5.09}$. The Error Symbol increases from 30 onwards and becomes constant at 350.

Thus Coding helps to make the Error symbol stable. The Error symbol does not increases after 350, but remains constant.

On X – Axis Total Symbol are taken from $0.2\text{x}10^5$ to $2\text{x}10^5$. On Y – Axis SER is taken. The SER increases from 1.26 to 1.84. Then it remains somewhat constant in the range of 1.75 to 1.88. Thus coding helps to stabilize the SER.

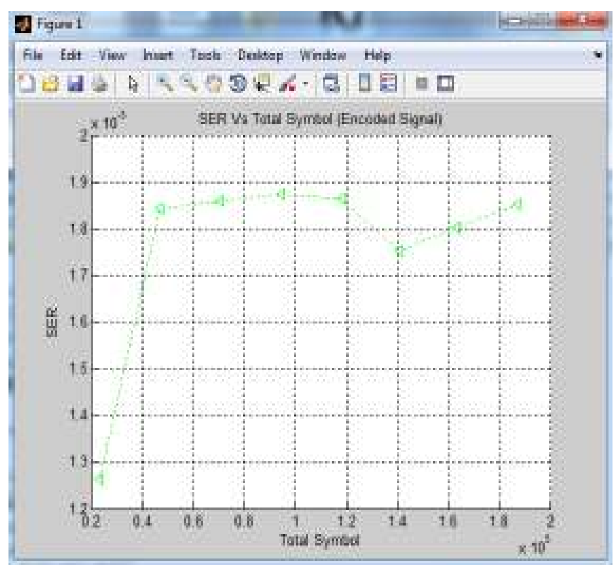

Figure 4.11 SER Vs Total Symbol in the Encoded Signal.

V. Conclusion

We presented the simulated model of DAB based on OFDM system, communicating in AWGN and Rayleigh channel. The performance analysis of the system shows that degradation in limited only to a certain part of the spectrum and also encoding and decoding at the Transmitter and receiver respectively lowers the degradation. This work used the previous work of [25] as reference.

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