

# Design and Development of Coded OFDM based Digital Audio Broadcasting System using Concatenated Convolutional Turbo Codes

T.Nagarjuna<sup>1</sup>, K.Nehru<sup>2</sup>, A.Usharani<sup>3</sup>, L.Raviteja<sup>4</sup>

<sup>1</sup>Assistant Professor, Department of Electronics and Communication Engineering,  
Institute of Aeronautical Engineering, Hyderabad – 500 043, India.  
nagarjuna473@gmail.com

<sup>2</sup>Professor, Department of Electronics and Communication Engineering,  
Institute of Aeronautical Engineering, Hyderabad– 500 043, India.

<sup>3</sup>Assistant Professor, Department of Electronics and Communication Engineering,  
Institute of Aeronautical Engineering, Hyderabad – 500 043, India.

<sup>4</sup>Assistant Professor, Department of Electronics and Communication Engineering,  
Institute of Aeronautical Engineering, Hyderabad – 500 043, India.

**Abstract—** Digital Audio Broadcasting (DAB) is an amazing technology, achieving its promise of certainly delivering high quality digital audio in the most vindicate mobile and fixed receiver environments. In this paper, the dissection of Bit Error Rate (BER) for Digital Audio Broadcasting (DAB) process with different channel coding techniques using Coded Orthogonal frequency division multiplexing (C-OFDM) in Matlab-Simulink Environment. The interpretation is carried out for convolution and turbo codes in an Additive White Gaussian Noise Channel (AWGN) depending on code generator polynomials with distinct constraint lengths. Finally, at a certain point the intricacy of the decoder is so high that it becomes tough to execute and also there is no considerable reduction in BER. In addition the coding gain is possible with turbo codes with affordable coding and decoding complexity with reduction in BER. BER for a finest music signal is considered to be  $10^{-3.5}$  by using turbo coding, it is approximately achieved with an  $E_b/N_o$  of 7 dB is attained using turbo coding, when distinguish with convolution coding which features high computational complexness.

**Keywords:** Coded Orthogonal frequency division multiplexing, Additive White Gaussian Noise Channel, Digital Audio Broadcasting, Bit Error Rate.

## I. INTRODUCTION

The computerized radio framework DAB is an extremely offbeat and general sight and sound communicate framework. It is an advanced technique for conveying radio administrations from the studio to various beneficiaries and guarantees to give the sound of reduced circle quality, with less impact from multipath contortion and interchannel, Co channel impedances. It is additionally appropriate for portable gathering and give high quality against multipath gathering. Other than that it gives astounding advanced sound administrations in mono, two-channel or multi channels. it provides other data services such as travel and traffic information[1].

If a digital signal is transmitted between two systems, the signal gets distorted because of the addition of noise to the transmitted signal. We cannot increase the ratio beyond this limit. Hence, for a fixed value of  $E_b/N_o$ . We have to use some kind of coding in order to improve the quality of the transmitted signal [2].

Another disadvantage of using coding is that we can reduce the required value of  $E_b/N_o$  for a fixed bit error rate. This, in turn, reduces the transmitted power and the size of the antenna. Error control coding is a good way to achieve a very low bit-rate transmission of error after the noisy-band limited channel. This is theoretically possible to achieve as stated in Shannon channel theorem was explained[2]. The detection and correction of transmission errors, the extra bits are added to the data bits at the time of transmitting the signal. These extra bits are known as parity bits. These parity bits detector sometimes correct the error[3].The need of flexibility while connected to the network is fortify the magnification of wireless communication. OFDM is well suited for Single Frequency Networks (SFN) due to extension of cycles duration was explained [4].

This proposed paper implements OFDM application standard called Digital Audio Broadcasting (DAB) system model in Matlab/Simulink environment. The attainment is considered over an AWGN channel. C-OFDM technique is studied using conventional turbo codes is computed and bit error rates (BER) are calculated and analysis is based on a different channel coding parameters [5].

## II. SYSTEM MODEL

### A. Proposed Block Diagram

The Figure 1 shows that both audio and data services are used. The analog signal is concealed and correlated to channel coding like convolutional and turbo codes and then bit streams are 16-QAM mapped. The data can then be passed to the OFDM generator by making use of IFFT algorithm the high data rate, bit stream is break into 'M' parallel data streams of less data rate and separately modulated onto orthogonal sub carriers. Because of the property of orthogonality, the sub carriers are 90 degrees phase with each other so the Inter Symbol Interference is Zero. Theoretically and at last the symbol is provided with cyclic prefix and this frame structure is transmitted through AWGN Channel. Simultaneously the Figure 2 shows the Simulink model for convolutional turbo encoder and Viterbi and turbo code decoder and Simulink model runs in normal mode.

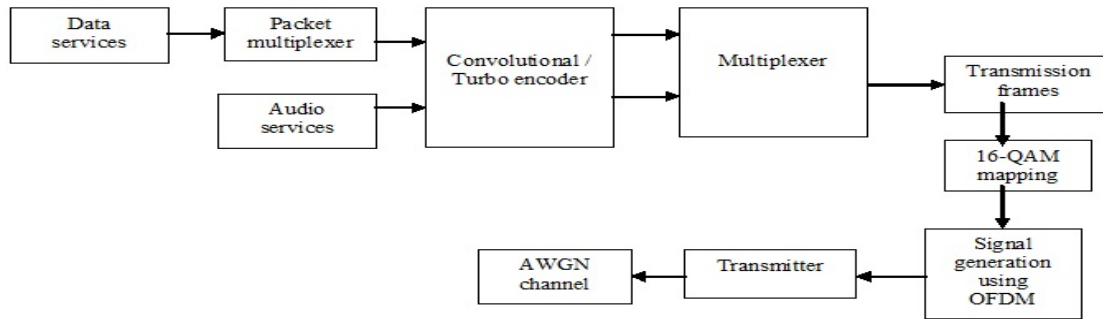


Fig 1. Proposed Block Diagram of Digital Audio Broadcasting System

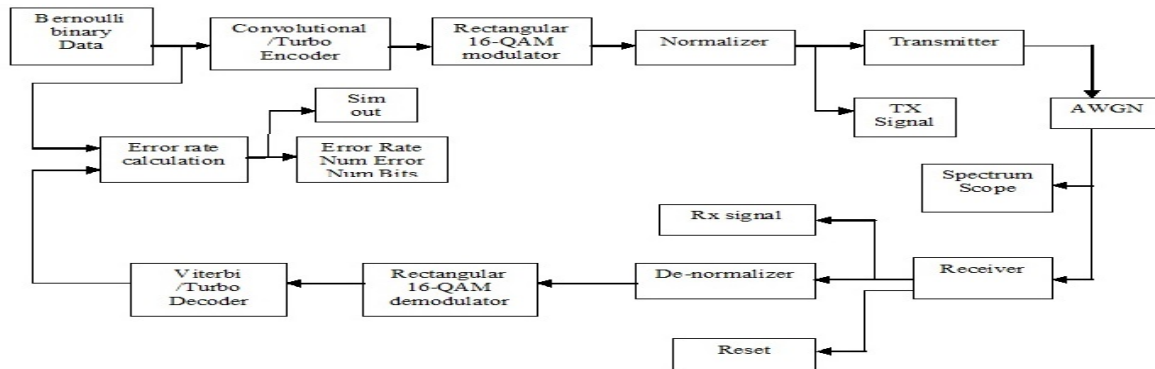


Fig 2. Proposed Simulink Block Diagram for DAB System using Convolutional/Turbo Encoder and Viterbi/Turbo decoder

### B. DAB transmission modes

The main reason to use Coded Orthogonal Frequency Division Multiplexing (C-OFDM) is to improve the spectrum efficiency. The Table 1 shows the duration of transmission frame for different transmission modes of DAB [6].

TABLE 1: DAB Transmission Modes

Transmission modes	Duration of Transmission frame	Number of FIBs per Transmission frame	Number of CIFs per Transmission Frame
I	96	12	4
II	24	3	1
III	24	4	1
IV	48	6	4

This paper uses transmission mode-I for simulation. Table2 shows different transmission modes for DAB with no of subcarriers and spacing between them and length of FFT in OFDM and each transmission mode has different radio frequency which will be useful in many applications such as satellite digital radio [5].

TABLE 2: DAB Transmission Modes Including FFT Length and Subcarrier Spacing

Modes of Transmission	No of Subcarriers	Spacing between subcarrier	Length of FFT	Maximum radio Frequency
I	1536	1 KHz	2048	375 MHz
II	384	4 KHz	512	1.5 GHz
III	192	8 KHz	256	3 GHz
IV	768	2 KHz	1024	750 MHz

### III. CHANNEL CODING

#### A. Encoder for Convolutional Coding

The figure 3 shows the essential unit for the convolutional encoding is tapped shift register with (N+1) stages. The tap gains  $G_0, G_1, \dots$  etc. are nothing but binary digits 1s and 0s. The binary digits 0s and 1s represent an open and short circuit. The message bits enter one by one into the tapped shift register and then mixed by mod-2 addition to form the encoded bit  $x$ .

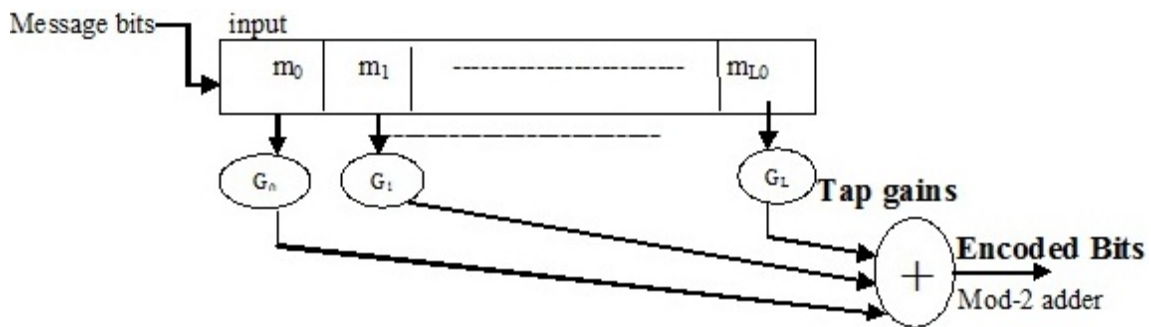


Fig3. Block Diagram of Convolutional Encoder

#### B. Viterbi Algorithm

Viterbi decoding technique is the final processing to be performed at the receiver. Let  $M$  represents the message vector and  $N$  be the corresponding code vector. Let this code vector be applied to the input of discrete memory less channel. Let  $T$  be the received code vector. This may be same or different form the transmitted code vector  $N$ . The decoder is expected to make an  $\hat{M}$  of the message from the received vector  $T$ . There is a one to one correspondence between  $M$  and  $N$ . The decoding rule developed to choose the estimate  $\hat{N}$  when  $T$  is received will be called optimum if and only if the probability of receiving  $T$  when  $N$  was sent. The  $\log P(Y/X)$  is called as log – likli hood function.

#### C. Turbo Codes

It has its unique iterative from hence the turbo code is listed as a separate category under the structured sequences. These codes achieve a  $E_b/N_o = 0.7\text{dB}$  for BPSK modulation and code rate of  $1/2$  in AWGN channel with bit error probability of  $10^{-5}$ .

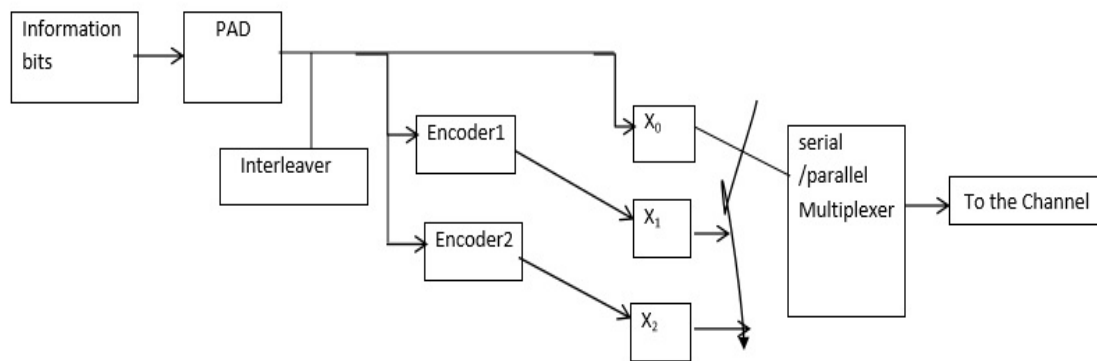


Fig 4. Turbo Encoder

The figure 4 explains the encoding operation in that the interleaver. The two concerns in interleaver design are its size and map.

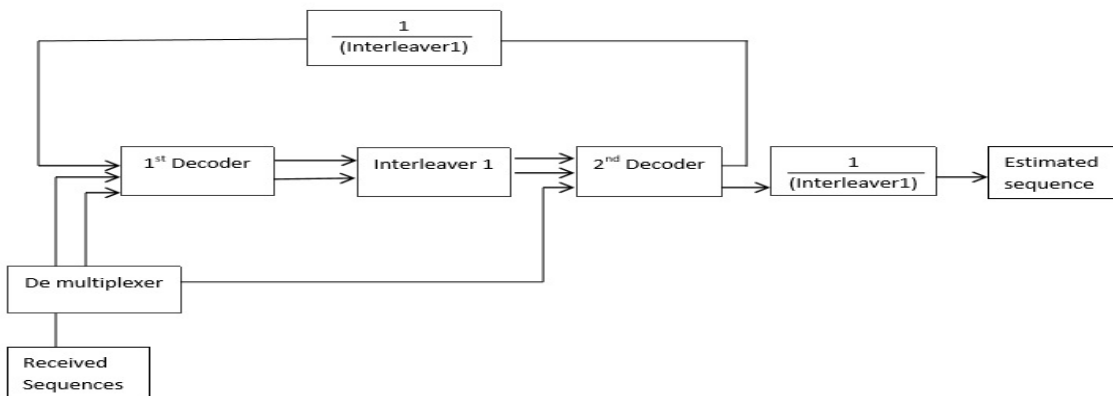


Fig 5. Turbo Decoder

For turbo codes the figure 5 shows two encoded sequences and decoding each one of them to get a first estimate of the information sequence. This estimate sequences are applied to soft decision input [6].

#### IV. SIMULATION RESULTS

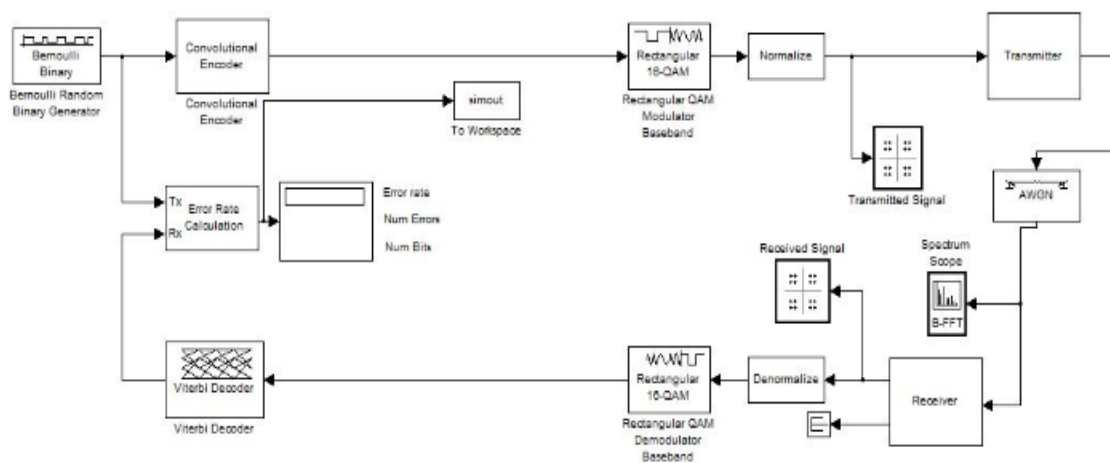


Fig6. Simulink diagram for DAB system transceiver using Convolution coding & Viterbi decoding.

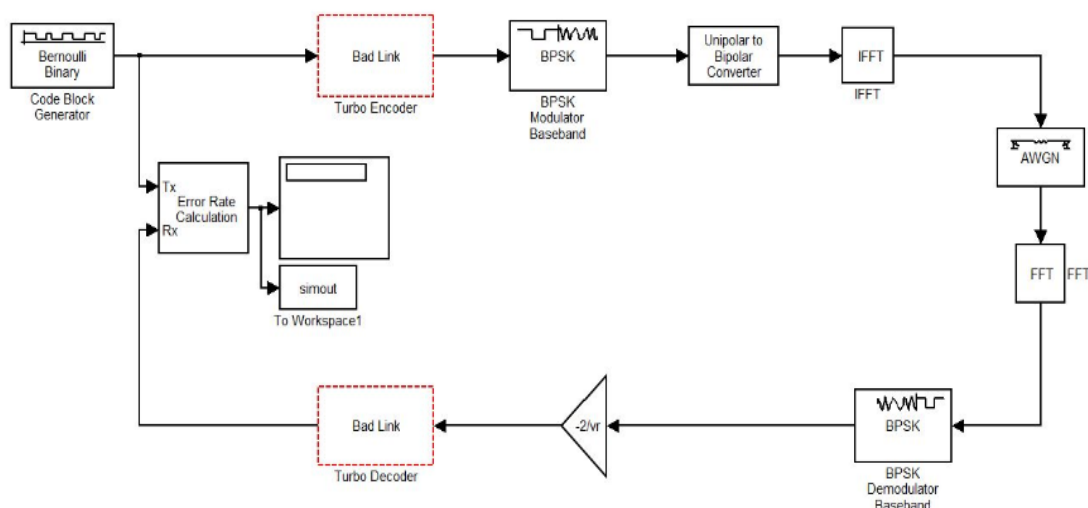


Fig 7. Simulink diagram for DAB transceiver using Turbo encoder and decoder.

The Simulink model shown in figure 6 and figure 7 uses a frame based processing. It is not possible in practice to work with only one frame rate throughout the entire DAB system due to the complexity. Simulink supports multiple frame rates. The convolutional decoder performs the inverse operations and employs a 4-bit soft decision Viterbi decoder. The Figure 6 shows the OFDM transmitter maps 1536 complex 16-QAM symbols to the corresponding sub-carriers and zero padding to 2048 subcarriers and then performs an IFFT operation to obtain a time domain representation of the symbols

TABLE 3: Simulation Parameters

Transmission mode	Model
Subcarriers used	1536
Duration of Transmission frame	96 ms
OFDM symbols per transmission frame	76
Sampling time	0.48828
Length of frame	196608
Fast Fourier transform length	2048
Cyclic prefix	504
Length of OFDM	2552
Channel coding schemes	Convolutional and turbo coding
Modulation	16-QAM
Channel	AWGN

TABLE 4: Channel Coding Parameters

Channel coding types	Constraint length	Octet format of code generator polynomial	Feedback polynomial
Convolutional coding	3	7,5	7
	4	15,13	15
Turbo coding	5	34,27	34
	6	71,57	71
	7	171,133	171

Digital Audio Broadcasting system has four modes of transmission modes here transmission mode- I is selected with different modulation technique and tested both on matlab/ simulink model for different channel parameters.

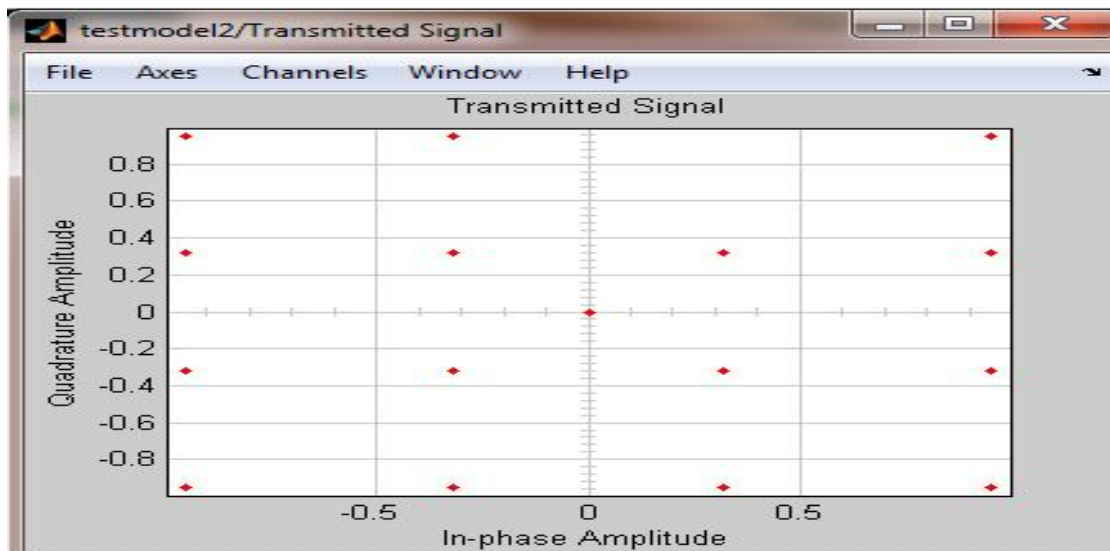


Fig 8. Transmitted Signal of Digital Audio Broadcasting System

The figure 8 shows the transmitted signal from the output of AWGN channel and Whichever rectangular QAM constellation has been equivalent to superimposing two ASK signals on quadrature carriers with both Inphase and Quadrature phase components. Rectangular constellations are preferred because of its simplicity in modulation and demodulation.

In the M - QAM constellation the adjacent symbols in the transmitter constellation should not differ more than one bit. It is also achieved by converting the input symbols to reflect binary coded symbols and then mapping them to the particular QAM constellation. This intermediate step can be skipped altogether with the help of Look-Up-Table approach this intermediate step can be skipped and LUT translates the input symbol to desired position in the constellation.

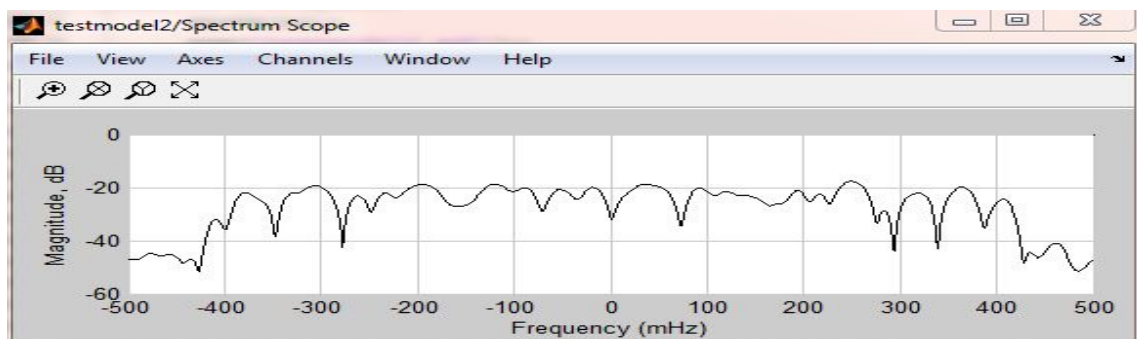


Fig 9. Spectrum Scope

The figure 9 shows the output of spectrum scope with the help of Spectrum Analyzer block. The scope block accepts input signals with one or more input ports with the following characteristics of a signal such as fixed point data type or floating point data type, signal in discrete sample time, signal with real or complex values. In this paper the spectrum analyzer block runs in Accelerator simulation mode.

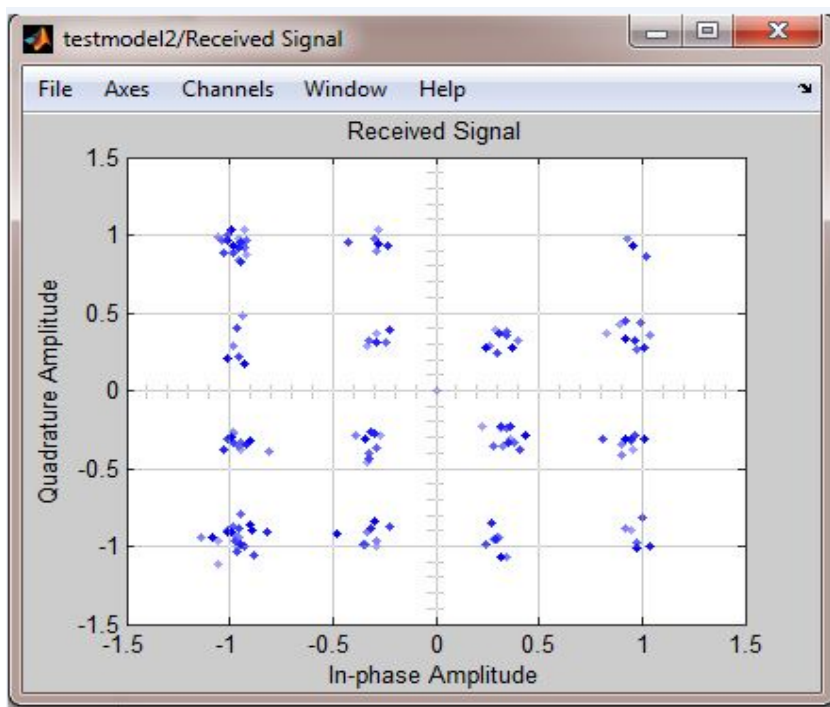


Fig 10. Received signal

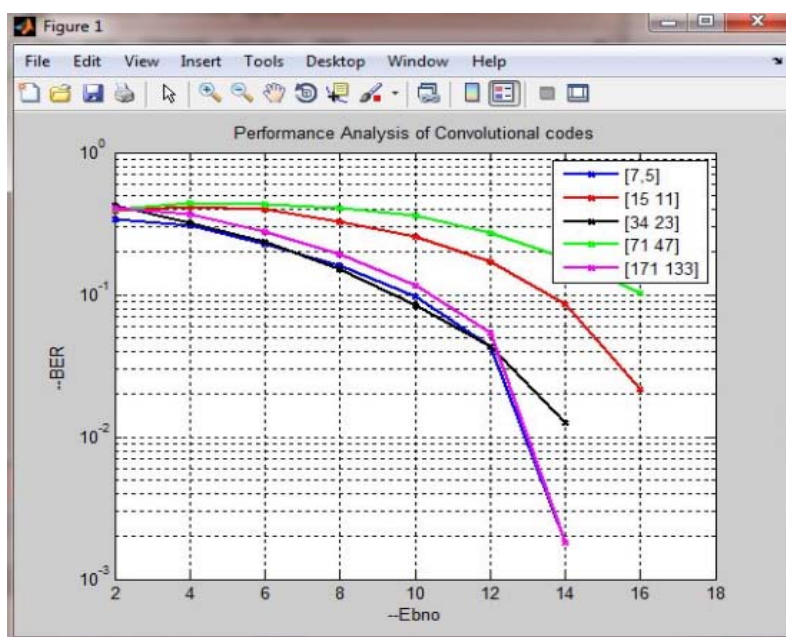


Fig 11. Simulated output of Digital Audio Broadcasting System

For the constraint length  $K=7$  corresponding code generator polynomial  $[171,133]$  as the value of energy bit per noise  $E_b/N_0$  increases the bit error rate is dramatically reduced Therefore the turbo codes have an astounding performance of bit error rate at low  $E_b/N_0$ .

### V. CONCLUSION AND FUTURE SCOPE

Thus the Digital Audio Broadcasting system using Coded OFDM is implemented using a convolutional turbo code encoder and Viterbi decoder for different channel constraints over an AWGN channel and Bit-Error-Rate (BER) is measured. Convolutional codes are error-detecting codes used to reliable transmit digital data over unreliable communication channel system to channel noise. The parallel concentrated codes also called turbo codes has solved the structure of dilemma and randomness through concentrated codes and interleaving respectively. Convolutional codes are easy to implement than turbo codes. They produce randomness in coding due to interleavers which are absent in convolutional codes. With the help of turbo code the bit error rate is

reduced. Using turbo coding, it is virtually concluded with an  $E_b/N_0$  of 7 dB and achieved by employing turbo coding, when correlated to convolutional coding which is of high computational complexity.

The work in this paper can be extended in a number of directions such as Femtocell. The Viterbi decoding of the conventional codes can be improved, just by using 8 bit soft decision decoding with the adaptation of quantization levels. Another possibility could be use of adaptive symbols and frame synchronization using different phase reference symbols and last use of MPEG-4 audio coding may be applied to source coding.

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### Author Biographies



Nagarjuna T received his Bachelor Degree in Electronics and communication Engineering from Narayana Engineering College, JNTU University in 2011 and he obtained his Master Degree in Communication Systems from Loyola institute of technology, Anna University in 2013. He awarded as Anna university Rank holder[20] for M.E degree in 2014 and his main research areas are MIMO, OFDM. He has published papers on these topics in various international journals.



Nehru K received his Bachelor Degree in Erode Sengunthar Engineering College, Anna University in 2005. He obtained his Master Degree in R.M.K Engineering, Anna University in 2007. He also obtained outstanding master student at that year. He is obtained his Ph.D in 2014 at Faculty of Information and Communication Engineering, Anna University, India. His main research interest is in the area of Low Power VLSI, Testing of VLSI Circuits, FPGA Design, CAD for VLSI, Signal processing. He has published papers on these topics in various international journals.



Usha Rani A received her Bachelor Degree in Electronics and communication Engineering from Dr SGIET, Markapur, and JNTU University. And she obtained her Master Degree in Systems and signal processing from JNTUH college of Engineering, JNTU University and her main research interest is in the area of antennas and electromagnetic fields. She has published papers on these topics in various international journals.



M.Lakshmi Ravi teja working as an assistant professor in Institute of Aeronautical Engineering. He have pursued M.Tech in Sensor Systems Technology at VIT university Vellore and pursued his B.Tech in Electronics and Communication Engineering at Vignan's Lara Institute of Technology And Science.