Receiver Synchronization for Digital Audio Broadcasting system based on Phase Reference Symbol

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***Abstract*--The Eureka-147 Digital Audio Broadcasting (DAB) system is a new digital radio technology for broadcasting radio stations that provides high-quality audio and data services to both fixed and mobile receivers and employs coded orthogonal frequency division multiplexing (COFDM) technology. In this paper, we present the efficient synchronization method based on Phase reference symbol (PRS) for the DAB signal at the receiver. Since DAB uses OFDM digital transmission technique, the synchronization block plays an important role in determining bit error rate (BER) performance of DAB system in different channels. The result shows that the synchronization method proposed locates precisely each DAB frame even at low signal to noise ratio (SNR), so that the demodulation can be performed frame by frame, symbol by symbol.**

***Index Terms*—Eureka-147, DAB, OFDM, phase reference symbol, synchronization, transmission frame, BER, digital transmission, multipath fading, SNR,**

I. Introduction

The new digital radio system DAB (Digital Audio Broadcasting) was developed within the European Eureka-147 project [1], mainly to replace the existing AM and FM audio broadcast services in many parts of the world. It was developed in the 1990s by the Eureka 147/DAB project. The new concepts of digital broadcasting such as perceptual audio coding (MPEG-1/2), OFDM channel coding and modulation, the provision of a multiplex of several services and data transmission protocols makes DAB to provide high-quality digital audio services (mono, two-channel or multichannel stereophonic) along with programme-associated data and a multiplex of other data services (e.g. travel and traffic information, still and moving pictures, etc.) [2].Use of coded orthogonal frequency division multiplexing (COFDM) technology makes DAB system very well suited for mobile reception providing very high robustness against multipath reception. It also enables the system to operate in single frequency networks (SFNs) for high frequency efficiency.Synchronization is a very important in now-a-days digital communication systems.

All digital communication systems require proper synchronization for decoding of the received signal in order to produce the original information transmitted. In this paper we have described in detail the synchronization method based on phase reference symbol for the DAB signal that improves performance of DAB system in different transmission channels. The synchronization block is used to locate precisely each DAB frame, so that the demodulation can be performed frame by frame, symbol by symbol. In this paper we developed a DAB mode-II base-band transmission system based on Eureka-147 standard [1]. A frame based processing is used in this work.

Following this introduction the remaining part of the paper is organized as follows. Section II presents the DAB system standard. In Section III, the details of the modeling and simulation of the Synchronization method is presented based on phase reference symbol (PRS). Then, simulation results have been discussed in Section IV. Finally, Section V provides the conclusion.

II. SYSTEM SPECIFICATIONS

The working principle of the DAB system is illustrated in conceptual block diagram shown in Fig. 1. At the input of the system the analog signals such as audio and data services are MPEG layer-II encoded and then scrambled. In order to ensure proper energy dispersal in the transmitted signal, individual inputs of the energy dispersal scramblers is scrambled by modulo-2 addition with a pseudo-random binary sequence (PRBS), prior to convolutional coding [1]. The scrambled bit stream is then subjected to forward error correction (FEC) employing punctured convolutional codes with code rates in the range 0.25-0.88. The coded bit-stream is then time interleaved and multiplexed with other programs to form Main Service Channel (MSC) in the main service multiplexer. The output of the multiplexer is then combined with service information in the Fast Information Channel (FIC) to form the DAB frame. Then after QPSK mapping with frequency interleaving of each subcarriers in the frame, $\pi/4$ shifted differential QPSK modulation is performed. Then the output of FIC and MSC symbol generator along with the Phase Reference Symbol (PRS) which is a dedicated pilot symbol generated by block named synchronization symbol generator is passed to OFDM signal generator. This block is the heart of the DAB system. Finally, the addition of Null symbol to the OFDM signal completes the final DAB Frame structure for transmission.

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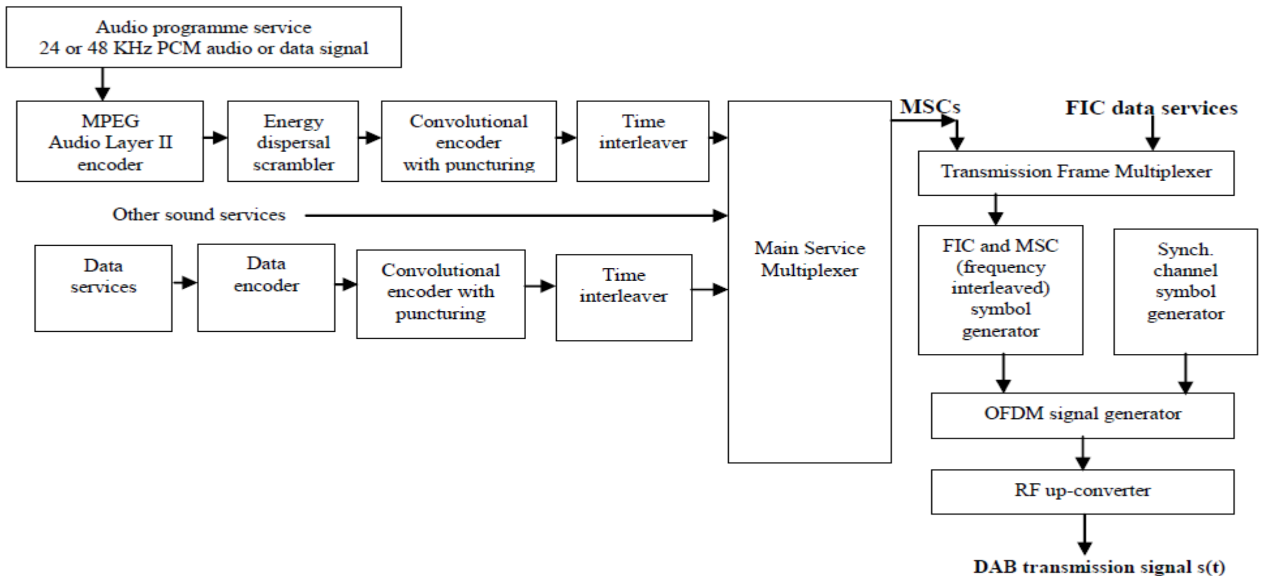

Figure 1. Complete DAB transmitter block diagram [1].

A. Channel coding, muliplexing and transmission frame

The channel coding is based on a convolutional code with constraint length 7. The octal forms of the generator polynomials are 133, 171, 145 and 133, respectively. The mother code has the code rate R =1/4, that is for each data bit a_i the encoder produces four coded bits $x_{0,i}$, $x_{1,i}$, $x_{2,i}$, and $x_{3,i}$. The individual programme (audio and data) are initially encoded, error protected by applying FEC and then time interleaved. These outputs are then combined together to form a single data stream ready for transmission. This process is called as Multiplexing. In DAB several programmes are multiplexed into a so-called ensemble with a bandwidth of 1.536 MHz.

The DAB signal frame has the structure shown in Fig. 2 that helps in efficient receiver synchronization. The period T_F of each DAB transmission frame is of 24 ms or an integer multiple of it.

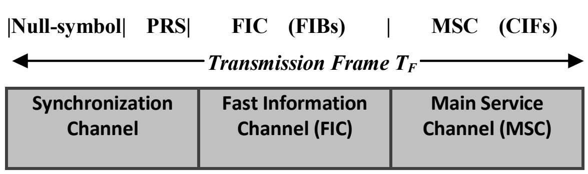

Figure 2. DAB transmission signal frame structure.

B. COFDM

DAB uses COFDM technology that makes it resistant to multipath fading effects and inters symbol interference (ISI). OFDM is derived from the fact that the high serial bit stream data is transmitted over large (parallel) number sub-carriers (obtained by dividing the available bandwidth), each of a different frequency and these carriers are orthogonal to each other. OFDM converts frequency selective fading channel into N flat fading channels, where N is the number of sub-carriers. Othogonality is maintained by keeping the carrier spacing multiple of 1/Ts by using Fourier transform methods, where Ts is the symbol duration. Since channel coding is applied prior to OFDM symbol generation which accounts for the term 'coded' in COFDM.

C. DAB Transmission modes

The Eureka 147 DAB [1] system has four transmission modes of operation named as mode-I, mode-II, mode-III, and mode-IV, each having its particular set of parameters as shown in Table-I.

TABLE I. SYSTEM PARAMETERS FOR THE FOUR DAB MODES

System Parameter	Mode -I	Mode -II	Mode -III	Mode -IV
No. of sub-carriers	1536	384	192	768
OFDM symbols/frame	76	76	153	76
Transmission frame duration	196608 T	49152T 24 ms	49152 T 24 ms	98304 T 48 ms
Null-symbol duration	2656 T 1297 ms	664 T 324 µs	345 T 168 µs	1328 T 648 µs
OFDM symbol duration	2552 T 1246 ms	638 T 312 µs	319 T 156 µs	1276 T 623 µs
Inverse of carrier spacing	2048 T 1 ms	512 T 250 µs	256 T 125 µs	1024 T 500 µs
Guard interval	504 T 246 µs	126 T 62 µs	63 T 31 µs	252 T 123 µs
Max. RF	375 MHz	1.5GHz	3 GHz	750MHz
Sub-carrier spacing	1 kHz	4 kHz	8 kHz	2 kHz
FFT length	2048	512	256	1024

The use of these transmission modes depends on the network configuration and operating frequencies. This makes the DAB system operate over a wide range of frequencies from 30 MHz to 3 GHz. Transmission mode –II is designed principally for Terrestrial DAB for small to medium coverage areas at frequencies below 1.5 GHz (UHF L-Band).

III. THE SIMULATION MODEL

Fig. 3 presents the complete block diagram of the DAB system which was modeled and simulated by us in MATLAB environment. The main objective of this simulation study is to explain the synchronization method based on phase reference symbol. The simulation parameters are obtained from Table I for transmission mode-II. A frame based processing is used in this simulation model. The system model was exposed to AWGN channel, Rayleigh fading channel and Rice channel to test the effectiveness of the synchronization block. The important blocks of the simulation model is discussed in detail as follows:

A. Phase reference symbol generator

According to DAB standard the first OFDM symbol (without taking account Null symbol) in the transmission frame is the phase reference symbol which helps in receiver synchronization. Since it occurs once in a frame therefore the detection of this symbol can be used for frame synchronization. It serves as reference for the differential modulation for the next OFDM symbols in the transmission frame. The phase reference symbol is defined [1] by the following expression:

$$z_{l,k} = \begin{cases} e^{j\varphi_k} & for -\frac{K}{2} \le k < 0 \; and \; 0 < k \le \frac{K}{2} \\ 0 & for \; k = 0 \end{cases} \quad (1)$$

Where

$$\varphi_{k=\frac{\pi}{2}}(h_{i,k-k'}+n) \quad (2)$$

The values of indices i, k' and the parameter n are given as functions of the carrier index k for all the DAB transmission modes. The values of the parameter $h_{i,j}$ is given as a function of its indices i and j [1].

B. Synchronization

The synchronization block is used to locate precisely each DAB frame, so that the demodulation can be performed frame by frame, symbol by symbol. In the DAB frame shown in Fig. 2 the first is the synchronization channel which consists of two symbols.

One is the Null symbol having duration TNULL. During null symbol period no information is transmitted. Null symbol gives coarse time synchronization. It gives the rough frame timing by envelope detection of the received signal i.e., detecting the null symbol by comparing average signal power during null symbol period TNULL with a set threshold. From the received signal, a data block of size 664 (equal to TNULL) samples is taken to measure the average signal power [4]. When this average signal power is less than half of the average transmitted signal power the null symbol has been detected, which indicates the start of the new frame. This method of frame synchronization based on null symbol detection is not suited for low SNR conditions because high noise power will provide incorrect frame timing estimate. Therefore phase reference symbol detection is ideally well suited for correct symbol timing and frame timing. Fig. 4 illustrates the process of receiver synchronization.

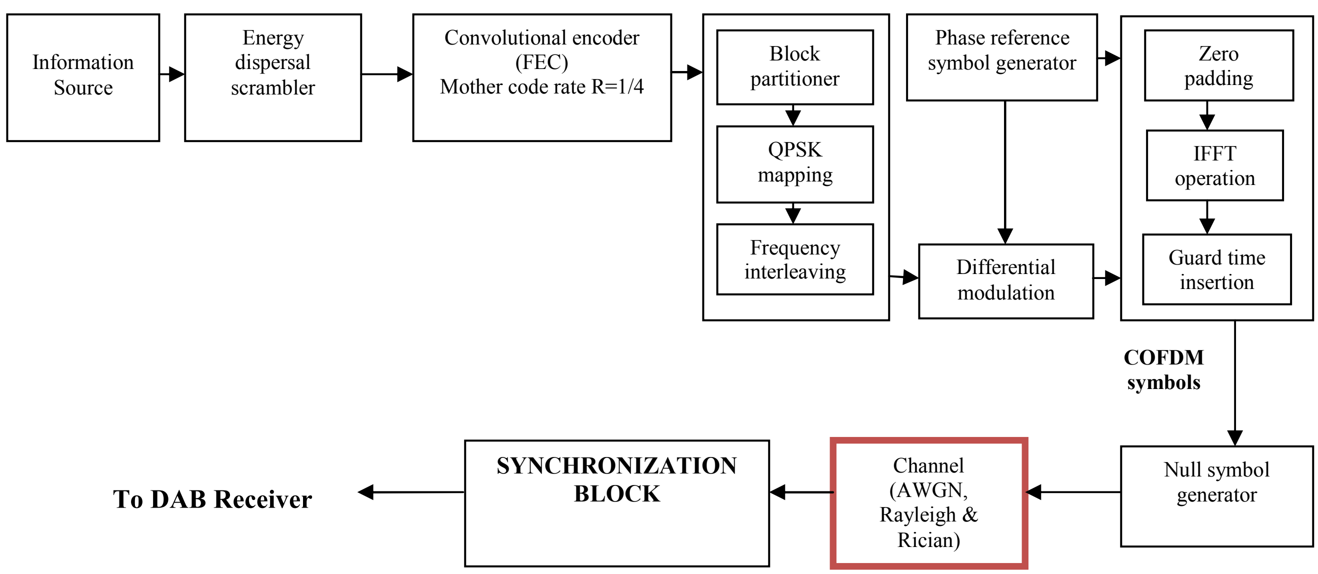

Figure 3. Block diagram of DAB system simulated.

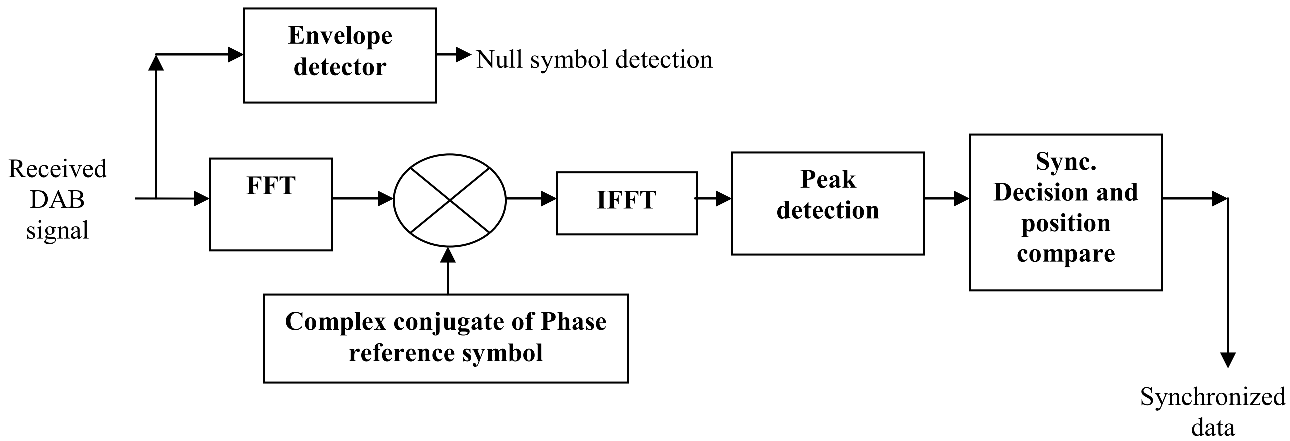

Figure 4. Block diagram of Symbol and Frame synchronization [13], [4].

Fine time synchronization or symbol timing synchronization [4] is performed by calculating the Channel Impulse Response (CIR) based on the actually received time frequency phase reference symbol (PRS) and the specified PRS stored in the receiver. To estimate the CIR, training Sequences (PRS in case of DAB system) are used. This means that a part (or the whole) of the transmitted signal is known from the receiver. As the receiver knows which signal is supposed to be observed, it can evaluate the distortion induced by the propagation channel and the modulation& demodulation stages.

Fine time synchronization is based on the phase reference symbol which is the dedicated pilot symbol in each DAB transmission frame. Since the modulation each carrier is known [12], multiplication of received PRS with complex conjugate of PRS at the receiver results in cancellation of the phase modulation of each carrier. The phase reference symbol can be converted to impulse signal or CIR can be obtained by an IFFT operation of the resultant product as illustrated in (3) below

$$\text{CIR} = \text{IFFT}\ \{\text{Received PRS} \bullet \text{PRS}^*\} \tag{3}$$

Where PRS* is the complex conjugate of the phase reference symbol. The peak of the impulse signal obtained from (3) will give position of the start of the PRS compared to a set threshold (T) providing symbol timing as well as frame timing. According to Fig. 4 from the received signal a data sample block of FFT length is taken. Then FFT operation is performed on the block to convert the samples into frequency domain. Since FFT window length is 512 and size of PRS at the receiver is 384 (mode-II) therefore zero padding removal and data rearrangement has to be done [4]. The resulting sample block is of size 384 same as PRS. Now sample block is multiplied by the complex conjugate of the PRS known at the receiver which is then transformed into impulse signal in time by performing IFFT operation on the product.

The highest peak detection will indicate the start position of the PRS. To get a precise synchronization decision the peak obtained from every sample block taken from the received signal is compared to set threshold level (T). When the detected peak is less than the threshold level, then the peak found is not the desired peak and does not indicate the accurate start of the PRS. So the loop process has to be continued by taking the next sample block till the desired peak is obtained. The peak will be greater than the threshold only for the sample block which has phase reference symbol in it, since PRS have a high correlation with itself.

IV. SIMULATION RESULTS AND DISCUSSION

In this section we have presented the simulation results along with analysis under worst signal to noise ratio (SNR) conditions for AWGN channel, Rayleigh fading channel and Rice channel. The results are shown for transmission mode-II and the simulation parameters are taken as per the DAB standard [1].

First of all the Threshold level will be determined by observing the magnitude of the highest peak obtained by multiplication of the PRS with its complex conjugate and IFFT applied to the product, both in presence and absence of noise. Fig. 5 presents the phase reference impulse symbol in presence and absence of noise. It is observed that the highest peak is obtained in the absence of noise therefore the threshold level was set to be greater than half the magnitude of this peak. This ensures that noise peak will not be mistaken as desired peak during peak detection. Threshold level was set to be T= 140.

Fig. 6 shows the successful detection of the desired peak in AWGN channel with SNR of 20dB. It may be easily evaluated form Fig. 6 that the highest peak is located at sample index 791. According to DAB standard the first symbol in the DAB frame is a Null symbol of size 664 zeros followed by a guard interval of 126 samples of PRS. Thus sum of null symbol and guard interval samples equals 790, therefore the peak is located exactly at the starting point of useful phase reference symbol. After successful verification of fine time synchronization, the DAB system will be tested for peak detection under worst SNR of -11dB. Fig. 7 presents the peak detection with SNR of -11 dB in AWGN channel.

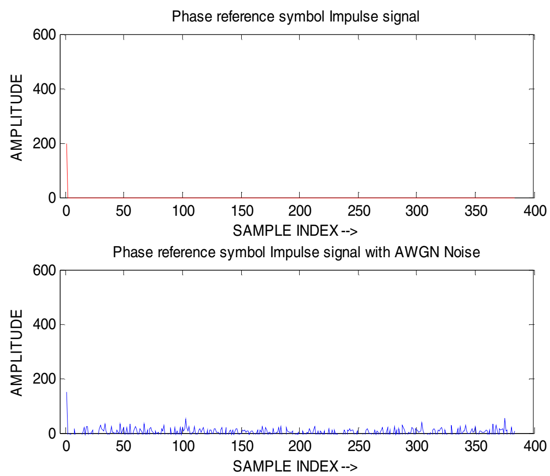

Figure 5. Threshold level determination.

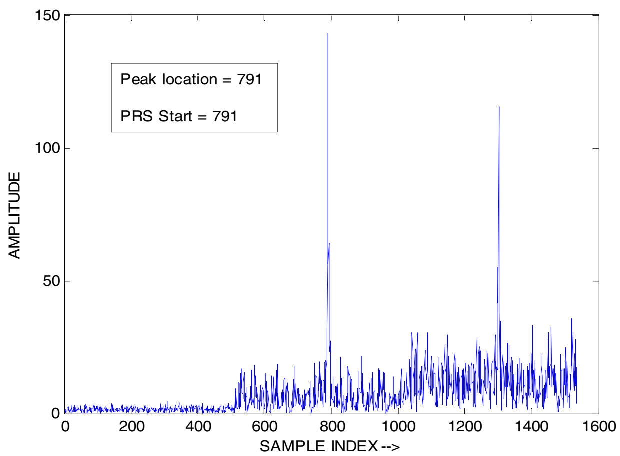

Figure 6. Desired peak detection in AWGN channel with 20 dB SNR.

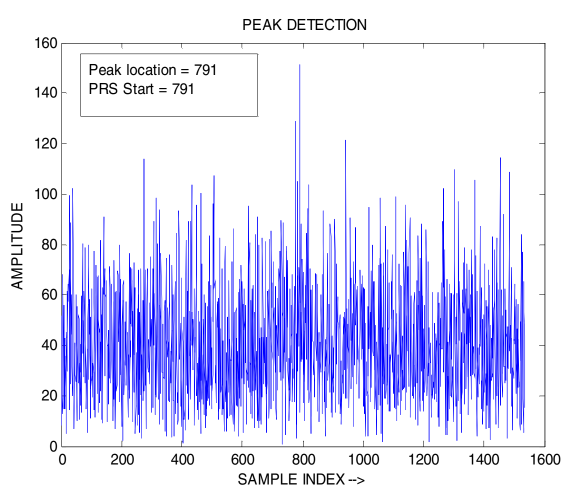

Figure 7. Desired peak detection in AWGN channel with -11 dB SNR.

From the Fig. 7 it can be evaluated that the highest peak is located at sample index 791 even at worst SNR at -11 dB providing correct fine time synchronization. The peak is located exactly at the starting point of useful phase reference symbol.

Fig. 8 and Fig. 9 presents the peak detection in Rayleigh fading channel with doppler shift of 40 Hz with SNR of 20 dB and -11 dB, respectively. It is observed that in both figures we have successful peak detection with correct position of PRS. From this point, the DAB frame and hence the ODFM symbols can now be demodulated to extract the original information.

After analyzing the peak detection in AWGN channel and Rayleigh channel, the fine time synchronization using PRS in Rician channel will be investigated next. Fig. 10 and Fig. 11 presents the peak detection in Rician fading channel with Doppler shift of 40 Hz with SNR of 20 dB and -11 dB, respectively. It is observed that in both figures we have successful peak detection.

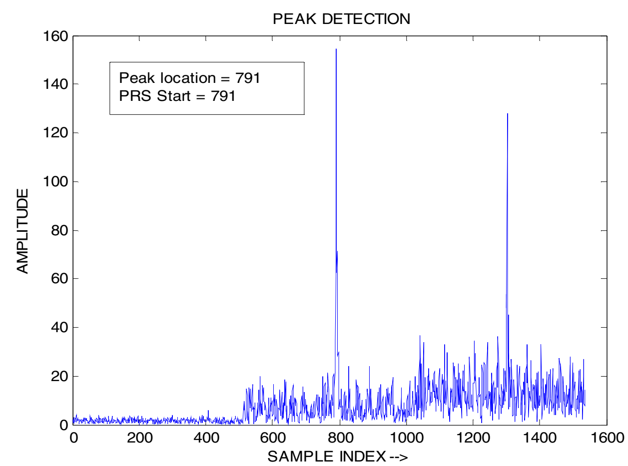

Figure 8. Desired peak detection in Rayleigh fading channel with 20 dB SNR.

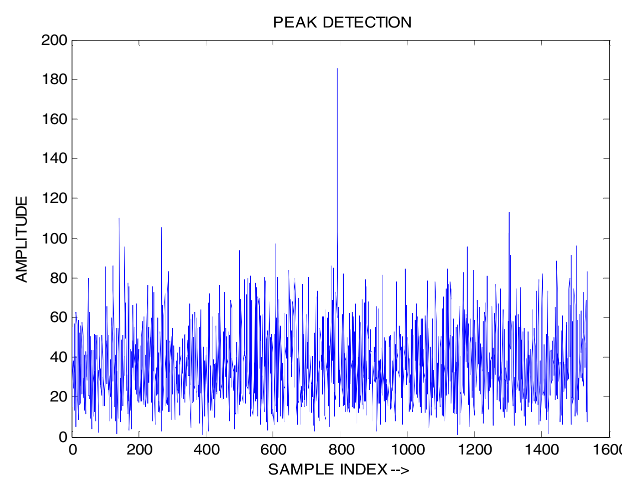

Figure 9. Desired peak detection in Rayleigh fading channel with -11 dB SNR.

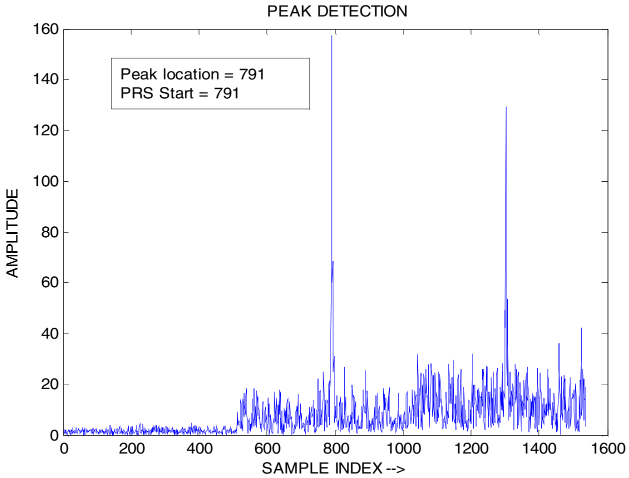

Figure 10. Desired peak detection in Rician channel with 20 dB SNR.

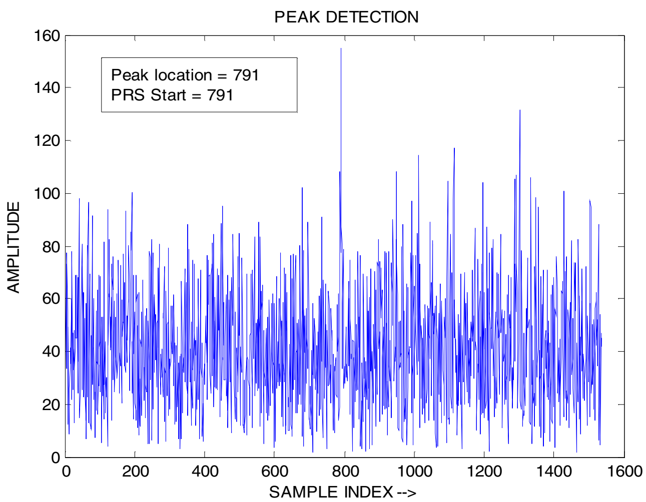

Figure 11. Desired peak detection in Rician channel with -11 dB SNR.

V. CONCLUSION

The proposed synchronization method of fine time synchronization using phase reference symbol provided correct frame synchronization even in the low SNR condition in all the three channels. It was observed that in all channel condition successful peak detection was obtained indicating exact position of PRS. From this position, the DAB frame and hence the ODFM symbols can be demodulated to extract the original information.

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