Carrier Time Division Multiplexing

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Research Article

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Abstract

Carrier Time Division Multiplexing is a new method that makes a specific bandwidth for a digital modulated signal to virtually appear as a single frequency, this can be approached by time divide the modulated carriers so that each modulated carrier can be independently extracted and demodulated, the application of this concept in a communication system can increase the transmission medium utilization.

I. Introduction

Virtual Bandwidth is a new concept where the bandwidth of a digital modulated signal is required to virtually appears as a single frequency or with a bandwidth that is much smaller than the original signal bandwidth, in this paper Carrier Time Division Multiplexing will be introduced where it achieves the virtual bandwidth concept, Simulation results and comparison with other techniques are illustrated.

I I. Carrier Time Division Multiplexing Overview

Carrier Time Division is a Multiplexing method for digital modulation techniques where a number of carriers are time divided so that for each carrier at least two equispaced samples can be extracted with in its period to comply with Nyquist rate and Discrete Fourier Transform (DFT) process, the sampling rate Fs must be Fs ≥ 2*Fmax where Fs is the Sampling rate (Samples/sec) and Fmax is the largest frequency component in the signal that is going to be sampled.

Carrier time division process is to free time periods (TF) for each symbol in each carrier so that no any other signals values exist in these periods except for the previous signals as shown in Fig. 1.

Figure 1 is an example of a carrier time division multiplexed signals (3 QAM modulated signals), the first carrier frequency is 100 KHz, the second 125 KHz and the third 150 KHz, the time period from T0 to the end of the symbol time contains symbol 1 for signal 1, from T1 to T4 contains symbol 1 for signal 2 and from T2 to T3 contains symbol 1 for signal 3, the total symbol time (Tt) in this example is [3 / (4*100 KHz)] = 7.5 usec where only ¾ of the full sinusoidal cycle is used because of the fact that DFT requires equispaced samples and carrier time division requires free time period to be placed at the beginning and end of each symbol, this leads to the difficulty of using samples spaced by π and its multiples which if used will result in some constrains in the DFT process, as an example, a sine wave with zero phase and equispaced samples spaced by π or its multiples will have zero value for all its samples which leads to false results for the DFT process at the receiver, so it is not possible to use a full cycle sinusoidal symbol.

The processes of demodulating and demultiplexing of the first symbol for the received signal occurs by sampling the received signal at the free time periods of the first symbol for the first signal then processing a Fast Fourier Transform (FFT) or a direct low pass filtering of the samples to demodulate, for demultiplexing an Inverse Fast Fourier Transform (IFFT) and a subtraction process from the total signal or a direct subtraction process when direct low pass filtering is used to demultiplex, the same processes are repeated for the demodulation and demultiplexing of each symbol.

The minimum required bandwidth of a digital QAM equals to 1/TS where Ts is the Symbol time, Fig. 2 (a) shows the bandwidth allocation for a normal QAM signal and Fig. 2 (b) shows the bandwidth of a carrier time division multiplexed QAM, Fig. 2 (c) shows the single frequencies (Virtual Bandwidth) of the carrier time division multiplexing QAM in the demodulation and demultiplexing processes, as shown in Fig. 2 huge bandwidth saving can be attained by well allocating the carriers frequencies and the free time periods.

A block diagram for a simple carrier time division multiplexing transmitter is shown in Fig. 3, the serial data stream is split into N-1 sub-streams and each stream is modulated using a digital modulation scheme, Inverse Fast Fourier Transform (IFFT) is used to transform the results to time domain, symbols in time domain are shifted and centred to produce a carrier time division multiplexed signal, then the total sampled signal is converted to a continuous one by using a digital to analog converter (DAC), an optional RF modulator can be used if the operating frequency of the system is so high.

A block diagram for a simple carrier time division multiplexing receiver is shown in Fig. 4, the samples are extracted from the received signal at the free time periods of each symbol, a Fast Fourier Transform (FFT) is used to transform the time domain signal to its frequency domain so it can be demodulated, at the same time an Inverse Fourier Transform (IFFT) is used to convert the same signal to its time domain so it can be subtracted from the total received signal, the same processes processed for extracting the other symbols.

I I I. Simulation Results

In this simulation, three 16-QAM modulated signals with 5 symbols for each will be transmitted at 100 KHz, 100.1 KHz and 100.2 KHz respectively with 134 KHz approximated bandwidth for each and 7.5 uSec symbol time duration, Additive White Gaussian Noise (AWGN) will be added to the signals to simulate a communication channel.

In Fig. 5a constellation of 16-QAM with Min. Distance = 2 is shown to modulate the transmitted digital bits of signal 1, 2 and 3 as shown in Fig. 7, a theoretical bandwidth is shown in Fig. 6 (a), the bandwidth equals to 1/Ts where Ts is the active symbol time, Fig. 6 (b) shows the time domain of the signal.

Figure 8 showing the modulated 16-QAM signals in time domain with an AWGN noise added to the summation of the three signals, the free periods are not visible due to its short time periods compared with the signal time periods.

Figure 9 is the simulated frequency bandwidth for the sum of the signals using FFT zero padding and it is approximately equals to the theoretical bandwidth shown in Fig. 6 (a).

Figure 10 and Fig. 11 are the simulation results for different cases, FFT processes requires equispaced samples of the sampled signal, when a 1.5π spaced samples are used, two more samples are required to perform equispaced samples FFT process, many ways can be used to get those two more samples, by using sinusoidal waves properties, sin (t) = -sin (t-π) and sin (t) = sin (t-2π), this means it is possible to get another additional sample with in the time period T from knowing only one sample, this process is known as sample copy, the second method is to shift the second sample backward and make 4 point zero padding FFT.

Each of those two methods have a different Symbol Error Rate (SER) for a specific Signal To Noise Ratio (SNR), as shown in Fig. 10 where the Symbol Error Rate % (SER) percentage is shown for the case of 2 Samples (1 Shifted Backward) with FFT Padding to 4 points with different Signal To Noise Ratios (SNR) and Fig. 11 showing the Symbol Error Rate % (SER) for the case of 4 Samples (2 real + 2 copied).

Figure 12 is an example of a 16-QAM Symbol Error Rate % (SER) Percentage for 16-QAM BASEBAND with different Signal to Noise Ratio (SNR) for comparison.

I V. Conclusions

Carrier Time division virtual bandwidth is more noise sensitive compared to other methods due to accumulated noise in each subtraction process but on the other hand it has great bandwidth savings compared with other techniques, as shown in the simulation results, three signals are send, frequency spaced by 100 Hz and there spectrum are totally interfered by each other, the three signals are successfully demultiplexed and demodulated.

The only drawback of carrier time division virtual bandwidth method is its high sensitivity to noise, but with the aid of using non-uniform discrete Fourier transform (N-DFT), the high sensitivity to noise can be lowered.

The application of non-uniform discrete Fourier transform is left for a future study.

Declarations

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Data availability statements (DAS):

The manuscript does not contain human or animal studies

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Figures

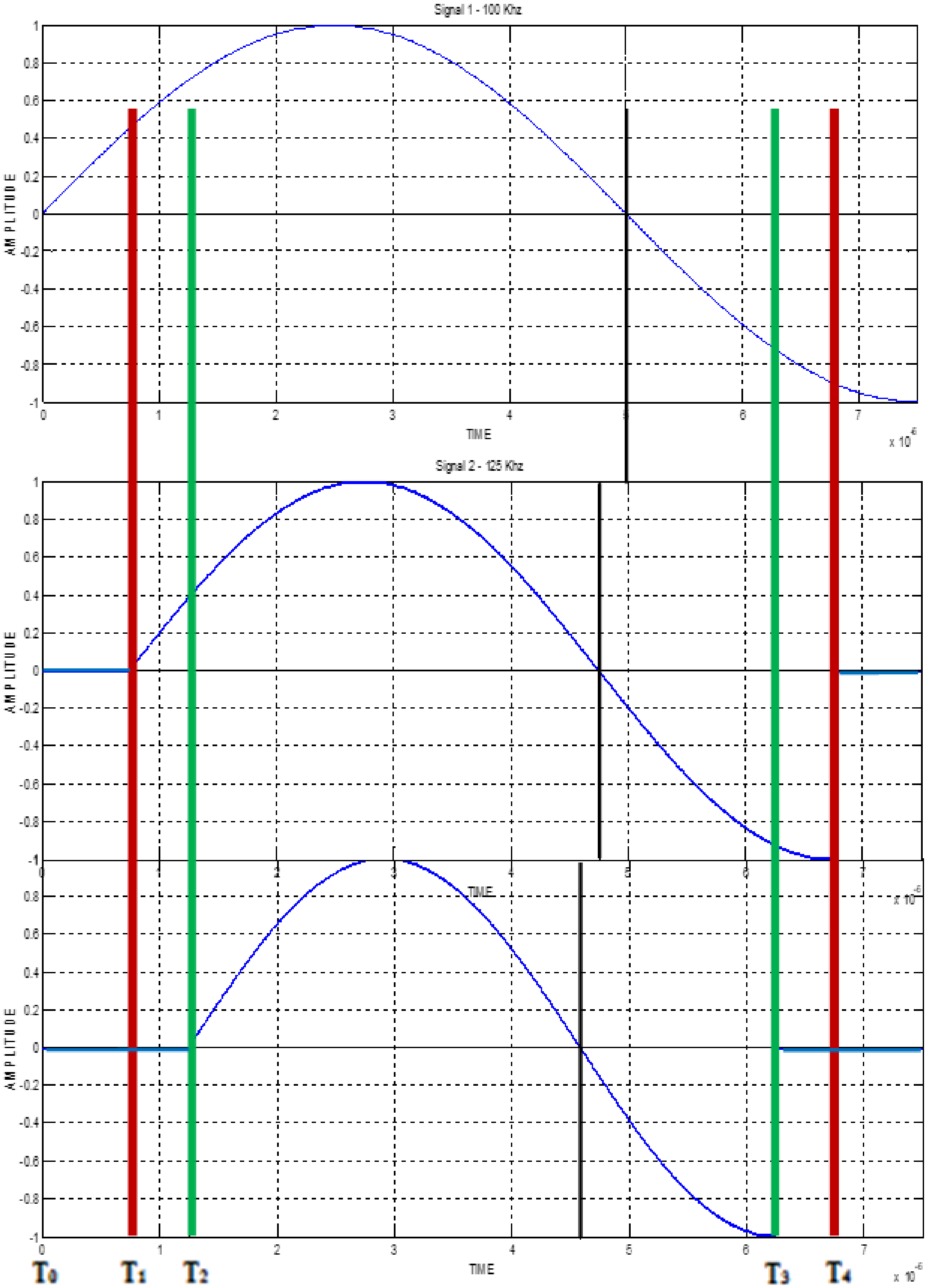


Figure 1

The construction of carrier time division multiplexed signals

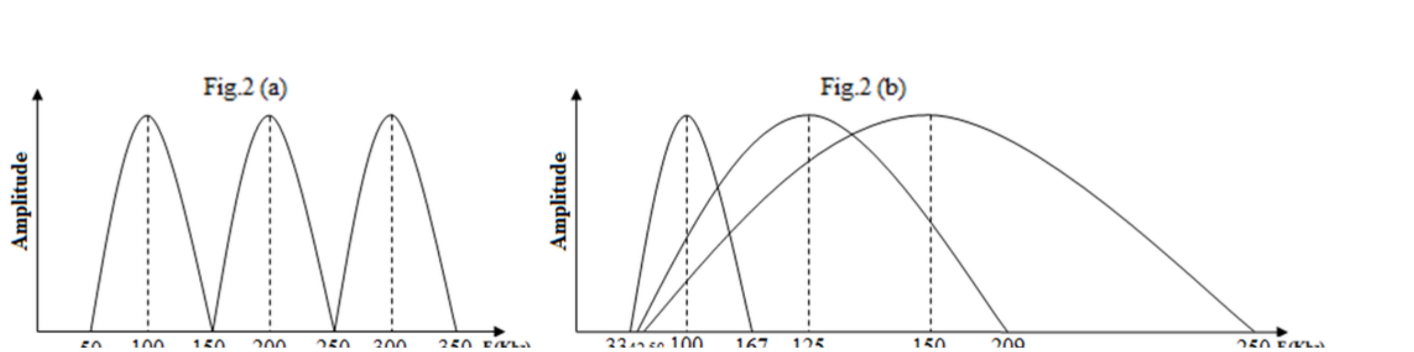


Figure 2

(a) Normal QAM Modulation Spectral Allocation and Bandwidth

(b) Carrier Time Division Multiplexed Signals Bandwidth

(c) Carrier Time Division Demultiplexed Signals Virtual Bandwidth (Single Frequencies)

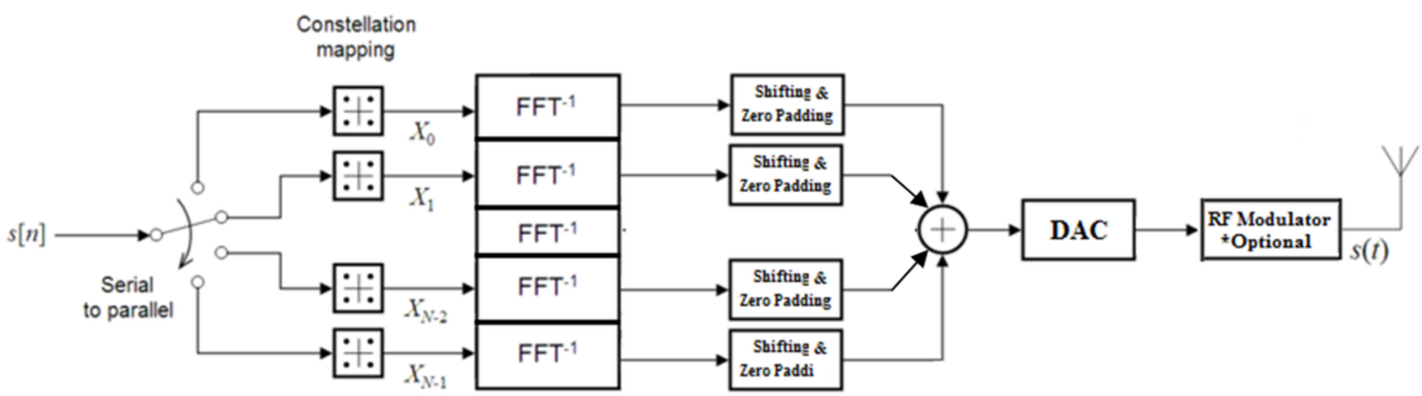


Figure 3

Block diagram for a Carrier Time Division Multiplexing (CTDM) Transmitter

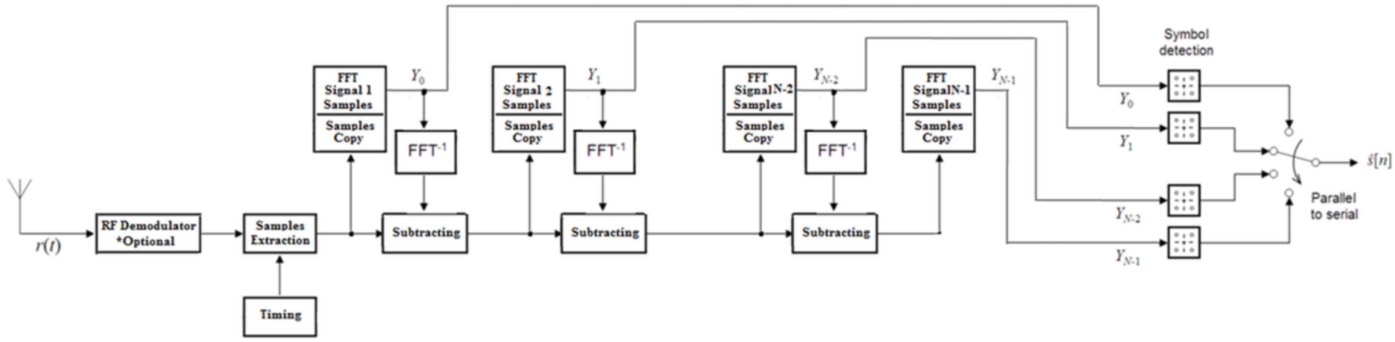


Figure 4

Block diagram for a Carrier Time Division Multiplexing (CTDM) Receiver

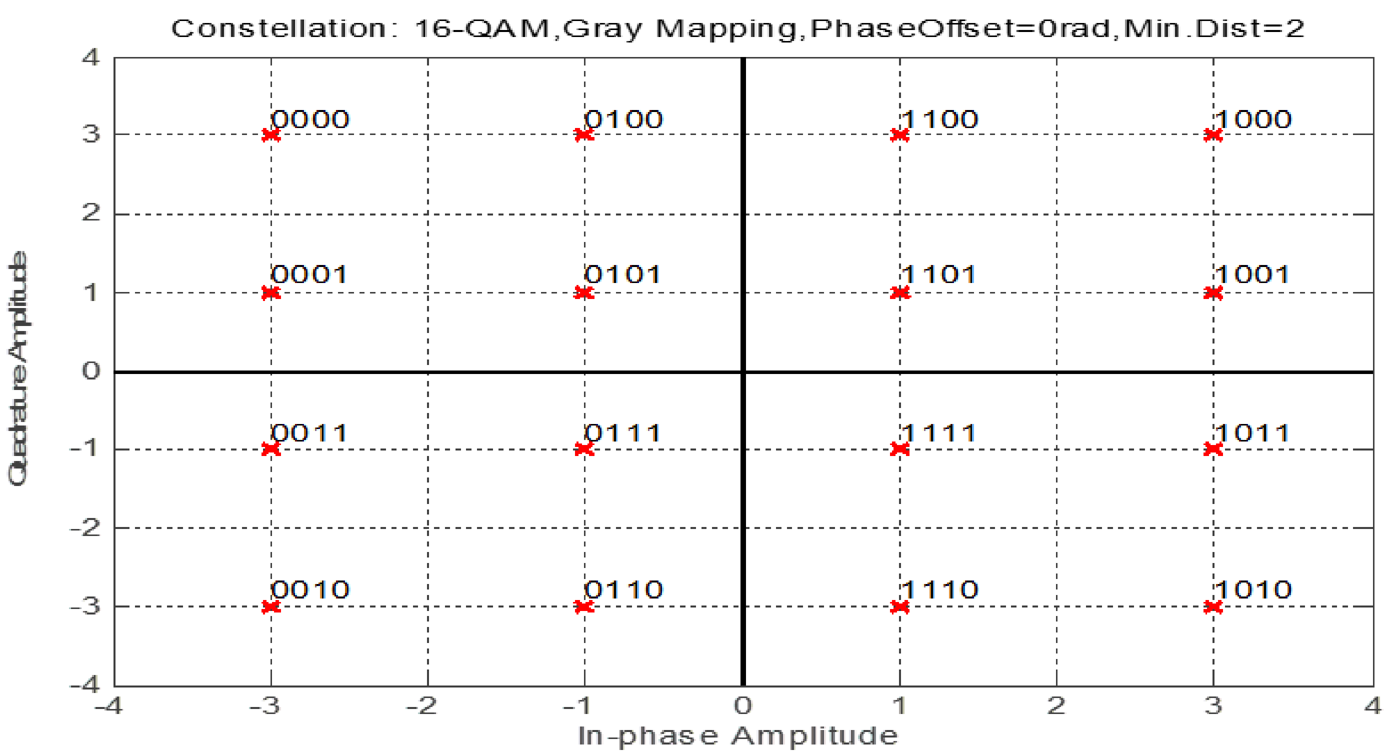


Figure 5

Constellation of 16-QAM with Min. Distance=2

Figure 6

(a) Signal Bandwidth allocation (Frequency Domain)

(b) Time Domain

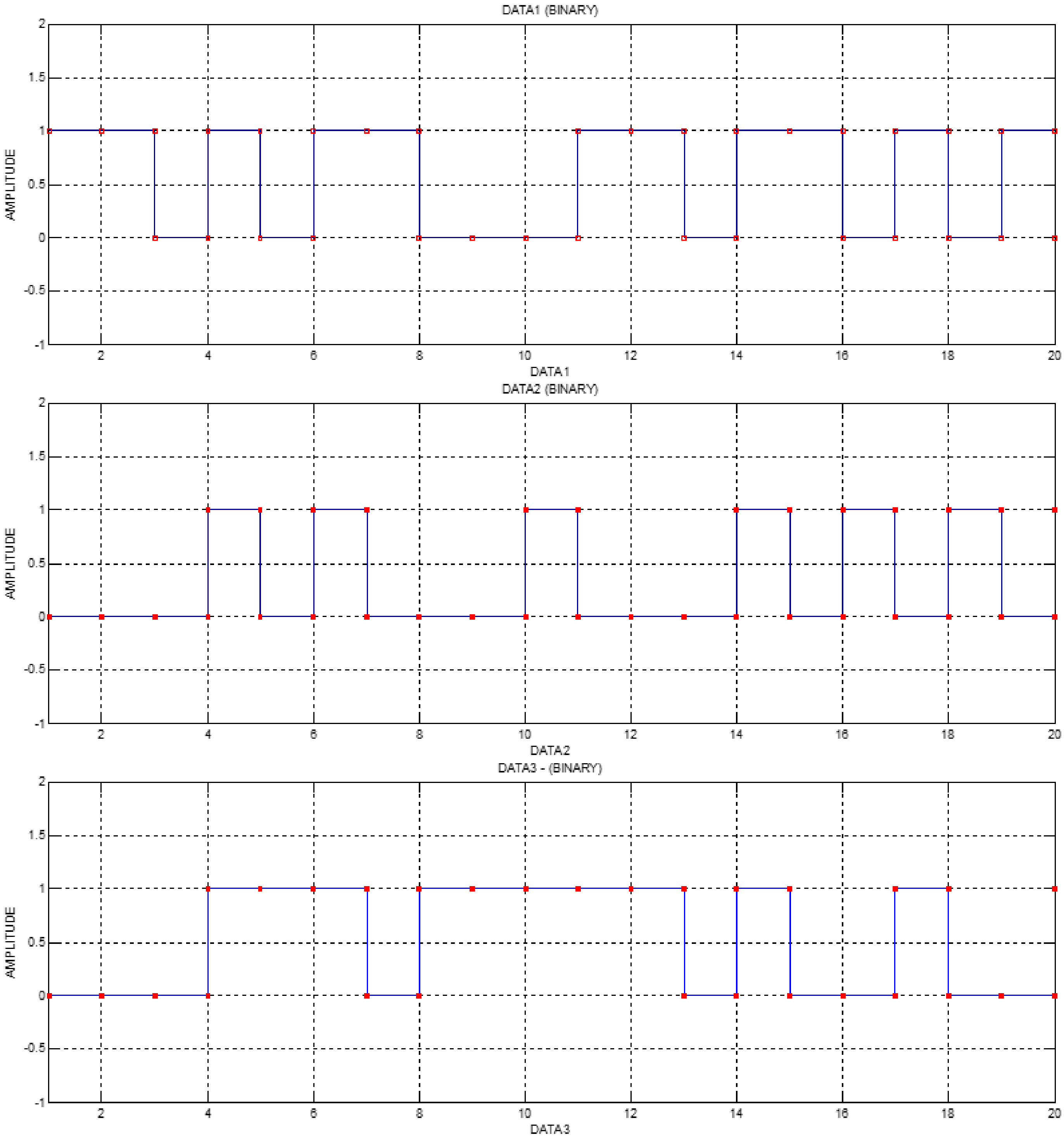


Figure 7

transmitted digital bits for Signal 1, 2 and 3

Figure 8

16-QAM modulated Signals 1, 2 and 3

& summation with AWGN noise

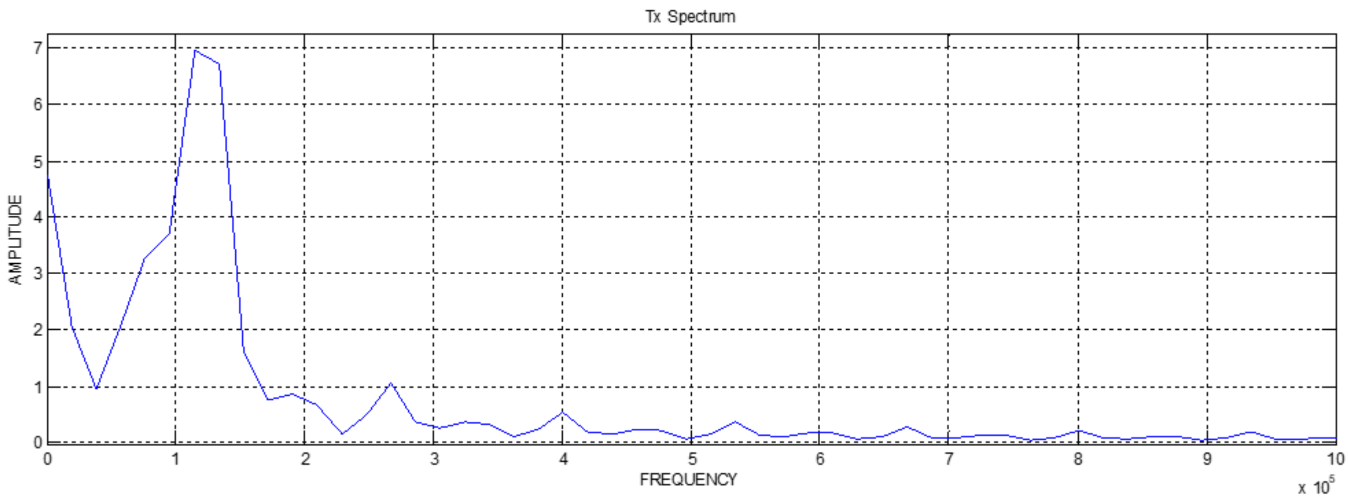
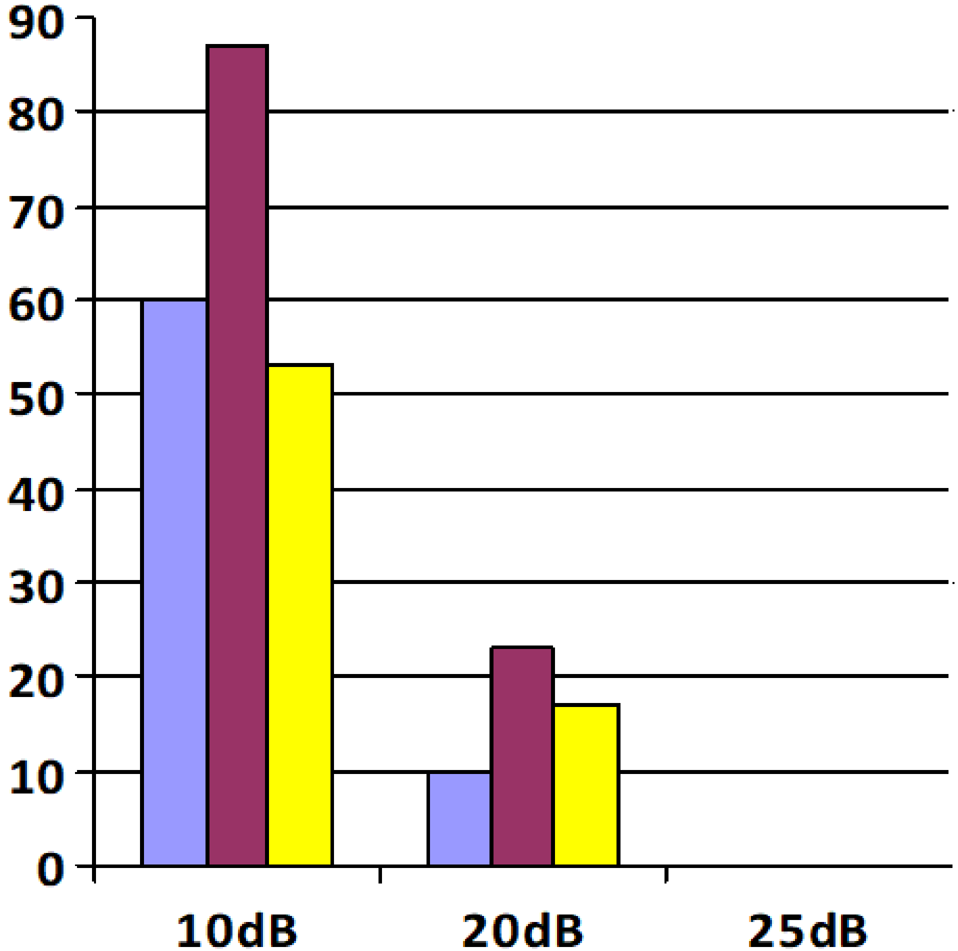


Figure 9

Frequency Spectrum for the total sum of the signals without the added AWGN noise, using FFT Zero padding

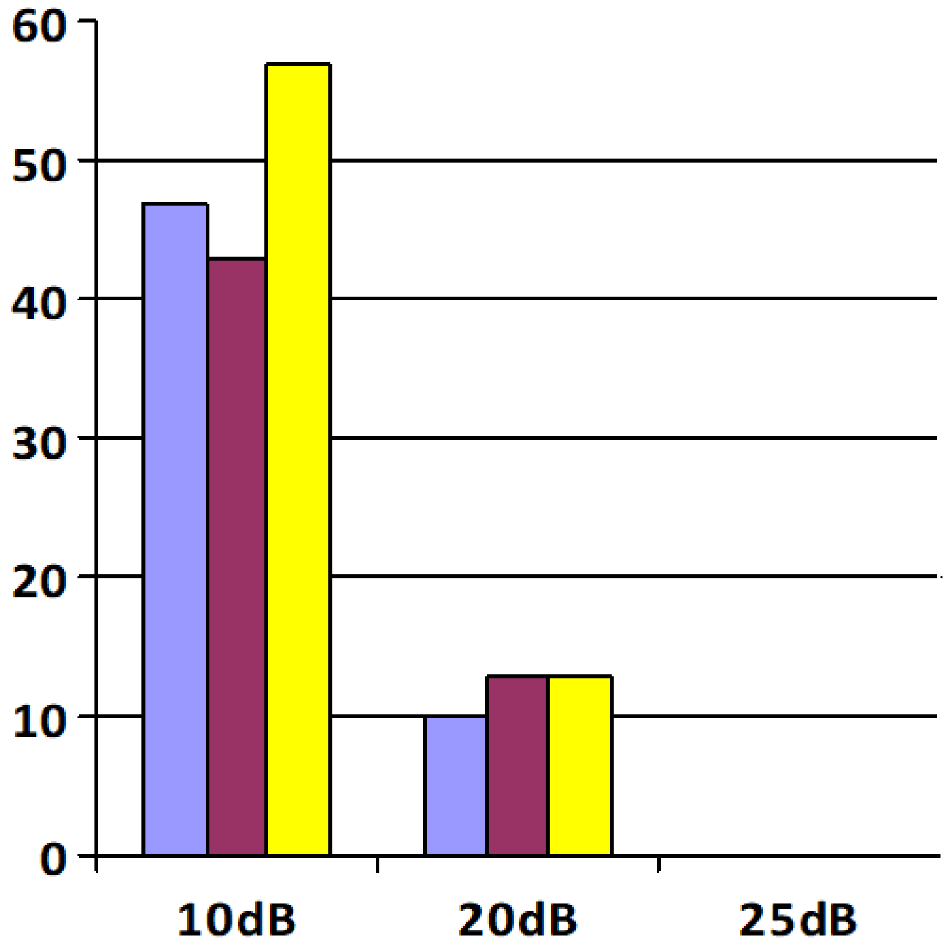


	10dB	20dB	25dB
Signal 1	60	10	0
Signal 2	87	23	0
Signal 3	53	17	0

Signal 1 Signal 2 Signal 3

Figure 10

Symbol Error Rate % (SER) for the case of 2 Samples (1 Shifted Backward) with FFT Padding to 4 points, SNR = 10dB, 20dB and 25dB

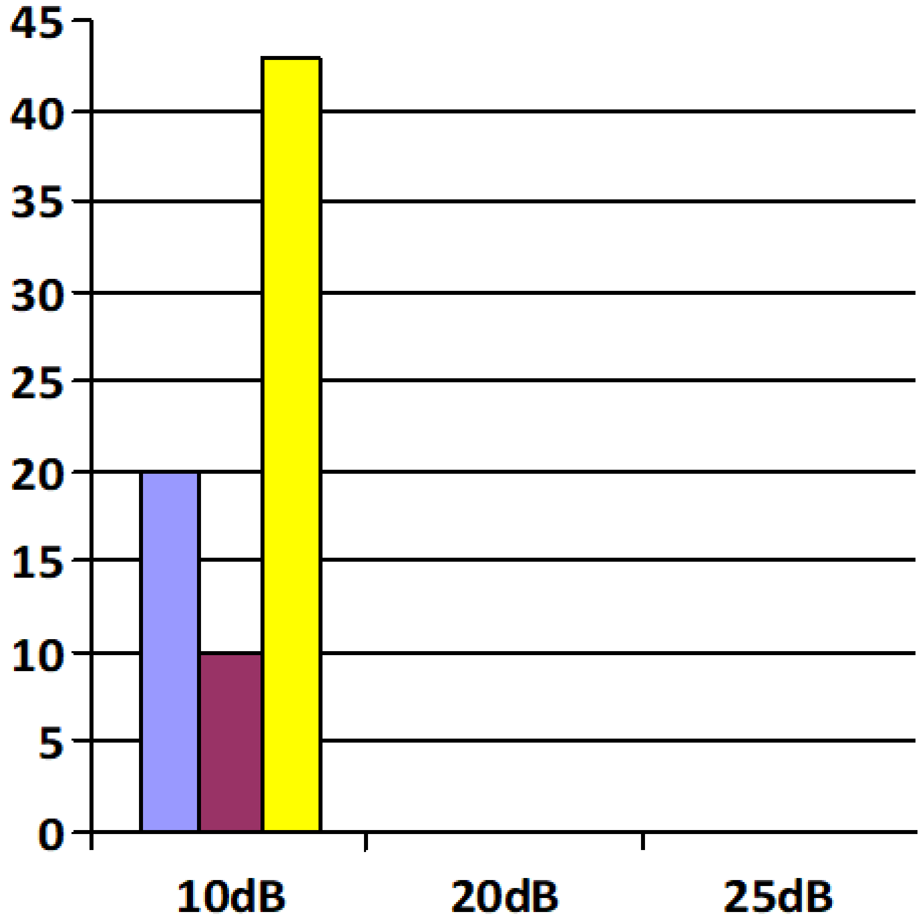


	10dB	20dB	25dB
Signal 1	47	10	0
Signal 2	43	13	0
Signal 3	57	13	0

Signal 1 Signal 2 Signal 3

Figure 11

Symbol Error Rate % (SER) for the case of 4 Samples (2 real + 2 copied), SNR = 10dB, 20dB and 25dB



	10dB	20dB	25dB
Signal 1	20	0	0
Signal 2	10	0	0
Signal 3	43	0	0

Signal 1 Signal 2 Signal 3

Figure 12

Symbol Error Rate % (SER) for 16-QAM BASEBAND, SNR = 10dB, 20dB and 25dB