

A REPORT  
ON  
**ENHANCED SPREAD SPECTRUM ACCESS TECHNOLOGIES FOR IOT AND M2M  
COMMUNICATION**

BY  
**Jay Kamat (2018A8PS0409P)**

AT  
**ISRO Space Applications Centre, Ahmedabad.**

A Practice School-I Station of  
**BIRLA INSTITUTE OF TECHNOLOGY AND SCIENCE, PILANI**  
**(June, 2020)**

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**Prepared in the partial fulfillment of the  
Practice School-1 Course No.s  
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## **ACKNOWLEDGEMENTS**

I am using this opportunity to thank PSD for allotting me this station. I have had a great learning experience so far. All theory that I keep reading in the books has translated to real life simulations also, giving me a clearer understanding of Communication Systems

I would like to thank Mr. Chandra Prakash of ISRO SAC Ahmedabad for his timely guidance and care. He has always been very approachable and helpful whenever I have had any difficulties in the project.

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## **ABSTRACT**

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Communication

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Key Words: Fourier Transform, Signal-to-Noise Ratio, Noise, Multiplexing, Modulation, Demodulation, Carrier Wave.

### Abstract

This aim of this project is to understand and simulate the working of a Burst Modulator-demodulator system. It involves understanding several types of digital modulation schemes, multiplexing schemes, and simulating the modulator block as well as carrier acquisition module block in MATLAB. Several methods to improve the estimation accuracy of this module have also been implemented.

Signature of Student

Signature of PS Faculty

Date: 25/06/2020

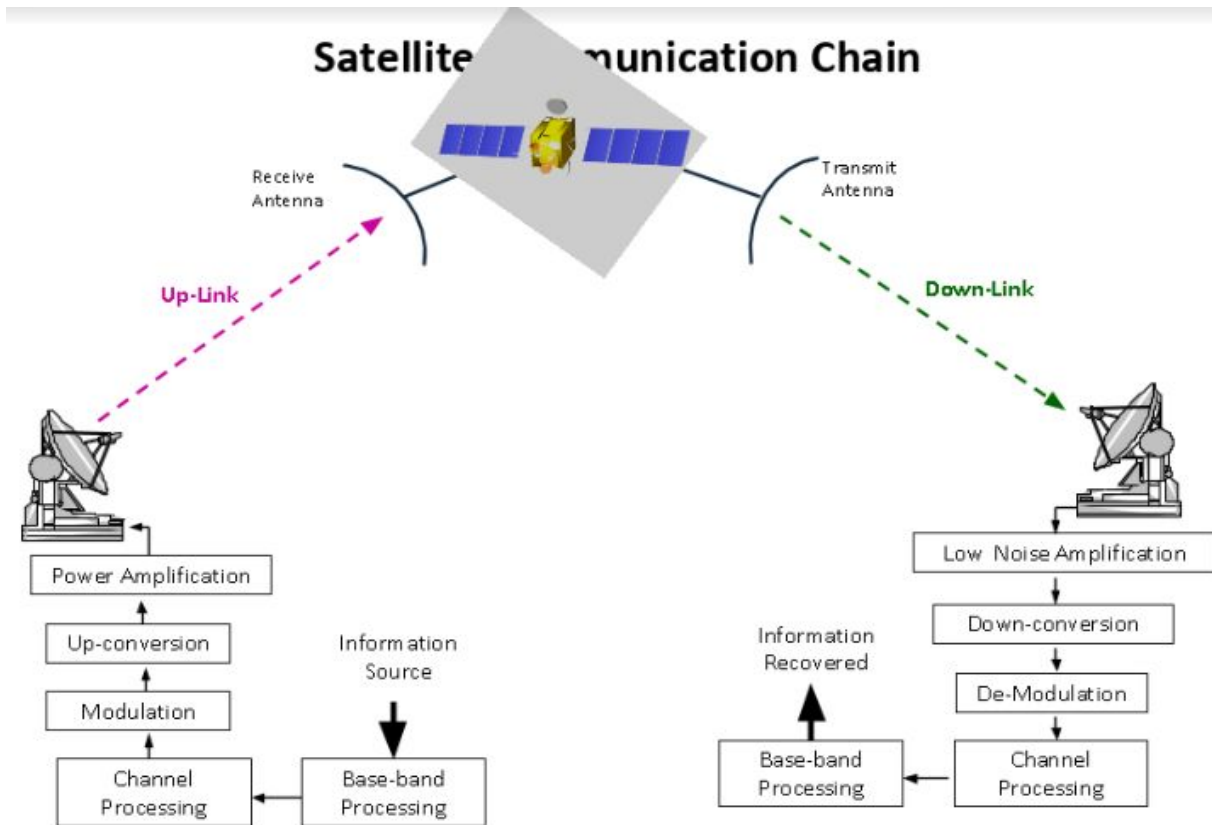
Date: 25/06/2020

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## INTRODUCTION

The process of satellite communication involves many stages as shown in the diagram below



This entire process involves transmission and reception of digital signals from one ground station to another. The purpose of each of the blocks is given below:

- **Baseband Processing block**

During Transmission: This block takes as input the low frequency digital information signal, and encodes it with a code signal, so that the message can be decoded only by the target user upon reception.

During Reception : This block first corrects the received signal for frequency shifts caused during reception, owing to doppler effect. After this, the block tries to decode the received signal by locally generating a replica of the code signal, and constantly auto-correlating it with the input signal till a maxima is encountered. A maxima shows us that the code signal generated locally is correct and in sync with the original code signal. Now we can easily decode this signal by performing XOR operation on it with the locally generated code signal. It performs all these operations iteratively using a control system (PLL and DLL tracking loops).

- *Channel Processing block*

During transmission: It assigns specific frequency channels to the incoming signals from the Baseband Processing block, according to the satellites to which the information is to be uplinked. Now the signal is forwarded to the modulation block for modulation.

During reception: It segregates the incoming signals into different channels, based on the frequency bands allotted to them. Now the signal is sent for Baseband Processing.

- *Modulation and Demodulation Block*

Modulation: The digital signal is converted from digital to analog by many modulation schemes like ASK, FSK, PSK. It now shifts from [Low frequency to Intermediate frequency](#).

Demodulation: It converts the received analog signal to digital domain. I will explain steps for demodulation in detail later in this document (through (i) Costas Loop or (ii) Squaring Loop). Signal shifts from [Intermediate frequency to Low frequency](#).

- *Up-Conversion and Down-Conversion block*

Up-conversion: The modulated analog signal is shifted to a higher frequency, by using an RF Mixer. It contains a local oscillator, which produces a high frequency sinusoid, which is multiplied to the modulated signal in time domain. By doing that, we shift this signal to [Radio Frequencies](#). Now such signals can be received by quarter wavelength antennas of realistic small sizes, like those present on the satellite.

Down conversion: The RF level signal is again multiplied by the local oscillator sinusoid. It is now passed through a low pass filter to retrieve the modulated signal sent by the satellite. It gets converted to [Intermediate frequency](#)

- *Power Amplification and Low Noise Amplification*

Low Noise Amplifier: The signal received by the ground station antenna has SNR value nearly equal to 1. In this case, we need to use an LNA in order to amplify the signal, while keeping noise nearly the same as before. Characteristics of LNA include high gain, low noise figure, and matched input impedance with the antenna

Power Amplifier: The RF signal required to be sent to the satellite must have high power so that it can survive atmospheric attenuation while reaching the satellite. Hence, a power amplifier is used.

Our project involved understanding and simulating the Modulation-Demodulation block of this chain. More specifically, we simulated Modulation, and the Carrier Acquisition Module of the Demodulator. The main aim was to increase the performance of the

Carrier Acquisition Module. Let us understand more about Modulation Schemes to get a better understanding of the project.

## Modulation

Modulation is the process of superimposing the message on a carrier wave. The carrier wave always has a high frequency, preferably in the Radio Frequency range.

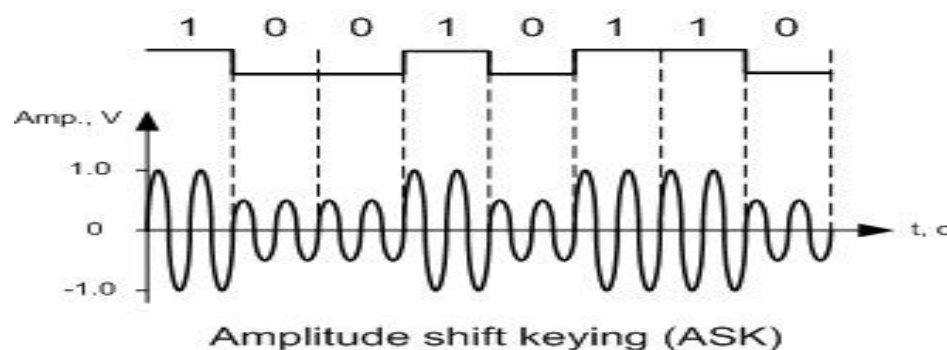
Reason:

- 1) The size of antenna required for reception of a signal is inversely proportional to the frequency of the signal. If we were to send our low frequency information signal as it is, it would require very long and inconvenient antennas for receiving them. Hence it is beneficial to use RF Carriers, so that antennas can become portable.
- 2) Another reason for modulating a signal is Multiplexing. Multiplexing involves utilizing our communication channel effectively to send information signals to multiple users at a given time, without allowing their respective information signals to interfere with each other. One particular method of doing this is called Frequency Multiplexing, where each user is allotted different RF Frequency carriers, so that information doesn't get mixed up between users.

Modulation is of two types - Analog Modulation and Digital Modulation. Analog modulation consists of Amplitude Modulation, Frequency Modulation, and Phase Modulation. Whereas Digital Modulation consists of Amplitude Shift Keying, Frequency shift keying, and Phase shift keying. The types of Digital Modulation are explained below.

- 1) Amplitude Shift Keying:

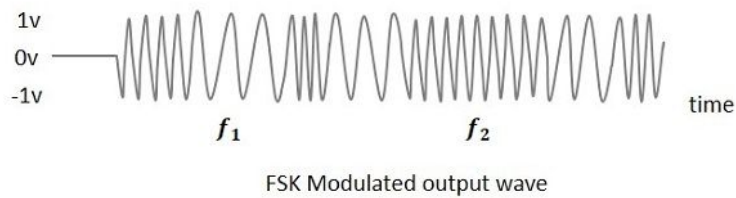
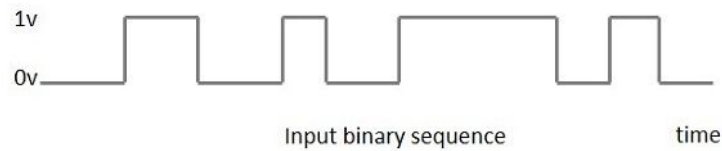
It involves changing the amplitude of the carrier wave, depending on whether there is a '1' or a '0' occurring in the bitstream, as shown below.



- 2) Frequency Shift Keying:

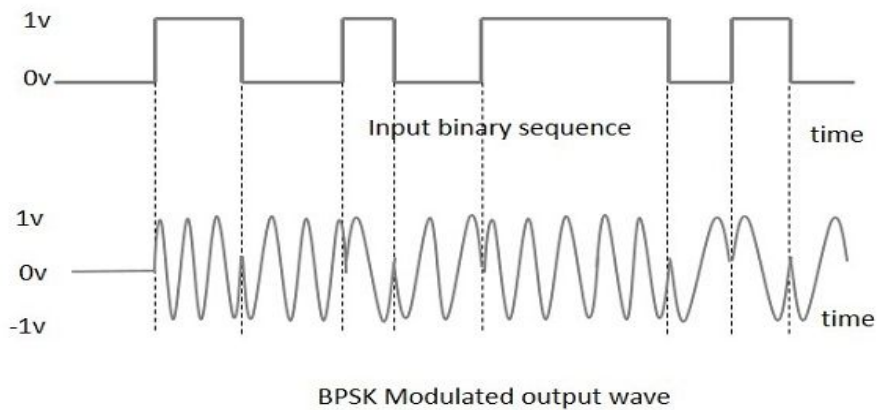
It involves changing the frequency of the carrier wave, depending on whether there is a '1' or a '0' occurring in the bitstream, as shown below.



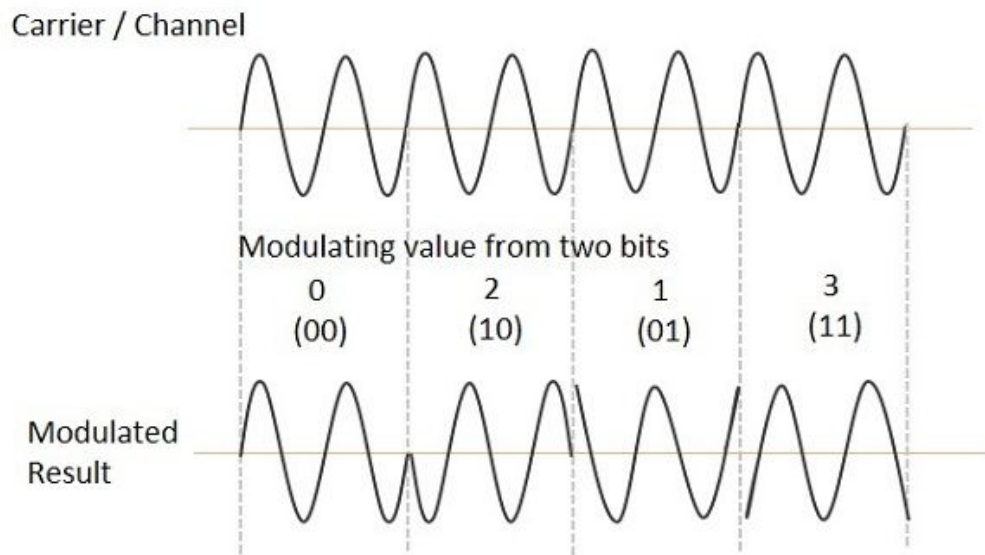


### 3) Phase Shift Keying:

It involves changing the phase of the carrier wave, depending on whether there is a '1' or a '0' occurring in the bitstream, as shown below.



The above figure shows Binary Phase Shift Keying, where one symbol corresponds to one bit. We could also have chosen one symbol as two bits. Therefore, the combinations would have been (0,0) (0,1) (1,0) and (1,1). Hence there will be 4 phases of carrier signal corresponding to each symbol. This is called as Quadrature Phase Shift Keying (QPSK)



Out of all these Digital Modulation Schemes, BPSK and QPSK were chosen for our project as they are the most energy efficient digital modulation schemes. They also tend to have the lowest Bit Error Rate.

## MULTIPLEXING

Once we have decided upon the kind of modulation scheme, we should now focus on how the same communication channel must be utilized to communicate with multiple users. Care should be taken that the messages do not get mixed up between the users, so that effective communication can take place. This process is called Multiplexing. There are three different ways in which we can multiplex our signals.

### 1) Time Division Multiple Access (TDMA)

In this scheme, data is sent as a short burst of signal from time to time. Each user knows the approximate time at which he/she would receive the message. The channel remains off for the remaining time. This ensures power savings. However the main issue is that the clocks at both transmitter and receiver end must be synchronous.

### 2) Frequency Division Multiple Access (FDMA)

In this particular scheme, each user is allotted a band of frequency, within which they can receive their messages. In this way, messages between users are not mixed up

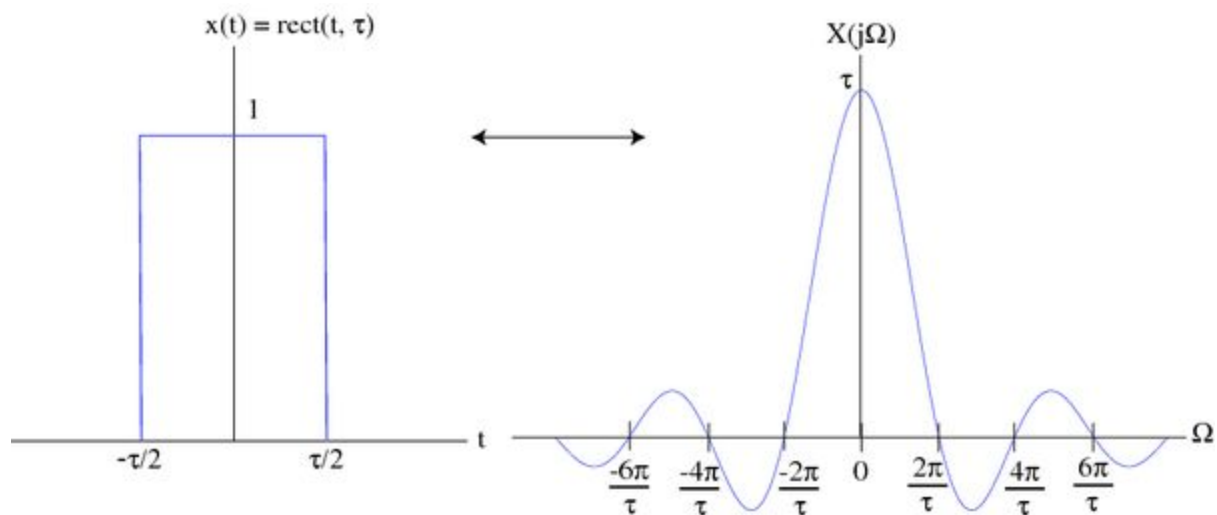
### 3) Code Division Multiple Access (CDMA)

In this scheme, the sent message is encoded, and only the correct receiver can decode the message. Hence although everyone receives the message, only the target user can actually use the message.

We are focusing on the TDMA Multiplexing scheme because it is the most commonly used Multiplexing scheme, and has no bandwidth limitation in terms of frequency. It is also power efficient.

### PULSE SHAPING

A sharp rectangular pulse in the time domain results in a sinc wave in the frequency domain, which spreads across the spectrum infinitely, as shown below



However in real life, there are limitations regarding the available bandwidth. If the spread in frequency domain exceeds the available bandwidth, signal energy is dissipated in other bands, which reduces the power of the signal. Hence in order to avoid such consequences, we pass the plain binary signal through a Pulse Shaping filter, through a Root Raised Cosine filter in particular, so that the signal is contained within the available bandwidth without loss of information. This helps us minimise Inter-Symbol interference if we appropriately position the lobes of our signal by adjusting sampling frequency.

## MAIN TEXT

In this part, I shall explain several parts of my code along with snippets.

The code initially asks for several kinds of inputs. Let us consider a particular case for viewing the working and results clearly. Values are purposely taken small so that the graphs are clearly visible.

Enter 1 for BPSK, or 2 for QPSK:

1

Enter Bit Rate:

100

Enter tolerable acquisition error:

2

Centre Frequency of carrier must be lesser than 25598

Enter Centre Frequency of Carrier:

40

Enter duration of signal:

1

### 1) Signal Generator

In this section, I first generated a random combination of 0's and 1's. Now this bitstream is converted to a continuous time signal and sent further for pulse shaping.

```
N = duration * bit_rate; %Number of bits to be generated
n = randi([0 1], 1, N);
```

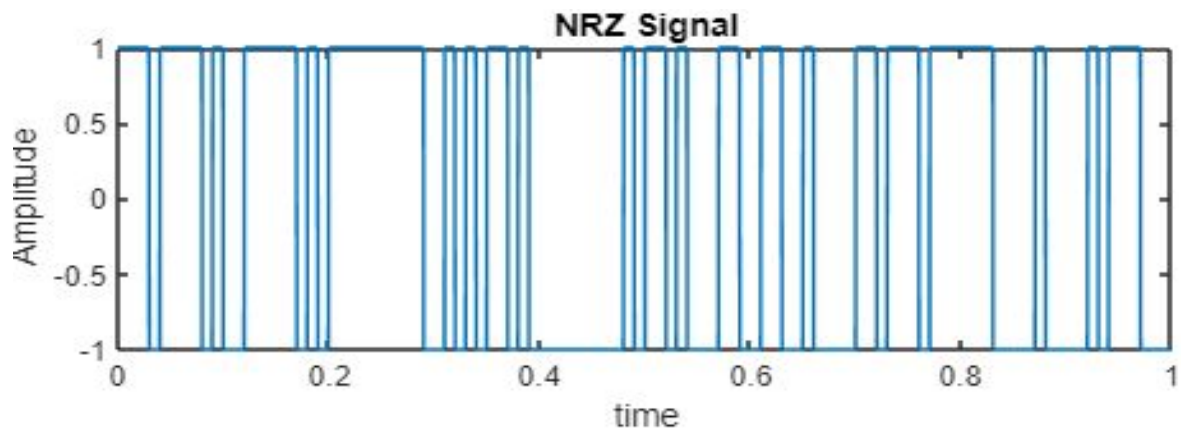
```
%*****Mapping of bits in NRZ form*****
```

```
for index = 1:N
    if n(index) == 0
        nn(index) = -1
    else
        nn(index) = 1
    end
end
```

```

*****Conversion to Continuous time BPSK NRZ signal*****
fsb = fs/bit_rate ; %Samples per bit
i = 1; %Index of input bits
st = 0 : 1/fsb : N-1/fsb; %Time
for j = 1 : length(st) %Index of time array
    if st(j) <= i
        signal(j) = nn(i);
    else
        signal(j) = nn(i);
        i = i + 1;
    end
end
end

```



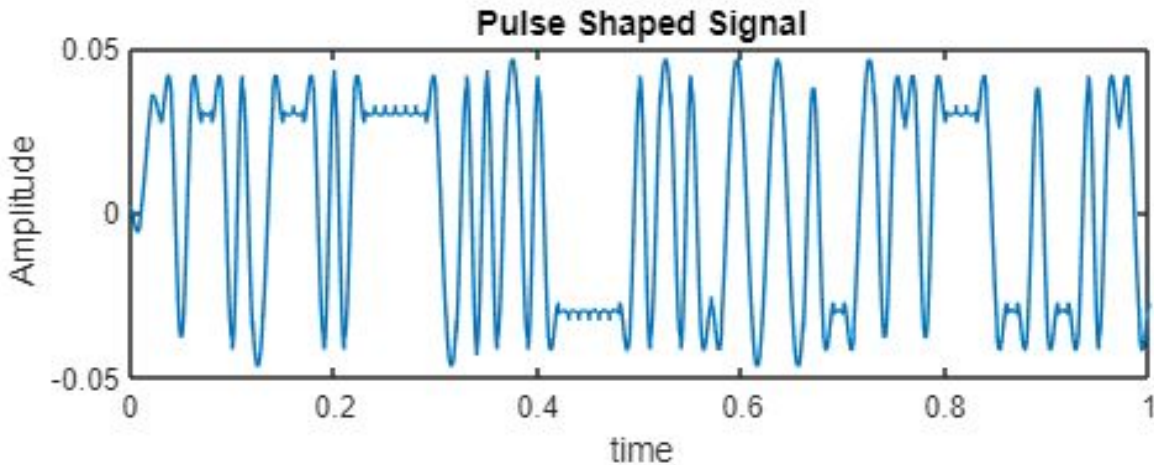
## 2) Pulse Shaping Filter

The NRZ signal is now passed through the pulse shaping filter, which shapes the pulses as shown below

```

rrc_filt =
comm.RaisedCosineTransmitFilter("FilterSpanInSymbols",4,"RolloffFactor",0.35,"OutputSamplesPerSymbol",1024);

```



### 3) Modulation of Signal

This segment modulates the bit stream into BPSK and/or QPSK as per specification

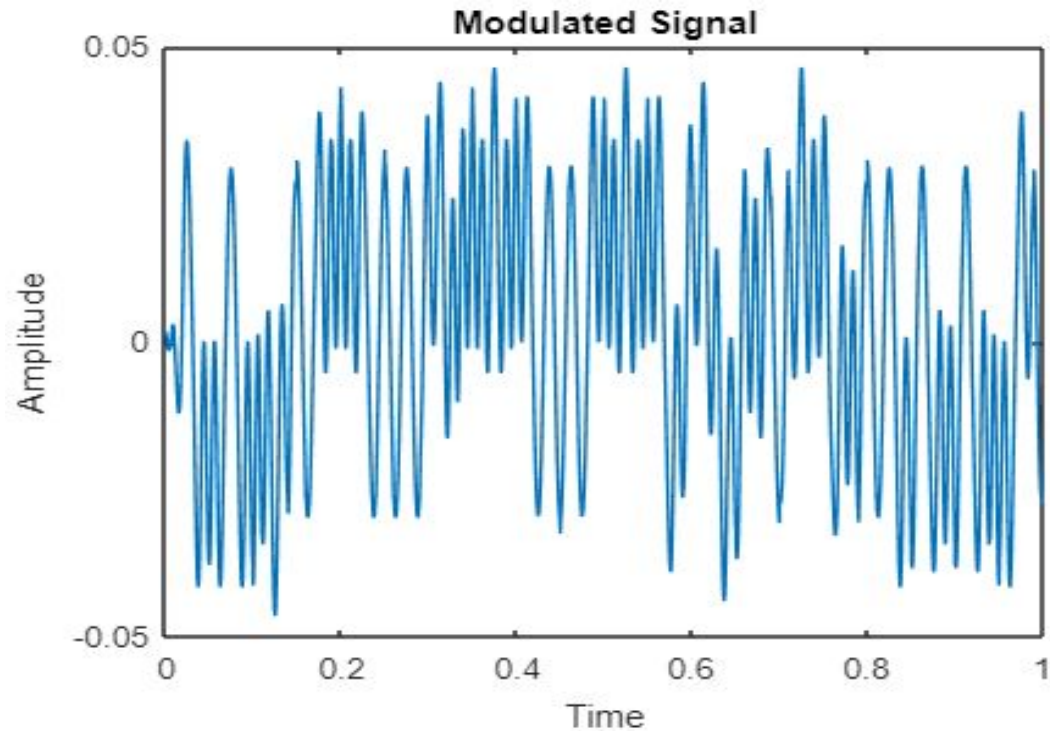
```
%*****BPSK Modulation of rrc_signal*****
```

```
carrier0 = cos(2*pi*fc*t); %Carrier Wave of 900 Hz
carrier1 = cos(2*pi*fc*t + pi/4)
carrier2 = cos(2*pi*fc*t + 3*pi/4);
carrier3 = cos(2*pi*fc*t + 5*pi/4);
carrier4 = cos(2*pi*fc*t + 7*pi/4);
```

```
x_bpsk = rrc_sig .* carrier0;
```

```
%*****QPSK Modulation of rrc_signal*****
```

```
for a = 1:fsb:length(st)-fsb
    if(signal(a)== -1 || signal(a+fsb) == -1)
        x_qpsk(a:a+2*fsb-1) = rrc_sig(a:a+2*fsb-1) .* carrier1(a:a+2*fsb-1);
    elseif(signal(a)== -1 || signal(a+fsb) == 1)
        x_qpsk(a:a+2*fsb-1) = rrc_sig(a:a+2*fsb-1) .* carrier2(a:a+2*fsb-1);
    elseif(signal(a)== 1 || signal(a+fsb) == -1)
        x_qpsk(a:a+2*fsb-1) = rrc_sig(a:a+2*fsb-1) .* carrier3(a:a+2*fsb-1);
    else
        x_qpsk(a:a+2*fsb-1) = rrc_sig(a:a+2*fsb-1) .* carrier4(a:a+2*fsb-1);
    end
end
```

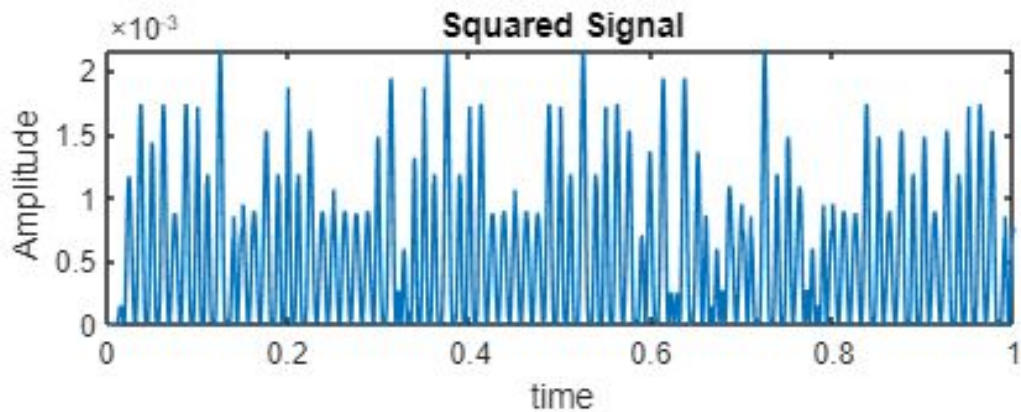


#### 4) Squaring

The entire Carrier Acquisition Module starts here. We devise a strategy to retrieve the carrier frequency from the modulated signal. For that, we first square the received modulated signal:

$$(m(t) \cos(2 * \pi * f_c * t))^2 = \frac{1}{2} + \frac{1}{2} \cos(2 * \pi * 2f_c * t)$$

Squaring the message signal  $m(t)$  yields 1 since  $m(t)$  can only achieve values 1 and -1.



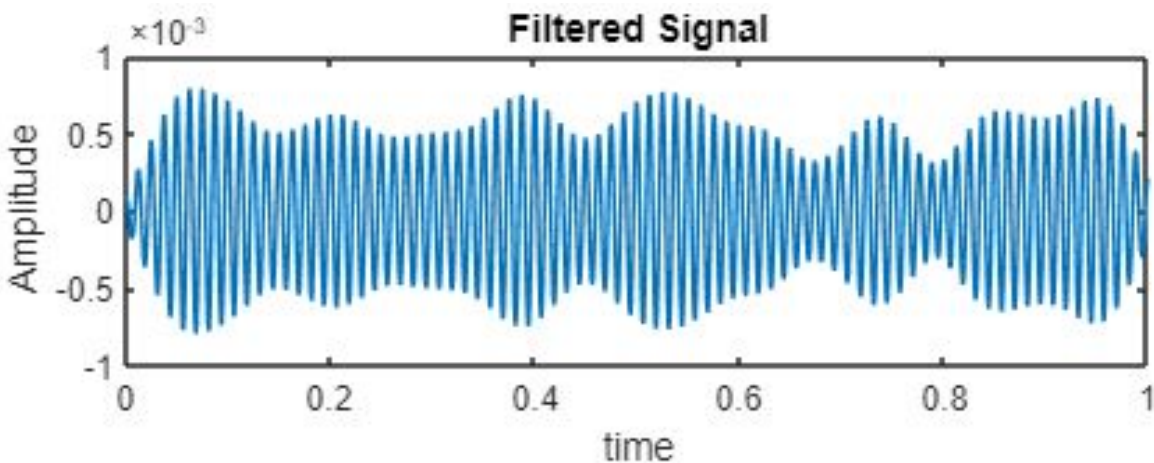
## 5) Bandpass Filtering

Now, when we have squared this signal, we can see that  $2 f_c$  frequency component appears, and any dependence on the message signal has vanished after squaring. Now we will extract the  $2 f_c$  component and filter out the DC component using a bandpass filter.

This bandpass filter is set around  $2f_c$ , and its width is decided by the **acquisition error** as input by the user

```
x_sq = x .* x;           %Squared Signal BPSK

x_sq_filt = bandpass(x_sq, [2*(fc-delta_fc) 2*(fc+delta_fc)], fs); %Bandpass
filtered output
```



## 6) Decimation

Now that we have obtained the  $2 f_c$  component, we should discuss a few things about Discrete Fourier Transforms before proceeding ahead. This will also make the need for Decimation clearer.

MATLAB uses the `fft` function to compute the fourier transform of a signal. This fourier transform is actually Discrete Fourier Transform, wherein frequencies are discrete in terms of bins. The resolution of each bin is given by  $\frac{f_s}{nfft}$ , where  $f_s$  is the sampling frequency while  $nfft$  is the number of samples in the fourier domain. If bin resolution is for example, say 1000, then any frequency between 1500 to 2500 will show up on 2000, while any frequency between 2500 to 3500 will show up on 3000. Hence, frequency estimation error increases if bin resolution is high.

In order to tackle this problem, we decimate (downsample) the squared, filtered signal, so that accuracy of estimation increases. After decimating this signal, we examine the fourier plot of it,



and try to find the index with maximum magnitude of fourier transform. Hence, we locate the 2<sup>nd</sup> component, and estimate  $f_c$ .

```
%Filtered Waveform
FILTERED = abs(fft(decimated_op, nfft));
FILTERED = FILTERED(1:nfft/2);
[Max, k] = max(FILTERED); %finding bin of interest
```

\*\*\*\*\*Showing results of BPSK Modulated Signal\*\*\*\*\*

Actual carrier frequency is 40.000000

Estimate (without any interpolation) = 40.000000

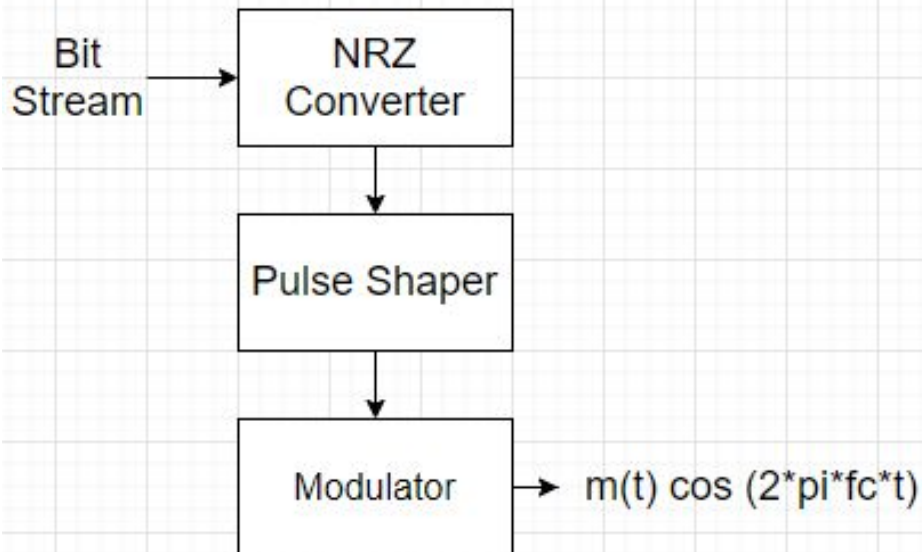
Estimate after interpolation = 50.803983

The interpolation here is done by taking the weighted mean of the  $k$ th,  $k-1$ th and  $k+1$ th index, with weights as their fourier transform magnitudes ( $k$  being the frequency with maximum fourier transform magnitude).

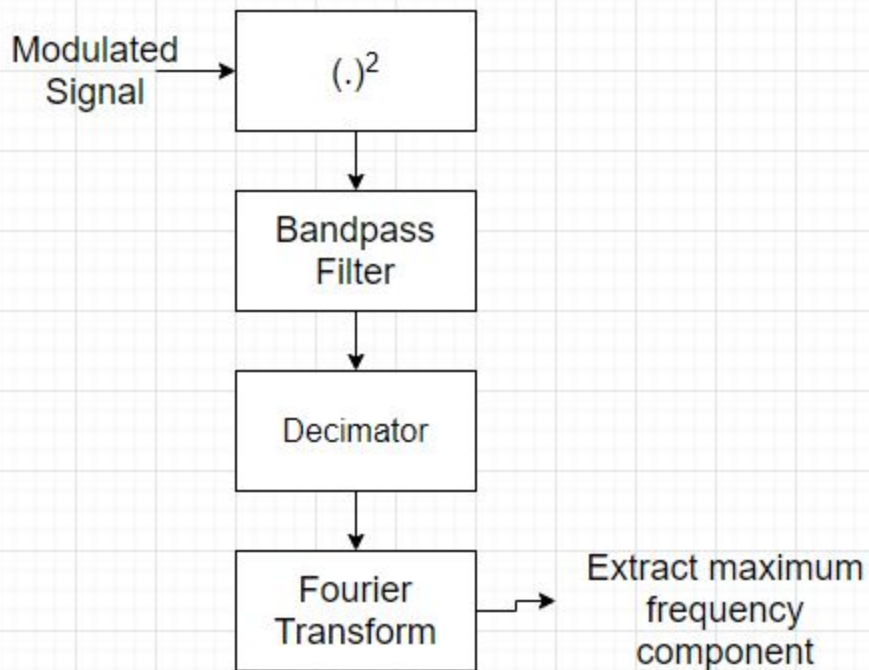
Frequency =  $k_{\text{effective}} \times f_s/n_{\text{fft}}$

Interpolation works better when the desired frequency is located between bins.

## Modulation Flowchart



## Carrier Acquisition Module Flowchart



## **CONCLUSION AND RECOMMENDATION**

I would like to conclude by saying that the Carrier Acquisition module works perfectly well. It gives an estimation error of about 50 Hz, which is further improved by interpolation. All schemes chosen in the report were deliberately chosen keeping in mind their performance. For example, Binary Phase Shift Keying and Quadrature Phase shift Keying were used because of their highest energy efficiency and low bit error rates. Square Carrier Acquisition Module was used in order to make the system indifferent to the algebraic sign of the message signal.

In this way, we have ensured that the system gives best performance, by careful consideration of all schemes and using the best available. We recommend the readers to do the same, but any improvement is welcome.

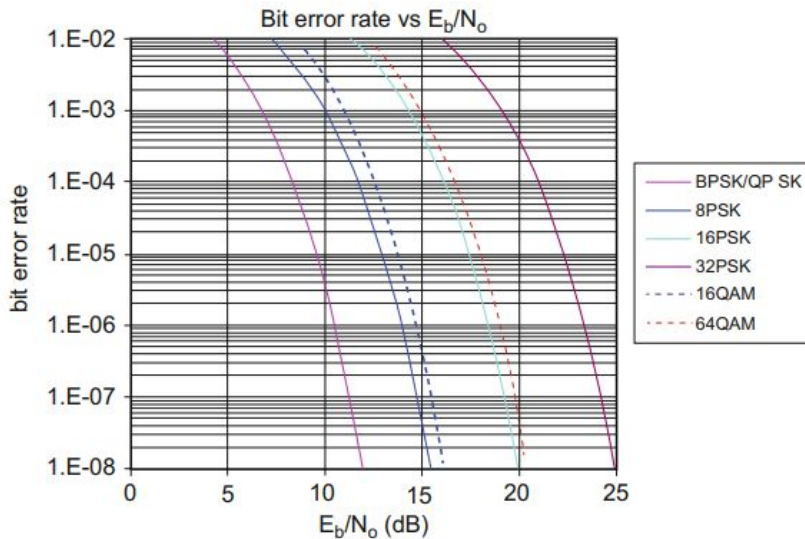
## APPENDIX

### 1) Steps involved in choosing a particular modulation scheme:

- We first fix the Bit Error Rate (BER) at which the transmission is acceptable to us
- Now we need to find out the Channel SNR (C/N level) corresponding to all modulation schemes for the chosen BER value
- These parameters are related as follows:

$$C/N_{dB} = E_b/N_{0\ dB} + \left(\frac{f_b}{B}\right)_{dB} + Coding\ Gain_{dB}$$

- We mark a horizontal line on BER vs  $E_b/N_{0\ dB}$  curve, and lookup the respective  $E_b/N_{0\ dB}$  corresponding to all modulation schemes. We choose those which are feasible for us.
- Next we have to decide the options regarding data rate  $f_b$  and channel bandwidth  $B$  that are available to us, corresponding to each of the chosen BER and  $E_b/N_{0\ dB}$
- We must now select an appropriate filter for Down Conversion purposes
- High Channel SNR requires high power consumption. If power consumption is a problem, we can decrease it at the cost of frequency Bandwidth  $B$ . This is done by introducing Forward Error Correction.



BER versus  $E_b / N_o$  for a range of modulations appropriate to satellite links

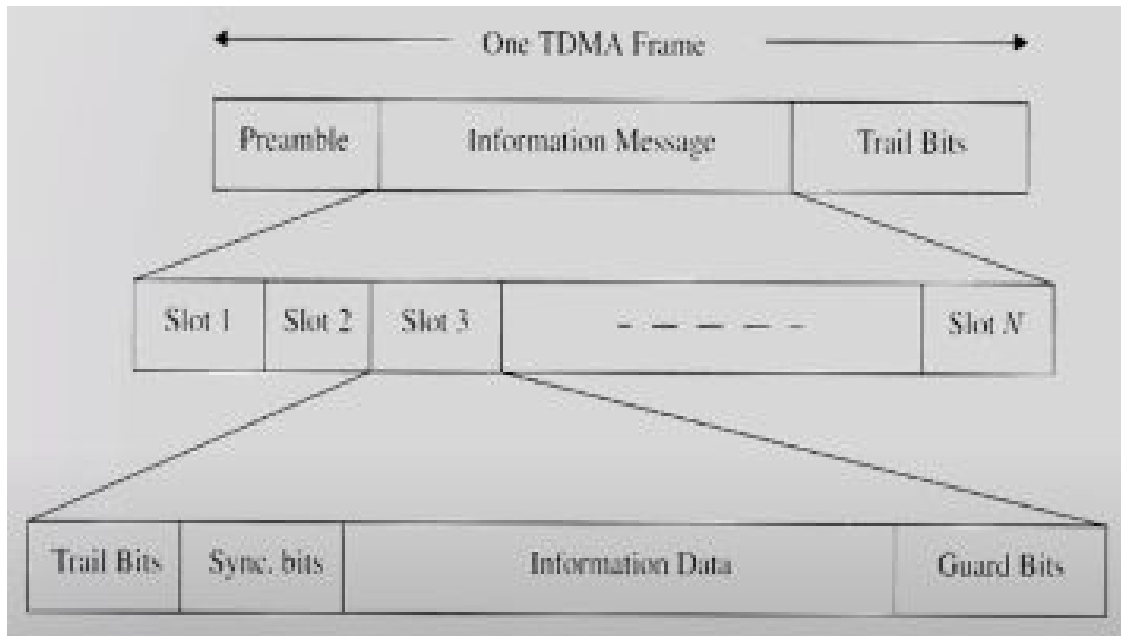
**Table 1** DVB-S2 performance at the QEF threshold

DVB-S2 MOD	FEC rate	Spectral efficiency (bps/Hz)	Ideal $E_s / N_o$ dB	IDEAL $E_b / N_o$ dB	Including modem margin $E_b/N_o$ dB	M
QPSK	1/4	0.490243	-2.35	0.75	1.95	4
QPSK	1/3	0.656448	-1.24	0.59	1.79	4
QPSK	2/5	0.789412	-0.30	0.73	1.93	4
QPSK	1/2	0.988858	1.00	1.05	2.25	4
QPSK	3/5	1.188304	2.23	1.48	2.68	4
QPSK	2/3	1.322253	3.10	1.89	3.09	4
QPSK	3/4	1.487473	4.03	2.31	3.51	4
QPSK	4/5	1.587196	4.68	2.67	3.87	4
QPSK	5/6	1.654663	5.18	2.99	4.19	4
QPSK	8/9	1.766451	6.20	3.73	4.93	4
QPSK	9/10	1.788612	6.42	3.89	5.09	4
8PSK	3/5	1.779991	5.50	3.00	4.20	8
8PSK	2/3	1.980636	6.62	3.65	4.85	8
8PSK	3/4	2.228124	7.91	4.43	5.63	8
8PSK	5/6	2.478562	9.35	5.41	6.61	8
8PSK	8/9	2.646012	10.69	6.46	7.66	8
8PSK	9/10	2.679207	10.98	6.70	7.90	8
16APSK	2/3	2.637201	8.97	4.76	5.96	16
16APSK	3/4	2.966728	10.21	5.49	6.69	16
16APSK	4/5	3.165623	11.03	6.03	7.23	16
16APSK	5/6	3.300184	11.61	6.42	7.62	16
16APSK	8/9	3.523143	12.89	7.42	8.62	16
16APSK	9/10	3.567342	13.13	7.61	8.81	16
32APSK	3/4	3.703295	12.73	7.04	8.24	32
32APSK	4/5	3.951571	13.64	7.67	8.87	32
32APSK	5/6	4.119540	14.28	8.13	9.33	32
32APSK	8/9	4.397854	15.69	9.26	10.46	32
32APSK	9/10	4.453027	16.05	9.56	10.76	32

- Forward Error Correction or Coding involves sending some redundant information along with the signal to ensure error free communication through a noisy channel. It provides us with additional gain called **Coding gain**. It has a distinct value for every value of **BER**

## 2) Burst Transmission

Data is transmitted in burst mode while using the TDMA scheme (Time Division Multiple Access). This means that data is sent in packets for short time durations. For the remaining time, the line stays on sleep mode. This helps us save a lot of power. The format of each packet is as follows:



Preamble => It contains address information, as well as clock synchronization bits. It is used by base station users to identify each other

Information Message => This segment is divided into slots for each user. The *Trail bits* are used to awake the system and raise power level for message transmission. *Sync bits* are again used to synchronize the clock of sender and receiver. *Guard bits* are used to separate different slots to avoid interference.

Trail bits => These are used in the end to lower power level when the transmission is complete.

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- 2) [https://www.researchgate.net/publication/271253098\\_Design\\_and\\_Implementation\\_of\\_B psk\\_Modulator\\_and\\_Demodulator\\_Using\\_Vhdl](https://www.researchgate.net/publication/271253098_Design_and_Implementation_of_B psk_Modulator_and_Demodulator_Using_Vhdl)
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- 5) <https://ieeexplore.ieee.org/document/6420723>

## GLOSSARY

- 1) **Modulation:** In electronics and telecommunications, modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal, with a modulating signal that typically contains information to be transmitted
- 2) **Demodulation:** Demodulation is extracting the original information-bearing signal from a carrier wave
- 3) **Signal to Noise Ratio:** Signal-to-noise ratio is a measure that compares the level of a desired signal to the level of background noise. SNR is defined as the ratio of signal power to the noise power, often expressed in decibels. A ratio higher than 1:1 indicates more signal than noise
- 4) **Bit Error Rate:** In digital transmission, the number of bit errors is the number of received bits of a data stream over a communication channel that have been altered due to noise, interference, distortion or bit synchronization errors. The bit error rate is the number of bit errors per unit time.
- 5) **Radio Frequency:** Radio frequency is the oscillation rate of an alternating electric current or voltage or of a magnetic, electric or electromagnetic field or mechanical system in the frequency range from around 20 kHz to around 300 GHz
- 6) **Low Pass Filter:** A low-pass filter is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency.
- 7) **Bandpass Filter:** A bandpass filter passes all signals contained within a band specified around a certain frequency.