Bangladesh University of Engineering and Technology

Project title

Modeling and Performance analysis of a RF Satellite

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Abstract

In this project, the modeling and performance of the satellite downlink system using 16 QAM digital modulation techniques is analyzed and investigated. The Quadrature Amplitude Modulation (QAM) is an important modulation scheme with many practical applications, including current and future wireless technologies. More spectrally efficient digital modulation scheme, such as M-ary quadrature amplitude modulation (QAM), is an attractive technique to achieve high-rate transmission without increasing the bandwidth. And exact evaluation of bit error probability for M-ary QAM can be obtainable for arbitrary M. For the case of M = 16 QAM, this project will show some practical results using tests on Matlab Simulink satellite system model to investigate the effect of antenna diameter, carrier frequency and altitude on BER.

Introduction

Satellite communications are used to provide communication links between different points on the Earth by receiving a signal from a transmitting earth station. Satellite communications play a vital role in the global communications system. Nearly 2,000 satellites are orbiting the Earth relaying analog and digital signals carrying audio, video and data to and from one or many locations around the world. Specially in the recent years, personal wireless communication is getting more and more popular and is continuing to grow at an exponential rate. This growth has triggered a tremendous demand for not only higher transmission capacity, but also greater coverage area and better quality of service. This demand is being served by employing digital modulation, which provides noise immunity and robustness to channel deterioration and a better error detection and correction control. A communications satellite is an orbiting artificial earth satellite that receives a signal from a transmitting earth station, amplifies and potentially processes it, then transmits it back to a ground station. Communications information does not start and/or end in the satellite itself. The satellite is an active relay, similar to the propagation function of towers used in terrestrial microwave communications. Recent theoretical studies of communication systems show much interest in high-level modulation, such as M-ary Quadrature amplitude modulation (M-QAM), and most related works are based on the simulations. In this paper, a simulation model to study various QAM modulation

M-ary Quadrature Amplitude Modulation

Quadrature amplitude modulation is the combination of amplitude shift keying and phase shift keying. Quadrature amplitude modulation is a system in which data is transferred by changing some aspect of a carrier signal, or the carrier wave, usually a sinusoid in response to a data signal. In the case of QAM, the amplitude of two waves of the same frequency, 90° out-of-phase with each other in quadrature are changed modulated or keyed to represent the data signal. In quadrature amplitude modulation, a signal obtained by summing the amplitude and phase modulation of a carrier signal modulated sine and cosine waves or quadrature waves are used for the data transfer [3]. The mathematical representation of the M-ARY QAM is:

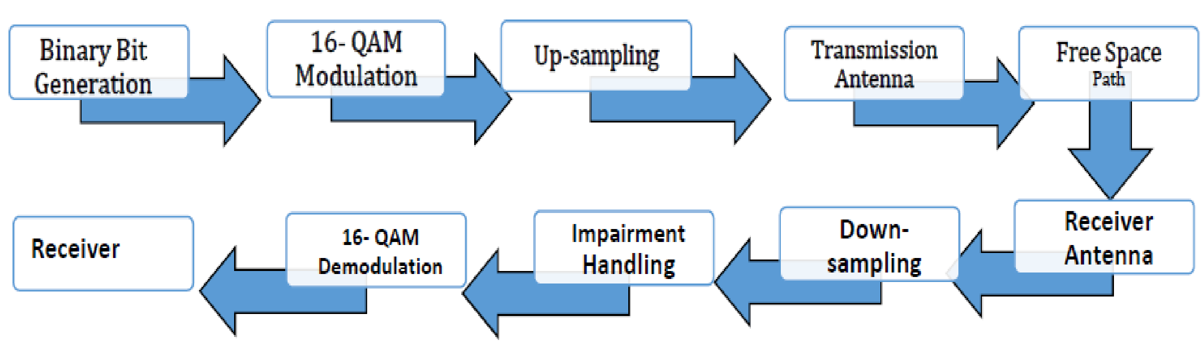
$$s_i(t) = a_i \cos(2f_c t) + b_i \sin(2f_c t) \qquad where : 0 \leq t \leq T\,;\, i = 1,2, \ldots . M$$

Phase modulation (analog PM) and phase-shift keying (digital PSK) can be regarded as a special case of QAM, where the magnitude of the modulating signal is a constant, with only the phase varying. This can also be extended to Frequency modulation (FM) and frequency-shift keying (FSK), for these can be regarded as a special case of phase modulation.

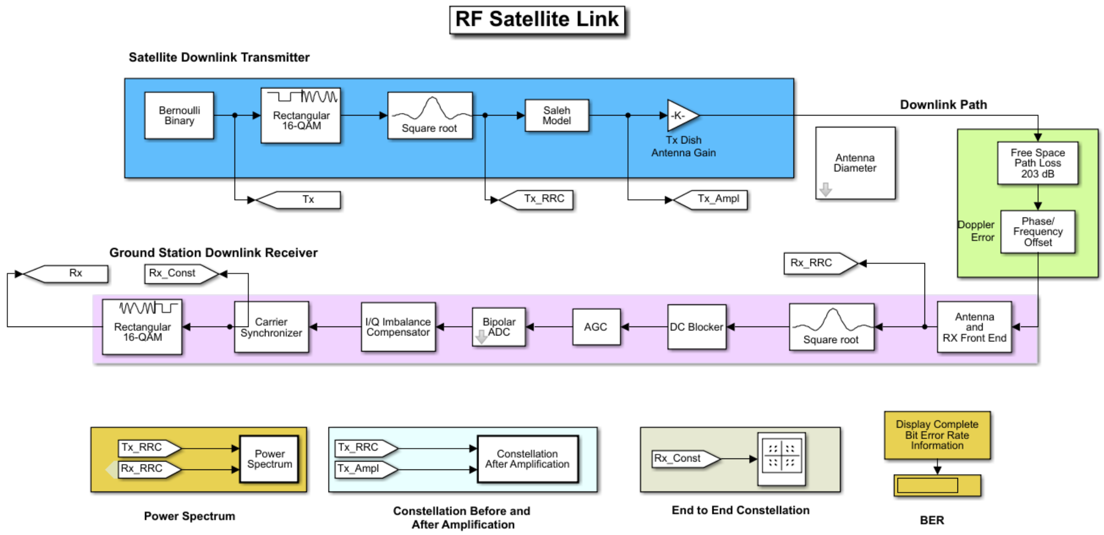
Simulation Summary

Binary Bit Generation > 16-QAM Modulation > Up-sampling > Transmission Antenna > Free Space > Receiver Antenna > Down-sampling> Impairment Handling > Demodulation > Received Binary Bit

Work Flow:



Simulink Model of the Satellite Link



Internal Subsystems

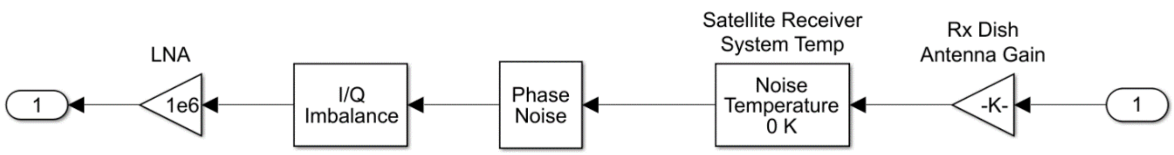


Fig: Antenna and RX Front End

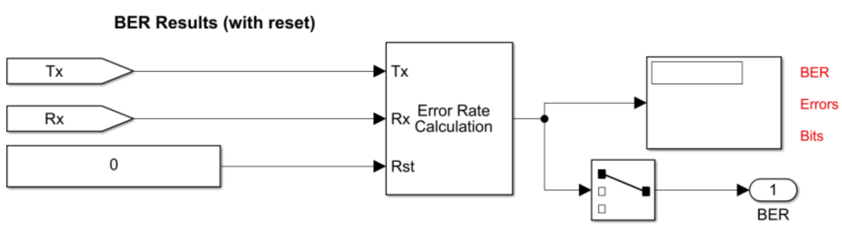


Fig: Bit Error Rate Calculation Block

Simulink Block Summary

1.Bernoulli Binary Generator

The Bernoulli Binary Generator block generates random binary numbers using a Bernoulli distribution. Use this block to generate random data bits to simulate digital communication systems and obtain performance metrics such as bit error rate. The Bernoulli distribution with parameter p produces zero with probability p and one with probability 1-p. The Bernoulli distribution has mean value 1-p and variance p(1-p). The Probability of zero parameter specifies p and can be any real number in range [0, 1].

The output signal can be a column or row vector, two-dimensional matrix, or scalar. The number of rows in the output signal corresponds to the number of samples in one frame and is set by the Samples per frame parameter. The number of columns in the output signal corresponds to the number of channels and is set by the number of elements in the Probability of zero parameter

Output: Output data signal, returned as a scalar, vector, or matrix.

Probability of zero

Probability of zero must be in the range of [0, 1]. The number of elements in the Probability of zero parameter corresponds to the number of independent channels output from the block. The Bernoulli distribution with parameter *p* produces zero with probability *p* and one with probability *1-p*. We assign it to 0.5

Source of initial seed

When the Source of initial seed parameter is set to Auto and the Simulate using parameter is set to Code generation, the random number generator uses an initial seed of zero. In this case, the block generates the same random numbers each time it is started. So, we set it to Auto for testing various cases with the same random numbers inside the model.

Sample time

Output sample time, specified as -1 or a positive scalar that represents the time between each sample of the output signal. We set it to 1e-5

Samples per frame

Samples per frame in one channel of the output signal, specified as a positive integer. We assign 400 in this field.

2.Rectangular QAM Modulator Baseband

The Rectangular QAM Modulator Baseband block modulates using M-ary quadrature amplitude modulation with a constellation on a rectangular lattice. The output is a baseband representation of the modulated signal. This block accepts a scalar or column vector input signal.

M-ary number

The number of points in the signal constellation. It must have the form 2^K for some positive integer K.

We choose 16 as M.

.

Input Type

When we set the Input type parameter to Bit, the block accepts binary-valued inputs that represent integers. The block collects binary-valued signals into groups of K = log2(M) bits where K represents the number of bits per symbol. We choose the input type as 'bit' as we are injecting binary bits into the QAM Modulator.

Constellation ordering

Determines how the block maps each symbol to a group of output bits or integer. If Constellation ordering is set to Gray and K is even, the block uses a Gray-coded constellation.

Normalization method

Determines how the block scales the signal constellation. Choices are Min. distance between symbols, Average Power, and Peak Power. We assign Average Power as a Normalization method.

Average power, referenced to 1 ohm (watts)

The average power of the symbols in the constellation, referenced to 1 ohm. This field appears only when the Normalization **method** is set to Average Power. We set the average power as 1 referred to 1 ohm.

Phase offset (rad)

The rotation of the signal constellation, in radians. This is set to zero(0).

3.Raised Cosine Transmit Filter

The Raised Cosine Transmit Filter blocks up samples and filters the input signal using a normal raised cosine FIR filter or a square root raised cosine FIR filter. The **Filter shape** parameter determines which type of filter the block uses; choices are Normal and Square root. Square root type FIR filter is chosen in our simulation.

Filter shape
To specify the filter shape as square root or Normal. We choose square root.

Roll Off factor
The Roll-off factor parameter is the filter's roll off factor. It must be a real number between 0 and 1. The roll-off factor determines the excess bandwidth of the filter. For example, a roll off factor of .5 means that the bandwidth of the filter is 1.5 times the input sampling frequency. The assigned roll off factor is 0.3.

Filter span in symbols
Because the ideal raised cosine filter has an infinite impulse response, the block truncates the impulse response to the number of symbols that the Filter span in symbols parameter specifies. The Filter span in symbols, N, and the Output samples per symbol, L, determine the length of the filter's impulse response, which is L * Filter span in symbols + 1. The assigned filter span is 6.

Output samples per symbol
Specify the number of output samples for each input symbol. The default is 8. This property accepts an integer-valued, positive scalar. The number of taps for the raised cosine filter equals the value of this parameter multiplied by the value of the **Filter span in symbols** parameter.

Linear amplitude filter gain
Specify a positive scalar value that the block uses to scale the filter coefficients. By default, the block normalizes filter coefficients to provide unit energy gain. If specified a gain other than 1, the block scales the normalized filter coefficients using the gain value you specify.

3. Memoryless Nonlinearity

The Memoryless Nonlinearity block applies memoryless nonlinear impairments to a complex baseband signal. Use this block to model memoryless nonlinear impairments caused by signal amplification in the radio frequency (RF) transmitter or receiver.

Nonlinearity modeling method

Nonlinearity modeling method, specified as Cubic polynomial, Hyperbolic tangent, Saleh model, Ghorbani model, Rapp model, or Lookup table. Our choice for the simulation is the Saleh Model.

Input scaling (dB)

Input signal scaling factor in decibels, specified as a scalar. This parameter scales the power gain of the input signal

AM/AM parameters [alpha beta]

AM/AM parameters for Saleh model, used to compute the amplitude gain for an input signal, specified as a two-element vector.

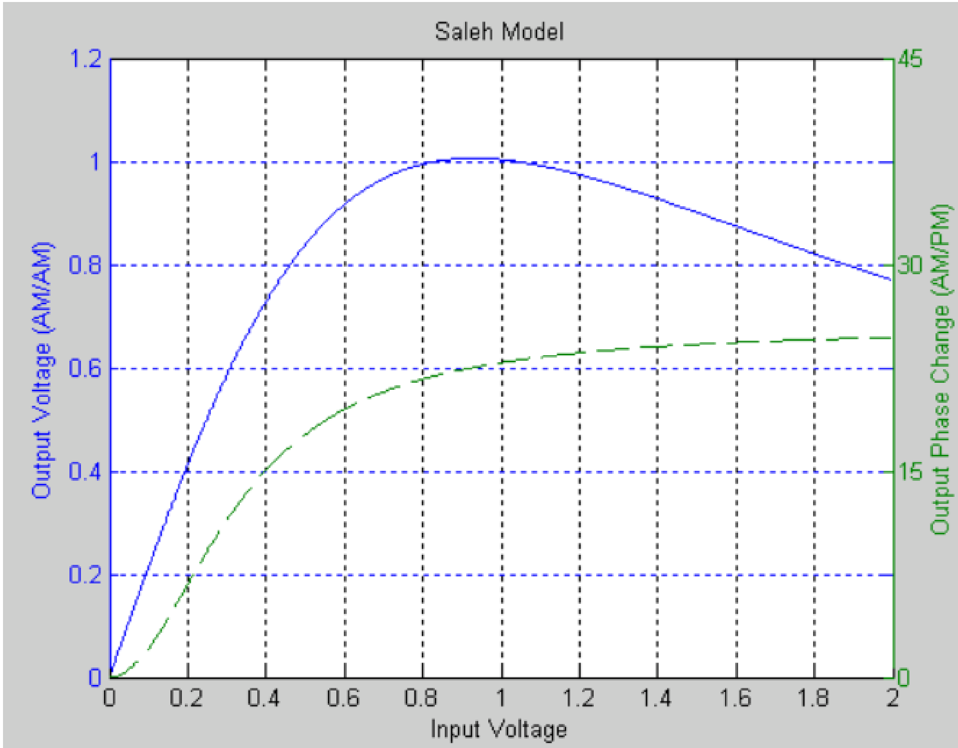
AM/PM parameters [alpha beta]

AM/PM parameters for Saleh model, used to compute the phase change for an input signal, specified as a two-element vector.

Memoryless nonlinear impairments distort the input signal amplitude and phase. The amplitude distortion is amplitude-to-amplitude modulation (AM/AM) and the phase distortion is amplitude-to-phase modulation (AM/PM).

Saleh Model Method

This figure shows the AM/AM behavior (output voltage versus input voltage for the AM/AM distortion) and the AM/PM behavior (output phase versus input voltage for the AM/PM distortion) for the Saleh model method.



The AM/AM parameters, α_{AMAM} and β_{AMAM}, are used to compute the amplitude distortion of the input signal by using

$$F_{\mathrm{AMAM}}(u) = \frac{\alpha_{\mathrm{AMAM}} \times u}{1 + \beta_{\mathrm{AMAM}} \times u^2},$$

where u is the magnitude of the scaled signal.

The AM/PM parameters, α_{AMPM} and β_{AMPM}, are used to compute the phase distortion of the input signal by using

$$F_{\mathrm{AMPM}}(u) = \frac{\alpha_{\mathrm{AMPM}} \times u^2}{1 + \beta_{\mathrm{AMPM}} \times u^2},$$

where u is the magnitude of the scaled signal. The α and β parameters for AM/AM and AM/PM are similarly named but distinct.

Output scaling (dB)

Output signal scaling factor in decibels, specified as a scalar. This parameter scales the power gain of the output signal. We choose 30db as output scaling.

4. Transmitter Antenna Gain

The antenna gain is parameterized by the antenna diameter. We created a mask to take the antenna diameter as user input. This input is used to measure antenna gain. The Gain block multiplies the input by a constant value (gain). The input and the gain can each be a scalar, vector, or matrix.
We specify the value of gain in the Gain parameter as a variable "TxGain".

5. Antenna Diameter (User-defined Block)

This block prompts users to give the antenna diameters as input. Underlying the mask, there are two functions working to find out the transmitter and receiver antenna gain and save them in the base workspace of Simulink. These values are assigned to the antenna gains.

6. Free Space Path Loss

The Free Space Path Loss block applies a free space path loss to a complex signal, specified as a scalar or column vector. The block simulates the loss of signal power due to the distance between the transmitter and receiver. The Mode parameter indicates whether one wants to specify the loss in decibels or as a computation that is based on distance and the RF signal frequency. The output signal is returned as a scalar or column vector. This output is the same dimension and data type as the input signal.

The free-space path loss, L, in decibels is

$$L = 20\log_{10}\left(\frac{4\pi R}{\lambda}\right)$$

We can measure this via Matlab code also.
Code:

```
1. fc = 4e9;
2. lambda = physconst('LightSpeed')/fc;
3. R = 35600000;
4. L = fspl(R,lambda)
```

Output: 195.518

In our model, we used the 'Distance and Frequency' as the loss calculation mode. The other two inputs were distance (between the satellite and the ground station) and the carrier frequency of the link. Changing this parameter updates the Free Space Path Loss block. For 35600 Km distance and 4000MHz carrier frequency, the path loss was 196dB.

7. Phase/Frequency Offset (Doppler Error)

This block Rotates the signal to model Doppler error on the link. The Doppler effect or Doppler shift is the change in frequency of a wave in relation to an observer who is moving relative to the wave source. Fast-moving satellites can have a Doppler shift of dozens of KHz relative to a ground station. To execute this error the Phase/ Frequency Offset block is used in this RF satellite model. The block allows one to select one of two values of Doppler which comes in the form of phase offset or frequency offset. The selection updates the Phase/Frequency Offset (Doppler Error) block. The default setting is 0 Hz which means no Doppler on the link.

If the input signal is $u(t)$, then the output signal is

$$y(t) = u(t) \cdot \left(\cos\left(2\pi \int_0^t f(\tau)d\tau + \varphi(t) \right) + j \sin\left(2\pi \int_0^t f(\tau)d\tau + \varphi(t) \right) \right),$$

where $f(t)$ is the frequency offset, and $\varphi(t)$ is the phase offset.

The discrete-time output is given by

$$y(0) = u(0)(\cos(\varphi(0)) + j \sin(\varphi(0))) \text{ and}$$

$$y(i) = u(i)\left(\cos\left(2\pi \sum_{n=0}^{i-1} f(n)\Delta t + \varphi(i) \right) + j \sin\left(2\pi \sum_{n=0}^{i-1} f(n)\Delta t + \varphi(i) \right) \right),$$

where $i > 0$, and Δt is the sample time.

In our model, we set the offset values to zero as no doppler error is encountered.

9. Receiver Antenna Gain

The antenna gain is parameterized by the antenna diameter. We created a mask to take the antenna diameter as user input. This input is used to measure antenna gain. The Gain block multiplies the input by a constant value (gain). The input and the gain can each be a scalar, vector, or matrix.

We specify the value of gain in the Gain parameter as a variable “RxGain”.

10. Satellite Receiver System Temperature

The Receiver Thermal Noise block applies receiver thermal noise to a signal. The block simulates the effects of thermal noise on a signal. The Specification method parameter enables the specification of thermal noise based on noise temperature, noise figure, or noise factor. The output signal is returned as a scalar or column vector. This output is the same dimension and data type as the input signal. Here is an initial seed value for the random number generator, specified as a scalar. Basically, it adds white Gaussian noise that represents the effective system temperature of the receiver.

The setting of noise temperature at 0K is to view the other RF impairments without the perturbing effects of noise.

In our model, we set the specification mode to be noise temperature which is fixed at 20K (very low noise level) with an initial seed value of 67987.

11. Phase Noise:

Phase noise is the frequency-domain representation of random fluctuations in the phase of a waveform, corresponding to time-domain deviations from perfect periodicity. The Phase Noise block adds phase noise to a complex signal. This block emulates impairments introduced by the local oscillator of a wireless communication transmitter or receiver. The block generates filtered phase noise according to the specified spectral mask and adds it to the input signal.

Parameters:

- Phase noise level (dBc/Hz)- The phase noise level in decibels relative to carrier per hertz (dBc/Hz), specified as a vector of negative scalars. The Phase noise level (dBc/Hz) and Frequency offset (Hz) parameters must have the same length.
- Frequency offset (Hz)-Frequency offset in Hz, specified as a vector of positive increasing values.
- Sample rate (Hz)- To avoid aliasing, the sample rate must be greater than twice the largest value specified by Frequency offset (Hz).
- Initial seed- This block uses the Random Source block to generate noise. Every time one rerun the simulation, the block reuses the same initial seed.

For a scalar frequency offset and phase noise level specification, an IIR digital filter computes the spectrum mask. The spectrum mask has a 1/f characteristic that passes through the specified point.

For a vector frequency offset and phase noise level specification, an FIR filter computes the spectrum mask. The spectrum mask is interpolated across log10(f). It is flat from DC to the lowest frequency offset, and from the highest frequency offset to half the sample rate.

In our model, frequency offset(100Hz) and phase noise level(-100dBc/Hz) are scalar with a sample rate of 2e5Hz and the initial seed value is 2137.

12. I/Q Imbalance

I/Q imbalance (in-phase and quadrature imbalance) is a performance-limiting issue in the design of direct conversion receivers. It occurs due to mismatches between the parallel sections of the receiver chain dealing with the in-phase and quadrature signal paths. A direct-conversion RF front-end suffers from two major drawbacks: one is
In phase and quadrature imbalance and the other is DC offset.

This block introduces DC offset, amplitude imbalance, or phase imbalance to the signal.
a. Amplitude imbalance - It applies gain to the in-phase signal and the quadrature signal. For example, if the gain of in-phase signal is 1.5dB and -1.5 dB in the quadrature signal then, the amplitude imbalance will be 3dB.
b. Phase imbalance -It rotates the in-phase signal and the quadrature signal. We set the phase imbalance 10 degree in our model.
c. In-phase DC offset - Adds a DC offset to the in-phase signal amplitude. This offset changes the received signal constellation diagram but does not cause errors on the link unless combined with thermal noise or other RF impairments. In our model, we set the value 2e-9.
d. Quadrature DC offset - Adds a DC offset to the quadrature signal amplitude. This offset causes errors on the link even when not combined with thermal noise or another RF impairment. This offset also causes a DC spike in the received signal spectrum. We set the value to be 3e-8.

13. LNA

LNA (Low Noise Amplifier) applies low noise amplifier gain. LNA amplifies the small DC offsets so that they are visible on the constellation diagram with much larger axis limits. So,if the dc offsets are not compensated from the signal, it will help to notice the received signal constellation diagram for a large DC offset and the DC spike in the received signal spectrum. In our model the gain value is set to be 1e6.

14. DC Blocker

This block compensates for the DC offset in the I/Q Imbalance block. The dc offsets shift the constellation diagram of the received signal. So,to compensate such imbalances we need to add a dc blocker that removes dc component from the signal.

The algorithm for estimating dc offset is set IIR in our model which uses a recursive estimate based on a narrow, lowpass elliptic filter. This algorithm typically uses less memory than FIR and is more efficient.

The normalized bandwidth is set to be 0.0005 which has to be a real scalar greater than 0 and less than and the filter order is 3.

15. AGC Block

The automatic gain controller (AGC) block adaptively adjusts its gain to achieve a constant signal level at the output.

Step size

Specify the step size for gain updates as a double-precision or single-precision real positive scalar. The default is 0.01.

If we increase Step size, the AGC responds faster to changes in the input signal level. We assign 0.01 as step size in the block

Desired output power (W)

Specify the desired output power level as a real positive scalar. The power level is specified in Watts referenced to 1 ohm. The default is 1. The assigned value is 1 in our case.

Averaging Length

To specify the length of the averaging window in samples as a positive integer scalar. An increase in **Averaging length** reduces execution speed. We put 256 in this field.

Maximum power gain (dB)

To Specify the maximum gain of the AGC in decibels as a positive scalar. The default is 60.

If the AGC input signal power is very small, the AGC gain will be very large. This can cause problems when the input signal power suddenly increases. We use Maximum power gain (dB) to avoid this by limiting the gain that the AGC applies to the input signal. In this simulation we used 400dB as maximum gain.

16. Analog to Digital Converter

Quantize the input signal with a bipolar ADC. The output levels are floating point numbers in the set of -FSV:LSB:FSV-LSB, where LSB=FSV/2^(N-1) and N is the number of bits. This block is useful to simulate quantization and saturation effects of an ADC on the input signal. Under-lying the block, there are two operations going on. Firstly, the analog signal is quantized and then limited to a certain value. N is 16 and FSV is 1.1.

17. I/Q Imbalance Compensator

The I/Q Imbalance Compensator mitigates the effects of an amplitude and phase imbalance between the in-phase and quadrature components of a modulated signal.

Source of compensator coefficient

If it is set to be estimated from input signal, the compensator calculates the coefficients from the input signal, which is the case with this simulation.

Initial compensator coefficient

To specify the initial coefficient used by the internal algorithm to compensate for the I/Q imbalance. The default value is 0+0j.

Source of adaptation step size

To specify the source of the adaptation step size as Property or Input port. If set to Property, specify the step size in the Adaptation step size field. We set it to property.

Adaptation step size

To specify the step size of the adaptation algorithm as a real scalar. This parameter is available only when the Source of adaptation step size is set to Property. The default value is 0.00001. And we set it to the default value.

18. Carrier Synchronizer

The Carrier Synchronizer block compensates for carrier frequency and phase offsets using a closed-loop approach for BPSK, QPSK, OQPSK, 8-PSK, QAM, and PAM modulation schemes.

Modulation

To specify QAM as the modulation type.

Modulation phase offset

To specify the method used to calculate the modulation phase offset as either Auto or Custom.

Auto applies the traditional offset for the specified modulation type.

Modulation	Phase offset (rad)
BPSK,QAM or PAM	0
QPSK or OQPSK	pi/4
8PSK	pi/8

Samples per symbol

To specify the number of samples per symbol as a positive integer scalar which is 1 in our simulation.

Damping factor

To specify the damping factor of the loop as a positive real finite scalar. This is assigned as 0.707

Normalized loop bandwidth

To specify the normalized loop bandwidth as a real scalar between 0 and 1. The bandwidth is normalized by the sample rate of the carrier synchronizer block.

Observations

The main output parameter in our simulations scheme is the bit error rate (BER). This is shown as

BER = Total error bits / Total Number of bits

We parameterized three independent variables to vary and saw the BER behavior according to the change of the respective variables.

The three variables are:

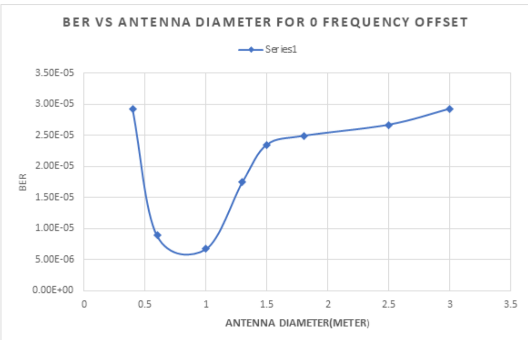
1. Antenna Diameter (Antenna Gain)
2. Carrier Frequency
3. Altitude of the satellite

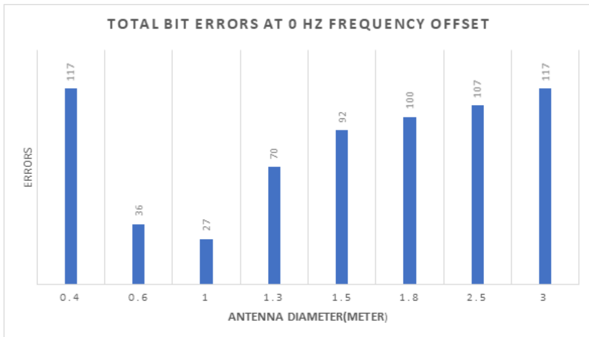
1. BER vs Antenna Diameter

To see the change in BER, we set up two different cases of frequency offset.

a. 0 Hz frequency offset with 5-degree phase offset

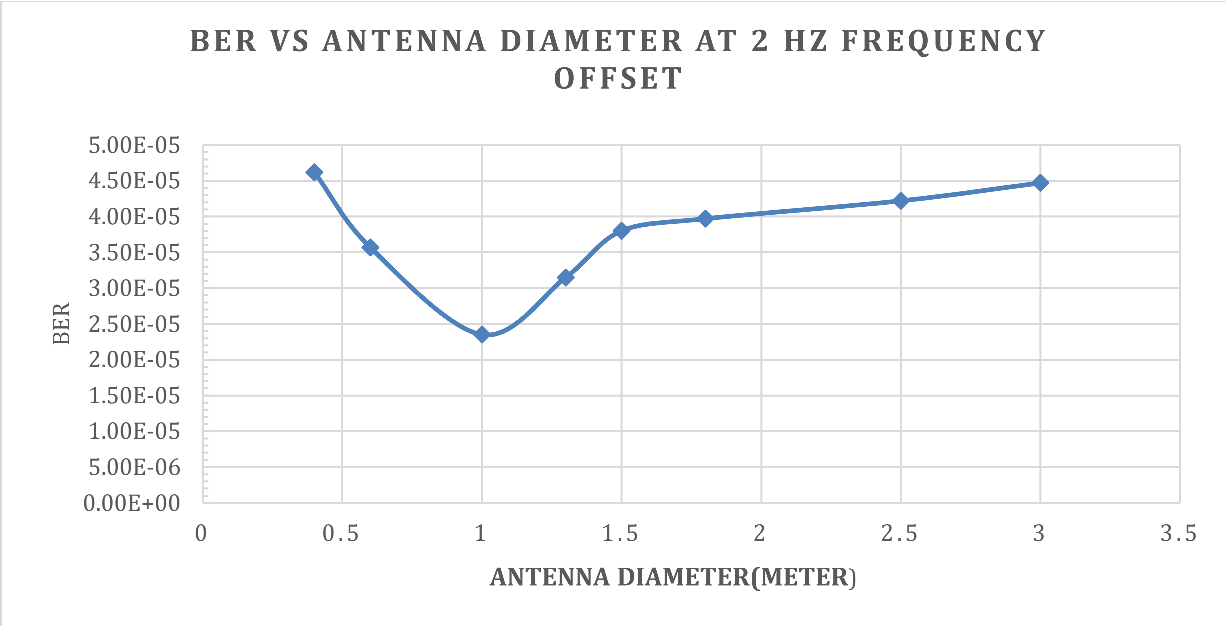
We ran through the antenna diameter inside the simulation and found out the following figure for BER vs antenna diameter.

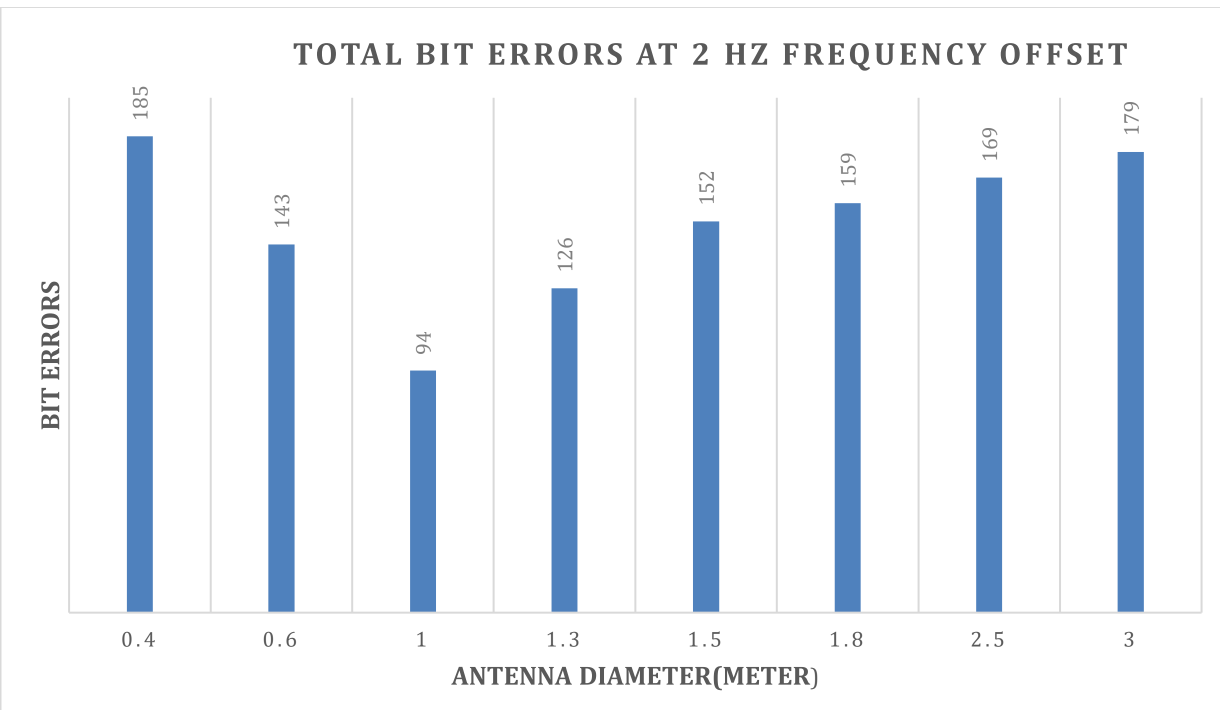




The lowest BER is obtained at 1 meter antenna diameter. For we know, antenna gain increases linearly with antenna diameter, one would assume increase in BER as antenna diameter increases. But the main factor here is the SNR. As antenna gain increases, so does the noise gain into the system. As noise increases, SNR of the system decreases and as a result, we can see an increase in BER after the case of 1 meter diameter.

b. 2 Hz frequency offset with 5 degree phase offset

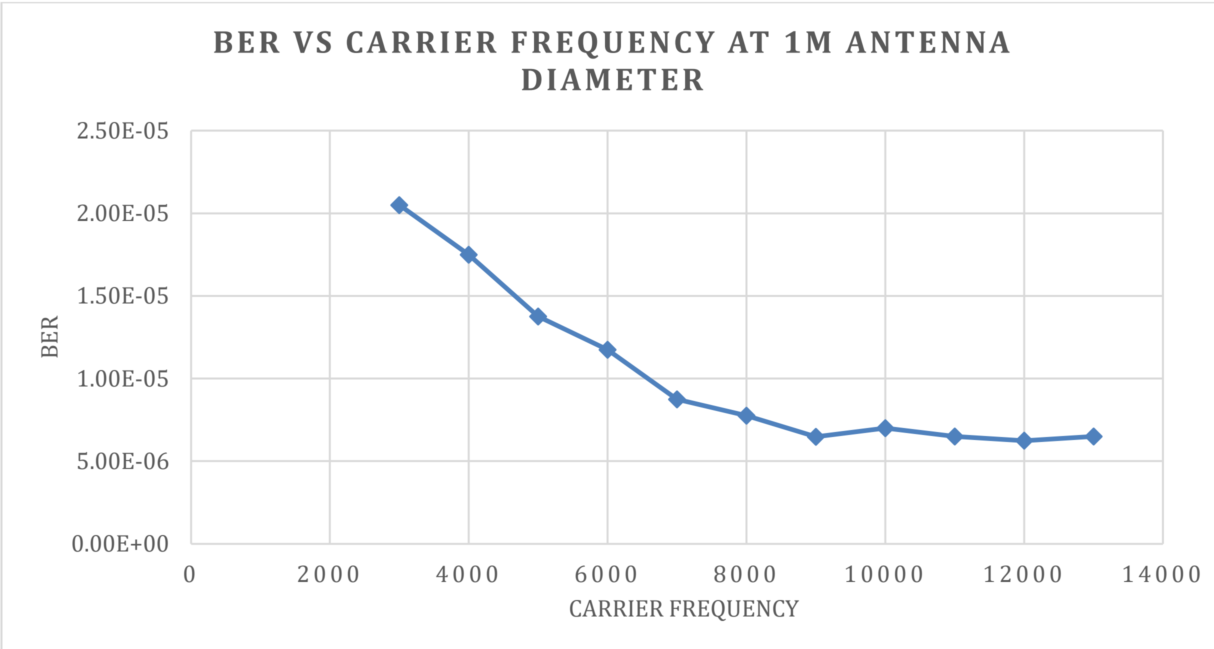


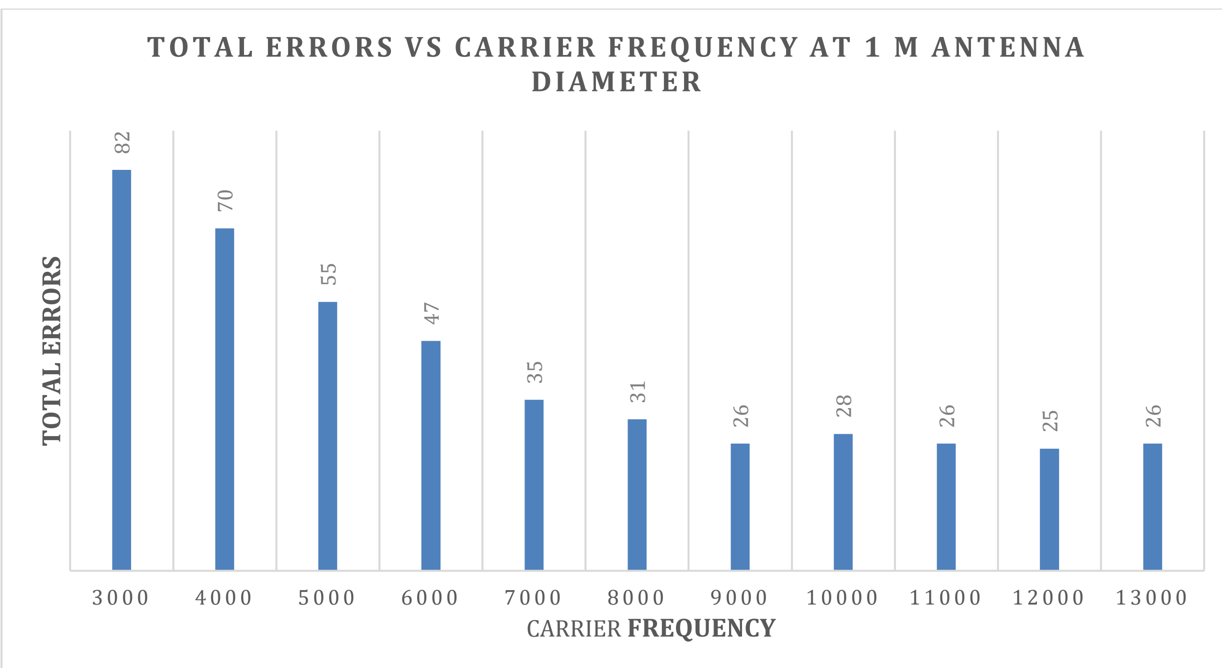


We can see a similar behavior as case a. But the BER is slightly high for each case of antenna diameter. This is because of the 2Hz frequency offset inserted into the system.

2. **BER vs Carrier Frequency**

At an antenna length of 1 meter, we simulated the system for different carrier frequencies and plotted BER against it.

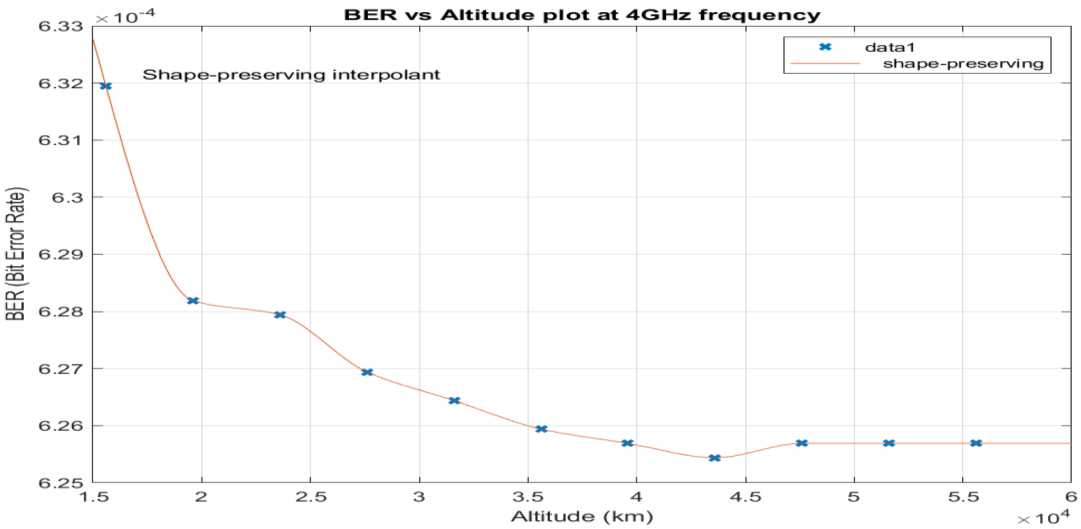




The system works better in the case of 12000 MHz which is the known Ku band. Normally, 0.9, 1.2 or 1.8 meter dishes are used for Ku-Band applications. We took our dish antenna diameter to be 1meter.

In the range of 4GHz(C band) the system has a higher bit error rate.

3. **BER vs altitude of the satellite plot**



Apparently, one will assume that BER will increase as altitude increases. But in our case, it is the opposite. Since, we are only accounting for attenuation (which does nothing to the noise character of the signal), the SNR is not affected here. That's why we don't see an assumed increasing BER in this case.

Constellation diagram

For QAM modulation, a rectangular diagram is used to plot the symbols. We have M = 16 and N = 4 in the simulation system. So we can see how the constellation diagram behaves for no offset and 5Hz frequency offset.

Not much information can be obtained from the constellation diagram if we want to see the effects of minimal change in the system. Only a big change in the system could be understood easily from CD.

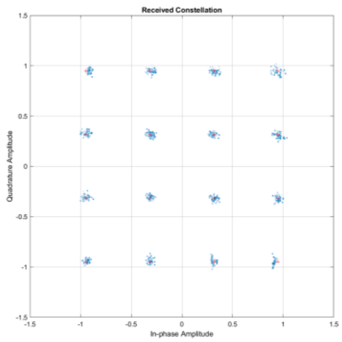


Figure: 0 frequency offset

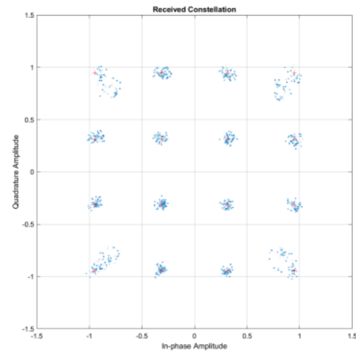
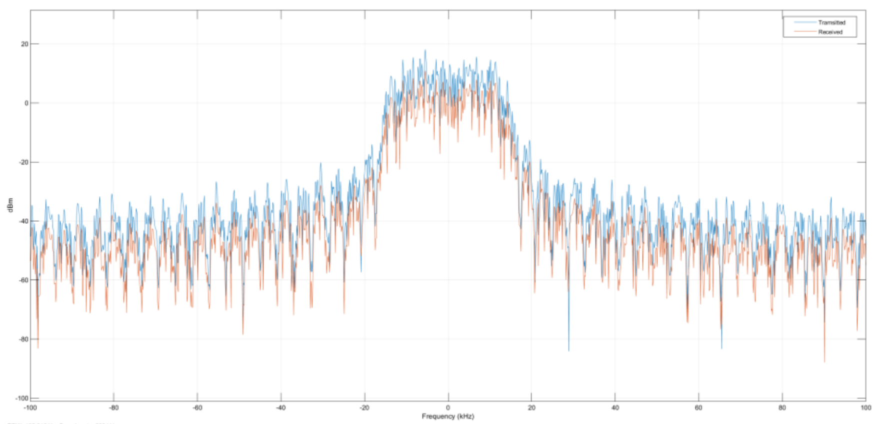


Figure: 5Hz frequency offset

Power Spectrum

The transmitted and received power spectrum is shown below. It is of not much importance quantitatively.



Conclusion

In this simulation-oriented project, we got familiar with the satellite downlink transmitter and receiver sides. The impairments that could occur and their possible remedies were practiced using Simulink blocks. Most of the block were built-in, we created one custom block to calculate antenna gain from antenna diameter. This project is a hands-on experience on satellite communication, modulation techniques, channel design and impairment handling.

References:

1. https://www.researchgate.net/publication/326294872_Modelling_and_Performance_Analysis_of_RF_Satellite_Link_System_Using_16_QAM
2. https://www.mathworks.com/help/comm/ug/add-saleh-mode-of-memoryless-nonlinearity-to-16qam-signal.html
3. Free Space Path loss - https://en.wikipedia.org/wiki/Free-space_path_loss
4. **Thermal Noise Formulas & Calculator -** https://www.electronics-notes.com/articles/basic_concepts/electronic-rf-noise/thermal-noise-calculations-calculator-formulas.php
5. **Phase Noise -** https://en.wikipedia.org/wiki/Phase_noise
6. **IQ Imbalance -** https://en.wikipedia.org/wiki/IQ_imbalance
7. **Constellation diagram -** https://youtu.be/Zh7qRwt2dOo