# REVIEW OF EFFICIENT COMBINATION OF DIGITAL MODULATION TECHNIQUES AND CODING SCHEMES FOR OPTIMIZED PERFORMANCE IN SATELLITE COMMUNICATION


**BY**

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## CERTIFICATON

This project work conducted by Musa Abdullahi Usman has been read and approved by the undersigned as having fulfilled part of the requirements for the award of post Graduate Diploma in Electronics and Communication in Physics Department of Nasarawa State University Keffi.

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## DECLERATION

I hereby declare that this project has been written by me and it is a report of my research work. It has not been presented in any previous application for state diploma or degree. All quotations are indicated and sources of information specifically acknowledged by means of references.

Sign

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## ABSTRACT

Satellite system was traditionally believed to be power – limited on its downlink channels. However, today's increase in bandwidth – demanding application have also makes satellite system to be bandwidth – limited (spectrum). The fundamental strive for efficiency faced by design engineers is further reinforced by the challenge of striking balance between power and bandwidth which are the two critical communication resources of meeting the system performance requirements without severe financial implications. This paper expresses methodologies for making the most efficient combination of coding scheme and digital modulation technique under bandwidth – limited is and power – limited conditions in satellite link capacity. One of the method adopted in proving this solution is to poise a trade – off of the probability of error ($P_E$) and the ratio of bit energy to noise-power spectral density ($E_b/N_O$). Matlab was used to solve the huge mathematical complexities therein. The implementation of this mechanism in satellite industry will yield great financial saving whose second order effects are keeping satellite services providers in business, meet customers' demand and optimize satellite systems operations.

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# CHAPTER ONE

# INTRODUCTION

## 1.1 BACKGOUND OF THE STUDY

Satellite is one of the greatest means of communication carrying a chuck of and data streams from one part of the horizon to other as compared to other medium. In this mother age, communication satellite network are an indispensable part of the major telecommunication system. Satellite inter-connects one nodes and provides same better advantages in application than the traditional communication systems such as combine massive data connections, mobile communication and direct communication for last-mile users, television and other broadcasting for pubic. To provides the optimum quality service (QoS) different types of design techniques need to be consider for deferent purpose like distinct types of modulation and coding f(channel and service ) techniques b are used for specific purpose, link budget calculation, selection of radio frequency, (RF) e c t. Other dominating factors are: permitted earth station size and complexity, the size and the shape of the service area e t c. (Margaret2012)

The satellite communication market has recorded huge successes for decades due to the regular and consistent patronage from two measure in branches of the industry; telecommunication and broadcasting industries. The two measure resource in communication systems industry are the transmit power and channels bandwidth, hence, efficiency of these becomes a highly significant factor (Adeyem, 2013)

Insufficiency in the commercial viable frequency spectrum recourse give rise to high cost of operating satellite system, this research will therefore focused on providing steps(methodologies) using basic coded modulation scheme, to achieve a compromise between bandwidth efficient and power efficient communication techniques that justify economics and technical basic (Adeyemi, 2013)

## 1.2 PROBLEM STATEMEN

Due to increase in bandwidth-ding applications, making scarcity in commercially viable frequency spectrum resource to grow. There by becomes bandwidth-limited (spectrum). This contribute to a service financial implication in communication system industry due to high operational cost while utilizing the frequency spectrum resource coupled with the power-limited sources on the downlink channel of the satellite system.

Therefore, to meet the system performance requirement without high cost operating, a strive for efficiency on the cause of striking and balance between the power and available bandwidth, which are the two critical communication resources could proper solution the challenge.

## 1.3 AIM AND OBJECTIVE

### 1.3.1 Aim of the student

The aim of the student is to review the efficient utilization of digital modulation and coding scheme for optimized performance in satellite communication

### 1.3.2 Objectives of the study

i. To provide methodologies for efficient combination of coding scheme and digital modulation technique under bandwidth-limited and power –limited condition in satellite link capacity.

ii. The strike balance between the power and bandwidth which are the two major communication resources towards meeting the system performance.

iii. To provide mechanism that will yield great financial in satellite industry while meeting customer’s demands.

## SCOPE AND LIMITATION

This work focused basically on the efficiency, power limited, bandwidth –limited (spectrum), coding and digital modulation in satellite link system. The following are the basic tools consider and analyzed, Basic satellite link budget channel capacity and M-ray signaling.

## SIGNIFICANCE OF THE STUDY

Due to growing demand from both broadcasting and telecommunication industries, operation under efficient combination of coding scheme and digital modulation technique by utilizing the available limited power and bandwidth will go a long way in boosting the tremendous successes recorded so far in the satellite communication sector by yielding great financial saving whose second order effect are keeping the satellite service provide in business, meeting the customer’ demands while optimizing the satellite system operations.

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# CHAPTER TWO

# LITERATURE REVIEW

## 2.1 COMMUNICATION

Communication is the basic process of exchanging. The basic components of electronic communication system are:

1) Transmitter
2) Communication channel
3) Receiver

A **Transmitter** is a collection circuits designed to convert the information in to a signal suitable for transmission over a given communication

A Receiver is a collection of a electronic circuits designed to convert the signal back to the original information.

The communication channel is the physical medium by which the electronic signal is transmitted from one place to another.

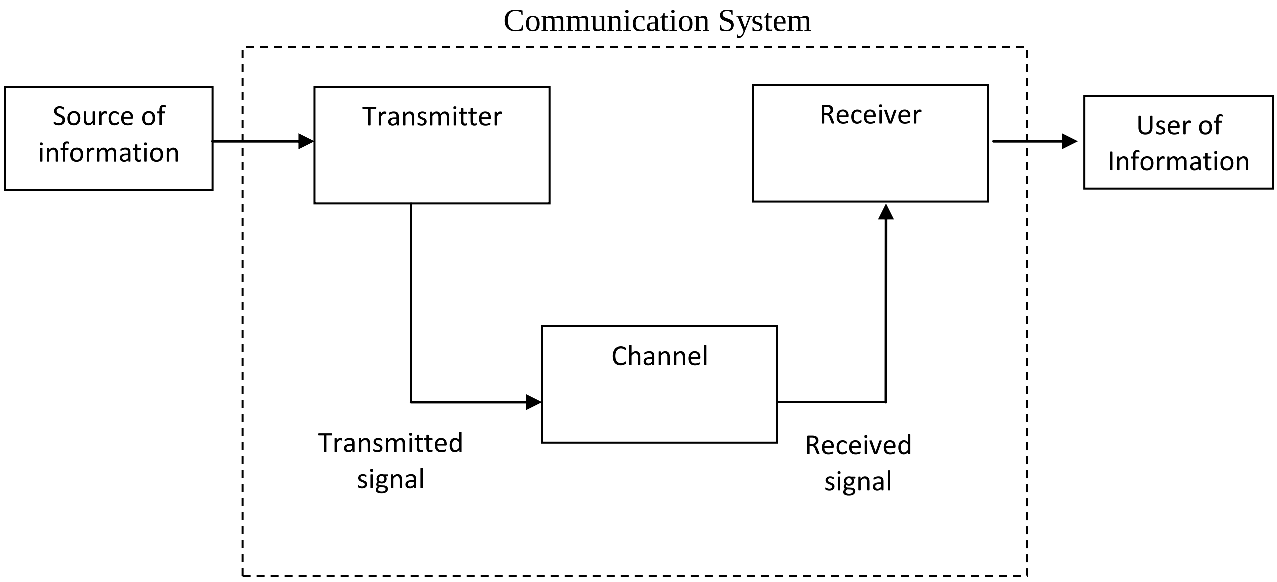


**Figure: 2.1: A Communication System (Umar, 2016)**

The communication channel is the central to operation of the communication system.

There are two types of communication channels:

- Linear (e.g. mobile radio) or nonlinear (e.g. satellite)
- Time invariant (e.g fiber) or time varying (e.g. mobile radio)

The information-carrying capacity of a communication system is proportional to the channel bandwidth. In pursuit for wider bandwidth, we have:

- Copper wire : 1MHz
- Coaxial caplet: 100MHz
- Microwave: GHz
- Optical fiber: THz

They use light as signal carrier; and have highest capacity among all practical signals. (Umar, 2019)

### 2.1.1 Type of Communication System

Communication system may be categorized based on their infrastructure and specifications of the signals they transmit. The infrastructure pertains to the channel used and the hardware design of the transmitting and receiving equipment. The signal specifications signify the nature and the type of the transmitter signal.

The type of communication based on their infrastructure and the signal specifications are discussed below:

***Based on physical infrastructure***

Based on physical structure, there are types of communication systems:

– Line communication systems

– Radio communication systems

Line communication system is the physical link called hardware channel between the transmitter and the receiver in line communication systems.

Radio communication system, there is no physical link, and natural resources such as space and water are used as software channels.

A system can be one or two ways transmission system. For instance, consider a TV system in which a user can receive the signals and view available channels. A TV receiver cannot transmit the signal. Thus, T V transmission is referred to as Simplex in technical terms.

A tow ways transmission, consider telephony as another example, in this case, one can simultaneously send and receive signals, hence in technical term this is called Duplex. As rule, a communication system can be a simplex or duplex but not both.

Therefore baseband on the physical of a communication system, you can define two group only one specification from each group is required to decide the type of communication system.

These are:

- Line/ radio communication
- Simplex/duplex communication

For instance, a TV communication system is a combination of the radio and simplex communication systems and landline telephony is a combination of the line and radio communication system. However, overseas or long distance telephony is carried out through satellite and the system is called radiotelephony as it makes use of radio waves for transmission and reception.

***Based on signal specification***

The signal specification is used to decide the types of communication system include:

- Nature of the baseband of information signal
- Nature of the transmitted signal

Based on the nature of the baseband signal, there are two types of communication systems:

- Analog and digital communication systems

Based on the transmitted signal can either be transmitted as it is without modulation or through a carrier signal with modulation.

The two systems can be categorized as:

- Baseband communication system
- Carrier communication system

Therefore, the four types of communication system categorized based on the signal specification are:

- Analog communication system
- Digital communication system
- Baseband communication system
- Carrier communication system

Of the four, at least two types are required to specify a particular communication system:

- Analog/digital communication system
- Baseband/carrier communication system

A particular communication system is either an analog or a digital communication system at a time. For instance, TV transmission is an analog communication system while High Definition Television (HDTV) is a digital communication system. The internet is another example of a digital communication system.

Similarly, a particular communication is either a baseband or carrier communication system. Examples of baseband communication system are landline telephony and fax. Examples of carrier communications systems are TV transmissions, radio broadcast and cable TV.

Consider a television communication system it is a radio-simplex-analog-carrier communication system while telephony is landline-duplex-analog baseband communication system.

## 2.2 DIGITAL COMMUNICATION

The Term Digital Communication covers a broad area of communication techniques, including digital transmissions and digital radio. Digital transmission of digital pulses between two or more points in a communication system. Digital radio is the transmission of digital modulated analog carriers between two or more points in a communication system.

Digital signals are discrete in the time and value. Digital signals are signals that represented by binary numbers “1” or “0”. They are sampled, quantized and encoded version of continuous time signals which they represent.

(i) Advantages of digital communication system

- Reliable communication: less sensitivity to changes in environmental conditions.
- Easy multiplexing
- Easy processing like encryption and compression

- Easy system performance monitoring QOS monitoring
- Integration of transmission and switching
- Signal regeneration operation, operation at low SNR, superior performance.
- Integration of services leading to ISDN (integration services Digital Network )

(ii) Disadvantage of Digital communication system

- Increased bandwidth: 64KB for a $4H_z$ charnel, without compression
- Need for precision timing: bit, character, frame synchronization needed.
- Analogue to digital, digital to analogue conversions: very often nonlinear ABC and DAC used, some performance degration
- Higher complexity (umar, 2016 ).

(iii) The basic model of digital communication system:

**Format:** Transform the source information to digital symbol

**Modulation:** The process by which symbol are converted to waveform

**Source encode:** using A/D conversion and remove unwanted information

**Encrypt:** prevent unauthorized users from understanding the message and from injecting false information into the system.

**Channel encoding:** reduce the pe and SNR ( pay of the expenses on BN and complicity)

**Frequency spread:** produce a signal that is less vulnerable to interference and provides the private of the communication.

**Multiple Access:** combination signal that might have different characteristics or may originates from different sources so they may share one or more communication sources.

(iv) Applications of digital communication system

- Terrestrial microwave telecommunication e.g. the use of optical fibre, QAM-256 for bandwidth-efficient modulation.
- Satellite system e.g QPSK, proposed modulation coded 8-psk or 16-QAM
- Bradcast system
- Private mobile radio. (Umar, 2016)

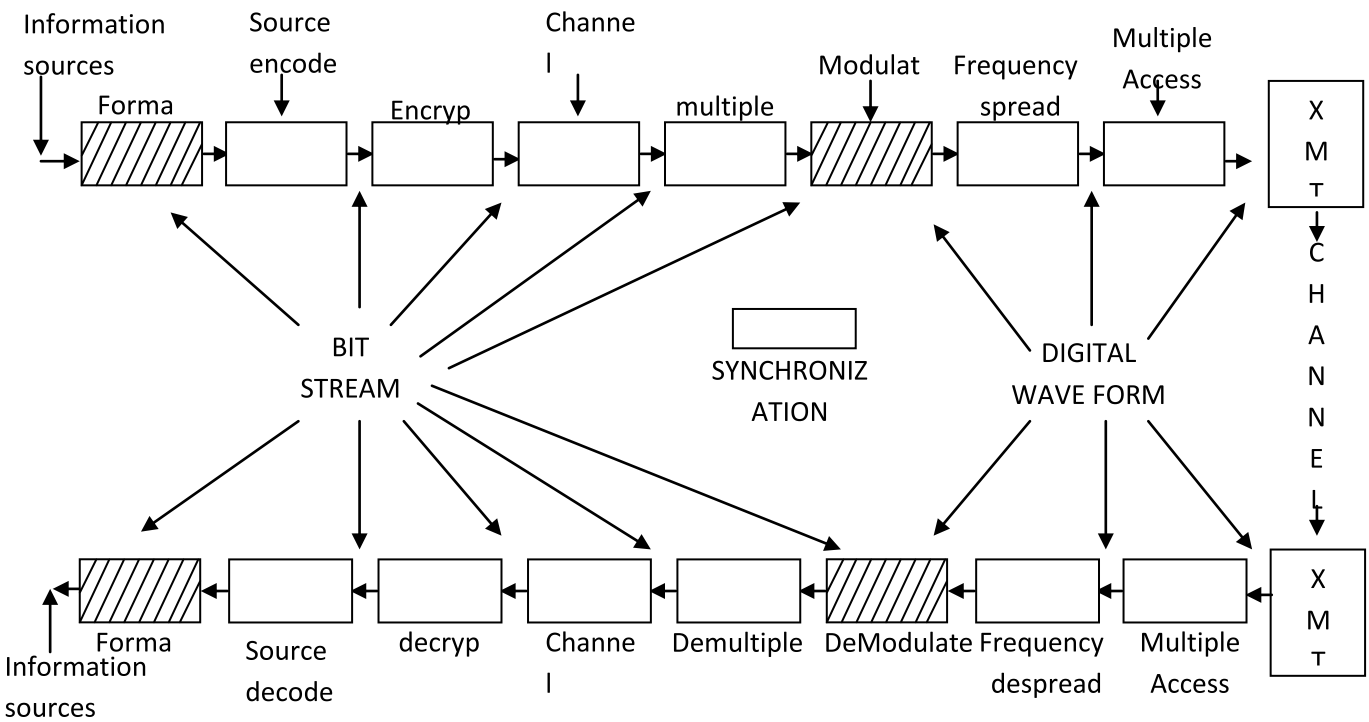


**Figure 2.2: Basic Model of Digital Communication System (Umar, 2016)**

## 2.3 SATELLITE COMMUNICATION SYSTEM

In generation, a satellite is anything that orbits something as for example, the moon orbit the earth. In a communication context, a satellite is a specialized wireless receiver/transmitter that is launched by a rocket and place in orbit around the earth.

### 2.3.1 Type of satellite Systems

There are three type of communication satellite system. They are categorized based on the or according to the type of orbit they follow:

(a) Geostationary satellite

Orbit the earth directly over the equator approximately 22000miles up. At this altitude, one complete trip round the earth (relative to the sum) take 24hours. Thus satellite remain over the same spot on the earth's surface at all time.A geostationary satellite can be accessed using a disk antenna aimed at the spot in the sky where the satellite hovers.

Geostationary orbit only one exist and is circular in nature, it lies in the earth's equatorial plane and so all geostationary satellites have zero inclination. Its widely used because earth station antenna do not have to trace them. (kwaha, 2016)

(b) A low-earth-orbit (LEO) satellite system

It employ a large fleet of "bird" each in a circular orbit at a constant altitude of a few hundred miles. The orbit take the satellite over or nearly over the geographical poles. Each revolution take approximately 90minutes to a few hours. The fleet is arrange in such a way that, from any point on the surface at anytime, at least one satellite is on a line of sight.

The entire system operates in such a manner similar to the way a cellular telephone functions. The main difference is that the transponders or wireless receiver/transmitter, are moving rather than fixed and are in space rather than on the earth. (kwaha, 2016).

(c) Elliptical orbit satellites

These revolve around the earth in the elliptical orbit. These satellite move rapidly when they are near perigee, or their lowest altitude; they move slowly when they are near apogee or their highest altitude. Such "birds" are used by amateur radio operators and by some commercial and government services. They require directional antennas whose orientation must be constantly adjusted to follow the satellite path across the sky. (Margaret, 2012)

## 2.4 MODULATION

### 2.4.1 Definition:

Modulation is superimposing of information contents of a modulation signal on a carrier signal (which is of high frequency) by varying the characteristic of carrier signal according to the modulating signal (Umar, 2016). It can be said to be a process in which the baseband signal modifies another high-frequency signal called the carrier (Kwaha, 2016).

### 2.4.2 Types of Modulation

Modulation can be among and can be digital

***Analog Modulation***

Analog modulation is the process of converting an analog input signal into a signal that is suitable for RF transmission. The analog carrier signal is modulated by analog information signal so that information bearing analog signal can travel larger distance without the fear of loss due to absorption.

The analog modulation is of two types:

(i) Amplitude Modulation

In this type of modulation the strength of the carrier signal is varied with the modulating signal.

(ii) Angle modulation

This further classified as Frequency Modulation (FM) and phase Modulation (PM).

**Frequency Modulation:** in this type of modulation, the frequency of the carrier signal is varied with the modulation signal.

**Phase Modulation:** in this type of modulation the phase of the carrier signal is varied with the modulating signal. It is the variant of the frequency modulation.

The analog carrier signal is modulated by digital information signal, it is also considered as digital to analog conversion.

***Digital Modulation***

This means an analog signal of carrier is converted by a digital data bit stream

There are two types of bits in binary, these are

- Logic 0 (Low)

- Logic 1 (high)

This method is use to convert signal to analog and the responding demodulating is applied to convert analog signal to digital signal. Here the analog signal bearing information is transmitted by digital method.

### 2.4.3 Basic Modulation Technique

The choice for digital modulations basically revolve round tremendous merits of digital system which include lower design costs, increase system capacity using more efficient schemes; robust error control coding techniques to combat noise and interference. A typical carrier signal S(t)can be represented as

$$S\ (t)=A\cos\ (2\pi ftŧ\varphi) \qquad (2.1)$$

Where A is the amplitude, f is the carrier frequency and …… is the signal phase. Carrier signal is characterized by these three parameters. Modulation is simply varying (Modulating) one or two of the three key parameters of electromagnetic wave in accordance with the data to be sent. In digital modulations, switching (keying) the amplitude, frequency or phase of the carrier in accordance with the baseband signal (data) before transmission over the satellite link amount to amplitude shift keying ASK, frequency shift keying (FSK) and phase shift keying (PSK) respectively (Adeyemi,2013).

The basic goal of modulation is to squeeze as much data into the least amount of spectrum possible. The objective known as the spectral efficiency, measures how quickly data can be transmitted in an assigned bandwidth. The unit of the measurement is bit per second per Herz (bps/Hz). (Frenzel, 2012)

Among the techniques that emerged to achieve and improve spectral efficiency are:

### *Amplitude Shift Keying (ASK) and Frequency Shift Keying (FSK)*

There are three basic ways to modulate a sine wave carrier; modifying the amplitude, frequency or phase. More sophisticated methods combines two or more of these variations to improve the spectral efficiency. These basic modulation forms are still in used today (Frenzel, 2012).

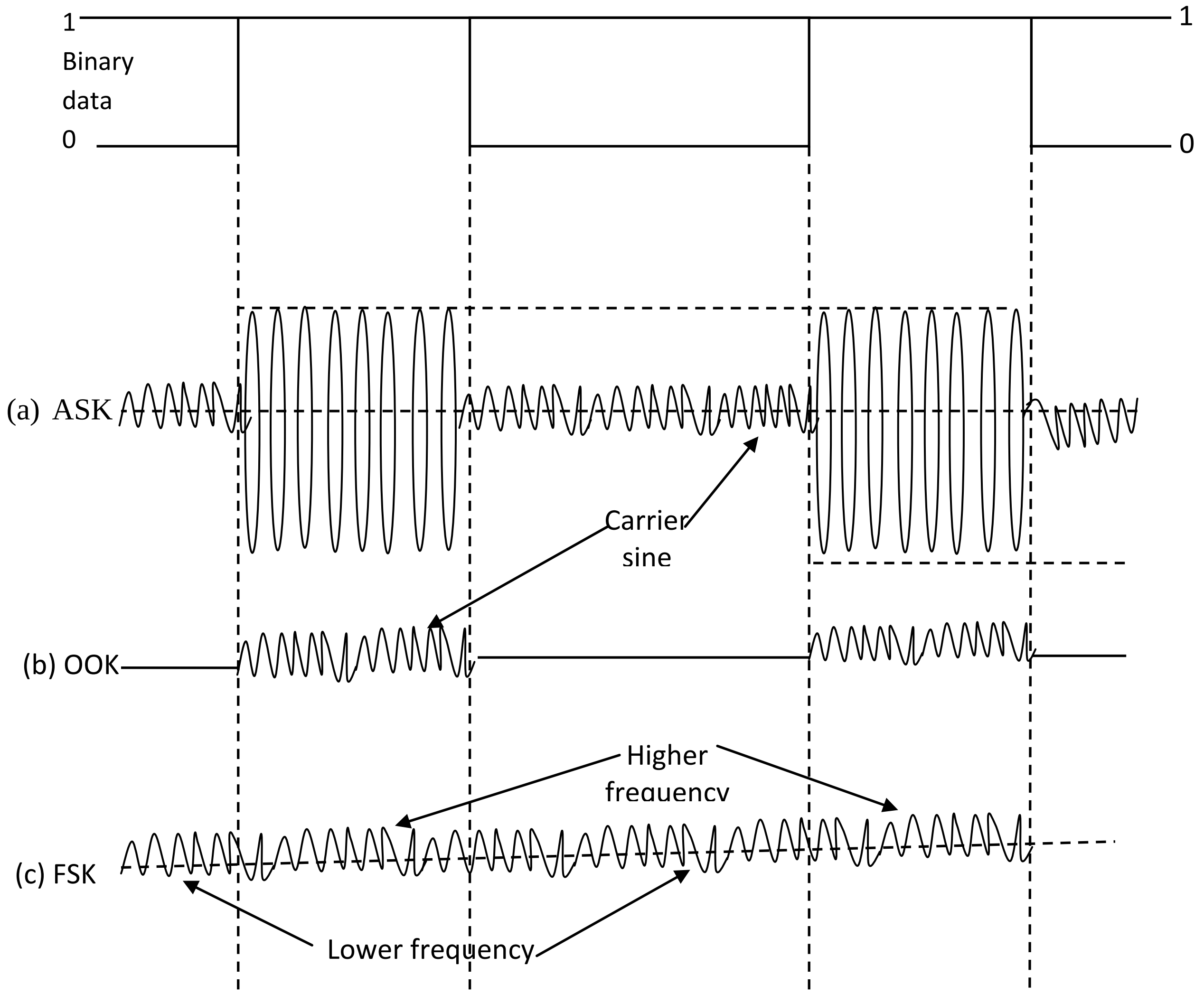


**Figure 2.3: Three basic digital modulation formats are still very popular with Low-data-rate short- range wireless application: ASK (i) On-Off shift keying (OOK) (ii) and Frequency Shift Keying FSK (iii). These waveforms are coherent as the binary state change occurs at carrier zero crossing point (Frenzel, 2012).**

Figure 2.3 shows a basic serial digital signal of binary zeros and ones the corresponding AM and FM signals resulting from modulation. There are two types of AM signal: On-off keying (OOK) and amplitude shift keying (ASK). In figure 2.3 (i), the carrier ampliittude is shifted between two amplitude levels too produce ASK. In figure 2.3 (ii) the binary signal turns the carrier off and on to create OOK. AM produces sidebands above and below the carrier equal to the highest frequency content including any harmonics for binary pulse modulating signals.

Frequency shift keying (FSK), shift the carrier between two different frequencies called the mark and space frequencies or fm and fs (figure 2.3 iii). FM produces multiple sideband frequencies above and below the carrier frequency. The bandwith produce is a function of the highest modulating frequency including harmonics and the modulation index, which is:

$$M = \Delta f\,(T)$$

(2.2)

$\Delta f$ is the frequency deviation or shift between the mark and space frequencies, or:

$\Delta f = fs - fm$

T is the bit interval of the data or the reciprocal of the data (1 per bit/s).

The smaller values of m produce fewer sidebands. A popular version of FSK called minimum shift keying (MSK) specifies m = 0.5. smaller values are used such as m = 0.3. there are two ways to further improve the spectral efficiency for both ASK and FSK. Frist select data rate, carrier frequencies and shift frequencies so as there are no discontinuities in the sine carrier when changing from binary state to another. These discontinuities produce glitches that increase the harmonic content and bandwith. The idea is to synchronize the stop and times of the binary data with when the sign carrier is transitional in amplitude or frequency at the zero crossing points. This is called continuous phase or coherent operation.

Both coherent ASK/OOK and coherent FSK have fewer harmonic and a narrower band with than non-coherent signals.

A second technique is to filter the binary data prior to modulation. This round the signal off, lengthening the rise and fall times and reducing the harmonic content. A special Gaussian and rise cosine low pass filters are used for this purpose (Frenzel, 2012)

***Binary phase shift keying (BPSK) AND Quadrature phase keying (QPSK)***

A very popular digital modulation scheme, binary phase keying (BPSK), shift the carrier sine wave $180^0$ for each change in binary state (figure 2.4). BPSK is coherent as the phase transaction occurs at zero crossing point. The proper demodulation of BPSK requires the signal to be compared to a sine carrier of the same phase. This involves carrier recovery and other complex circuitry, (Frenzel, 2012).

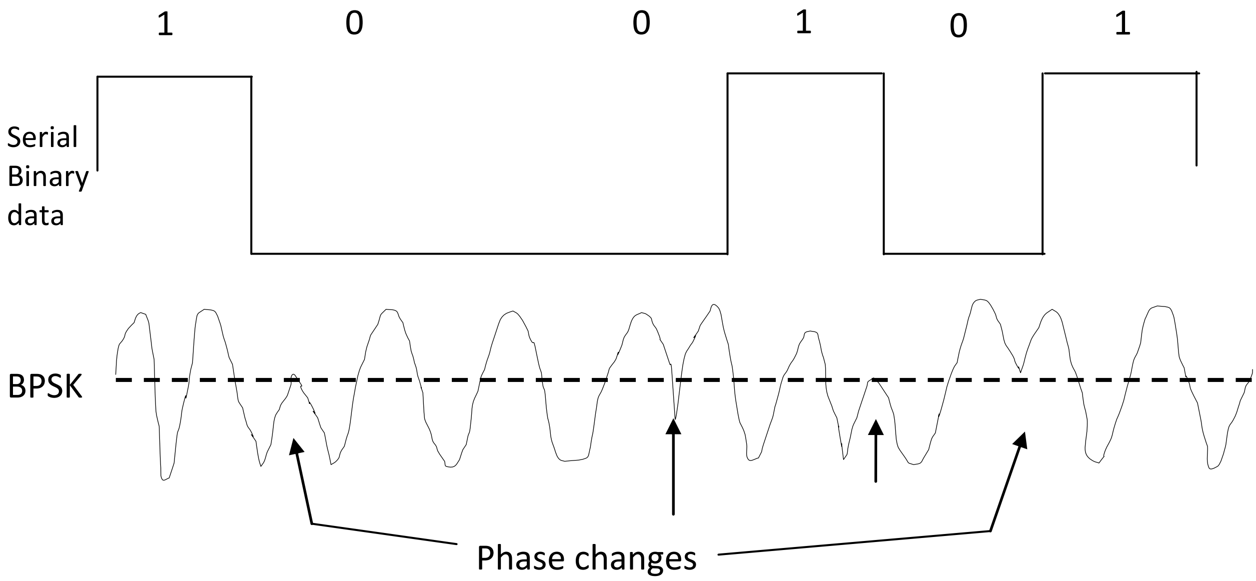


Figure 2.4: In Binary Phase keying. Note how a binary 0 is $0^0$ while a Binary 1 is $180^0$.

The phase changes when the binary state switches so the signal is coherent. (Frenzel, 2012).

A simpler version is differential BPSK or DPSK where the received bit phase is compared to the previous bit signal. BPSK is very spectrally efficient in that you can transmit at a data rate equal to the bandwidth or 1 bit per $H_z$.

In a popular variation of BPSK, quadrature PSK (QPSK), the modulator produces two sine carrier $90^0$ apart. The binary data modulates each phase, producing four unique sine signals shifted by $45^0$ from one another. The places are added together to produce the final signals. Each unique pair of the bits generates a carrier with a different phase (table 2.1).

Table 2.1: carrier phase shift foe each pair of bits represented (Frenzel, 2012)

| Bit pairs | Phase (degree) |
|---|---|
| 00 | 45 |
| 01 | 135 |
| 11 | 225 |
| 10 | 315 |

Figure 2.5 illustrates QPRS with the phasor diagram where the phasor represents the carrier sine amplitude peak and its position indicates a phase. A constellation diagram in 2.5 (ii) shows same information. QPSK is very spectrally efficient since carrier phase represents two bits of data. The spectral efficiency is 2bit per $H_z$, meaning twice the data rate can be achieved in the same bandwidth as BPSK.

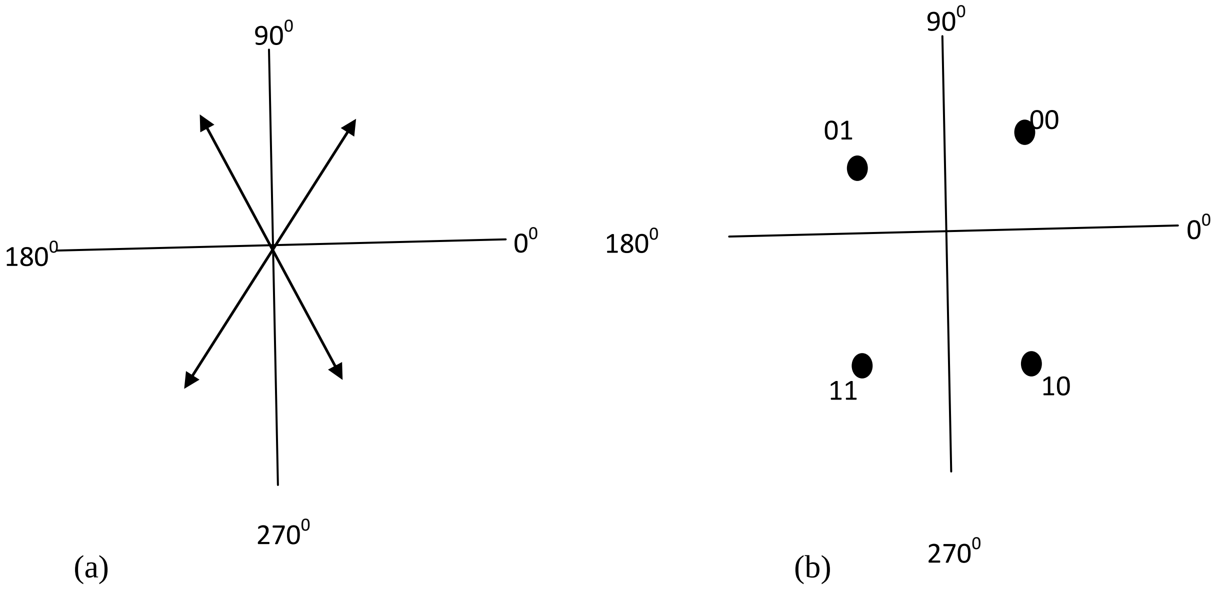


**Figure 2.5: modulation can be represented without time domain waveforms. For example, QPSK can be represented with a phasor diagram (a) or a constellation diagram (b) both of which indicate phase amplitude magnitudes. (Frenzel, 2012).**

***Data Rate and Baud Rate***

The maximum theoretical data rate or channel capacity (C) in bit/s is a function of the channel bandwidth (B) channel in Hz and the signal –to noise ratio (SNR).

$$C = b\log_2 (1+SNR) \qquad (2.3)$$

This is called the Shannon- Hartley law. The maximum data rate for a given bit error rate (BER) (Adeyemi, 2013).

However, if a signal is corrupted by noise, one may think that amplifying the signal further will reduce the noise. No, because while amplify the signal, the noise is amplified (Kwaha, 2016).

Another key factor is the baudrate or the number of modulation symbols transmitted per second. It can be amplitude, a frequency, a phase or some communication of them. Basic binary transmission uses one bit per symbol.

In ASK, a binary 0 is one amplitude and a binary is another amplitude. In FSK, a binary 0 is one carrier frequency and binary 1 is another frequency. BPSK uses $0^0$ shift for a binary 0 and $180^0$ shift for a binary 1. In each of these cases there is one bit per symbol.

Date rate in bit/s is calculated as the reciprocal of the bit time ($t_b$):

$$\text{Bits/s} = 1/\, t_b \qquad (2.4)$$

With symbol bit, rate is the same as the bit rate. However, if you transmit more bit per symbol, the baud rate is slower than the bit rate by a factor equal to the number of bits per symbol. For example, if 2 bits per symbol are transmitted, the baud rate is the bit rate divided by 22. For instance, with QPSK a 70Mb/s data stream is transmitted at baud rate of 35symbols per second.

### *Multiple phase shift keying (M-PSK)*

QPSK produces two bits per symbol, making it very spectrally efficient. QPSK can be referred to as 4-PSK because there are four amplitudes phase combinations. By using smaller phase shifts, more bits can be transmitted per symbol. Some popular variations are 8- PSK and 16-PSK.

8-PSK uses eight symbols with constant carrier amplitude 45shifts between them, enabling three bits to be transmitted for each symbol. 116-PSK uses $225^0$ shifts of constant amplitude carrier signals. This arrangements results in a transmission of 4 bits per symbol.

While multiple phase shifts keying (M-PSK) is much more spectrally efficient, the greater the number of smaller phase shifts, the more difficult the signal is to demodulate in the present of noise. The benefit of M-PSK is that the constants carrier amplitude means more efficient non-linear power amplification can be used. (Frenzel, 2012)

### *Quadrature Amplitude Modulation (QAM)*

The creation of symbols that are some combination of amplitude and phase can carry the concept of transmitting more bits per symbol further. This method is called quadrature amplitude modulation (QAM). For example, 8QAM uses four carrier phases plus two amplitude levels to transmit 3bits per symbol. Other popular variations are 16QAM and 256QAM, which transmit 4, 6 and 8 bits per symbol respectively.

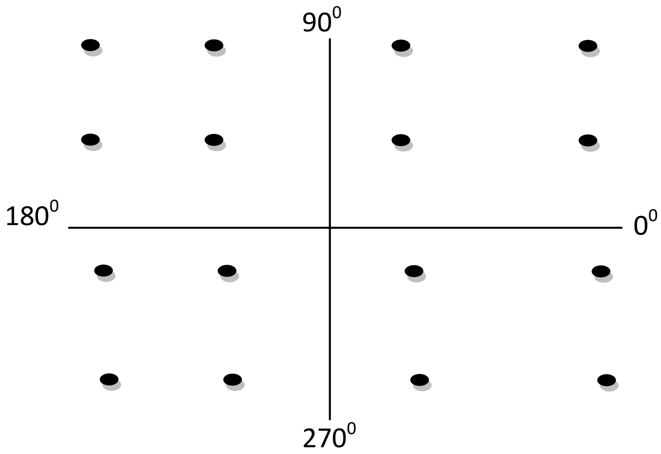


Figure 2.6: 16QAM uses a mix of amplitudes and phases to achieve 4 bits/$H_z$. in this example there are 3 amplitudes and 12 phase shifts (Frenzel, 2012).

While QAM is enormously efficient of spectrum, it is more difficult to demodulate in the presence of noise, which is mostly random amplitude variation. Linear power amplification is required. QAM is noise, which is mostly random amplitude variation. Linear power amplification is required.QAM is very widely used in cable TV, wi-fi wireless local-area networks (LANs), satellites and cellular telephone systems to produce maximum data rate in limited bandwidth

***Amplitude Phase Shift Keying (APSK).***

Amplitude phase shift keying (APSK), a variations of both M-PSK and QAM was created in response to need for an improved QAM. Higher level of QAM such as 16QAM and above have many different amplitude levels as well as phase shifts. These amplitude levels are susceptible to noise.

Furthermore, these multiple level require linear power amplifier (PAs) that less efficient than non-linear (e.g. Class C). The fewer the number of amplitude levels of the smaller the difference between the amplitude levels, the greater the chance to operate in nonlinear region of the PA to boost power level.

APSK uses fewer amplitude levels, it essentially arranges the symbols into two or more concentric rings with constant phase off-set ___. For example, 16 APSK uses double –ring PSK format (Fig 2.7). This is called 4-12 16APSK with four symbols in the center ring and 12 in the outer ring.

(Frenzel, 2012).

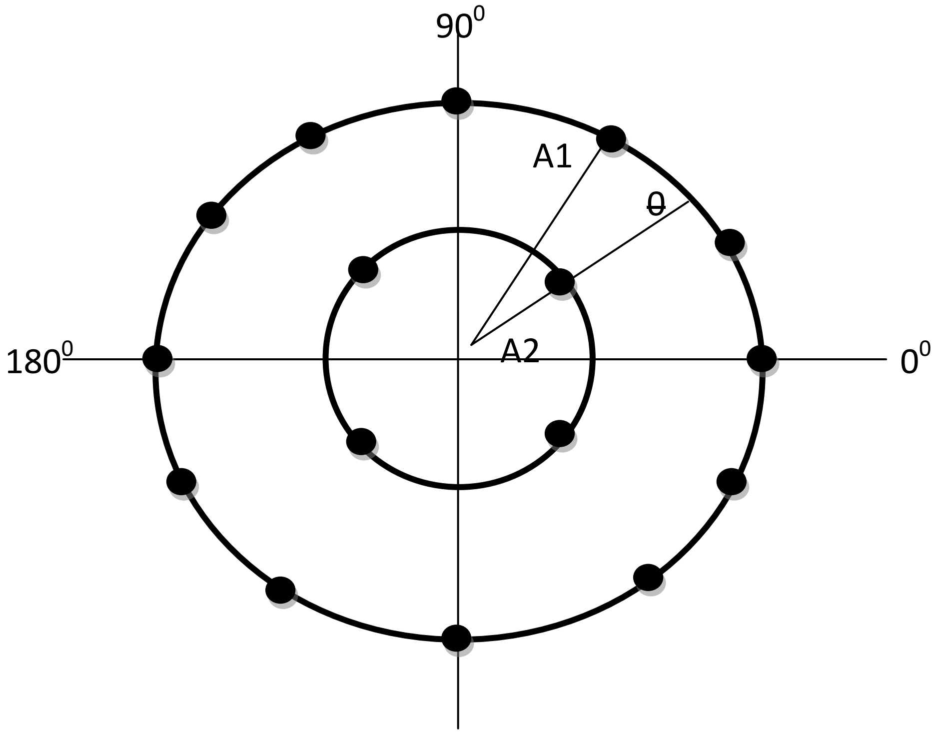


Figure 2.7: 16APSK uses two amplitude levels, A1 and A2, plus 16 different phase positions with an offset of $\theta$. This technique is widely used in satellite, (Frenzel, 2012).

Two close amplitude levels allow the amplifier to operate closer to the non linear, improving efficiency as well as well as power output. APSK is used primarily in satellite since it is a good fit with the popular travelling wave tube (TWT) PAs.

Other modulation systems use in satellite communications are:

## 2.5 M-ARRY SIGNALLING

In an M-arry signaling scheme, each of the possible M signals, known as a symbol comprising m bits, is mapped to signal waveform during transmission of interval Ts (adeyemi, 2013).the transmitted wave thereby represents a symbol of mbits with Ts duration. M and m are related as

$$M= 2^{m} \text{ or } m =\log_2 M \qquad (2.5)$$

Where m is an integer representing the number of bits per symbol.for data rate R bps, the bit duration Tb becomes

$$Tb=1/R \text{ sec,} \qquad (2.6)$$

It follows that

$$Ts =mTb \text{ sec,} \qquad (2.7)$$

Where Ts is the symbol duration.

From equation 2.6 and 2.7 ,we have

$$R=m/Ts \text{ bps,} \qquad (2.8)$$

An expression relating the symbol rate Rs with the data R is

$Rs = R/Log_2 M$ bps, (2.9)

The minimum required band width for MPSK are given as

$B_{MPSK} = 1/T = Rs$ Hz, (2.10)

$B_{MPSK} = 1/T = MRs$ Hz, (2.11)

When a digital scheme transmit m bits over a period of Ts occupying B (Hz) bandwidth, its bandwidth efficiency p becomes

$p_{MPSK} = R/B \;\; Log_2 M/BT = 1/BT$ bps,/Hz (2.12)

where Tb is the bit duration in seconds. For M-PSK and M-QAM modulated signals, the bandwidth efficiency using the Nyquist filtering at the baseband where the B is equal to Rs, p is given as

$p = R/B \;\; log_2 M$ bps,

where bandwidth efficiency increases as M increases, hence, M-PSK and M-QAM are said to be bandwidth- efficiency. Contrary to MQAM modulations, in MFSK modulations, the bandwidth efficiency, p is given as

$p_{MFSK}$ = log2 M/M bps/Hz (2.13)

Bandwidth efficiency decreases as M increases, hence, MFSK modulations scheme is not a bandwidth efficient but bandwidth expensive.

At the receiving end, the demodulator is expected to make a correct decision on the received $S/N_0$ waveform based on predetermined conditions. The probability that it will 0) make a symbol error $P_E$ (M) is

$$P_E(M) \cong 2Q\left[\frac{\sqrt{2Es}}{No}\sin\left(\frac{\pi}{M}\right)\right] \quad (2.15)$$

Where Q (x) is the Q – function sometimes called the complementary error function equation 2.15 is valid for the MPSK modulation techniques. For MPSK signaling system, symbol error probability $P_E$ is related to bit error probability $P_B$ by

$$PB \cong \frac{PE}{log2M} = PE/m \quad (2.16)$$

For MPSK, the symbol error probability is given as

$$P_E(M) \leq M - \frac{1}{2}^{m-1}\ {}^{1}/_{2}{}^{m} - 1 \quad (2.17)$$

$P_E$ can be expressed in terms of $P_B$

$$P_B/\ P_E = 2^{m-1}/_{2}{}^{m} - 1 \quad (2.18)$$

Coding gain G in dB, is the measure of power saving between when a system is coded and when un-coded, given the same modulation and BER.

$$G\ (dB) = \left[\left(\frac{Eb}{No}\right) \text{uncoded} - \left(\frac{Eb}{No}\right)\text{coded}\right] \quad (2.19)$$

A channel encoder with code rate k/n, processes a data rate r to produces a channel rate Rc which is sent to the modulator which further yields a symbol rate Rs.

$$Rc = (m/k)R \text{ bps} \quad (2.20)$$

$$Rs = \frac{Rc}{logM} \quad (2.21)$$

Due to coding process, Pr/No arriving at demodulator's input is given by

$$\frac{Pr}{No} = \frac{Ec}{No} \mathrm{R} = \frac{Es}{No} R \text{ dB}, \qquad \text{Hz} \tag{2.22}$$

Where Ec and are the channel and symbol energy respectively.

The denominator yields channel error probability Pc given as

$$\mathrm{Pc} = \frac{PE}{m} \tag{2.23}$$

(Adeyemi, 2013)

## 2.6 CHANNEL CAPACITY

Channel capacity C defines the maximum number of bits per second (bps) that can be reliably sent over a channel (Gibson, 2002). The Shannon-Hartley theorem expressed channel capacity in the light of additive White Gaussian Noise (AWGN) as

$$C = B\log_2 (1 + SNR) \text{ bps},$$

Where C is the highest number of bit can be reliably sent/transmitted per second, SNR is the signal – to – noise, which is the ratio of the receiver power and noise power and B is the channel bandwith in Hz (Adeyemi, 2013). Equation is called Shannon-Hartley Law. The main significance of equation 2.25 is to take into cognizance cognizance that there is a limit of data rate that can be sent over a nosy channel. Equation 2.24 can also be written as:

$$C = B\log_2 \left(1 + \frac{Pr}{NoB}\right) \text{ bps}, \tag{2.25}$$

Where Pr is the average received power in watts (W) and No is the noise spectral density in watts per Hz (W/Hz). In digital communications, equation 2.25 can be expressed as

$$\frac{R}{B} = \log_2 \left(1 + \frac{Eb}{No} * \frac{R}{B}\right) \text{ bps}, \qquad (2.26)$$

Where R is data rate (bps), $E_b$ is the bit energy, such that a modulator expended an average power of $E_b / T_b$ watts on each bit, where $T_b$ is the bit duration. At the receiver, the ratio of the average received power to noise- power spectral density Pr/ No and the bit energy per noise power spectral density $E_b/N_o$ are related as follows

$$E_b/N_o = (Pr/No)(1/R) \text{ dB}, \qquad (2.27)$$

Where R is the data rate (bps) to band with ratio R/B is referred to as channel spectral efficiency or bandwidth efficiency. As band with B tends infinity or a spectral density tends zero, $E_b/N_o$ tends to $(\log_2 e)^{-1} = 0.693 = -1.6$ dB, which is called the shannon limit this is the value of $E_b / N_o$ below which system cannot operate at capacity. The spectral efficiency indicates the effective utilization of the bandwidth resource. If a message of data rate R (bps) is to be transmitted over channel capacity C (bps); if R< C, then small probability ($P_E$) can be assumed by implementing intelligent coding scheme. At R= c, the system is said to be boundary level, which is not recommended. When R > C, then no communication can be archived since the data rate is greater than the capacity, hence, it is described as unattainable region. A link is said to be power-limited if R> B, meaning the capacity can be increased by increasing the transmit power (Adeyemi, 2013).

Spectral efficiency and power efficiency are mathematically related to SNR as

SNR = ($E_b/N_o$) (R/B) (2.28)

Where $E_b/\ N_o$ is the link power efficiency and R/b is the special efficiency. (Adeyemi, 2013)

## 2.7 BASIC SATELLITE LINK BUDGET

Satellite link budget can simply be vied as a balanced sheet of power gains and losses in a space channel (propagating medium) between the satellite and the earth station. Link budget therefore becomes a basic tool for estimating system error performance ($P_E$), which is usually expressed as bit energy to noise spectral power $E_b/N_o$ in digital systems. The values of $E_b/\ N_o$ there should drive the choice of modulation scheme to be adopted in Gaussian noise environment. Since satellite link operate on line on site principles, free space path loss (FSLP) becomes mojor impairing parameter of consideration for signal transmission. The degrading effects of noise and rainfall warrant some reasonable link margin.

The generic satellite link operation is written as

$$Pr = PtGt\ Gr\ (\lambda/4\pi R)^2\ dB, \quad (2.29)$$

Where Pr is the received power in dB, Pt is the transmit power in dB, Gt is the transmit antenna gain in dB, Gr is the receiver antenna gain in dB, R is the distance between the transmitter and the receiver, $\lambda$ is the signal wavelength in meters. The squared terms is called the free space loss. Equation 2.29 can be expressed interms of noise spectral density.

$$Pr/\ KTs = PtGt/K\ (\frac{Gr}{Ts})\ (\frac{\lambda}{4\pi R})^2\ dB, \qquad Hz, \quad (2.30)$$

Where T is the receiver system temperature in K, K is the Boltzmann's constant (-228.6 dB W/ Hz), Gr/T is called the figure of merit and KTs represent the noise power spectral density $N_0$

$$Pr/N_o = Pr/KTs \quad (2.31)$$

(Adeyemi, 2013).

## 2.8 CODING FOR ERROR DETECTION AND CORRECTION

Environmental interference and physical defects in communication medium can cause random bit errors during data transmission. Error coding is a method of detecting and correcting these errors to ensure information is transferred intact from its source to its destination, (Charles, 1999).

Coding is therefore the process of redundant bits to the data bits before transmission, such that the redundant bits are used to correct he likely errors caused by the noisy channel (Adeyemi,2 013).

In analogue links, degradation takes the form of decrease in signal-to-noise ratio (SNR), while in digital systems; information degradation is measured in terms of bit error rate (BER) (Pratt et, a1, 2002).

One major advantage of digital systems has over analogues systems is the error detection and correction (EDC) mechanisms. The EDC techniques are used to improve the quality of digital signals.

The choice to adopt EDC technique is determined by system performance requirements and the nature of the transmission channel. EDC can either be implemented by requesting for retransmission of corrupted symbol using the Automated Request (ARQ) scheme or the receiver decides to detect and corrects the error without resource for retransmission using forward error correction (FEC) scheme. FEC scheme is usually adopted in real – time application and in satellite communications systems where bandwidth cost and latency are major concerns.

The transmission system of FEC scheme is traded for a complex receiving terminal. FEC coded are broadly categorized into block coding family comprising reed Solomon, hamming, BCH, cyclic and revolution coding family, (Adeyemi, 2013).

Error coding is used in many digital applications like computer memory, magnetic and optical data storage media, satellite and deep space communications, network communications and cellular telephone networks. Rather than transmitting digital data in raw bit to bit form, the data is encoded with extra bits at the source. The longer “the code war” is then transmitted and the receiver can then decode it to retrieve the desired information. The extra bits transforms the data into a valid code word in the coding scheme. The space of valid code words is smaller than the space of possible bit strings of that length; therefore the destination can recognize invalid code words. Figure 2.8 illustrates the code word space.

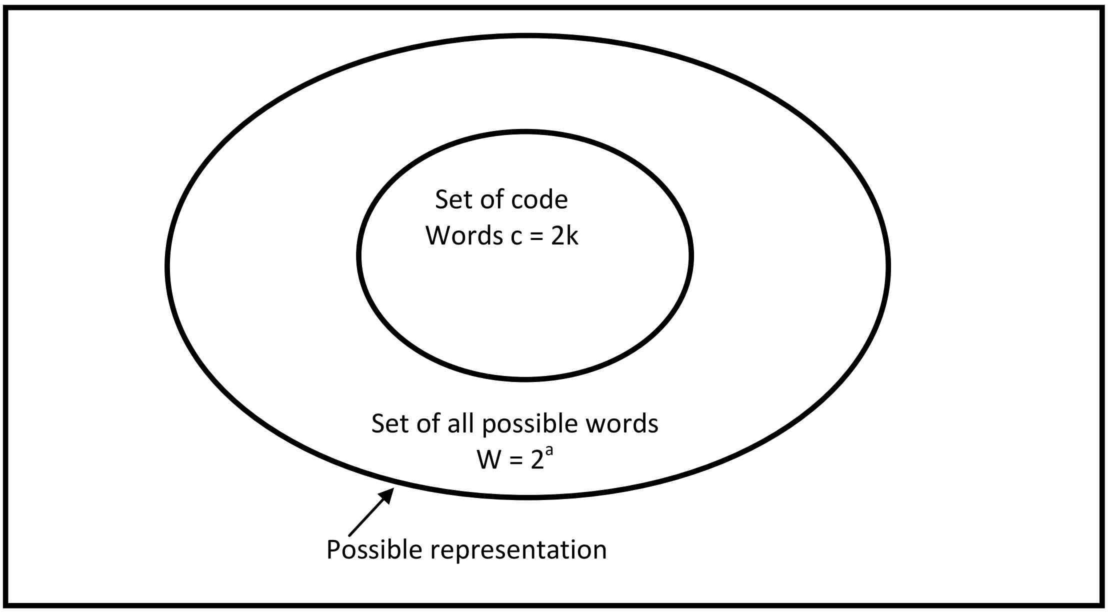


**Figure 2.8: code word space within all possible words, (Charles, 1999).**

If the errors are introduced during transmission, they will likely be detected during the decoding process at the destination because the cord word would be transformed into invalid bit string. Given a data string to be transmitted that is k bits long, there are 2k possible bit strings that the data can be. Error coding assumes the worst case scenario that the information to be encoded can be any of these bit strings. Therefore there will be 2k valid code words. The code word will be n bits long, where n>k. so just having extra bits in the data transmission eliminates many of the possible 2n bit strings as valid code words. (Charles, 1999)

### 2.8.1. Types of Error in Communication Channel

When choosing a coding scheme for error protection, the types of errors that tend to occur on the communication channel must be considered.

There are two types of errors that occur on a communication channel:

- Random bit errors
- Burst errors

A channel that usually has random bit errors will tend to have isolated bit Hips during data transmissions and the bit errors are independent of each other. A channel with burst errors will tend to have clumps of bit errors that occur during one transmission. Error codes have been developed to specifically protect against both random and burst errors.

### 2.8.2 Major Type of Coding Schemes

There are two major types of coding schemes:

- Linear block codes
- Convolutional codes.

The linear block codes are characterized by segmenting a message into separate blocks of a fixed length and encoding each block one at a time for transmission.

Convolution codes encode the entire data stream into one long code word and transmits and transmit it in pieces.

***Linear Block Codes***

Linear block codes are so named because each code word in the set is linear combination of a set of generator code words. If the messages are k bits long, and the code words are n bits long (where n>k) there are k linearly independent code words of length n that form a generator matrix: To encode any message of k bits, you simply multiply the message vector u by the generator matrix to produce a code word vector v that is n bits long.

Linear block codes are very easy to implement in '_hardware and since they are algebraically determined, they can be decoded in constant time. They have high code rates, usually above 0.9S.The have low coding overhead, but they have limited error correction capabilities. They are very useful in situations where the BER of the channel is relatively low, bandwidth availability is limited in the transmission and it is easy to retransmit data.

***Convolutional codes***

Are more difficult to decode because they are encoded using finite state machines that have branching paths for encoding each bit in the data sequence. A well-known process for decoding convolution codes quickly is the Viterbi Algorithm. The algorithm is a maximum likelihood decoder, meaning that the output code word from decoding a transmission is always the one with the highest probability of being the correct word transmitted from the source. The drivers of coding and modulation decisions are power-limited system and bandwidth-efficient system respectively. Thus coding operations are partitioned into:

- Source coding
- Channel coding

As shown in Figure 2.9, channel coding increase the bandwidth due to the addition of redundancy bits which is traded for good quality source coding involves the removal of unwanted information ' a from the message, which invariably reduces bandwidth within a satisfactory quality for decoding. If the symbol decision is made at the detector/demodulator, then the such process is considered a hard decision. A decision is said to be soft if such symbol decisions are made at the decoder. The key parameters derivable from the received signal-to-noise ratio (SNR) are the bandwidth efficiency R/B and the bit energy to spectral noise density ratio (Eb/No). The choice of coding and modulation has the capacity to deliver high quality digital information (ITU Report, 2010). The combination of coding and modulation is called coded modulation. The system spectral efficiency p, of a coded modulation determines the bandwidth B requirement. ldeal coded modulation should have as minimum error probability PB as possible with low Eb/N0 but with high spectral efficiency p. From equation 2.26, SNR is a product of Eb/N where R is the transmitted data rate (bps) (Charles, 1999).

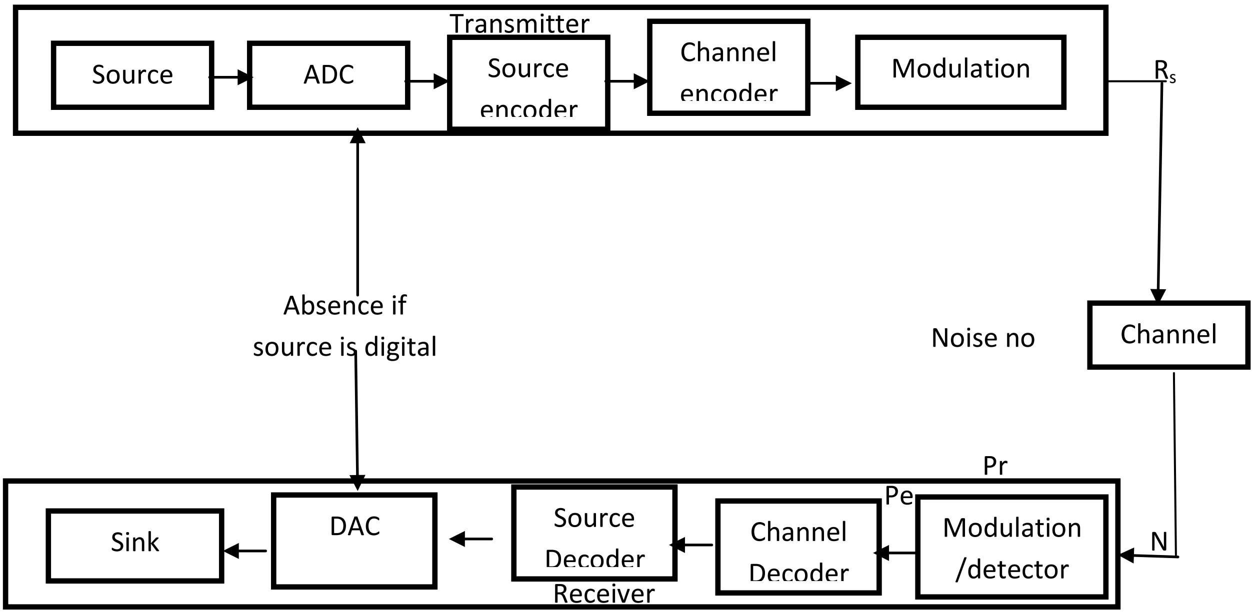


**Figure 2.9 coding and modulation system (Adeyemi, 2013).**

Channel encoder adds redundancy bit (n k) to the information bits (k) in order to curtail transmission errors. In order word, k bits data is assigned to an n-bit codeword, denoted by (k,n) code, resulting to k/n code rate r. The symbol of m bits, with Ts sec symbol duration is then mapped into one of the possible M waveforms by M-arry modulator. The signal is then transmitted within the required bandwidth, B. (Adeyemi, 2013).

## 2.9 NOISE

Noise is simply referred to as unwanted random signals (Kwaha, 2016). It present degrades the performance of the communication system, (Umar, 2016). If signal is corrupted by noise, one may think that amplifying the signal further, will reduce the noise, no, because while amplifying the signal, the noise too will be amplified. (Kwaha, 2016)

Noise that occurs such that it occupies the' whole bandwidths are called Additive Whilte Gaussian Noise AWGN. Such noise cannot be band limited (Kwaha, 2016).

### 2.9.1 Types 0f Noise

In communication some sources that generate noise may be natural such as electric storms, stellar radiation (galactic Noise), lightening discharges and other atmospherics. While some are Man-made from electrical equipment e. g arching contact in electrical machinery, (Umar, 2016).

Few of them are mentioned, these are

***Thermal Noise***

It occurs due to the random motion of electrons in a conductor by the thermal agitation of atoms. This is perhaps the most fundamental type of noise since it is always present in a conductor without the external application of electrical energy (except at a temperature of absolute zero), (Umar, 2016).

When in motion, electrons perform collisions of all sorts even with imperfection or even lattice Sites. Note that these collisions occur in all real conductors and it causes the electron density throughout the conductor to vary randomly. This is what causes resistance. It causes a random voltage in the conductor.

Generally in a communication system, when these electrons move in the conductor(wire) they move at velocity of approximately 105 m/s (Kwaha, 2016).

The noise voltage is given as

$$E^2n = 4RKTBn \tag{2.32}$$

Where R is the resistance of the conductor in Q, T is the temperature of the conductor in K, Bn is the noise Bandwidth in Hz and K is the Boltzman‘s constant"1.38 x 10'23 J/K

For a generator of emf$E_{volt}$ (rms) and internal resistance R, the available power is

$$\frac{E}{4R} \tag{2.33}$$

Applying this to the equation of available thermal noise power is

$$Pn = KTBn. \tag{2.34}$$

(Kwaha, 2016)

***Shot Noise***

This noise is encountered in active devices (valves, transistors and doides) and caused by the discrete nature of electrons and holes, flowing in semiconductors and other charge-transfer processes. Shot noise is in many ways similar to that thermal noise. They are both due to the random fluctuation of a large number of electrons, have uniform power spectral densities and furthermore, the mean square current in both cases is directly proportional to the bandwidth of the system.

The mean square noise component is proportional to the dc flowing. Generally, the mean square shot noise current is given by

$I^2n=21_{dc}q_eBn$ (2.35)

Where $I_{dc}$ is the dc in amperes, qe is the magnitude of electron charge (1.6x10‘mC) and En is the Bandwidth ih Hz. For thermal noise, using conductance G= l/R, the rms noise current In is $I^2n = 4GKTBn$ (2.36)

***Partition Noise***

Partition noise occurs whenever current has to divide between two or more electrodes (i.e Junction). This results from the random fluctuations in the divisions. Example ordinarily a diode is less noisy than a transistor because a transistor has three terminals and two junctions while diode has two terminals and one junction. This is why the inputs of most microwave devices contain a diode circuit.

***Flicker Noise***

At very low frequencies (below KHZ) a noise component exists. Its spectral density increases as the noise frequency decreases. It is called Flicker noise (or l/f noise). In vacuum tubes it manifests as a slow change in Oxide structure if the oxide coated cathode and also from migration impurity ions, through the oxide. In semiconductor, flicker noise arise from the fluctuation in the carrier densities (hole and electrons) which in turns affect the conductivity of the material. It follows therefore that a noise voltage is developed whenever dc current flows through a semiconductor. Although flicker noise is a low frequency effect, it bursts the sensitivity of microwave diode mixers used in Doppler radar systems. Because Doppler frequency output is in the low frequency range where flicker noise is significant.

***Burst Noise***

Burst noise is common in BJT. The name arises because the noise arises as a series of bursts at two or more levels. In an audio system, this noise produces popping sounds hence sometimes called ‘pop corn’ noise. lts spectral density also increases as the frequency decreases.

***Avalanche Noise***

The reverse bias characteristics of a diode exhibits a region where the reverse current, normally very small, increases very rapidly with a slight increase in the reverse voltage, this is called the avalanche region, This occurs because of the holes and electrons in the depletion region of the diode gain sufficient energy from the reverse bias field to ionize atoms by collision. This in a multiplicative process called avalanche. The collision process occurring to create the avalanche results in large noise spike E.g in Zenerdiode , the spectral density is flat.(Kwaha, 2016)

### 2.9.2 Signal-To-Noise Ratio (SNR)

In communication, signal-to-noise ratio is more considered than the value of the noise. The ratio is defined as a power ratio and given as the square of voltages.

$$SNR = Ps/Pn \qquad (2.37)$$

$$=V^2s/V^2n= \frac{thermal\ noise\ voltage\ from\ the\ source}{total\ noise\ voltage\ at\ the\ input} \qquad (2.38)$$

Where Vs is the noise voltage from the source and Vn is the noise voltage at the input. (Kwaha, 2016)

## 2.10 REVIEW OF RELATED LITERATURE

Ricardo et a1 (2006), investigated a design of power and spectrally efficient coded modulations based on amplitude phase shift keying (APSK) modulation with application to satellite broadband communication.

The work presented APSK as attractive modulation format for digital transmission over non-linear distortion and so it has been recently induced in the new standard for satellite digital video broadcasting names DVB-$S_2$. Assuming an ideal rectangular transmission pulse, for which no non-linear inter-symbol interference is present and perfect pre-compensation of the non-linearity, optimized the APSK constellation. In addition to the minimum distance criteria, a new Optimization is introduced based on channel capacity, this new method generates an optimum constellation for each spectral efficiency. To achieve power efficiency jointly with low bit error rate (BER) floor we adopt a powerful binary serially concentrated turbo-code coupled with optimal APSK modulations through bit-interleaved coded modulation. Tight approximation on the maximum-likelihood decoding error probability and results are compared with computer simulations. [n the current analysis is complemented with the effects related to satellite non-linear distortion effects with a band-limited transmission pulse and inducing demodulator timing amplitude and phase estimation errors. The proposed coded modulation scheme is shown to provide a considerable performance advantage compared to current standards for satellite multi-media and broadcasting systems.

Ricardo *et al* (2001), proposed a new class of l6-ary Amplitude and phase shift keying (APSK) coded modulations called double-ring modulation. The new modulation is shown to be power and spectrally-efficient, with interesting applications to satellite communications. A preliminary constellation parameter optimization, based on information-theoretic evaluation on channel capacity bounds has been validated by comprehensive computer simulations. Furthermore, a simple and yet effect pre-compensation technique to reduce the impact of non-linearities is propose and examined digital clock and carrier phase schemes have been derived analyzed for a turbo coded versioampn of the double ring APSK scheme,

Finally, the performance of the state-of the-art turbo new 16-ary digital modulation has been investigated and compared to the know TCM scheme. It is shown that, for the same coding scheme, pre-distorted double-ring APSK modulation largely outperform classical 16-QAM and 16-PSK over a typical satellite chemical, due to its intrinsic robustness against the high power amplifier (HPA) non-linear characteristics. The use of turbo codes, as shown in the paper, provides an additional gain over TCM of the equivalent rate for both liner and non-liner channels. A comparison of turbo code performance with a concentrated TCM and Reed-Solomon scheme is also made. The proposed coded modulation scheme is shown to provide a considerable performance advantage for the future satellite multiline and broadcasting system.

Xingyu xiang (2013), investigated the optimization of a coded modulation system with shaped constellation. The research indicate that conventional communication system transmit signals that are selected from a single constellation with uniform probability. However, study further disclosed that information theories results suggest that performance may be improved by shaping the constellation such that lower- energy signals are selection more frequently than higher-energy signals. In this dissertation an energy-efficient approaches for shaping the techniques for system that use a combination of amplitude-phase shift keying (APSK) and low-density parity check (LDPC) coding. Such as the system used by the DVB – $S_2$ standards.

The system implementation requires that a subset of the bits at the output of the LDPC encoder are passed through a non-liner shaping encoder whose output bits are more likely to be zero than a one.

The constellation as indicated in the paper. Is petitioned into a plurality of sub-constellations, each with a different average single energy and the shaping bits are used to select the sub-constellation modulation, LDPC decoder, and shaping codes are optimized with respect to information rate, while the design of the LDPC code is optimized with for shaped modulation with assistance of extrinsic-information transfer (exit) charts. The rule, for labeling the constellation with bits is optimized using a novel hybrid cost function and binary switching algorithm.

Simulation results show that the combination of constellation shaping LDPC code optimization, and optimized bit labeling can achieve a gain in excess of IdB an Additive white Gaussian noise (AWGN) channel at the rate of a 3 bits/symbol compared with a system that adheres directly to the DVB-$S_2$ standard.

# CHAPTER THREE

# METHODOLOGY

## 3.1 CODED MODULATION SELECTION METHODOLOGY

Supposed the modulation scheme and possible coding are required to meet certain performance requirements such as bit error probability $p_B$ and data rate R are know. The available bandwidth B and the received signal power to noise –power spectral density $P_f/N_o$ are know. There is a trade-off between the spectral efficiency R/B and the energy efficiency $E_b$ / $N_o$ achieve performance requirement $p_B$ (Adeyemi, 2013)

### 3.1.1 Step 1:

Determine whether the system is bandwidth-limited or power-limited by comparing the bit rate R (bps) with the bandwidth B (Hz). If the bit rate greater than the bandwidth b (i.e if R> B), then the link is said to be bandwidth-limited.(Adeyemi, 2013)

Then apply or think of MPSK modulation, since it is much more spectrally efficient, especially when using smaller phase shifts, more bits can be transmitted:

From equation (2.5) it can be seen that

$M=2^m$ or $m = \log^2 m$

Where m is an integer representing the number of bits per unit symbol. M is known as the symbol comprising of m bits.

But if the data rate R is less than the bandwidth B (i.e R< B), then the link is said to be power-limited which leads to insufficient use of bandwidth. Then in this care, think of MFSK modulation since it is not bandwidth efficient.

### 3.1.2 Step 2:

Determine the minimum bandwidth W using the following

Expressing relating the symbol rate $R_s$ with the rate R i.e equation

$Rs = R/Log_2 m$

Expression for minimum required bandwidth for MPSK and MFSK given as

$B_{mpsk} = I/T = Rs$ Hz,

$B_{mfsk}$ $M/T = MRs$ Hz,

Ensure that the minimum required bandwidth W is not greater than the available bandwidth B. if otherwise, select the smallest value of M (i.e the symbol consisting of m bits) that satisfy the condition.

### 3.1.2 Step 3:

Verify how bandwidth efficient the design is, using the following equations

$Pmpsk = R/B = \log_2 M/BTs = I/BT_b$ bps,/Hz

Where Tb = symbol duration in seconds.

$Ts = mT_b$ = symbol duration: and

B= available bandwidth.

The bandwidth efficiency using the Nyquist filteing at the baseband where the B isequal to Rs (B = Rs). P is given as

$P = \frac{R}{B}\ \frac{R}{RS} = 10g2m$ bps.

Where R is the transmission bit rate (Gibson, 2002).

From the equation it can be seen that, efficiency increases, hence M-PSK and M-QAM are said to be bandwidth-efficient.

**3.1.4 Step 4:**

Since the received signal power to noise – power special density $\frac{Pr}{No}$ is known (as supposed). Then obtain the value of the power efficiency from the relationship of the ratio of average received power to noise-power spectral density $\frac{Pr}{No}$ and nit energy noise-power spectral density which is given from equation (2.27)

$$\frac{Eb}{NO} = \frac{\Pr 1}{No\ R}$$

Which is valid for both MPSK and MFSK modulation scheme.

3.1.5 step 5:

Verify if the performance requirement is not using equation (2.16) given as

$$PE\ (M) \cong 20Q[\sqrt{\frac{2Es}{NO}}\ \mathrm{Sin}\ (\frac{\pi}{M}))]$$

For MPSK, the signaling system, symbol error probability $P_E$ is related to bit error probability $P_B$ by

$$P \approx \frac{PE}{Log\ M} = \frac{PE}{M}$$

For MPSK the symbol error probability is given from equation 2.18:

$$PE\ (M) \leq \frac{M-1}{2}\ \exp\ (-\frac{Es}{2M})$$

Where

Q(X) is the Q-function sometime called the complementary error function Q(x) can be solved using single mat Lab code "qfunc (x)"

If the answer satisfies the specified bit error probability $P_{B,}$ then the modulation scheme is considered alright without coding. This scenario is referred to as encoded symbol

### 3.1.6. Step 6:

A coded symbol become necessary when application are critical and the required performance tends to gulp more bandwidth and power than what is available. Bandwidth-limited and yet power limited system requires performance improvement using coding mechanism such that the system can operate within the available satellite bandwidth.

### 3.1.7 Step 7:

Take advantage of any available liberty for coding and choose the appropriate rate R (K/n).

Where

n=n-bits code word

k=information bit (data)

(meaning k bits data is assigned to an n-bit code word).This choice of code rare R should increase the transmission bandwidth outside the available channel bandwidth.

Supposed a real –time Digital TV (DTV) application over the satellite which requires modulation and coding. And we presume a modulation scheme whose bandwidth is calculated as 12MHz while the available channel bandwidth is 15MHz, creating a space of 3MHz.

Then:

- Find the % tolerance as follows

$$\frac{\text{Channel bandwidth} - \text{modulation bandwidth}}{\textit{modulatiion band width}} \quad \text{x } 100 \quad \%$$

In this case, we have 25% tolerance.

Since channel coding enlarges bandwidth, take the advantage of this liberty (3MHz space at 25% tolerance) for the coding

To choose the code rate R, the code rate therefore must not be less than:

$$\frac{100\% - 25\%\ tolerance}{100\%} = 0.75$$

In other word, the code must not gulp more than 25% of the carrier bandwidth. This means 3/4, 5/6, 7/8 FEC are the suitable code rates. The task left is to save power at the expense of the bandwidth at our liberty in order to meet the BER performance requirement.

**3.1.8 Step 8:**

Calculate the coding gain G which is a measure of power saving between a coded and un coded system.

To do that, employ equation (2.19):

$$G = [(\frac{Eb}{No}) \quad - (\frac{Eb}{No}) \ ] \quad (dB)$$

Uncoded coded

Then the channel encoder with code rate r (k/n)) processes the data rate R to produce a channel rate Rc which is sent to the modulator which further rate a symbol rate Rs:

Rc = (n/k) R     (bps)

Rs = $\frac{Rc}{log2M}$

Due to coding process, the average received power to noise-power spectral density, (p/ No) arriving at the demodulators input is given by

$$\frac{Pr}{No} = \frac{Ec}{No}\ \text{Rc} = \frac{Es}{No}\ \text{Rs} \qquad \text{(dB)}$$

Where Ec and Es are the channel and symbol energy respectively

The modulation then yields channel error probability Pc is given as

Pc = $\frac{Pc}{M}$

# CHAPTER FOUR
# RESULTS AND DISCUSSIONS

## 4.1 Effect of Efficient Modulation (with coded/uncoded symbol)

The outcomes of steps 1 to 5 are not base on coded system only, rather uncoded symbol is also considered. This is because conclusion as to whether the system or modulation technique is considered alright with or without coding could only be determined at step 5 base on the value of the symbol error probability calculated at that stage for any decision to be taken as to whether to forge ahead by employing coding mechanism (when the application is critical) or not.

It can be seen clearly from step 1, that if the bit rate R is greater than bandwidth the link is considered bandwidth limited which implies the need for increase in its channel capacity, C by increasing the transmit power-limited system; the bit rate is less than the bandwidth leading to inefficient application of the bandwidth, consequently promoting low channel capacity. However, to compromise between these two conditions, selection is made for modulation technique for power-limited system and MPSK modulation technique for bandwidth –limited system. Since MFSK is spectrally inefficient and so requires less power and MPSK is much more efficient spectrally.

Step 2: Determination of the required bandwidth is our ultimate guide towards ensuring that the available bandwidth is greater than the minimum required bandwidth for communication to be achieved. This step has a direct link/relationship with the extent of efficiency is the design which controls or depends on the value of M. hence, M is always adjusted to meet/satisfy the condition for the minimum required bandwidth.

Step 3: The effect of the verification of the extent of design efficiency enables identification of modulation technique that is most bandwidth efficient and those that are not. Since for

MPSK and M- Q where bandwidth efficiency decreases with increase in value of M which is contrary to M-FSK where bandwidth efficiency decrease with increasing M, hence,

Step 4: in digital system such as this, information degradation is in terms of bit error rate. (BER) (T, pratt, et, al, 2002). Energy efficiency is therefore needed for certain performance required to be delivered, since the received single power spectral density are known. Compared with ideal code modulation this value is presumable and relatively limited, otherwise, R is adjusted within the range required range of minimum required band with for the communication until the condition is satisfied.

Step 5: this is outcome of step 1 to step 4.recall that the ideal coded modulation should have as minimum error possibility $p_B$ as possible with limited (low) energy efficiency value but with high spectral efficiency, therefor, verification of the performance requirement to carry out at this stage by determining the symbol error probability using approximate equations for either MPSK or MFSK as the case may be the product is then used to calculate the bit error probability accordingly, if the value conforms with specified bit error probability $P_B$ then the design is considered alright without coding (uncoded symbol) on the contrary, step 6,7 and 8 are employed.

Step 6: if the calculated value of symbol error probability did not satisfy the specified bit error probability $P_B$, the application is therefore considered critical and thus the system tends to absorb or gulp more power than the available. Hence the require coding mechanism to archive the performance required.

Step 7: critical choice is made of the approximate code rate that is within the suitable range of error correction. However, calculation of tolerance is carried out to predict the maximum value of  code that is needed for the system to gulp within the% percentage tolerance basically, the code rate are obtained by converting the % percentage tolerance to its equivalent in decimal places, then converting to range of suitable for use as rates in forward error correction (FEC) the task is to determine the suitable code rates a to be accommodated or gulped by the available channel capacity of such design.

**Step 8:** finally the task is to save the power at the expense of any available bandwidth so as to archive the BER performance requirement. The mechanism by computing the difference in the power between the uncoded and coded system know saving or coding gain.

# CHAPTER FIVE

# SUMMARY, CONCLUSIONAND RECOMMENDATION

## 5.1 SUMMARY

Efficient utilization of digital modulation and coding technique to optimize the performance of the satellite links capacity has been the focus of research. However, the major challenge in satellite communication lies on the effective usage of the two key factors/resources or satellite communication system, that is power limited and spectral inefficient due to growing demand of power as result of increase in band with demanding applications.

Review of some related literatures were made so as to serve the as guide and basic for comparison towards meeting the aim of this research. Brief descriptions on some basic principles in satellite communication were also considered.

The objectives of this research are to strike the balance between the band with-limited and power-limited conditions, provision of mechanism that will yield great saving financially while meeting the customers' demands. Discussion in details was made on coded modulation selection methodology where series of eight steps were critically analyzed for the aforementioned objectives to be achieved. The mechanism further disclose the effects of effective combination of coding and digital modulation scheme that yield a tremendous band width utilization and coding gain. This coding gain base on the study, has shown a remarkable energy saving which improves link available and the transponder can then operates with distortion. The proposed outcome indicates the greater improvement in meeting the system performance requirement without Sevier financial implications as compared with previous work.

## 5.2 CONCLUSIONS

Digital technology has a tremendous merits and so the digital modulation technique, focused on this research is effective utilization of code modulation technique assuming the bit error probability and dot rate R are known.

Channel coding is like an insurance against error probability even if the error do not occur most of time. However when they occur, coding is used to combat such distortion to ensure integrity.

From the related literature, Ricardo *et al* (2001), investigates the design of power and spectrally efficient coded modulation information and minimum Eucidean distance criteria, under the simplified assumption of rectangular shaped transmission pulse. However, the research could not provide solution on how to achieve effective satellite communication under band with limited and power limited conditions. As long as long distance criteria. Ricardor *et al* (2006), in his work proposed 16-ary amplitude and phase shift keying coded modulation which is more power and spectrally- efficient compared with that of Recardo *et al* (2001) above. This is due to its intrinsic robustness against high power amplifier (HPA) non-linear characteristics. More so, the use of turbo code as shown in the paper provides an additional gain over TCM of the equivalent rate for both linear and non-linear channels. Again compassion of the turbo performance with a concatenated TCM and Reed-Solomon scheme is also made. The proposal coded modulation scheme is shown to provide considerable performance advantage for future satellite multimedia and broadcasting system, yet investigation failed to address the severe financial implication attached by the proposed design.

Investigation of Xingyu Xiang (2013), of the optimization of coded modulation system with shaped constellation. Simulation result, show that the combination of constellation shaping, low density parity check (LDPC) code optimized bit labeling can archive a gain in excess of IdB in the additive white Guassian noise (AWGN) channel at the rate of a 3 bits/symbol compared with a system adheres directly to the DVB.S2standard. This investigation could not proffer solution as to low severe financial implication could be avoided while archiving effective satellite link capacity under limited and band with (spectrum).

In this research of effective combination of digital modulation technique and coding scheme for optimized performance, it is therefore concluded that is a trade-off between spectral efficiency and energy (power) efficiency to archive certain performance requirement, this provided a considerable performance advantage compared with the aforementioned literatures, this provided a considerable performance advantage compared with the aforementioned literatures in terms of satellite multimedia and broadcasting system with great financial savings whose second order effects are keeping the satellite providers in business, meet customers' demands while optimizing satellite system operations.

## 5.3 RECOMMENDATION

The following recommendation were considered after the due research;

- Consideration the current financial crises in our contemporary society, minimizing the severe financial implications involve in satellite in satellite industries should be an optimum priority of the authority or service provides as that will go a long way in keeping the customers which will in turn sustain or keep the service providers in business. Thus, implementation of this mechanism in satellite industries will yield great financial savings.
- The ultimate desire of any link should be:

  Designing of an almost destruction free system with minimal transmitting power to deliver large data, using nominal spectrum band with at affordable low cost while maintaining the required bit error probability;

  The choice of digital modulation technique; this is due to its tremendous merits which includes lower design cost, increased system capacity using more efficient modulation scheme; robust error control coding technique to noise and interference.

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