IN THE NAME OF ALLAH THE MOST BENEFICENT AND THE MOST MERCIFUL

DESIGN AND IMPLEMENTATION OF QUADRATURE PHASE SHIFT KEYING MODEM

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FROM HOLLY SCRIPTURE

"In the Name Of Allah, the Compassionate, the Merciful"

Praise to Allah, the lord of creation. The compassionate, the merciful. The king of the judgment day! You alone we pray, and to you alone we ask for help. Guide us to straight. The path of those whom you have favored, not of those who have incurred your wrath, nor of those who have gone astray.

DEDICATIONS

This project is dedicated to our parents who has made us capable of doing this job and always been praying for our success. May Allah grant them long life.

ACKNOWLEDGEMENT

All thanks to Allah Almighty who gave us the faith, hope and ability to complete this project successfully.

It is Allah who makes and breaks and gives ability to his humble men to do things which they cannot even imagine of doing. So all praise is for Allah and all greatness is for Allah.

Thanks to our Project advisor Group Captain (Retd) Muzaffar Ali for without his valuable advice and motivation we never would have been able to accomplish our goals.

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PREFACE

This project is about a new era of human communication where modulation technologies become information skyways, a new avenue to send ideas and masses of information to remote locations in ways most of us would never have imagined.

This project report on the Design and Development of the Quadrature Phase Shift Keying Modem represents not only an endless effort and hard work but it also encapsulates in it a lesson for life which only one of the authors, of this preface to the report, would know.

The aim of the project was to acquire understanding of modern day modulation and demodulation techniques and to study closely one of them with respect to its design and implementation. Thus, this flow can be observed in this report which starts of with the basics of digital communications.

In Chapter 1 the very need for modulation is discussed. In chapter 2 all the basic types of modulation schemes are explored and brief introductions along with diagrams is given. In Chapter 3 the authors of the report narrow down themselves to the Phase Shift Keying Technique and describe its various forms like the BPSK and the QPSK etc. In Chapter 4 hardware design of the modem is discussed.

The completion of this project report has been made possible by contribution from a lot of people which include parents, teachers, friends and seniors. Their names demand a whole new report to be published which at this stage is not very convenient. So thanks to all who have helped us and may Allah have Mercy on them.

CHAPTER 1

INTRODUCTION

1.1 Digital Communications:

The term digital communications covers a broad area of communications techniques, including digital transmission and digital radio. Digital transmission is the transmittal of digital pulses between two or more points in a communications system. Digital radio is the transmittal of digitally modulated analog carriers between two or more points in a communications system. Digital transmission systems require a physical facility between the transmitter and receiver, such as a metallic wire pair, a coaxial cable, or an optical fiber cable. In digital radio systems, the transmission medium is free space or Earth's atmosphere.

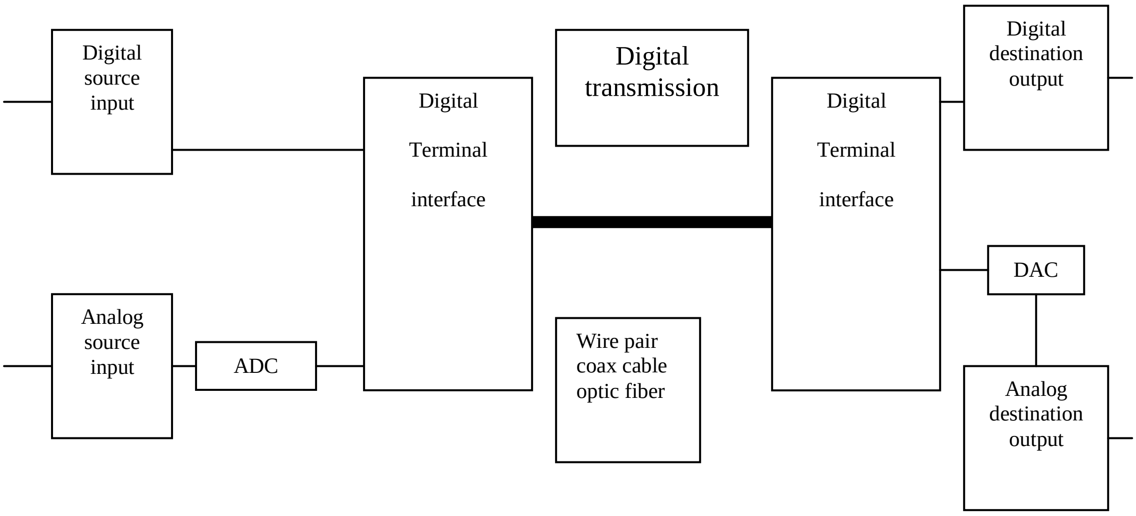

Fig.1 Digital Communication Systems: (a) digital transmission

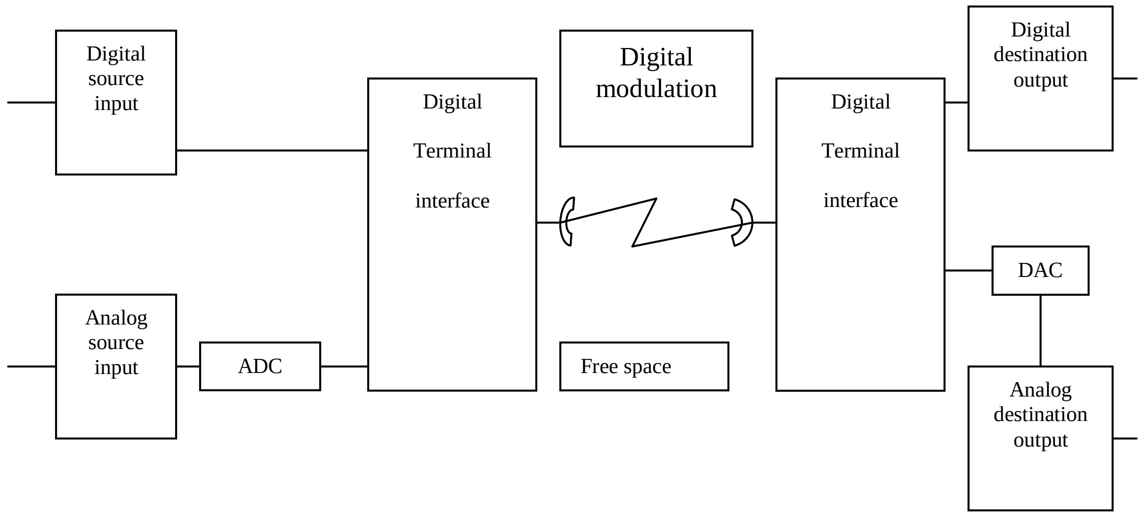

Fig.2 Digital Communication Systems: (b) digital radio

In a digital transmission system, the original source information may be in digital or analog form. If it is in analog form, it must be converted to digital pulses prior to transmission and converted back to analog form at the receive end. In a digital radio system, the modulating input signal and the demodulated output signal are digital pulses. The digital pulses could originate from a digital transmission system, from a digital source such as a mainframe computer, or from the binary encoding of an analog signal.

1.2 Shannon limit for information capacity:

The information capacity of a communications system represents the number of independent symbols that can be carried through the system in a given unit of time. The most basic symbol is the binary digit (bit). Therefore, it is often convenient to express the information capacity of a system in bits per second (bps). In 1928, R. Hartley of Bell Telephone Laboratories developed a useful relationship among bandwidth, transmission time, and information capacity. Simply stated Hartley's law is

$$\mathbf{I \;\alpha\; B x T}$$

Where

I = information capacity (bps)

B = bandwidth (Hz)

T == transmission time (s)

From Equation above it can be seen that the information capacity is a linear function of bandwidth and transmission time and is directly proportional to both. If either the bandwidth or the transmission time is changed, a directly proportional change in information capacity will occur.

In 1948, C. E. Shannon published a paper in the Bell System Technical Journal relating the information capacity of a communications channel to bandwidth and signal-to-noise ratio. Mathematically stated, the Shannon limit for information capacity is

$$\mathbf{I = B \log_2[1+S/N)}$$

where

I = information capacity (bps)

B = bandwidth (Hz)

S/N = signal-to-noise power ratio (unit less)

For a standard voice-band communications channel with a signal-to-noise power ratio of 1000 (30 dB) and a bandwidth of 2.7 kHz, the Shannon limit for information capacity is

$$I = 2700 \log_2[1+1000)$$
$$= 26.9 \text{ kbps}$$

Shannon's formula is often misunderstood. The results of the preceding example indicate that 26.9 kbps can be transferred through a 2.7 kHz channel. This may be true, but it cannot be done with a binary system. To achieve an information transmission rate of 26.9 kbps through a 2.7 KHz channel, each symbol transmitted must contain more than one bit of information. Therefore, to achieve the Shannon limit for information capacity, Digital transmission systems that have more than two output conditions (symbols) must be used. These systems include both analog and digital modulation techniques and the transmission of both digital and analog signals.

1.3 Digital Radio:

One property that distinguishes a digital radio system from a conventional AM, FM, or PM radio system is that in a digital radio system the modulating and demodulating signals are digital pulses rather than analog waveforms. Another difference between digital and analog radio systems is the type of modulation used. Digital radios, however, use analog carriers just as conventional systems do.

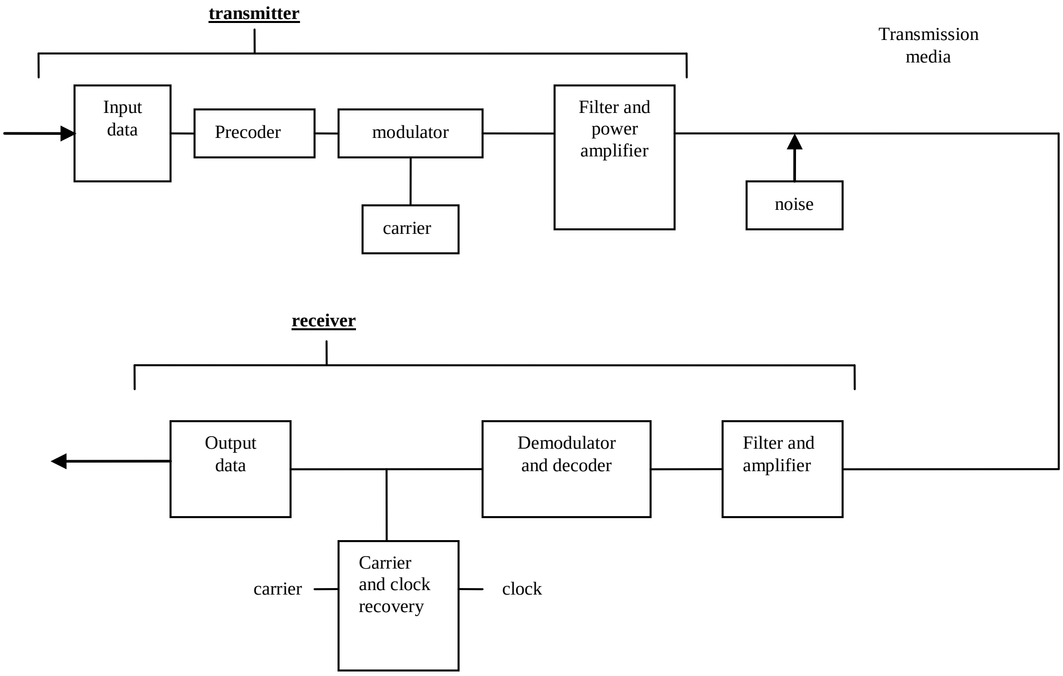

Fig.3 Simplified block diagram of a digital radio system

Figure shows a simplified block diagram for a digital radio system. In the transmitter, the pre-coder encodes or groups the incoming data into a control word that modulates the carrier signal, The modulated carrier is shaped (filtered), then amplified. In the receiver, the pulse shaping network further filters the modulated wave, rejects out-of-'band noise, and suppresses interference from adjacent channels.

1.4 Digital amplitude modulation:

The simplest digital modulation technique is double-sideband, full-carrier amplitude modulation with a binary modulating signal. Mathematically, amplitude modulation by a binary signal is

$$\mathbf{V(t) = A/2\ [1 + m(t)]\ (\cos \omega_c t)}$$

where

$\omega_c t$ = carrier angular frequency (rad/s)

m(t) = binary modulating signal

In above equation, the modulating signal is a standard binary data waveform, m(t) = ±1. Thus, for 100% modulation v(t) is either on or off; what is referred to as on-off keying (OOK) modulation.

It is generally recognized that with amplitude modulation the carrier itself contains no information. Consequently, suppression of the carrier yields DSB-SC (double-sideband, suppressed-carrier) modulation, which is stated mathematically as

$$\mathbf{V(t)_{dsb} = A\ m(t)\ (\cos \omega_c t)}$$

When m(t) equals either 0 or 1, again we have OOK modulation. An OOK waveform can be demodulated either coherently or non-coherently with little difference in performance. The use of amplitude modulated analog carriers to transport digital information is a relatively low-quality, low-cost method of digital radio and is seldom used in high capacity, high-performance systems.

1.5 Why modulate?

Digital modulation is the process by which digital symbols are transformed into waveforms that are compatible with the characteristics of the channel. In the case of base band modulation, these waveforms usually lake the form of shaped pulses.. But in the case of band pass modulation the shaped pulses modulate a sinusoid called a carrier wave, or simply a carrier; for radio transmission the carrier is converted to an electromagnetic (EM) field through space is accomplished with the use of a carrier for radio transmission of base band signal. The transmission of EM fields through space is accomplished with the use of antennas. The size of the antenna depends on the wavelength λ and the application.

For cellular telephones, the antennas are typically λ /4 in size, where wavelength is equal to c/f, and c, the speed of light is 3 x 108 m/s. Consider sending a base band signal by coupling it to an antenna directly without a carrier wave. How large would the antenna have to be? Let us size it by using the telephone industry benchmark of λ /4 = 2.5 x 104 m = 15 miles. To transmit a 3,000 Hz signal through space, without carrier wave modulation, an antenna that spans 15 miles would be required. However, if the base hand information is first modulated on a higher frequency carrier, for example a 900 MHz carrier wave or band pass modulation is an essential step for all systems involving radio transmission.

Band pass modulation can provide other important benefits in signal transmission. If more than one signal utilizes a single channel, modulation may be used to separate the different signals. This technique is known as FDM. Modulation can be used to minimize the effects of interference. A class of such modulation schemes, known as spread spectrum modulation, requires a system bandwidth much larger than the minimum band width that would be required by the message. Modulation can also be used to place a signal in a frequency band where design requirements, such as filtering and amplification can be easily met. This is the case when radio frequency (RF) signals are converted to an intermediate frequency (IF) in a receiver.

formatting

Character
Coding sampling
Quantization PCM

Source coding

Predictive coding Block coding Variable length coding Synthesis/analysis coding Lossless compression Lossy compression

Baseband signaling

PCM waveform NRZ,RZ Phase encoded Multilevel binary M-ary pulse modulation PAM,PPM,PDM

Equalizatio

Max likelihood sequence estimation(MLSE)
Equalization with filters
Transversal or decision feedback Preset or adaptive
Symbol spaced or fractionally spaced

Bandpass Signaling

coherent

PSK
FSK
ASK
CPM
Hybrids

Noncoherent

DPSK
FSK
ASK
CPM
Hybrids

Channel coding

Waveforms

M-ary signaling Antipodal Orthogonal Trellis-coded modulation

Structured sequences

Block
Convolution
Turbo

Encryption

Block
Data stream

Synchronization

Frequency synchronization Phase synchronization Symbol synchronization Frame synchronization Network synchronization

Multiplexing/Multiple Access

Frequency division(FDM/FDMA) Time Division(TDM/TDMA) Code division(CDM/CDMA) Space division(SDMA) Polarization division(PDMA)

Spreading

Direct sequence(DS)
Frequency hopping(FH)
Timing hopping(TH)
Hybrids

Fig.4 Basic digital communication transformation

CHAPTER 2

DIGITAL BAND PASS MODULATION TECHNIQUES

2.1 Band pass modulation:

Band pass modulation (either analog or digital) is the process by which information signals is converted to a sinusoidal waveform for digital modulation, such a sinusoid of duration T is referred to as a digital symbol. The sinusoid has just three features that can be used to distinguish it from other: amplitude, frequency, and phase. Thus band pass modulation can be defined as the process whereby the amplitude, frequency, or phase of an RF carrier, or a combination of them, is varied in accordance with the information to be transmitted. The general form of the carrier wave is

$$\mathbf{s(t) = A(t) \cos \theta(t)}$$

Where A(t) is the time-varying amplitude and θ(t) is the time varying angle. It is convenient to write:

$$\mathbf{\theta (t) = \omega t + \varphi(t)}$$

So that

$$\mathbf{s(t) = A(t) \cos [\omega_0 t + \varphi(t)]}$$

Where ω_0 is the radian frequency of the carrier and φ(t) is phase. The terms f and ω each is used to denote frequency. When f is used, frequency in hertz is intended, when ω is used, frequency in radians per second is intended. The two frequency parameters are related by

$$\mathbf{\omega = 2 \pi f}$$

The basic band pass modulation/demodulation types are listed in figure 4. When the receiver exploits knowledge of the carrier phase to detect the signals, the process is called coherent detection; when the receiver does not utilize such phase reference information the process is called non-coherent detection. In general communications, the terms demodulation and detection are often used interchangeably, although demodulation emphasizes waveform recovery, and detection entails the process of symbol decision. In ideal coherent detection, there is available at the receiver a prototype of each possible arriving signal. These prototype waveforms attempt to duplicate the transmitted signal set in every respect even RF phase. The receiver is then said to be phase locked to the incoming signal.

During demodulation, the receiver multiplies and integrates the incoming signal with each of its prototype replicas. Under the heading of coherent modulation/demodulation in the figure 1 are listed phase shift keying (PSK), frequency shift keying (FSK), amplitude shift keying (ASK), and continuous phase modulation (CPM), and hybrid combinations. The basic band pass modulation formats are discussed in this chapter. The specialized format of offset quadrature PSK (OQPSK) will be discussed in a later chapter.

Non-coherent demodulation refers to systems employing demodulators that are designed to operate without knowledge of the absolute value of the incoming signal's phase; therefore, phase estimation is not required. Thus the advantage of non-coherent over coherent systems is reduced complexity, and the price paid is increased probability of error. In figure 1 the modulation demodulation types that are listed in the non-coherent column, DPSK, FSK, ASK, CPM, and hybrids are similar to those listed in the coherent column. We had implied that phase information is not used for non-coherent reception, how do you account for the fact that there is a form of phase shift keying under the no-coherent heading? It turns out that an important form of PSK can be classified as non coherent (or differentially coherent) since it does not require a reference in phase with the received carrier. This "pseudo-PSK", termed differential PSK (DPSK), utilizes phase information of the prior symbol as a phase reference for detecting the current symbol.

2.2 Phasor representation of a sinusoid:

Using a well-known trigonometric identity called Euler's theorem, we introduced the complex notation of a sinusoidal wave as follows:

$$\mathbf{e^{j\omega 0t} = \cos \omega_0 t + j \sin \omega_0 t}$$

One might be more comfortable with the simpler, more straightforward notation $\cos\omega_0 t$ or $\sin \omega_0 t$. What possible benefit can there be with the complex notation? We will see that this notation facilitates our description of how real-world modulators and demodulators are implemented. For now, let us point to the general benefits of viewing a carries wave in the complex form of equation (1.1).

First, within this compact form, $e^{j\omega 0t}$,is contained the two important quadrature components of any sinusoidal carrier wave, namely the in-phase (real) and the quadrature (imaginary) components that are orthogonal to each other. Second, the un-modulated carrier wave is conveniently represented in a polar coordinate system as a unit vector or phasor rotating counterclockwise at the constant rate of ω_0 radians/s, as depicted in figure 5. As time is increasing we can visualize the time-varying projections of the rotating phasor on the in-phase (I) axis and the quadrature (Q) axis.

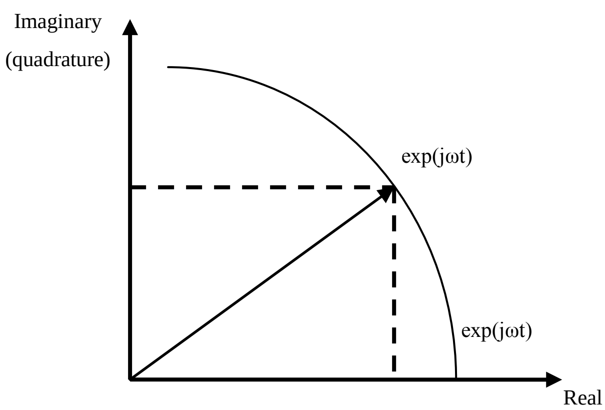

Fig. 5 Phasor representation of a sinusoid

These Cartesian axes are usually referred to as I channel and Q channel respectively and the projections on them represent the signal components (orthogonal to each other) associated with those channels. Third, when it comes time to modulate the carrier wave with information, we can view this modulation as a methodical perturbation of the rotating phasor (and its projections).

For example, consider a carrier wave that is amplitude (AM) with a sinusoid having amplitude of unity and a frequency ω_m where

$$\omega_m << \omega_0$$

the analytical form of the transmitted waveform is:

$$s(t) = Re \{ e^{j\omega 0t} (1 + e^{j\omega mt} /2 + e^{-j\omega mt})\}$$

Where Re[x] is the real part of the complex quantity [x]. Figure 6 illustrates that the rotating phasor ejω0t of figure 5 is now perturbed by two side band terms $e^{j\omega mt}$ /2 rotating counterclockwise and $e^{-j\omega mt}$ rotating clockwise. The sideband phasors are rotating at a much slower speed than the carrier-wave phasor. The net result of the composite signal is that the rotating carrier-wave phasor now appears to be growing longer and shorter pursuant to the dictates of the sidebands, but its frequency stays constant hence, the term "amplitude modulation."

Another example to reinforce the usefulness of the phasor view is that of frequency modulating (FM) the carrier wave with a similar sinusoid having a frequency of ωm radians/s. The analytical representation of narrowband FM (NFM) has an appearance similar to AM and is represented by

$$s(t)=Re\{e^{j\omega ot}(I-\beta e^{-j\omega mt}/2+\beta e^{j\omega mt} /2)\}$$

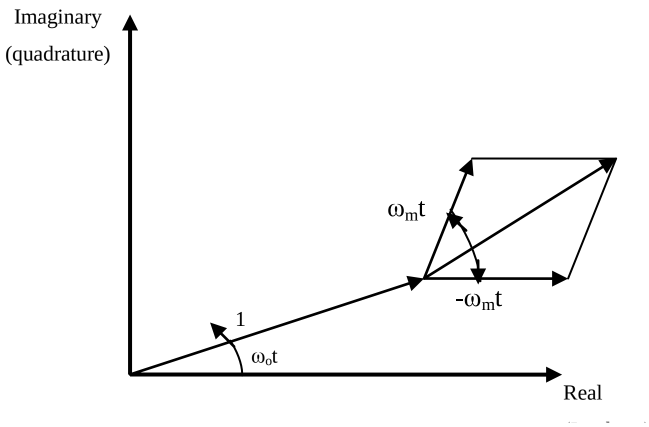

Fig. 6 Phasor representation of a sinusoid

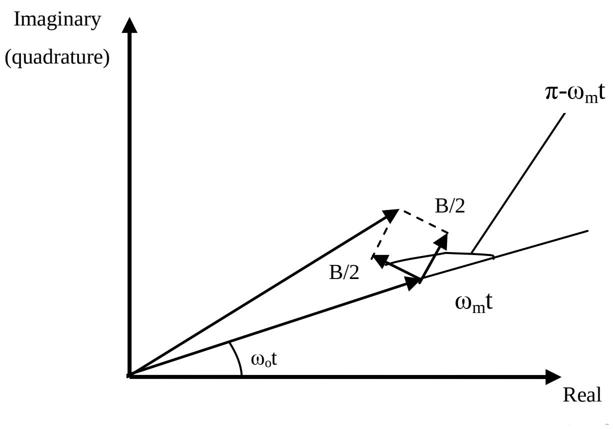

Fig. 7 Phasor representation of a sinusoid

Where β is the modulation index [1] figure 7 illustrates that the rotating carrier-wave phasor is again perturbed by two sideband terms, but because one of the sideband terms carries a minus sign in equation (1.6), the clockwise and counterclockwise rotating sideband phasors have a different symmetry than in the case of AM. In the case of AM the sideband symmetry results in the carrier-wave phasor growing longer and shorter with time. In NFM, the sideband symmetry (90° different than AM) results in the carrier-wave phasor speeding up and slowing down according to the dictates of the sidebands, but the amplitude stays essentially constant, hence, the term "frequency modulation".

Figure 8 illustrates examples of the most common digital modulation formats: PSK, FSK, ASK, and a hybrid combination of ASK and PSK (ASK/PSK or APK). The first column lists the analytic expression, the second is a typical pictorial of the waveform versus time, and the third is a vector schematic, with the orthogonal axes labeled [ψ(t)}. In the general M-ary signaling case, the processor accepts k source bits (or channel bits if there is coding) at a time and instructs the modulator to produce one of any available set of M=2k waveform types. Binary modulation, where k=1, is just a special case of M-ary modulation. In figure 4, we represented a carrier wave as a phasor rotating in a plane as the speed of the carrier-wave frequency ω_o radian/s. In figure 7, the phasor schematic for each digital modulation examples represent a constellations of information signals (vectors or points in the signaling), where time is not represented. In other words, the constantly rotating aspect of un-modulated carrier wave has been removed, and only the information bearing phasor positions, relative to one another, are presented.

2.3 Phase shift keying:

Phase shift keying (PSK) was developed during the early days of the deep space program; PSK. is now widely used in both military and commercial communications systems. The general analytic expression for PSK is

$$\mathbf{Si(t) = (2E/T)^{1/2} \cos[\omega_o t + \varphi_i(t)]}$$

$$i = 1,\ldots\ldots M, \quad \& \quad 0 \le t \le T$$

where the phase term, φi(t),will have M discrete values, typically given by

$$\boldsymbol{\varphi_i(t) \;\; = 2\pi i/M} \quad i=1,\ldots.M$$

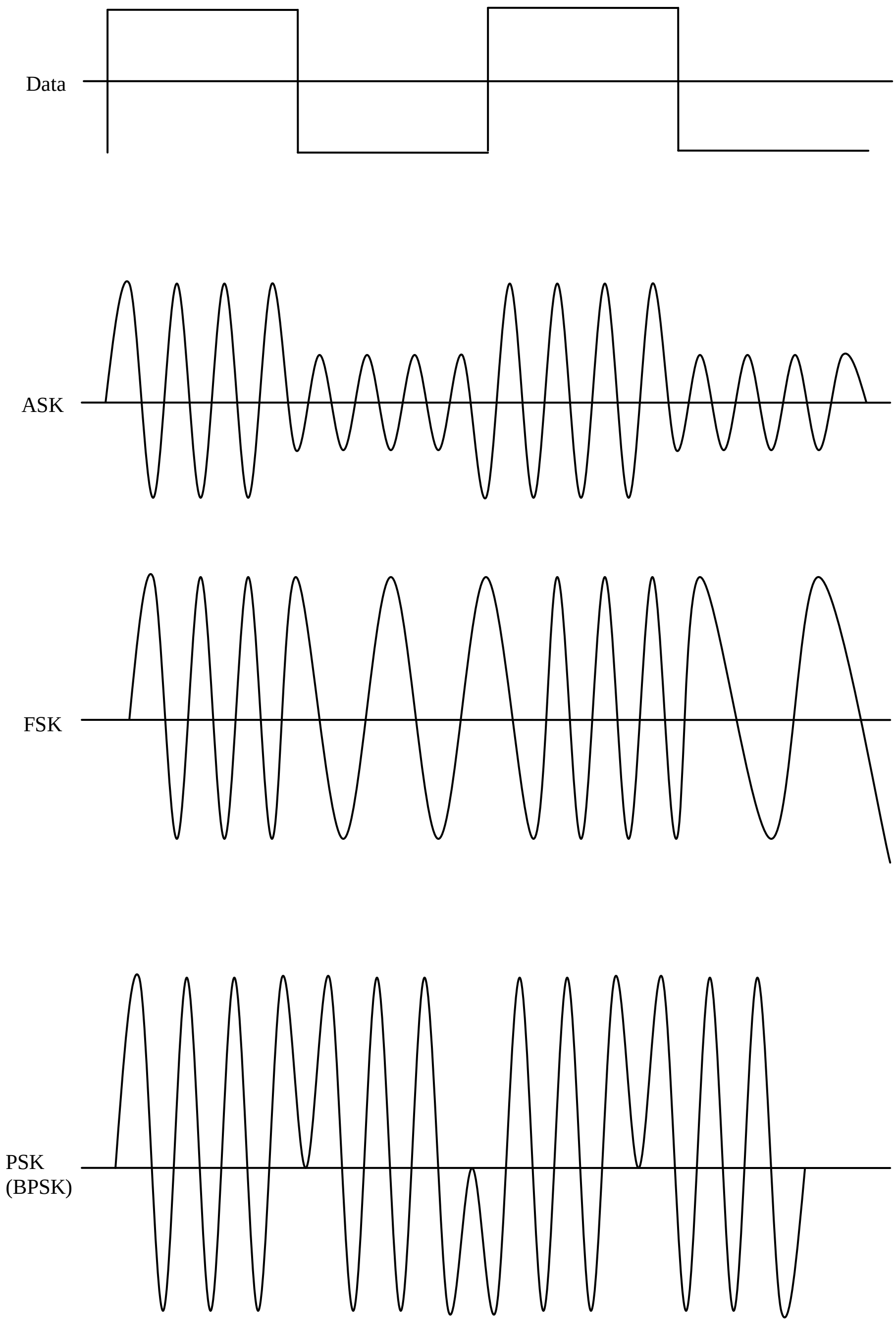

Fig. 8 Digital modulation

For the binary PSK (BPSK) example in figure 5, M is 2. The parameter E is symbol energy, T is symbol time duration, and $0 \leq t \leq T$. In BPSK modulation, the modulating data signal shifts the phase of the waveform $S_i(t)$ to one of the two states, either zero or π (180°). The waveform sketch in figure 5 shows a typical BPSK waveform with its abrupt phase changes at the symbol transitions; if the modulating data stream were to consist of alternating ones and zero, there would be such an abrupt change at each transition. The signal waveforms can be represented as vectors or phasors on a polar plot; the vector length corresponds to the signal amplitude, and the vector direction for the general M-ary case correspond to the signal phase relative to the other M-1 signals in the set. For the BPSK example, the vector picture illustrates the two 180° opposing vectors. Signal sets that can be depicted with such opposing vectors are called antipodal signal sets.

2.4 Amplitude shift keying:

For the ASK example in figure 8, the general analytic expression is:

$$\mathbf{S_i(t) = (2Ei(t)/T)^{1/2} \cos[\omega_o t + \varphi]}$$

$$i = 1,\ldots\ldots.M, \quad \& \quad 0 \leq t \leq T$$

where the amplitude term $(2E_i(t)/T)^{1/2}$ will have M discrete values, and the phase term φ is an arbitrary constant. In figure 8, M has been chosen equal to 2, corresponding to two waveform types. The ASK waveform sketch in the figure can describe a radar transmission example, where the two signal amplitude states would be $(2E/T)^{1/2}$ and zero. The vector picture utilizes the same phase-amplitude polar coordinates as the PSK example. Here we see a vector corresponding to the maximum-amplitude state, and a point at the origin corresponding to the zero amplitude state. Binary ASK signaling was one of the earliest forms of digital modulation used in radio telegraphy at the beginning of this century. Simple ASK is no longer widely used in digital communications systems, and thus it will not be treated in detail here.

2.5 Amplitude phase keying:

The combination of ASK and PSK (APK) example in figure 8, the general analytic expression:

$$Si(t) = (2E_i(t)/T)^{1/2} \cos[\omega_o t + \varphi(t)]$$

$$i = 1, \ldots\ldots M, \quad \& \quad 0 \leq t \leq T$$

Illustrates the indexing of both the signal amplitude term and the phase term. The APK waveform picture in figure 8 illustrates some typical simultaneous phase and amplitude changes at the symbol transition times. For this example, M has been chosen equal to 8, corresponding to eight waveforms (8-ray). The figure illustrates a hypothetical eight-vector signal set on the phase-amplitude plane. Four of the vectors are at one amplitude, and the other four vectors are at different amplitude. Each of the vectors is separated by 45°. When the set of M symbols in the two dimensional signal space are arranged in a rectangular constellation, the signaling is referred to as quadrature amplitude modulation (QAM).

The vector picture for each of the modulation types described in figure 8 (except the FSK case) is characterized on a plane whose polar coordinates represent signal amplitude and phase. The FSK case assumes orthogonal FSK and is characterized in a Cartesian coordinate space, with each axis representing a frequency tone ($\cos\omega_i t$) from the M-ary set of orthogonal tones.

2.6 Frequency shift keying:

The general analytical expression for FSK modulation is:

$$S_i(t) = (2E/T)^{1/2} \cos[\omega_o t + \varphi_i]$$

$$i = 1, \ldots\ldots M, \quad \& \quad 0 \leq t \leq T$$

where the frequency term ω_i has M discrete values, and the phase term φ is an arbitrary constant. The FSK waveform sketch in figure 8 illustrates the typical frequency changes at the symbol transitions. At the symbol transitions, the figure depicts a gentle shift from one frequency (tone) to another. This behavior is only true for a special class of FSK called continuous phase FSK (CPFSK). In the general MFSK case, the change to a different tone can be quite abrupt, because there is no requirement for the phase to be continuous. In this example, M has been chosen equal to 3, corresponding to the same number of waveform types (3-ary); note that this M=3 choice for FSK has been selected to emphasize the mutually perpendicular axes. In practice, M is usually nonzero power of 2 (2, 4, 8, 16…..). The signal set is characterized by Cartesian coordinates, such that each of the mutually perpendicular axes presents a sinusoid with a different frequency. As described earlier, signal sets that can be characterized with such mutually perpendicular vector are called orthogonal signals. Not all FSK signaling is orthogonal.

For any FSK signal set, in the process of meeting this criterion, a condition arises on the spacing between the tones in the set.

Frequency shift keying (FSK) is a relatively simple low performance form of a digital modulation. Binary FSK is a form of constant-amplitude angle modulation similar to conventional frequency modulation except that the modulating signal is a binary pulse stream that varies between two discrete voltage levels rather than a continuously changing analog wave form. The general expression for a binary FSK signal is

$$\mathbf{v(t)= V_c \cos [(\omega_c + V_m(t)\Delta\omega\} \, t \,]}$$

where

$v(t)$ = binary FSK wave form
V_c = peak un-modulated carrier amplitude
ω_c = radian carrier frequency
$V_m(t)$ = binary digital modulating signal
$\Delta\omega$ = change in radian output frequency

From the equation it can be seen that the binary FSK the carrier amplitude V_c remains constant with modulation. However, out put carrier radian frequency (ω_c)

shifts by an amount equal to + $\Delta\omega/2$. The frequency shifts ($\Delta\omega/2$) is proportional to the amplitude and polarity of the binary input signal. For example, a binary 1 could be +1 volt, and a binary 0 could be -1 volt, producing frequency shifts of $+\Delta\omega/2$ and $-\Delta\omega/2$, respectively. In addition, the rate at with the carrier frequency shifts is equal to the rate of change of the binary input signal. (that is, the input bit rate). Thus, the output carrier frequency deviates (shifts) between $\omega_c +\Delta\omega/2$ and $\omega_c -\Delta\omega/2$ at a rate equal to fm.

2.6.1 FSK transmitter:

With binary FSK, the center or carrier frequency is shifted by the binary input data. Consequently, the output of binary FSK modulator is the step function in the time domain. As the binary input signal changes from logic 0 to logic 1, and vice versa, the FSK output shift between two frequencies; a mark or logic 1 frequency and a space or logic 0 frequency. With binary FSK, there is a change in the output frequency each time the logic condition of the binary input signal changes. Consequently, the output rate of change id equal to input rate of change. In digital modulation, the rate of change at the input to the modulator is called the bit rate and has the units of bits per second (bps). The measure of the rate of change at the out put of the modulator is called baud and is equal to the reciprocal of the time of one output signaling element. In essence, baud is line speed in symbols per second. In binary FSK, the input and output rates of change are equal; therefore, the bit rate and baud rate are equal. A simple binary FSK transmitter is shown in figure.

2.6.2 Bandwidth consideration of FSK:

As with all electronic communication systems, band width is one of the primary considerations when designing a binary FSK transmitter. FSK is similar to conventional frequency modulation and so can be described in a similar manner.

Fig shows a binary FSK modulator, which is very similar to a conventional FM modulator, and is very often a voltage controlled oscillator (VCO). The fastest input rate of change occurs when the binary input is a series of alternating 1's and 0's; namely, a square wave. Constituently, if only the fundamental frequency of the input is considered , the highest modulating frequency is equal to one half the input bit rate.

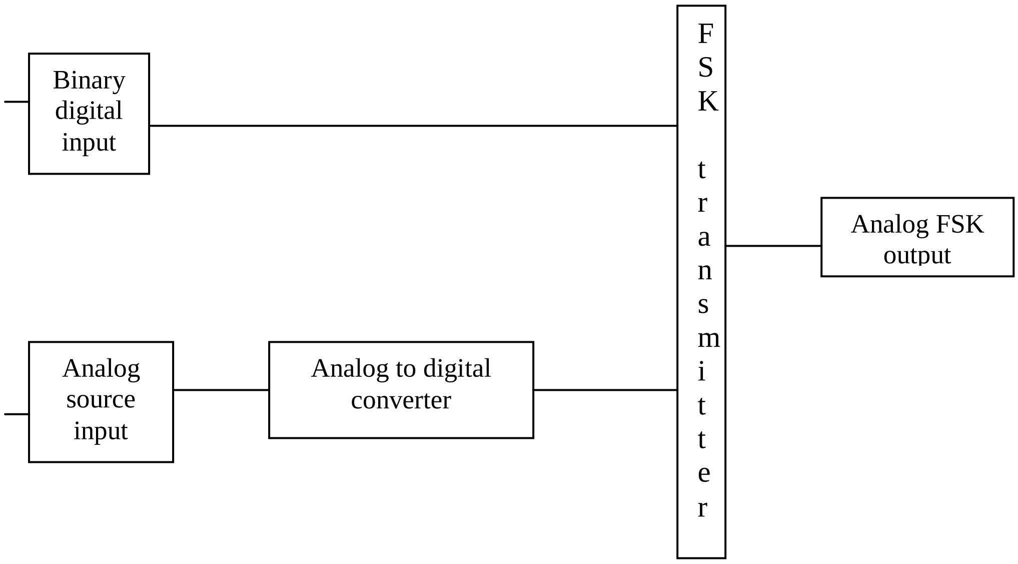

Fig.9 Binary FSK Transmitter

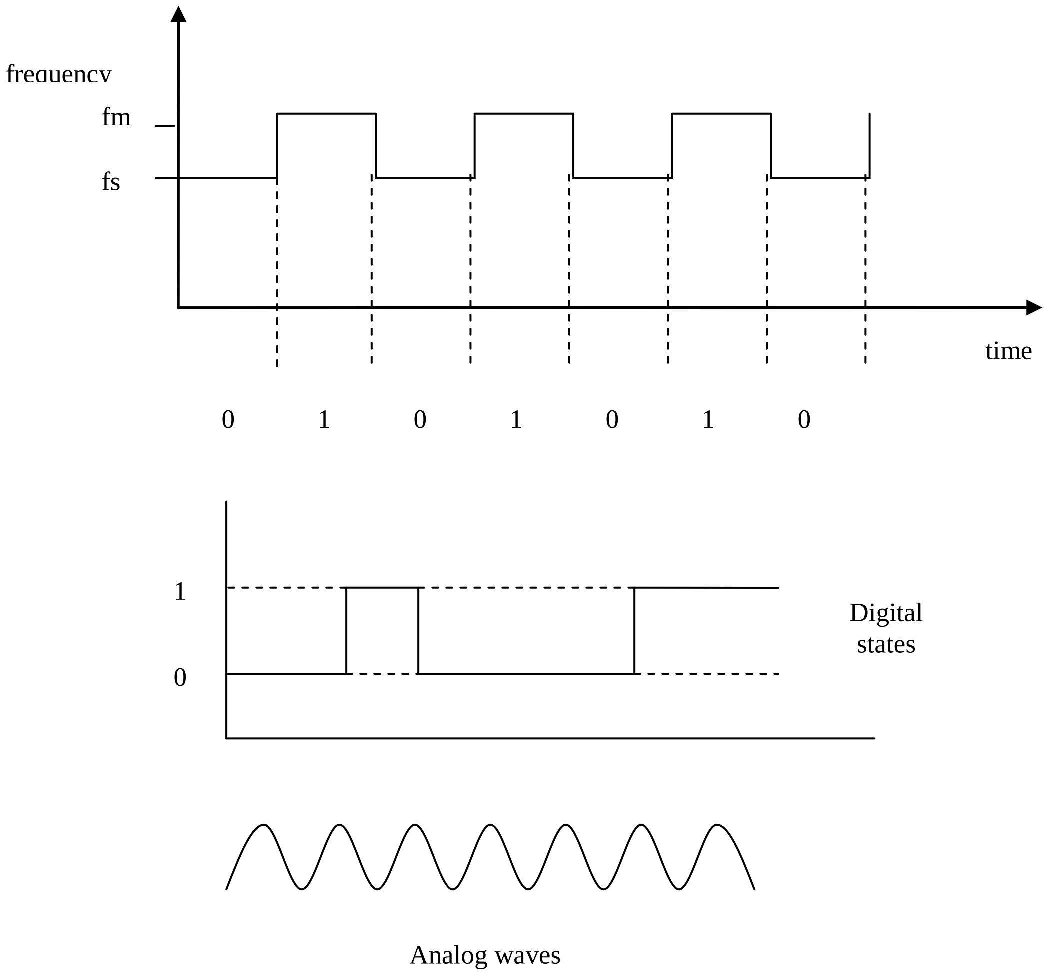

Fig 10. Binary FSK Transmitter.

The rest frequency of the VCO is chosen such that it falls half way between the mark and space frequency. A logic 1 condition at the input shift the VCO from its rest frequency ti the mark frequency, and a logic 0 condition at the input shifts the VCO from its rest frequency to space frequency. Consequently, as the input binary signal changes from a logic 1 to logic 0, and vice versa, the VCO out put frequency shifts or deviates back and forth between the mark and space frequencies.

In a binary FSK modulator. Δf is the peek frequency deviation of the carrier and is equal to the difference between the rest frequency and either the mark or space frequency.(or half the difference between the mark and the space frequencies). The peek frequency deviation depends on the amplitude of the modulating signal. In a binary digital signal, all logic 1's have the same voltage and all logic 0's have the same voltage; consequently, the frequency deviation is constant and always at its maximum value.

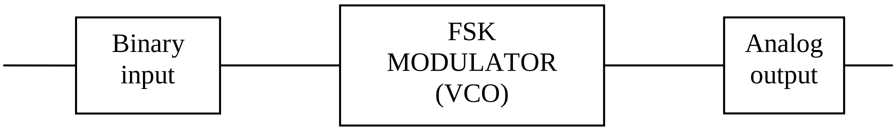

Fig. 11 FSK modulator

The output of a FSK modulator is related to the binary input as shown in the figure where a logic 0 corresponds to space frequency f_s; a logic 1 corresponds to mark frequency f_m and f_c is the carrier frequency. The required peek frequency deviation , Δf, is given as

$$\mathbf{\Delta f = \left| \frac{f_m - f_s}{2} \right| = 1/4t_b \text{ (Hz, minimum)}}$$

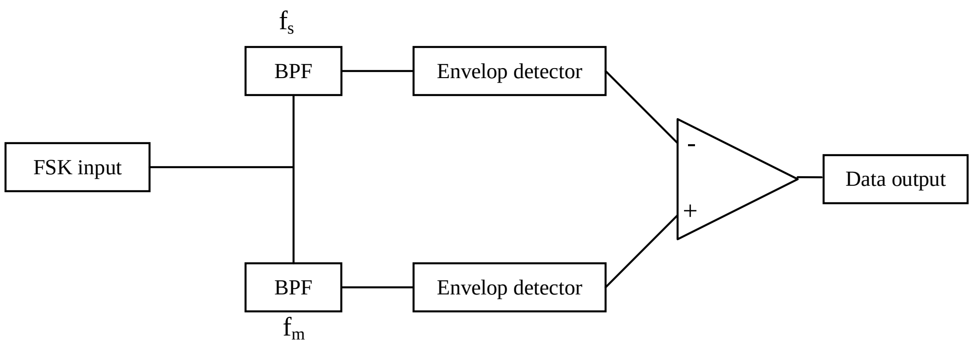

Fig. 12 Non-coherent FSK demodulator

where t_b is the time of one bit in second and fm and fs, are expressed as

$$\mathbf{f_m = f_c - \Delta f = f_c - 1/4t_b}$$

$$\mathbf{f_s = f_c + \Delta f = f_c + 1/4t_b}$$

From figure it can be seen that FSK. consists of two pulsed sinusoidal waves of frequency and Pulsed sinusoidal waves have frequency spectrums that are sin x/x functions. Assuming that the zero crossings contain the bulk of the energy, the bandwidth for FSK can be approximated as

$$\mathbf{BW = fm + 1/tb - (fs - 1/tb) = fm - fs + 2/tb}$$

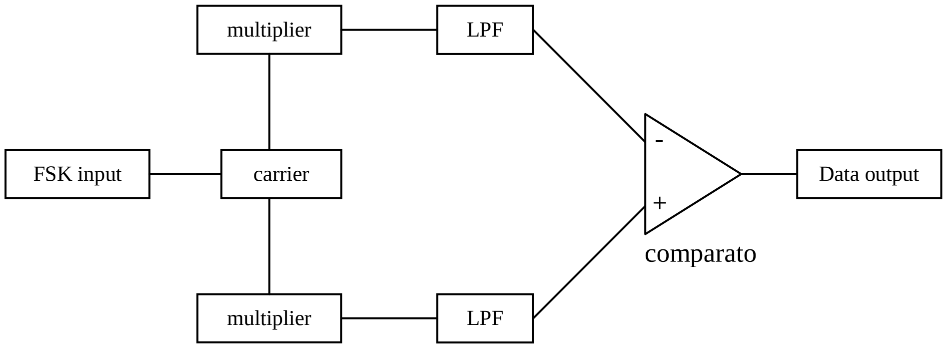

Fig. 13 Coherent FSK demodulator

FSK demodulation is quite simple with a circuit such as the one shown in Figure. The FSK input signal is applied to the inputs of both band pass filters (BPFs). The respective filter passes only the mark or only the space frequency on to its respective envelope detector. The envelope detectors, in turn, indicate the total power in each pass band and the comparator responds to the larger of the two powers. This type of FSK detection is referred to as non-coherent detection; there is no "frequency involved in the demodulation process that is synchronized either in phase, frequency, or both with the incoming FSK signal.

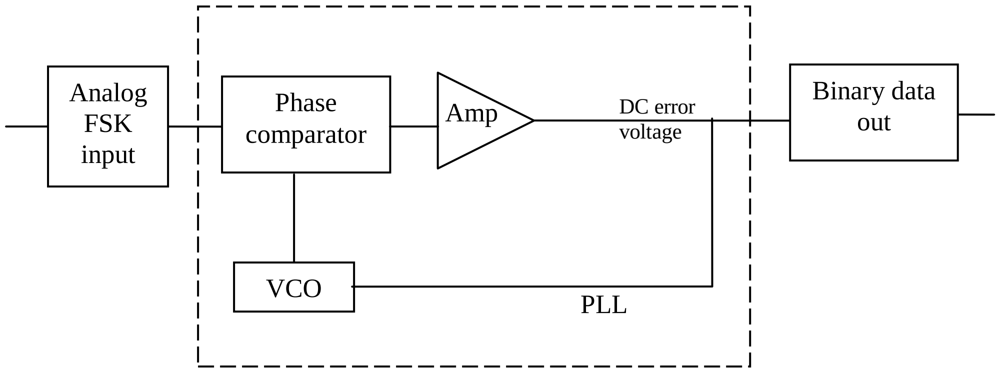

Fig.14 PLL-FSK demodulator

Figure 5 shows the block diagram for a coherent FSK receiver. The incoming FSK signal is multiplied by a recovered carrier signal that has the exact same frequency and phase as the transmitter reference. However, the two transmitted frequencies (the mark and space frequencies) are not generally continuous; it is not practical to reproduce a local reference that is coherent with both of them. Consequently, coherent FSK detection is seldom used.

The most common circuit used for demodulating binary FSK signals is the phase-locked loop (PLL), which is shown in block diagram form in Figure 5. A PLL-FSK demodulator works similarly to a PLL-FM demodulator. As the input to the PLL shifts between the mark and space frequencies, the dc error voltage at the output of the phase comparator follows the frequency shift. Because there are only two input frequencies (mark and space), there are also only two output error voltages. One represents logic 1 and the other logic 0. Therefore, the output is a two-level (binary) representation of the FSK input. Generally, the natural frequency of the PLL is made equal to the center frequency of the FSK modulator. As a result, the changes in the dc error voltage follow the changes in the analog input frequency and are symmetrical around 0 V.

Binary FSK has a poorer error performance than PSK or QAM and, consequently, is seldom used for high-performance digital radio systems. Its use is restricted to low-performance, low-cost, asynchronous data modems that are used for data communications over analog, voice-band telephone.

2.7 Minimum shift keying FSK:

Minimum shift-keying FSK (MSK) is a form of continuous-phase frequency shift keying (CPFSK). Essentially, MSK is binary FSK except that the mark and space frequencies are synchronized with the input binary bit rate. Synchronous simply means that there is a precise time relationship between the two; it does not mean they are equal. With MSK, the mark and space frequencies are selected such that they are separated from the center frequency by an exact odd multiple of one-half of the bit rate [f_m and $f_s = n (f_b/2)$, where n = any odd integer]. This ensures that there is a smooth phase transition in the analog output signal when it changes from a mark to a space frequency, or vice versa. It can be seen that when the input changes from logic 1 to logic 0, and vice versa, there is an abrupt phase discontinuity in the analog output signal. When this occurs, the demodulator has trouble following the frequency shift; consequently, an error may occur.

CHAPTER 3

PHASE SHIFT KEYING TECHNIQUES

3.1 Phase shift keying :

Phase shift keying (PSK) is another form of angle-modulated, constant-amplitude digital modulation. PSK is similar to conventional phase modulation except that with PSK the input signal is a binary digital signal and a limited number of output phases are possible.

3.2 Binary phase shift keying :

With binary phase shift keying (BPSK), two output phases are possible for a single carrier frequency ("binary" meaning "2"). One output phase represents a logic 1 and the other a logic 0. As the input digital signal changes state, the phase of the output carrier shifts between two angles that are 180° out of phase. Other names for BPSK are phase reversal keying (PRK) and bi-phase modulation. BPSK is a form of suppressed-carrier, square-wave modulation of a continuous wave (CW) signal.

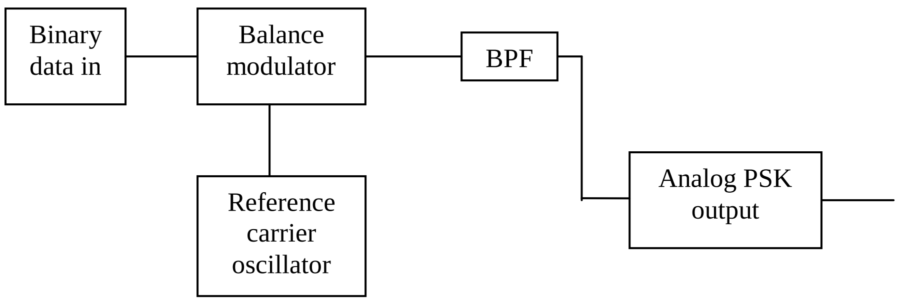

Fig 15.1. BPSK Modulator

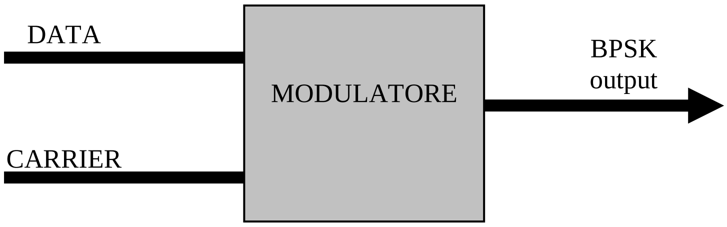

Fig 15.2 BPSK Modulator

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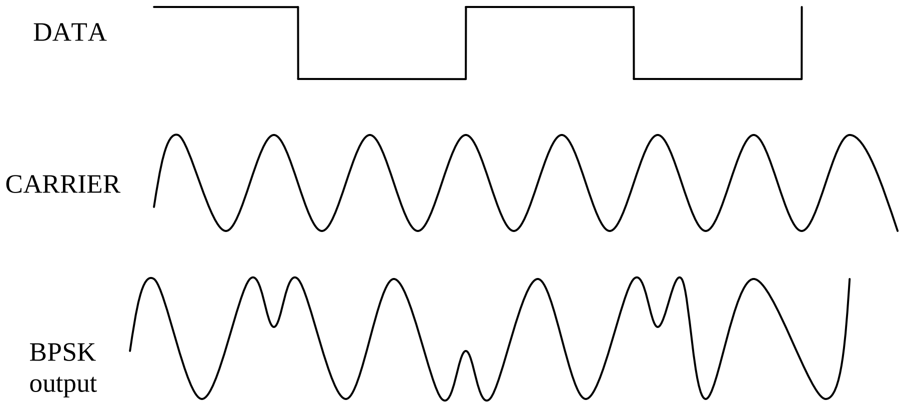


Fig 16. BPSK output wave forms

In Binary Phase Shift Keying a single data channel modulates the carrier. A single bit transition, 1 to 0 or 0 to 1, causes a 180 degree phase shift in the carrier. Thus, the carrier is said to be modulated by the data.

Figure shows a simplified block diagram of a BPSK modulator. The balanced modulator acts as a phase reversing switch. Depending on the logic condition of the digital input, the carrier is transferred to the output either in phase or 180° out of phase with the reference carrier oscillator.

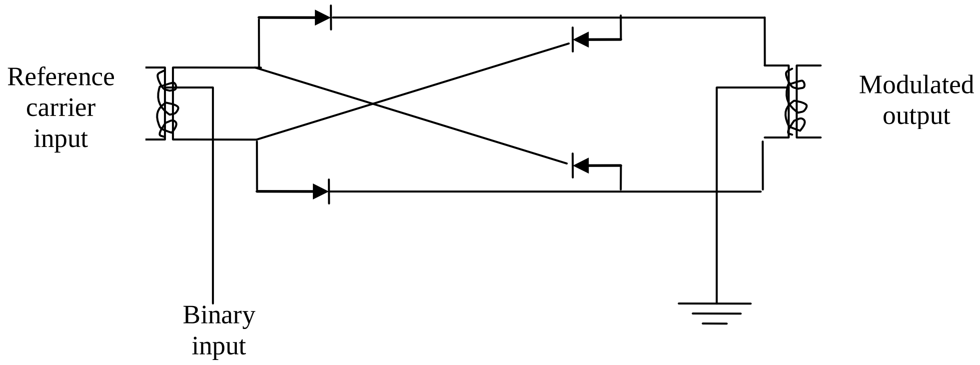


Fig 17 .BPSK circuit diagram

Figure 7 shows the schematic diagram of a balanced ring modulator. The balanced modulator has two inputs: a carrier that is in phase with the reference oscillator and the binary digital data. For the balanced modulator to operate properly, the digital input voltage must be much greater than the peak carrier voltage. This ensures that the digital input controls the on/off state of diodes Dl-D4. If the binary input is a logic 1 (positive voltage), diodes Dl and D2 are forward biased and on, while diodes D3 and D4 are reverse biased and off. With the polarities shown, the carrier voltage is developed across transformer T2 in phase with the carrier voltage across Tl. Consequently, the output signal is in phase with the reference oscillator.

If the binary input is a logic 0 (negative voltage), diodes Dl and D2 are reverse biased and off, while diodes D3 and D4 are forward biased and on. As a result, the carrier voltage is developed across transformer T2 180° out of phase with the carrier voltage across Tl. Consequently, the output signal is 180° out of phase with the reference oscillator.

If the binary input is a logic 0 (negative voltage), diodes D1 and D2 are reverse biased and off, while diodes D3 and D4 are forward biased and on. As a result, the carrier voltage is developed across transformer T2 180^0 out o phase with the carrier voltage across T1. Consequently, the output signal is 180^0 out of phase with the reference oscillator.

fig shows the truth table, phasor diagram, and constellation diagram for a

BPSK modulator. A constellation diagram, which is sometimes called a signal state-space diagram, is similar to a phasor diagram except that the entire phasor is not drawn. In a constellation diagram, only the relative positions of the peaks of the phasors are shown.

3.2.1 Band width considerations of BPSK:

A balanced modulator is a product modulator; the output signal is the product of the two input signals. In a BPSK modulator, the carrier input signal is multiplied by the binary data. If +1 V is assigned to a logic 1 and -1 V is assigned to a logic 0, the input carrier (sin $\omega_c t$) is multiplied by either a +1 or -1. Consequently, the output signal is either $+1\sin\omega_c t$ or $-1\sin\omega_c t$; the first represents a signal that is in phase with the reference oscillator, the later a signal that is 180° out of phase with the reference oscillator. Each time the input logic condition changes, the output phase changes. Consequently, for BPSK, the output rate of change (baud) is equal to the input rate of change (bps), and the widest output bandwidth occurs when the input binary data are an alternating 1/0 sequence. The fundamental frequency (f_a) of an alternative 1/0 bit sequence is equal to one-half of the bit rate ($f_b/2$). Mathematically, the output phase of a BPSK modulator is

$$\textbf{output} = \underbrace{\sin \omega_c t}_{\text{unmodulated currier}} \times \underbrace{\sin \omega_a t}_{\text{fundamental frequency}}$$

The output spectrum from a BPSK modulator is simply a double-sideband, suppressed-carrier signal where the upper and lower side frequencies are separated from the carrier frequency by a value equal to one-half of the bit rate. Consequently, the minimum bandwidth (f_n) required to pass the worst-case BPSK output signal is equal to the input bit rate.

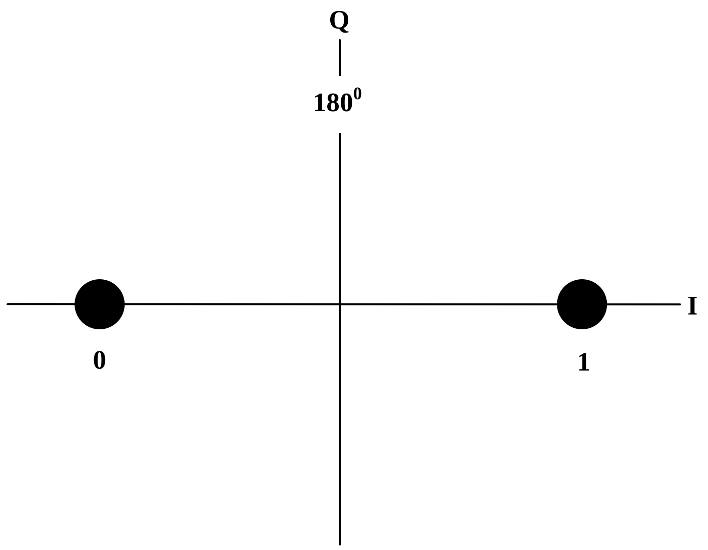


Fig 18. Phasor representation of BPSK

Binary Input	Output Phase
Logic 1, Logic 0	0 Degree, 180 Degree

Fig 19. BPSK Truth Table

3.2.3 BPSK RECIEVER :

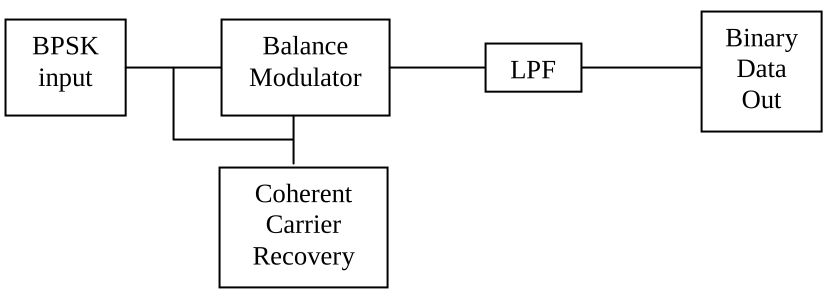


Fig 20. BPSK demodulator

The input signal may be $+\sin\omega_c t$ or $-\sin\omega_c t$. The coherent carrier recovery circuit detects and regenerates a carrier signal that is both frequency and phase coherent with the original transmit carrier. The balanced modulator is a product detector; the output is the product of the two inputs (the BPSK signal and the recovered carrier). The low-pass filter (LPF) separates the recovered binary data from the complex demodulated signal. Mathematically, the demodulation process is as follows.

For a BPSK input signal of $+\sin\omega_c t$ (logic 1), the output of the balanced modulator is

$$\textbf{Output} = (\sin\omega_c t)(\sin\omega_c t) = \sin^2\omega_c t$$

Leaving **Out put = +1/2 V = logic 1**

It can be seen that the output of the balanced modulator contains a positive voltage [+(1/2)V] and a cosine wave at twice the carrier frequency ($2\,\omega_c$). The LPF has a cutoff frequency much lower than $2\,\omega_c$ and, thus, blocks the second harmonic of the carrier and passes only the positive constant component. A positive voltage represents a demodulated logic 1. For a BPSK input signal of $-\sin\omega_c t$ (logic zero), the output of the balanced modulator is

$$\textbf{Output} = (-\sin\omega_c t)(\sin\omega_c t) = -\sin^2\omega_c t$$

Leaving **Out put = -1/2 V = logic 0**

The output of the balanced modulator contains a negative voltage [-(1/2)V] and a cosine wave at twice the carrier frequency ($2\,\omega_c t$). Again, the LPF blocks the second harmonic of the carrier and passes only the negative constant component. A negative voltage represents a demodulated logic 0.

3.3 M-ary Encoding:

M-ary is a term derived from the word "binary." M is simply a digit that represents the number of conditions possible. The two digital modulation techniques discussed thus far (binary FSK and BPSK) are binary systems; there are only two possible output conditions. One represents a logic 1 and the other a logic 0; thus, they are M-ary systems where M = 2. With digital modulation, very often it is advantageous to encode at a level higher than binary. For example, a PSK system with four possible output phases is an M-ary system where M = 4. If there were eight possible output phases, M - 8, and so on. Mathematically,

$$\mathbf{N = log\ 2\ M}$$

where, N - number of bits

M = number of output conditions possible with N bits

For example, it 2 bits were allowed to enter a modulator before the output were allowed to change,

$2 = \log_2 M$ and $2^2 = M$ thus, $M = 4$

An M = 4 indicates that with 2 bits, four different output conditions are possible. For N=3, M = 2^3 or 8, and so on.

3.4 Quaternary phase shift keying

Quaternary phase shift keying (QPSK), or quadrature PSK as it is sometimes called, is another form of angle-modulated, constant-amplitude digital modulation. QPSK is an M-ary encoding technique where M= 4 (hence, the name "quaternary," meaning "4"). With QPSK four output phases are possible for a single carrier frequency. Because there are four different output phases, there must be four different input conditions. Because the digital input to a QPSK modulator is a binary (base 2) signal, to produce four different input conditions, it takes more than a single input bit. With 2 bits, there are four possible conditions: 00, 01, 10, and 11. Therefore, with QPSK, the binary input data are combined into groups of 2 bits called dibits. Each dibit code generates one of the four possible output phases. Therefore, for each 2-bit dibit clocked into the modulator, a single output change occurs. Therefore, the rate of change at the output (baud rate) is one-half of the input bit rate.,

3.4.1 QPSK Transmitter :

A block diagram of a QPSK modulator is shown in Figure 10. Two bits (a dibit) are clocked into the bit splitter. After both bits have been serially inputted, they are simultaneously parallel outputted. One bit is directed to the I channel and the other to the Q channel. The I bit modulates a carrier that is in phase with the reference oscillator (hence, the name "I" for "in phase" channel), and the Q bit modulates a carrier that is 90° out of phase or in quadrature with the reference carrier (hence, the name "Q" for "quadrature" channel).

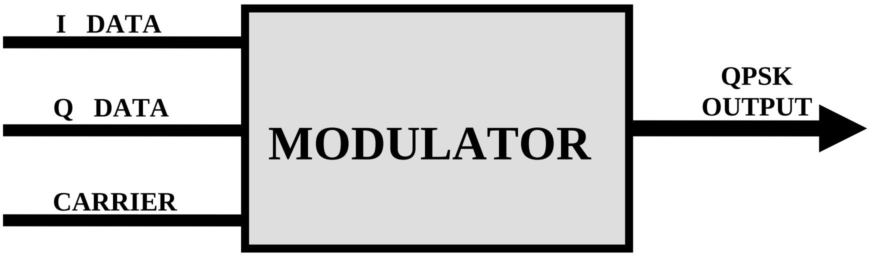


Fig 21. QPSK modulator

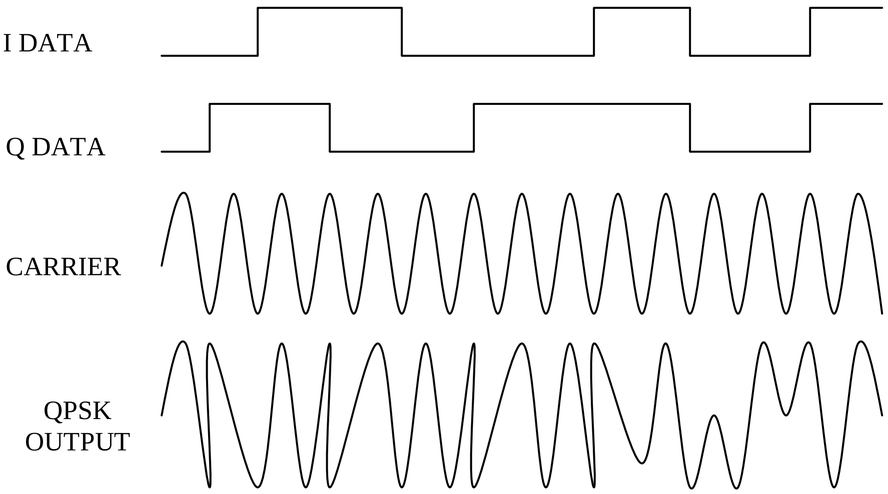


Fig 22 QPSK output wave forms

Quadrature Phase Shift Keying- Two data channels modulate the carrier. Transitions in the data cause the carrier to shift by either 90 or 180 degrees. This allows customers to transmit two discrete data streams, identified as I channel (In phase) and Q channel (Quadrature) data.

It can be seen that once a digit has been split into the I and Q channels, the operation is the same as in a BPSK modulator. Essentially, a QPSK modulator is two BPSK modulators combined in parallel. Again, for a logic 1 = +1 V and a logic 0 = -1 V, two phases are possible at the output of the I balanced modulator (+sin *Wgt* and -sin $w_c t$), and two phases are possible at the output of the Q balanced modulator (+cos $w_c t$ and -cos $w_c t$). When the linear summer combines the two quadrature (90° out of phase) signals, there are four possible resultant phasors given by these expressions:

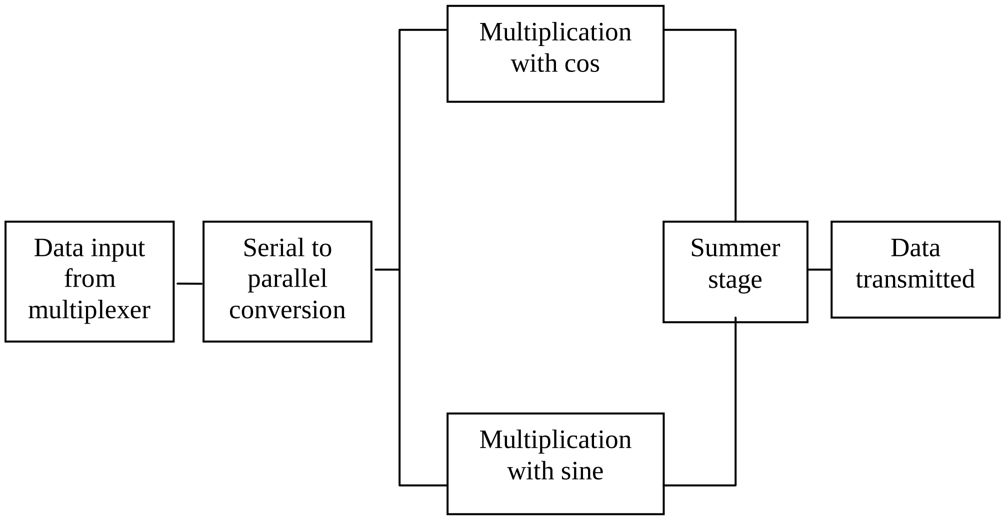


Fig 23. QPSK modulator block diagram

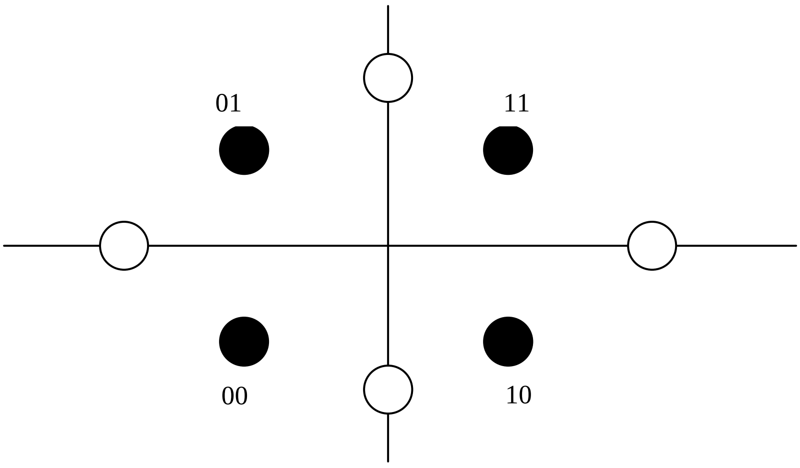


Fig 24. QPSK constellation diagram

Binary input Q	I	QPSK output
0	0	-135 degree
0	1	-45 degree
1	0	135 degree
1	1	45 degree

Fig 25. QPSK truth table

In Figure 12 it can be seen that with QPSK each of the four possible output phasors has exactly the same amplitude. Therefore, the binary information must be encoded entirely in the phase of the output signal. This constant amplitude characteristic is the most important characteristic of PSK that distinguishes it from QAM, which is explained later in this chapter. Also, from figure it can be seen that the angular separation between any two adjacent phasors in QPSK is 90°. Therefore, a QPSK signal can undergo almost a +45° or -45° shift in phase during transmission and still retain the correct encoded information when demodulated at the receiver. Figure 13 shows the output phase-versus-time relationship for a QPSK. modulator.

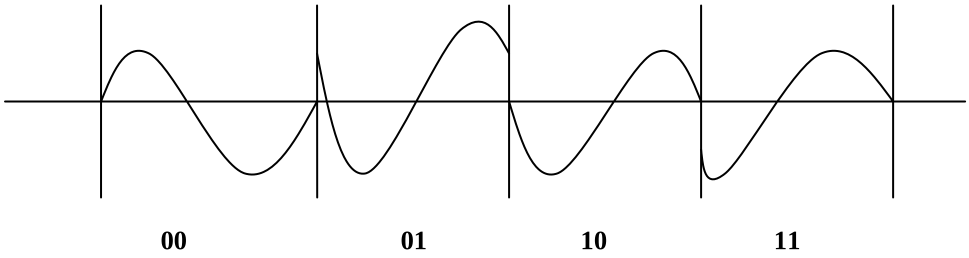
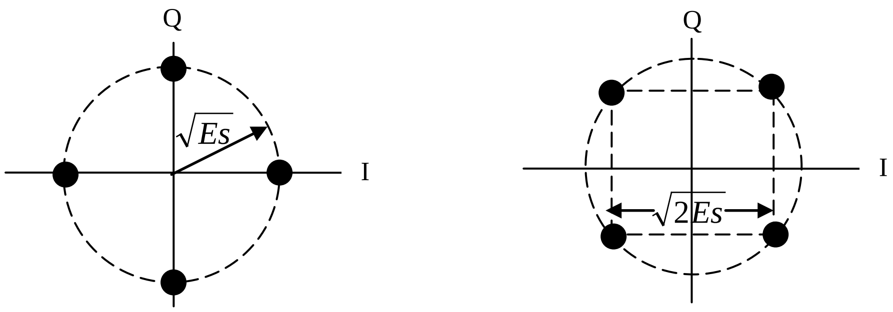


Fig 26. Output phase verses time relationship of QPSK modulator



(a) QPSK where carrier phases are 0,90,180,270
(b) QPSK where carrier phases are 45,135,225,315

Fig 27. Energy diagram of QPSK constellation

3.4.2 Bandwidth considerations of QPSK:

With QPSK because the input data are divided into two channels the bit rate in either the I or the Q channel is equal to one-half of the input data rate (fb/2). (Essentially, the bit splitter stretches the I and Q bits to twice their input bit length.) Consequently, the highest fundamental frequency present at the data input to the 1 or the Q balanced modulator is equal to one-fourth of the input data rate. As a result, the output of the I and Q balanced modulators requires a minimum double-sided Nyquist bandwidth equal to one-half of the incoming bit rate. Thus, with QPSK, a bandwidth compression is realized (the minimum bandwidth is less than the incoming bit rate). Also, because the QPSK output signal does not change phs.se until 2 bits (a dibit) have been clocked into the bit splitter, the fastest output rate of change (baud) is also equal to one-half of the input bit rate. As with BPSK, the minimum bandwidth and the baud are equal.

Carrier Phase Shifts Corresponding to Various Input Bit Pairs

Information bits	Phase shift
1 1	π/4
0 1	3 π /4
0 0	-3 π/4
1 0	- π /4

The worse-case input condition to the I or Q balanced modulator is an alternative 1/0 pattern, which occurs when the binary input data has a 1100 repetitive pattern. One cycle of the fastest binary transition (a 1/0 sequence) in the I or Q channel takes the same time as 4 input data bits. Consequently, the highest fundamental frequency at the input and fastest rate of change at the output of the balanced modulators is equal to one-fourth of the binary input bit rate.

The output of the balanced modulators can be expressed mathematically as

$$\textbf{Output} = (\sin w_c t)(\sin w_a t)$$

The output frequency spectrum extends from f_c+fb/4 to f_c-fb/4 and the minimum bandwidth *(/N)* is

$$(fc+fb/4) - (fc-fb/4) = f_b/2$$

3.4.3 QPSK Receiver :

The block diagram of a QPSK receiver is shown in Figure 16. The power splitter directs the input QPSK signal to the I and Q product detectors and the carrier recovery circuit. The carrier recovery circuit reproduces the original transmit carrier oscillator signal. The recovered carrier must be frequency and phase coherent with the transmit reference carrier. The QPSK signal is demodulated in the I and Q product detectors, which generate the original I and Q data bits. The outputs of the product detectors are fed to the bit combining circuit, where they are converted from parallel I and Q data channels to a single binary output data stream.

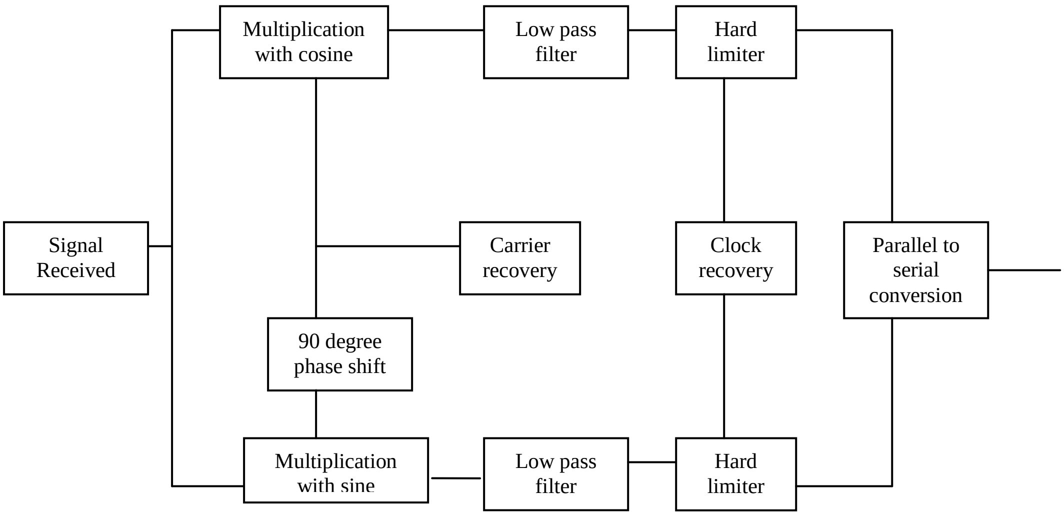


Fig 28. Block diagram of QPSK receiver

The incoming QPSK signal may be any one of the four possible output phases shown in Figure. To illustrate the demodulation process, let the incoming QPSK signal be -sin $w_c t$ + cos $w_c t$. Mathematically, the demodulation process is as follows.

The receive QPSK signal (-sin $w_c t$ + cos $w_c t$) is one of the inputs to the I product detector. The other input is the recovered carrier (sin $w_c t$). The output of the I product detector is

$$\mathbf{I=(-\sin w_c t + \cos w_c t)(\sin w_c t)}$$

After filtering **I= -1/2 V (logic 0)**

Again, the received QPSK signal (-sin w_ct + cos w_cf) is one of the inputs to the Q product detector. The other input is the recovered carrier shifted 90° in phase (cos w_ct). The output of the Q product detector is

$$\mathbf{Q=(-\sin w_c t + \cos w_c t)(\cos w_c t)}$$

After filtering **I= +1/2 V (logic 1)**

The demodulated I and Q bits (0 and 1, respectively) correspond to the constellation diagram and truth table for the QPSK modulator shown in Figure

3.5 Errors in digital communication:

It is noted that one of the most important advantages of digital communications is that it permits very high fidelity. In this section we shall investigate this more closely. We shall consider in detail only BPSK systems, and comment on the alternative modulations.

In the absence of noise, the signal, V, from a BPSK system can take one of two values, . In the ideal case, if the signal is greater than 0, the value that is read is assigned 1. If the signal is less than 0, then value that is read is assigned 0. When noise is present, this distinction between + v_b(with the threshold at 0) becomes blurred. There is a finite probability of the signal dropping below 0, and thus being assigned 0, even though a 1 was transmitted. When this happens, we say that a bit-error has occurred. The probability that a bit-error will occur in a given time is referred to as the bit-error rate (BER). In actuality, we may decide that our threshold of deciding whether the signal is interpreted as a 0 or a 1 is set at vb/2 such that any signal detected between a 0 is read if –vb< V< 0 and a 1 is read if vb< V< 0
We suppose (without loss) that the signal V, which has the signal levels. + vb noise N of variance σ^2 The probability that an error will occur in the transmission of a 1 is:

P = {N + vb<0} = P { N < -vb}
= ½ -1/2crf(vb)

$$= \frac{1}{2}\,\mathrm{crfc}(vb)$$

Similarly the probability that an error will occur in the transmission of a 0 is:

$$\frac{1}{2}\,\mathrm{crfc}(vb)$$

This is an important result as it gives us an expression for the probability of error without reference to which value (a 1 or a 0) is transmitted. It is usual to write these expressions in terms of the ratio of (energy per bit) to (noise power per unit Hz). The power S in the signal is, on average and the total energy in the signaling period T is vb^2t . The average energy per bit is therefore:

$$E_b = (v_b^2T + v_b^2T)/2 = v_b^2T$$

For BPSK, the signaling period T is half the reciprocal of the bandwidth B, i.e. 1/2T=B; thus with using the expressions we have:

$$P(\mathrm{error}) = \frac{1}{2}\,\mathrm{crfc}\left(\sqrt{Eb/En}\right)$$

All coherent detection schemes give rise to error rates of the form in equation. For example, QPSK has twice the error probability of BPSK, reflecting the fact that with a quadrature scheme, there are more ways an error can occur. Narrow-band FSK. has an error probability rather worse than QPSK, although its numerical value depends on the exact scheme used.

Incoherent demodulation schemes always have a higher probability of error than coherent schemes. Incoherent schemes are forms of power detection, i.e. Produce an output proportional to the square of the input. Power detection always decreases the SNR. It is quite easy to see why this is so. Suppose the input, X, is of the form X = V + N, as before. The input SNR is

$$SNR_{in} = V^2/N^2$$

If we square the input, the output is

$$X^2 = (V + N)^2 = V^2 + 2VN + N^2$$

$$VN >> N^2$$

Assuming the SNR is high, and the SNR of the output is

$$SNR_{out} = (V^2)^2/(2VN)^2$$

$$V^2/4N^2$$

$$SNRin/4$$

This decrease in the signal-to-noise ratio causes an increase in the error probability. Although poorer, however, their performance is good nonetheless. This explains the widespread use of incoherent FSK.

Error rates are usually quoted as bit error rates (BER). The conversion from error probability to BER is numerically simple: BER = P(error). However, this conversion assumes that the probabilities of errors from bit-to-bit are independent. This may or may not be a reasonable assumption. In particular, loss of timing can cause multiple bit failures that can dramatically increase the BER. When signals travel along the channel, they are being attenuated. As the signal is losing power, the BER increases with the length of the channel. Regenerators, placed at regular intervals, can dramatically reduce the error rate over long channels. To determine the BER of the channel with N regenerators, it is simplest to calculate first the probability of no error. This probability is the probability of no error over one regenerator, raised to the Nth power:

$$P\{\text{No error over N regenerators}) = (1 - P(\text{error}))^N$$

Assuming the regenerators are regularly spaced and the probabilities are independent. The BER is then determined simply by:

$$P[\text{ error over N regenerators}] = 1 - P[\text{No error over N regenerators}]$$

This avoids having to enumerate all the ways in which the multiple systems can fail.

CHAPTER 4

HARDWARE DESIGN OF MODEM

As it is with all the fields of practical sciences and engineering the practical side of the picture is a bit different from theory and the major portion of implementing some thing practically means fighting the practical constraints.

4.1 Modulator Design :

4.1.1 Carrier Generator :

For any band pass modulation the basic need is the carrier wave. There could be two possible solutions for generation of carrier. Either place an independent circuit that generates the carrier wave of desired frequency or use of the frequency generator equipment. So in this hardware design instead of placing and independent circuit the carrier has been taken from the signal generator. The sinusoidal (fm) wave of 2MHz is used as a carrier wave for modulation process.

4.1.2 Clock Generation :

The schematics of the practical modem are formed keeping in mind that it is a digital one and thus the foremost requirement of the circuit would be a stable clock. The clock is generated using the standard sine wave. It is used for both clock generation and as a carrier.

The clock signal for the modulator has been derived from the carrier signal. The IC used is 4520. The IC is basically a divider circuit. The output of the IC gives 4 outputs, that is fm/2,fm/4,fm/8 and fm/16. Three outputs from the four have been utilized. These are fm/2,fm/8 and fm/16 from pin no 3,5 and 16. the input of 2MHz was given on pin 1 of the IC.

4.1.3 Data Input to Modulator :

The data input for the modulator has been taken from any serial device, as in most of the cases the data will not be available in two discreet bit streams. Normally a PCM stream is available. This serial input can be assumed as a PCM stream. The data is then converted from serial to parallel using IC 4015. The CD4015BC contains two identical, 4-stage, serial input parallel-output registers with independent “Data”, “Clock,” and “Reset” inputs. The logic level present at the input of each stage is transferred to the output of that stage, at each positive-going clock transition. Logic high on the “Reset” input resets all four stages covered by that input. All inputs are protected from static discharge by a series resistor and diode clamps to VDD and VSS.

Since it has two identical shift registers, so any one of them can be used for conversion of data. From this IC the data stream has been broken in to two discreet data streams. These data streams can be represented as I channel and Q channel respectively.

4.1.4 Formation of Digits :

The basic requirement of any QPSK modulator is the input in form of Dibits. So it is there necessary to make dibits from the incoming PCM stream. Although the data stream has already been converted from serial to parallel, still there is a need to make the dibits from these two data streams. The dibits actually gives the value of 1 or -1 which allows the modulator to judge and give a phase shift to the carrier signal as required. Since the dibits are coming from two channels so we can have four possible combination. That are (1,1),(1,-1),(-1,-1),(-1,1). Now plotting them on graph gives us the position in the phasor diagram form making an angle difference of 90 degrees from each other, means 45,135,225 and 315 degrees.

4.1.5 4 Phase Inputs:

The carrier frequency now onward used for the QPSK modem is fm/8. Two inputs fm/2 and fm/8 are fed to IC 4015. Here fm/2 is used as the clock pulse and fm/8 as the data. Since the out put is the data depending on the clock we get four outputs of the same frequency but of different phases, with a phase difference of 90 degrees. These four carrier waves are then fed to the modulator IC 4512 .

4.1.6 Modulation :

Modulation is done with the help of IC 4512. The input of the IC are the four carrier waves with 90 degree of phase shift and two input of the Dibits. The carrier keep a track of incoming data that is the dibits and the phase and gives the output from pin 14. This output is the modulated data and have a general shape of pulses.

4.1.7 Pulse shaping and Filtering :

Pulse shaping and removal of the DC component is a necessary issue. It is done with the help of LC filter. A filter is placed at the output of the modulator to shape the pulse and to remove the DC components. After passing through the filter circuit the signal goes weak and thus need the amplification.

At this stage where the modulator and demodulator are not placed far a part the amplifier has not been used, however a simple no inverting amplifier for the desired frequency can also be used for batter performance of the hardware.

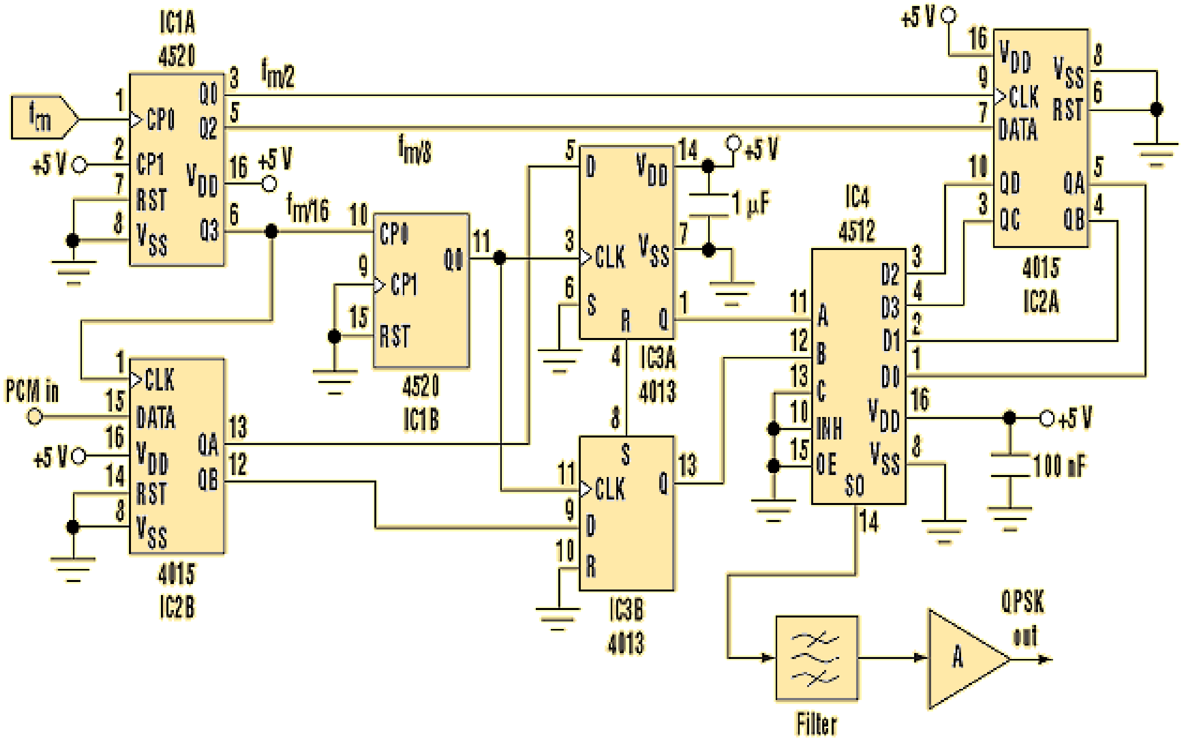


Fig 29. Hardware Diagram of QPSK modulator

4.8 Demodulator Design:

4.8.1 Amplifier circuit :

Since the incoming signal into the demodulator is quite week so an amplifier circuit is necessary to make it amplified enough to carrey out recovery of the data. In this demodulator circuit a simple no inverting amplifier has been used which amplifies the signal before actually putting it into demodulator stage.

4.8.2 Phase generator :

First step in any demodulator is the generation of same carrier frequency that was used in the modulator. Due to unstable components available in the market two different frequency divider circuit has not been used, rather the required frequency has been taken from the modulator block.

The frequency fm/2 and fm/8 has been used and fed to IC 4015 as it was used in modulator circuit to generate frequency with phase difference of 90 degrees. These four outputs has been passed on to IC 4052 which is dual four channel analog multiplexed and demultiplexer. These inputs are given on pin 11,15,14 and 12. the other inpurts to the IC are from pin 9 and 10. These inputs are from IC 4520 which is actually giving a feed back from the output of the multiplexer as a clock and UART as an enabling input. The output of multiplexer IC 4052 is also feed to IC 4015. 4015 uses clock of fm/2 and output of multiplexer as data and gives 4 outputs. 3 of them are used as frequency with 0 phase is not required.

4.8.3   Bits generator :

After amplification the input signal of frequency fm/8 is sent to IC 4013, which is basically a shift register IC. The output of the IC using one of the inner combination is then feed to 3 XOR gates. Other inpout to XOR gates is the output frequency from the phase generator. Now where both the inouts of XOR are same the output comes to be 1, otherwise it is 0. so that means when the phases are same on the inpout of gate the output comes to be 1. the outputs from these three XOR oare feed to two TL082 IC which is working as a comparator at this stage, the three outputs are compared with each other to give the true combination of output bits.

4.8.4 Bit timing recovery :

Bit timing recovery is again an issue to be handled carefully. The phase generator also provides the timing clock using IC 4520 and the multiplexed output from IC 4052. This timing clock is applied to IC 4076 along with the bits information. The IC is a flip flop and passes on the bits keeping in view the timing provided by the applied clock.

4.8.5 PCM Output :

The output of the IC 4076 is then passed on to multiplexer IC 4052 as in put to be multiplexed with the clock signal same as for bit timing. This IC multiplex the two input bit streams into a single output in form of PCM stream. This is the demultiplexed data available as a final output of demodulator.

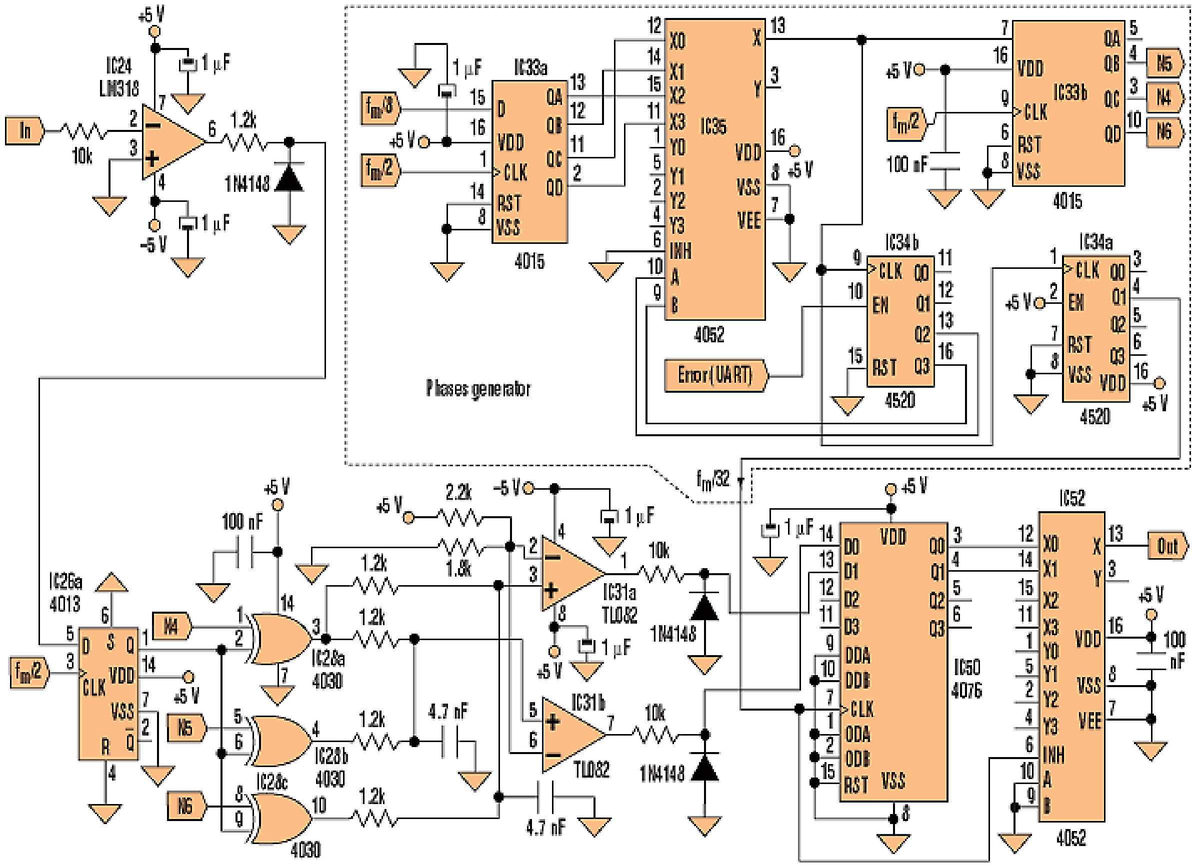


Fig 30. Hardware Diagram of QPSK demodualtor

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