Communications Perhubungan

EEE 332

Semester II 2010/2011

Lecturer

Dr Mohd Fadzil Bin Ain

Room: 3.25 or 1.21 (Programme Chairman Room)

Telephone: ext. 6603

Email: mfadzil@eng.usm.my

Course Outline

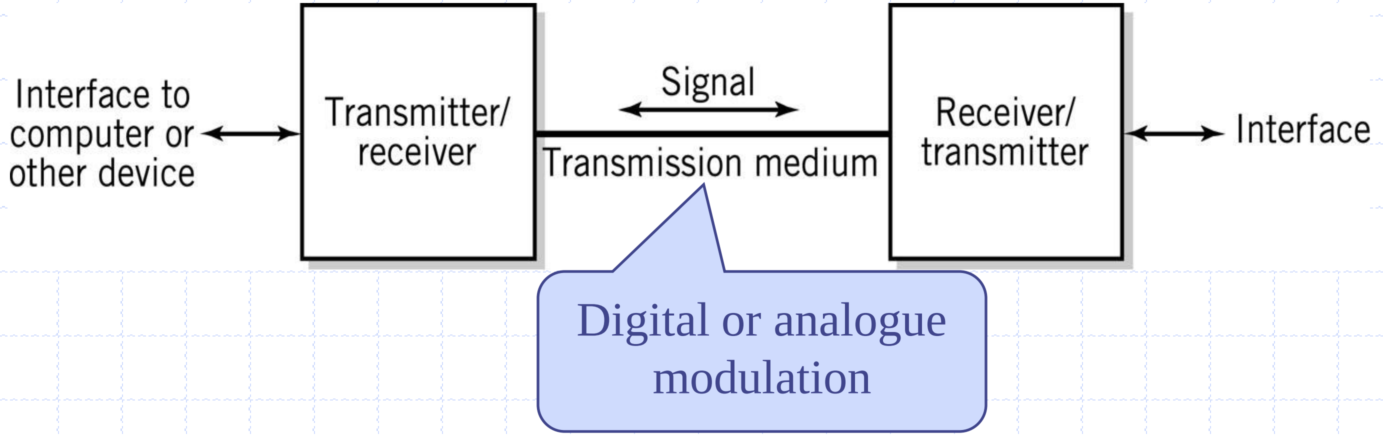
- Digital Modulation Techniques
  - Information capacity, bits, Bit Rate, Baud and M-*ary* Encoding
  - Amplitude Shift Keying
  - Frequency Shift Keying
  - Phase Shift Keying
  - Quadrature Amplitude Modulation
  - Bandwidth Efficiency
  - Carrier Recovery
  - Clock Recovery
  - Differential Phase Shift Keying
  - Probability of error and Bit Error Rate
  - Error Performance

Introduction

- Electronic communication is the transmission, reception and processing of information with the use of electronic circuits.
- Traditional electronic communications system that use analogue modulation are rapidly being replaced with modern digital modulation system.

What is digital communication?

- *Digital communication* is a rather ambiguous term that could have different meanings to different people.
- The property that distinguishes digital communication system from conventional analogue system is the nature the modulating signal.



Why Digital Modulation?

- Digital modulation offers many advantages over analogue modulation:
  - Increased channel capability
  - Greater accuracy in the presence of the noise and distortion
  - Ease of handling
- In the digital transmission, bits are transmitted at a rate of kilobits, megabits or gigabits per second and a certain number of bit represent a symbol.
- It is unimportant if the amplitude or shape of the received signal is distorted as long as the receiver can clearly distinguish one symbol from the other.

Why Digital Modulation?

Power Efficiency

- Ability of a modulation technique to preserve the fidelity of the digital message at low power levels
- Designer can increase noise immunity by increasing signal power
- Power efficiency is a measure of how much signal power should be increased to achieve a particular BER for a given modulation scheme

Bandwidth Efficiency

- Ability to accommodate data within a limited bandwidth
- Tradeoff between data rate and pulse width

Information capacity

- ***Information capacity*** is a measure of how much information can be propagated through a communication system and is a function of bandwidth and transmission time.
- ***Information capacity*** is represents the number of independent symbols that can be carried through a system in a given unit of time.
- It is often convenient to express the information capacity of a system as a ***bit rate***.

Information capacity

- Shannon limit for information capacity is:

$$I = B\log_2\left(1+\frac{S}{N}\right)$$

$$I = 3.32B\log_{10}\left(1+\frac{S}{N}\right)$$

Where:

I = information capacity (bps)
B = bandwidth (Hz)
S/N = signal-to-noise power ratio (unitless)

- For a standard telephone circuit with S/N of 3000 (30 dB) and BW of 2.7 kHz, Shannon limit for information capacity is:

$$I = 3.32(2700)\log_{10}\left(1+1000\right) = 26.9kbps$$

Baud Rate

- Baud Rate and Bit Rate:
  - Baud is the term that is often misunderstood and confused with bit rate (bps). Bit rate refers to the rate of change of a digital information signal, usually binary.
  - Baud refers to the rate of change of a signal on the transmission medium after encoding and the modulation have occurred (transmission rate, modulation rate or symbol rate). Baud is expressed as:

$$baud = \frac{1}{t_S}$$

Where:

Baud = symbol rate (baud per second)
t_S = time of signalling element (seconds)

Minimum Bandwidth

- Minimum theoretical BW necessary to propagate a signal is called *Nyquist BW* or *Nyquist frequency*. Thus, bit rate related to the BW as:

$$f_b = 2B$$

Where:

f_b is the bit rate in bps and B is the ideal Nyquist BW.

- However, if more than two levels are used for signaling, more than one bit may be transmitted at a time, it is possible to propagate a bit rate that exceeds 2B. The Nyquist channel capacity is:

$$f_b = 2B\log_2 M$$

Where:

f_b = channel capacity (bps)
B = minimum Nyquist BW (hertz)
M = number of discrete signal or voltage

- M-ary Encoding
  - M-ary derived from word binary and M represents a digit that corresponds to the number of conditions, levels or combinations possible for a given number of binary variables.
  - For example, digital signal with four possible conditions (voltage levels, frequencies, phases and so on) is M-ary system where M = 4. If there are eight possible conditions, M = 8.
  - The number of bits necessary to produce a given number of conditions can be expressed as:

$$N = \log_2 M$$

Where:

N = number of bits necessary

M = number of conditions, levels

- The minimum BW to pass the M-ary digitally modulated carrier is:

$$B = \frac{f_b}{\log_2 M}$$

- If N (the number of bits encoded) is substituted for $\log_2 M$, we have:

$$B = \frac{f_b}{N}$$

- Since baud is the encoded rate of change, it also equals the bit rate divided by the number of bits encoded into one signaling element. Thus:

$$Baud = \frac{f_b}{N}$$

- The baud and the ideal Nyquist BW have the same value and true for all forms of digital modulation except FSK.

Data and Signals

- Analog **data**
  - Takes on continuous values. Ex. Voice or video
- Digital **data**
  - Takes on discrete values. Ex. Text and integers

- Analog **Signal**
  - Continuously varying electromagnetic wave representing data carried over a variety of medium
- Digital **Signal**
  - Sequence of voltage pulses representing data transmitted over a wire medium

Data and Signals

Table 2-1

Five combinations of data and signals

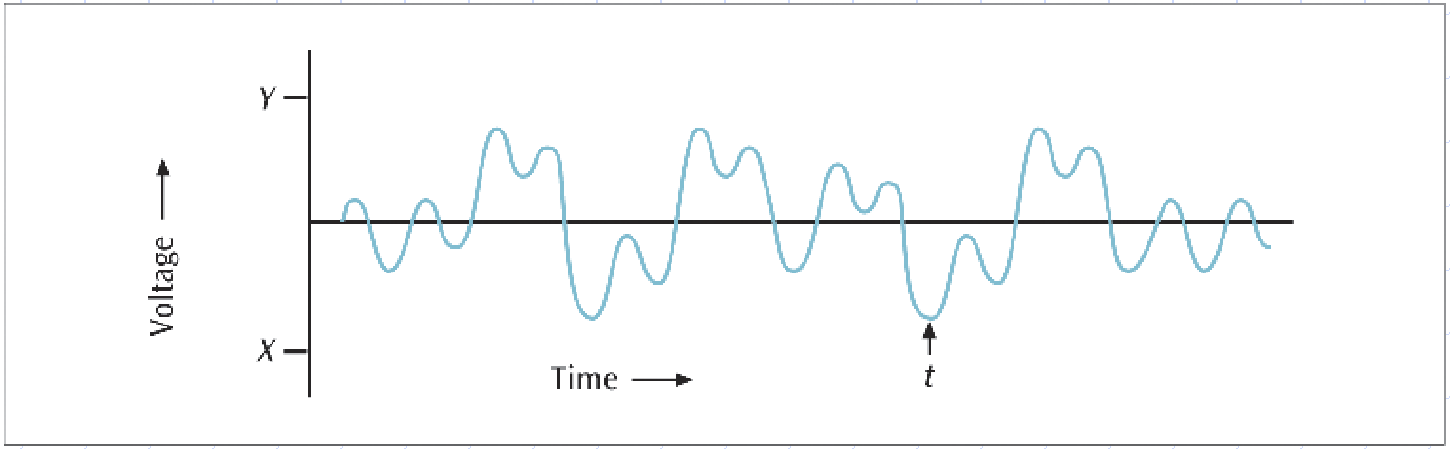
Data	Signal	Common Conversion Technique	Common Devices	Common Systems
Analog	Analog	Amplitude modulation Frequency modulation	Radio tuner TV tuner	Telephone Cable TV Broadcast TV AM and FM Radio
Digital	Digital	NRZ-L NRZI Manchester Differential Manchester Bipolar-AMI 4B/5B	Digital encoder	Local area networks Telephone systems HDTV
Digital	Analog	Amplitude shift keying Frequency shift keying Phase shift keying	Modem	Dial-up Internet access DSL Cable modems
Analog	Digital	Pulse code modulation Delta modulation	Codec	Telephone systems Music systems
Analog or Digital	Analog	Spread spectrum technology	Spread spectrum encoder	Cordless telephones Wireless LANs

Analog versus Digital

Analog is a continuous waveform

Examples: (naturally occurring) music and voice.

Figure 2-1
A simple example of an analog waveform



Analog versus Digital

It is harder to separate noise from an analog signal than it is to separate noise from a digital signal.

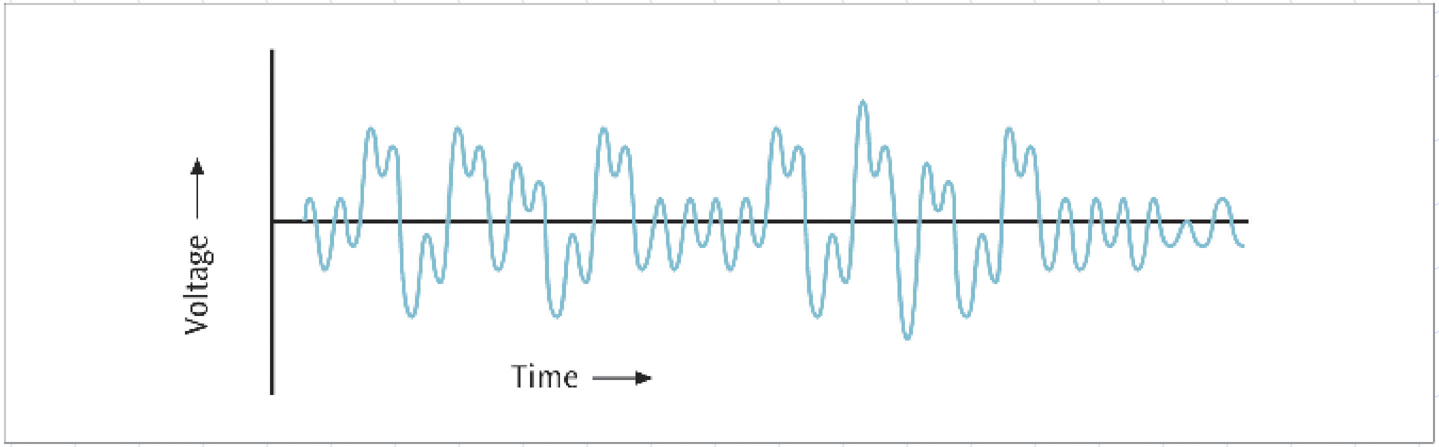
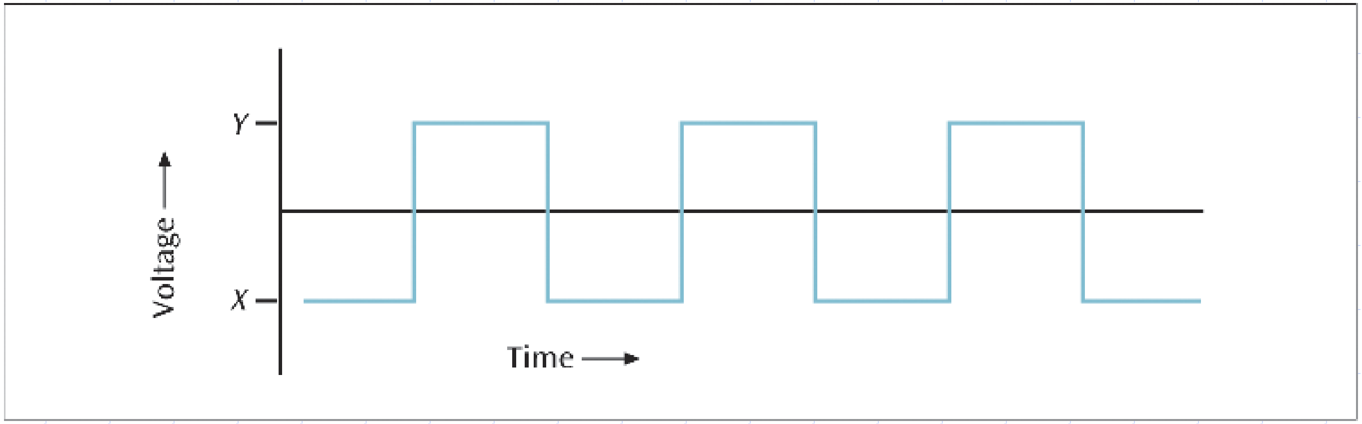


Figure 2-2
The waveform of a symphonic overture with noise

Analog versus Digital

Digital is a discrete or non-continuous waveform with examples such as computer 1s and 0s.

Figure 2-3
A simple example of a digital waveform



Noise in a digital signal

You can still discern a high voltage from a low voltage.

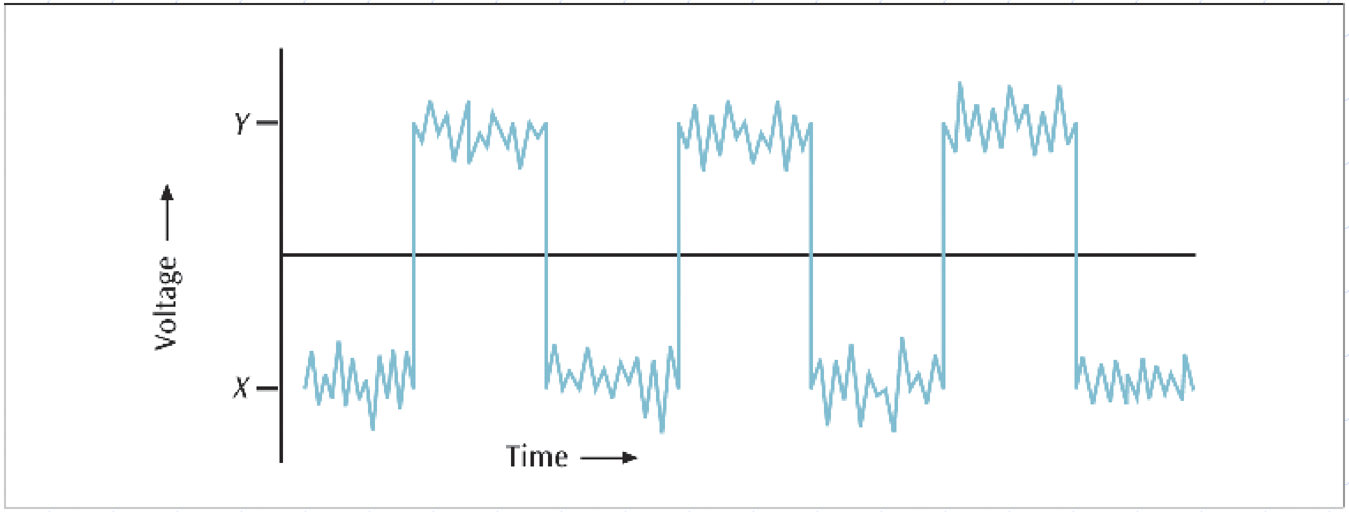
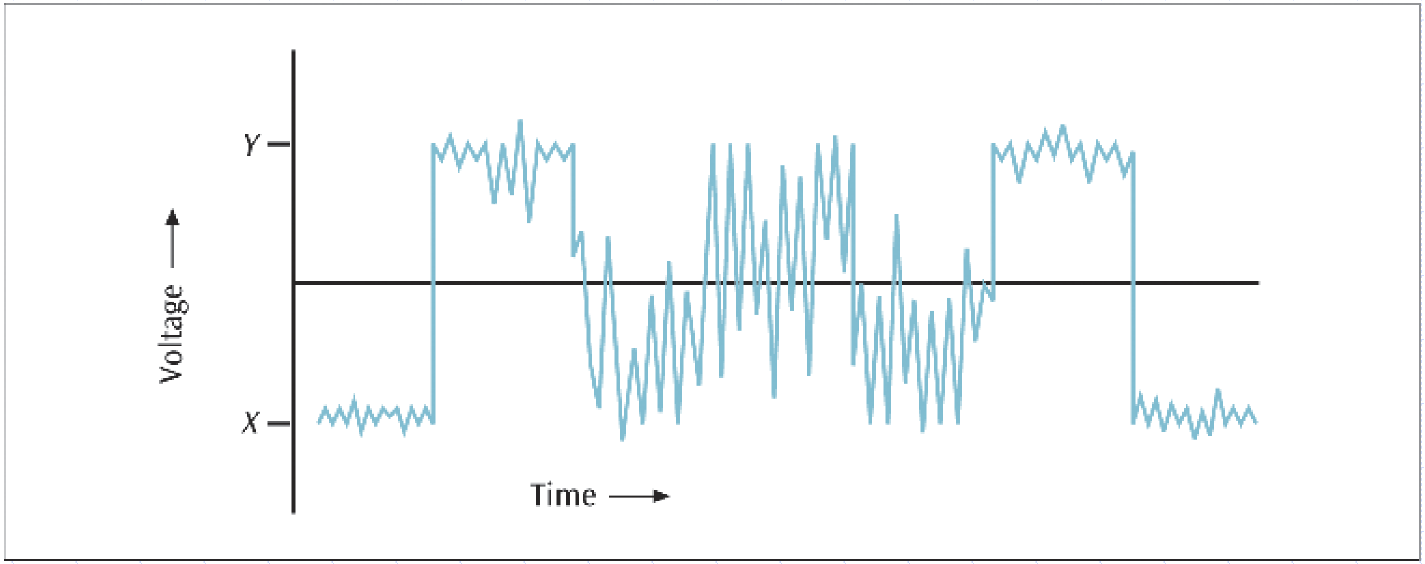


Figure 2-4
A digital signal with some noise introduced

Noise in a digital signal

Too much noise - you cannot discern a high voltage from a low voltage.

Figure 2-5
A digital waveform with noise so great that you can no longer recognize the original waveform



Fundamentals of Signals

All Signals Have Three Components

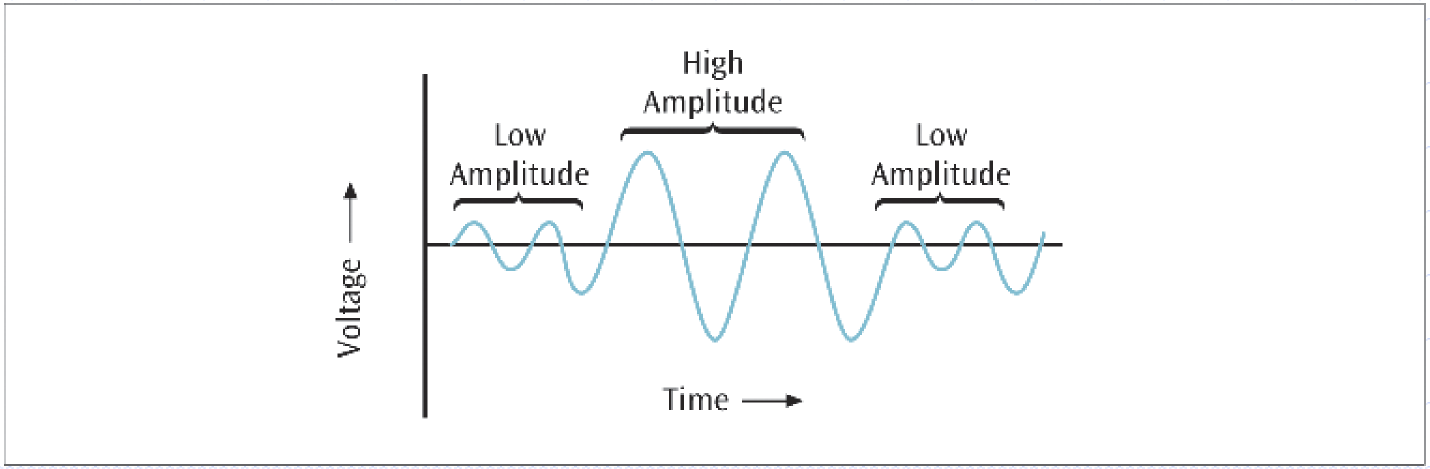
- Amplitude
- Frequency
- Phase.

Fundamentals of Signals

Amplitude

height of the wave above or below a given reference point.

Figure 2-6
A signal with two different amplitudes



Fundamentals of Signals

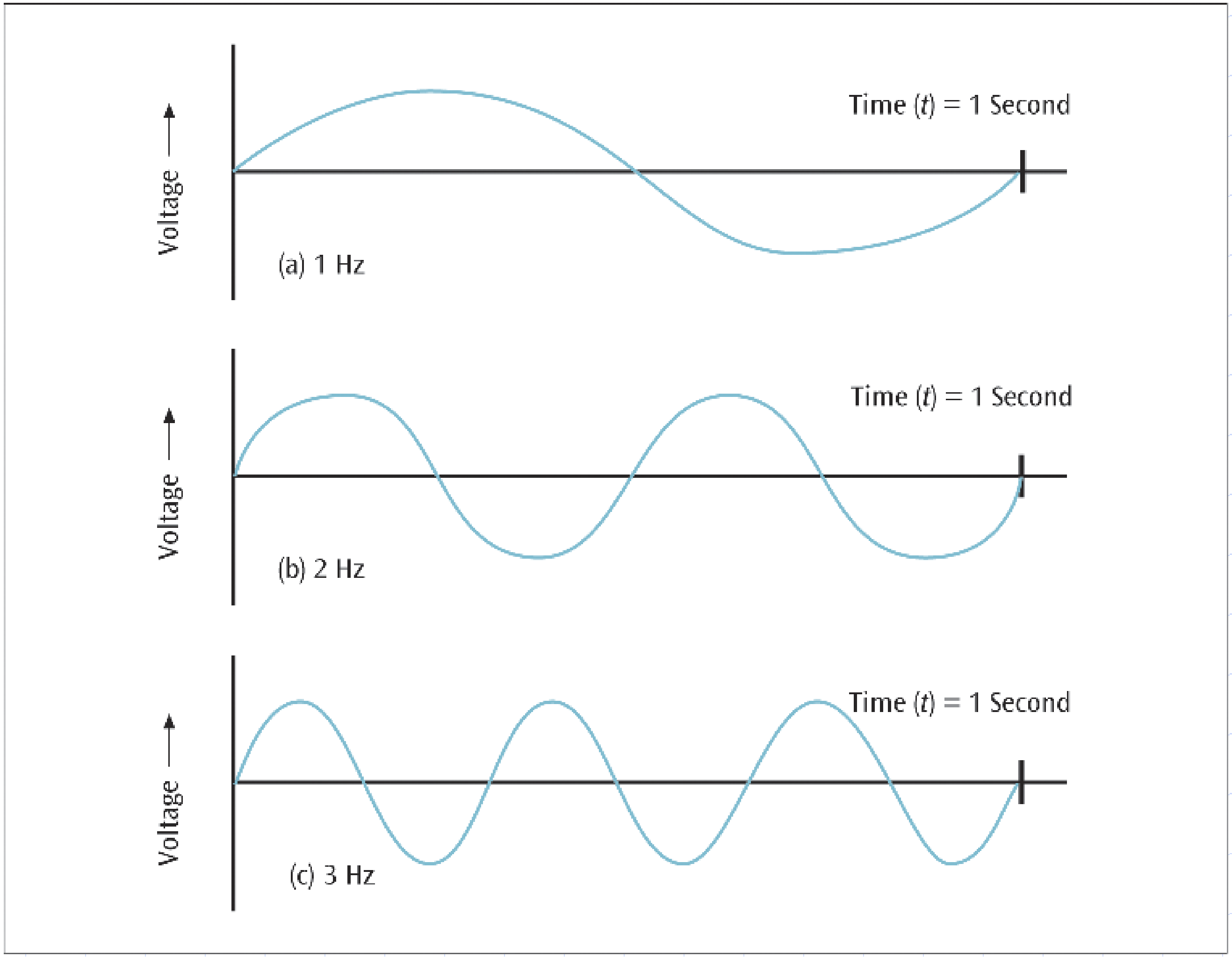
Frequency
Number of times a signal makes a complete cycle within a given time frame

Spectrum
Range of frequencies that a signal spans from minimum to maximum

Bandwidth
Absolute value of the difference between the lowest and highest frequencies of a signal.

Fundamentals of Signals

Figure 2-7
Three signals of 1 Hz, 2 Hz, and 3 Hz



Fundamentals of Signals

Frequency

Example - consider an average voice:

The average voice has a frequency range of roughly 300 Hz to 3100 Hz.

The spectrum would be 300 - 3100 Hz

The bandwidth would be 2800 Hz.

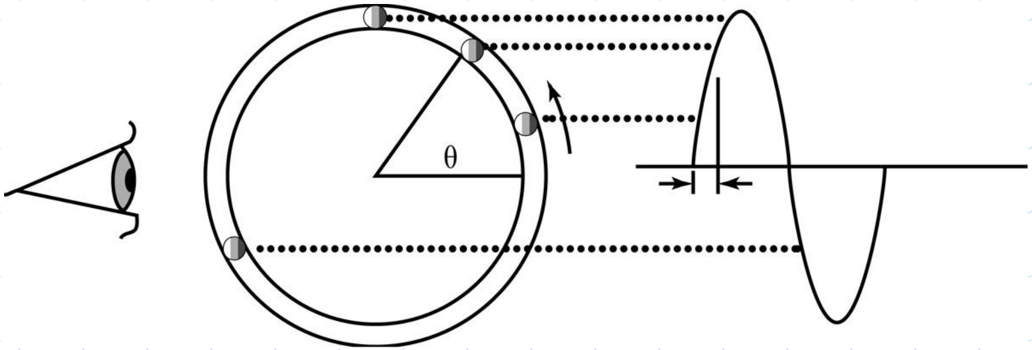
Fundamentals of Signals

Phase

The phase of a signal is the position of the waveform relative to a given moment of time or relative to time zero.

A change in phase can be any number of angles between 0 and 360 degrees.

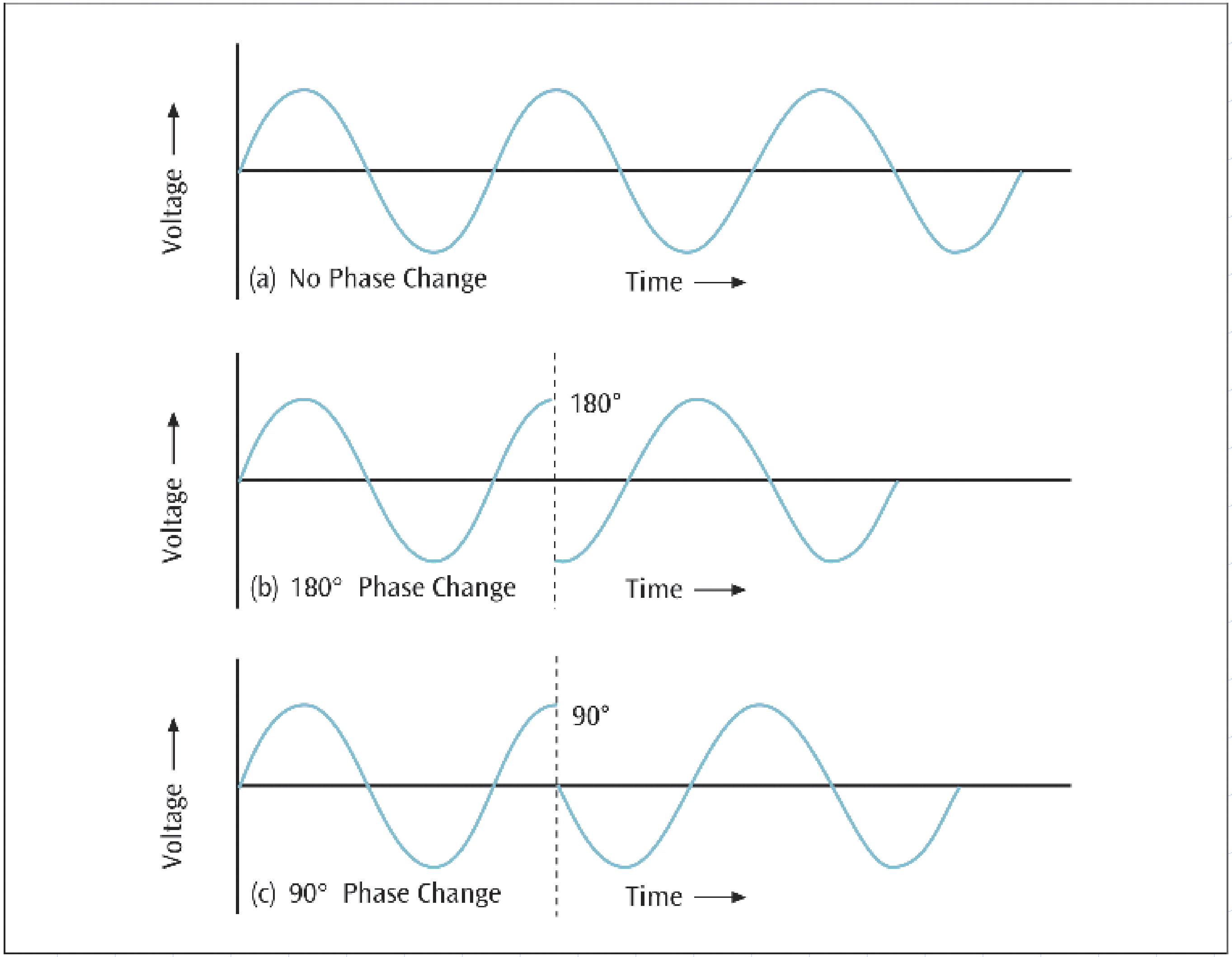
Phase changes often occur on common angles, such as 45, 90, 135, etc.



Fundamentals of Signals

Figure 2-8

A sine wave showing (a) no phase change, (b) a 180-degree phase change, and (c) a 90-degree phase change



Converting Data into Signals

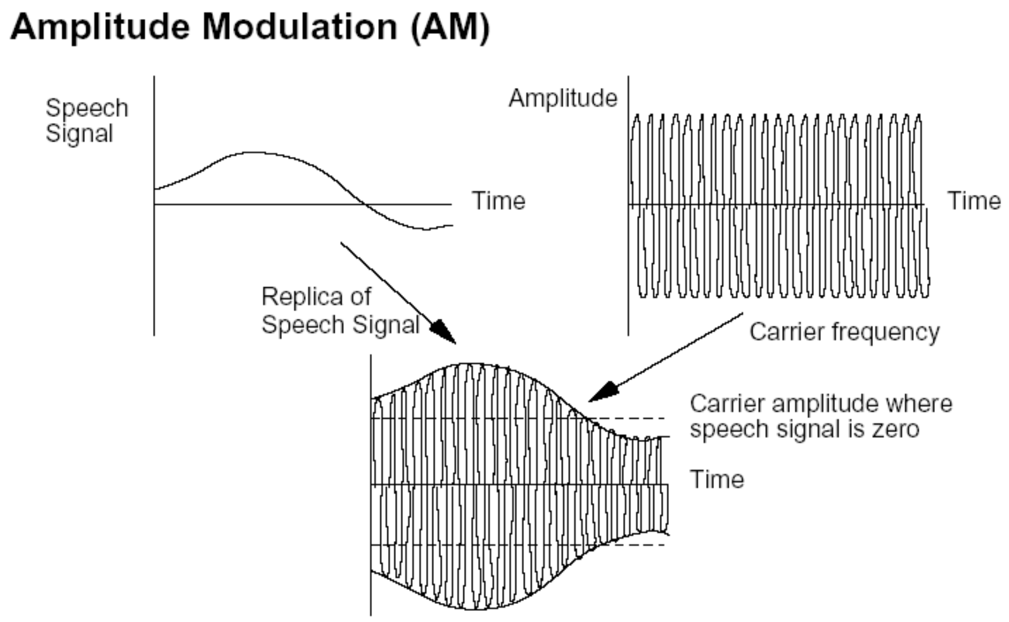
Reason For Conversion:

- Digital data → Digital Signal
  - Easy and simple to implement
- Analog data → Digital Signal
  - Allows the use of digital transmission and switching equipment
- Digital data → Analog Signal
  - Allows use of the public telephone system
  - Allows use of optical fiber
- Analog Data → Analog Signal
  - Easy
  - Telephone system was primarily analog
  - Conversion from low frequency to higher frequency

Converting Data into Signals

- An exact terminology is ***Modulation***.
- Two type of modulation: Analogue and Digital Modulation

Example of analogue modulation



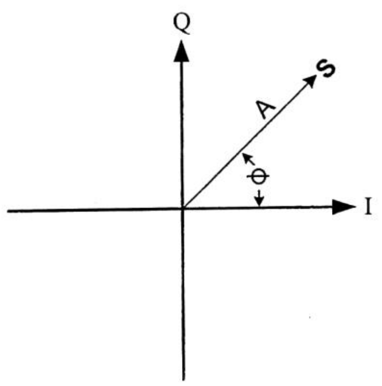
Digital Modulation Techniques

- Assume the following signal is digital:

$$V(t) = V\sin(2\pi f t + \theta)$$

ASK FSK PSK

QAM



- Digital modulation method in general can easily be viewed with a phasor diagram. **I** is the in-phase (**0** degree) reference plane, while **Q** is the quadrature (**90** degree) reference plane.
- The signal **S** can vary in phase (ϕ) and amplitude (**A**).

Amplitude Shift Keying (ASK)

- The simplest digital modulation technique where a binary information signal directly modulates the amplitude of an analogue carrier, sometime called Digital Amplitude Modulation.
- Mathematically, ASK is:

$$V_{(ask)}(t) = [1 + V_m(t)]\left[\frac{A}{2}\cos(\omega_c t)\right]$$

Where:

$V_{(ask)}$ = amplitude shift keying wave
$V_m(t)$ = digital information signal (Volts)
$A/2$ = unmodulated carrier amplitude (Volts)
ω_c = analogue carrier radian frequency

Amplitude Shift Keying (ASK)

- The modulating signal is $V_m(t)$ is a normalised binary waveform, where +1V = logic 1 and –1V is logic 0. Therefore, for a logic 1 input, $V_m(t)$ = +1V and we have:

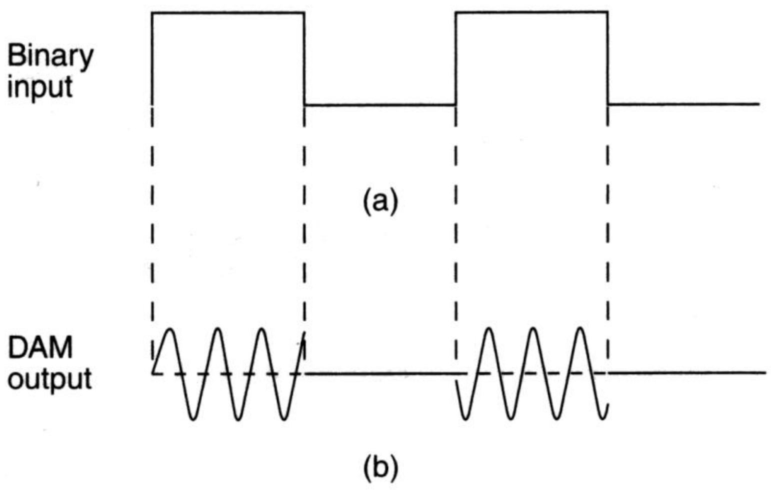
$$V_{(ask)}(t) = [1+1]\left[\frac{A}{2}\cos(\omega_c t)\right] = A\cos(\omega_c t)$$

- And for logic 0, $V_m(t)$=-1V, we have:

$$V_{(ask)}(t) = [1+(-1)]\left[\frac{A}{2}\cos(\omega_c t)\right] = 0$$

Amplitude Shift Keying (ASK)

- Thus, the modulated ASK is either carrier frequency or 0. Hence, the carrier is either *on* or *off*.
- That is why the ASK is sometime referred as ON OFF Keying or OOK.



Amplitude Shift Keying (ASK)

- ASK waveform is the same as the rate of change of the binary input (bps); thus the bit rate equals the baud. With ASK the bit rate is also equal to the minimum Nyquist bandwidth. Setting N = 1: The minimum Nyquist bandwidth is;

$$B = \frac{f_b}{N} = f_b$$

$$Baud = \frac{f_b}{N} = f_b$$

- The use of the ASK to transport the digital information is a relatively low quality except for very low speed telemetry circuit.

Amplitude Shift Keying (ASK)

Example:

- Determine the baud and the minimum bandwidth necessary to pass a 10 kbps binary signal using ASK:

Solution:

- For ASK, N=1:

$$B = \frac{10000}{1} = 10000$$

$$Baud = \frac{10000}{1} = 10000$$

Frequency Shift Keying (FSK)

- Frequency Shift Keying or FSK is another relatively simple, low performance type of digital modulation.
- FSK is a form of constant amplitude angle modulation similar to the standard frequency modulation (FM) except the modulating signal is binary signal that varies between two discrete voltage levels rather than a continuously changing analogue waveform.
- Consequently, FSK sometime called *binary* FSK (BFSK). The general expression for FSK is:

$$V_{(fsk)}(t) = V_c \cos\left[2\pi(f_c + V_m(t)\Delta f)t\right]$$

Where:

$V_{(fsk)}(t)$ = binary FSK waveform
V_c = peak analogue carrier amplitude (volts)
f_c = analogue carrier centre frequency (hertz)
Δf = peak change in the analogue carrier frequency (hertz)
$V_m(t)$ = binary input signal (volts)

Frequency Shift Keying (FSK)

- From the FSK equation, it can be seen that the peak shift in the carrier frequency (Δf) is proportional to the amplitude of the binary input signal ($V_m(t)$) and the direction of the shift is determine by the polarity.
- The modulating signal is a normalised binary waveform where again a logic 1 is +1V and logic 0 is –1V.
- For logic 1 input:

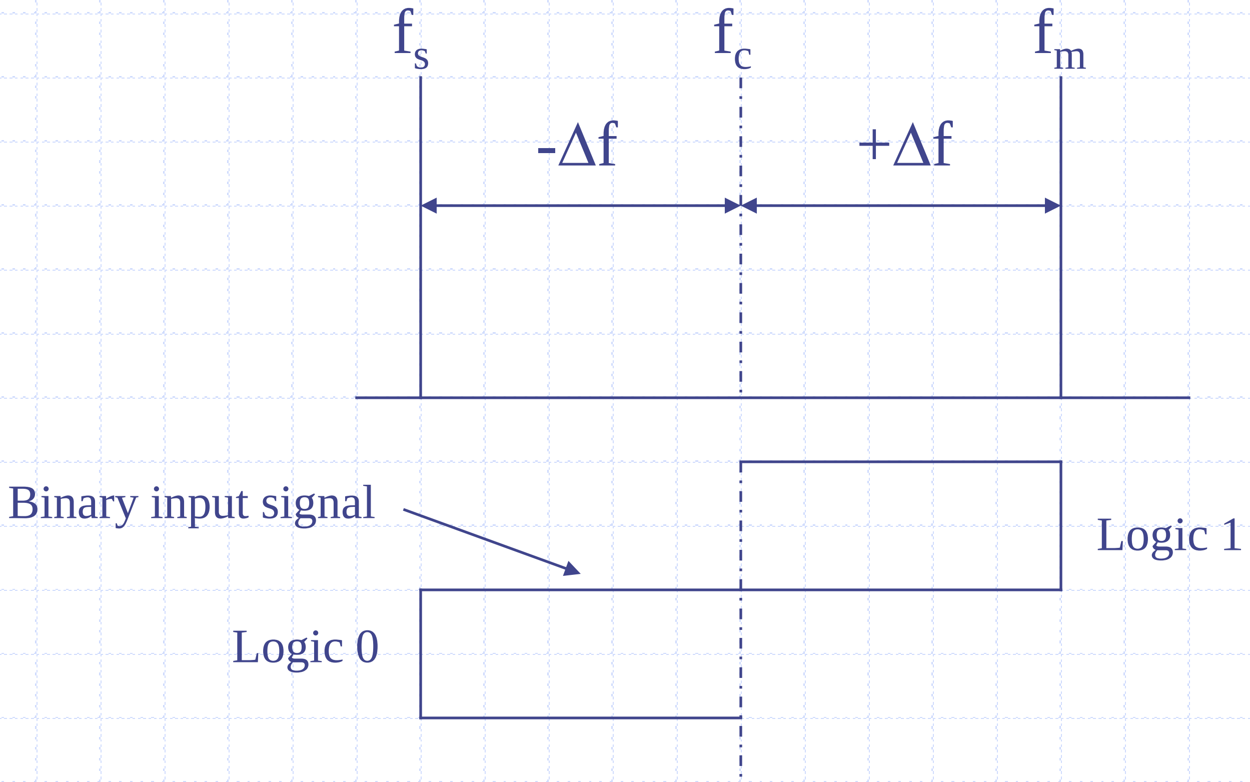
$$V_{(fsk)}(t) = V_c \cos\left[2\pi(f_c + \Delta f)t\right]$$

- For logic 0 input:

$$V_{(fsk)}(t) = V_c \cos\left[2\pi(f_c - \Delta f)t\right]$$

Frequency Shift Keying (FSK)

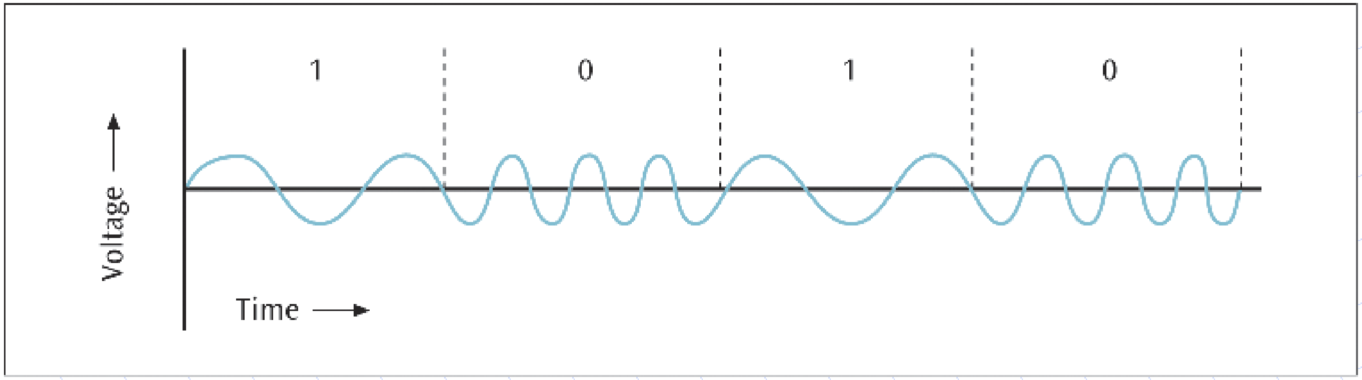
- With binary FSK, the carrier centre frequency is shifted (deviated) up and down in the frequency domain by the binary input signal.



Frequency Shift Keying (FSK)

One frequency encodes a 0 while another frequency encodes a 1

Figure 2-17
Simple example of frequency shift keying



Frequency Shift Keying (FSK)

FSK Bit Rate, Baud and Bandwidth

- The baud rate of the binary FSK can also be determined by substituting N = 1, therefore:

$$Baud = \frac{f_b}{N} = f_b$$

- The minimum bandwidth for FSK is given as:

$$B = \left|\left(f_s - f_b\right) - \left(f_m - f_b\right)\right| = \left|f_s - f_m\right| + 2f_b$$

- Since the $|f_s - f_m|$ equals $2\Delta f$, the minimum bandwidth can be approximated as:

$$B = 2\left(\Delta f + f_b\right)$$

Frequency Shift Keying (FSK)

Example:

- Determine: (a) the peak frequency (b) Minimum bandwidth (c) Baud rate for a binary FSK signal with a mark frequency of 49 kHz, a space frequency of 51 kHz and input bit rate at 2 kbps.

Solution:

(a) The peak frequency deviation is;

$$\Delta f = \frac{|49kHz - 51kHz|}{2} = 1kHz$$

(b) The minimum bandwidth is;

$$B = 2(1000 + 2000) = 6kHz$$

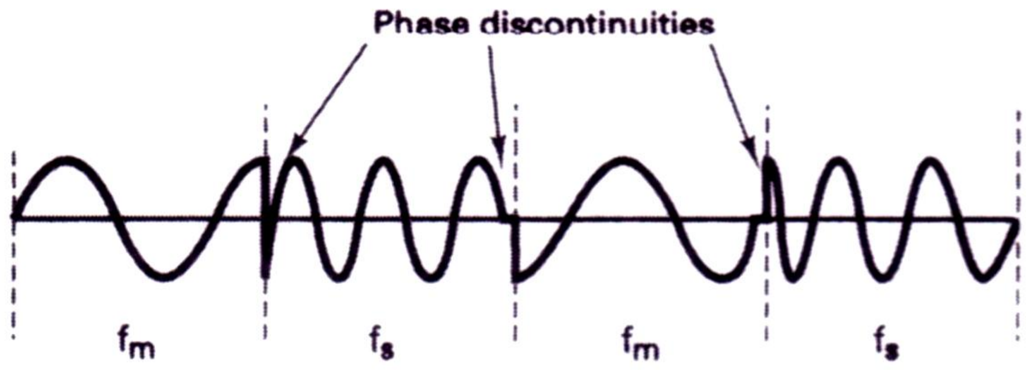
(c) For FSK, N = 1, and the baud rate is;

$$Baud = \frac{2000}{1} = 2000$$

Continuous FSK

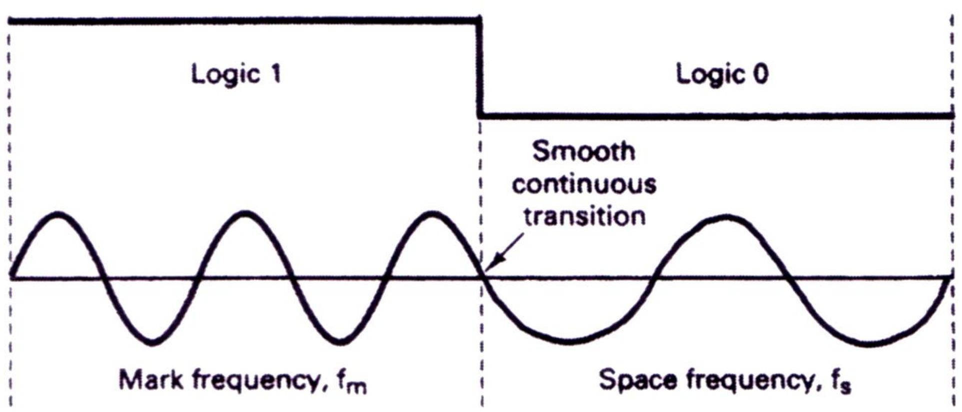
- Continuous Phase Frequency Shift Keying (CP-FSK) is also binary FSK except the mark and space frequencies are synchronise with the input binary rate.
- The mark and space frequencies are selected such that they are separated from the centre frequency by an exact multiple of one-half the bit rate (f_m and $f_s=n(f_b/2)$), n is an integer.
- CP-FSK has a better bit-error performance than conventional binary FSK for a given signal-to-noise ratio.
- However, CP-FSK requires synchronization circuits and therefore more expensive to implement.

Continuous FSK



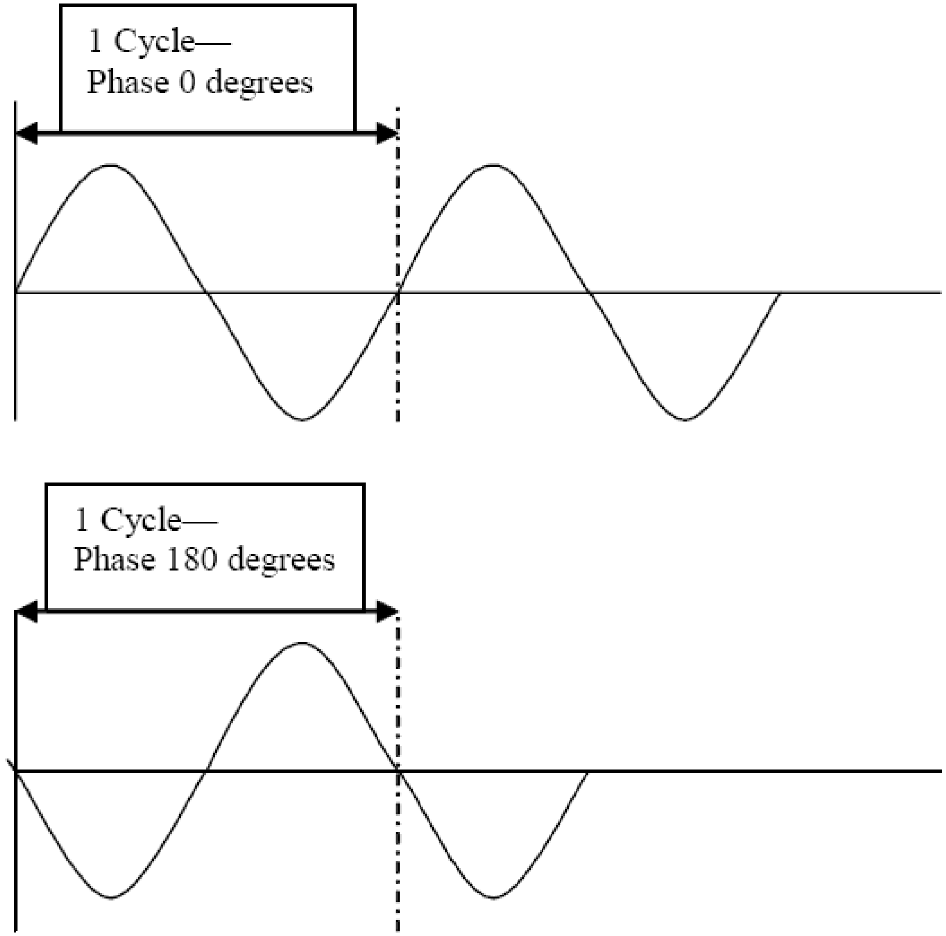
FSK

CP-FSK



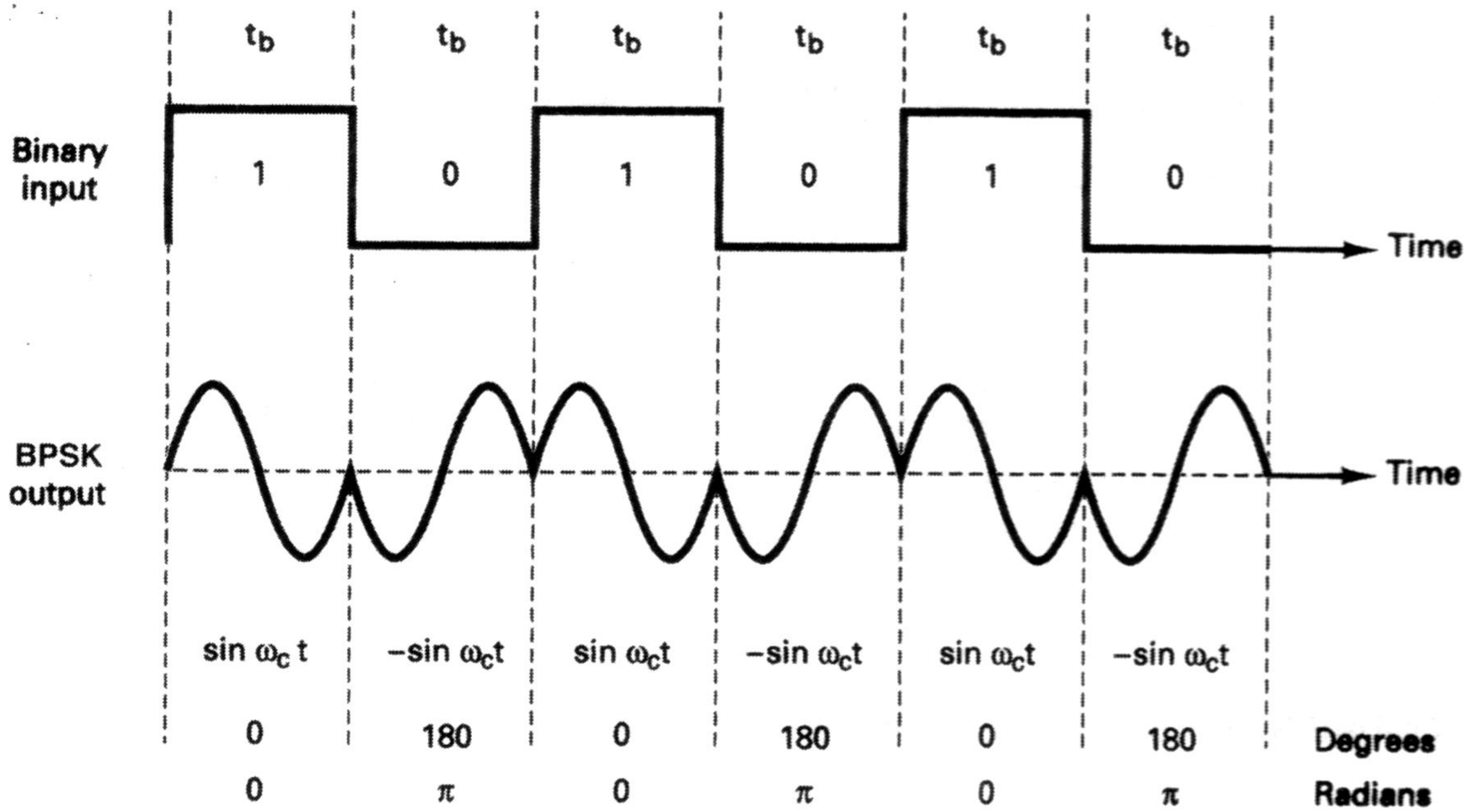
Phase Shift Keying

- Phase shift keying (PSK) is another form of angle-modulated constant amplitude digital modulation.
- In this modulation method a sinewave is transmitted and the phase of the sinewave carries the digital data.
- For a "0", a 0 degrees phase sine wave is transmitted. For a "1", a 180 degrees sine wave is transmitted as shown in the next figure.
- In order to correctly detect the phase of each symbol, this technique requires phase synchronization between the receiver's and transmitter's phase.
- This complicates the receiver's design.



Phase change of the sinewave carrier for BPSK

BPSK signal



Output phase-versus-time relationship for a BPSK modulator

Bandwidth of BPSK

- Mathematically, the output of a BPSK modulator is proportional to

$$BPSK_{output} = [\sin(2\pi f_a t] \times [\sin(2\pi f_c t]$$

Where:
f_a = maximum fundamental frequency of binary input (hertz)
f_c = reference carrier frequency (hertz)

- Solving for the trig identity for the product of two sine functions,

$$\frac{1}{2}\cos[2\pi(f_c - f_a)t] - \frac{1}{2}\cos[2\pi(f_c + f_a)t]$$

Bandwidth of BPSK

- Thus the minimum double-sided Nyquist bandwidth (B) is:

$$f_c + f_a - (f_c - f_a) \text{ or } f_c + f_a - f_c + f_a = 2f_a$$

- Because:

$$f_a = \frac{f_b}{2}$$

f_b = input bit rate

- Therefore, the minimum double-sided Nyquist bandwidth is:

$$B = \frac{2f_b}{2} = f_b$$

Bandwidth of BPSK

Example:

- For a BPSK modulator with a carrier frequency of 70 MHz and an input bit rate of 10 Mbps, determine the maximum and minimum upper and lower side frequencies, draw the output spectrum, determine the Nyquist bandwidth and calculate the baud.

Solution

- An output of BPSK is:

$$BPSK_{output} = [\sin(2\pi f_a t] \times [\sin(2\pi f_c t]$$

$$= [\sin 2\pi (5MHz)t] \times [\sin 2\pi (70MHz)t]$$

$$= \frac{1}{2}\cos 2\pi (70MHz - 5MHz)t - \frac{1}{2}\cos 2\pi (70MHz + 5MHz)t$$

Lower side freq Upper side freq

Bandwidth of BPSK

- Minimum lower side frequency:

$$LSF = 70MHz - 5MHz = 65MHz$$

- Maximum lower side frequency:

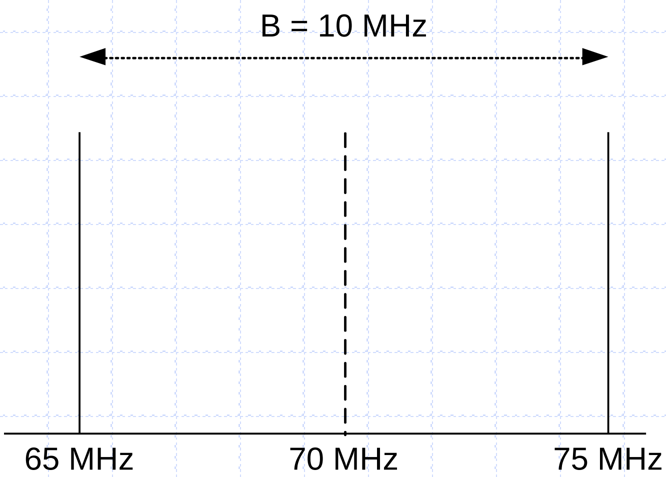
$$USF = 70MHz + 5MHz = 75MHz$$

- The bandwidth is:

$$B = 75MHz - 65MHz = 10MHz$$

- The baud = f_b = 10 megabaud

Bandwidth of BPSK

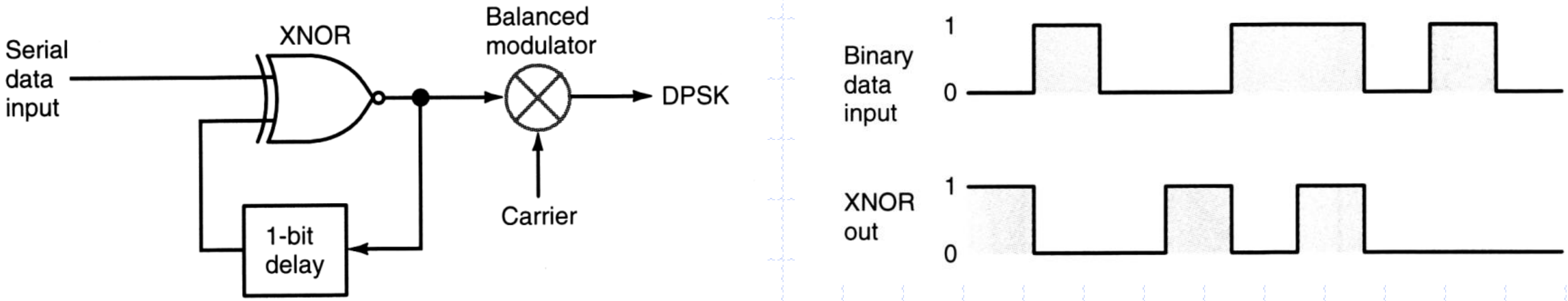


Output spectrum of the BPSK

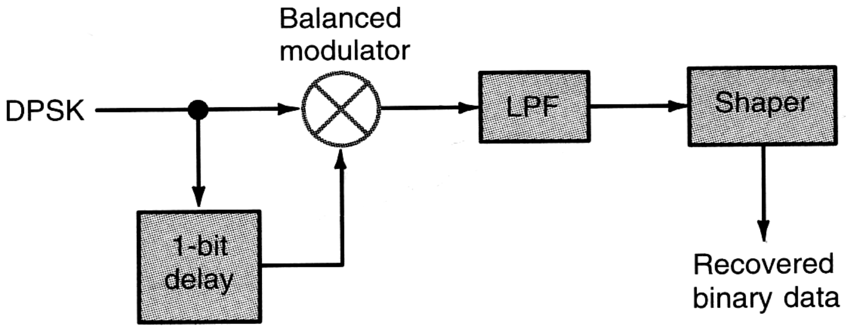
Differential PSK

- A version of binary PSK. DPSK is used to simplify the demodulation process where there is no absolute carrier phase reference.
- Instead of the synchronisation technique as in the BPSK, the transmitted signal becomes the phase reference where the phase of the received bit is compared to the phase of the previously received bit.
- The DPSK can be generated from the BPSK by a process known as differential phase coding, in which the serial bit stream passes the XNOR as shown in the next figure.
- The XNOR output is then applied to 1 bit delay before applied back to the other input of the XNOR.

DPSK



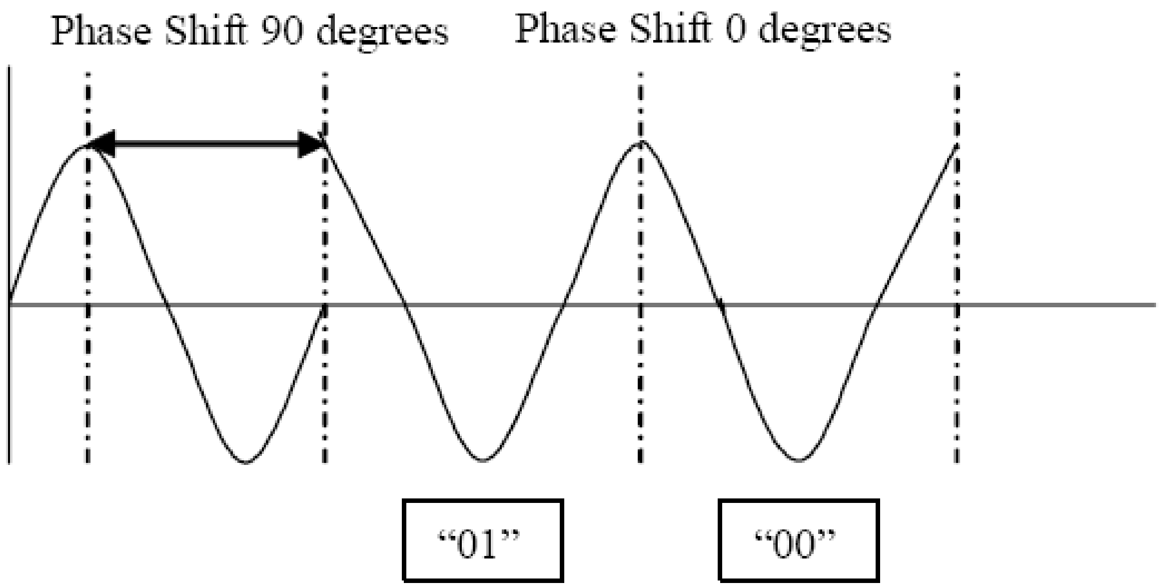
Generation of DPSK



Demodulation of DPSK

Quadrature or Quaternary PSK

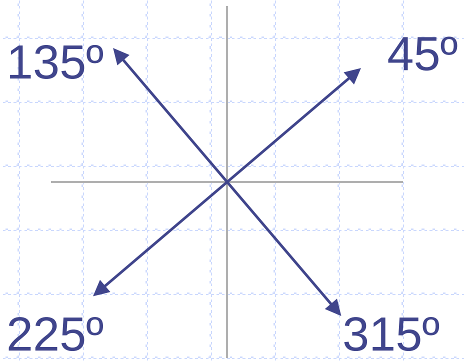
- The disadvantage of the BPSK and DPSK is that the speed of data transmission in a given bandwidth is limited.
- To increase the data rate without increasing the bandwidth is to encode more than one bit per phase change.
- QPSK adds two more phases: 90 and 270 degrees. Now two symbols per bit can be transmitted.
- Each symbol’s phase is compared relative to the previous symbol; so, if there is no phase shift (0 degrees), the bits “00” are represented.
- If there is a phase shift of 180 degrees, the bits “11” are represented.
- The system for doing this is known as quadrature PSK or 4-PSK.
- The QPSK is another form of angle-modulated, constant amplitude digital modulation. QPSK is M-ary encoding scheme where N = 2 and M = 4 (quaternary meaning 4).
- With QPSK, four output phases are possible for a single carrier frequency therefore there will be four different input conditions.
- Each pair of serial bits called a *dibit* is represented by a specific phase. A 90º phase shift exists between each pair of bits.



Phase shift of sinewave carrier for QPSK

Quadrature or Quaternary PSK

Dibit	Phase Shift
00	45
01	135
11	225
10	315

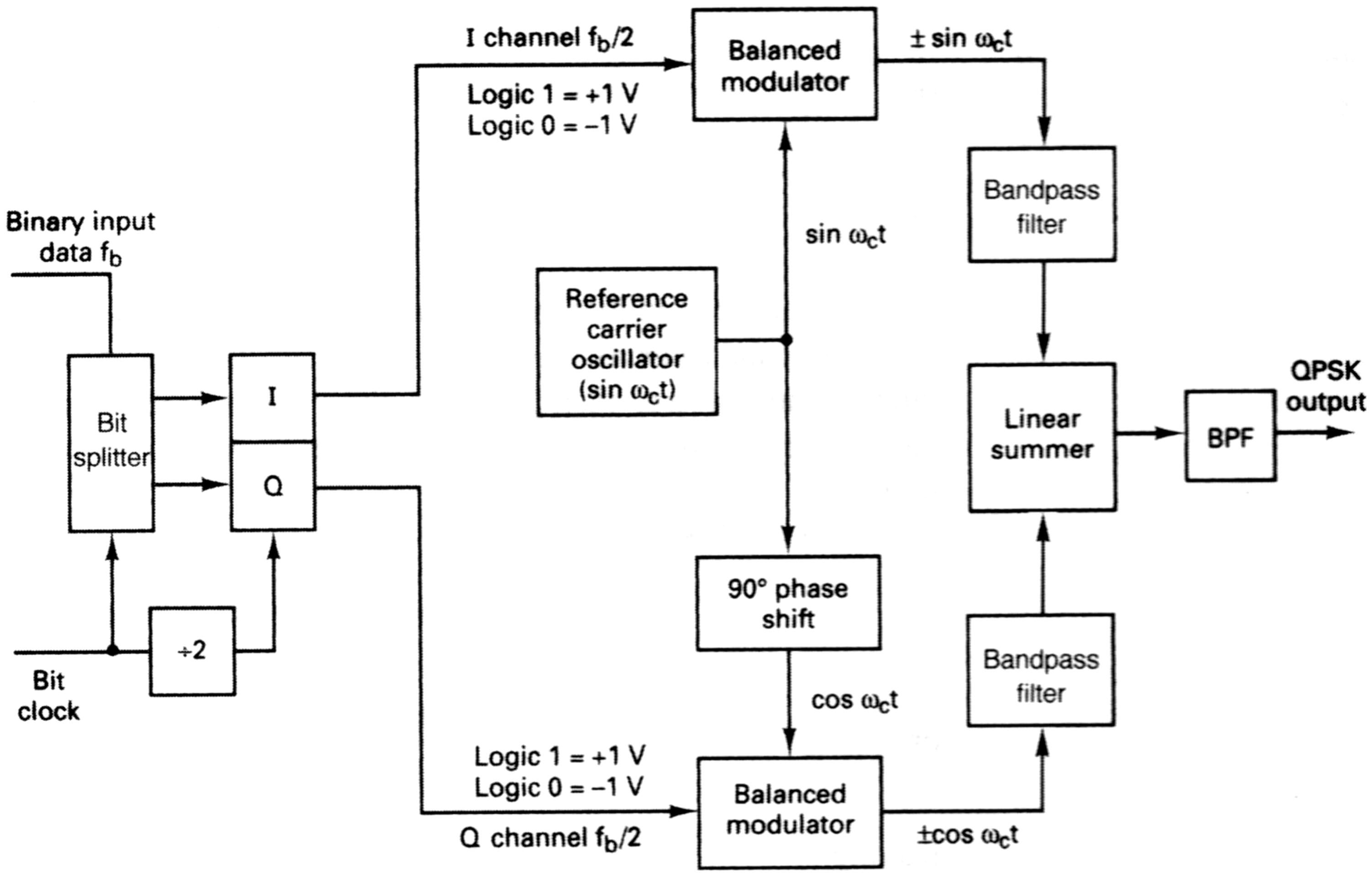


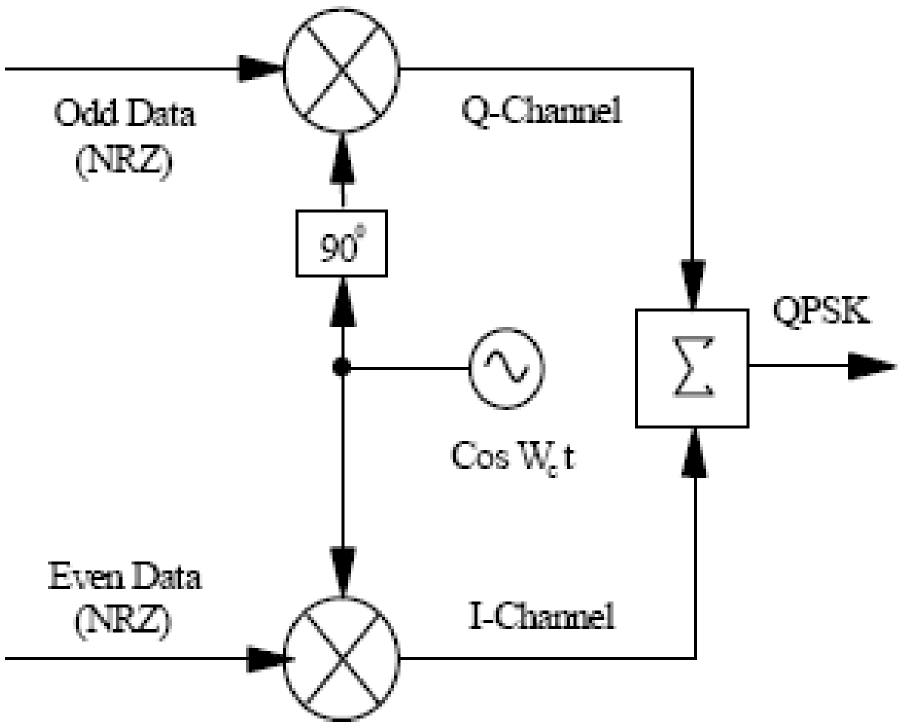
QPSK Modulator

- A block diagram of a QPSK modulator is shown in the next figure.
- A serially input binary data are split into two bits and 1 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 and the Q bit modulates a carrier that is 90° out of phase or in quadrature with the reference carrier.
- For a logic 1 = +1 V and a logic 0 = -1 V, two phases are possible at the output of the I balanced modulator ($+\sin\omega_c t$ and $-\sin\omega_c t$) and two phases are possible at the output of the Q modulator ($+\cos\omega_c t$ and $-\cos\omega_c t$).
- When the linear summer combines the two quadrature signals, there are four possible resultant phasors:

$$+\sin\omega_c t+\cos\omega_c t \qquad +\sin\omega_c t-\cos\omega_c t \qquad -\sin\omega_c t+\cos\omega_c t \qquad -\sin\omega_c t-\cos\omega_c t$$

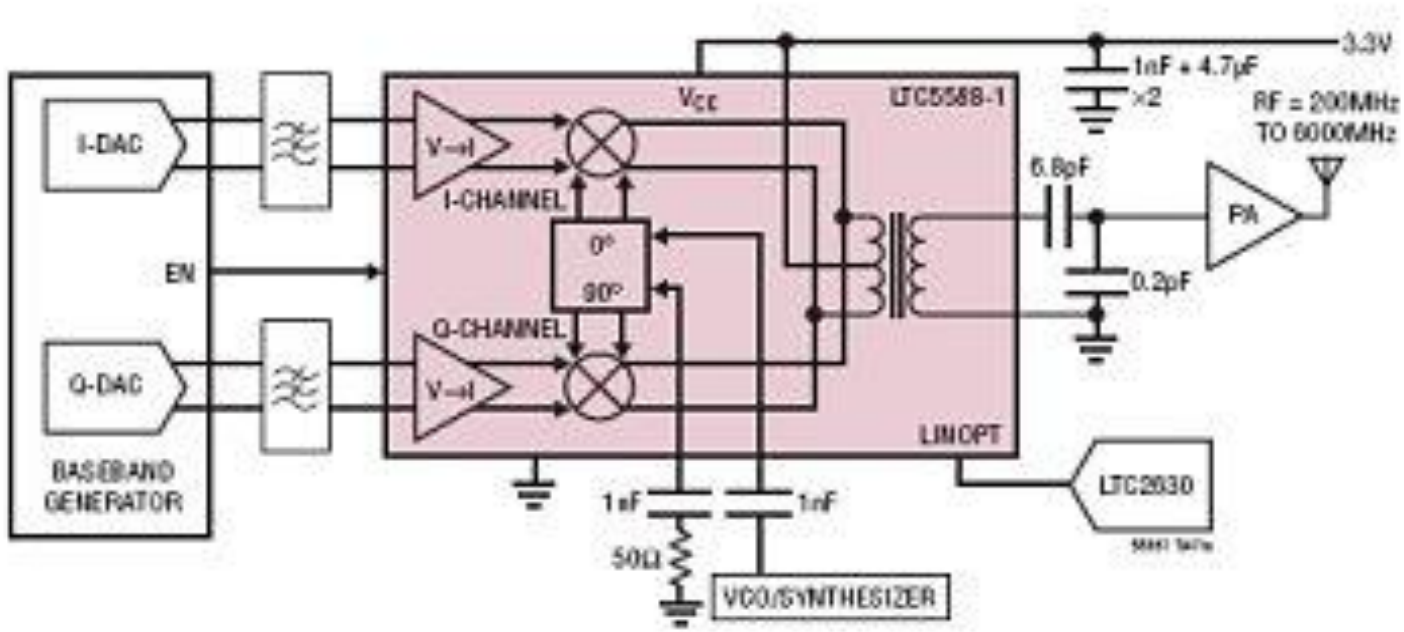
QPSK modulator





Simplified QPSK modulator

200MHz to 6000MHz Direct Conversion Transmitter Application



LTC5588-1 - 200MHz to 6000MHz Quadrature Modulator with Ultrahigh OIP3

QPSK Modulator

Example

- Construct the truth table, phasor diagram and constellation diagram for the QPSK system:

Solution

- For a binary data input of Q = 0 and I = 0, the two inputs to the I balanced modulator are –1 and sinω_ct and the two inputs to the balanced modulator are –1 and cosω_ct. Consequently, the outputs are:

$$I \text{ balanced modulator} = (\text{-}1)(\sin \omega_c t) = -1 \sin \omega_c t$$

$$Q \text{ balanced modulator} = (\text{-}1)(\cos \omega_c t) = -1 \cos \omega_c t$$

- The output of the summer is

$$\text{-}1 \cos \omega_c t \text{ -}1 \sin \omega_c t) = 1.414 \sin(\omega_c t - 135°)$$

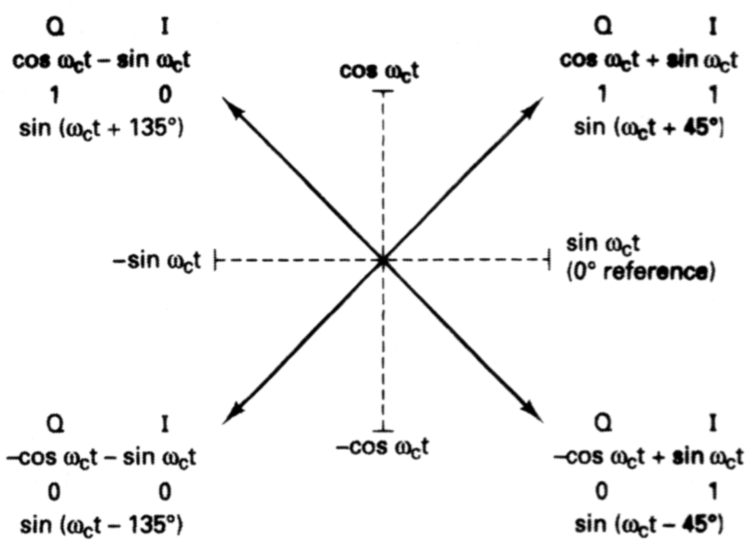
QPSK Constellation Diagram

Binary input		QPSK output phase
Q	I	
0	0	−135°
0	1	−45°
1	0	+135°
1	1	+45°

(a)

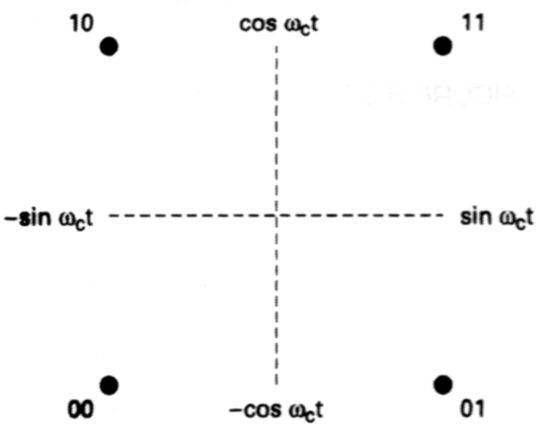
Truth Table

Phasor Diagram



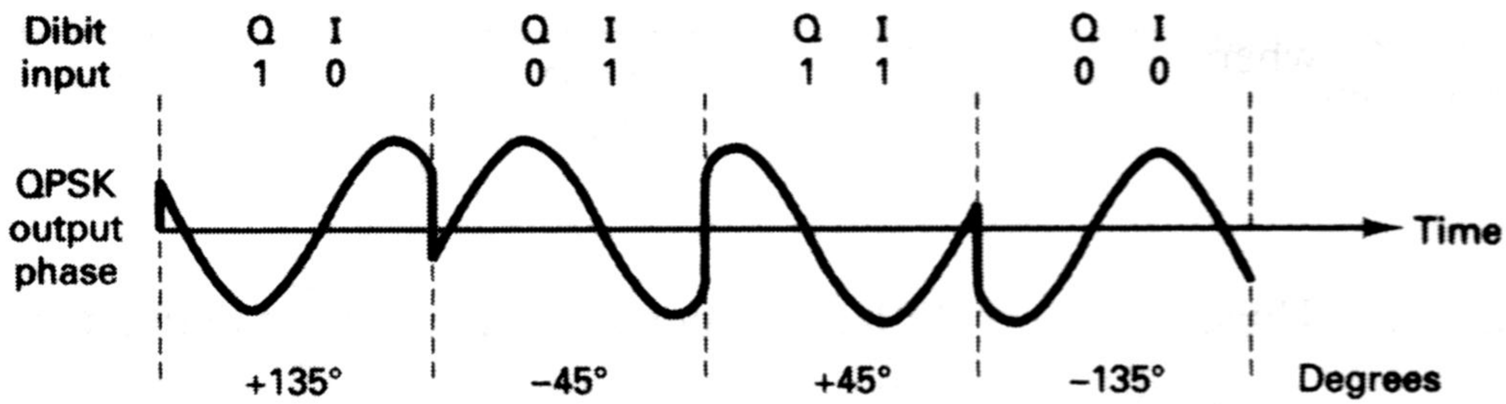
(b)

Constellation Diagram



(c)

QPSK Output phase

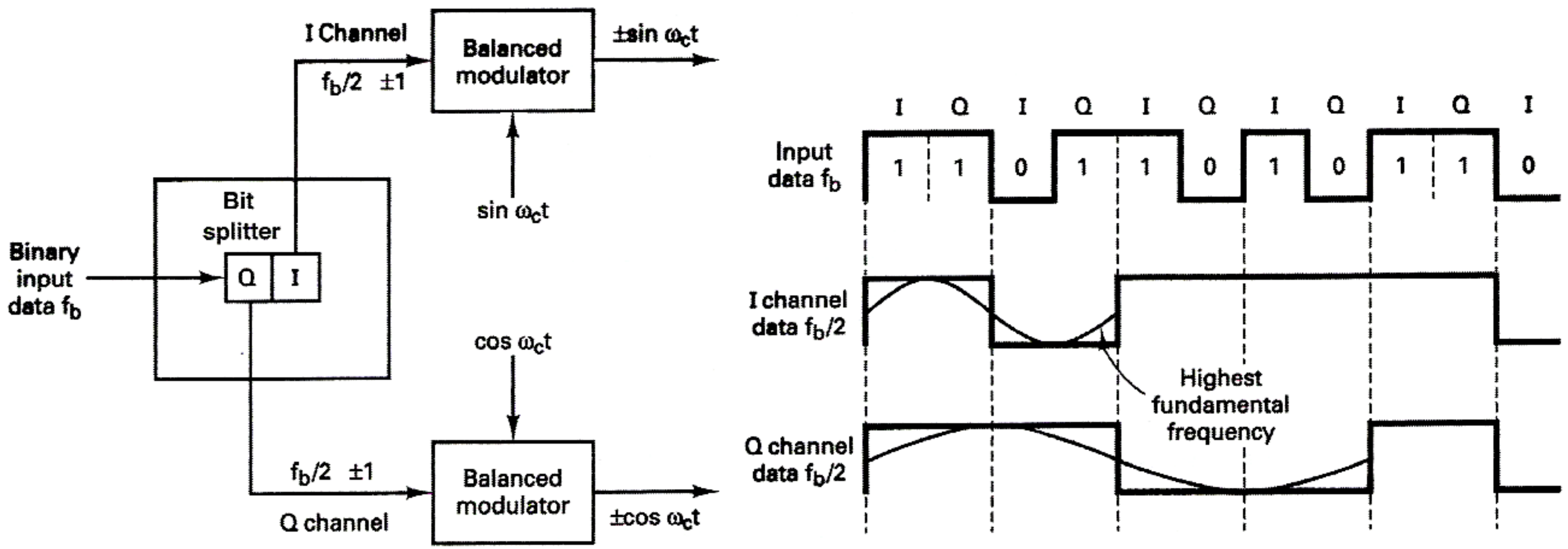


Output phase versus time of QPSK

QPSK Bandwidth

- With QPSK, because the input data are divided into two channels, the bit rate in either the I and Q channel is equal to one-half of the input data rate ($f_b/2$).
- Consequently, the highest fundamental frequency present at the input data rate one half of the $f_b/2 = f_b/4$.
- 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 (f_N = twice $f_b/4 = f_b/2$).
- Thus, with QPSK, a bandwidth compression is realised (minimum bandwidth is less than incoming bit rate).
- Because the QPSK output signal does not change phase until two bits 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.
- Therefore the minimum bandwidth and the baud are equal.

QPSK



Bandwidth consideration of BPSK

QPSK

The output of the balanced modulators can be expressed mathematically as

$$\text{output} = (\sin \omega_a t)(\sin \omega_c t)$$

where

$$\underbrace{\omega_a t = 2\pi \frac{f_b}{4} t}_{\substack{\text{modulating}\\ \text{signal}}} \quad \text{and} \quad \underbrace{\omega_c t = 2\pi f_c}_{\text{carrier}}$$

Thus,

$$\text{output} = \left(\sin 2\pi \frac{f_b}{4} t\right)(\sin 2\pi f_c t)$$

$$\frac{1}{2}\cos 2\pi\left(f_c - \frac{f_b}{4}\right)t - \frac{1}{2}\cos 2\pi\left(f_c + \frac{f_b}{4}\right)t$$

The output frequency spectrum extends from $f_c + f_b/4$ to $f_c - f_b/4$, and the minimum bandwidth (f_N) is

$$\left(f_c + \frac{f_b}{4}\right) - \left(f_c - \frac{f_b}{4}\right) = \frac{2f_b}{4} = \frac{f_b}{2}$$

Example 9-6

For a QPSK modulator with an input data rate (f_b) equal to 10 Mbps and a carrier frequency of 70 MHz, determine the minimum double-sided Nyquist bandwidth (f_N) and the baud. Also, compare the results with those achieved with the BPSK modulator in Example 9-4. Use the QPSK block diagram shown in Figure 9-17 as the modulator model.

Solution The bit rate in both the I and Q channels is equal to one-half of the transmission bit rate, or

$$f_{bQ} = f_{bI} = \frac{f_b}{2} = \frac{10\text{ Mbps}}{2} = 5\text{ Mbps}$$

The highest fundamental frequency presented to either balanced modulator is

$$f_a = \frac{f_{bQ}}{2} \text{ or } \frac{f_{bI}}{2} = \frac{5\text{ Mbps}}{2} = 2.5\text{ MHz}$$

The output wave from each balanced modulator is

$$(\sin 2\pi f_a t)(\sin 2\pi f_c t)$$

$$\frac{1}{2}\cos 2\pi(f_c - f_a)t - \frac{1}{2}\cos 2\pi(f_c + f_a)t$$

$$\frac{1}{2}\cos 2\pi[(70 - 2.5)\text{ MHz}]t - \frac{1}{2}\cos 2\pi[(70 + 2.5)\text{ MHz}]t$$

$$\frac{1}{2}\cos 2\pi(67.5\text{ MHz})t - \frac{1}{2}\cos 2\pi(72.5\text{ MHz})t$$

The minimum Nyquist bandwidth is

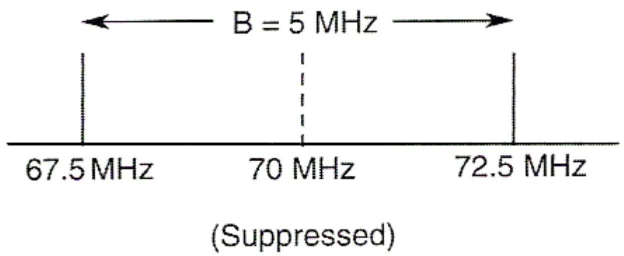
$$B = (72.5 - 67.5)\text{ MHz} = 5\text{ MHz}$$

The symbol rate equals the bandwidth; thus,

$$\text{symbol rate} = 5\text{ megabaud}$$

QPSK

The output spectrum is as follows:



$$B = 5 \text{ MHz}$$

It can be seen that for the same input bit rate the minimum bandwidth required to pass the output of the QPSK modulator is equal to one-half of that required for the BPSK modulator in Example 9-4. Also, the baud rate for the QPSK modulator is one-half that of the BPSK modulator.

The minimum bandwidth for the QPSK system described in Example 9-6 can also be determined by simply substituting into Equation 9-10:

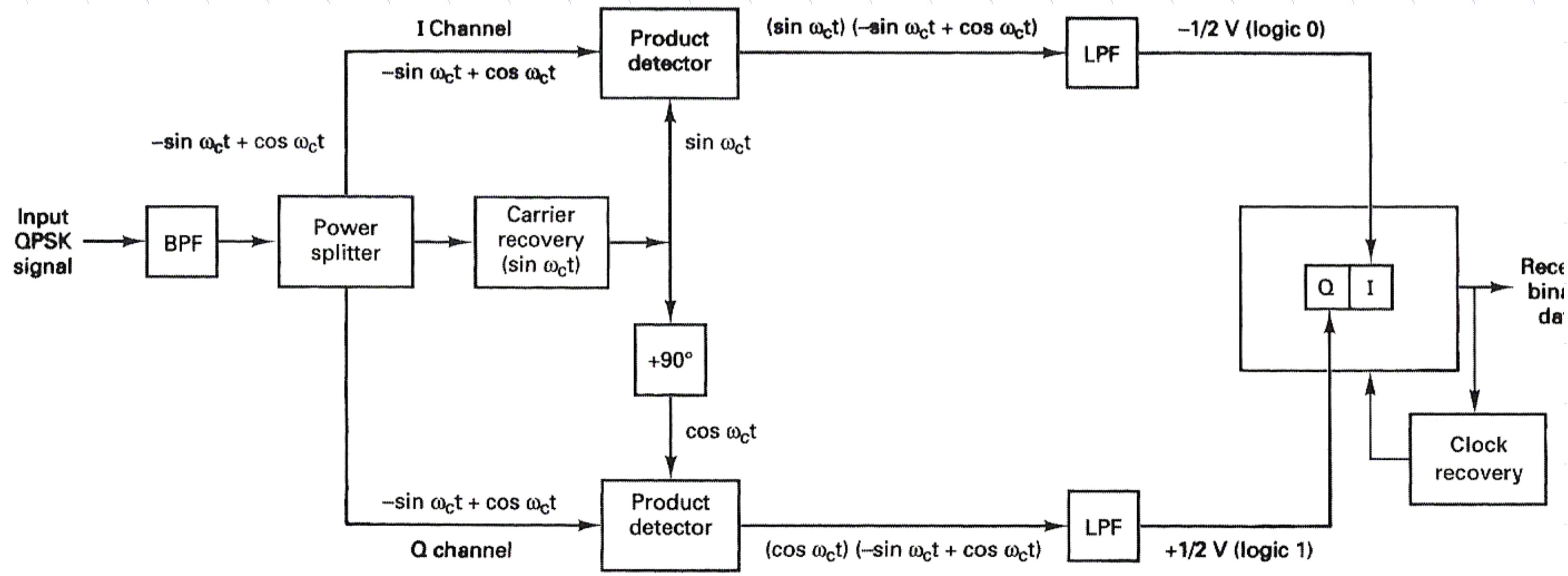
$$B = \frac{10 \text{ Mbps}}{2}$$
$$= 5 \text{ MHz}$$

QPSK

9-5-2-3 QPSK receiver. The block diagram of a QPSK receiver is shown in Figure 9-21. 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.

The incoming QPSK signal may be any one of the four possible output phases shown in Figure 9-18. To illustrate the demodulation process, let the incoming QPSK signal be $-\sin \omega_c t + \cos \omega_c t$. Mathematically, the demodulation process is as follows.

QPSK



QPSK demodulator

QPSK

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

$$I = \underbrace{(-\sin \omega_c t + \cos \omega_c t)}_{\text{QPSK input signal}} \underbrace{(\sin \omega_c t)}_{\text{carrier}} \tag{9-23}$$

$$= (-\sin \omega_c t)(\sin \omega_c t) + (\cos \omega_c t)(\sin \omega_c t)$$

$$= -\sin^2 \omega_c t + (\cos \omega_c t)(\sin \omega_c t)$$

$$= -\frac{1}{2}(1 - \cos 2\omega_c t) + \frac{1}{2}\sin(\omega_c + \omega_c)t + \frac{1}{2}\sin(\omega_c - \omega_c)t$$

$$I = -\frac{1}{2} + \overbrace{\frac{1}{2}\cos 2\omega_c t}^{\text{(filtered out)}} + \overbrace{\frac{1}{2}\sin 2\omega_c t}^{\text{(equals 0)}} + \frac{1}{2}\sin 0$$

$$= -\frac{1}{2}\text{V (logic 0)}$$

Offset QPSK (OQPSK)

- Is a modified form of QPSK where the bit waveforms on the I and Q channels are offset or shifted in phase from each other by one-half of a bit time.
- Because changes in the I channel occur at the midpoints of the Q channel bits and vice versa, there is never more than a single bit change in the dibit code and therefore, there is never more than a 90° shift in the output phase.
- In conventional QPSK, a change in the input dibit from 00 to 11 or 01 to 10 causes a corresponding 180° shift in the output phase.
- Therefore, an advantage of OQPSK is the limited phase shift must be imparted during modulation.
- A disadvantage is that changes in the output phase occur at twice the data rate in either the I and Q channels.
- Consequently, with OQPSK the baud and minimum bandwidth are twice that of conventional QPSK for a given transmission time.

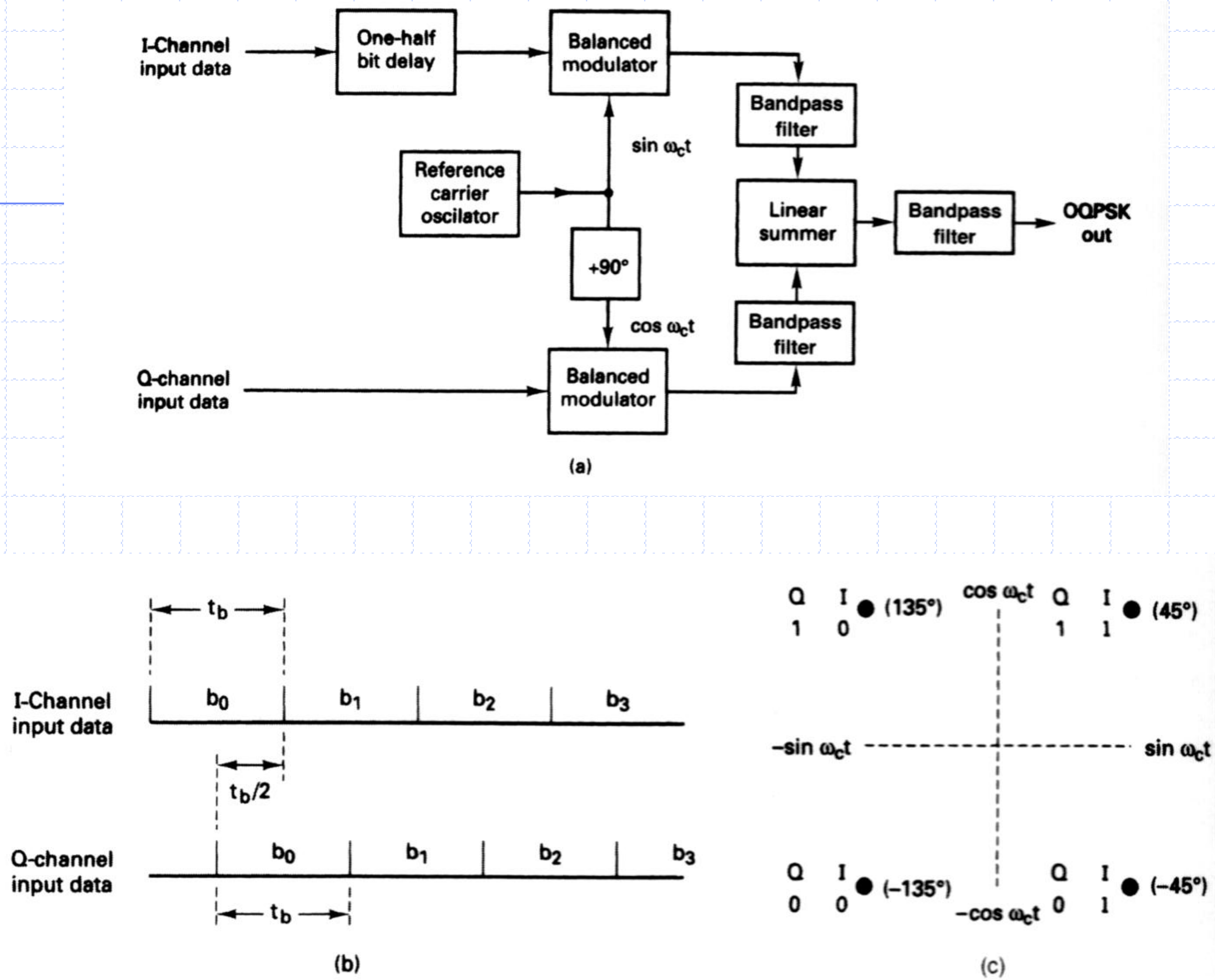


FIGURE 9-22 Offset keyed (OQPSK): (a) block diagram; (b) bit alignment; (c) constellation diagram

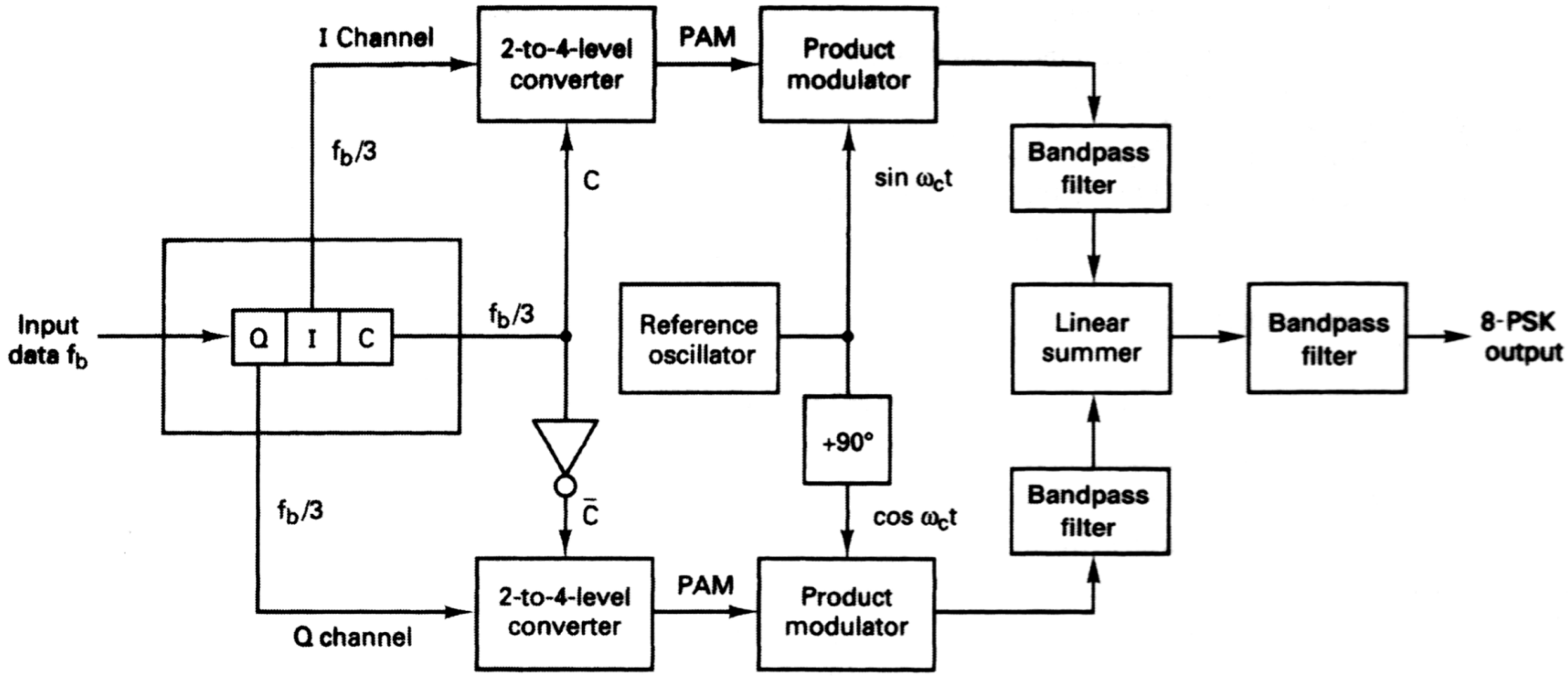
8-PSK

- With 8-PSK, three bits are encoded, forming tribits and producing eight different output phases.
- With 8-PSK, $n = 3$, $M = 8$ and there are eight possible output phases.
- To encode eight different phases, the incoming bits are encoded in groups of three, called tribits ($2^3 = 8$).

8-PSK Modulator

- The incoming serial bit stream enters the bit splitter where it is converted to a parallel, three channel output.
  - I is in-phase channel
  - Q is in-quadrature channel
  - C is control channel
- Consequently, the bit rate in each of the three channel is $f_b/3$.
- The bits in the I and C channels enter the I channel 2-to-4 level converter and the bits in the Q and C channels enter the Q channel 2-to-4 level converter.

8-PSK Modulator



8-PSK Modulator

- The 2-to-4 level converter are parallel-input digital-to-analog converters (DAC). With two input bits, four output voltages are possible.
- The I and Q bits determines the polarity of the output analog signal (logic 1 is +V and logic 0 is –V).
- Whereas the C or C bit determines the magnitude (logic 1 = 1.307 V and logic 0 = 0.541 V).
- Consequently, with two magnitudes and two polarities, four different output conditions are possible.

8-PSK Modulator

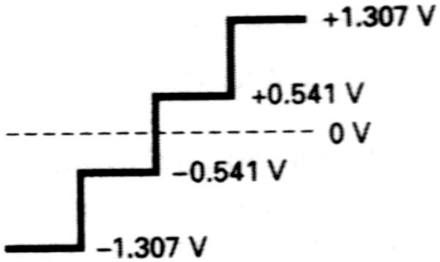
- Because C and C bits can never be the same logic state, the outputs from the I and Q 2-to-4 level converter can never have the same magnitude, although they can have the same polarity.
- The output of the 2-to-4 converter is an M-ary, Pulse Amplitude Modulated (PAM) signal where M = 4.

I	C	Output
0	0	−0.541 V
0	1	−1.307 V
1	0	+0.541 V
1	1	+1.307 V

(a)

Q	$\bar{C}$	Output
0	1	−1.307 V
0	0	−0.541 V
1	1	+1.307 V
1	0	+0.541 V

(b)



(c)

FIGURE 9-24 I- and Q-channel 2-to-4-level converters: (a) I-channel truth table; (b) Q-channel truth table; (c) PAM levels

8-PSK Modulator

Example 9-7

For a tribit input of Q = 0, 1 = 0, and C = 0 (000), determine the output phase for the 8-PSK modulator shown in Figure 9-23.

Solution The inputs to the I channel 2-to-4-level converter are I = 0 and C = 0. From Figure 9-24 the output is −0.541 V. The inputs to the Q channel 2-to-4-level converter are Q = 0 and $\overline{C}$ = 1. Again from Figure 9-24, the output is −1.307 V.

Thus, the two inputs to the I channel product modulators are −0.541 and $\sin \omega_c t$. The output is

$$I = (-0.541)(\sin \omega_c t) = -0.541 \sin \omega_c t$$

The two inputs to the Q channel product modulator are −1.307 V and $\cos \omega_c t$. The output is

$$Q = (-1.307)(\cos \omega_c t) = -1.307 \cos \omega_c t$$

The outputs of the I and Q channel product modulators are combined in the linear summer and produce a modulated output of

$$\begin{aligned} \text{summer output} &= -0.541 \sin \omega_c t - 1.307 \cos \omega_c t \\ &= 1.41 \sin(\omega_c t - 112.5°) \end{aligned}$$

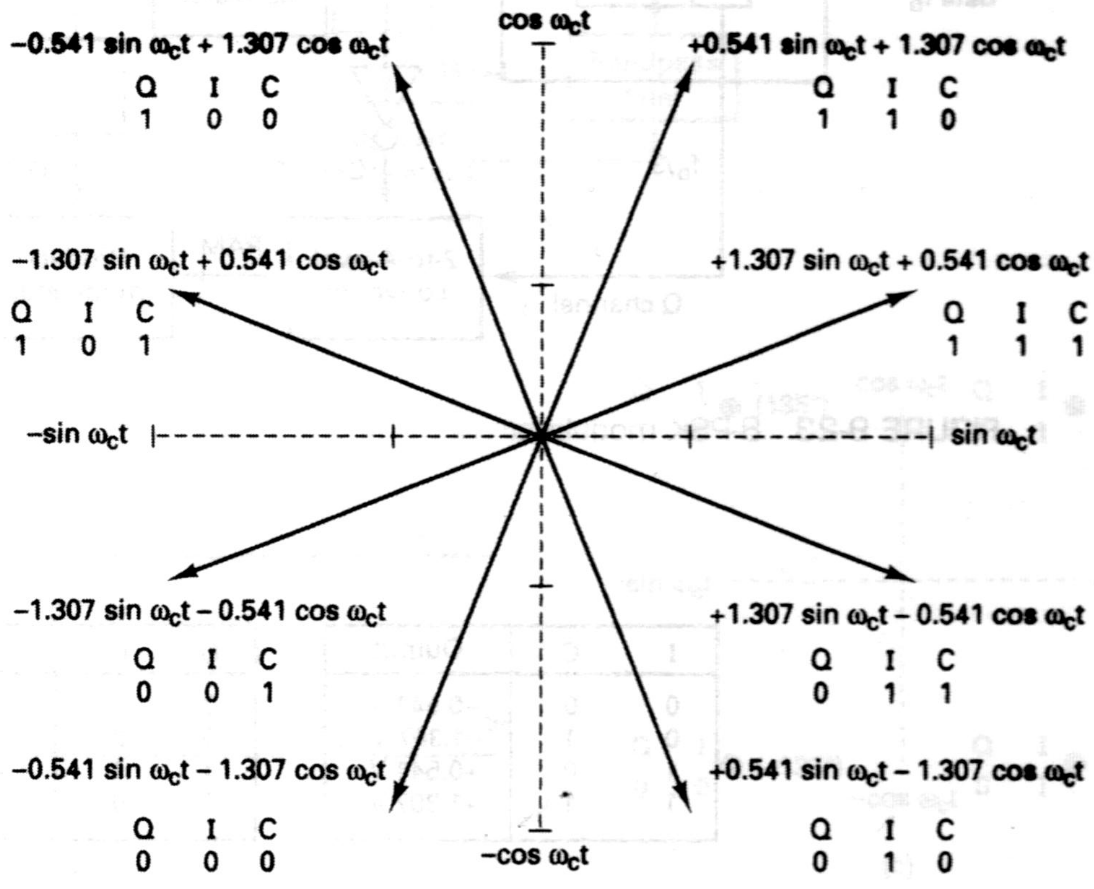
For the remaining tribit codes (001, 010, 011, 100, 101, 110, and 111), the procedure is the same. The results are shown in Figure 9-25.

8-PSK Modulator

Binary input Q	I	C	8-PSK output phase
0	0	0	−112.5°
0	0	1	−157.5°
0	1	0	−67.5°
0	1	1	−22.5°
1	0	0	+112.5°
1	0	1	+157.5°
1	1	0	+67.5°
1	1	1	+22.5°

(a)

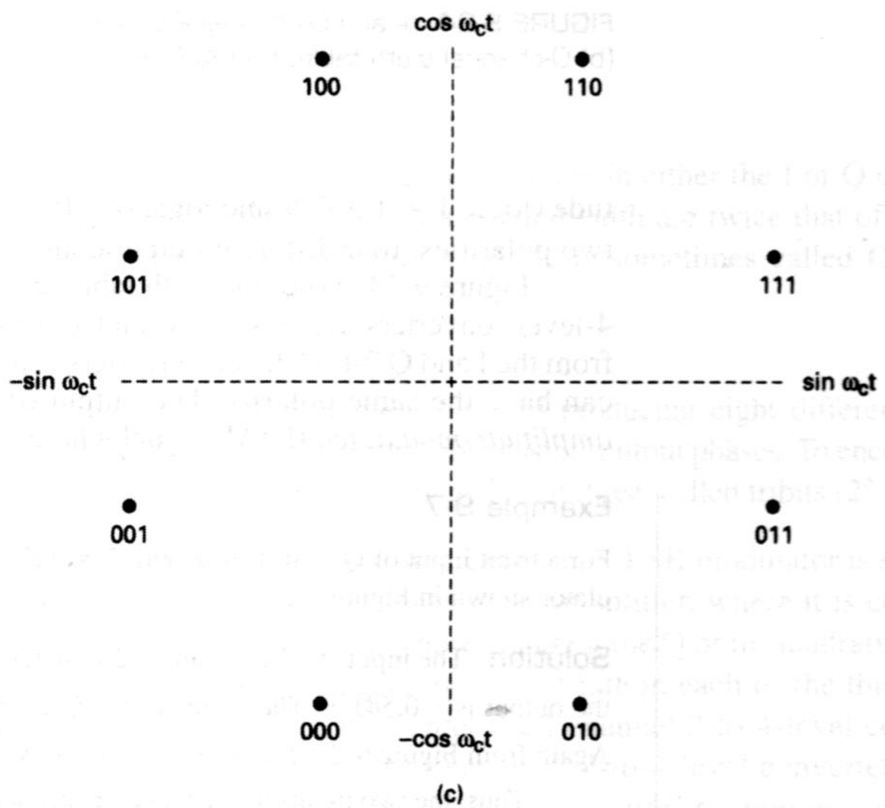
8-PSK truth table



(b)

Phasor Diagram

8-PSK Modulator

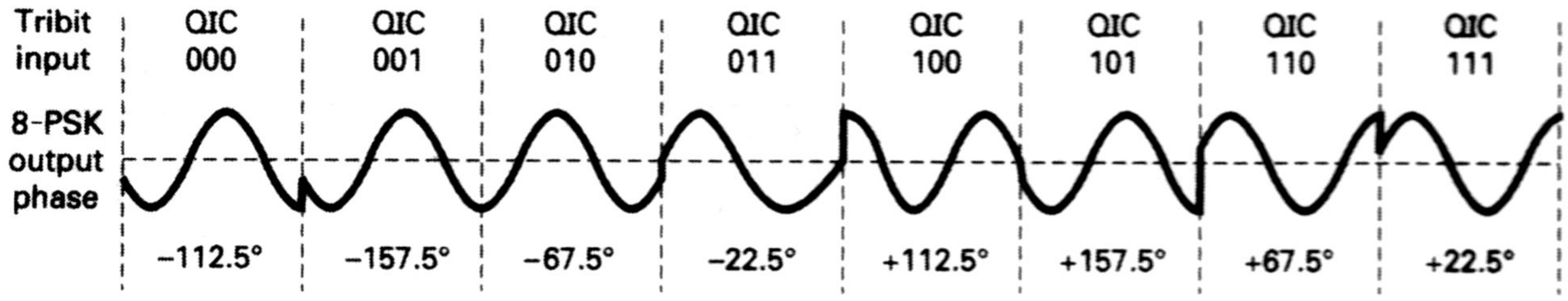


Constellation diagram

The constellation diagram shows that angular separation between any two adjacent phasors is 45°, half what it is with QPSK.

8-PSK can undergo almost ±22.5° phase shift during transmission.

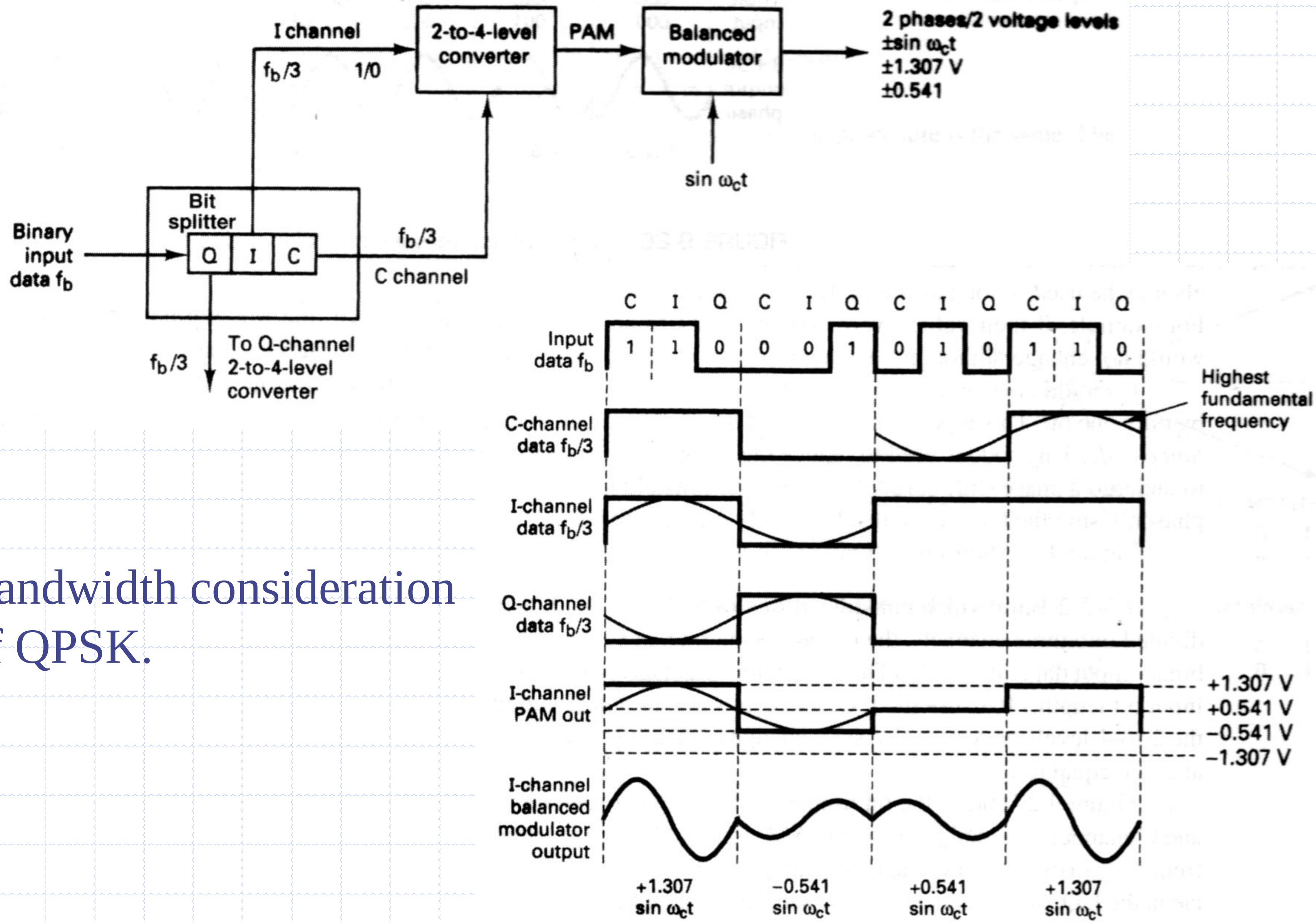
8-PSK Modulator



Phase versus time relationship for an 8-PSK

Bandwidth of 8-PSK

- Because the data are divided into three channels, the bit rate in the I, Q or C channel is equal to one-third of the binary input data rate ($f_b/3$).
- Because the I,Q and C are outputted simultaneously and in parallel, the 2-to-4 level converters also see a change in their input at a rate of $f_b/3$.
- The next figure shows that the highest fundamental frequency in the I,Q or C channel is equal to one-sixth the bit rate of the binary input.
- There is one change in phase at the output for every three data input bits. Consequently, the baud for 8-PSK equals $f_b/3$, the same as the minimum bandwidth.



Bandwidth consideration of QPSK.

Bandwidth of 8-PSK

Mathematically, the output of the balanced modulators is:

$$\theta = (X \sin \omega_a t)(\sin \omega_c t)$$

where

$$\underbrace{\omega_a t = 2\pi \frac{f_b}{6} t}_{\text{modulating signal}} \quad \text{and} \quad \underbrace{\omega_c t = 2\pi f_c t}_{\text{carrier}}$$

and

$$X = \pm 1.307 \text{ or } \pm 0.541$$

Thus,

$$\theta = \left(X \sin 2\pi \frac{f_b}{6} t\right)(\sin 2\pi f_c t)$$

$$= \frac{X}{2} \cos 2\pi\left(f_c - \frac{f_b}{6}\right)t - \frac{X}{2} \cos 2\pi\left(f_c + \frac{f_b}{6}\right)t$$

The output frequency spectrum extends from $f_c + f_b/6$ to $f_c - f_b/6$, and the minimum bandwidth (f_N) is

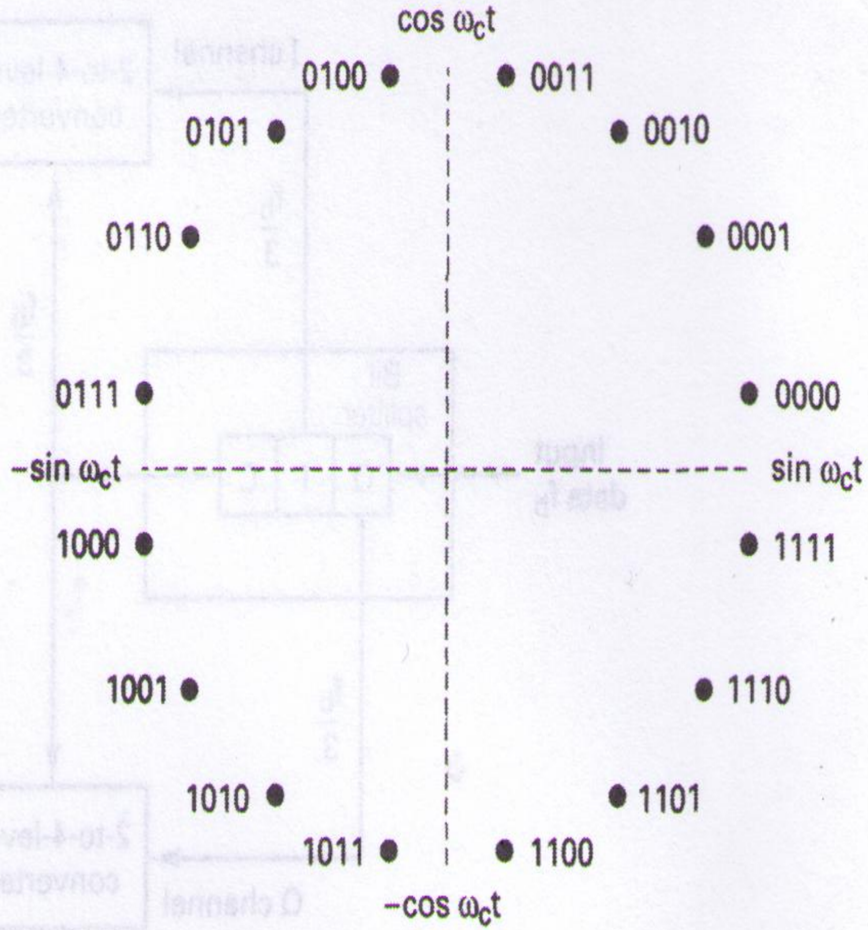
$$\left(f_c + \frac{f_b}{6}\right) - \left(f_c - \frac{f_b}{6}\right) = \frac{2f_b}{6} = \frac{f_b}{3}$$

16-PSK

- 16-PSK is M-ary encoding technique where M=16 giving 16 different output phases.
- The baud rate is fb/4
- 4 bits can be encoded with angular separation is 22.5° between adjacent channel therefore can undergo 11.25° phase shift during transmission.
- However, if more data is encoded i.e 6 bits, the angular separation is only 5.6° giving 64 output phases, the receiver has the difficulty to discern the phase and signaling element.

Bit code	Phase	Bit code	Phase
0000	11.25°	1000	191.25°
0001	33.75°	1001	213.75°
0010	56.25°	1010	236.25°
0011	78.75°	1011	258.75°
0100	101.25°	1100	281.25°
0101	123.75°	1101	303.75°
0110	146.25°	1110	326.25°
0111	168.75°	1111	348.75°

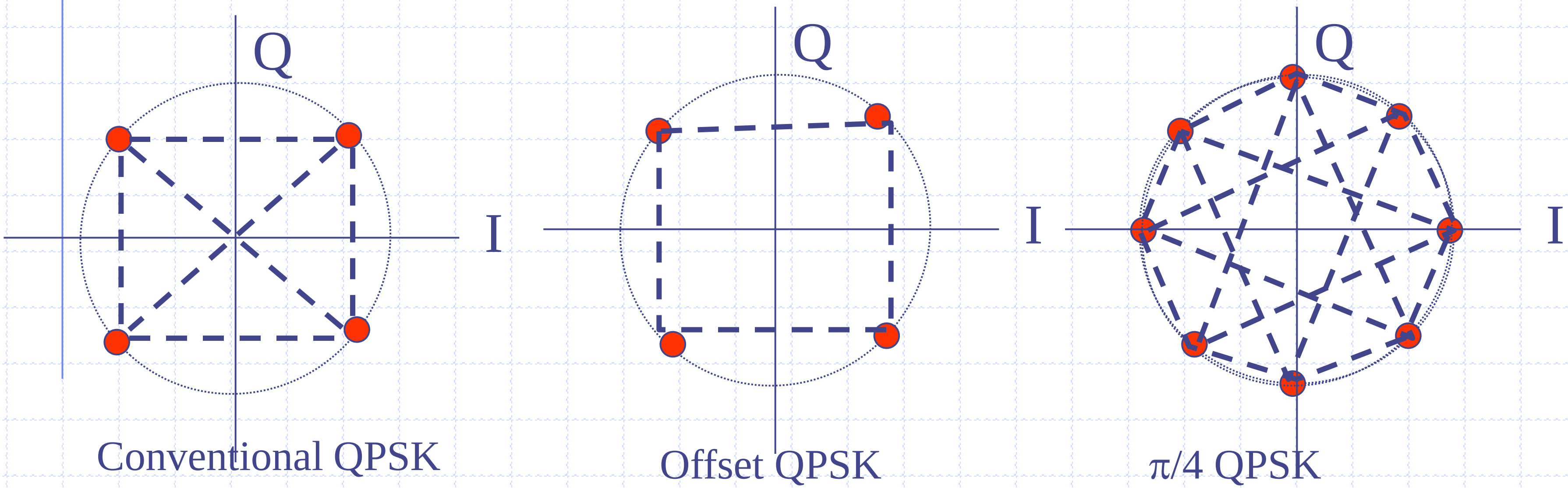
(a)



(b)

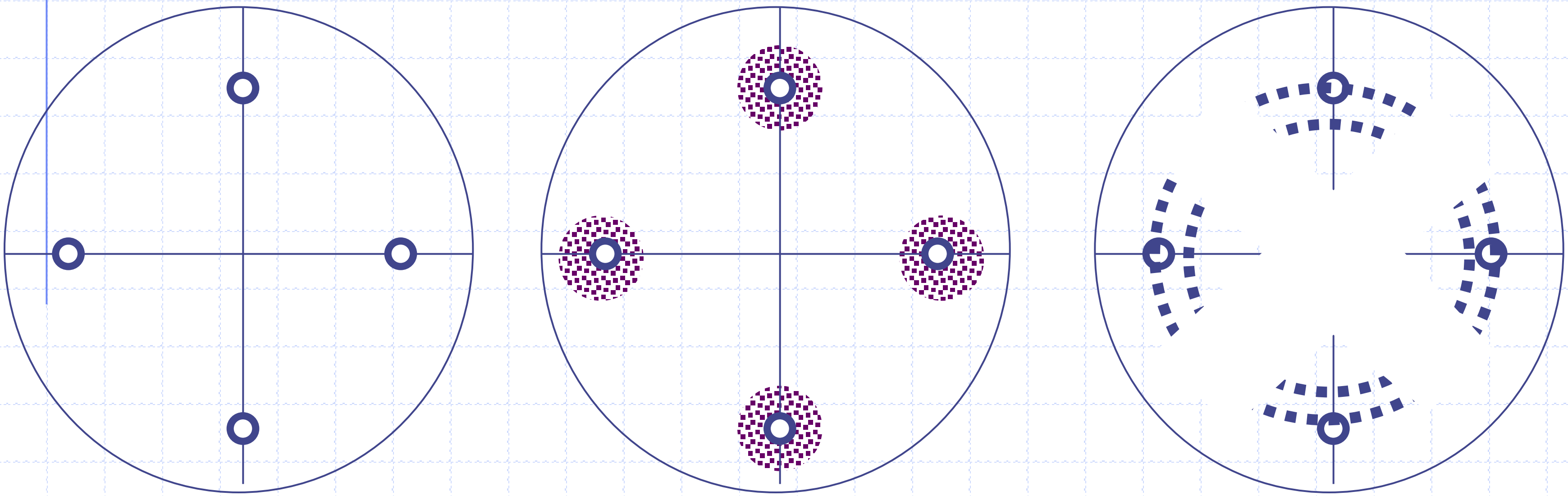
16-PSK truth table and constellation diagram

Types of QPSK



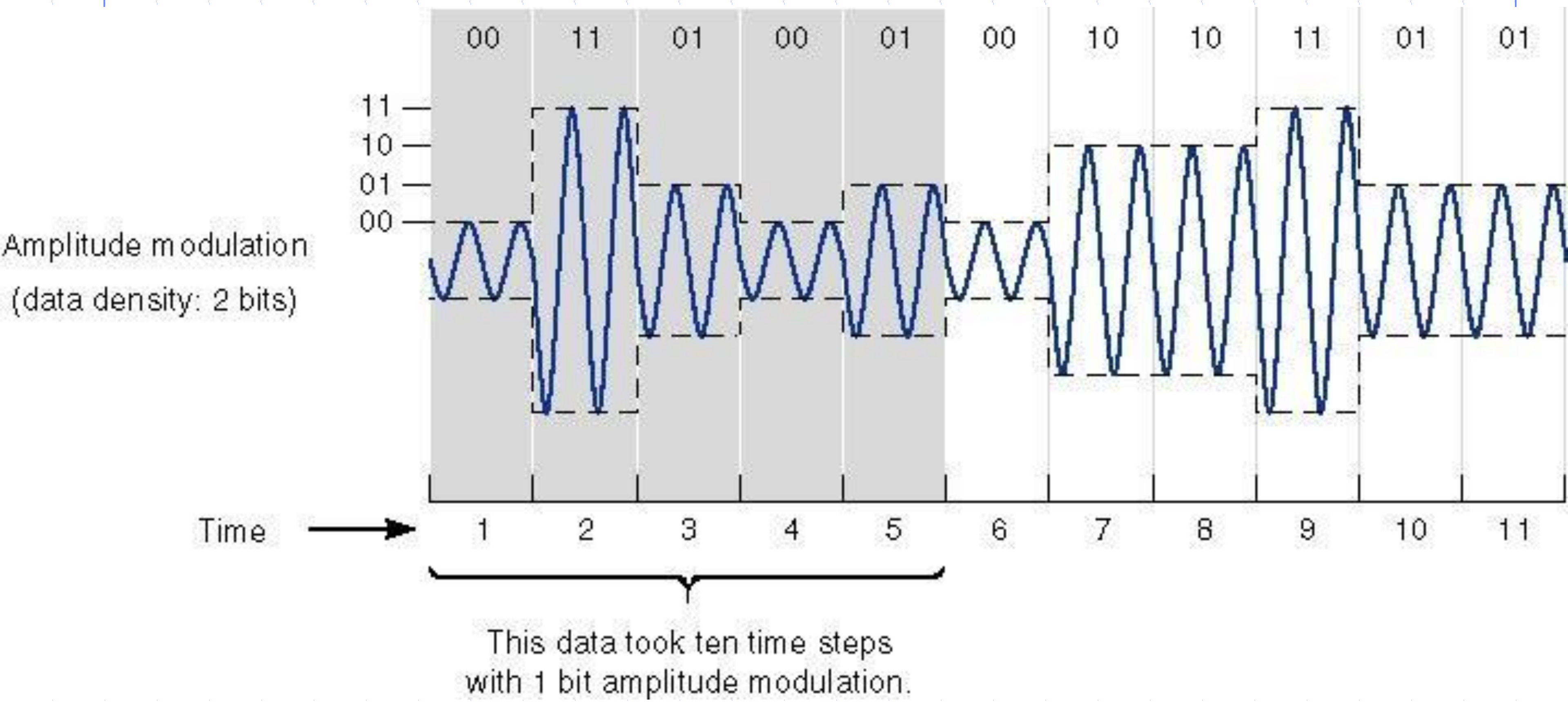
- Conventional QPSK has transitions through zero (i.e. 180^0 phase transition). Highly linear amplifiers required.
- In Offset QPSK, the phase transitions are limited to 90^0, the transitions on the I and Q channels are staggered.
- In $\pi/4$ QPSK the set of constellation points are toggled each symbol, so transitions through zero cannot occur. This scheme produces the lowest envelope variations.
- All QPSK schemes require linear power amplifiers

Distortions



Perfect channel White noise Phase jitter

Sending Multiple Bits Simultaneously



Sending Multiple Bits Simultaneously

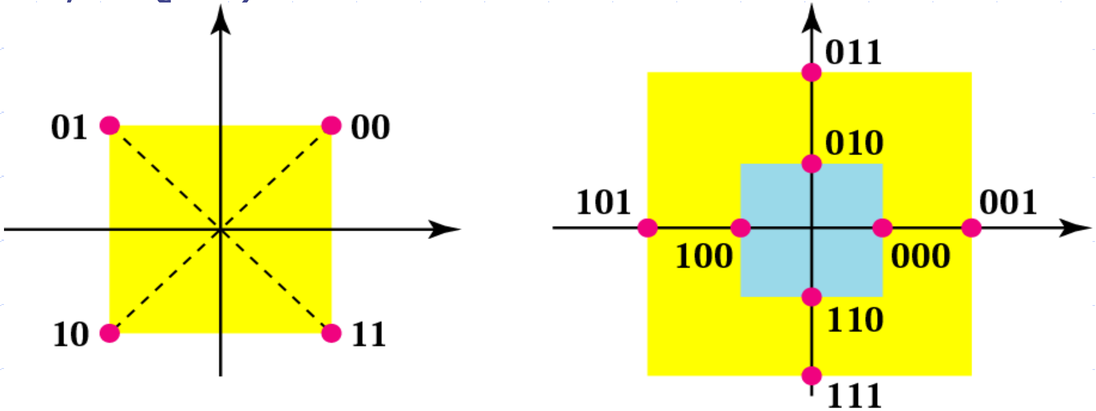
In practice, the maximum number of bits that can be sent with any one of these techniques is about five bits. The solution is to combine modulation techniques.

One popular technique is quadrature amplitude modulation (QAM) involves splitting the signal into eight different phases, and two different amplitude for a total of 16 different possible values.

Quadrature Amplitude Modulation

- PSK is limited by the ability of the equipment to distinguish between small differences in phases.
  - Limits the potential data rate.
- Quadrature amplitude modulation is a combination of ASK and PSK so that a maximum contrast between each signal unit (bit, dibit, tribit, and so on) is achieved.
  - We can have x variations in phase and y variations of amplitude
  - x • y possible variation (greater data rates)
- Numerous variations. (4-QAM, 8-QAM)

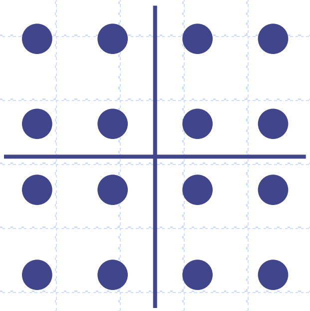
# of phase shifts > # of amplitude shifts



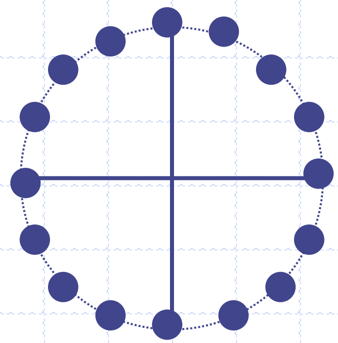
4-QAM
1 amplitude, 4 phases

8-QAM
2 amplitudes, 4 phases

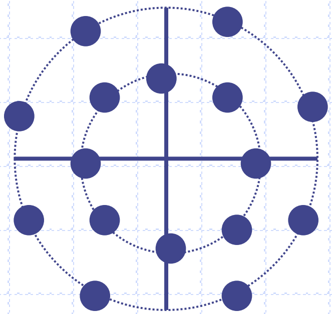
Multi-level (M-ary) Phase and Amplitude Modulation



16 QAM



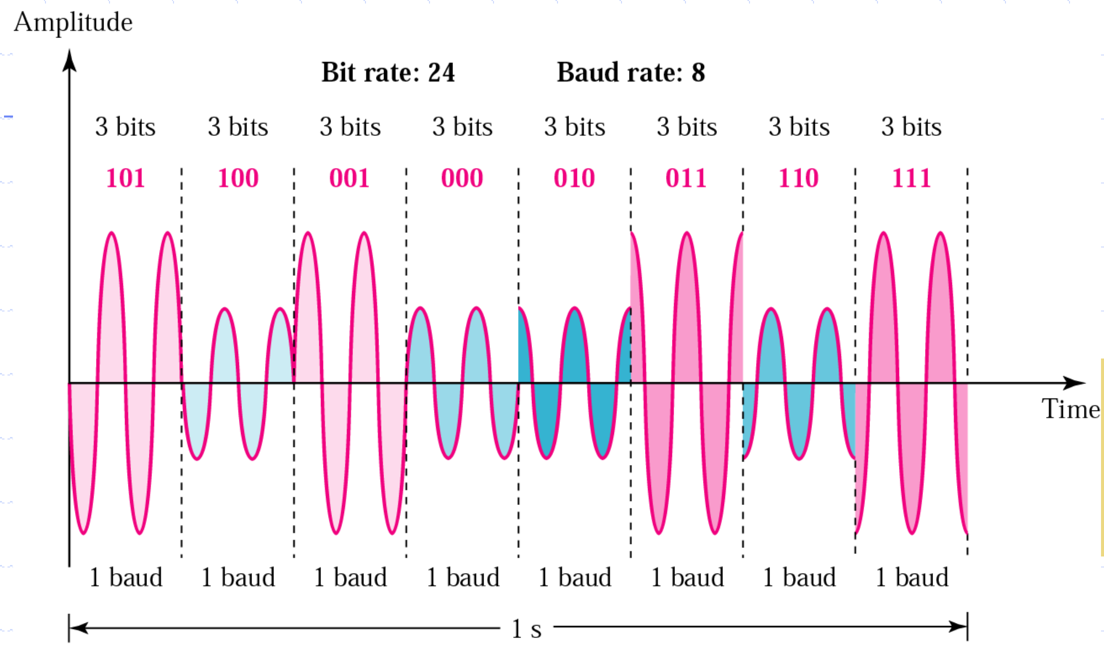
16 PSK



16 APSK

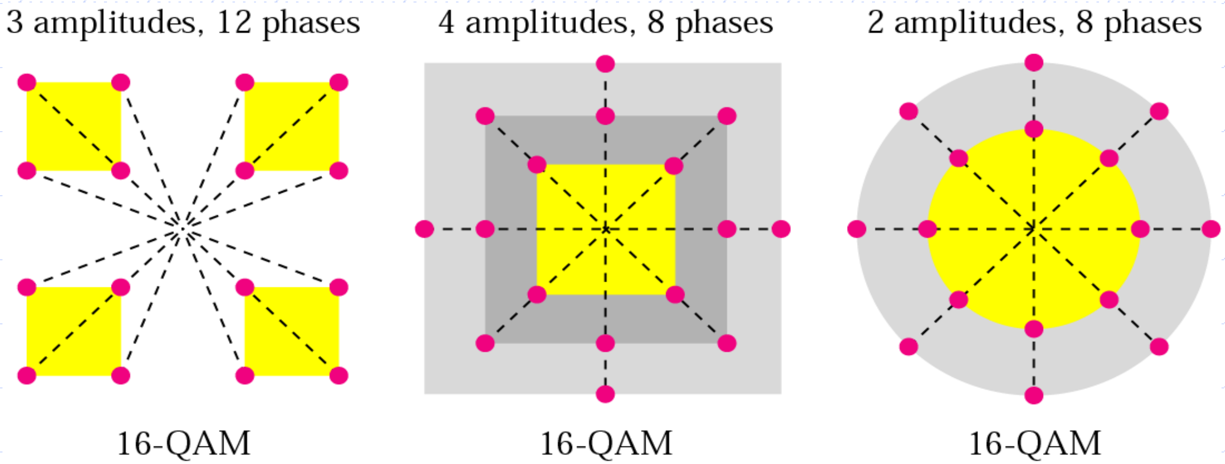
- Amplitude and phase shift keying can be combined to transmit several bits per symbol.
  - Often referred to as *linear* as they require linear amplification.
  - More bandwidth-efficient, but more susceptible to noise.
- For M=4, 16QAM has the largest distance between points, but requires very linear amplification. 16PSK has less stringent linearity requirements, but has less spacing between constellation points, and is therefore more affected by noise.

8-QAM and 16-QAM

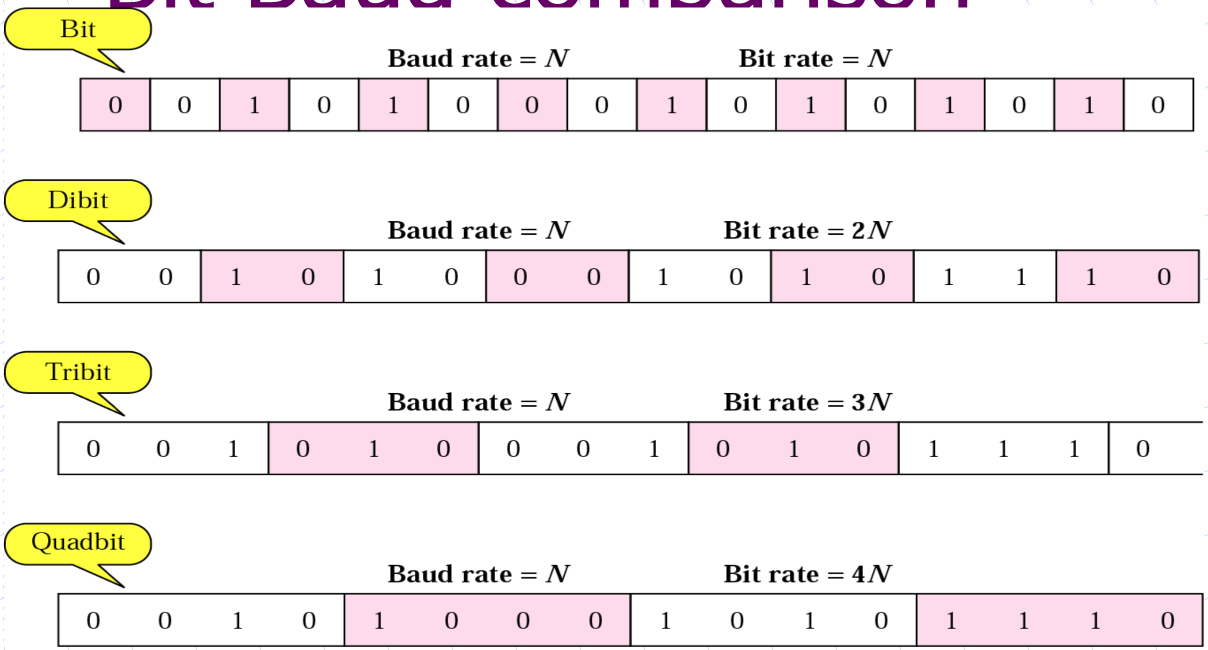


Second example, recommendation of OSI. not all possibilities are used, to increase readability of signal, measurable differences between shifts are increased

First example handles noise best
Because of ratio of phases to amplitudes
ITU-T recommendation.



Bit Baud comparison



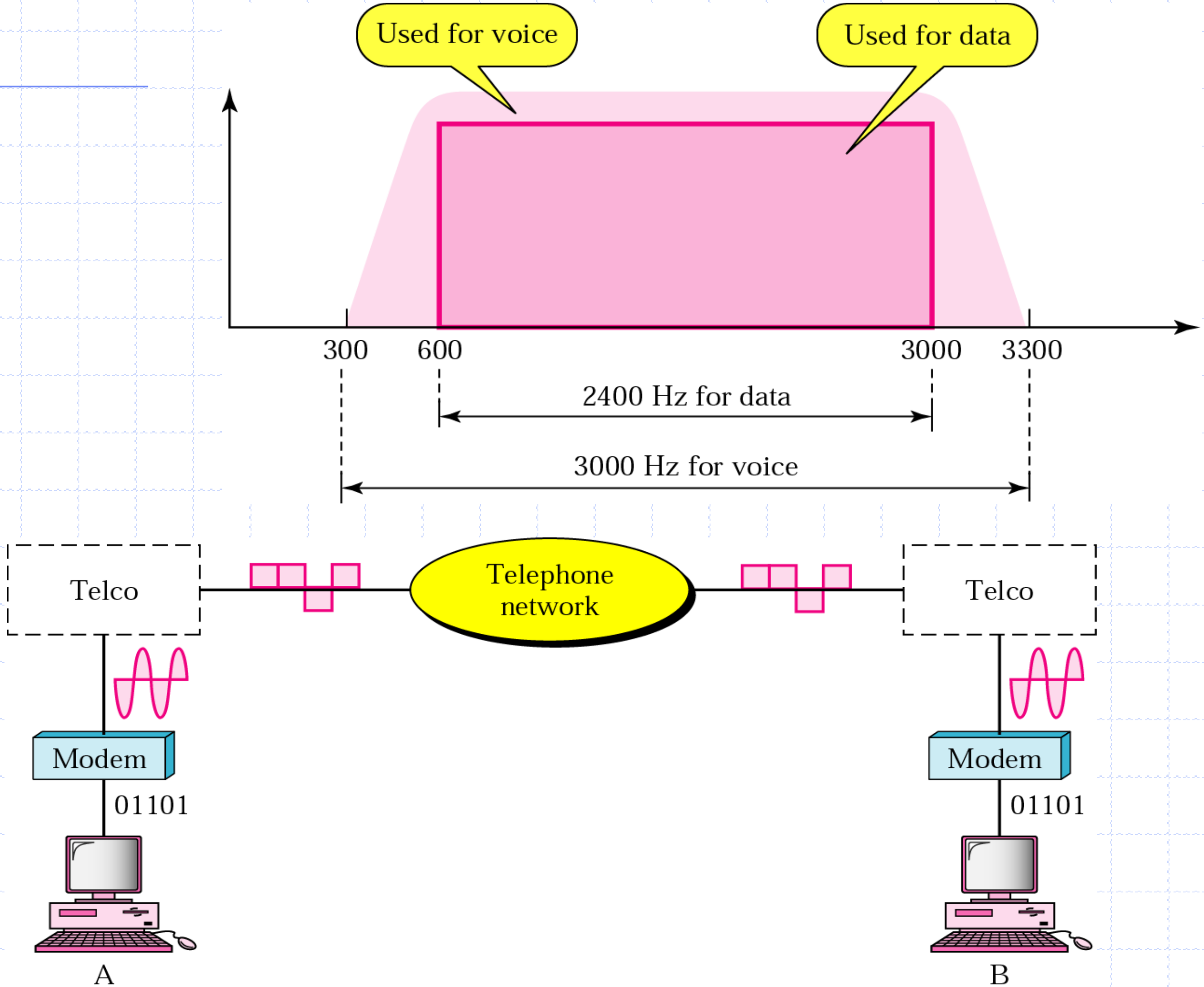
- Assuming a FSK signal over voice-grade phone line can send 1200 bps, it requires 1200 signal units to send 1200 bits (each frequency shift represents one bit, baud rate 1200)
- Assuming 8-QAM, baud rate is only 400 to achieve same data rate.

Modulation	Units	Bits/Baud	Baud rate	Bit Rate
ASK, FSK, 2-PSK	Bit	1	N	N
4-PSK, 4-QAM	Dibit	2	N	2N
8-PSK, 8-QAM	Tribit	3	N	3N
16-QAM	Quadbit	4	N	4N
32-QAM	Pentabit	5	N	5N
64-QAM	Hexabit	6	N	6N
128-QAM	Septabit	7	N	7N
256-QAM	Octabit	8	N	8N

Bit Baud comparison (examples)

- A constellation diagram consists of eight equally spaced points on a circle. If the bit rate is 4800 bps, what is the baud rate?
  - The constellation indicates 8-PSK with the points 45 degrees apart. Since $2^3 = 8$, 3 bits are transmitted with each signal unit. Therefore, the baud rate is 4800 / 3 = 1600 baud
- What is the bit rate for a 1000-baud 16-QAM signal.
  - A 16-QAM signal has 4 bits per signal unit since $\log_2 16 = 4$. Thus, (1000)(4) = 4000 bps
- Compute the baud rate for a 72,000-bps 64-QAM signal.
  - A 64-QAM signal has 6 bits per signal unit since $\log_2 64 = 6$.
  - Therefore, 72000 / 6 = 12,000 baud

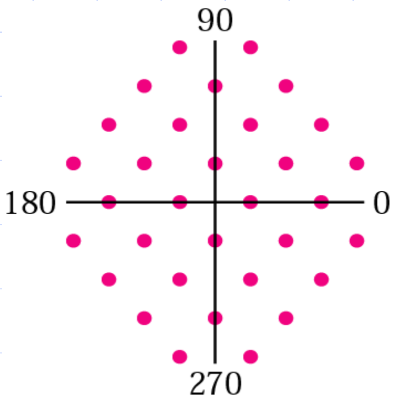
Phone modems

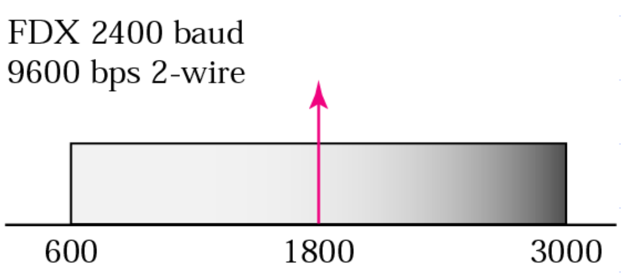


V-series (ITU-T Standards)

- V-32.
  - Uses a combined modulation and encoding technique: Trellis coded Modulation.
  - Trellis = QAM + a redundant bit
  - 5 bit (pentabit) = 4 data + 1 calculated from data.
  - A signal distorted by noise can arrive closer to an adjacent point than the intended point (extra bit is therefore used to adjust)
  - Less likely to be misread than a QAM signal.

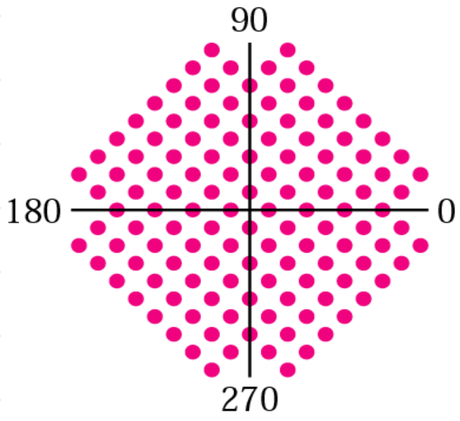
2400 baud • 4 bit = 9600 bps

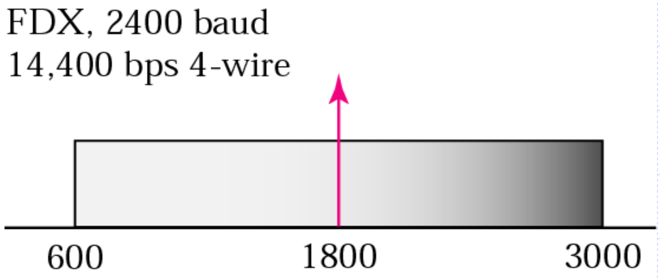




V-series (cont.)

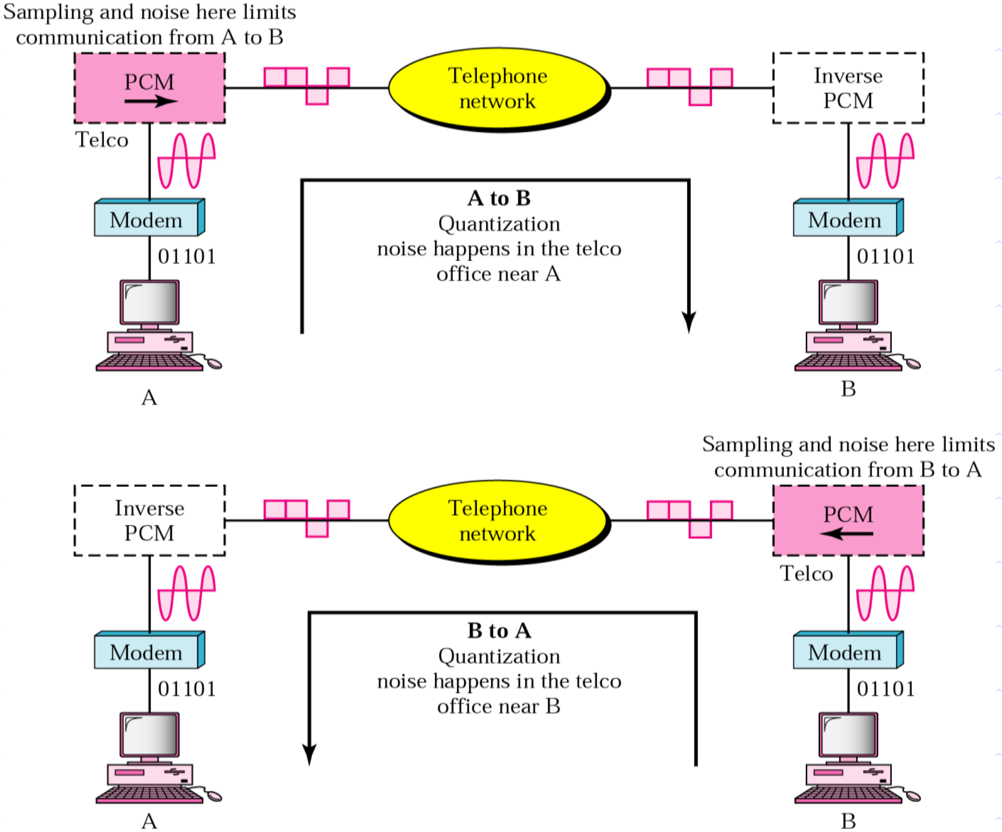
- V-32.bis
  - First ITU-T standard to support 14,400 bps transmission.
  - Uses 128-QAM
    - (7 bits/baud with 1 bit for error control)
    - at a rate 2400 baud 2400 * 6 = 14,400 bps
  - Adjustment of the speed upward or downward depending on the quality of the line or signal
- V-34 bis
  - Bit rate of 28,800 bps with 960-point constellation to 1664-point constellation for a bit rate of 33,600 bps.





V-series (cont.)

- V-90
  - Traditionally modems have a limitation on data rate (max. 33.6 Kbps)
  - V-90 modems can be used (up to 56Kbps) if using digital signaling.
    - For example, Through an Internet Service Provider (ISP)
  - V-90 are asymmetric
    - Downloading rate (from ISP to PC) has a 56 Kbps limitation.
    - Uploading rate (from PC to IST) has a 33.6 Kbps limitation.



V-90/92 (cont.)

- V-90
  - In uploading, signal still to be sampled at the switching station. Limit due to noise sampling.
  - Phone company samples at 8000 times/sec with 8 bits (including bit error)
    - Data rate = 8000 * 7 = 56Kbps
  - In download, signal is not affected by sampling.
- V-92
  - Speed adjustment
  - Upload data at 48Kbps.
  - Call waiting service