# RADIO PROPAGATION MODELS AND NARROWBAND DIGITAL MODULATION

## A) PROPAGATION MODELS FOR WIRELESS COMMUNICATION SYSTEM

### INTRODUCTION

The wireless communication system posses several challenges for the reliable and a high-speed communication. It is not receptive of noise channel and other channel hindrance, but these obstacle changes with time in unforeseeable ways due to user movement. We will characterize in detail the variation in the received signal power over the distance due to path loss and shadowing. Path loss models describe the signal attenuation between a transmitter and receiver antenna as a function of propagation distance and other parameters which is caused by the dissipation of the power radiated by the transmitter as well as effects of the propagation channel. Shadowing is caused by obstruction between the transmitter and the receiver that attenuate the signal power through absorption, reflection, scattering, and diffraction. A very important practical issue is to test and validate the ability of the "smart" antenna array to meet performance requirements. For this purpose, a channel model is needed to take into account the temporal and spatial characteristics of radio propagation.

### WIRELESS CHANNEL

The wireless signal proliferates in space, based on the rule of physics. An electromagnetic Radio Frequency (RF) signal which proceed in a medium suffers an attenuation (path loss) based on the nature of the medium. In addition, thesignal experiences objects and gets reflected, refracted, diffracted, and scattered. The cumulative effect results in the signal getting absorbed, signal travel across multiple paths, signal's frequency being shifted due to relative motion between the source and objects (Doppler Effect), thus are getting modified in a sufficient way. It is clear that the radio frequency signal is a space-time-frequency signal.

### CHARACTERISTICS OF WIRELESS CHANNEL

The main characteristics of wireless communication channel are as follows:

1. Path loss
2. Fading and shadowing
3. Interference
4. Doppler shift

#### Path loss

Path loss can be expressed as the ratio of power of transmitted signal to the power of the same signal received by the receiver on a given path. It is a function of the propagation distance.

- Estimation of path loss is very important for designing and deploying wireless communication networks.
- Path loss depends on the number of factors such as the radio frequency used and the nature

of the terrain.

• The free space propagation model is the simplest path loss model in which there is a direct-path signal between the transmitter and the receiver with no atmosphere attenuation or multipath components. In this model, the relationship between the transmitted power ***Pt*** and the received power ***Pr*** is given by

$$P_r = P_t G_t G_r \left(\frac{\lambda}{4\Pi d}\right)^2$$

Where,

- $\mathbf{G_t}$ is the transmitter antenna gain
- $\mathbf{G_r}$ is. the receiver antenna gain
- **d** is the distance between the transmitter and receiver
- **λ** is the wavelength of the signal

Two-way model also called as two path models is widely used path loss model. The free space model gives a detail amount of above assumes that there is only one single path from the transmitter to the receiver. It is actually experienced that the signal reaches the receiver through the multiple paths. The two-path model struggles to capture this phenomenon. The model assumes that the signal reaches the receiver through two paths, one a line-of-sight and the other the path through which the reflected wave is received. According to the two-path model, the power which is received is given by

$$P_r = P_t G_t G_r \left(\frac{h_t h_r}{d^2}\right)^2$$

Where,

- $\mathbf{P_t}$ is the transmitted power
- $\mathbf{G_t}$ represent the antenna gain at the transmitter
- $\mathbf{G_r}$ represent the antenna gain at the receiver
- **d** is the distance between the transmitter and receiver
- $\mathbf{h_t}$ is the height of the transmitter
- $\mathbf{h_r}$ are the height of the receiver

**Fading.**

Fading mentions the fluctuations in strength of the signal when the signal is received at the receiver. Fading can be classified into two types –

• Fast fading/small scale fading and

• Slow fading/large scale fading

Fast fading refers to the swift fluctuations in the amplitude, phase or multipath delays of the received signal, due to the interference between the multiple versions of the same transmitted signal arriving at the receiver at slightly different time interval. The time between the reception of the first version of the signal and the last echoed signal can be expressed as delay spread. The multipath propagation of the transmitted signal, which causes fast fading, is because of the three propagation mechanisms, namely –

• Reflection

• Diffraction

• Scattering

The multiple signal paths may sometimes add constructively or sometimes destructively at the receiver causing a variation in the received signal's power level. The received single envelope of a fast-fading signal is said to follow a Rayleigh distribution to see if there is no line-of-sight path between the transmitter and the receiver.

**Slow Fading**

The name Slow Fading itself indicates that the signal fades away slowly. The features of slow fading are as given below.

- Slow fading occurs when objects that partially absorb the transmission lie between the transmitter and the receiver.
- Slow fading is so called because the duration of the fade may last for multiple seconds or minutes.
- When the receiver is inside a building and the radio wave passes through the walls of a building slow fading occurs. The blocking object causes an irregular variation in the power of received signal.
- Slow fading may cause the received signal power to vary, though the distance between the transmitter and receiver remains the same.
- Slow fading can also be expressed as the shadow fading since the objects that cause the fade, which may be large buildings or other structures, block the direct transmission path from the transmitter to the receiver.

**Interference.**

Interference is the sum of all signal contributions that are neither noise not the wanted signal. Let's understand how its effect, its type and what possible source for it.

**Effects of Interference.**

- Interference is an important limiting factor in the performance of cellular systems.
- Interference degrades the quality of the signal.
- It initiates bit errors in the received signal.
- Bit errors are partly recoverable by means of the channel coding and the error correction mechanisms.
- The situation of the interference is not reciprocal to the uplink and downlink direction.
- Mobile stations and base stations are introduced to different interference situation.

**Sources of Interference**

- When another mobile is present in the same cell.
- When a call is in progress in the neighbouring cell.
- When other base stations are operating on the same frequency.
- When any non-cellular system leaks energy into the cellular frequency band.

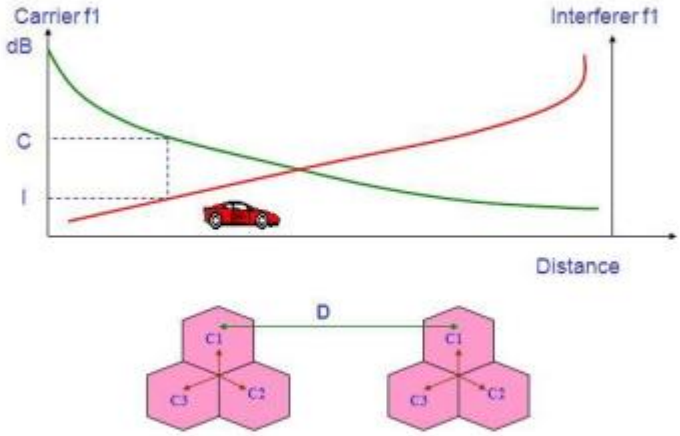



**Fig-1**: Interference

**Types of Interference**
There are two types of system generated interference
1. Co-channel interference
2. Adjacent channel interference

**Co-Channel Interference**
• Co-channel interference occurs the because of frequency reuse, i.e., several cells use the same set of frequency.
• These cells are called co-channel cells.
• Co-channel interference cannot be combated by increasing the power of the transmitter. This is because an increase in carrier transmit power increases the interference to neighbouring cochannel cells.

• To reduce the co-channel interference, the cells must be separated by a minimum distance to provide sufficient isolation due to propagation or reduce the footprint of the cell.
• Some factors other than reuse distance that influence co-channel interference are antenna type, directionality, height, site position etc.

**Adjacent channel interference**
• Interference concluding from the signals which are adjacent in frequency to the desired signal is called adjacent channel interference.
• Adjacent channel interference results from imperfect receiver filters which allow nearby frequencies to leak into the pass band.
• Adjacent channel interference can be minimized through channel assignments and careful filtering.
• By keeping the frequency separation between each channel in a given cell as large as possible, the adjacent interference may be reduced considerably.

**DOPPLER SHIFT**
The Doppler Effect is named after Austrian physicist Christian Doppler who proposed it in 1842. Doppler shift is referred as the change in frequency of a wave for an observer moving

relative to the source of the wave. It is heard when a vehicle sounding a siren or horn approaches, passes, and recedes from an observer. The frequency is higher at the instant when it is emitted. The frequency is identical at the instant of passing by, and it is lower during the recession. For the waves that propagate in a medium like sound waves, where the velocity of the source and of the observer is corresponding to that of the medium in which the waves are transmitted. The total Doppler Effect may result from the motion of the observer, motion of the source, or motion of the medium. Each of these effects is examined separately. For the waves which do not require any medium, such as gravity or light in general relativity, only the relative difference in velocity between the source and the observer needs to be considered.

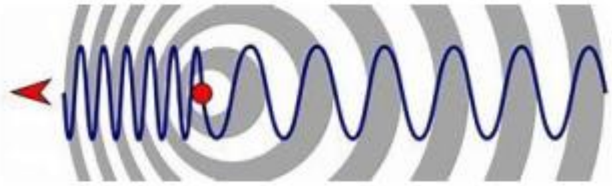


**Fig-2**: Doppler Shift

**Radio propagation model**

The radio propagation model is an experimental mathematical formulation for the characterization of radio wave propagation as a function of distance frequency and other conditions. A single model is usually developed to anticipate the behavior of propagation for all identical links under identical constraints. Create with the goal of formalizing the way radio waves are propagated from one place to another, such models typically predict the path loss along a link or the effective coverage area of a transmitter.

The propagation models are classified mainly into two types:

1. Outdoor propagation model.
2. Indoor propagation model.

**Outdoor propagation models**

In the mobile communication system radio transmission often takes place over discontinuous terrain. The terrain profile may vary from a simple curved earth profile to a highly mountainous profile. The presence of trees, buildings, and other obstacles also must be taken into account. A number of propagation models are available to predict path loss over irregular terrain. Some commonly used outdoor propagation models are now discussed.

**Longley-Rice Model**

The Longley-Rice model is application to point-to-point communication systems in the frequency range from 40 MHz to 100 GHz, over different kinds of terrain. The main transmission loss is predicted using the path geometry of the terrain profile and the refractive of the troposphere. Geometric optics techniques are used to predict signal strengths within the radio horizon. The Longley-Rice model is also available as a computer program to calculate large – scale median transmission loss relative to free space loss over irregular terrain for frequencies between 20 MHz and 10 GHz. The Longley-Rice method operates in two modes. For the availability of complete path terrain, the path-specific parameters can be easily determined and the prediction is called a point-to-point mode prediction. On the other hand, if the terrain path profile is not available, the Longley-Rice method provides techniques to estimate the path-specific parameters, and such a prediction is called an area mode prediction. There have been many predictions and corrections to the Longley-Rice model since its original publication. One important modification deals with radio propagation in urban areas and this particularly relevant to mobile radio. This modification introduces an excess term as an allowance for the additional attenuation due to urban clutter near the receiving antenna. This extra term called, urban factor, has been derived by comparing the prediction by the original Longley-Rice model with those obtained by Okumura.

**Okumura model**

Okumura model is one of the most widely used models for signal prediction in urban areas. This model is applicable for frequencies in the range 150 MHz to 1920 MHz and distances of 1km to 100 km. It can be used for base station antenna ranging from 30 m to 1000 m. Okumura developed a set of curve giving the median attenuation relative to free space (Amu) in an urban area over a quasi-smooth terrain with a base station effect antenna height (hte) of 200 m and the height of mobile antenna (hre) of 3m. To determine path loss using Okumura's model, the free space path loss between the points of interest is first determined, and then the value of Amu (f,d) is added to it along with correlation factors to account for the type of terrain.

**Hata Model:**

It is an empirical formulation of graphical path loss data provided by Okumura and is valid from 150MHz-1500MHz. Hata established empirical mathematical relationships to describe the graphical information given by Okumura. Hata's formulation is limited to certain ranges of input parameters and is applicable only over quasi-smooth terrain. The following expressions have considerably enhanced the practical value of the Okumura method, although Hata's formulations do not include any of the path specific corrections available in the original model.

**Durkin's Model**

It provides a perspective in to the nature of propagation over irregular terrain and the losses occur due to obstacles in a radio path. The demerit of this model is it cannot predict propagation effects due to foliage, buildings, and other human structures and doesn't support multi path communication. This model provides anticipations which satisfy well with measurements when the base station antenna is above rooftop height, giving mean output errors of about 3db with standard deviations in the range 4-8 db. However, the performance degrades as hb reaches hr and is quite poor when hb<<hr. The model produces much bigger errors in the microcellular situation.

**Indoor Propagation Models**

It provides a alternative in to the nature of propagation over irregular terrain and the losses occurred due to obstacles in a radio path. The disadvantage of this model is it cannot assumes propagation effects due to foliage, buildings, and other manmade structures and does not supports multi path communication.

**Free Space Path Loss**

The free space path loss model is not directly related with the indoor propagation. As it is required to compute the path loss at a close-in reference distance as desired by the models. The free space model gives a measure of path loss as a function of T-R separation when the receiver and transmitter are under the LOS range in a free space environment. The model is defined by equation given below, which depicts the path loss as a positive quantity in dB:

$$PL(d) = -10\log\left[\frac{G_t G_r \lambda^2}{(4\pi)^2 d^2}\right]$$

Where, Gt and Gr are the individual ratio gains of the transmitting and receiving antennas respectively, l gives the wavelength in meters, and d is the T-R separation in meters. When antennas are removed, we assume that Gt = Gr= 1. The free space path loss equation gives desired results only if the receiving antenna is in the far-field or Fraunhofer region of the transmitting antenna. The far-field denoted as the distance df given by equation below.

$$d_f = \frac{2D^2}{\lambda}$$

Here, D = largest linear dimension of the antenna. Additionally, for a receiver to be assumed in the far-field of the transmitter, it must satisfy df >> D and df >> λ

**Log-Distance Path Loss**

The log-distance path loss model assumes the path loss variations takes place exponentially with distance. The path loss in dB is given by equation (7.3).

$$\overline{PL}(d) = \overline{PL}(d_0) + 10n\log\left(\frac{d}{d_0}\right)$$

Where n gives the path loss exponent, d defines the T-R separation in meters, and do defines the close-in reference distance in meters. PL (do) is calculated using the free space path loss equation mentioned above. The value d0 should be considered such that it is in the far-field of the transmitting antenna; however, some small relative to any practical distance used in the mobile communication system. The path loss exponent value n varies according to the environment. In free space environment, n is equal to 2. In practice, the value of n is calculated using empirical data.

**Log-Normal Shadowing**

One major drawback of the log-distance path loss model is that it does not counts for shadowing effects which can be caused by changing degrees of clutter between the transmitter and receiver. The log-normal shadowing model tries to overcomes this. The log-normal shadowing model assumes path loss as a function of T-R separation.

$$PL(d) = \overline{PL}(d_0) + 10n\log\left(\frac{d}{d_0}\right) + X_\sigma$$

Where, Xs denotes the zero-mean Gaussian random variable with standard deviation s. Both Xs and s are defined in dBs. The random variable Xs tries to counteract for random shadowing effects that can result from clutter. The values n and s are considered from empirical data.

**Addition of Attenuation Factors to LogDistance Model**

Many researchers have attempted to enhance the logdistance model by taking into account certain additional attenuation factors based upon measured data. One relevant example is the attenuation factor model proposed by Seidel and Rappaport. The attenuation factor model includes a special path loss exponent and a floor attenuation factor to give an estimation of indoor path loss. The model is described in equation mentioned below:

$$PL(dB) = PL(d_0) + 10n_{sf} \log\left(\frac{d}{d_0}\right) + FAF$$

PL

where nsf provides the exponential path loss for a same floor measurement and FAF defines a floor attenuation factor completely based on the counts of floors between transmitter and receiver. Both nsf and FAF are approximated from empirical data. A quite familiar model was developed by Devasirvatham. Devasirvatham's model has an additional loss factor which improves exponentially with distance. The modified path loss equation is given below:

$$PL(d) = PL(d_0) + 20\log\left(\frac{d}{d_0}\right) + \alpha d + FAF$$

Where, α denotes an attenuation factor in dB/m for a defined channel. A third model includes additional attenuation factors. This new model was developed by Motley and Keenan and is of the form shown below:

$$PL(d) = PL(d_0) + 10n\log(d) + kF$$

Where, k gives the number of floors within the transmitter and receiver and F is the individual floor loss factor. The main point of difference between Motley and Keenan's model and the one developed by Seidel and Rappaport is that Motley and Keenan give an individual floor loss factor which is later multiplied by the number of floors separating transmitter and receiver. Seidel and Rappaport proposed a table comprising of floor attenuation factors which changes are based upon the number of floors separating the transmitter and receiver respectively.

**An Additive Path Loss Model**

An additional path loss model which has been found out by researchers is named as an additive path loss model. In this model, individual losses occurred due to obstructions between transmitter and receiver are approximated and added together. Researchers have proposed tables of recorded average attenuation values for different obstructions including walls, floors, and doors. However, maximum of the recorded information is related to only a few carrier frequencies. Furthermore, the resulting attenuations are not equal among various researchers.

# B) NARROWBAND DIGITAL MODULATION

**Objectives:**

(a) Quantitatively describe the relationship between a bit and a symbol, and the bit rate and the symbol rate (baud).

(b) Describe how digital information is conveyed using various digital modulation techniques (ASK or OOK, FSK, PSK and QAM) and recognize their waveforms, and constellations.

(c) Calculate the bandwidth of an ASK, OOK, FSK, PSK, or QAM signal.

(d) Using a constellation diagram analyze an M-ary PSK or QAM signal to determine its symbols and bits per symbols.

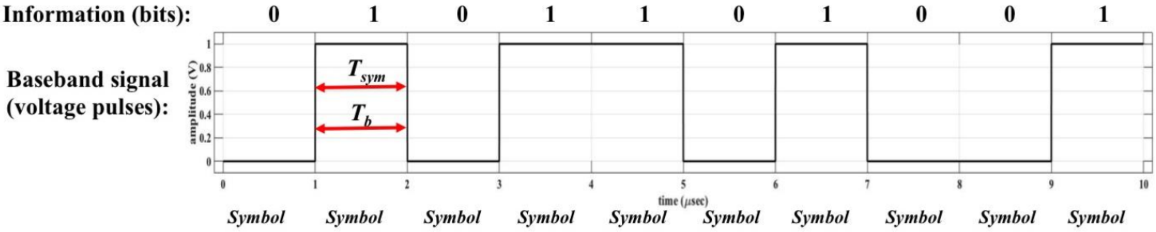
(e) Discuss the effect of noise on M-ary PSK and how Quadrature Amplitude Modulation (QAM) helps overcome these detrimental effects.

## 1. Information and Symbols

For a digital communication, the information we wish to send from the transmitter to the receiver are the bits. Bits of information can represent anything from ASCII characters in a Microsoft Word document, to numeric values that represent samples from an audio signal, to numeric values that represent the colors of pixels in a digital image. The information is carried in the bits that are transmitted, but we don't actually transmit bits; we transmit *waveforms that represent bits*. These waveforms are commonly referred to as *symbols*. Symbols are the physical means by which bits move from transmitter to receiver, and exactly how it is done depends on the communication medium being used.

If we wish to send bits over a wire, we usually use voltage pulses. For example, a high pulse may represent a 1-bit and a low pulse (or no pulse) may represent a 0-bit (or vice versa). In this case, the voltage pulses are the symbols, and each pulse carries 1 bit of information. Using voltage pulses, the transmitter is sending one of two possible symbols (e.g. a high pulse or a low pulse), and the process of sending digital information with voltage pulses forms a baseband (low frequency) signal.

Usually, we are concerned about how fast the information is being transmitted, and this relates to *symbol rate*, $R_{sym}$, which is the number of symbols per second being transmitted. Symbol rate is sometimes referred to as *baud* or *baud rate*, but they all mean the same thing. The figure below shows the relation between information (bits) and symbols (voltage pulses) for an example transmission.




In the figure above, there are 10 total bits being transmitted, and they are carried in the 10 symbols shown. The time it takes to send one symbol, $T_{sym}$, is 1 μsec as shown. The symbol rate is the inverse of the time to transmit one symbol, and the bit rate is the inverse of the time to transmit one bit, i.e.,

$$R_{sym} = \frac{1}{T_{sym}} \quad \text{and} \quad R_b = \frac{1}{T_b}.$$

In this example, since $T_{sym}$ is 1 μsec, then $R_{sym}=1/10^{-6} = 1 \times 10^6$ symbols/sec. And, since each symbol carries 1 bit of information (that is, 1 bit/symbol), the bit rate is $1 \times 10^6$ bits/sec = 1 Mbps. In general, the symbol rate and bit rate are related by:

$$R_b = N \cdot R_{sym},$$

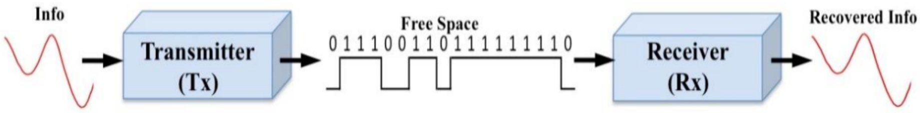
where $N$ is the number of bits per symbol. In the figure above, $N = 1$ bit/symbol, leading to $R_b$=1 Mbps. There is a relationship between the number of possible symbols that could be transmitted, and the number of bits per symbol. The number of symbols available for the transmitter to transmit is variable $M$: that is, there are possible $M$ symbols, and the relationship is:

$$N = \log_2(M).$$

For the example in the previous figure, there are two possible symbols for the transmitter to transmit, and so $N = \log_2(2) = 1$ bit/symbol.

**2. Digital Signal Frequency Spectrum**

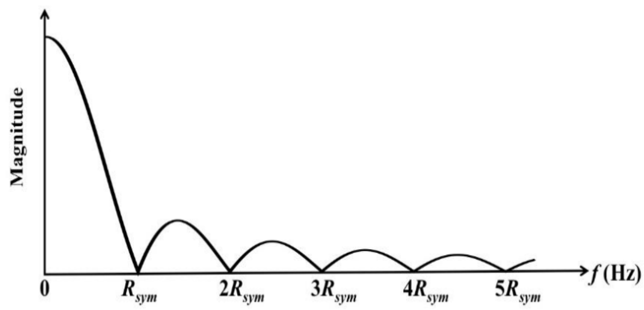
In Lesson 21, it was mentioned that in many cases, we wished to convert analog signals into digital signals to take advantage of the benefits of digital technologies. Samples of the analog signal were converted into bits and the bits were then used to create a binary voltage waveform that represented the bits. If we then wanted to transmit this digital waveform through free space, then all we need to do is connect the wire carrying the voltage pulses to an antenna, right?




No, it is not that easy. The binary voltage waveforms to which we are so accustomed are, typically, voltage pulses that alternate between 0V (for a 0-bit ) and 5V (for a 1-bit). It just so happens that the frequency content in these voltage pulses is predominantly very low (a baseband signal), and just like was pointed out for voice signals (which also have low frequency content), an antenna needed to transmit this kind of signal through free space would be impractically large.

For a large number of seemingly random 0-bit and 1-bit voltage pulses, as is normally the case in digital communication, the frequency spectrum of the pulses would take the following shape during the transmission. In this figure, the largest frequency content is at 0 Hz, and at regular intervals the frequency content goes to zero magnitude. This occurs at multiples of the symbol rate (or multiples of the bit rate because the bit rate is equal to the symbol rate), in Hz.

Frequency spectrum for random voltage pulses (a baseband signal—primarily low frequencies)

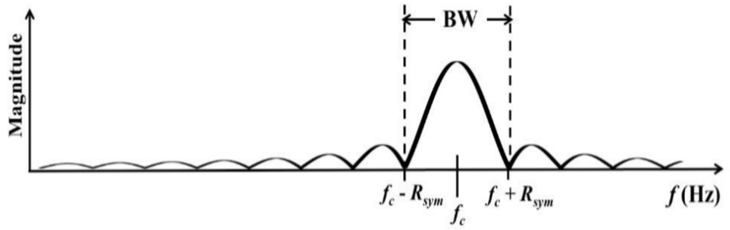



For example, if the symbol rate was 500 symbols/sec (so the bit rate was 500 bps), then the frequency content's magnitude would be zero at 500 Hz, 1000 Hz, etc. This plot of frequency content is much different than that of a signal composed of sinusoids! There are no spikes! Nevertheless, most of the frequency content is at very low frequencies. The frequency content does continue out to an infinite frequency, although the magnitude drops dramatically at higher frequencies. In a perfect world, we'd say the bandwidth of voltage pulses approaches ∞ Hz, but for digital signals, we'll use the *null-bandwidth* as our calculated bandwidth. The null-bandwidth is defined as the amount of the frequency spectrum (in Hz) from the maximum magnitude (which here occurs at 0 Hz) to where the spectrum first goes to a magnitude of 0 (called a *null*, here at $R_{sym}$ Hz). The bandwidth is given by:

$$BW = f_2 - f_1 = R_{sym} - 0 = R_{sym} \text{ Hz}$$

We must come up with a method to transmit the baseband digital information (1s and 0s) using electromagnetic waves, but since the frequency content is primarily low frequencies, as pointed out earlier, the antenna size would be impracticably large. Digital modulation techniques solve this problem. As you recall, one goal of modulation is to upshift the frequency spectrum of the information signal to allow transmission through free space using a reasonably-sized antenna. With digital modulation, the transmitted signal's frequency spectrum would then look like the following.

Frequency Spectrum for modulated voltage pulses (now a band pass signal—primary frequency content at a much higher frequency than the voltage pulses, centered at the carrier frequency)




Like in analog amplitude modulation, the information signal's frequency spectrum is shifted up by $f_c$ Hz, and there is a mirror image of the frequency content on the left side of $f_c$. The transmission bandwidth (using the null-bandwidth definition along with the fact that there is now a null to the left and right of the carrier frequency) is

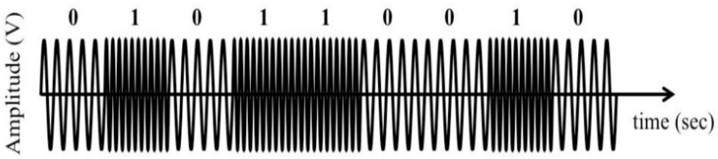
$$BW = f_2 - f_1 = \left(f_c + R_{sym}\right) - \left(f_c - R_{sym}\right) = 2R_{sym} \text{ Hz.}$$

Note that the bandwidth of the modulated signal is *twice* the bandwidth of the baseband signal (the voltage pulses).

### 3. Binary Digital Modulation

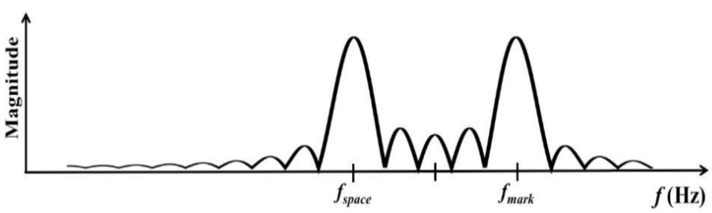
*Binary digital modulation* refers to types of modulation where there are two symbols, and so each symbol carries 1 bit of information. Recall the equation for a high frequency carrier: $v_c(t)=V_c \cos(2\pi f_c t + \theta)$. As discussed in Lesson 18, we can use an information signal (message) to modulate a carrier by varying its amplitude, frequency, or phase. So, how do we go about representing digital information (1s and 0s) with modulation? Just as we can vary amplitude, frequency, and phase of a high-frequency carrier in accordance with an analog information (message) waveform, we can do the same with a digital waveform. Since bit values "shift" between 0s and 1s, digital modulation techniques that vary the carrier's amplitude, frequency, and phase are referred to as "**shift keying**."

**3.1 <u>Frequency Shift Keying (FSK)</u>** Frequency-shift keying (FSK) is a frequency modulation scheme in which digital information is transmitted through discrete frequency changes (shifts) of a carrier wave. The simplest form of FSK is Binary FSK (BFSK), in which a carrier's frequency is shifted to a low frequency or a high frequency to transmit 0s and 1s. The plot below shows a sample FSK signal along with the associated bits.

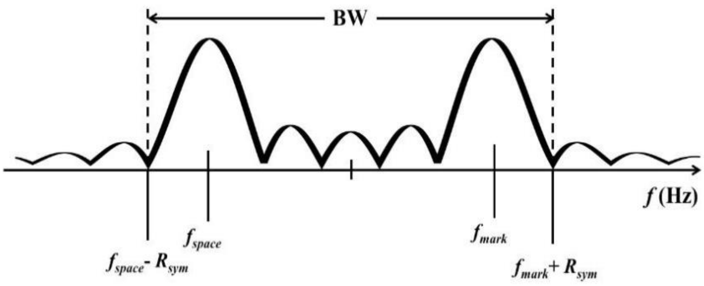



FSK was used "back in the day" with dial-up modems to connect your home computer to your Internet service provider over your analog phone. With a modem, a 0-bit was represented with a lower frequency carrier of 1070 Hz and a 1-bit was represented with a higher carrier frequency of 1270 Hz. The lower frequency, binary 0, was called the "space" frequency while the higher frequency, binary 1, was called the "mark" frequency. The terms mark/space were a throwback to the days of Morse code or flashing light communications.

In the frequency domain, we use two carrier frequencies and consider FSK to be two different digital transmissions, one at the mark frequency (the higher frequency) and one at the space frequency (lower frequency). The resulting FSK frequency plot would look like the following. This figure is two copies of the frequency plot on the previous page, one centered at $f_{mark}$ and one centered at $f_{space}$.




To determine the bandwidth for FSK modulation, we take a closer look at the frequency spectrum around the mark and space frequencies. We use the null-bandwidth definition to compute the bandwidth as shown below.



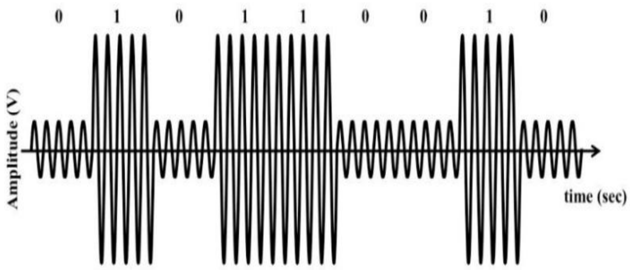

In the figure, the bandwidth effectively runs from the first null to the left of $f_{space}$ to the first null to the right of $f_{mark}$. Mathematically, we can compute the FSK bandwidth as:

$$BW = f_2 - f_1 = \left(f_{mark} + R_{sym}\right) - \left(f_{space} - R_{sym}\right) = f_{mark} - f_{space} + 2R_{sym}.$$

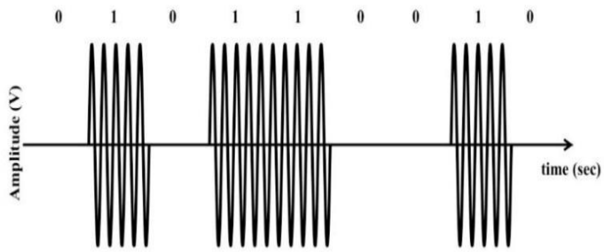
Or, for FSK, since the symbol rate is equal to the bit rate, we could also use the equation:

$$BW = f_2 - f_1 = \left(f_{mark} + R_b\right) - \left(f_{space} - R_b\right) = f_{mark} - f_{space} + 2R_b$$

**3.2 <u>Amplitude Shift Keying (ASK) and On-Off Keying (OOK)</u>** Amplitude Shift Keying is a form of amplitude modulation that represents digital data as shifts in the amplitude of a carrier wave: for example, small amplitude for a 0-bit, and larger amplitude for a 1-bit. We have seen what an ASK signal looks like in a previous chapter, repeated below.




The simplest digital modulation scheme is a form of ASK called *On-Off keying* (OOK). This is analogous to flashing light communication. In OOK, a carrier is transmitted for a 1-bit and nothing is transmitted for a 0-bit; this is the same as saying that the smaller ASK amplitude is 0.

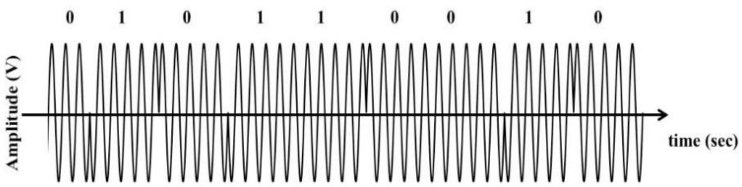



Note that in all forms of ASK, the frequency and phase of the carrier are the same for both symbols; it is the amplitude that changes.

**3.3 Binary Phase Shift Keying (BPSK)** Phase shift keying (PSK) is a form of phase modulation where the carrier's phase shifts to one of a finite set of possible phases based on the bits that are input. For binary phase shift keying (BPSK), the carrier phase is shifted between one of only two phases (typically 0° and 180°) depending on whether a 0-bit or a 1-bit is being transmitted. For example:

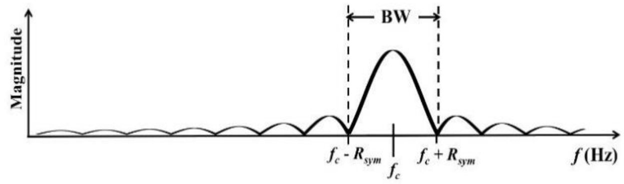
0-bit: the symbol transmitted is $V_c \cos(2\pi f_c t)$.
1-bit: the symbol transmitted is $V_c \cos(2\pi f_c t + 180°) = -V_c \cos(2\pi f_c t)$

Using $\pm\cos(2\pi f_c t)$ as the symbols is not the only type of BPSK modulation there is, as long as the phases are 180° apart…for example, you could also use $\pm \sin(2\pi f_c t)$. A sample BPSK transmission is shown in the following figure. As you look at the figure, you'll notice that for us, it is probably harder to see the transitions in phase for 0-bits and 1-bits, but rest assured that a BPSK receiver can do it easily.

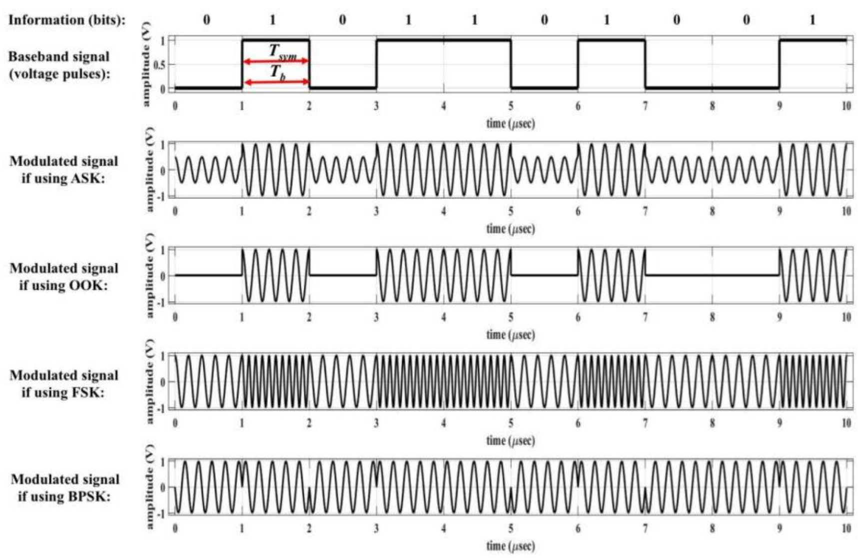



It is important to point out that in PSK, the amplitude of all output symbols is the same; it is the phase of the output symbols that are different, while in ASK, both symbols have the same phase, but different amplitudes.

The bandwidth associated with ASK, OOK, or BPSK is what we have seen before, $BW = 2R_{sym}$, as shown in the figure below. In the case of ASK, OOK, or BPSK, since $N = 1$ bits/symbol, $BW = 2R_{sym} = 2R_b$. For example, for ASK, OOK, or BPSK, if the symbol rate is 600,000 symbols/sec, the bitrate is 600 kbps, and so the bandwidth is 2(600,000) = 1.2 MHz.




Let us now summarize the types of binary digital modulation that have been introduced so far. On a wire, the symbols take the form of voltage pulses, which are, for example, a high pulse for a 1-bit and a low (or no) pulse for a 0-bit. In FSK, ASK, OOK or BPSK, the symbols take the form of a high frequency carrier that has its frequency or amplitude or phase altered based on whether a 0-bit or a 1-bit is being transmitted. In binary modulation, the number of symbols that can be transmitted ($M$) is two ($M = 2$) and each symbol represents one bit of data ($N = 1$ bit per symbol). For binary modulation, the time duration of a bit is the same as the time duration of a symbol ($T_b = T_{sym}$). The following figure depicts the relationship between bits and symbols for voltage pulses (the baseband signal), and FSK, ASK, OOK and BPSK (the modulated signals).

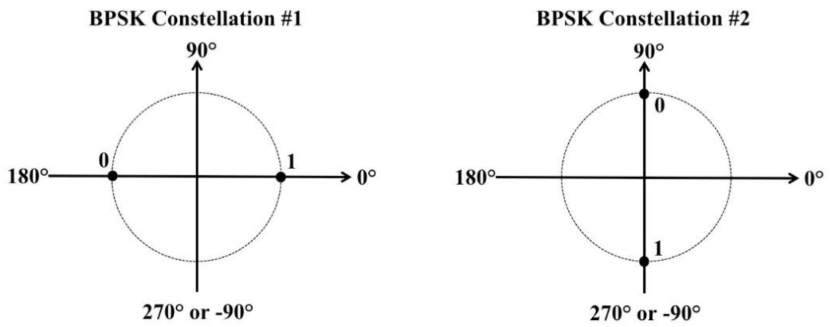


Up to this point we have discussed binary digital modulation ($M$=2) with one bit per symbol ($N$=1), which means that at any time, only one of two possible symbols would be transmitted. But it is possible to have a modulation scheme with more than two symbols. This means that with each symbol, more than one bit is transmitted at a time. These types of modulation are referred to as $M$-ary digital modulation.

### 4. *M*-ary Digital Modulation

Before launching into more complicated digital modulation, we'll introduce a graphical way to relate what the output symbols are, and the bits that each symbol represents. This is called a *constellation diagram*. A constellation diagram is a plot of the phase and relative amplitude of the output symbols for a digital modulation system, in polar coordinates. In terms of the symbol's phase, 0° is along the positive x-axis, and phase increases as you move counterclockwise around the x-y plane. The symbol's relative amplitude is measured as distance from the origin of the plot. The possible output symbols are represented with large dots, and adjacent to them are the bits they represent. Symbols that have the same amplitude are the same distance from the origin (you can think of them as laying on the same circle around the origin). All symbols with the same phase would fall on the same line segment that originates at the origin and goes out at a certain angle.

For example, here are two possible BPSK systems' constellation diagrams. In BPSK, the output symbols both have the same amplitude (both of the symbols are equidistant from the origin), but their phases are 180° apart. There are other possible combinations of two carrier phases that might be used (such as +90° and -90°), but the actual constellation used is not important, as long as the transmitter and receiver use the same constellation.

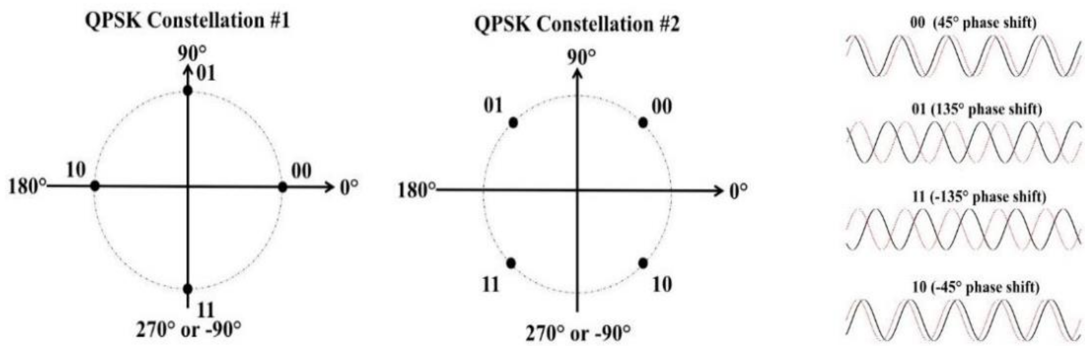



Note that BPSK transmits 1 bit per symbol, so only one bit value is placed next to each symbol.

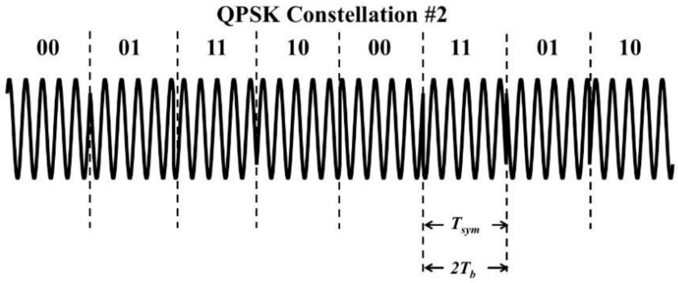
If it is desired to get the information from the transmitter to the receiver faster, we could increase the number of bits per second (bps) that are transmitted. The cost of increasing the bitrate (besides requiring more complex components) is that it increases the transmission bandwidth: recall that for ASK, OOK or BPSK, $BW = 2R_{sym} = 2R_b$, and from Chapter 19, that bandwidth can be expensive! Is there a way to transmit a higher bitrate but using a smaller transmission bandwidth? The answer is yes, using $M$-ary digital modulation.

In $M$-ary modulation, we can preserve bandwidth if we **keep the symbol rate the same** and **increase the number of bits per symbol**. For example, instead of transmitting just 2 possible phase shifts (0˚and 180˚), we could transmit one of 4 possible phase shifts per symbol. This is called quadrature phase shift keying (QPSK).

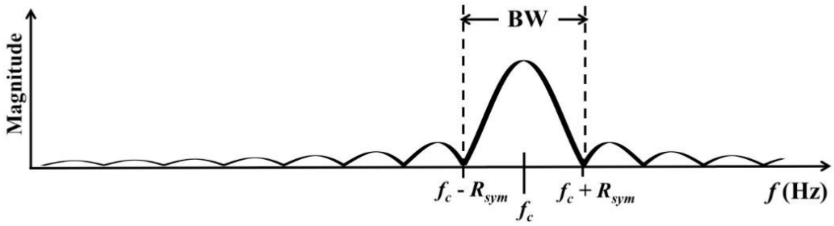
**4.1 <u>Quadrature Phase Shift Keying (QPSK)</u>** In QSPK, there are **4 symbols** ($M$ = 4) and there are **2 bits per symbol** ($N = 2 = \log_2 M$). Two of the possible constellation diagrams for QPSK are shown in the following figure[1], and the four symbols from QPSK Constellation #2 are shown to the right of this constellation. The carrier with a phase of 0˚ is plotted in a dashed line with each symbol for reference. The four symbols in the right-hand constellation are:
$V_c \cos(2\pi f_c t + 45°)$, $V_c \cos(2\pi f_c t + 135°)$, $V_c \cos(2\pi f_c t - 135°)$ and $V_c \cos(2\pi f_c t - 45°)$.



The following figure is a plot of the use of QPSK constellation #2 in the previous figure to transmit the bit stream 0001111000110110. Also shown is the bit duration, and the symbol duration for QPSK.




The frequency spectrum for *M*-ary modulation schemes is shown in the figure below, which is the same one that appeared earlier in the chapter for ASK, OOK and BPSK. But if the bit rate is constant, the benefit of transmitting more than one bit in a symbol can be seen in the fact that the nulls are closer to the carrier frequency, since $R_{sym} = R_b / N$.




From the figure, it is seen that the bandwidth for QPSK ($N = 2$) is given by

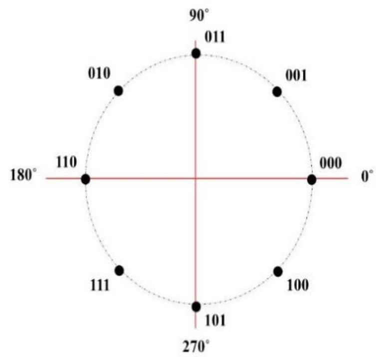
$$BW = 2R_{sym} = \frac{2R_b}{N} = 2\left(\frac{R_b}{2}\right) = R_b \text{ Hz.}$$

For example, if bitrate is 600 kbps, $BW = 2(600{,}000)/2 = 600$ kHz, half that of ASK, OOK or BPSK!

[1] The points in the picture for QPSK Constellation #2 are labeled using *gray code* where only one bit changes between adjacent coordinates. Gray code is used to minimize the number of bits that could be received in error.

### 4.2 *M*-ary PSK

We can further increase the number of bits per symbol by increasing the number of possible phase shifts. The *M* in *M*-ary refers to the number of symbols. Consider the 8-PSK constellation to the right (one of many possible 8-PSK constellations[2]).

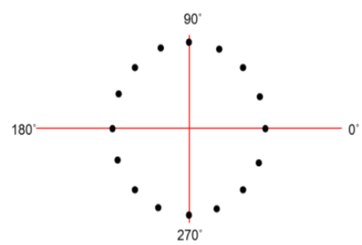



How many bits per symbol are transmitted? There are 8 symbols ($M = 8$), so $N = \log_2 M = \log_2 8 = 3$ bits/symbol. This is also evident from the diagram because the three bits associated with each symbol appear next to the symbol.

What is the bandwidth for 8-PSK? Since $N$=3 bits/symbol, bandwidth is given by

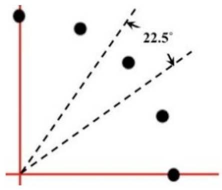
$$BW = 2R_{sym} = \frac{2R_b}{N} = \frac{2R_b}{3}.$$

For example, if the bitrate is 600 kbps, bandwidth for 8-PSK is $BW = 2(600{,}000)/3 = 400$ kHz, even less than for QPSK!



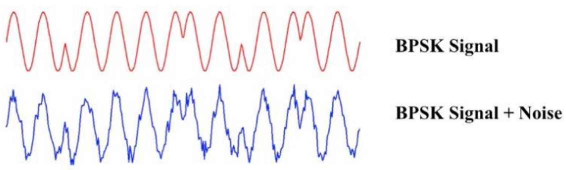

We could further increase to 4 bits/symbol using 16-PSK. Here, $M = 16$ and $N = 4$ bits/symbol. A 16-PSK constellation is shown to the right, where each phase is separated by 360°/16 = 22.5°. More complex *M*-ary PSK modulation is possible: 32-PSK, 64-PSK, etc., but it becomes more susceptible to noise as the symbols get closer together. As a reminder, for PSK, all of the symbols have the same carrier frequency and amplitude; it is their phase that is different. For that reason, on a PSK constellation diagram, all of the symbols appear on a circle about the origin.

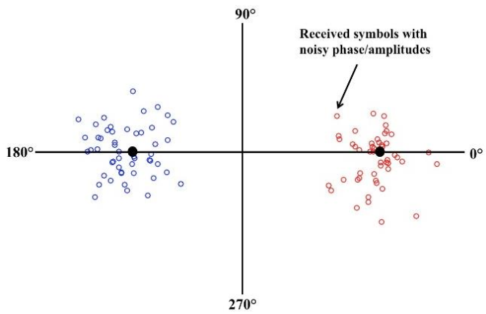
To demodulate any type of PSK, a receiver must determine the phase of the received symbol. For 16-PSK, the receiver must determine the phase within ±11.25°, since the phases are separated by 22.5°. A portion of the constellation diagram for 16-PSK is shown to the right, indicating the wedge of phase values that separates one of the symbols from the adjacent symbols.




**Noise Effects**

Recall that the number one most limiting factor in communication systems is noise. In <u>all</u> transmissions, the received signal will be degraded by noise. The following figure shows a BPSK signal and the same signal corrupted by noise. You might imagine that it is harder for a receiver to determine the correct phase (correct symbol) that was transmitted for the noisy signal.

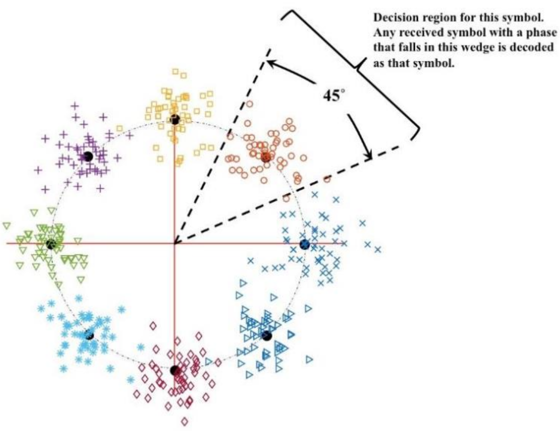



This noise corruption can be depicted in the constellation diagram to the right, where the two transmitted BPSK symbols are indicated in the two large black circles (phase = 0° and phase = 180°), and some noisy received symbols are the smaller red and blue circles.

A BPSK receiver must make a decision to determine the phase of a received signal to determine the corresponding bit. If the noise is severe enough, a receiver might make a mistake and decide that it had received a 0-bit when it actually received a 1-bit. These are called *bit errors*. Now, consider the same noise in the presence of an 8-PSK signal. Is it easier for the receiver to make bit errors?

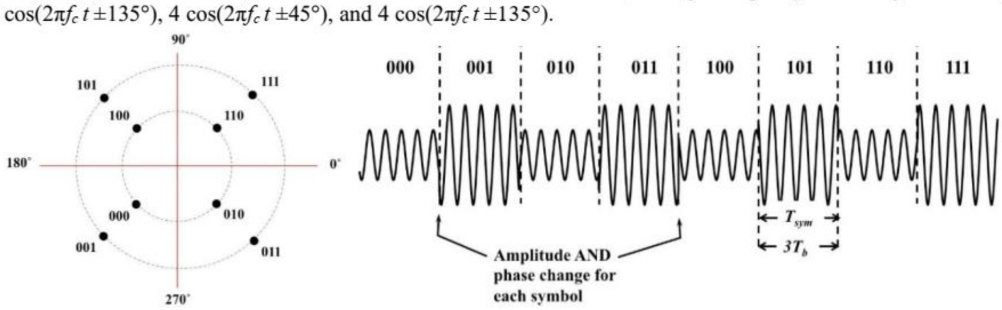
The answer is yes, since more phases are used in PSK, making the symbols closer together makes it easier for the receiver to make bit errors (see the figure to the right). But, of course, the advantage of more symbols is a narrower bandwidth, if the bit rate is held constant. There is a way to use more symbols in modulation while reducing the chances of making bit errors; by using symbols that have different amplitudes AND phases so they are more spread out.




**4.3 <u>Quadrature Amplitude Modulation (QAM)</u>**

In order to increase the distance between symbols in a constellation, another option is to modulate both the amplitude and the phase of the carrier. This is called *Quadrature Amplitude Modulation* (QAM).

**4.3.1 <u>8-QAM</u>** An 8-QAM constellation is shown below (one of many possible 8-QAM constellations). The eight symbols along with the 3-bit digital words corresponding to each are shown to the right of the constellation. This system uses 2 possible amplitudes and 4 possible phases. In 8-QAM, the duration of a symbol is three times the duration of a bit (since each symbol carries 3 bits). Note that there are both phase <u>and/or</u> amplitude changes for <u>each</u> symbol. For the system with the constellation shown below, the eight output symbols might be 2 cos($2\pi f_c t$ ±45°), 2 cos($2\pi f_c t$ ±135°), 4 cos($2\pi f_c t$ ±45°), and 4 cos($2\pi f_c t$ ±135°).



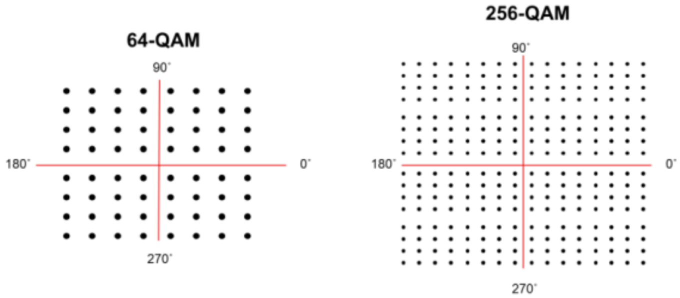

What is the bandwidth for 8-QAM? The same as for 8-PSK, since the bandwidth for all digital modulation types (except for FSK) is given by

$$BW = 2R_{sym} = \frac{2R_b}{N} = \frac{2R_b}{3}$$

And it doesn't stop there.

[2] The points in this picture are labeled using *gray code* where only one bit changes between adjacent symbols.

**4.3.2 <u>Higher level QAM signals</u>** QAM signaling can be extended to have a larger number of symbols, which then allows a much higher bit rate in the same bandwidth (because there are more bits per symbol). 64-QAM and 256-QAM are common in cable modems, satellites, and high-speed fixed broadband wireless. Some possible constellations are in the following figure.




In 256-QAM, you find that for each symbol you are transmitting (there are 256 symbols), there are 8 bits of information. Assuming the symbol rate remains constant, then for the same bandwidth you are sending 8 times more information when you use 256-QAM than when you use OOK, ASK or BPSK. For 256-QAM, if the bitrate is 600 kbps, the bandwidth is 2(600,000)/8 = 150 kHz.

Now that's powerful!

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