UNIT 3 DATA ENCODING AND MULTIPLEXING

Structure **Page Nos.**

3.0 INTRODUCTION

You might be aware from the description in the previous blocks, how data is transmitted through a channel (wired or wireless) in the form of an analog or digital signal. In this unit, we will elaborate on the techniques to produce these types of signals from analog or digital data. We will look at, in brief, on the following encoding techniques [*Figure1*]:

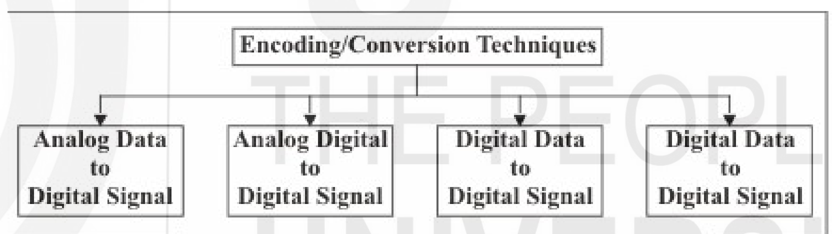


Figure 1: Encoding/Conversion Techniques

- Analog data as an analog signal
- Analog data as a digital signal
- Digital data as an analog signal
- Digital data as a digital signal

We will confine ourselves to, a treatment of the most useful and common techniques only, and will skip over the many other kinds of encoding of data that can be performed.

3.1 OBJECTIVES

After going through this unit, you should be able to describe:

- what encoding means;
- why encoding is needed;
- some different types of encoding;
- mechanisms and techniques used for encoding, and
- when each type of encoding is used with examples.

There are several other types of encoding as well as encoding techniques that will not be discussed in this unit.

3.2 ENCODING

Usually, we cannot send a signal containing information directly over a transmission medium, at least not for long distances. For example, the sound of our voice can only travel a few hundred meters. If we want to send our voice to the next city, we have to be able to transform our voice, to, an electrical signal that can then, be sent that long distance. Then, we would also need to perform the reverse conversion at the other end.

So, we see that, to send a signal over a physical medium, we need to encode or transform the signal in some way so that the transmission medium can transmit it. The sender and the receiver must agree on what kind of transformation has been done so that, it can be reversed at the other end and the actual signal or information can be recovered. The information content in a signal is based upon having some changes in it. For example, if we just send a pure sine wave, where the voltage varies with time as a sine function, then, we cannot convey much information apart from the fact that there is some entity at the other end that is doing the transmitting.

The information content of a signal is dependent on the variation in the signal. After sending one cycle of a sine wave there is no point in sending another cycle because we will not be able to convey anything more than what we have already done. What is needed is to modify the sine wave in some way to carry the information we want to convey. This is called modulation.

You know that there are two kinds of signals : analog and digital. Likewise, there are two types of information that we may want to transmit, again analog and digital. So, we get four basic combinations of encoding that we may need to perform, depending on the signal type as well as the information type. We shall look at each of these four types briefly in this unit and will also study the ways in which the actual encoding can be done.

3.3 ANALOG-TO-ANALOG MODULATION

Let us first look at a situation where our signal and information are both of the analog type. The act of changing or encoding the information in the signal is known as **modulation.** A good example of such encoding is radio broadcasting. Here we primarily want to send sound in some form over the atmosphere to the receiving radio sets. The sound is converted into an analog electrical signal at the source and is used to encode the signal which is the base frequency at which the transmission is being done. The reverse process is performed at the radio set to recover the information in electrical form, which is then converted back to sound so that, we hear what was being said at the radio station.

Let us, formulate another definition of modulation. When a low frequency information signal is encoded over a higher frequency signal, it is called modulation [Ref.3]. The encoding can be done by varying amplitude (strength of a signal), period (amount of time, in seconds, a signal needs to complete a cycle) and frequency (number of cycles per second) and phase (position of waveform relative to zero).

Notice the simulations of the word modulation to the word modifying. For instance, an audio signal (one that is audible to human ear) can be used to modify an RF (Radio Frequency) carrier, when the amplitude of the RF is varied according to the changes inthe amplitude of the audio, it is called amplitude modulation (AM), and when the frequency is varied, it is called FM (frequency modulation).

Although, it can be dangerous to use analogies as they can be carried too far, we can understand this process with an example. Suppose we want to send a piece of paper with something written on it to a person a hundred meters away. We will not be ableto throw the paper that far. So, we can wrap it around a small stone and then throw the stone towards the person. He can then unwrap the paper and read the message.
Here, the stone is the signal and the piece of paper is the information that is used to change the stone. Just as sending the stone alone would not have conveyed much to our friend, similarly sending only the base signal (also called carrier signal) will notconvey much information to the recipient.

Therefore, there are three different ways in which this encoding of the analog signalwith analog information is performed. These methods are:

- **Amplitude modulation** (AM), where the amplitude of the signal is changeddepending on the information to be sent (*Figure 2 (a)*).

- **Frequency modulation**, where the frequency of the signal is changeddepending on the information to be sent (*Figure 2 (b))*.

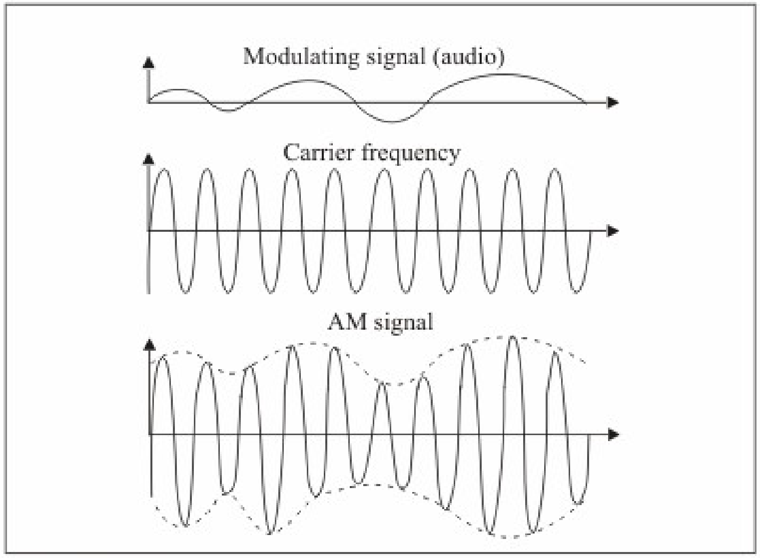
- **Phase modulation,** where it is the phase of the signal that is changed accordingto the information to be sent.

Amplitude Modulation

Now, let us go into the details. In this type of modulation the frequency and phase ofthe carrier or base signal are not altered. Only the amplitude changes and we can seethat the information is contained in the envelope of the carrier signal. It can be proud/demonstrated that, the bandwidth of the composite signal is twice that of the highest frequency in the information signal that modulates the carrier.

By international agreement, for radio transmission using AM, 9 KHz is allowed as thebandwidth. So, if a radio station transmits at 618 KHz, the next station can only transmit at 609 or at 627 KHz. It will be clear that the highest audio frequency that can be carried over AM radio is thus 4.5 KHz, which is sufficient for most voice and musical pieces. Actually, the modulating signal is centred around the carrier frequency and extends the composite signal both ways in equal measure. Each of these is called a sideband and therefore, AM radio transmission is dual sideband. This is actually wasteful because the spectrum is unnecessarily used up, but the cost and complexity associated with eliminating one sideband, and performing single sideband AM modulation, have led to the technique not being used widely.

AM radio transmission has been assigned the frequency range of 530 to 1700 KHz. The quality of AM transmission is not very good as noise in the channel (the atmosphere, here) can easily creep into the signal and alter its amplitude. So, the receiving end will likely find a fair amount of noise in the signal, particularly at higher, short wave frequencies. That is why you often get a lot of static in short waveor even medium wave broadcasts.



AM Bandwidth

Figure 2 (a): Amplitude modulation

Frequency Modulation

In contrast to Amplitude Modulation, here it is the frequency of the base signal that is altered depending on the information that is to be sent. The amplitude and phase of the carrier signal are not changed at all. It can be shown that this results in a bandwidth requirement of 10 times the modulating signal, centred on the carrier frequency.

This method is the less susceptible to noise and gives the best performance of all data encoding types as far as the quality of the transmitted signal is concerned. Although digital encoding methods may give better performance over multiple hops (because in their case, the original signal can be accurately reconstructed at each hop), FrequencyModulation (FM) is the best as far as single hop transmission goes.

FM radio transmission has been allocated the spectrum range 88 MHz to 108 MHz. As a good stereo sound signal needs 15 KHz of bandwidth, this means that FM transmission has a bandwidth of 150 KHz. To be safe from interference from neighbouring stations, 200 KHz is the minimum separation needed. As a further safety measure, in any given area, only alternate stations are allowed and the neighbouring frequencies are kept empty. This is practical because FM transmission is line of sight (because of the carrier frequency) and so, beyond a range of about 100Km or so, the signal cannot be received directly (this distance depends on various factors). We can therefore, have at most 50 FM stereo radio stations in a given geographical area.

As in the case of AM transmission (having 2 sub bands), FM also has multiple sidebands maybe two or more, which is really wasteful but is tolerated in the interestsof simplicity and cost of the equipment.

In television transmission the bandwidth needed is 4.5 MHz in each sideband. The video signal is sent using amplitude modulation. The audio signal goes at a frequency that is 5.5 MHz over the video signal and is sent as a frequency modulated signal. To reduce the bandwidth requirement, the video signal is sent using a method called vestigial sideband that needs only 6 MHz, instead of the 9 MHz that would have been needed for dual sideband transmission, though, it is more than the 4.5 MHz that a puresingle sideband transmission would have needed.

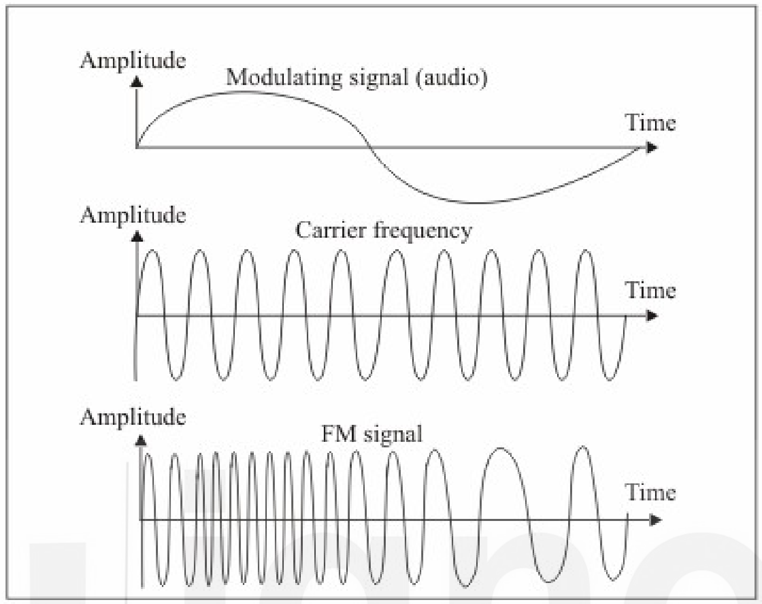


Figure 2 (b): Frequency modulation

Phase Modulation

As you would have guessed by now, in Phase Modulation (PM) the modulating signalleaves the frequency and amplitude of the carrier signal unchanged but alters its phase. The characteristics of this encoding technique are similar to FM, but the advantage is that of reduced complexity of equipment. However, this method is not used in commercial broadcasting.

☞ Check Your Progress 1

1) Why do we need modulation? Would it be right to simply send the informationas the signal itself ?

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2) Why is it that there are at most 50 FM radio stations in a particular geographicalarea? Justify with calculations.

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3) Why is AM the most susceptible to noise, of the three types of modulation?

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3.4 ANALOG TO DIGITAL MODULATION

Natural phenomena are analog in nature and can take on any of the potentially infinite number of values. For example, the actual frequencies contained in a sound made by a human is an analog signal, as is the amplitude or loudness of the sound. One example of coding analog data in digital form is when, we want to record sound on digital media such as a DVD or in other forms such as MP3 or as a ".wav" file. Here, the analog signal, that is, the musical composition or the human voice is encoded in digital form. This is an example of analog to digital encoding.

While, in the case of analog to analog encoding, the motivation was to be able to transmit the signal for long distances, here, the main reason is to be able to change the information itself to digital form. That digital signal can then be transmitted, if necessary, by any suitable method.

One method of encoding is what is called Pulse Code Modulation (PCM). This gives very good quality and when used in conjunction with error correction techniques, can be regenerated at every hop on the way to its final destination. This is the big advantage of digital signals. At every hop, errors can be removed and the original signal reconstructed almost exactly, barring the few errors that could not be corrected.

The first step in PCM is, to convert the analog signal into a series of pulses (*Figure 3*). This is called Pulse Amplitude Modulation (PAM). To do this the analog signal is sampled at fixed intervals and the amplitude at each sample decides the amplitude of that pulse. You can see that at this step, the resulting signal is still actually an analog signal because the pulse can have any amplitude, equal to the amplitude of the original signal at the time the sample was taken. In PAM, the sampled value is held for a small time so that the pulse has a finite width. In the original signal the value occurs only for the instant at which it was sampled.

One question that arises here is, how many samples do we have to take? We would not want to take too many samples as that would be wasteful. At the same time, if, we take too few samples, we may not be able to reconstruct the original signal properly. The answer comes from Nyqist's theorem, which states that, to be able to get back the original signal, we must sample at a rate, that is, at least twice that of the highest frequency contained in the original signal.

Let us do some calculations to see what this means in practice. In the next unit, you will see that a voice conversation over a telephone has a range of frequencies from 300 Hz to 3300 Hz. Because the highest frequency is 3300 Hz, we need at least 6600 samples a second to digitise the voice to be sent over a phone line. For safety we take 8000 samples per second, corresponding to a sampling interval of 125 microseconds.

The next stage in the digitization of the signal is quantisation. In this process, the analog values of the sampled pulses are quantised, that is, converted into discrete values. For example, if the original analog signal had an amplitude ranging from +5v to –5v, the result of PAM might be to produce pulses of the following amplitudes (each sample is taken at say, intervals of 125 microseconds)

+4.32, +1.78, -3.19, -4.07, -2.56, +0.08 and so on.

In quantisation, we may decide to give the amplitude 256 discrete values. So, they will range from –127 to +128. The actual amplitude of +5v to –5v has to be represented in this range, which means each value is of 39 mV nearly. By this token, the first value of +4.32v lies between 110 and 111, and we can decide that such a value will be taken as 110. This is quantisation. The analog values now become thefollowing quantised values:

110, 45, -81, -104, -65, 2 and so on.

The above discrete values are now represented as 8 binary digits with, 1 bit giving the sign while the other 7 bits represent the value of the sample.

In the final stage of the encoding, the binary digits are then transformed into a digital signal using any of the digital to digital encoding mechanisms discussed later in this unit. This digital signal is now the representation of the original analog signal.

PCM is commonly used to transmit voice signals over a telephone line. It gives goodvoice quality but, is a fairly complex method of encoding the signal. There is a simpler method that can be used, but we have to pay the price in terms of the lower quality of the encoded signal. This is called Delta modulation. Here, samples are taken as described above, but instead of quantising them into 256 or more levels, onlythe direction of change is retained. If the sample is larger in value than the previous one, it is considered a 1 while otherwise it is considered a 0 bit. So the sequence above would be encoded as:

1, 0, 0, 0, 1, 1 and so on.

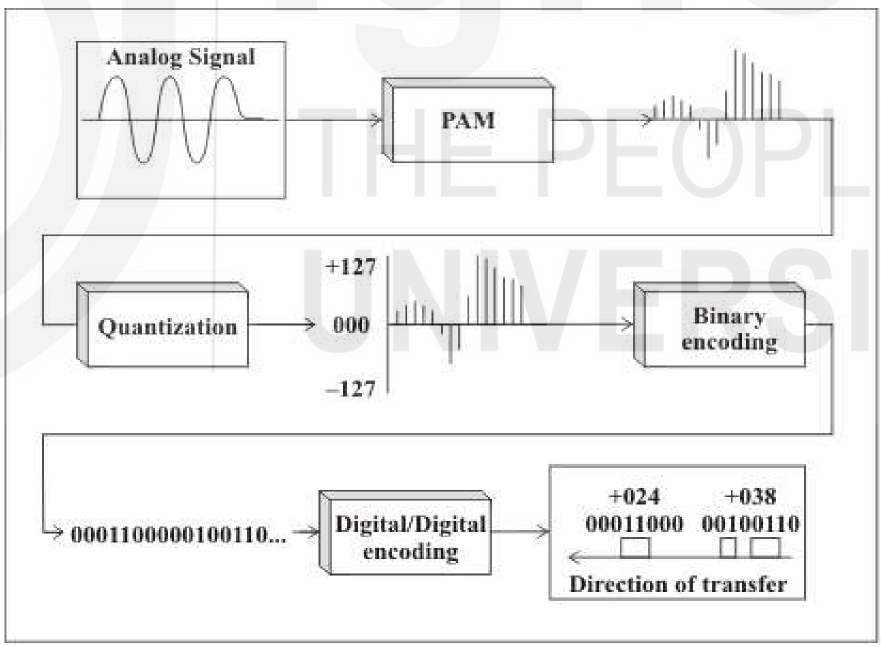

Figure 3: Pulse code modulation

☞ Check Your Progress 2

1) Why is PAM a necessary pre-requisite to PCM? Why can we not use PCMdirectly ?

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2) What would be the minimum sampling interval needed for reconstructing a signal where the highest frequency is 1 KHz ?

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3) Consider a signal where the amplitude varies from +6.4v to –6.4v. If we want to quantise it into 64 levels, what would be the quantised values correspondingto signals of –3.6v and +0.88v ?

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3.5 DIGITAL TO ANALOG MODULATION

So far, we have looked at the codification of analog data, where the source information could take up any value. We also have need for the reverse type of encoding, where binary digital values have to be encoded into an analog signal. This may seem simple because after all, we only have to give two distinct values to the analog signal, but here you need to realise that, we have to be able to distinguish between successive digital values that are the same, say three 1's in succession or two0's in succession. These should not get interpreted as a single 1 or a single 0! This sort of problem did not exist in the case of analog to analog encoding.

One example where such encoding is needed is, when we need to transmit between computers over a telephone line. The computers generate digital data, but the telephone system is analog. So, this situation is a bit different from the transmissionof voice over the telephone, where the source signal is also analog.

In this kind of encoding, one important factor of interest is the rate at which we can transfer bits. Clearly, there will be some kind of limit to the amount of information that we can send in a given time. This depends on the available bandwidth of the carrier signal, that is, the variation that we can permit around the nominal frequencyas well as on the specific technique of encoding that we use.

Here it is important to understand two aspects of the rate of transfer of information. The first is the bit rate, that tells us the number of individual bits that are transmitted per second. The second, and more important from the transmission angle, is the baudrate. This is the number of signal transitions per second that are needed to represent those bits. For a binary system, the two are equal. In general, the bit rate cannot be lower than the baud rate, as there has to be at least one signal transition to represent abit.

We have seen in analog to analog encoding that, the source signal modulates the baseor carrier signal to convey the information that we want to transmit. A similar principle is also used in digital to analog encoding. The carrier signal has to be modulated, but this time by the digital signal, to produce the composite signal that is transmitted. However, while in the analog case there were a potentially infinite number of values that could have been transmitted, here we have only two values (a 0 or a 1) that will modulate the signal.

If, there is no modulation, the carrier signal will be a pure sine wave at the frequency of transmission. You will realise that the baud rate that can be supported also depends on the quality of the line and the amount of random noise that is present in the medium. Like in analog-to-analog encoding, there are three properties of the sine wave that we can alter to convey information. These are the amplitude, the frequencyand the phase. In addition, we can also use a combination of amplitude and phase change to encode information more efficiently. Modulation is also referred to as shift keying in this type of encoding technique.

There are three types of digital to analog modulation (*Figure 4*)

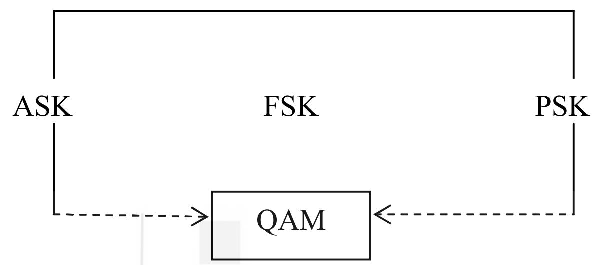


Figure 4: Digital to analog modulation techniques

Now we will take up each one individually.

Amplitude Shift Keying (ASK)

In this kind of encoding, abbreviated as ASK, the amplitude of the analog signal is changed to represent the bits, while the phase and the frequency are kept the same. The bigger the difference between the two values, the lower the possibility of noise causing errors. There is no restriction, however, on what voltage should represent a 1or a 0, and this is left to the designers. For the duration of a bit, the voltage should remain constant.

One method could be to represent the high voltage as a 1 and the negative high voltage as a 0, although the reverse convention could also be used. Sometimes, to reduce the energy needed for transmission, we can choose to represent a 0 or a 1 by azero voltage, and the other bit by the high voltage. This scheme is called on-off-keying (OOK).

Like amplitude modulation in the analog case, ASK is vulnerable to noise. Amplitudechanges easily as a result of the noise in the line, and if, the quantum of noise is great enough, it can result in an error. This would mean that a 0 gets interpreted as a 1 at the receiving end, or vice versa. Noise can occur as spikes or as a constant voltage that shifts the amplitude of the received signal. Even thermal noise, that is fairly constant, can cause a 0 to be interpreted as a 1 or the other way round.

To find the bandwidth requirement of an ASK encoded signal on an ideal, noiseless, line, we have to do a Fourier analysis of the composite signal. This yields harmonics on either side of the carrier frequency, with frequencies that are shifted by an odd multiple of half the baud rate. As most of the energy is concentrated in the first harmonic, we can ignore the others. It can then be shown that the bandwidth requiredfor ASK encoding is equal to the baud rate of the signal. So if we want to transfer data at 256 Kbps, we need a bandwidth of at least 256 KHz.

In a practical case, there will be noise in the line and this ideal cannot be achieved. This increases the bandwidth requirement by a factor that depends on the line qualityand the carrier amplitude. ASK is a simple encoding method but is not commonly used by itself because of efficiency and quality issues.

Frequency Shift Keying (FSK)

Just as in the case of analog to analog encoding, we can encode a digital signal by changing the frequency of the analog carrier. This is frequency shift keying (FSK). When a bit is encoded, the frequency changes and remains constant for a particular duration. The phase and the amplitude of the carrier are unaltered by FSK. Just as inFM, noise in the line has little effect on the frequency and so, this method is less susceptible to noise.

It is possible to show that an FSK signal can be analysed as the sum of two ASK signals. One of these is centred around the frequency used for a 0 bit and the other for the frequency used for a 1 bit. From this, it is easy to show that the bandwidth needed for an FSK signal is equal to the sum of the baud rate and the difference between the two frequencies. Thus, the requirement of FSK is a higher bandwidth that helps us to avoid the noise problems of ASK.

FSK is not much used in practice because of the need for higher bandwidth and thecomparative complex requirement for changing the frequency.

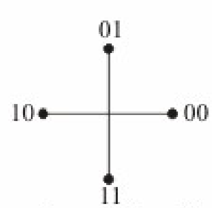
Phase Shift Keying (PSK)

We can also encode by varying the phase of the carrier signal. This is called phase shift keying (PSK) and here, the frequency and amplitude of the carrier are not altered. To send a 1 we could use a phase of 0 while we could change it to 180 degrees to represent a 0. Such an arrangement is not affected by the noise in the line,because that affects amplitude rather than the phase of the signal.

This quality of PSK can be used to achieve more efficient encoding. For example, instead of having only 2 phases, we could have four phases, 0, 90 degrees, 180 degrees and 270 degrees. Each of these phase shifts could represent 2 bits at one go, say, the combinations 00, 01, 10 and 11 respectively. Such a scheme is called 4-PSK. The concept can be extended to higher levels and we could have 8-PSK to send groups of 3 bits in one go. This method can be extended further, but at some point the sensitivity of the communication equipment will not be enough to detect the small phase changes and we are then limited for that reason.

Dibit	Phase
00	0
01	90
10	180
11	270

Dibit
(2 bits)



Constellation diagram

Figure 5(a) : 4 PSK

Dibit	Phase
000	0
001	45
010	90
011	135
100	180
101	225
110	270
111	315

Tribits
(3 bits)

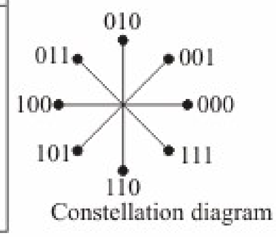


Figure 5(b): 8 PSK

Clearly, the scheme is more efficient than ASK because, we can now achieve a higherbit rate from the same bandwidth. It can be shown that the bandwidth required in 2- PSK is the same as that needed in ASK. But the same bandwidth at 8-PSK can transmit thrice the number of bits at the same baud rate.

Why can we not apply the same trick in ASK? We could have 4, 8 or more amplitude levels to transmit 2, 3 or more bits in one signal transition. The reason is that, ASK is vulnerable to noise and that makes it unsuitable for using many different amplitude levels for encoding. Similarly, FSK has higher bandwidth requirements and so we do not use this kind of technique there as, there is not much to be gained.

Quadrature Amplitude Modulation (QAM)

To further improve the efficiency of the transmission, we can consider combining different kinds of encoding. While, at least one them has to be held constant, there is no reason to alter only one of them. Because of its bandwidth requirements, we would not bother to combine FSK with any other method. But it can be quite fruitful to combine ASK and PSK together and reap the advantages. This technique is called Quadrature Amplitude Modulation (QAM).

We have already seen how ASK is vulnerable to noise. That is why the number of different amplitude levels is small while there may be more phase shift levels possible. There are a large number of schemes possible in the QAM method. For example 1 amplitude and 4 phases is called 4-QAM. It is the same as 4-PSK because there is no change in the amplitude. With 2 amplitudes and 4 phases we can send 3 bits at a time and so this scheme is called 8-QAM (*Figure 6*).

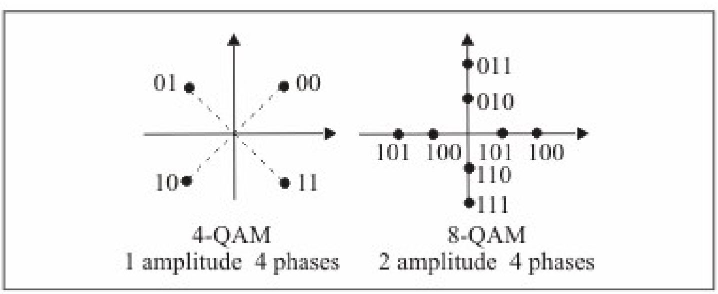


Figure 6: 4-QAM and 8-QAM constellations

In practical application we use 3 amplitudes and 12 phases or 4 amplitudes and 8 phases. Although, this way, we should be able to send 4 bits at a time (this is 32-QAM), to guard against errors, adjacent combinations are left unused. So, there is a smaller possibility of noise affecting the amplitude and causing problems with the signal received. As a further safeguard, in some schemes, certain amplitudes and phases are coupled, so that, some amount of error correction is possible at the physicallayer itself.

Note, that the errors we have talked of here are at the physical layer. By using the appropriate techniques, at the data link layer we achieve error free transmission irrespective of the underlying mechanisms at the physical layer. That is not something that will be looked at in this unit.

It can be shown that QAM requires the same bandwidth as ASK or PSK. So, we are able to achieve a much higher bit rate using the same baud rate. That is why QAM is the method of encoding used currently in data communication applications.

☞ **Check Your Progress 3**

1) What is the difference between the bit rate and the baud rate? Explain with an example.

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2) If our equipment is able to reliably detect differences of voltage of up to 0.5 v, and the probability is 99.74% that noise will cause a change in the amplitude ofour signal of no more than 0.2 v, then how many amplitude levels can one haveif the peak signal strength is to vary from between +3v to –3v ?

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3) If our equipment can reliably detect phase changes of up to 20^{O}, how many phase levels can we use in a PSK system ?

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4) Given the characteristics (of questions 2 and 3 above) and using only half the possible combinations for safety, what level of QAM can be achieved ?

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3.6 DIGITAL TO DIGITAL ENCODING

Let us, now look at the last combination of encoding possible, from digital to digital signals. A simple example is when sending data from a computer to a printer, both of which understand digital signals. The transmission has to be for short distances over the printer cable and occurs as a series of digital pulses. Another example of such encoding is for transmission over local area networks such as an Ethernet, where the communication is between computer and computer.

This kind of encoding is really of three types and we will look at each of the types here. Again, there can be many other mechanisms other than the ones discussed in this unit, but we will look at only the methods that are used in data communication applications. These types are unipolar, polar and bipolar techniques.

Unipolar Encoding

This is a simple mechanism but is rarely used in practice because of the inherent problems of the method. As the name implies, there is only one polarity used in the transmission of the pulses. A positive pulse may be taken as a 1 and a zero voltage can be taken as 0, or the other way round depending on the convention we are using (*Figure 7*). Unipolar encoding uses less energy and also has a low cost.

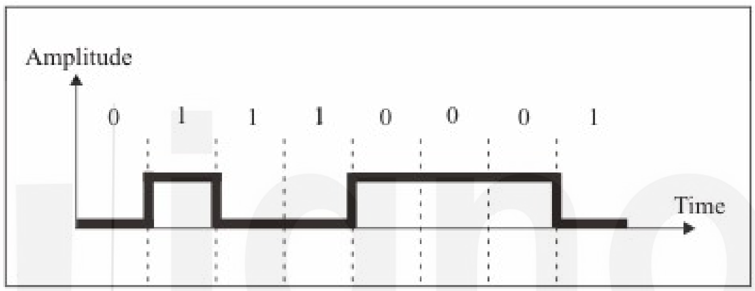

Figure 7: Unipolar encoding

What then are the problems that preclude widespread use of this technique? The two problems are those of synchronisation and of the direct current (DC) component in thesignal.

In any digital communication system the transmitter and receiver have to be synchronised or all information may be lost. For example, in determining when to consider the next bit to have started, a very useful mechanism is of using signal transition to indicate start of the next bit. But, if there is a string of bits of the same value, then this method does not work.

In such a case, we have to rely on the clock of the receiver to determine when a bit hasended. This has potential problems because sometimes there can be delays at the transmission end which may stretch the time of transmission, causing the receiver to wrongly believe that there are more bits than is actually the case. Moreover, drift in the clock of the receiver can also lead to difficulties. So, we cannot directly rely on the transmitter and receiver clocks being accurate enough to achieve synchronisation. One way out of this is, to send a separate timing pulse along with the signal on a separate cable, but this increases the cost of the scheme.

Another problem with unipolar encoding is that the average voltage of the signal is non-zero. This causes a direct current component to exist in the signal. This cannot be transmitted through some components of the circuit, causing difficulties in reception.

Polar

Unlike unipolar schemes, the polar methods use both a positive as well as a negative voltage level to represent the bits. So, a positive voltage may represent a 1 and a negative voltage may represent a 0, or the other way round. Because both positive and negative voltages are present, the average voltage is much lower than in the unipolar case, mitigating the problem of the DC component in the signal. Here, we will look at three popular polar encoding schemes.

Non-return to Zero (NRZ) is a polar encoding scheme which uses a positive as well asa negative voltage to represent the bits. A zero voltage will indicate the absence of any communication at the time. Here again there are two variants based on level and inversion. In NRZ-L (for level) encoding, it is the levels themselves that indicate the bits, for example a positive voltage for a 1 and a negative voltage for a 0. On the other hand, in NRZ-I (inversion) encoding, an inversion of the current level indicates a 1, while a continuation of the level indicates a 0.

Regarding synchronisation, NRZ-L is not well placed to handle it if, there is a string of 0's or 1's in the transmission. But NRZ-I is better off here as a 1 is signaled by the inversion of the voltage. So every time a 1 occurs, the voltage is inverted and we know the exact start of the bit, allowing the clock to be synchronised with the transmitter. However, this advantage does not extend to the case where there is a string of 1's. We are still vulnerable to losing synchronisation in such a case.

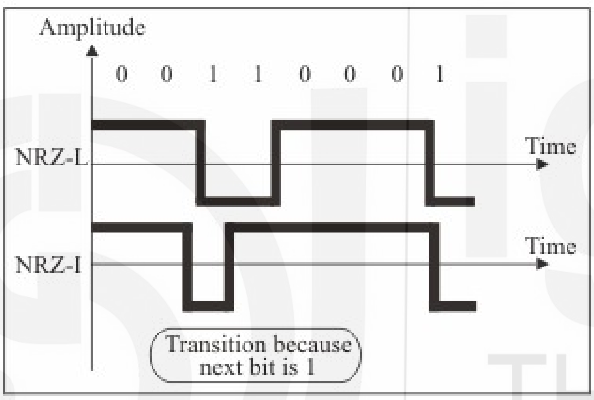

Figure 8: NRZ - L and NRZ I encoding

Return to Zero (RZ) is a method that allows for synchronisation after each bit. This is achieved by going to the zero level midway through each bit. So a 1 bit could be represented by the positive to zero transition and a 0 bit by a negative to zero transition. At each transition to zero, the clocks of the transmitter and receiver can be synchronised. Of course, the price one has to pay for this is a higher bandwidth requirement.

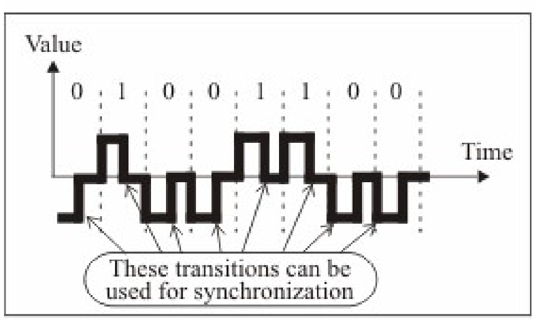

Figure 9: RZ encoding

A third polar method is biphase.encoding. While in RZ, the signal goes to zero midway through each bit, in biphase it goes to the opposite polarity midway through each bit. These transitions help in synchronisation, as you may have guessed. Again this has two flavours called Manchester encoding and Differential Manchester encoding. In the former, there is a transition at the middle of each bit that achieves synchronisation. A 0 is represented by a negative to positive transition while a 1 is represented by the opposite. In the Differential Manchester method, the transition halfway through the bit is used for synchronisation as usual, but a further transition atthe beginning of the bit represents the bit itself. There is no transition for a 1 and there is a transition for a 0, so that a 0 is actually represented by two transitions.

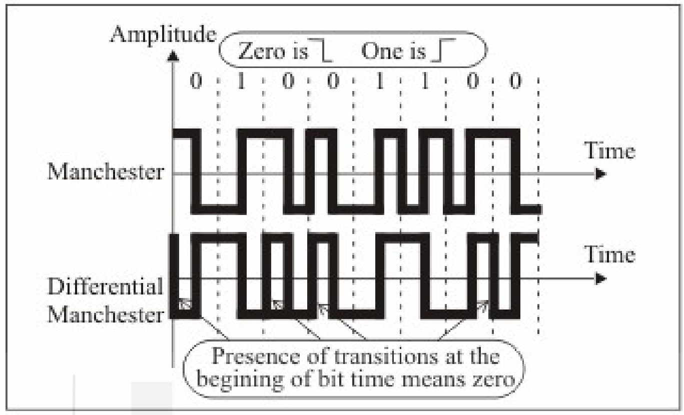

Figure 10: Manchester and differential manchester encoding

Bipolar

This encoding method uses three levels, with a zero voltage representing a 0 and a 1 being represented by positive and negative voltages respectively. This way the DC component is done away with because, every 1 cancels the DC component introducedby the previous 1. This kind of transmission also ensures that every 1 bit is synchronised. This simplest bipolar method is called Alternate Mark Inversion (AMI). The problem with Bipolar AMI is that synchronisation can be lost in a long string of 0's.

One way of synchronising 0's is to use the High Density Bipolar 3 (HDB3) method. This changes the pattern after every four consecutive 0's. The change is to represent the 0 by a positive or negative signal instead of the zero voltage. How do we prevent this from being wrongly interpreted as a 1? It is done by using the same voltage level as was used for the previous 1. As in bipolar encoding, a real 1 bit is represented by the opposite voltage level every time, there is no possibility of confusing the 0 with a
1. So we deliberately violate the convention to achieve synchronisation, a scheme which works because, the violation can be logically detected and interpreted for synchronisation.

HDB3 changes the pattern of four 0's depending on the number of 1's since the last such substitution and the polarity of the last 1. If the number of 1's since the last substitution is odd, the pattern is violated at the fourth 0. So if the previous 1 was positive, the 0 is also made positive to avoid confusion with a 1. If the number of 1's since the last substitution is even, the first and fourth zeroes have a pattern violation.

Another method of achieving 0 synchronisation is called 8-Zero Substitution (B8ZS). Here, a string of 8 zeroes is encoded depending on the polarity of the previous actual 1bit that occurred. If the previous 1 was sent as a negative voltage, the zeroes are sent as three 0's followed by a negative, positive, zero, positive, negative voltage sequence. A similar but opposite pattern violation is performed for the case where thelast 1 was positive. This way a deliberate pattern violation is used to achieve synchronisation.

☞ **Check Your Progress 4**

1) Which type of encoding is most effective at removing the DC component in thesignal and why ?

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2) How does return to zero ensure synchronisation irrespective of the data that istransmitted? What is its disadvantage ?

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3) How can we reconstruct the correct signal inspite of pattern violation in bipolarencoding ?

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3.7 SUMMARY

In this unit, you have seen the need for data encoding and seen that there are four types of data encoding. These arise from the fact that there are 2 source signal types and 2 transmission types. You have seen the different kinds of techniques used for each type of data encoding – analog-to-analog, analog to digital, digital to analog and digital-to-digital. The bandwidth needed, noise tolerance, synchronisation methods and other features of each different technique of encoding have been elaborated upon.

3.8 SOLUTIONS/ANSWERS

Check Your Progress 1

1) Please see the text (Refer Section 3.3, 3.4, 3.5)

2) The spectrum allocation for FM is 88 to 108 MHz, which is 20 MHz. Since abandwidth of 200 KHz is needed for each station, we can have 100 stations in an area. But as discussed, only alternate stations are allowed in one particulararea, to prevent any possibility of interference between them. This means that in any on geographical area, we can have at most 50 radio stations.

3) Please see the text (Refer Section 3.5)

Check Your Progress 2

1) Please see the text.

2) From Nyqist's theorem, for reconstructing a 1 KHz, signal, we must

sample atleast 2000 times a second, corresponding to a sampling interval of 500 microseconds. However, to be safe and to take care of noise and other perturbations, we may like to sample at a somewhat more frequent interval, such as 400 microseconds.

3) The voltage interval is 12.8v, that has to be quantised into 64 levels. Each level will therefore be 0.2v. Starting from 0v, we find that –3.6v will correspond to alevel of –18, while 0.88v will correspond to a level of +4.

Check Your Progress 3

1) Please see the text.

2) A voltage level of 0 can become anything from –0.2v to 0.2v because of noise. The next detectable level will be 0.7v which can become anything from 0.5v to 0.9v because of noise but will still remain detectable as a separate level. Similarly the next levels that will always remain detectable are 1.4v, 2.1v, 2.8vand 3.5v, making for a total of 10 levels from –3.6v to +3.6v.

3) We can have 18 levels as there are a total of 360 degrees available for shifting.

4) With 10 amplitude levels and 18 phase levels we can have 180 QAM combinations, of which we can then use only 90 for safety.

Check Your Progress 4

1) Bipolar encoding eliminates the DC component entirely because each 0 is represented by a 0 voltage while, a 1 is represented alternately by a positive and a negative voltage. So each 1 cancels out the DC component of the voltage introduced by the previous 1.

2) Please see the text.

3) Please see the text.

3.9 FURTHER READINGS

This unit has tried to cover a vast area in the very little space available. For a more detailed treatment of data encoding, you may refer to the following books:

1) *Computer Communications and Networking Technologies*, by Michael A Galloand William M Hancock, Thomson Asia, Second Reprint, 2002.

2) *Introduction to Data Communications and Networking,* by Behrouz Forouzan,Tata McGraw-Hill, 1999.

3) *Networks,* Tirothy S. Ramteke, Second Edition, Pearson Education, New Delhi, 2004.