B.Sc. Computer Science
SEMESTER II
CS2CRT03: Data Communication

Unit IV: Analog Transmission

Converting digital data to a bandpass analog signal.is traditionally called digital to analog conversion. Converting a low-pass analog signal to a bandpass analog signal is traditionally called analog-to-analog conversion

Digital to Analog Conversation

Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.

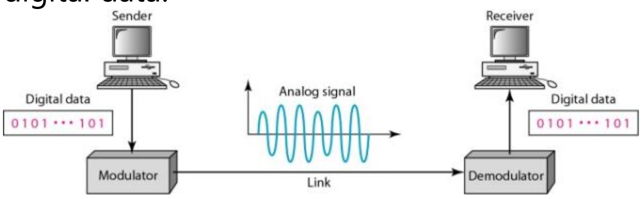


Figure shows the relationship between the digital information, the digital-to-analog modulating process, and the resultant analog signal.

Modulation of Digital Data

Digital Modulation provides more information capacity, high data security, quicker system availability with great quality communication. Hence, digital modulation techniques have a greater demand, for their capacity to convey larger amounts of data than analog modulation techniques.

There are many types of digital modulation techniques and also their combinations, depending upon the need.

Bit Rate & Baud Rate

Both Bit rate and Baud rate are generally used in data communication,

Bit rate is the transmission of number of bits per second. On the other hand, Baud rate is defined as the number of signal units per second.

The formula which relates both bit rate and baud rate is given below:

Bit rate = Baud rate x the number of bit per baud.

Let's see the difference between Bit Rate and Baud Rate:

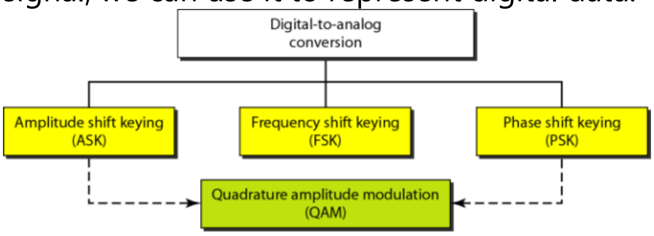
S.No	Bit Rate	Baud Rate
1	Bit rate is defined as the transmission of number of bits per second.	Baud rate is defined as the number of signal units per second.
2	Bit rate is also defined as per second travel number of bits.	Baud rate is also defined as per second number of changes in signal.
3	Bit rate emphasized on computer efficiency.	While baud rate emphasized on data transmission.
4	The formula of **Bit Rate** is: = baud rate x the number of bit per baud	The formula of **Baud Rate** is: = bit rate / the number of bit per baud
5	Bit rate is not used to decide the requirement of bandwidth for transmission of signal.	While baud rate is used to decide the requirement of bandwidth for transmission of signal.

Carrier signal

In analog transmission, the sending device produces a high-frequency signal that acts as a base for the information signal. This base signal is called the carrier signal or carrier frequency. The receiving device is tuned to the frequency of the carrier signal that it expects from the sender. Digital information then changes the carrier signal by modifying one or more of its characteristics (amplitude, frequency, or phase). This kind of modification is called modulation (shift keying).

Types of Digital-To-Analog Conversion

A sine wave is defined by three characteristics: amplitude, frequency, and phase. When we vary anyone of these characteristics, we create a different version of that wave. So, by changing one characteristic of a simple electric signal, we can use it to represent digital data.



Any of the three characteristics can be altered in this way, giving us at least three mechanisms for modulating digital data into an analog signal:

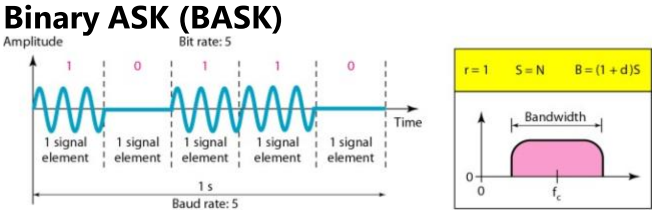
- Amplitude shift keying (ASK),
- Frequency shift keying (FSK), and
- Phase shift keying (PSK).

In addition, there is a fourth (and better) mechanism that combines changing both the amplitude and phase, called quadrature amplitude modulation (QAM).
QAM is the most efficient of these options and is the mechanism commonly used today

ASK

In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes.

Binary ASK (BASK)



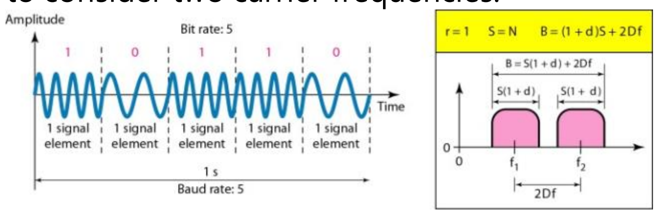
Although we can have several levels (kinds) of signal elements, each with a different amplitude, ASK is normally implemented using only two levels. This is referred to as binary amplitude shift keying or on-off keying (OOK). The peak amplitude of one signal level is 0; the other is the same as the amplitude of the carrier frequency. Figure gives a conceptual view of binary ASK.

FSK

In frequency shift keying, the frequency of the carrier signal is varied to represent data.
The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes. Both peak amplitude and phase remain constant for all signal elements.

Binary FSK (BFSK)

One way to think about binary FSK (or BFSK) is to consider two carrier frequencies.



In Figure we have selected two carrier frequencies, f_1 and f_2. We use the first carrier if the data element is 0; we use the second if the data element is 1. However, note that this is an unrealistic example used only for demonstration purposes. Normally the carrier frequencies are very high, and the difference between them is very small

PSK

In phase shift keying, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes. Today, PSK is more common than ASK or FSK.

Binary PSK (BPSK)

The simplest PSK is binary PSK, in which we have only two signal elements, one with a phase of 0°, and the other with a phase of 180°.

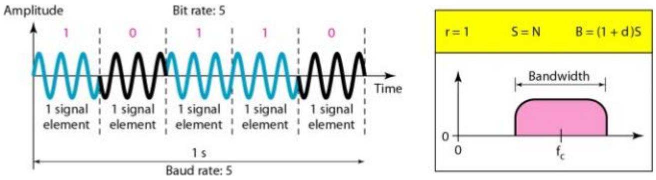


Figure gives a conceptual view of PSK. Binary PSK is as simple as binary ASK with one big advantage-it is less susceptible to noise. In ASK, the criterion for bit detection is the amplitude of the signal; in PSK, it is the phase. Noise can change the amplitude easier than it can change the phase. In other words, PSK is less susceptible to noise than ASK. PSK is superior to FSK because we do not need two carrier signals. Bandwidth in Figure also shows the bandwidth for BPSK. The bandwidth is the same as that for binary ASK, but less than that for BFSK. No bandwidth is wasted for separating two carrier signals.

QAM

PSK is limited by the ability of the equipment to distinguish small differences in phase. This factor limits its potential bit rate. We have been altering only one of the three characteristics of a sine wave at a time; but we can alter or combine ASK and PSK.

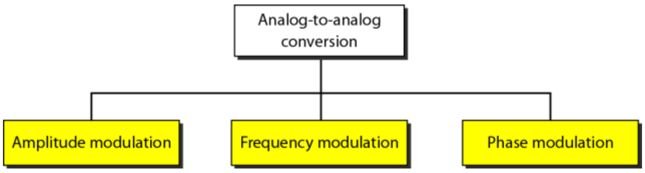
QAM←ASK+PSK

The idea of using two carriers, one in-phase and the other quadrature, with different amplitude levels for each carrier is the concept behind **Quadrature Amplitude Modulation (QAM).**

Analog To Analog Modulation

Analog-to-analog conversion, or analog modulation, is the representation of analog information by an analog signal. One may ask why we need to modulate an analog signal; it is already analog. Modulation is needed if the medium is bandpass in nature or if only a bandpass channel is available to us.

An example is radio. The government assigns a narrow bandwidth to each radio station. The analog signal produced by each station is a low-pass signal, all in the same range. To be able to listen to different stations, the low-pass signals need to be shifted, each to a different range.



Analog-to-analog conversion can be accomplished in three ways:

- Amplitude modulation (AM),
- Frequency modulation (FM), and
- Phase modulation (PM).

FM and PM are usually categorized together.

Amplitude Modulation

In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal. The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information.

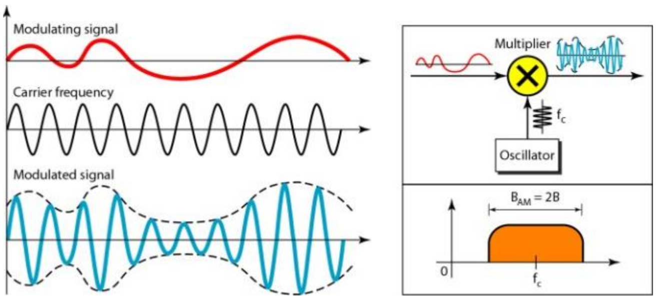
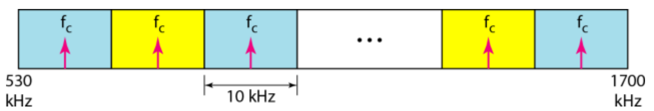


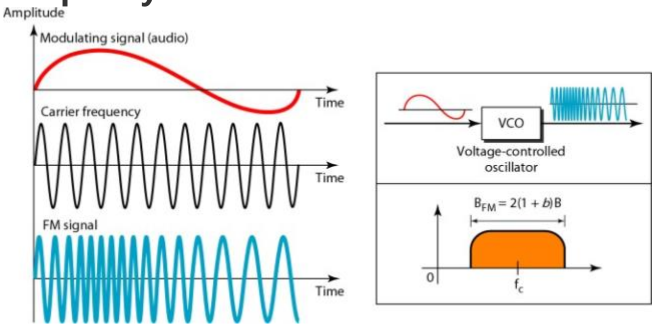
Figure shows how this concept works. The modulating signal is the envelope of the carrier. As in Figure, AM is normally implemented by using a simple multiplier because the amplitude of the carrier signal needs to be changed according to the amplitude of the modulating signal.

Bandwidth for AM

The total bandwidth required for AM can be determined from the bandwidth of the audio signal: $B_{AM} = 2B$.



Frequency Modulation

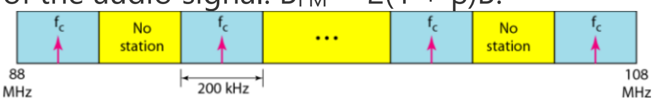


In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly.

Figure shows the relationships of the modulating signal, the carrier signal, and the resultant FM signal. As in Figure, FM is normally implemented by using a voltage-controlled oscillator as with FSK. The frequency of the oscillator changes according to the input voltage which is the amplitude of the modulating signal.

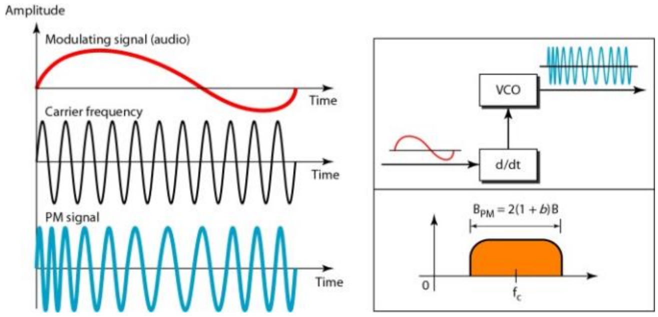
Bandwidth for FM

The total bandwidth required for FM can be determined from the bandwidth of the audio signal: $B_{FM} = 2(1 + \beta)B$.



Phase Modulation

In PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and frequency of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly. It can proved mathematically that PM is the same as FM with one difference.



In FM, the instantaneous change in the carrier frequency is proportional to the amplitude of the modulating signal; in PM the instantaneous change in the carrier frequency is proportional to the derivative of the amplitude of the modulating signal. Figure shows the relationships of the modulating signal, the carrier signal, and the resultant PM signal.

Bandwidth for PM

The total bandwidth required for PM can be determined from the bandwidth and maximum amplitude of the modulating signal: $B_{PM} = 2(1 + \beta)B$.

Bandwidth Utilization

In real life, we have links with limited bandwidths. The wise use of these bandwidths has been, and will be, one of the main challenges of electronic communications. However, the meaning of *wise* may depend on the application. Sometimes we need to combine several low-bandwidth channels to make use of one channel with a larger bandwidth. Sometimes we need to expand the bandwidth of a channel to achieve goals such as privacy and anti-jamming. We explore these two broad categories of bandwidth utilization:

- multiplexing and
- Spreading.

In multiplexing, our goal is efficiency; we combine several channels into one. In spreading, our goals are privacy and anti-jamming; we expand the bandwidth of a channel to insert redundancy, which is necessary to achieve these goals.

Bandwidth utilization is the wise use of available bandwidth to achieve specific goals.

Efficiency can be achieved by multiplexing; privacy and anti-jamming can be achieved by spreading.

Multiplexing

Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared.

Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link.

As data and telecommunications use increases, so does traffic. We can accommodate this increase by continuing to add individual links each time a new channel is needed; or we can install higher-bandwidth links and use each to carry multiple signals.

Today's technology includes high-bandwidth media such as optical fiber and terrestrial and satellite microwaves. Each has a bandwidth far in excess of that needed for the average transmission signal. If the bandwidth of a link is greater than the bandwidth needs of the devices connected to it, the bandwidth is wasted. An efficient system maximizes the utilization of all resources; bandwidth is one of the most precious resources we have in data communications.

In a multiplexed system, *n* lines share the bandwidth of one link.

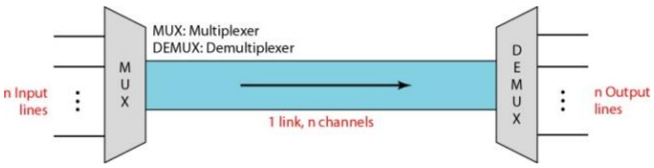
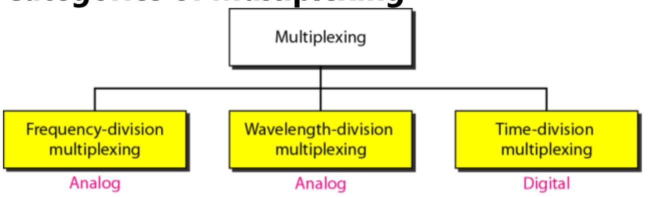


Figure shows the basic format of a multiplexed system. The lines on the left direct their transmission streams to a multiplexer (MUX), which combines them into a single stream (many-to-one).

At the receiving end, that stream is fed into a demultiplexer (DEMUX), which separates the stream back into its component transmissions (one-to-many) and directs them to their corresponding lines.

In the figure, the word **link** refers to the physical path. The word **channel** refers to the portion of a link that carries a transmission between a given pair of lines. *One link can have many (n) channels.*

Categories of Multiplexing



There are three basic multiplexing techniques:

- Frequency-division multiplexing
- Wavelength-division multiplexing, and
- Time-division multiplexing.

The first two are techniques designed for analog signals, the third, for digital signals

FDM

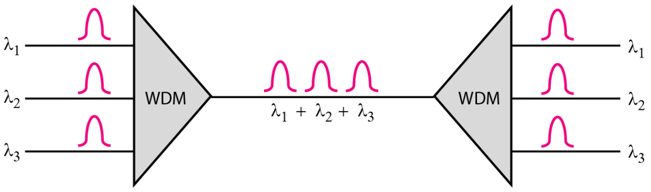
Frequency-Division Multiplexing (FDM) is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted.

In FDM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. These bandwidth ranges are the channels through which the various signals travel. Channels can be separated by strips of unused bandwidth-guard bands-to prevent signals from overlapping. In addition, carrier frequencies must not interfere with the original data frequencies.

Figure gives a conceptual view of FDM. In this illustration, the transmission path is divided into three parts, each representing a channel that carries one transmission.

WDM

Wavelength-division multiplexing (WDM) is designed to use the high-data-rate capability of fiber-optic cable. The optical fiber data rate is higher than the data rate of metallic transmission cable. Using a fiber-optic cable for one single line wastes the available bandwidth.



Multiplexing allows us to combine several lines into one. WDM is conceptually the same as FDM, except that the multiplexing and demultiplexing involve optical signals transmitted through fiber-optic channels. The idea is the same: We are combining different signals of different frequencies. The difference is that the frequencies are very high.

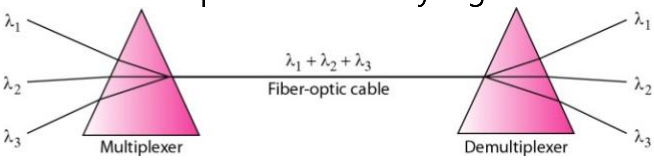
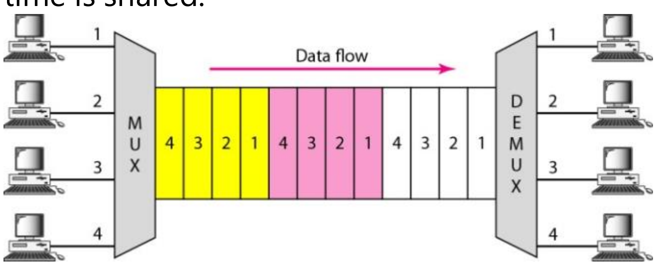


Figure gives a conceptual view of a WDM multiplexer and demultiplexer. Very narrow bands of light from different sources are combined to make a wider band of light. At the receiver, the signals are separated by the demultiplexer.

TDM/Synchronous TDM

Time-Division Multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a line Instead of sharing a portion of the bandwidth as in FDM, time is shared.



Each connection occupies a portion of time in the link. Figure gives a conceptual view of TDM. Note that the same link is used as in FDM; here, however, the link is shown sectioned by time rather than by frequency. In the figure, portions of signals 1, 2, 3, and 4 occupy the link sequentially. Note that in Figure we are concerned with only multiplexing, not switching.

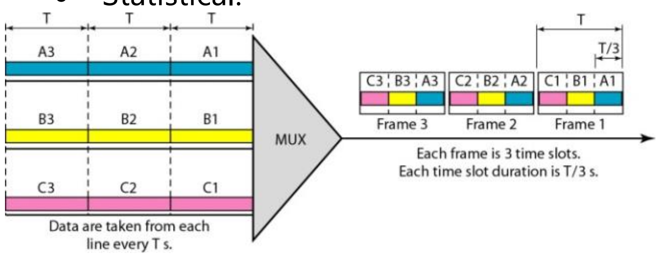
This means that all the data in a message from source 1 always go to one specific destination, be it 1, 2, 3, or 4. The delivery is fixed and unvarying, unlike switching.

We also need to remember that TDM is, in principle, a digital multiplexing technique.
Digital data from different sources are combined into one timeshared link. However, this does not mean that the sources cannot produce analog data; analog data can be sampled, changed to digital data, and then multiplexed by using TDM.

TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

We can divide TDM into two different schemes:

- Synchronous and
- Statistical.



In synchronous TDM, each input connection has an allotment in the output even if it is not sending data.

Digital Signal Services

Telephone companies implement TDM through a hierarchy of digital signals, called digital signal (DS) service or digital hierarchy.

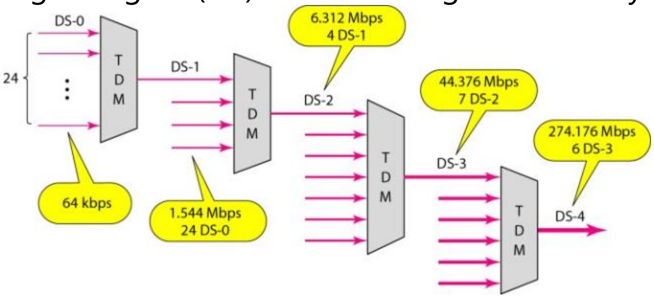
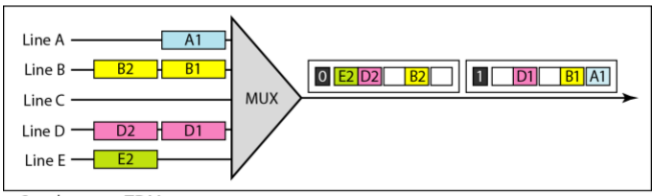


Figure shows the data rates supported by each level.

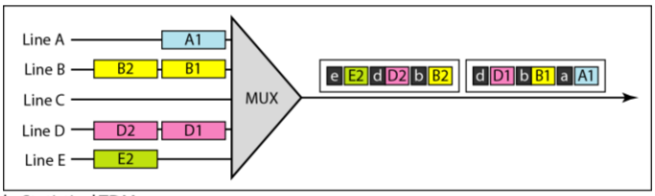
Statistical TDM

As we saw in the previous section, in synchronous TDM, each input has a reserved slot in the output frame. This can be inefficient if some input lines have no data to send.

In **Statistical Time-Division Multiplexing**, slots are dynamically allocated to improve bandwidth efficiency. Only when an input line has a slot's worth of data to send is it given a slot in the output frame. In statistical multiplexing, the number of slots in each frame is less than the number of input lines. The multiplexer checks each input line in Round-Robin (RR) fashion; it allocates a slot for an input line if the line has data to send; otherwise, it skips the line and checks the next line.



a. Synchronous TDM



b. Statistical TDM

Figure shows a synchronous and a statistical TDM example. In the former, some slots are empty because the corresponding line does not have data to send. In the latter, however, no slot is left empty as long as there are data to be sent by any input line.

Spread Spectrum

Multiplexing combines signals from several sources to achieve bandwidth efficiency; the available bandwidth of a link is divided between the sources. In spread spectrum (SS), we also combine signals from different sources to fit into a larger bandwidth, but our goals are somewhat different.

Spread spectrum is designed to be used in wireless applications (LANs and WANs). In these types of applications, we have some concerns that outweigh bandwidth efficiency.

In wireless applications, all stations use air (or a vacuum) as the medium for communication. Stations must be able to share this medium without interception by an eavesdropper and without being subject to jamming from a malicious intruder (in military operations, for example).

To achieve these goals, spread spectrum techniques add redundancy; they spread the original spectrum needed for each station. If the required bandwidth for each station is *B*, spread spectrum expands it to B_{SS}' such that

$$B_{SS} >> B.$$

The expanded bandwidth allows the source to wrap its message in a protective envelope for a more secure transmission.

An analogy is the sending of a delicate, expensive gift. We can insert the gift in a special box to prevent it from being damaged during transportation, and we can use a superior delivery service to guarantee the safety of the package.

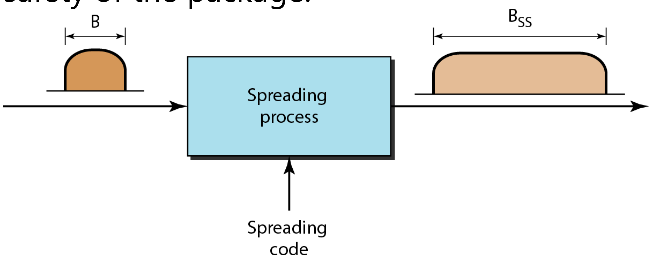


Figure shows the idea of spread spectrum. Spread spectrum achieves its goals through two principles:

1. The bandwidth allocated to each station needs to be, by far, larger than what is needed. This allows redundancy.
2. The expanding of the original bandwidth *B* to the bandwidth Bss must be done by a process that is independent of the original signal. In other words, the spreading process occurs after the signal is created by the source.

After the signal is created by the source, the spreading process uses a spreading code and spreads the bandwidth.

The figure shows the original bandwidth *B* and the spreaded bandwidth B_{SS}. The spreading code is a series of numbers that look random, but are actually a pattern.

There are two techniques to spread the bandwidth:

- Frequency hopping spread spectrum (FHSS)
- Direct sequence spread spectrum (DSSS).

FHSS

The Frequency Hopping Spread Spectrum (FHSS) technique uses *M* different carrier frequencies that are modulated by the source signal. At one moment, the signal modulates one carrier frequency; at the next moment, the signal modulates another carrier frequency. Although the modulation is done using one carrier frequency at a time, M frequencies are used in the long run. The bandwidth occupied by a source after spreading is $B_{FHSS} >> B$.

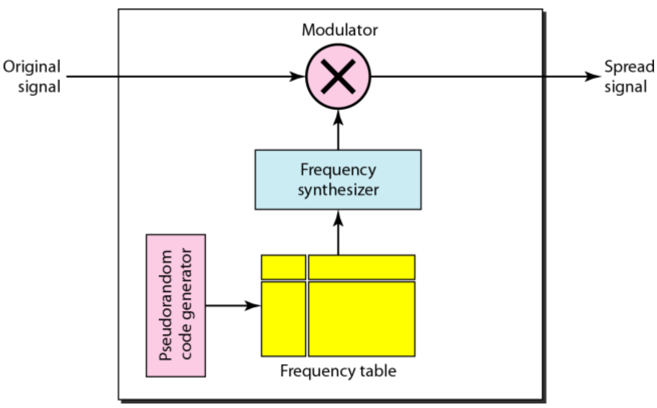
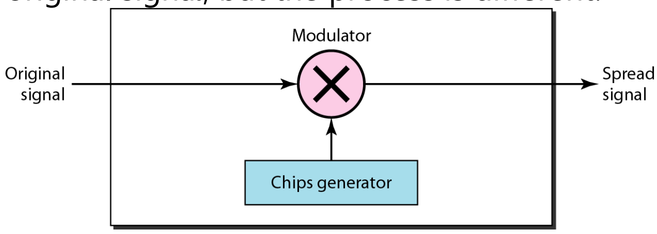


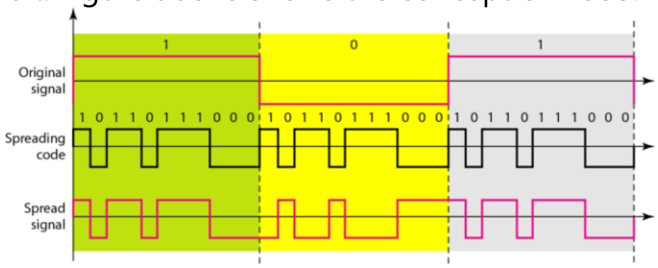
Figure shows the general layout for FHSS. A pseudorandom code generator, called pseudorandom noise (PN), creates a k-bit pattern for every hopping period T_h. The frequency table uses the pattern to find the frequency to be used for this hopping period and passes it to the frequency synthesizer. The frequency synthesizer creates a carrier signal of that frequency, and the source signal modulates the carrier signal.

DSSS

The Direct Sequence Spread Spectrum (DSSS) technique also expands the bandwidth of the original signal, but the process is different.



In DSSS, we replace each data bit with 11 bits using a spreading code. In other words, each bit is assigned a code of 11 bits, called chips, where the chip rate is 11 times that of the data bit. Figure above shows the concept of DSSS.



The spreading code is 11 chips having the pattern 10110111000 (in this case). If the original signal rate is *N*, the rate of the spread signal is *llN*. This means that the required bandwidth for the spread signal is 11 times larger than the bandwidth of the original signal. The spread signal can provide privacy if the intruder does not know the code. It can also provide immunity against interference if each station uses a different code.