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DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

Question Set-2 Communication Systems

Date:-31/03/2020

A. Objectives Questions:

Choose the correct answer:

1. In Pulse code modulation system,
2. In modulation carrier is :
3. FM transmitting and receiving equipment as compared to AM equipment is:
4. Which one of the following is analog?
5. Which one of the following is a digital modulation technique?
6. SSB can be generated by:
7. Under ordinary circumstances, impulse noise can be reduced in:
8. PAM stands for:

B. Short Answer Types Questions:

1. What is modulation? Explain the benefits of modulation.
2. With the help of neat waveforms explain natural PAM sampling and flat top sampling.
3. Draw the block diagram of communication system and explain the function of each block.

C. Long Answer Types Questions:

1) Define noise. Explain detailed classification of noise. What are the sources of noise?
2) Explain the term bandwidth and information capacity.
3) Compare the following digital modulation technique on the basis of bandwidth Requirement and S/N ratio: 1) FSK and 2) PSK
4) What do you mean by bit rate and baud rate?
5) What are the advantages of Single Sideband transmission?
6) Draw the block diagram of FM receiver and explain each block.
7) Write short notes on any two of the following:

a) PWM b) PLL demodulator c) DPSK d) Modulation of DSB-SC

Solutions:

A. Objectives Questions:

1) Large bandwidth is required.
2) Voltage for which frequency, phase or amplitude is varied.
3) Costly
4) PWM
5) all of these (PSK, DM, PCM)
6) all of these (filter method, phase cancellation method, good attenuation characteristics)
7) FM only
8) Pulse Amplitude Modulation

B. Short Answer Types Questions

1) Modulation and Demodulation

The transmission of information and reception of meaningful information can be successfully achieved with the help of two processes. The two processes are

a) Modulation b) Demodulation

Modulation : Modulation is the process of combining the low-frequency Audio waves with a very-high frequency radio waves. The low-frequency wave is called Modulating Wave. The very-high frequency radio wave which carries the low frequency audio wave information is called a Carrier Wave. The resultant wave obtained is called Modulated Carrier.

Demodulation : Demodulation involves recovering the low-frequency audio wave From the Modulated Carrier Wave. This process is performed at the receiving end. It is the reverse process of modulation.

Carrier Wave : Carrier wave is a high frequency radio wave produced using radio-frequency oscillators. The radio frequency ranges from 3kHz to 300GHz. In radio transmission, Carrier waves in the radio frequency range from hundreds of kHz to few MHz are preferred.

Need and benefits for Modulation

Audio frequency signals are low-frequency signals.There are disadvantages in transmitting the unmodulated low frequency signals during communication. They are

1) Low frequency signals cannot propagate over long distances. They are short range signals

2) If there are many transmissions of low frequency signals directly modulation they interfere and the information at the receiving end will not be clear.

3) The antenna length required for the transmission of audio-frequency signals is around 75 m which is practically very large.

Thus a low frequency signal cannot be transmitted effectively and efficiently without modulation. Hence radio frequency carrier waves are modulated by low frequency signals and are transmitted to reach longer distances. Even the antenna size required for the transmission of the radio frequency waves is of reasonable size.

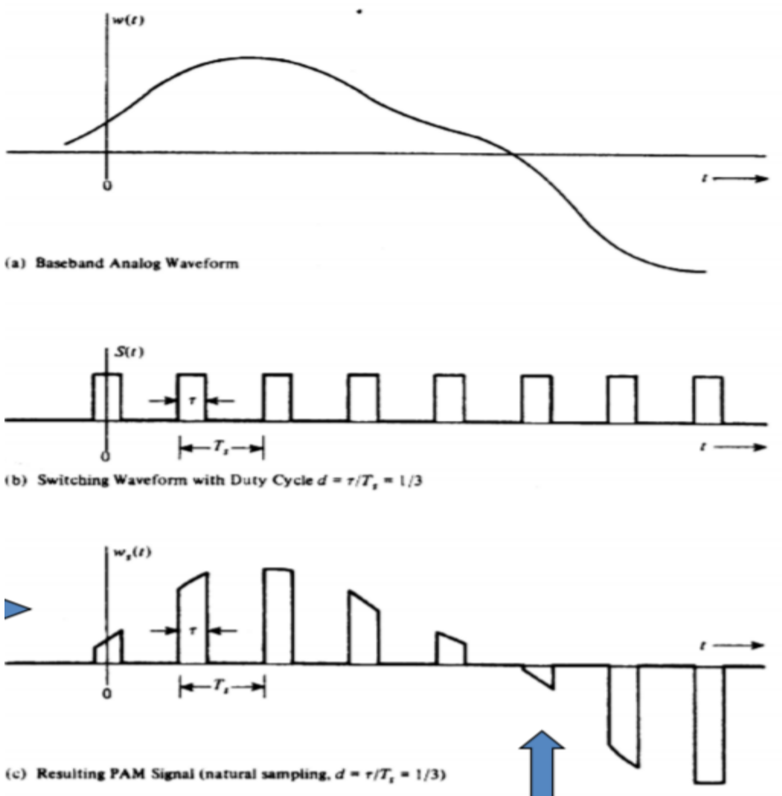
1) Natural PAM sampling and flat top sampling.

NATURAL SAMPLING:

Natural sampling is performed by multiplying w(t) by a train of pulses:
$W_s(t) = w(t) s(t)$ $$
where ,Natural sampling takes a slice of the waveform and the top of the slice

preserves the shape of the waveform.

PAM using natural sampling is as shown below:



(a) Baseband Analog Waveform

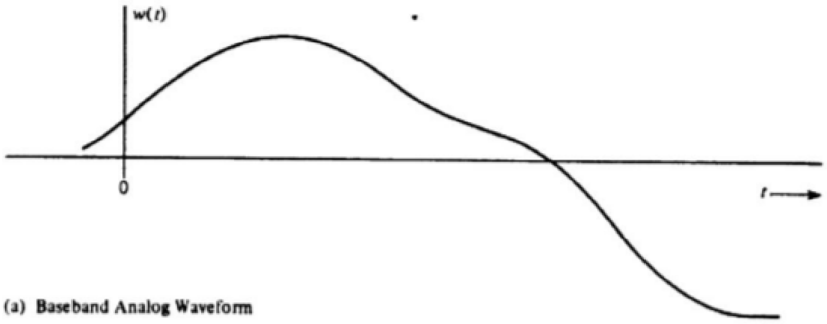
(b) Switching Waveform with Duty Cycle $d = \tau/T_s = 1/3$

(c) Resulting PAM Signal (natural sampling, $d = \tau/T_s = 1/3$)

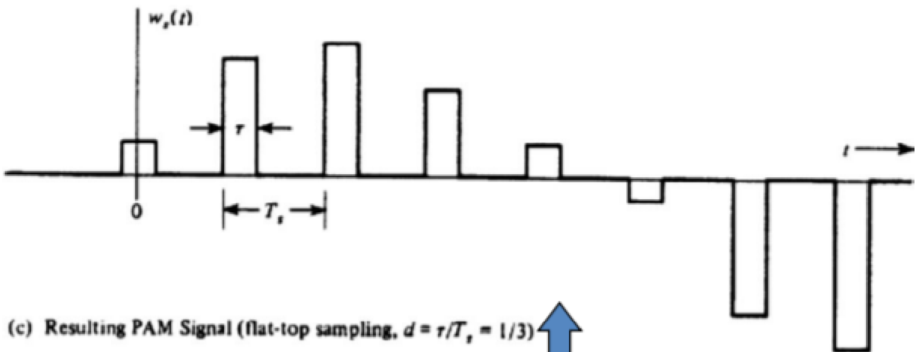
PAM using Natural sampling

FLAT TOP SAMPLING:

In flat top sampling, the top of the samples remains constant and equal to the instantaneous value of the modulating signal at the start of the sampling. Thus the amplitude of the pulse after sampling is kept constant and the top of the sampled pulse do not follow the contour of the modulating signal unlike Natural sampling.
The duration of each sample is τ and the sampling rate is $fs=1/Ts$
Therefore, $Ts=1/fs$
Sample and hold circuit is used for the generation of the sampled signal to attain flat top sampling.



(a) Baseband Analog Waveform



(c) Resulting PAM Signal (flat-top sampling, $d = \tau/T_s = 1/3$)

Pulse Amplitude Modulation using Flat Top sampling

Shannon's Sampling theorem:

- Sampling theorem states that in any pulse modulation system if the sampling rate of the samples exceeds twice the maximum signal frequency, then this ensures the reconstruction of the original signal in the receiver with minimum distortion.

- Sampling theorem can be expressed as given below: $f_s \geq 2f_m$ Where, fs is the sampling frequency and fm is the maximum modulating signal frequency

- Sampling is a process of translating continuous analog signal into discrete analog signal, where the sampled signal is the discrete time representation of the original analog signal.

Aliasing:

- Aliasing is an effect that causes different signals to become indistinguishable from each other during sampling.

- Signal loss may occur due to aliasing effect.

Now, we may state the sampling theorem for strictly band limited signals of finite energy into two equivalent parts:

- A band limited signal of finite energy, which only has frequency components less than **W** Hertz, is completely described by specifying the values of the signal at instants of time separated by **1/2W** seconds
- A band limited signal of finite energy which only has frequency components less than **W** Hertz , may be completely recovered from a knowledge of its sample taken at the rate of **2W** samples per second.

The sampling rate of **2W** samples per second for a signal bandwidth of **W** Hertz is called the Nyquist rate, its reciprocal **1/2W** measured in seconds is called the Nyquist interval.

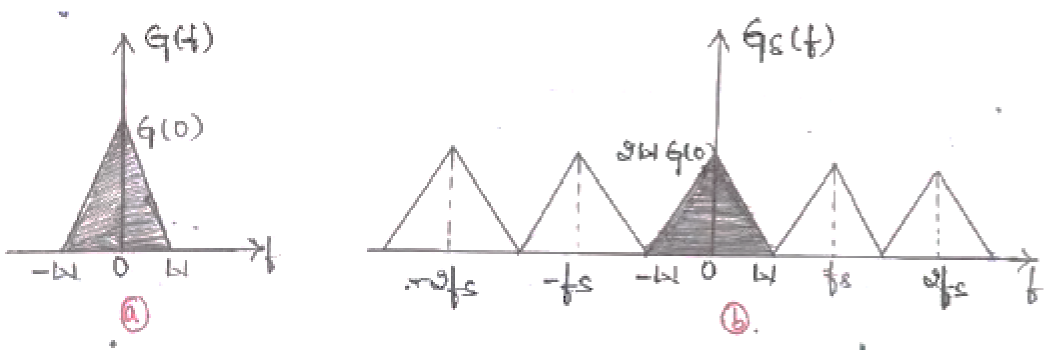
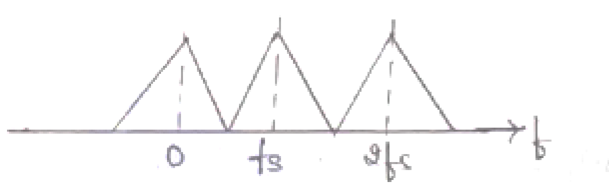


Fig: a) spectrum of a strictly band limited signal g(t)

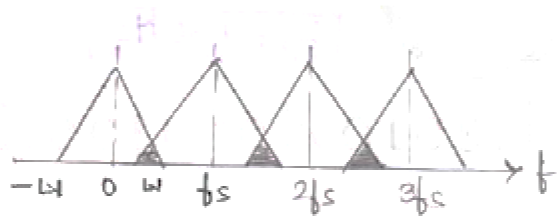
b) spectrum of a sampled version of g(t) for T_s = $\frac{1}{2W}$

NOTE : The concept of under sampling and over sampling is explained below.

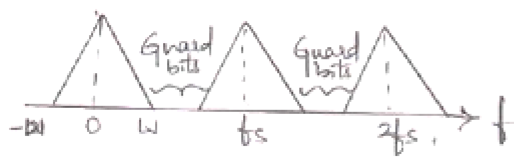
- When sampling frequency $f_r =2W$ then this type of sampling is called correct sampling and here there is no aliasing effect seen in this mechanism i.e when $f_s = 2W$.



- When $f_s < 2W$ then it is under sampling and there will be aliasing effect induced here.



- When $f_s > 2W$ then it is over sampling and there will no aliasing effect.



The effect of aliasing can be reduced by :

- Pre-alias filter must be used to limit band of frequency of the required signal fm Hz.
- Sampling frequency fs must be selected such that sampling frequency is greater than twice the maximum modulating signal frequency.

2) The block diagram of communication system and explain the function of each block.

The elements of basic communication system are as follows

- Information or input signal
- Input Transducer
- Transmitter
- Communication channel or medium
- Noise
- Receiver
- Output Transducer

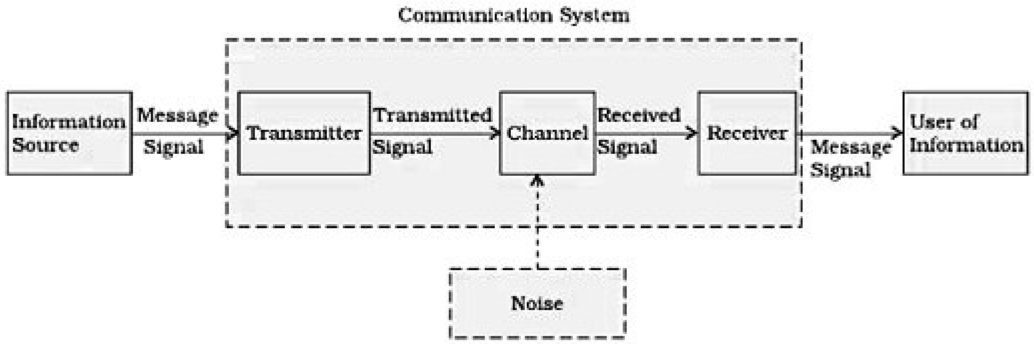


FIGURE 15.1 Block diagram of a generalised communication system.

Fig.1 Block diagram of Communication system

1. Information or input signal

- The communication systems have been developed for communicating useful information from one place to other
- The information can be in the form of sound signal like speech or music or it can be in the form of pictures.

2. Input Transducer

- The information in the form of sound, picture or data signals cannot the transmitted as it is.
- First it has to be converted into a suitable electrical signal.
- The input transducers commonly used in the communication systems are microphones, TV etc.

3. Transmitter

- The function of the transmitter block is to convert the electrical equivalent of the information to a suitable form
- It increases the power level of the signal. The power level should be increased in order to cover a large range. The transmitter consists of the electronics circuits such as amplifier, mixer, oscillator, and power amplifier.

4. Communication channel or medium

- The communication channel is the medium used for the transmission of electronic signals from one place to the another.
- The communication medium can be conducting wires, cables, optical fibres or free space. Depending upon the type of the communication medium, two types of the communication system will exist

  a. Wire communication or line communication

  b. Wireless communication or radio communication

5. Noise

- Noise is an unwanted electrical signal which gets added to the transmitted signal when it is travelling towards receiver.

- Due to noise, the quality of the transmitted information will degrade. One added the noise cannot be separated out from the information

- Hence noise is ab big problem in the communication systems.

6. Receiver

- The reception is exactly the opposite process of transmission. The received signal is amplified and demodulated and converted in a suitable form
- The receiver consists of the electronic circuits like mixer, oscillator, detector and amplifier.

7. Output Transducer

- It consists of the electrical signal at the output of the receiver back to the original form i.e. sound or TV pictures.
- The typical example of the output transducers are loud speakers, picture tubes etc.

C. Long Answer Types Questions:

1) Noise, its classifications & types

Types Of Noise In Communication

In electrical terms, noise is defined as the unwanted form of energy which tends to interface with the proper reception and the reproduction of transmitted signals. Electronic Devices unwanted random addition to the signal are considered as Noise. Their are various types of Noise presents. The Acoustic Noise is observed when the signals are converted into sound which is generally know as snow in TV or video Images. In signalling processing or computing noise can be considered random unwanted data without meaning that is, data that is not being used to transmit signal, but is simply produced as an unwanted by-product of other activities. “signal to noise ratio” is sometimes used to refer to the ratio of useful to irrelevant information in an exchange. Here in this post, Description of Noise of with Classification are explained in detail. If you are interested you can find out more details on various topics from Note Section.

“In common use, the word noise means any unwanted sound”. Unwanted signal are called noise.

Noise

Classification of Noise:

There are several way to classify Noise, but conveniently Noise is classified as 1)ExternalNoise

2) Internal Noise

External Noise:

External noise is defined as the type of Noise which is general externally due to communication system. External Noise are analysed qualitatively. Now, External Noise may be classified as

a) Atmospheric Noise : Atmospheric Noise is also known as static noise which is the natural source of disturbance caused by lightning, discharge in thunderstorm and the natural disturbances occurring in the nature.

b) Industrial Noise : Sources of Industrial noise are auto-mobiles, aircraft, ignition of electric motors and switching gear. The main cause of Industrial noise is High voltage wires. These noises is generally produced by the discharge present in the operations.

c) Extraterrestrial Noise : Extraterrestrial Noise exist on the basis of their originating source. They are subdivided into
i) Solar Noise
ii) Cosmic Noise

Internal Noise:

Internal Noise are the type of Noise which are generated internally or within the Communication System or in the receiver. They may be treated qualitatively and can also be reduced or minimized by the proper designing of the system. Internal Noises are classified as

1) Shot Noise : These Noise are generally arises in the active devices due to the random behaviour of Charge particles or carries. In case of electron tube, shot Noise is produces due to the random emission of electron form cathodes.

2) Partition Noise : When a circuit is to divide in between two or more paths then the noise generated is known as Partition noise. The reason for the generation is random fluctuation in the division.

3) Low- Frequency Noise : They are also known as FLICKER NOISE. These type of noise are generally observed at a frequency range below few kHz. Power spectral density of these noise increases with the decrease in frequency. That why the name is given Low- Frequency Noise.

4) High- Frequency Noise : These noises are also known TRANSIT- TIME Noise. They are observed in the semi-conductor devices when the transit time of a charge carrier while crossing a junction is compared with the time period of that signal.
5) Thermal Noise : Thermal Noise are random and often referred as White Noise or Johnson Noise. Thermal noise are generally observed in the resistor or the sensitive resistive components of a complex impedance due to the random and rapid movement of molecules or atoms or electrons.

2) The term bandwidth and information capacity.

Information Capacity, Bits, and Bit Rate I α B x t, where I= information capacity (bits per second) B = bandwidth (hertz) t = transmission time (seconds), it can be seen that information capacity is a linear function of bandwidth and transmission time and is directly proportional to both. If either the bandwidth or the transmission time changes, a directly proportional change occurs in the information capacity. The higher the signal-to-noise ratio, the better the performance and the higher the information capacity. Mathematically stated, the Shannon limit_for information capacity ,where I = information capacity (bps) B = bandwidth (hertz) N S = signal-to-noise power ratio (unitless) For a standard telephone circuit with a signal-to-noise power ratio of 1000 (30 dB) and a bandwidth of 2.7 kHz, the Shannon limit for information capacity is I = (3.32)(2700) log10 (1 + 1000) = 26.9 kbps

Shannon's formula is often misunderstood. The results of the preceding example indicate that 26.9 kbps can be propagated through a 2.7-kHz communications channel. This may be true, but it cannot be done with a binary system. To achieve an information transmission rate of 26.9 kbps through a 2.7-kHz channel, each symbol transmitted must contain more than one bit.

3) The following digital modulation technique on the basis of bandwidth

Requirement and S/N ratio:

1) FSK and 2) PSK

if the information signal is digital and the amplitude (lV of the carrier is varied proportional to the information signal, a digitally modulated signal called amplitude shift keying (ASK) is produced. If the frequency (f) is varied proportional to the information signal, frequency shift keying (FSK) is produced, and if the phase of the carrier (0) is varied proportional to the information signal, phase shift keying (PSK) is produced. If both the amplitude and the phase are varied proportional to the information signal, quadrature amplitude modulation (QAM) results. ASK, FSK, PSK, and QAM are all forms of digital modulation:

FSK Bit Rate, Baud, and Bandwidth , it can be seen that the time of one bit (tb) is the same as the time the FSK output is a mark of space frequency (ts). Thus, the bit time equals the time of an FSK signaling element, and the bit rate equals the baud. The baud for binary FSK can also be determined by substituting N = 1 : baud = fb / 1 = fb

The minimum bandwidth for FSK is given as

$B = |(fs – fb) – (fm – fb)| = |(fs– fm)| + 2fb$

and since $|(fs– fm)|$ equals $2\Delta f$, the minimum bandwidth can be approximated as

$B = 2(\Delta f + fb)$ (2.15) 14

where B= minimum Nyquist bandwidth (hertz)

Δf= frequency deviation $|(fm– fs)|$ (hertz) fb = input bit rate (bps)

Bandwidth considerations of BPSK. In a BPSK modulator. the carrier input signal is multiplied by the binary data. If + 1 V is assigned to a logic 1 and -1 V is assigned to a logic 0, the input carrier (sin ωc t) is multiplied by either a + or - 1 . The output signal is either + 1 sin ωc t or -1 sin ωc t the first represents a signal that is in phase with the reference oscillator, the latter a signal that is 180° out of phase with the reference oscillator. 27 Each time the input logic condition changes, the output phase changes. Mathematically, the output of a BPSK modulator is proportional to

BPSK output = $[\sin(2\pi fa\ t)] \times [\sin(2\pi fc\ t)]$ (2.20)

where fa = maximum fundamental frequency of binary input (hertz)

fc = reference carrier frequency (hertz) Solving for the trig identity for the product of two sine functions,

$0.5\cos[2\pi(fc – fa)t] – 0.5\cos[2\pi(fc + fa)t]$

Thus, the minimum double-sided

Nyquist bandwidth (B) is fc + fa fc + fa -(fc + fa) or -fc + fa 2fa

and because fa = fb / 2, where fb = input bit rate,

where B is the minimum double-sided Nyquist bandwidth.

the output phase-versus-time relationship for a BPSK waveform. Logic 1 input produces an analog output signal with a 0° phase angle, and a logic 0 input produces an analog output signal with a 180° phase angle.

b) What do you mean by bit rate and baud rate?

Solution:

Baud and Minimum Bandwidth Baud refers to the rate of change of a signal on the transmission medium after encoding and modulation have occurred. Hence, baud is a unit of transmission rate, modulation rate, or symbol rate and, therefore, the terms symbols per second and baud are often used interchangeably. Mathematically, baud is the reciprocal of the time of one output signaling element, and a signaling element may represent several information bits. Baud is expressed as baud = s t 1 (2.7) where baud = symbol rate (baud per second) ts = time of one signaling element (seconds) 6 The minimum theoretical bandwidth necessary to propagate a signal is called the minimum Nyquist bandwidth or sometimes the minimum Nyquist frequency. Thus, fb = 2B, where fb is the bit rate in bps and B is the ideal Nyquist bandwidth. The relationship between bandwidth and bit rate also applies to the opposite situation. For a given bandwidth (B), the highest theoretical bit rate is 2B. For example, a standard telephone circuit has a bandwidth of approximately 2700 Hz, which has the capacity to propagate 5400 bps through it. However, if more than two levels are used for signaling (higher-than-binary encoding), more than one bit may be transmitted at a time, and it is possible to propagate a bit rate that exceeds 2B. Using multilevel signaling, the Nyquist formulation for channel capacity is fb = B log2 M (2.8) where fb = channel capacity (bps) B = minimum Nyquist bandwidth (hertz) M = number of discrete signal or voltage level can be rearranged to solve for the minimum bandwidth necessary to pass M-ary digitally modulated carriers B = M f b 2 log (2.9) 7 If N is substituted for log2 M, reduces to B = N f b (2.10) where N is the number of bits encoded into each signaling element. In addition, since baud is the encoded rate of change, it also equals the bit rate divided by the number of bits encoded into one signaling element. Thus, Baud = N f b (2.11) By comparing Equation 2.10 with Equation 2.11 the baud and the ideal minimum Nyquist bandwidth have the same value and are equal to the bit rate divided by the number of bits encoded.

4)

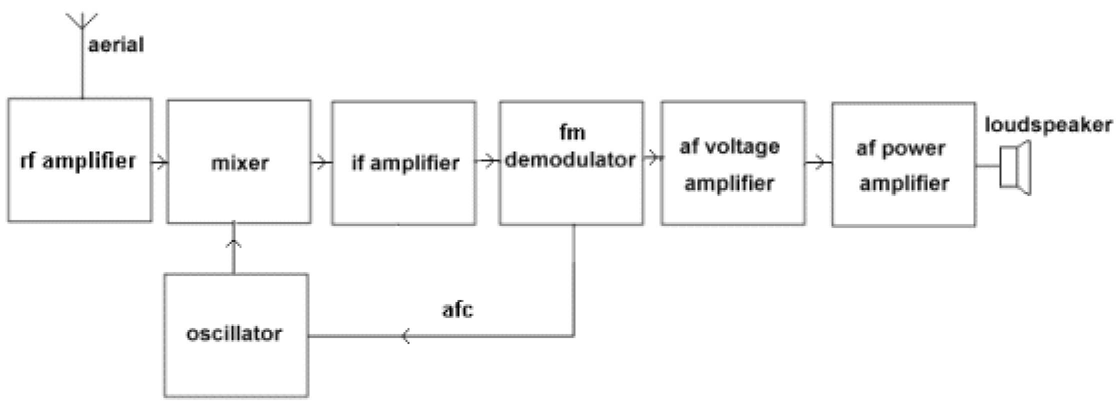
The advantages of single-sideband modulation mainly include the following.

1. Less bandwidth requirement as SSB requires a BW of fm. This will allow more number of signals to be transmitted in the same frequency range.

2. Lots of power saving. This is due to the transmission of only one sideband component. At 100% modulation, the percent power saving is 83.33%
3. Reduced interference of noise. This is due to the reduced bandwidth. As the bandwidth increases, the amount of noise added to the signal will increase.
4. It is used in the applications where the power saving and low bandwidth requirements are important. The application areas are land and air mobile communication, telemetry, military communications, navigation and amateur radio. Many of these applications are point to point communication applications.

6) The block diagram of FM receiver and explain each block.

Solution:



Most of these blocks are discussed individually, and in more detail, on other pages.

See filters, mixers, frequency changers, am modulation and amplifiers.

The f.m. band covers 88-108 MHz.

There are signals from many radio transmitters in this band inducing signal voltages in the aerial.

The rf amplifier selects and amplifies the desired station from the many.

It is adjustable so that the selection frequency can be altered.

This is called TUNING.

In cheaper receivers the tuning is fixed and the tuning filter is wide enough to pass all signals in the f.m. band.

The selected frequency is applied to the mixer.

The output of an oscillator is also applied to the mixer.

The mixer and oscillator form a FREQUENCY CHANGER circuit.

The output from the mixer is the intermediate frequency (i.f.)

The i.f. is a fixed frequency of 10.7 MHz.

No matter what the frequency of the selected radio station is, the i.f. is always 10.7 MHz.

The i.f. signal is fed into the i.f. amplifier.

The advantage of the i.f. amplifier is that its frequency and bandwidth are fixed, no matter what the frequency of the incoming signal is.

This makes the design and operation of the amplifier much simpler.

The amplified i.f. signal is fed to the demodulator.

This circuit recovers the audio signal and discards the r.f. carrier.

Some of the audio is fed back to the oscillator as an AUTOMATIC FREQUENCY CONTROL voltage.
This ensures that the oscillator frequency is stable in spite of temperature changes.

The audio signal voltage is increased in amplitude by a voltage amplifier.

The power level is increased sufficiently to drive the loudspeaker by the power amplifier.

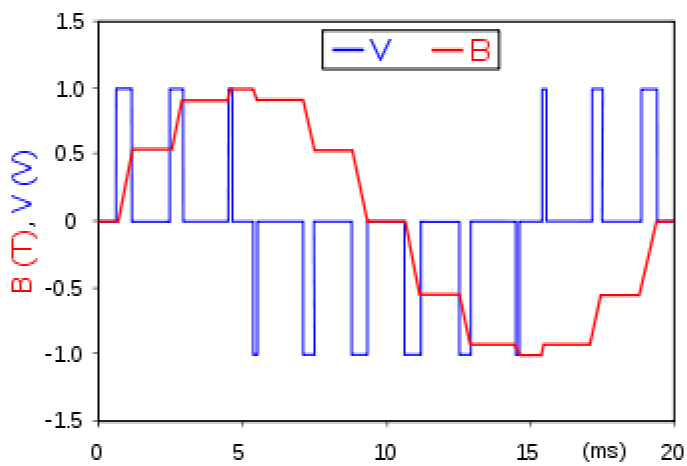
7) Write short notes on any two of the following:

a) PWM

Solution:

Pulse width modulation (**PWM**), or **pulse-duration modulation** (**PDM**), is a method of reducing the average power delivered by an electrical signal, by effectively chopping it up into discrete parts. The average value of voltage (and current) fed to the load is controlled by turning the switch between supply and load on and off at a fast rate. The longer the switch is on compared to the off periods, the higher the total power supplied to the load. Along with MPPT maximum power point tracking, it is one of the primary methods of reducing the output of solar panels to that which can be utilized by a battery. PWM is particularly suited for running inertial loads such as motors, which are not as easily affected by this discrete switching, because they have inertia to react slow. The PWM switching frequency has to be high enough not to affect the load, which is to say that the resultant waveform perceived by the load must be as smooth as possible.

The rate (or frequency) at which the power supply must switch can vary greatly depending on load and application. For example, switching has to be done several times a minute in an electric stove; 120 Hz in a lamp dimmer; between a few kilohertz (kHz) and tens of kHz for a motor drive; and well into the tens or hundreds of kHz in audio amplifiers and computer power supplies. The main advantage of PWM is that power loss in the switching devices is very low. When a switch is off there is practically no current, and when it is on and power is being transferred to the load, there is almost no voltage drop across the switch. Power loss, being the product of voltage and current, is thus in both cases close to zero. PWM also works well with digital controls, which, because of their on/off nature, can easily set the needed duty cycle. PWM has also been used in certain communication systems where its duty cycle has been used to convey information over a communications channel.



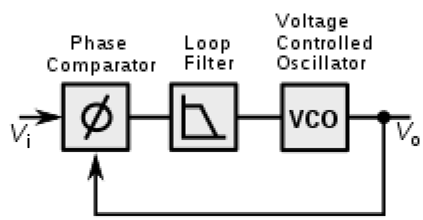
b) PLL demodulator

Solution:

A **phase-locked loop** or **phase lock loop** (**PLL**) is a control system that generates an output signal whose phase is related to the phase of an input signal. There are several different types; the simplest is an electronic circuit consisting of a variable frequency oscillator and a phase detector in a feedback loop. The oscillator generates a periodic signal, and the phase detector compares the phase of that signal with the phase of the input periodic signal, adjusting the oscillator to keep the phases matched.

Keeping the input and output phase in lock step also implies keeping the input and output frequencies the same. Consequently, in addition to synchronizing signals, a phase-locked loop can track an input frequency, or it can generate a frequency that is a multiple of the input frequency. These properties are used for computer clock synchronization, demodulation, and frequency synthesis.

Phase-locked loops are widely employed in radio, telecommunications, computers and other electronic applications. They can be used to demodulate a signal, recover a signal from a noisy communication channel, generate a stable frequency at multiples of an input frequency (frequency synthesis), or distribute precisely timed clock pulses in digital logic circuits such as microprocessors. Since a single integrated circuit can provide a complete phase-locked-loop building block, the technique is widely used in modern electronic devices, with output frequencies from a fraction of a hertz up to many gigahertz.



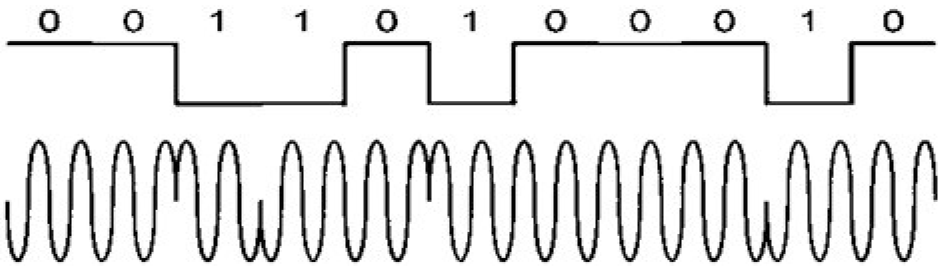
c) DPSK

Solution:

The DPSK stands for "Differential phase-shift keying". It is one type of phase modulation used to transmit data by altering the carrier wave's phase. In this, the modulated signal's phase is moved e to the element of an earlier signal. The phase of the signal tracks the low or high state of the earlier element. This kind of phase-shift keying doesn't require a synchronous carrier on the demodulator.

The binary bits input series can be changed so that the next bit depends upon the earlier bit. So, the earlier received bits in the receiver are utilized for detecting the current bit.

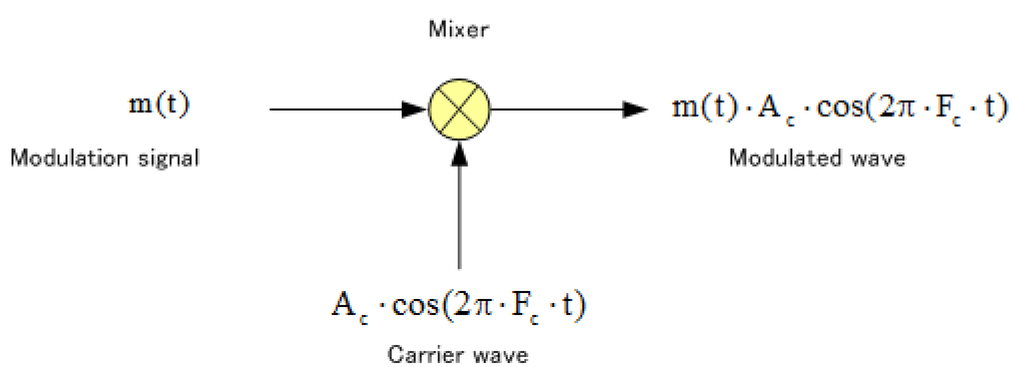
The above-shown figure is the **DPSK waveform**. From the above waveform, once the data-bit is '0', the signal's phase will not be inverted as well as continued. Once the data-bit is a '1', then the signal's phase will be inverted.



d) Modulation of DSB-SC

Solution:

Double sideband suppressed carrier modulation
At the beginning of the explanation of amplitude modulation, we explained the AM radio system, but the term for amplitude in the theoretical expression was complex. If the amplitude of the carrier wave is simply changed and mathematised, it is as follows. This modulation method is called DSB-SC (double sideband suppressed carrier modulation). As the name suggests, there's no wave carrier in the modulated wave.



$$S_{dsb-sc}(t) = K_{dsb-sc} \cdot m(t) \cdot A_c \cdot \cos(2\pi \cdot F_c \cdot t + \phi_c)$$

With DSB-SC, the amplitude of carrier wave Ac is shifted proportionally to the modulating signal m(t).

When the modulating signal m(t) is a single sine wave, it's as follows.

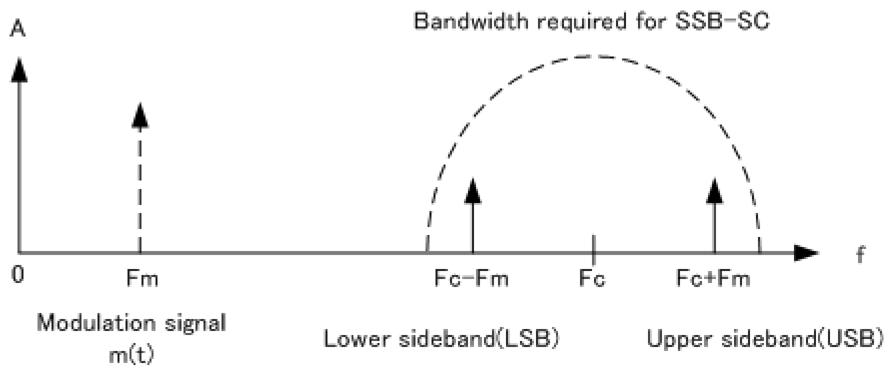
$$m(t) = A_m \cdot \cos(2\pi \cdot F_m \cdot t)$$

$$S_{dsb-sc}(t) = K_{dsb-sc} \cdot A_m \cdot \cos(2\pi \cdot F_m \cdot t) \cdot A_c \cdot \cos(2\pi \cdot F_c \cdot t + \phi_c)$$

If the initial phase ΦC of the carrier wave is 0, and Kdsb-sc*Am is modulation factor m, the result is as follows.

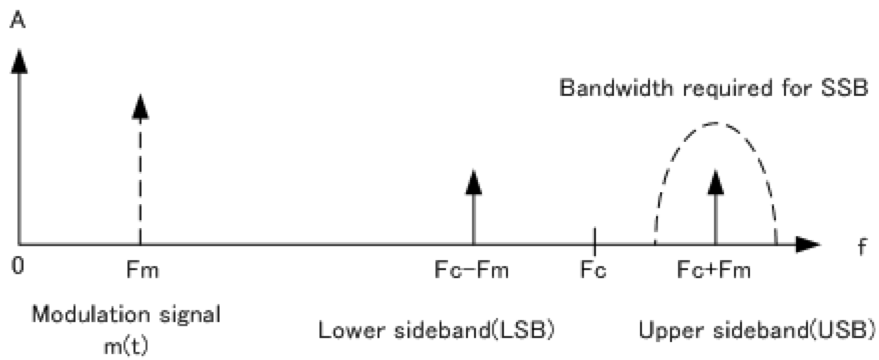
$$S_{dsb-sc}(t) = \frac{m \cdot A_c}{2} \cdot \left\{ \cos\left[2\pi \cdot \left(F_c + F_m\right) \cdot t\right] + \cos\left[2\pi \cdot \left(F_c - F_m\right) \cdot t\right] \right\}$$

The first equation shows a spectrum where only the modulation signal frequency Fm is separate on either side of the carrier frequency Fc. However note that with DSB-SC, there's no carrier wave. For this reason, DSB-SC is considered to have good electrical efficiency. However, synchronous detection is necessary on the receiving end, which involves a system with advanced technology and the associated costs.



◆Single sideband suppressed carrier modulation (SSB-SC)

The modulation method where only a single band of double sideband suppressed carrier modulation is transmitted is known simply as SSB. It offers even better electrical efficiency and frequency band efficiency than DSB. In addition, with digital modulation, modulation is performed at relatively low frequency, and this method is also used to up-convert the signal to a radio frequency. SSB-SC modulation can be achieved with various systems, but in terms of hardware, USB or LSB can be used as a filter. As with DSB-SC, the modulating signal spectrum is shifted directly to the carrier frequency band without loss.


A
Bandwidth required for SSB
f
0
Fm
Fc−Fm
Fc
Fc+Fm
Modulation signal
m(t)
Lower sideband(LSB)
Upper sideband(USB)