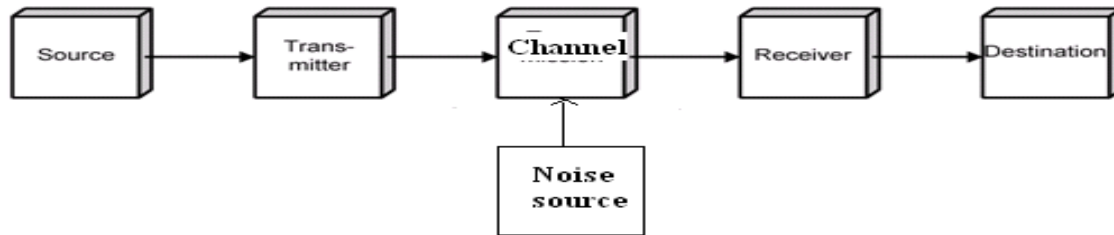


Q.2 a. With neat block diagram explain communication system model.

Answer:



The key elements of this model are:

- Source - generates information/message to be transmitted
- Transmitter - converts message signal into transmittable signals by modulating the signal
- Channel – media between source and destination
- Receiver - converts received signal into message signal
- Destination – message will be put in proper format

Source:

Source may be analog or digital signal.

Transmitter:

It will modulate the low frequency message signal using high frequency carrier before transmission. The different modulation technique are AM, FM and PM.

Channel:

It is the media between transmitter and receiver. It can be either wired or wireless.

Wireless system corresponds to free space and wired may be open wire, co-axial cable or optical fiber.

Receiver:

It is used to demodulate the received signal. The type of demodulation depends on the type of modulation used at transmitter.

Destination:

Here the message will be put in original form like voice, image or Text.

- b. What is noise? What are internal and external noise and briefly discuss them?

Answer:

Any unwanted signal in communication is referred as noise.

Noise generated within the system i.e with in transmitter and receiver is called internal noise.

Noise generated outside the system is called external noise

External noise may be classified into

- I. Atmospheric noise
- II. Extraterrestrial noise
- III. Manmade or Industrial noise

i) **Atmospheric noise**[static]: It is caused by lightning discharges in the thunderstorms and other natural electrical disturbances occurring in the atmosphere. This is random in nature.

ii) **Extraterrestrial noise**: These are two types:

a.Solar noise : This is the electrical noise emanating from the sun under quiet condition, there is a steady radiation of noise from the sun. This solar cycle repeats these electrical disturbances approximately every 11 years.

b.Cosmic noise: Distant stars are also suns and have high temperatures. These stars, therefore radiate noise in the same way as the sun. The noise received from these distant stars is thermal noise and is distributed uniformly over the entire sky.

iii) **Manmade noise**[**industrial noise**] :These electrical noises are produced by such sources as automobiles and aircraft ignition, electrical motors and switch gears, leakage from high voltage lines, fluorescent lights and numerous other heavy electrical machine.

Internal noise: Internal noise may be classified as

(i)Thermal noise or white noise or Johnson noise

(ii)Shot noise

(iii)Transit time noise

(iv)Miscellaneous internal noise

i) **Thermal noise**: The thermal noise is the random noise generated by in a resistor or the resistive component of complex impedance due to rapid and random motion of the molecules, atoms and electrons. The noise power generated in a resistor is proportional to the absolute temperature and bandwidth over which the noise is being measured.

Therefore $P_n = k.T.B$

Where, P_n = Noise power generated in the resistor

k = Boltzmann constant[1.38×10^{-23} J/K]

T = absolute temp

B = bandwidth in Hz

ii) **Shot noise** : This type of noise is found in communication receivers. Shot noises are due to active devices only. The fluctuating electrons in a vacuum tube constitute a randomly varying noise superimposed on the direct current of the output electrode of the amplifying device. When amplified, this noise sounds as a shower of lead shots falling on a metal sheet giving rise to its name shot noise.

iii) **Transit time noise**: At VHF range and beyond it the time taken by the electron from cathode to anode of a tube becomes comparable to the period of the signal; being amplified. This effect is known as transit time effect. It results in a type of noise called transit time noise.

(iv) **Miscellaneous internal noise**:

Here we have flicker noise and partition noise

- **Flicker noise**:

In the low freq range [below 5kHz], it is noticed that the random fluctuation of the anode current is much larger than that expected due to shot noise alone. This effect is due to the

random changes in emissivity of the cathode resulting in flicker noise. This type of noise is more pronounced at low frequency.

Partition noise: This noise is caused because of random distribution of the electrons i.e., the emitter current distributed at the junction as base and collector current if the process is random it results in partition noise

- c. Calculate the noise voltage at the input of a Television RF amplifier, using a device that has a 200-ohm equivalent noise resistance and a 300 ohm input resistor. The bandwidth of the amplifier is 6 MHz and temperature is 17°C.

Answer:

$$v_n = \sqrt{[4kT\delta f R_{tot}]}$$

$$v_n = \sqrt{[4 \times 1.38 \times 10^{-23} (17 + 273) \times 6 \times 10^6 (300 + 200)]} = 6.93 \times 10^{-6} \text{V}$$

- Q.3** a. Define amplitude modulation and modulation index in AM system also derive an expression for AM wave and find its required Bandwidth.

Answer:

Definition:

Amplitude Modulation is the process of varying amplitude of the high frequency carrier wave in accordance with the instantaneous value of the message signal.

Modulation index:

$m = E_m / E_c = \text{Amplitude of message signal} / \text{Amplitude of carrier wave}$

Expression for AM wave

Let $e_c = E_c \sin \omega_c t$ is the carrier signal

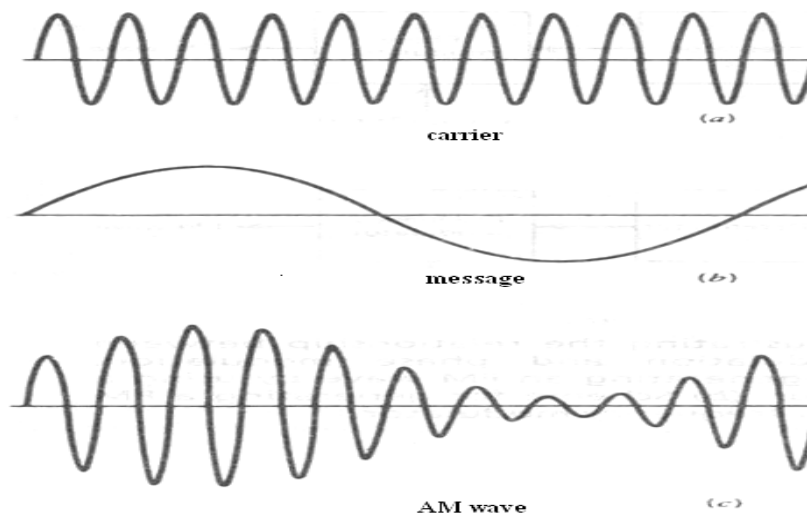
$e_m = E_m \sin \omega_m t$ is the message signal

where e_c and e_m are instantaneous value of carrier and message signals respectively.

E_c and E_m are amplitudes of carrier and message signals respectively.

ω_c and ω_m are angular frequencies of carrier and message signals respectively.

The carrier message and AM are as shown in figure.



In AM amplitude of the carrier is varied in accordance with the message signal hence the modulated wave is given by

$e = A \sin \omega_c t$ Where A is change in amplitude given by

$$A = E_c + e_m = E_c + E_m \sin \omega_m t = E_c [1 + m \sin \omega_m t]$$

Where $m = E_m / E_c$ called modulation index.

Therefore $e = E_c [1 + m \sin \omega_m t] \sin \omega_c t$

$$E = E_c \sin \omega_c t + m E_c / 2 [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$

The modulated signal consists of carrier, upper side band and lower sideband.

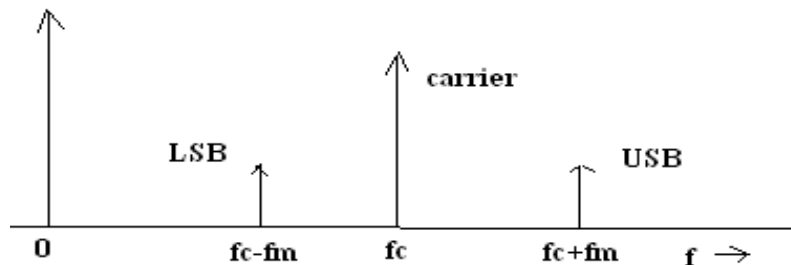


Figure: Spectrum of AM signal

The spectrum of the AM signal is as shown in figure.

From figure $BW = 2f_m$

- b. With suitable block diagram explain filter method of SSB generation.

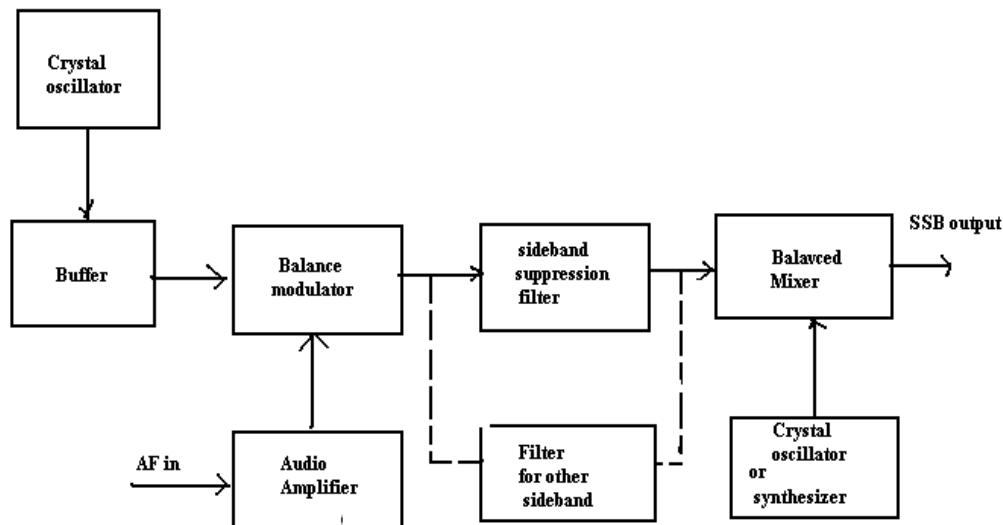
Answer:

The steps to be followed are

Step1: generate DSBSC using balanced modulator.

Step2: remove unwanted sideband using suitable filter.

The block diagram is as shown in figure



As shown in figure DSBSC is generated using BM (Balanced modulator). Which consists of both upper side band [USB] and lower side band [LSB].

We can select either LSB or USB using suitable filter.

In the mixer the frequency of the crystal oscillator or synthesizer is added to the SSB signal from the filter, the frequency thus being raised to the value desired for the transmission.

The output of the mixer is applied to the linear amplifier for further amplification before transmission.

- c. In an AM wave if the carrier and one of the sideband of is suppressed, calculate the percentage power saving for a depth of:
- (i) 100 percent modulation
 - (ii) 80 percent modulation and
 - (iii) 50 percent modulation

Answer:

$$\text{i) } P_t = P_c \left(1 + \frac{m^2}{2} \right) + P_c \left(1 + \frac{1^2}{2} \right) = 1.5P_c$$

$$P_{SB} = P_c \frac{m^2}{4} = P_c \frac{1^2}{4} = 0.25P_c$$

$$\text{Saving} = \frac{1.5 - 0.25}{1.5} = \frac{1.25}{1.5} = 0.833 = 83.3\%$$

$$\text{ii) } P_t = P_c \left(1 + \frac{0.8^2}{2} \right) = 1.32P_c$$

$$P_{SB} = P_c \frac{0.8^2}{4} = 0.16P_c$$

$$\text{Saving} = \frac{1.32 - 0.16}{1.32} = 0.878 = 87.8\%$$

$$\text{iii) } P_t = P_c \left(1 + \frac{0.5^2}{2} \right) = 1.125P_c$$

$$P_{SB} = P_c \frac{0.5^2}{4} = 0.0625P_c$$

$$\text{Saving} = \frac{1.125 - 0.0625}{1.125} = \frac{1.0625}{1.125} = 0.944 = 94.4\%$$

- Q.4** a. With neat circuit diagram explain the working of Phase discriminator type FM demodulation.

Answer:

With neat circuit diagram explain the working of Phase discriminator type FM demodulation.

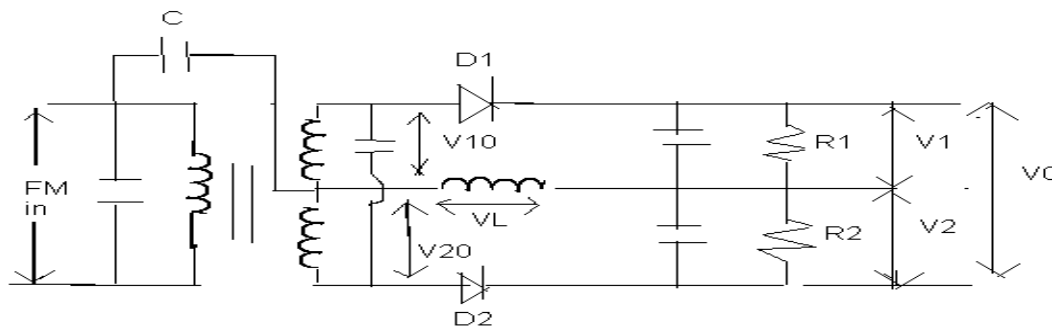


Fig : Ckt dig of phase discriminator

Fig shows the circuit diagram of phase discriminator. Here both the primary and the secondary voltages are

- a) Exactly 90 degrees out of phase when the input frequency f_{in} is equal to f_c .
- b) less than 90 degree out of phase when f_{in} is higher than f_c and
- c) More than 90 degree out of phase when f_{in} is below f_c .

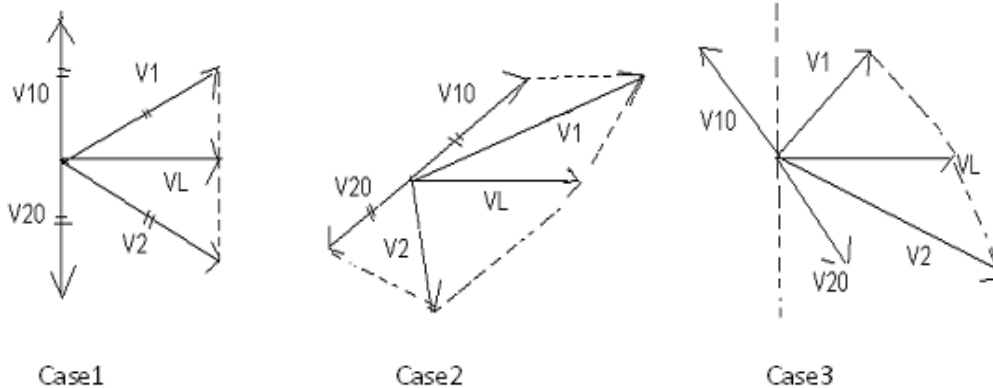
V_{10} & V_{20} represents outputs of a centre-tapped transformer, therefore $V_{10} = -V_{20}$

The output voltage, V_1 is the vector sum of voltages V_L & V_{10}

$$V_1 = V_{10} + V_L \quad (V_L = V_{in})$$

$$\text{Similarly, } V_2 = V_{20} + V_L$$

The vector diagram for different cases are as shown below



Case1: when input frequency $f_{in} = f_c$

Output voltage $V_1 =$ Output voltage V_2

$$V_o = V_1 - V_2$$

$$= 0 \quad \text{voltage No information}$$

Case2: when input frequency $f_{in} > f_c$

Output voltage $V_1 >$ Output voltage V_2

$$V_o = V_1 - V_2$$

$$= +ve \text{ voltage}$$

Case3: when input frequency $f_{in} < f_c$

Output voltage $V_1 <$ Output voltage V_2

$$V_o = V_1 - V_2$$

$$= -ve \text{ voltage}$$

- b. In which system pre-emphasis and de-emphasis are used? Also explain its operation.

Answer:

Pre-emphasis and de-emphasis are used in FM system.

PRE-EMPHASIS AND DE-EMPHASIS:

The spectral density of the noise at the receiver output has a square law dependence on the operating frequency. The spectral density of the message usually falls off appreciably at higher frequencies, on the other hand the spectral density of the output noise increases rapidly with frequency. Thus the SNR decreases at the output of the receiver for higher frequency signal. This clearly shows that the message is not utilizing the frequency band allotted to it in an efficient manner. For efficient utilization of BW the higher frequency components are boosted artificially at the transmitter and correspondingly cut at the receiver.

This boosting of the higher modulating frequencies at transmitter is termed as pre-emphasis and the compensation at the receiver is called de-emphasis.

The pre-emphasis is used at the FM transmitter before modulation in order to increase the signal level at higher frequencies. A simple HPF or differentiator network can be used as pre-emphasis.

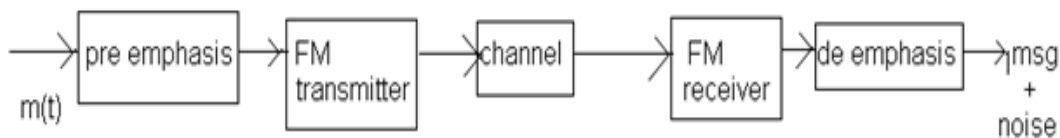
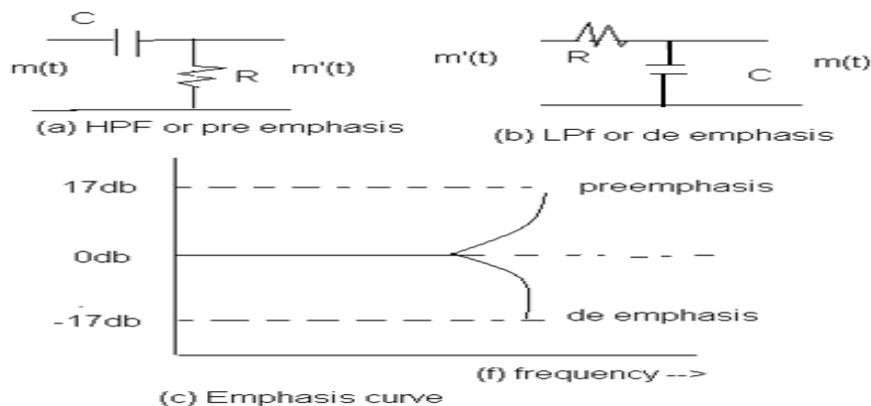


Fig: Use of pre emphasis & de emphasis in FM system



A de-emphasis is used at the FM receiver after demodulation in order to decrease the signal level at higher frequency. A simple LPF or integrator network can be used as de-emphasis.

Improvement factor:

It is defined as a ratio of noise power without emphasis and noise power with emphasis.

- c. An FM wave is represented by $v(t) = 10 \sin(6\pi 10^6 t + 6 \sin 1250\pi t)$. Find
- The carrier and modulating frequencies.
 - The modulation index.
 - Is it narrow band or wideband FM?
 - Maximum deviation of FM wave.
 - What power this FM will dissipate in a 10 ohm resistor?

Answer:

Given FM equation is

$$v(t) = 10 \sin(6\pi 10^6 t + 6 \sin 1250\pi t).$$

Comparing with standard equation $v(t) = A \sin(\omega_c t + m_f \sin \omega_m t)$, we have

$$i) f_c = \frac{6\pi 10^6}{2\pi} = 3\text{Mhz} \quad f_m = \frac{1250\pi}{2\pi} = 625\text{hz}$$

$$ii) m_f = 6$$

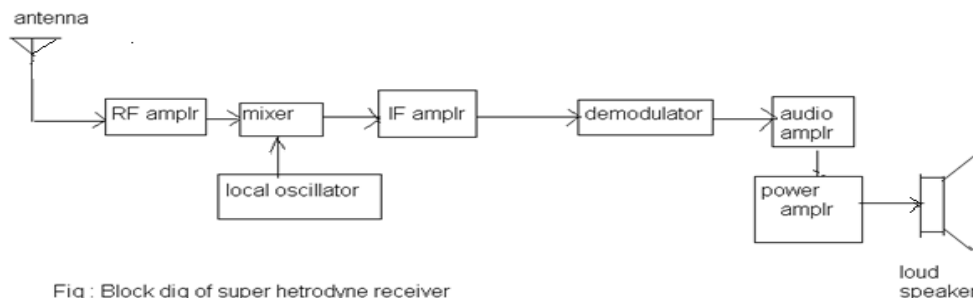
iii) As $m_f > 1$ it is wideband FM

$$iv) \text{ Deviation } \delta = m_f f_m = 6 \times 625 = 3750\text{hz}$$

$$v) P = \frac{V_{rms}^2}{R} = \frac{(10/\sqrt{2})^2}{10} = 5W$$

Q.5 a. With neat diagram explain the working of super- heterodyne receiver.

Answer:



- **Antenna:** It intersects the EM waves; it converts EM waves to electrical signal. Voltage induced in the antenna is coupled to RF amplifier.
- **RF amplifier:** It selects the desired signal using tuning circuit.
 - * It amplifies the received signal.
 - * It acts as matching network between antenna & input of mixer.
 - * It avoids any other spurious radiation from local oscillator.
 - * It consists of tuned amplifier.
 - * It improves SNR.
 - * It improves the sensitivity & selectivity.
- **Frequency conversion stage:** This consists of Local oscillator[LO] & mixer. The mixer being a non liner device it receives incoming signal f_s & the LO frequency

f_o & produce various inter modulation terms. The difference frequency voltage is picked up by the tuned circuit at the output of the mixer. This difference in frequency is called the intermediate frequency [IF].

$$IF = f_o - f_s$$

- **IF amplifier:** It consists of two or more stages of fixed frequency tuned voltage amplifier having 3 dB bandwidth of 10kHz for AM broadcast. This IF amplifier provides most of the receiving amplification and selectivity.
- **Demodulator:** Output of the last IF amplifier is fed to this demodulator, which is generally diode detector. Output of this detector is the original modulating signal.
- **AF amplifier:** Audio output from demodulator is fed to the AF amplifier which provides additional amplification. Usually one stage of audio voltage amplifier is used followed by one or more stages of audio power amplifier.
- **Loudspeaker:** Amplified audio output voltage of audio power amplifier is fed to loudspeaker through impedance matching transformer. The loudspeaker reproduces the original program [converts electrical signal to sound signal]

- b. What is stub? What are the limitation of single stub matching and also explain the working of double stub matching network.

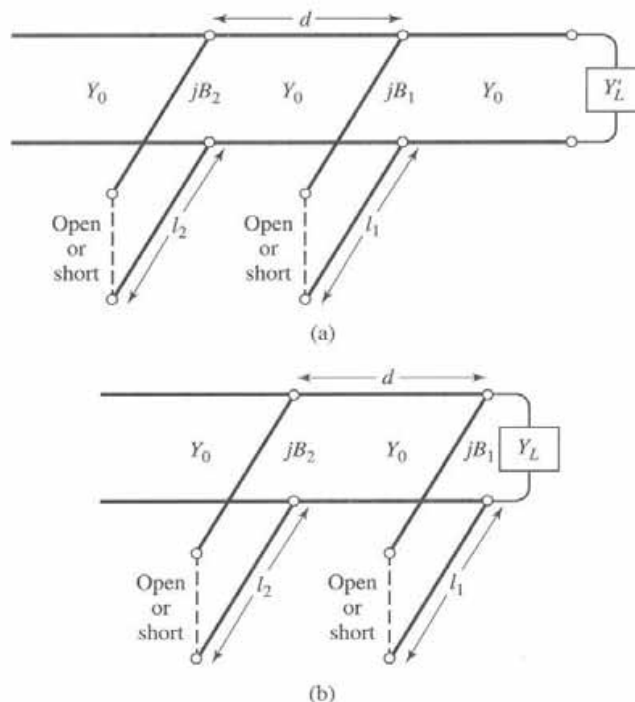
Answer:

Stub is a piece of transmission line used for impedance matching.

Disadvantage of Single stub matching:

It is a narrow band matching network.

Require a variable length of line between the load and the stub.



Steps for Double stub matching using smith chart:

1. The susceptance of the first stub, b_1 (or b_1'), moves the load Y_L to y_1 (or y_1').
2. These points lie on the rotated $1+jb$ circle; The circle whose amount of rotation from $1+jb$ circle is d wavelengths toward the load.
3. Transforming y_1 (or y_1') toward the generator through a length, d .
4. The resultant admittance y_2 (or y_2') must be on $1+jb$ circle.
5. The second stub then adds a susceptance to cancel the imaginary part of y_2 (or y_2').

Q.6 b. Representing general unmodulated wave $x = A \sin(\omega t + \phi)$ draw the basic modulation wave forms.

Answer: Page Number 82 of Text Book

Q.7 a. For a wave propagating in a parallel-plane wave guide define cut-off wavelength, group and phase velocity and also derive the relation between them.

Answer:

Cutoff wavelength

In a plane wave if a second wall is added to the first at a distance a from it, then, it must be placed at a point where the electric intensity due to the first wall is zero, i.e., at an integral number of half-wavelength away. Putting this mathematically, we have

$$a = \frac{m\lambda_n}{2}$$

Where a = distance between walls

λ_n = Wavelength in a direction normal to both walls

m = Number of half –wavelengths of electric intensity to be established between the walls (an integer)

Substituting for $\lambda_n = \frac{\lambda}{\cos \theta}$ gives

$$a = \frac{m(\lambda / \cos \theta)}{2} = \frac{m\lambda}{2 \cos \theta}$$

$$\cos \theta = \frac{m\lambda}{2a}$$

λ_p is the wavelength of the traveling wave which propagates down the waveguide. We have

$$\lambda_p = \frac{\lambda}{\sin \theta} = \frac{\lambda}{\sqrt{1 - \cos^2 \theta}} = \frac{\lambda}{\sqrt{1 - (m\lambda / 2a)^2}}$$

Cutoff wavelength is defined as the smallest free-space wavelength that is just unable to propagate, from above equation

$$1 - \left(\frac{m\lambda_o}{2a} \right)^2 = 0$$

$$\frac{m\lambda_o}{2a} = 1$$

$$\lambda_o = \frac{2a}{m}$$

where λ_o = cutoff wavelength.

Group and phase velocity in the waveguide

A wave reflected from a conducting wall has two velocities in a direction parallel to the wall, namely, the group velocity and the phase velocity. The former was shown as v_g and the latter as v_p . These two velocities have exactly the same meanings in the parallel-plane waveguide and must now be correlated and extended further.

$$\lambda_p = \frac{\lambda}{\sqrt{1 - (m\lambda/2a)^2}} = \frac{\lambda}{\sqrt{1 - [\lambda(1/\lambda_o)]^2}}$$

$$\lambda_p = \frac{\lambda}{\sqrt{1 - (\lambda/\lambda_o)^2}}$$

$$v_g v_p = v_c \sin \theta \frac{v_c}{\sin \theta}$$

$$v_g v_p = v_c^2$$

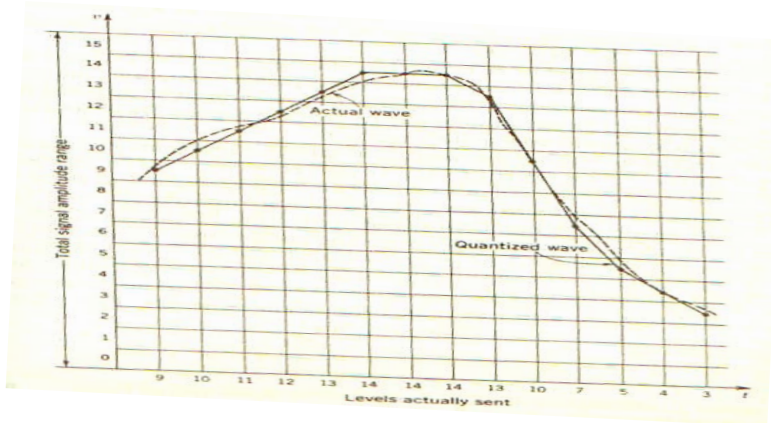
Thus the product of the group velocity and the phase velocity of a signal propagating in a waveguide is the square of the velocity of light in free space. Note that, in free space, phase and group velocities exist also, but they are then equal. It is now possible to calculate the two velocities in terms of the cutoff wavelength, again obtaining universal equations. we have

$$\begin{aligned} v_p &= f\lambda_p \\ &= f \frac{\lambda}{\sqrt{1 - (\lambda/\lambda_o)^2}} \\ &= \frac{v_c}{\sqrt{1 - (\lambda/\lambda_o)^2}} \end{aligned}$$

Substituting gives

$$\begin{aligned} v_g &= \frac{v_c^2}{v_p} = v_c^2 \frac{\sqrt{1 - (\lambda/\lambda_o)^2}}{v_c} \\ v_g &= v_c \sqrt{1 - \left(\frac{\lambda}{\lambda_o} \right)^2} \end{aligned}$$

- Q.8** a. Explain the working of PCM system and also mention the selection of number of bits in PCM system.

Answer:

Pulse-code modulation (PCM) is a digital representation of an analog signal where the magnitude of the signal is sampled regularly at uniform intervals, then quantized to a series of symbols in a numeric (binary) code. PCM has been used in digital telephone systems.

It is not necessary to transmit the exact amplitudes of the samples because the final receiver can detect only finite intensity differences. This means that the message signal can be approximated to a certain extent.

The quantization noise (or error) reduces with the step-size. But then, to cover a given range of signal values, we would need more levels. The quantization noise is the mean-square value of the individual quantization errors. For a sinusoidal waveform, the quantization noise is given by $(S^2 / 12)$, where 'S' is the step-size.

Trade-off:

We can reduce quantization error by reducing the step size, but it results in increasing the number of quantization levels and number of bits used to represent these levels. As the number of bits per sample increases the bandwidth also increases.

Hence in communication the selection of number bits is the compromise between bandwidth and quantization error. In practice 8 bit PCM is used.

- c. Calculate the capacity of a standard 4 kHz telephone channel with signal to noise ratio of 32 dB.

Answer:

$$(S/N)_{\text{ratio}} = \text{Antilog}(\text{SNR in db}/10) = \text{Antilog}(32/10) = 1585$$

$$C = \delta f \log_2(1 + S/N) = 3100 \times \log_2(1 + 1585)$$

$$= 3100 \times \log_2 1586 = 3100 \times 10.63$$

$$= 32,953 \text{ bits per second}$$

- Q.9** a. With neat diagram explain the channel translating equipment showing the formation of a basic 12-channel group B.

Answer:

The basic group consists of 12 adjacent 4-Khz channels, occupying the frequency range from 60 to 108 KHz. A low -level pilot is transmitted at 104.08 KHz, for regulating and monitoring purpose. Figure (a) shows a channel arrangement for a basic group B.

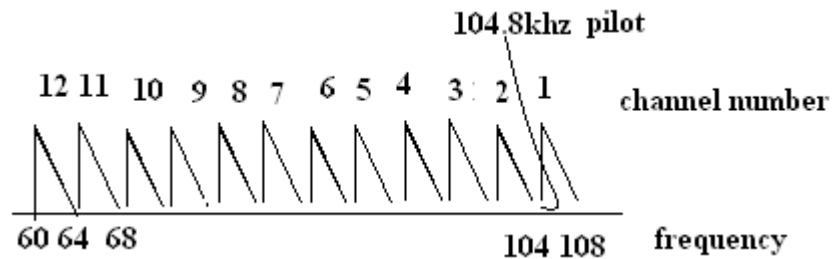


Fig (a): channel arrangement in basic group B

Figure (b) shows a simplified block diagram of channel translation equipment and shows how a basic group is assembled. The process is a repetitive one of producing adjacent lower sidebands, with a frequency separation of 900 hz between the channels.

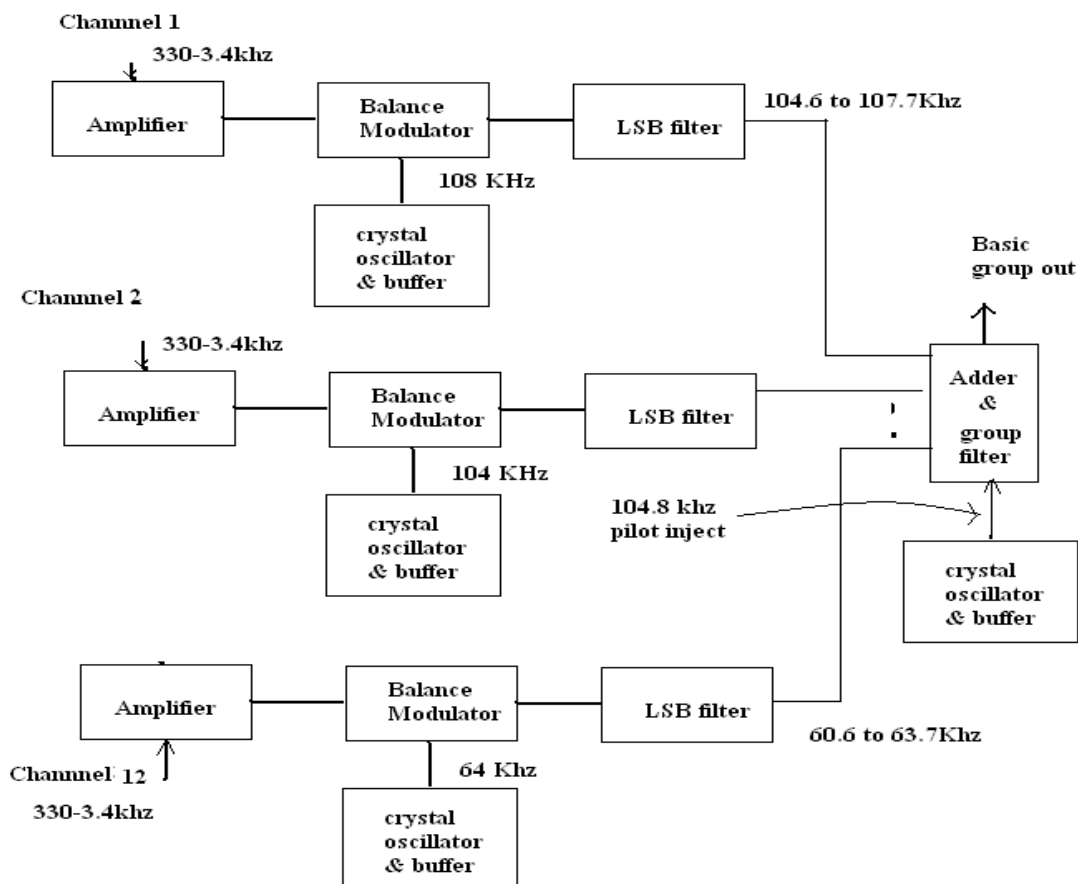


Figure (b): Channel translating equipment showing the formation of a basic 12 channel group B

Text Book

Electronic Communication Systems, George Kennedy and Bernard Davis, Fourth Edition (1999), Tata McGraw Hill Publishing Company Ltd.