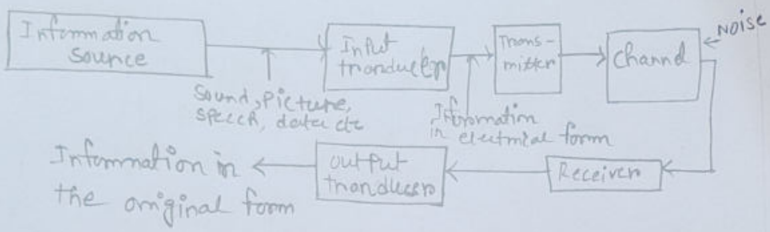
1) ⓐ Draw the block diagram of a Communication system and briefly explain the function of each block.



Block diagram of a Communication system

Explanation of the function of each block:

1) Information Source:- The function of information source is to Produce required message which has to be transmitted.

2) Input Tranducer:- It is used to convert the original message signal into a time varying electrical signal.

3) Transmitter :- The function of transmitter is to process the electrical signal from different aspects. For example in radio broadcasting the electrical signal obtained from Sound signal, is processed to restrict its range of audio frequencies and is often

amplifies modulation is the main functi-
on of it.

4) Channel :- The function of channel is to provide a physical connection between the transmitter and receiver, there are two types of channel point is point of broadcast channel.

5) Receiver:- the main function of the receiver is to reproduce the message signal in electrical form from the distorted received signal.

6) Destination:- It is the final stage which is used to convert an electrical message signal into its original from.

(b) what is base band transmission?

Baseband transmission is transmission of the encoded signal using its own baseband frequencies, i.e. without any shift to higher frequency ranges. It is used for short distances. The message signal generated from the information source is known as baseband signal. This baseband signal may be a combination of 2 or more message signals. If the baseband signal is transmitted directly, then it is known as baseband transmission.

The baseband signal cannot usually be transmitted through space by radio because the antennas required are too long and multiple baseband signals transmitting simultaneously would interfere with one another.

(c) Explain the need for modulation in a communication system.

We need modulation for the following reason-

(i) Practicality of antenna:- We know that in case when free space is used as transmitting medium (i.e. channel) messages are transmitted and received with the help of antennas. For efficient radiation & reception the transmitting & receiving antennas must have lengths comparable to a quarter wavelength of the frequency used. For example in Am broadcast systems, the maximum audio frequency transmitted from a radio station is of orders of 5KHz. If this message audio signal is transmitted without modulation, then the height of the antenna required for an effective radiation & reception will be 1/4th of the wavelength given as,

$$l = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times5\times10^3} = 5\,Km.$$

Obviously, it will be totally impracticable to construct and install an antenna of such a height. However this height of the antenna may be reduced by modulation technique and yet effective radiation & reception is achieved. In modulation process audio signal at radio stations are translated to higher frequency spectrum, i.e. radio frequency range. These higher radio frequencies with the small wavelength act as for audio frequency (i.e. modulating signal). Thus the height of the antenna required is much reduces and becomes practical.

As an example, if an audio frequency is translated to a radio frequency carrier of frequency 3 MHz, the antenna height required would be,

$$l = \lambda/4 = \frac{c}{4f} = \frac{3\times10^8}{4\times3\times10^6} = 25m$$

This antenna height may be achieved practically.

(ii) To remove interference:- We know that the frequency range of audio signal is from 20Hz to 20KHz. In radio stations. In case there is no modulation all these stations transmitt audio 20Hz to 20KHz. Due to this transmission over same range the programmes of different station will get mixed up.

Hence in order to keep the various signals seperate, it is necessary to translate or shift them to different portions of the electromagnetic spectrum. Thus each station is allocated a band of frequency. This also overcomes the drawback of poor radiation efficiently at low frequency.

(iii) Reduction of noise:- Noise is the major limitation of any communication. Although noise can not be eliminated completely, but with the help of several modulation technique, effect of noise can be minimized.

2/(a) What is meant by the term amplitude modulation?

Amplitude modulation may be defined as the process in which amplitude of the carrier wave is varied according to the instantaneous value (amplitude of the modulating or baseband signal). It is a process by which the wave signal is transmitted by modulating the amplitude of the signal. It is often called Am and is commonly used in transmitting a piece of information through a radio carrier wave.

(b) Define modulation index for Am wave.
It is also known as Amplitude sensitivity. In Am signal the modulation index is defined as the measure of extent of amplitude variation about an unmodulated maximum carrier. It is represented by ma.

mathematically,

$$\text{modulation Index } m_a = \frac{|x(t)|_{max}}{\text{max carrier amplitude}}$$

or, $$\text{Modulation Index } m_a = \frac{|x(t)|_{max}}{A}$$

where $|x(t)|_{max}$ represents the maximum amplitude of modulating signal and A represents the maximum amplitude of carrier signals.

The modulation index is also known as depth of modulation degree of modulation or modulation factor.

The absolute value of ma multiplied by 100 is known as percentage modulation.

① Derive an expression for single-tone amplitude modulated wave.

Let us consider a single tone modulating signal [In single tone modulating signal, the amplitude modulation in which the modulating or baseband signal consist of only one (single) frequency, ie modulation is done by a single frequency or tone. This type amplitude modulation is known as single tone amplitude modulation] as,

$$x(t) = V_m \cos w_m t \quad \text{--- (1)}$$

which contains a single frequency w_m.

Let the carrier signal be,

$$c(t) = A \cos w_c t \quad \text{--- (2)}$$

we known that general expression for Am signal is $s(t) = [A + x(t)] \cos w_c t$

$$\text{or, } s(t) = A\cos w_c t + x(t) \cos w_c t \quad \text{--- (3)}$$

Putting the value of $x(t)$ in equation (3) we get, $s(t) = A\cos w_c t + V_m \cos w_m t \cos w_c t$

$$= A\left[1 + \frac{V_m}{A} \cos w_m t\right] \cos w_c t \quad \text{--- (4)}$$

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But we know that for Am, the modulation index ma is given as $m_a = \frac{|x(t)|_{max}}{A}$.

where $|x(t)|_{max}$ denotes the maximum amplitude of modulation signal and A is the maximum amplitude of carrier signal.

In this case we have,

$$|x(t)|_{max} = V_m$$

Therefore, $m_a = \frac{V_m}{A}$

Putting this value of ma in equation (4) we get,

$$s(t) = A[1 + m_a \cos\omega_m t]\cos\omega_c t \quad \text{—— (5)}$$

This is the desired expression for single-tone modulated signal.

The expression in equation (5) may be further simplifies to observe the frequency components present in Am signal.

$$s(t) = A\cos\omega_c t + Am_a \cos\omega_c t \cos\omega_m t$$

$$\text{or, } s(t) = A\cos\omega_c t + \frac{Am_a}{2}[2\cos\omega_c t \cos\omega_m t]$$

$$\text{or, } s(t) = A\cos\omega_c t + \frac{Am_a}{2}\cos(\omega_c + \omega_m) + \frac{Am_a}{2}\cos(\omega_c - \omega_m) \quad \text{—(6)}$$

Equation (6) reveals that the Am signal has three components as follow -

i) carrier frequency ω_c having amplitude A

ii) Upper sideband $(\omega_c + \omega_m)$ having amplitude $\frac{m_a A}{2}$

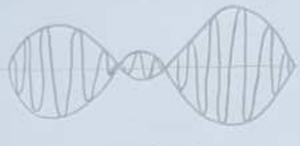
iii) Lower sideband $(\omega_c - \omega_m)$ having amplitude $\frac{m_a A}{2}$.

(d) Draw the waveform of an overmodulated Am wave and write the condition for over modulation.

Amplitude modulated signals are basically in three types -

(i) Under modulated
(ii) Over modulated
(iii) Prefect modulated

Overmodulation :- when modulation index (m_a) is greater than 1, i.e. $m_a = \frac{|m(t)|_{max}}{A} > 1$ i.e amplitude of message signal is greater than amplitude of carrier signal.



ma >> 1

3) (a) what is carson's rule?

Carson's rule :- It is also known as practical bandwidth. It provides a thumb formula to calculate the BW of single tone WBFM. According to this rule FM BW is given as twice the sum of the frequency deviation and highest modulating frequency. However it must be remembered that the rule is just an approximation. mathematically,

$$BW = 2(\Delta\omega + \omega_m)$$
$$= 2\omega_m\left(1 + \frac{\Delta\omega}{\omega_m}\right)$$
$$= 2\omega_m(1 + m_f)$$

Regarding the carson's rule, we can consider two special cases -

(i) $BW = 2(1+m_f)\omega_m$

If $\Delta\omega << \omega_m$, i.e., $m_f << 1$ as is the case for NBFm, then

$BW = 2\omega_m$

(ii) If $\Delta\omega >> \omega_m$, i.e., $m_f >> 1$ as is the case for WBFm, then

$BW = 2(m_f)\omega_m$
$= 2\Delta\omega$

(b) Explain the generation of narrowband Fm signals with suitable block diagram.

we known that the expressions for narrowband Fm and narrowband pm are given as by,

$s(t)_{Fm} = A\cos\omega_c t - Ak_f\, y(t)\sin\omega_c t$ —① &

$s(t)_{pm} = A\cos\omega_c t - Ak_p\, x(t)\sin\omega_c t$ —②

the above equation for Fm signal (eqn ①) Provide an idea to generate narrowband Fm. This means that the sideband terms are generated by a balanced modulator as in a DSB-sc system & then the carrier term is added to the sideband terms. Based upon the above equation we can show the method of generation of narrowband Fm and narrowband pm. It may be observed that the block diagram shown in Fig satisfy the corresponding equation for narrowband Fm and the block diagram is below-

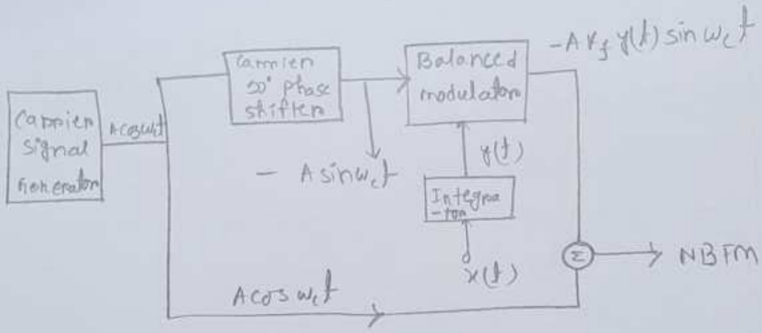


Fig:- Narrowband Fm Generator

(Q) A baseband or modulating signal $x(t) = 5\cos(2\pi \times 1000t)$ angle modulates a carrier $A\cos(\omega_c t)$.

i) Determine the modulation index and bandwidth for Fm system. ii) Find the change in the bandwidth and modulation index for Fm if modulating frequency is reduced to 5KHz. Assume $K_f = 15$ KHz/volt.

i) we know that,

$$m_f = \frac{\Delta\omega}{\omega_m} = \frac{K_f A_m}{\omega_m}$$

[K_f = Frequency sensitivity
A_m = max amplitude of modulating signal
ω_m = modulating frequency]

$$= \frac{15\times10^3 \times 5}{2\pi\times1000}$$

[$\because x(t) = A_m \cos\omega_m t$
$\therefore A_m = 5,\ \omega_m = 2\pi\times1000$]

$$= 11.936$$

we know that, according to the carson's rule,

$$B.\omega = 2(\omega_m + \Delta\omega)$$

[$\Delta\omega$ = Frequency deviation (Angular)]

$$= 2(2\pi\times1000 + 15\times10^3\times5)$$
$$= 162.566\times10^3 \text{ HZ}$$
$$= 162.566 \text{ KHZ}$$

$$B.\omega_l = 2(f_m + \Delta f) = 2(1000 + 15\times10^3\times5) = 152 \text{ KHZ}$$

[f_m = Angul. linear F.m. signal]
[$\because \omega_m = 2\pi f_m$]
[Δf = linear freq. deviation]

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(ii) Now, $m_f'' = \frac{\Delta\omega}{\omega_m} = \frac{k_f A_m}{\omega_m}$

$= \frac{15\times10^3 \times 5}{5\times10^3}$ [$\because$ given, $\omega_m = 5\times10^3 Hz$]

$= 15$

Now, $B\cdot\omega' = 2(\omega_m + \Delta\omega)$

$= 2(5\times10^3 + 5\times15\times10^3)$

$= 160\times10^3\ Hz$

$= 160 KHz$

$B\cdot\omega_l' = 2(f_m + \Delta f)$ [$\because f_m = \frac{\omega_m}{2\pi} = \frac{5\times10^3}{2\pi} = 795.77 Hz$]

$= 2(795.77 + 5\times15\times10^3)$

$= 151.59 KHz$

$\therefore$ the change in B·ω (Angular) $= (162.566 - 160)$ KHz

$= 2.566$ KHz

the change in $B\cdot\omega_l$ (linear) $= (152 - 151.59)$ KHz

$= 0.41 KHz$

the change in modulation index,

$m_f = m_f' - m_f''$

$= 11.936 - 15$; $= -3.064$

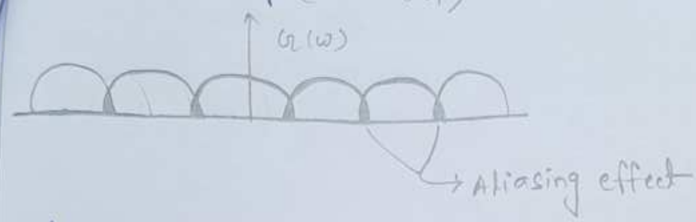
So, we can conclude that $B.W_l$ (linear Bandwidth) & B.W (Angular Bandwidth) is decreased by 0.41 KHz & 2.566 KHz respectively. The modulating index (m_f) is increased by 3.064.

1) (a) State sampling theorem.

A bandlimited continuous-time signal of finite energy may be completely represented in its samples and recovered back if the sampling frequency is $f_s \geq 2f_m$. Here f_s is the sampling frequency & f_m is the maximum frequency present in the signal. The sampling theorem states that a signal can be exactly reproduced if it is sampled at a frequency F, where F is greater than twice the maximum frequency in the signal.

(b) What is aliasing? How is it prevented?

Aliasing is the effect of overlapping frequency components resulting from unsufficiently large sample rate. In other words, it causes appearance of frequencies in the amplitude-frequency spectrum, that are not in the original signal. Aliasing effect is the effect of under sampling $(f < 2f_m)$



It can be prevented by following way -
we can:

i) ensure it is in the $[0, \frac{f_s}{2})$ range (e.g. through low-pass filtering) or

ii) Increase the sample rate.

This implies that we should know what range our signal is in before we sample it.

Remember: after aliasing creeped into the sampled signal, it is impossible to eliminate.

Anti-aliasing filter can overcome the aliasing effect.

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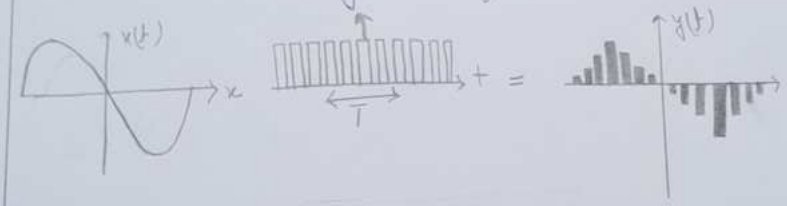
(3) Explain flat top sampling technique.

During transmission, noise is introduced at top of the transmission pulse which can be easily removed if the pulse is in the form of flat top. Here, the top of the samples are flat, i.e they have constant amplitude. Hence, it is called as flat top sampling or practical sampling. Flat top sampling makes use of sample and hold circuit. Here width of pulse is finite. Here, 2 switch are used (discharge switch, sampling switch).



Theoretically, the sampled signal can be obtained by convolution of rectangular pulse $p(t)$ with ideally sampled signal say $y_s(t)$ as shown in the diagram:

i.e, $y(t) = p(t) \times y_s(t)$ ------ (1)

To get the sampled spectrum, consider Fourier transform on both sides for equation (1)

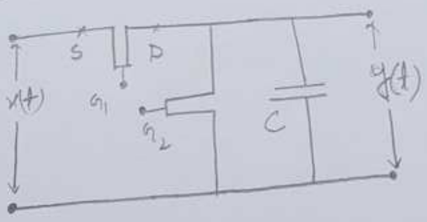
$Y[\omega] = F.T[p(t) \times y_s(t)]$

By the knowledge of convolution property

$Y[\omega] = P(\omega)\, y_s(\omega)$

Here, $P(\omega) = T\,Sa\left(\frac{\omega T}{2}\right) = 2\sin\omega T/\omega$

Sampling circuit,



(d) Find the Nyquist rate and Nyquist interval for the signal $x(t) = \cos(400\pi t)\cos(100\pi t)$.

$$x(t) = \cos(4000\pi t)\cos(1000\pi t)$$
$$= \frac{1}{2}\left[2\cos(4000\pi t)\cos(1000\pi t)\right]$$
$$= \frac{1}{2}\left[\cos 5000\pi t + \cos 3000\pi t\right] \quad \text{--- (I)}$$

So, there are 2 frequencies present be ω_1 & ω_2.

$$x(t) = \frac{1}{2}\left[\cos\omega_1 t + \cos\omega_2 t\right] \quad \text{--- (II)}$$

Comparing eqn (I) & (II) we get,

$\omega_1 = 5000\pi$	$\omega_2 = 3000\pi$
$\Rightarrow f_1 = \frac{5000\pi}{2\pi} = 2500\,Hz$	$\Rightarrow f_2 = \frac{3000\pi}{2\pi} = 1500\,Hz$

f_m = maximum frequency present in the signal

$\therefore$ we can see, $f_m = f_1 = 2500\,Hz$

$\therefore$ Nyquist rate, $f_s = 2f_m$
$= 2\times 2500$
$= 5000\,Hz$

Nyquist interval $= T_s = \frac{1}{2f_m} = \frac{1}{2\times 2500} = \frac{1}{5000}$
$= 2\times10^{-4}\,s$

5) (a) How is an analog signal converted to a digital signal in PCM system?

In the method of a PCM Generator or Transmitter we can see how to convert an analog signal to a digital signal, from a practical point of view. Fig. shows a practical block diagram of a PCM generator.

In PCM generator of fig. the signal $x(t)$ is first passed through the low-pass filter (LPF) of cut-off frequency f_m Hz. This low-pass filter blocks all the frequency components which are lying above f_m Hz.

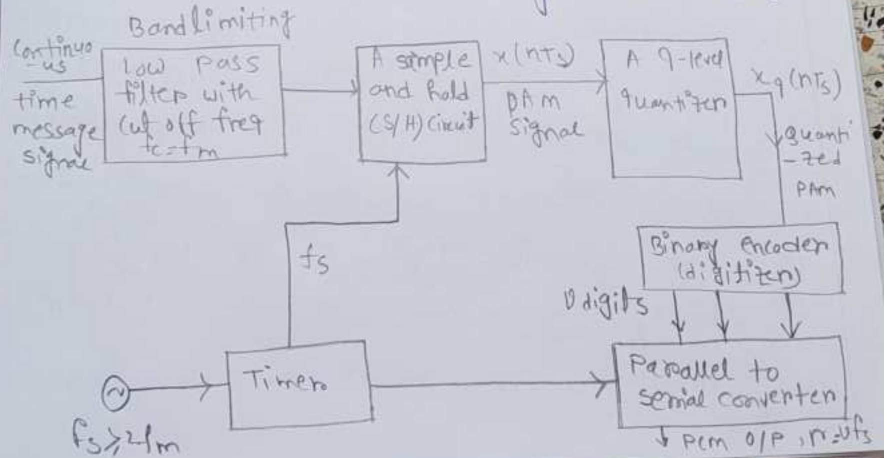


Fig: A Practical PCM generator

This means that now the signal $x(t)$ is bandlimited to f_m Hz. The Sample & hold circuit then samples this signal at a rate of f_s Hz. Sampling frequency f_s is selected sufficiently above nyquist rate to avoid aliasing i.e, $f_s \geq 4_m$

In fig., the output of sample and hold circuit is denoted by $x(nT_s)$. This signal $x(nT_s)$ is discrete in time and continuous in input $x(nT_s)$ with its fixed digital levels. It assigns any one of the digital level to $x(nT_s)$ which results in minimum distortion or error. Thus, output of quantizer is a digital level called $x_q(nT_s)$. Now the quantized signal level $x_q(nT_s)$ is passed through binary encoder. This encoder converts input signal to v digits binary word. Thus $x_q(nT_s)$ is converted to 'v' binary bits. This encoder is also known as digitizer.

Also an oscillator generates the clocks for S/H circuit and parallel-to-serial converter. In PCM generator S/H circuit, quantizer & encoder combinely form an analog-to-digital converter(ADC).

(b) Derive the expression for transmission bandwidth in a PCM system.

In this section, we shall evaluate the transmission bandwidth for PCM system. Let us assume that the quantizer use 'v' number of binary digits to represent each level. Then, the number of levels that may be represented by 'v' digits will be

$$q = 2^v$$

Here, 'q' represents total numbers of digital levels of a q-level quantizer.

For example if 'v' = 4 bits, the total number of levels will be

$$q = 2^4 = 16 \text{ levels.}$$

Each Sample is converted to 'v' binary bits, the number of bits per sample = v
We know that, number of samples per second = f_s.
Therefore' no of bits per second is expressed as (number of bits/second)
= (number of bits per samples) x (number of samples per second)
= v bits per sample x f_s samples per second
As a matter of fact, the number of bits per second is known as signalling rate of PCM and is denoted by 'r' i.e.
signaling rate in PCM $r = vf_s$
where, $f_s \geq 2f_m$
Also, since bandwidth needed for PCM transmission is given by half of the signalling rate.
'Therefore' transmission bandwidth in PCM

$BW \geq \frac{1}{2} r$

But, $r = v f_s$

Therefore, $BW \geq \frac{1}{2} v f_s$

Again since $f_s \geq 2f_m$

Hence, $BW \geq v f_m$

This is required expression for bandwidth of a PCM system.

© A television signal having a bandwidth of 4.2 mHz is transmitted using binary PCM system. Given that numbers of quantization level is 512. Determine transmission bandwidth and output signal to quantization noise ratio.

We know that the transmission channel bandwidth is given by,

$BW \geq vf_m \geq 9 \times 4.2 \times 10^6 \text{ Hz} \geq 37.8 \text{ mHz}$ [given, $f_m = 4.2$ mHz]

[we know,
$q = 2^v$
i.e., $512 = 2^v$
$\Rightarrow \log_{10} 512 = \log_{10} 2^v$

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$\Rightarrow v = \frac{\log_{10} 512}{\log_{10} 2} = 9$

$\therefore v = 9$ bits]

the o/p signal to quantization noise ratio is expressed as,

$\left(\frac{S}{N}\right) dB \leq 4.8 + 6v\ dB$

But $v = 9$

'Therefore' $\left(\frac{S}{N}\right) dB \leq 4.8 + 6 \times 9$

$\Rightarrow \left(\frac{S}{N}\right)_{dB} \leq 58.8\ dB$

[where, v = no. of binary bits]