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Internal Assesment Test - II

Sub:	Communication Theory							Code:	21EC44
Date:	07/ 08/ 2023	Duration:	90 mins	Max Marks:	50	Sem:	4^{th}	Branch:	ECE
Answer Any FIVE FULL Questions									

		Marks	OBE CO	OBE RBT
1.	Derive the equation for frequency modulated wave considering single tone modulating signal. Define frequency deviation and modulation index	[10]	CO1	L2
2.	Explain Wide band FM with relevant expressions. Plot the spectrum for any particular case of modulation index.	[10]	CO1	L2
3.	Find the carrier, modulating frequency, modulation index and maximum frequency deviation of a FM wave represented by s(t) =12sin (6×10^8 t+5 sin1250t) volts. What power will this FM wave dissipate in a 10Ω resistor? Plot the sample spectrum of the given FM signal.	[10]	CO2	L3
4	State and prove Sampling theorem	[10]	CO4	L2
5	With neat circuit diagram and relevant frequency response plots explain demodulation of FM by balanced slope detector	[10]	CO2	L2
6	Write the expression of single tone wideband FM modulated signal. Explain the two methods for calculating the transmission bandwidth of FM. An FM signal, has sinusoidal modulation with $\boldsymbol{fm}$=$\mathbf{15kHz}$ and modulation index $\boldsymbol{\beta}$=5.Using Carson's rule, find the transmission bandwidth.	[10]	CO2	L3
7	Explain in detail using relevant block diagrams FM stereo Multiplexing	[10]	CO2	L2
8	What is multiplexing? What are the different types of Multiplexing? Explain TDM with a neat block diagram.	[10]	CO4	L2

1. FM single tone

→ Frequency Modulation is a process of altering the frequency of carrier signal in accordance with the instantaneous values of message signal by keeping amplitude & phase of carrier constant.

Time domain expression:-

- Let the instantaneous value of carrier signal is

$$c(t) = A_c \cos 2\pi f_c t \longrightarrow (1)$$

- Let the instantaneous value of message signal is

$$m(t) = A_m \cos 2\pi f_c t \longrightarrow (2)$$

- We know that the standard equation of Angle modulated wave is given by, $S(t) = A_c \cos \theta_i(t) \longrightarrow (3)$

where $\theta_i(t)$ = Angle of FM wave (modulated wave)

- We know that the instantaneous frequency $f_i(t)$ of FM signal is given by $f_i(t) = f_c + k_f m(t)$ —— (4)

where, k_f = frequency sensitivity

$m(t)$ = message signal

- We know that the Angular frequency,

$$\omega_i(t) = \frac{d}{dt}\theta_i(t)$$

$$\Downarrow$$

$$2\pi f_i(t) = \frac{d}{dt}\theta_i(t)$$

$$\therefore f_i(t) = \frac{1}{2\pi}\frac{d}{dt}\theta_i(t)$$

—— (5)

Substitute $f_i(t) = f_c + k_f m(t)$ in equation (5) we get,

$\therefore \quad f_c + k_f m(t) = \frac{1}{2\pi}\frac{d}{dt}\theta_i(t)$

$\therefore \quad \frac{d\,\theta_i(t)}{dt} = 2\pi f_c + 2\pi k_f m(t) \quad —(6)$

⇒ Apply Integral on both sides of equation (6) we get

$$\int \frac{d\,\theta_i(t)}{dt} = \int [2\pi f_c + 2\pi k_f m(t)]\, dt$$

$$\Downarrow$$

$$\theta_i(t) = 2\pi f_c t + 2\pi k_f \int m(t)\,dt \quad —(7)$$

∴ The General equation of FM signal is

$S(t) = A_c \cos\theta_i(t)$ — using equation (7)

$$\boxed{S(t) = A_c \cos[2\pi f_c t + 2\pi k_f \int m(t)\,dt]} \quad —(8)$$

Equation (8) is the general equation of FM signal for any message signal $m(t)$.

for, $m(t) = A_m \cos 2\pi f_m t$

$$\int m(t)\,dt = \int A_m \cos 2\pi f_m t\, dt \qquad \left(\because \int \cos mx\,dx = \frac{\sin mx}{m}\right)$$

$$= \frac{A_m}{2\pi f_m} \cdot \sin 2\pi f_m(t) \quad —(9)$$

∴

$$S(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \times \frac{A_m}{2\pi f_m} \cdot \sin(2\pi f_m t)\right]$$

$$= A_c \cos\left[2\pi f_c t + \frac{k_f A_m}{f_m} \sin 2\pi f_m t\right]$$

$$\boxed{S(t) = A_c \cos[2\pi f_c t + \beta \sin 2\pi f_m t]} \quad —(10)$$

Equation (10) is the standard equation of FM signal for $m(t) = A_m \cos 2\pi f_m t$; where $\beta = \frac{k_f A_m}{f_m} = \frac{\Delta f_{max}}{f_m}$ ⇐ Modulation Index of FM signal.

3. PLL Nonlinear model

FM-demodulation using phase Locked Loop:- (PLL)

phase Locked Loop (PLL) is a negative feedback system that consists of three major components

<i> A Multiplier used as a phase detector (or) phase comparator.

<ii> A - voltage Controlled oscillator <VCO>

<iii> A - Loop filter, which is a Low pass filter <LPF>

The Block diagram of PLL is shown in Fig.1.

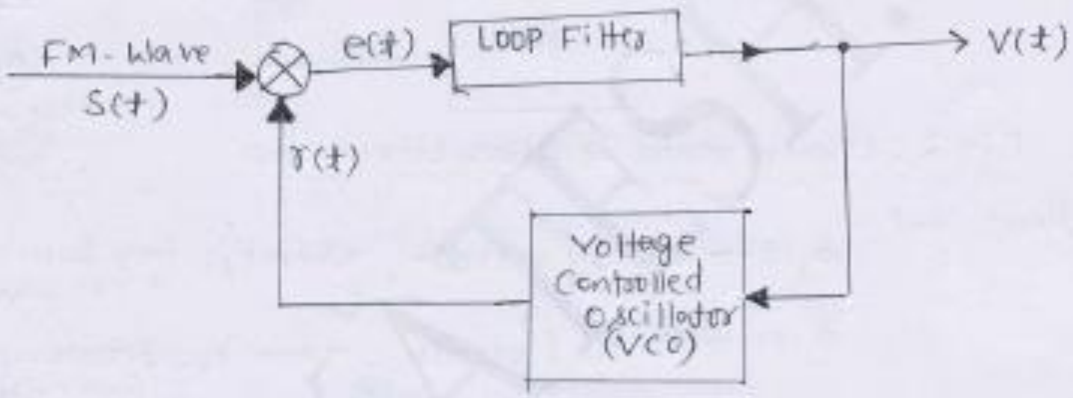

Fig.1: Block diagram of PLL

↳ The VCO output is defined as

$$r(t) = A_v \cos(2\pi f_c t + \phi_2(t)) \quad —① $$

where $\phi_2(t) = 2\pi K_v \int_0^t v(t)\,dt$.

↳ Then, the incoming signal (FM) (S(t)) and the VCO output r(t) are applied to the multiplier, then it gives error signal,

$$e(t) = r(t) \cdot S(t) \quad —② $$

where $S(t) = A_c \sin[2\pi f_c t + \phi_1(t)] \quad —③$

where $\phi_1(t) = 2\pi K_f \int_0^T m(t)\,dt \quad —④$

2A).

$$s(t) = 10sin\left(5 * 10^8 t + 4sin\,(1250t)\right)$$

Comparing with basic FM equation

$$s(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]$$

a) Modulation Index = 4

b) Carrier frequency = 79.6MHz

c) Frequency Deviation =index*fm =4*199 =796Hz

d) Power generated across 5ohm load =(Ac^2)/2R = 50/50 =1W

2B) fm=15kHz, deviation =50kHz

BT = 2(deviation+fm) =130kHz

4. FM stereo

stereo multiplexing is a form of frequency division multi--plexing <FDM> designed to transmit two separate signals [$m_R(t)$ and $m_L(t)$] via the same carrier.

FM-stereo system consists of

a> FM-stereo transmitter

b> FM-stero Receiver.

FM-stereo transmitter :-

- Let $m_L(t)$ and $m_R(t)$ denote the two message signals picked up by Left hand and Right hand microphones at the transmi--tting end of the system, as shown in Fig. 1.
- It uses a pilot carrier frequency, $f_c = 19$ KHZ. Frequency doubler - produces, sub carrier, $\cos(4\pi f_c t)$.

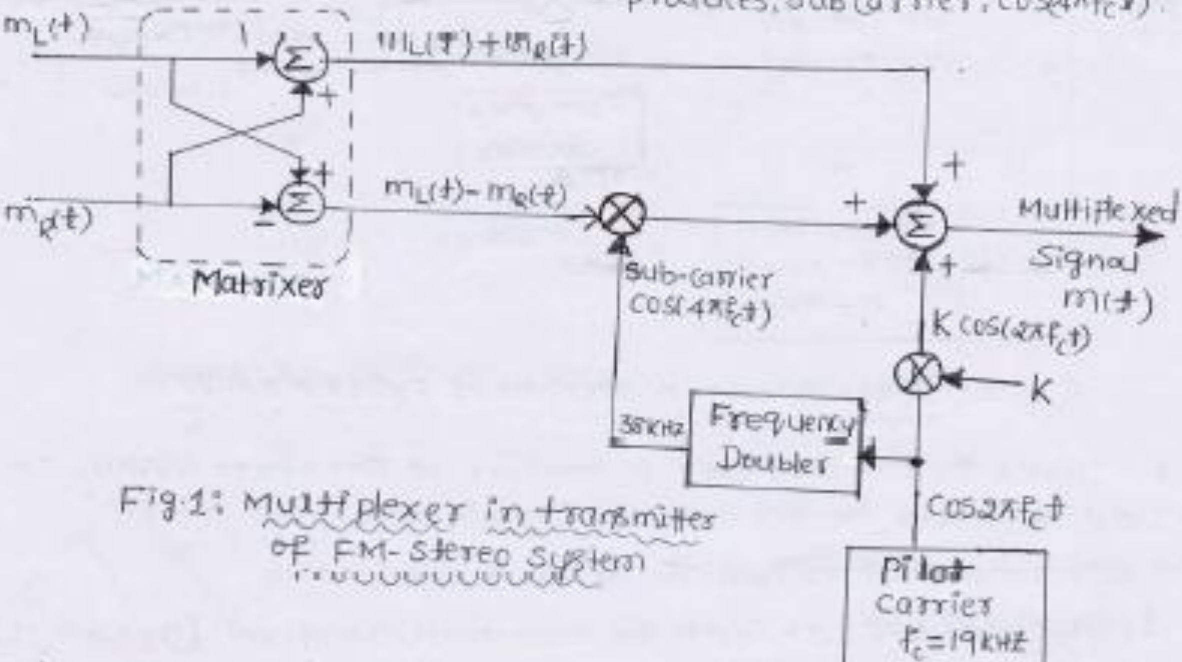

Fig 1: Multiplexer in transmitter of FM-Stereo System

- The Multiplexed signal $m(t)$, produced at the output of multiplexer in transmitter of FM stereo system is,

$$m(t) = \underbrace{[m_L(t) + m_R(t)]}_{\text{Baseband signal ①}} + \underbrace{[m_L(t) - m_R(t)]\cos 4\pi f_c t}_{\substack{\text{DSBSC ② Signal} \\ 2f_c = 38\text{KHZ}}} + \underbrace{K\cos(2\pi f_c t)}_{\substack{\text{Pilot ③ Carrier Signal.} \\ f_c = 19\text{KHZ}}} \quad \text{--- (1)}$$

- Multiplexed signal, m(t) consists of three different signals (28)

1. $[m_L(t)+m_R(t)] \Rightarrow$ sum of $m_L(t)$, $m_R(t)$ generated by the simple matrixer. It is baseband signal.
2. $[m_L(t)-m_R(t)]\cos(4\pi f_c t) \Rightarrow$ DSBSC-signal produced by the product modulator.
3. $K\cdot\cos(2\pi f_c t) \Rightarrow$ pilot carrier signal multiplied by a constant 'K'.

FM-stereo Receiver :-

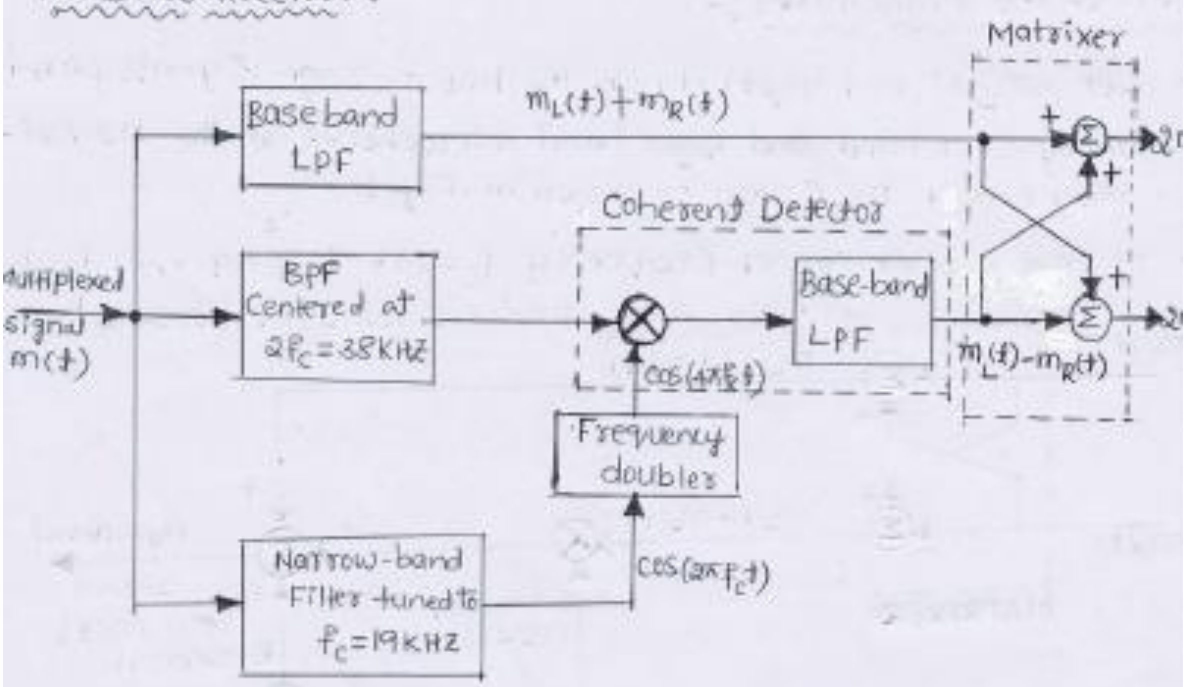

Fig 2: Demultiplexer in receiver of FM-stereo system

Fig. 2 shows the demultiplexer in receiver of FM-stereo system. It is used to recover the two message signals $m_L(t)$ and $m_R(t)$.

FM-stereo demultiplexer consists of 3-filters,

1. Baseband LPF : It selects the base-band component $[m_L(t)+m_R(t)]$ present in multiplexed signal m(t).
2. BPF : (Bandpass filter) :- It selects the DSBSC-signal.
3. Narrowband filter :- It selects the pilot carrier signal, $\cos(2\pi f_c t)$.

Frequency doubler produces the required subcarrier signal, $\cos(4\pi f_c t)$ for coherent detection of DSBSC-signal.

Coherent Detector, recovers the difference signal $[m_L(t)-m_R(t)]$.

Finally the Matrixer, produces the required signals $2m_L(t)$ and $2m_R(t)$.

6)

Statement: Sampling theorem states that any continuous time signal can be completely represented in its samples and recovered back if the sampling frequency is greater than or equal to twice the highest frequency component of base band signal.

That is Sampling frequency, $f_s \geq 2W$.

Where W= Highest frequency in base band continuous time signal.

This condition is also called Nyquist condition for sampling process.

Explanation and Proof:

* Consider an arbitrary signal g(t) of finite energy.

* Let '$g_\delta(t)$' denote the signal obtained by individually weighting the elements of a periodic sequence spaced T_s seconds. Therefore, sampled output $g_\delta(t)$ is given by,

$$g_\delta(t) = g(t) \cdot \delta_\delta(t) \quad \text{——— (1)}$$

* Let $\delta_\delta(t)$ denote the periodic impulse train and is represented as,

$$\delta_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \quad \text{——— (2)}$$

Substituting Eq (2) in Eq (1) we get

$$g_\delta(t) = g(t) \cdot \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

Using shifting property of impulse function

WKT, $g(t) \cdot \delta(t - nT_s) = g(nT_s)\,\delta(t - nT_s)$

$$\therefore \boxed{g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\,\delta(t - nT_s)} \quad \text{——— (3)}$$

For frequency domain consider,

$$g_\delta(t) = g(t) \cdot \delta_\delta(t)$$

Taking Fourier Transform on both sides, we get

$$G_\delta(f) = G(f) * \delta_\delta(f) \quad \text{——— (4)}$$

where,

$$S_\delta(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \qquad \text{(5)}$$

substituting Eqⁿ (5) in Eqⁿ (4) we get.

$$G_\delta(f) = G(f) * f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

From convolution property of impulse function

w.k.t, $G(f) * \delta(f - nf_s) = G(f - nf_s)$

$$\therefore \quad G_\delta(f) = f_s \sum_{n=-\infty}^{\infty} G(f - nf_s) \qquad \text{(6)}$$

Eqⁿ (6) can be rewritten as,

$$G_\delta(f) = f_s\, G(f) + f_s \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} G(f - nf_s) \qquad \text{(7)}$$

When the spectrum of $G_\delta(f)$ is passed through an LPF, then the 2^{nd} term in RHS of Eqⁿ (7) is eliminated resulting in

$$G_\delta(f) = f_s \cdot G(f)$$

$$\therefore \quad G(f) = \frac{1}{f_s} \cdot G_\delta(f) \qquad \text{(8)}$$

where $f_s = 2W$

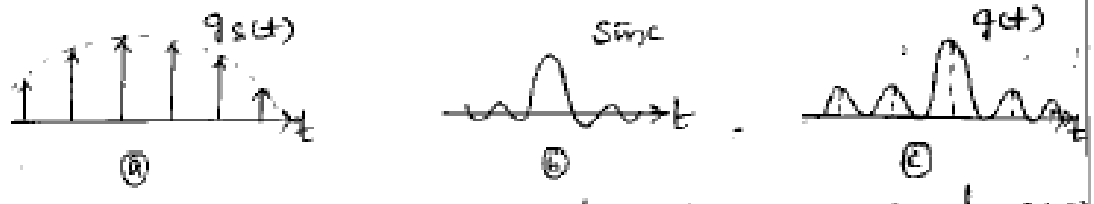

Fig : Recovering $g(t)$ signal from sequence of samples $g(nT_s)$.

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

1) A bandlimited 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 $\frac{1}{2W}$ seconds.

2) A bandlimited signal of finite energy, which only has frequency components less than "W" Hertz, may be completely recovered from a knowledge of its samples 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.

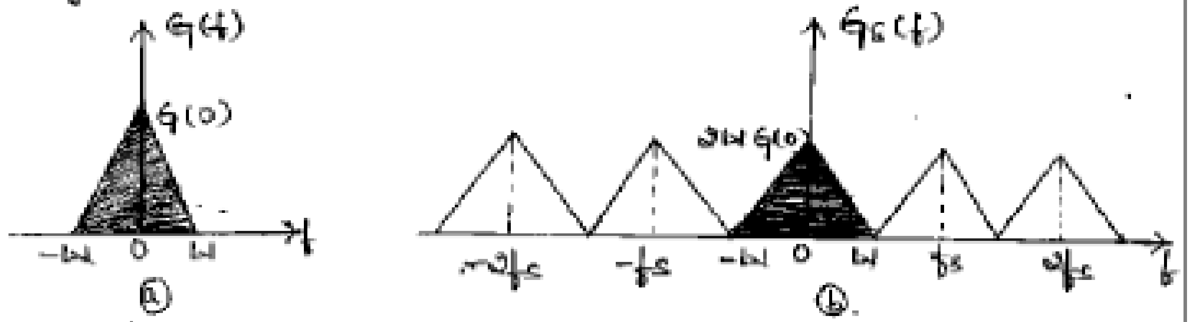

Fig : (a) Spectrum of a strictly band limited signal g(t).
(b) Spectrum of a sampled version of g(t) for $T_s = \frac{1}{2W}$.

7)b

- Less sensitive to noise.
- It is easier to integrate different services

    - video and the accompanying soundtrack, into the same transmission scheme.
- The transmission scheme can be relatively independent of the source.
- Circuitry for handling digital signals is easier to repeat
- Digital circuits are less sensitive to physical effects
    - such as vibration and temperature.
- Digital signals are simpler to characterize , this makes the associated hardware easier to design.
- Easy to Implement techniques like
        - Multiplexing
- Channel compensation
    - Equalization
    - Error correction
- There are techniques for removing redundancy from a digital transmission, so as to minimize the amount of information that has to be transmitted.
- Digital techniques make it easier to specify complex standards that may be shared on a worldwide basis.
- This allows the development of communication components with different features and their interoperation with a different component produced by a different manufacturer.

7 A

* TIME DIVISION MULTIPLEXING : [TDM]

Time Division Multiplexing is a method of transmitting and receiving independent signals over a common channel by means of synchronised switches at each end of transmission line so that each signal appears on the line only a fraction of time in an alternating pattern.

* fig(5) shows the block diagram of TDM system.

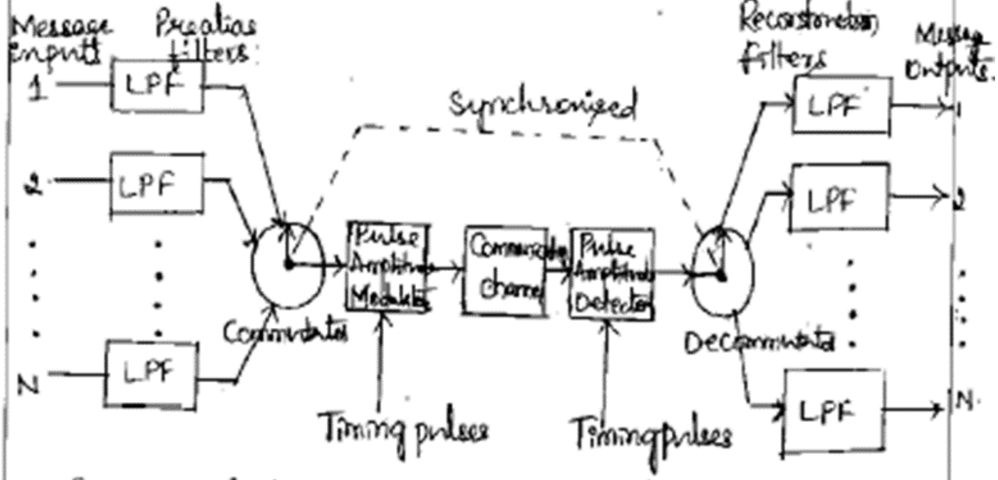

Fig 5 : Block Diagram of TDM system.

* The concept of TDM is illustrated in the fig(5). The Lowpass filters are used to remove high frequency components present in the message signal. The output of the pre-alias filters are then fed to a commutator, which is usually implemented using electronic switching circuitory.

* The function of commutator is as follows:

i) To take a narrow sample of each of the 'N' samples of input at a rate of $f_s \geq 2W$.

ii) To sequentially interleave (multiplex) these 'N' samples inside a sampling interval $T_s = 1/f_s$.

* The multiplexed signal is then applied to a pulse amplitude modulator whose purpose is to transform the multiplexed signal into a form suitable for transmission over a common channel.

* At the receiving end, the pulse amplitude demodulator performs the reverse operation of PAM and the decommutator distributes the signals to the appropriate low pass reconstruction filters. The decommutator operates in synchronisation with the commutator.