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SCHOOL OF ELECTRICAL AND ELECTRONICS ENGINEERING

DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

UNIT – I - Basics and Amplitude Modulation – SEC1304

I. Basics and Amplitude Modulation

Electromagnetic Spectrum and Communication Applications

The electromagnetic spectrum is the range of frequencies (the spectrum) of electromagnetic radiation and their respective wavelengths and energies. The electromagnetic spectrum covers electromagnetic waves with frequencies ranging from below one hertz to above 1025 hertz, corresponding to wavelengths from thousands of kilometres down to a fraction of the size of an atomic nucleus. This frequency range is divided into separate bands, and the electromagnetic waves within each frequency band are called by different names; beginning at the low frequency (long wavelength) end of the spectrum these are: radio waves, microwaves, infrared, visible light, ultraviolet, X-rays, and gamma rays at the high-frequency (short wavelength) end.

Table 1.1 Frequency spectrum

Frequency band	Name
3–30 kHz	Very low frequency (VLF)
30–300 kHz	Low frequency (LF)
300–3000 kHz	Medium frequency (MF)
3–30 MHz	High frequency (HF)
30–300 MHz	Very high frequency (VHF)
0.3–3 GHz	Ultrahigh frequency (UHF)
3–30 GHz	Superhigh frequency (SHF)
30–300 GHz	Extremely high frequency (EHF)
43–430 THz	Infrared (0.7–7 μm)
430–750 THz	Visible light (0.4–0.7 μm)
750–3000 THz	Ultraviolet (0.1–0.4 μm)

Elements of Communication Systems

The basic components of a communication system are information source, input transducer, transmitter, communication channel, receiver, output transducer, and destination.

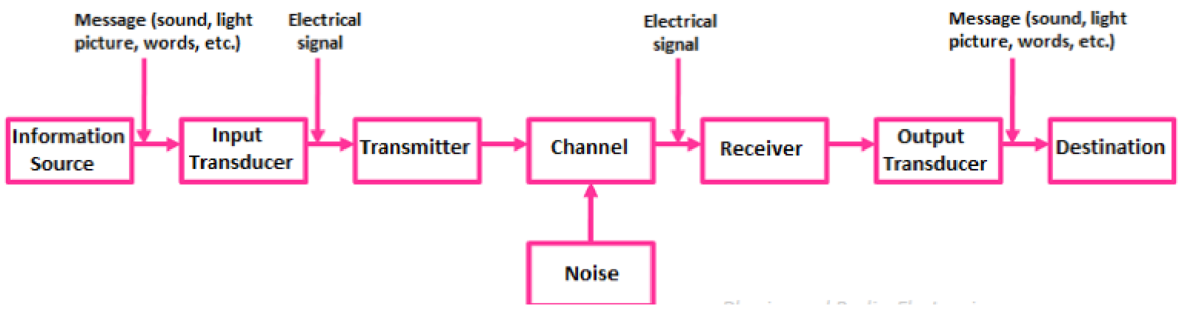


Fig 1.1 Basic Elements of Communication System

Information Source

The information generated by the source may be in the form of sound (human speech), picture (image source), words (plain text in some particular language such as English, French, German etc.) The message is the part of a communication which involves sending information from source to destination. Information is a meaningful data that the receiver consumes.

Input Transducer

A transducer is a device which converts one form of energy or signal into another form of energy or signal. The transducer is present at the input side and output side of the communication system. The transducer that is present at the input side of the communication system is called input transducer. Generally, the input transducer converts the non-electrical signal (sound signal or light signal) into an electrical signal. The best example of an input transducer is the microphone which is placed between the information source and the transmitter section. A microphone is a device which converts your voice signals (sound signals) into electrical signals.

Transmitter

The transmitter is a device which converts the signal produced by the source into a form that is suitable for transmission over a given channel or medium. Transmitters use a technique called modulation to convert the electrical signal into a form that is suitable for transmission over a given channel or medium. Modulation is the main function of a transmitter.

Communication Channel

The communication channel is a wired or wireless medium through which the signal (information) travels from source (transmitter) to destination (receiver).

Noise

Noise is an unwanted signal that enters the communication system via the communication channel and interferes with the transmitted signal. The noise signal (unwanted signal) degrades the transmitted signal (signal containing information).

Receiver

The receiver is a device that receives the signal (electrical signal) from the channel and converts the signal (electrical signal) back to its original form (light and sound) which is understandable by humans at the destination.

Output Transducer

The transducer that is present at the output side of the communication system is called output transducer. Generally, the output transducer converts the electrical signal into a non-electrical signal (sound signal, light signal, or both sound and light signal).

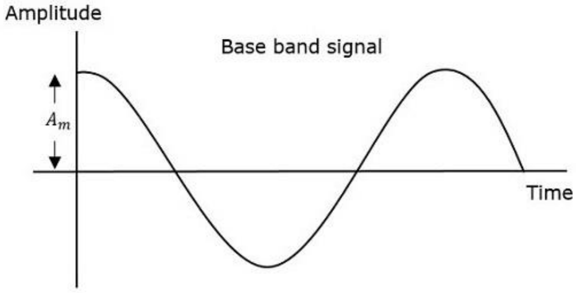
Need for Modulation

The primary purpose of modulation in a communication system is to generate a modulated signal which is well suited to the characteristics of transmission medium. The need for modulation is listed as follows:

- To reduce the antenna height
- To overcome hardware system limitations
- To reduce the interference, noise & distortions made when we transmit the signals with nearly same frequency in the audio frequency range (20-20k) Hz.
- To multiplex the more number of signals
- To the assignment of channel frequency
- To narrow banding the signal
- To reduce the complexity of the transmission system
- To increase the bandwidth of the signal

Amplitude Modulation

The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal." Which means, the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant. This can be well explained by the following figures.



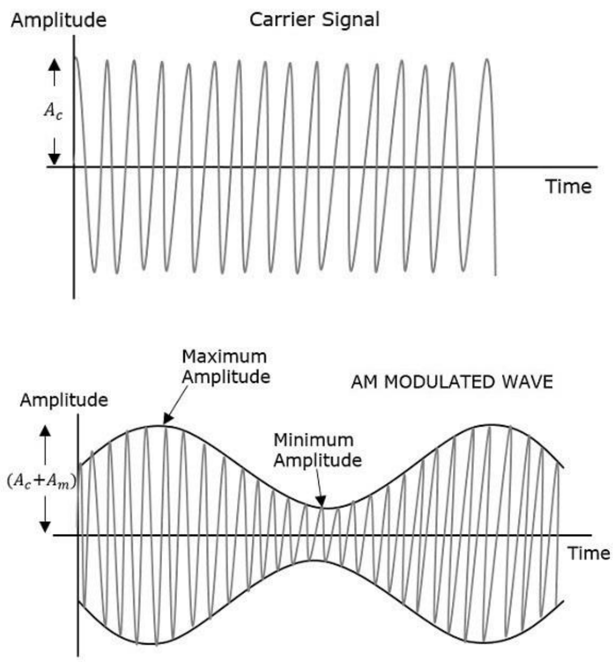


Fig 1.2 Amplitude Modulation

The first figure shows the modulating wave, which is the message signal. The next one is the carrier wave, which is a high frequency signal and contains no information. While, the last one is the resultant modulated wave.

It can be observed that the positive and negative peaks of the carrier wave, are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as Envelope. It is the same as that of the message signal.

Mathematical Expressions

Following are the mathematical expressions for these waves.

Time-domain Representation of the Waves

Let the modulating signal be,

$$m(t)=Am \cos(2\pi fmt)$$

and the carrier signal be,

$$c(t)=Ac \cos(2\pi fct)$$

Where,

Am and Ac are the amplitude of the modulating signal and the carrier signal respectively.

fm and fc are the frequency of the modulating signal and the carrier signal respectively.

Then, the equation of Amplitude Modulated wave will be

$$s(t)=[Ac+Am\cos(2\pi fmt)]\cos(2\pi fct)$$

Modulation Index of AM

A carrier wave, after being modulated, if the modulated level is calculated, then such an attempt is called as Modulation Index or Modulation Depth. It states the level of modulation that a carrier wave undergoes.

Rearrange the Equation 1 as below.

$$s(t)=Ac[1+(AmAc)\cos(2\pi fmt)]\cos(2\pi fct)$$

$$s(t)=Ac[1+\mu\cos(2\pi fmt)]\cos(2\pi fct) \text{ (Equation 2)}$$

Where, μ is Modulation index and it is equal to the ratio of Am and Ac. Mathematically, we can write it as

$$\mu=Am/Ac \text{ (Equation 3)}$$

Hence, we can calculate the value of modulation index by using the above formula, when the amplitudes of the message and carrier signals are known.

The modulation index or modulation depth is often denoted in percentage called as Percentage of Modulation. We will get the percentage of modulation, just by multiplying the modulation index value with 100.

For a perfect modulation, the value of modulation index should be 1, which implies the percentage of modulation should be 100%.

For instance, if this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure. It is called as Under-modulation. Such a wave is called as an under-modulated wave.

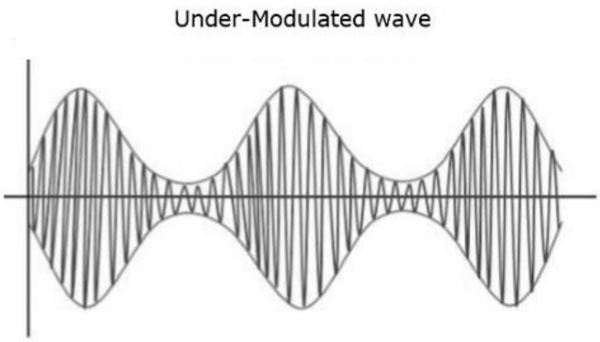


Fig 1.3 Under-Modulated Wave

If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an over-modulated wave. It would look like the following figure.

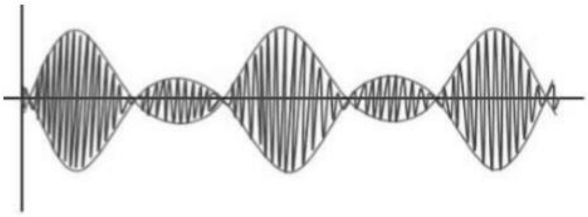


Fig 1.4 Over-Modulated Wave

As the value of the modulation index increases, the carrier experiences a 180o phase reversal, which causes additional sidebands and hence, the wave gets distorted. Such an over-modulated wave causes interference, which cannot be eliminated.

Frequency Spectrum

AM Power Distribution in AM_DSB_FC

Consider the following equation of amplitude modulated wave.

$$s(t) = A_c \cos(2\pi f_c t) + \frac{A_c \mu}{2}\cos[2\pi(f_c + f_m)t] + \frac{A_c \mu}{2}\cos[2\pi(f_c - f_m)t]$$

Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components.

$$P_t = P_c + P_{USB} + P_{LSB}$$

We know that the standard formula for power of cos signal is

$$P = \frac{{v_{rms}}^2}{R} = \frac{\left(v_m/\sqrt{2}\right)^2}{2}$$

Where,

vrms is the rms value of cos signal.

vm is the peak value of cos signal.

First, let us find the powers of the carrier, the upper and lower sideband one by one.

Carrier power

$$P_c = \frac{\left(A_c/\sqrt{2}\right)^2}{R} = \frac{{A_c}^2}{2R}$$

Upper sideband power

$$P_{USB} = \frac{(A_c\mu/2\sqrt{2})^2}{R} = \frac{A_c{}^2{}_\mu{}^2}{8R}$$

Similarly, we will get the lower sideband power same as that of the upper side band power.

$$P_{LSB} = \frac{A_c{}^2{}_\mu{}^2}{8R}$$

Now, let us add these three powers in order to get the power of AM wave.

$$P_t = \frac{A_c{}^2}{2R} + \frac{A_c{}^2{}_\mu{}^2}{8R} + \frac{A_c{}^2{}_\mu{}^2}{8R}$$

$$\Rightarrow P_t = \left(\frac{A_c{}^2}{2R}\right)\left(1 + \frac{\mu^2}{4} + \frac{\mu^2}{4}\right)$$

$$\Rightarrow P_t = P_c\left(1 + \frac{\mu^2}{2}\right)$$

We can use the above formula to calculate the power of AM wave, when the carrier power and the modulation index are known.

If the modulation index μ=1μ=1 then the power of AM wave is equal to 1.5 times the carrier power. So, the power required for transmitting an AM wave is 1.5 times the carrier power for a perfect modulation.

Generation of AM_DSB_SC using FET Balance Modulator

A balanced modulator is a device that modifies a signal, usually in the form of an amplitude modulated (AM) radio signal. It takes the original signal that has both sidebands and a carrier signal, and then modulates it so that only the sideband signals come through the output modulator. This creates a balanced signal, as there is less noise because the carrier signal has been removed.

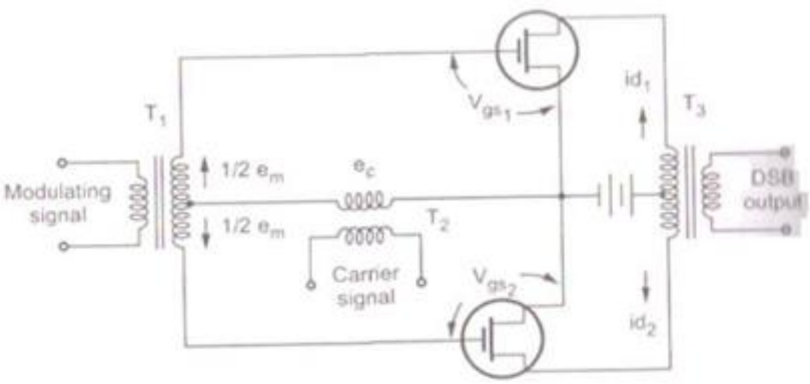


Fig 1.5 FET Balance Modulator

The balanced modulator can also be built using FETs. Figure shows the circuit diagram of balanced modulator using FETs. There are three transformers T1,T2 and T3. The carrier signal is applied to the center taps of the input transformer T1 and the output transformer T3 through the Transformer T1. The modulating signal is applied to the input transformer T1.The carrier signal is applied to the primary of transformer T2. This signal is further applied to two gates of FETs in phase through the secondary of T2. The modulating voltage appearing 180 degree out of phase at the gates, since these are the opposite ends of the center tapped transformer.

Consider that there is no modulating signal is applied. Then FET currents due to carrier signal are equal in amplitude but opposite in the directions. These opposite and equal currents are the primary of the output transformer cancel each other. Hence, no output is produced at the secondary of T3. Thus the carrier is suppressed.

When modulating signal is applied, the current id1 and id2 flow in the primary of T3 due to carrier signal as well as the modulating signal. The FET currents due to carrier are equal and opposite and cancel each other. Seems modulating signal is applied 180 degree out of phase at the gates, the FET currents due to modulating signal for equal but not opposite, hence do not cancel each other. Thus DSB output is produced by FET balanced modulator.

Generation of AM_SSB using Phase Shift Method

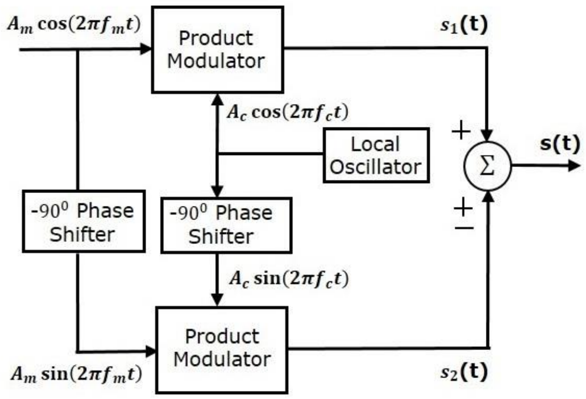


Fig 1.6 Generation of AM_SSB using Phase Shift Method

This block diagram consists of two product modulators, two -90^0 phase shifters, one local oscillator and one summer block. The product modulator produces an output, which is the product of two inputs. The -90^0 phase shifter produces an output, which has a phase lag of -90^0 with respect to the input.

The local oscillator is used to generate the carrier signal. Summer block produces an output, which is either the sum of two inputs or the difference of two inputs based on the polarity of inputs.

The modulating signal Am cos(2πfmt) and the carrier signal Ac cos(2πfct) are directly applied as inputs to the upper product modulator. So, the upper product modulator produces an output, which is the product of these two inputs.

The output of upper product modulator is

$$s_1(t) = A_m A_c \cos(2\pi f_m t)\cos(2\pi f_c t)$$

$$s_1(t) = \frac{A_m A_c}{2}\{\cos[2\pi(f_c + f_m)t] + \cos[2\pi(f_c - f_m)t]\}$$

The modulating signal Am cos(2πfmt) and the carrier signal Ac cos(2πfct) are phase shifted by 900 before applying as inputs to the lower product modulator. So, the lower product modulator produces an output, which is the product of these two inputs.

The output of lower product modulator is

$$s_2(t) = A_m A_c \cos\left(2\pi f_m t - 90^0\right)\cos\left(2\pi f_c t - 90^0\right)$$

$$s_2(t) = A_m A_c \sin(2\pi f_m t)\sin(2\pi f_c t)$$

$$s_2(t) = \frac{A_m A_c}{2}\{\cos[2\pi(f_c - f_m)t] - \cos[2\pi(f_c + f_m)t]\}$$

Add s1(t) and s2(t) in order to get the SSBSC modulated wave s(t)s(t) having a lower sideband.

$$s(t) = \frac{A_m A_c}{2}\{\cos[2\pi(f_c + f_m)t] + \cos[2\pi(f_c - f_m)t]\} +$$

$$\frac{A_m A_c}{2}\{\cos[2\pi(f_c - f_m)t] - \cos[2\pi(f_c + f_m)t]\}$$

$$s(t) = A_m A_c \cos[2\pi(f_c - f_m)t]$$

Subtract s2(t) from s1(t) in order to get the SSBSC modulated wave s(t) having a upper sideband.

$$s(t) = \frac{A_m A_c}{2}\{\cos[2\pi(f_c + f_m)t] + \cos[2\pi(f_c - f_m)t]\} -$$

$$\frac{A_m A_c}{2}\{\cos[2\pi(f_c - f_m)t] - \cos[2\pi(f_c + f_m)t]\}$$

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

Hence, by properly choosing the polarities of inputs at summer block, we will get SSBSC wave having a upper sideband or a lower sideband.

Generation of AM_SSB using Filter Method

This is the filter method of SSB suppression for the transmission. Fig

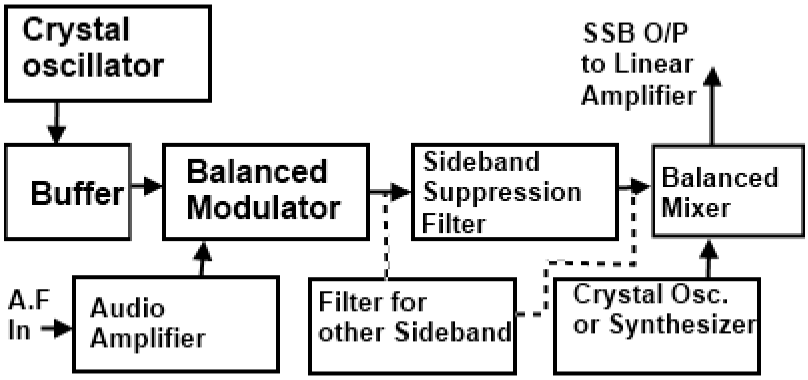


Fig 1.7 SSB Single Side Band Transmission Filter Method

A crystal controlled master oscillator produces a stable carrier frequency fc (say 100 KHz). This carrier frequency is then fed to the balanced modulator through a buffer amplifier which isolates these two stages.The audio signal from the modulating amplifier modulates the carrier in the balanced modulator. Audio frequency range is 300 to 2800 Hz. The carrier is also suppressed in this stage but allows only to pass the both side bands. (USB & LSB).A band pass filter (BPF) allows only a single band either USB or LSB to pass through it. This side band is then heterodyned in the balanced mixer stage with 12 MHz frequency produced by crystal oscillator or synthesizer depends upon the requirements of our transmission. So in mixer stage, the frequency of the crystal oscillator or synthesizer is added to SSB signal. The output frequency thus being raised to the value desired for transmission. Then this band is amplified in driver and power amplifier stages and then fed to the aerial for the transmission.

AM VSB

In case of SSB modulation, when a sideband is passed through the filters, the band pass filter may not work perfectly in practice. As a result of which, some of the information may get lost. Hence to avoid this loss, a technique is chosen, which is a compromise between DSB-SC and SSB, called as Vestigial Sideband (VSB) technique. The word vestige which means “a part” from which the name is derived. Vestigial Sideband Modulation or VSB Modulation is the process where a part of the signal called as vestige is modulated, along with one sideband. A VSB signal can be plotted as shown in the following figure.

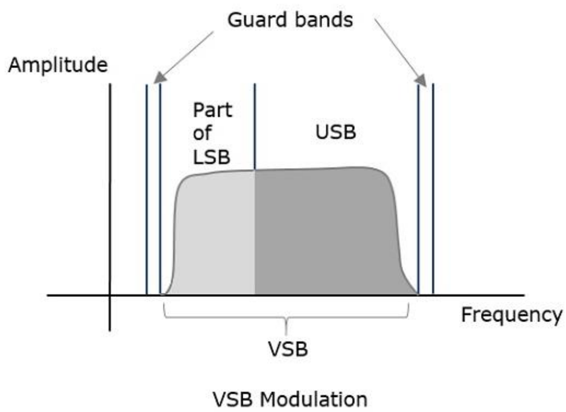


Fig 1.8 VSB Modulation

Along with the upper sideband, a part of the lower sideband is also being transmitted in this technique. A guard band of very small width is laid on either side of VSB in order to avoid the interferences. VSB modulation is mostly used in television transmissions.

Transmission Bandwidth

The transmission bandwidth of VSB modulated wave is represented as −

$$B=(f_{m}+ f_{v})\ Hz$$

Where,

fm = Message bandwidth

fv = Width of the vestigial sideband

Advantages

Following are the advantages of VSB −

- Highly efficient.
- Reduction in bandwidth.
- Filter design is easy as high accuracy is not needed.
- The transmission of low frequency components is possible, without difficulty.
- Possesses good phase characteristics.

Disadvantages

Following are the disadvantages of VSB −

- Bandwidth when compared to SSB is greater.
- Demodulation is complex.
- VSB Modulation − Application

The most prominent and standard application of VSB is for the transmission of television signals. Also, this is the most convenient and efficient technique when bandwidth usage is considered.

Comparison of AM Schemes

Table 1.2 Comparison of AM Schemes

Sr. No.	Parameter	SSB	DSB	ISB	VSB
1.	Power	less	Medium	High	High but less than ISB
2.	Bandwidth	f_m	$2 f_m$	$f_{m1}+f_{m2}$	f_m< Bw < $2f_m$
3.	Modulating inputs	1	1	2	1
4.	Use for	Radio communication	Radio communication	Telegraphy and telephony	Television
5.	Carrier suppression	Complete	Complete	Partly	No
6.	Sideband suppression	One sideband completely	No	One per channel	One sideband suppressed partially
7.	Transmission efficiency	Maximum	Moderate	Moderate	Moderate

Envelope Detector

The AM diode detector is an envelope detector – it provides an output of the envelope of the signal. As such the diode detector or demodulator is able to provide an output proportional to the amplitude of the envelope of the amplitude modulated signal.

The signal diode detector consists of two main elements to the circuit:

Diode / rectifier: The diode in the detector serves to that enhances one half of the received signal over the other. In many instances Schottky diodes are used for this form of detector, because signal levels may be low, and Schottky diodes have a much lower turn on voltage (typically around 0.2 V) than standard silicon diodes (typically around 0.7 or 0.7 V).

Low pass filter: The low pass filter is required to remove the high frequency elements that remain within the signal after detection / demodulation. The filter usually consists of a very simple RC network but in some cases It can be provided simply by relying on the limited frequency response of the circuitry following the rectifier. As the capacitor in the circuit stores the voltage, the output voltage reflects the peak of the waveform. Sometimes these circuits are used as peak detectors.

When selecting the value of the capacitor used int he circuit, it should be large enough to hold the peak of the RF waveform, but not so large that it attenuates any modulation on the signal, i.e. it should act as a filter for the RF carrier and not the audio modulation.

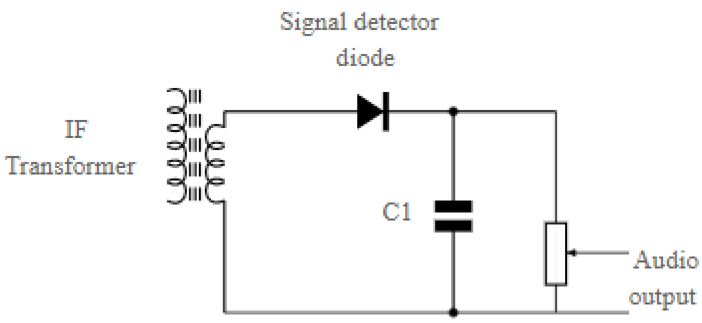


Fig 1.9 Circuit of an envelope detector as used in an AM radio receiver

The circuit typically has a relatively high source impedance. When linking the circuit to a following stage of the circuit, care should be taken not to land the detector too much otherwise the operation will be impaired.

Normally a resistor is placed across the capacitor - this may either be the load of the next stage, a volume control, or resistor in the circuit. This level of this should be determined by calculating the time constant of the capacitor and the load. This should be between the RF signal and audio modulation so that the RF is satisfactorily removed, but the audio modulation is left untouched.

It is worth noting in this circuit that the secondary of the transformer provides a DC return to ground. Sometimes when the AM signal detector is used using a capacitor connection tot he previous stage, then a resistor or choke (inductor) to ground must be used at the input so that a DC return path is provided. If not the circuit will not operate correctly.

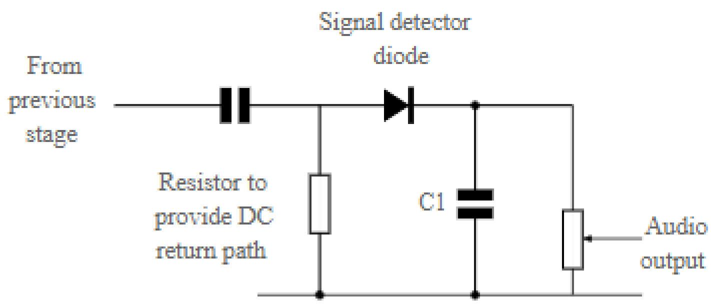


Fig 1.10 Capacitor coupled envelope signal detector showing resistor providing DC return path.

The value of the resistor on the input providing the DC return path is normally critical, but it can help provide the require match without absorbing too much signal.

AM diode detection process

In rectifying the RF signal, the AM diode detector provides an output equivalent to the envelope of one half of the signal, i.e. it is an envelope detector.

In view of the operation of the diode detector, it may sometimes be referred to as an envelope detector.

The incoming amplitude modulated RF signal consists of a waveform of both positive and negative going voltages as shown. Any audio transducer would not respond to this.

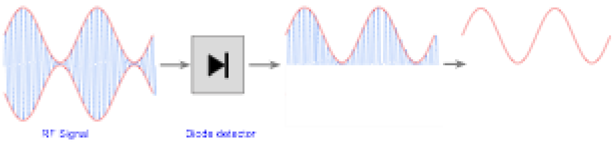


Fig 1.11 AM diode envelope detection process.

The diode envelope detector rectifies the waveform leaving only the positive or negative half of the waveform.

The high frequency element of this is then filtered out, typically using a capacitor which forms the low pass filter and effectively 'fills in' the high frequency elements, leaving a waveform to which a transducer like a pair of earphones or a loudspeaker could respond to and convert into sound waves.

Significance of RC time Constant in envelope Detector

By keeping the time constant, RC large, the capacitor discharging a small that is negligible hence spikes can be reduced.But the large values of RC create another problem called diagonal clipping. Hence we cannot increase it beyond the certain limit.

Choice of time constant RC

It is desired to keep the time constant RC very high as compared to time of carrier dev in order to minimize spikes fluctuation in detected envelope. On the other hand, if it is kept too high the discharge curve becomes approximately horizontal. In that case, negative pics of detected envelope baby completely or partially missing. Therefore the recovered baseband signal is distorted at negative peaks. This type of distortion is called diagonal clipping.

Diagonal clipping maybe caused

- When the time constant of detector is not selected properly
- Increase in depth of modulation index make the envelope slope steeper than the discharge path of capacitor
- To avoid diagonal clipping proper value of time constant may be obtained as follows
- During the non-conducting period of diode, the voltage across RC combination at an instant ―t‖ is given by

$$V_c(t) = V_0 e^{-t/R}$$

- The slope of capacitor discharge or rate of fall may be obtained by differentiating Vc(t) with respect to time t

$$\frac{dV_c(t)}{dt} = \frac{-V_c(t)}{RC}$$

- If the distortion is avoided, the decrease in capacitor voltage must follows envelope. We know the envelope of the modulated voltage signal which is given by VAM (t)= Vc[1+ ma sinωmt] sinωct

VAM=envelope voltage VAM= Vc[1+ ma sinωmt]

The slope of the envelope is given by

$$= \frac{dV_c}{dt} + \frac{d}{dt} V_c m_a sin m_m t = 0 + KV_c m_a m_m cos m_m t$$

Where K=proportionality constant

- The diagonal clipping is avoided, rate of fall for slope of capacitor is algebraically greater than or equal to slope of envelope.

TEXT / REFERENCE BOOKS

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Question Bank

Part A

1. Define modulation index of an AM signal
2. Draw the circuit diagram of an envelope detector
3. Draw the spectrum of AM.
4. Draw the spectrum of DSB-SC.
5. Define the transmission efficiency of AM signal.
6. Draw the phasor diagram of AM signal.
7. Advantages of SSB.
8. Disadvantages of DSB-SC.
9. Draw the spectrum of SSB
10. Distinguish between low level and high level modulator.

Part B

1. Explain the generation of AM signals using square law modulator.
2. Explain the detection of AM signals using envelope detector.
3. Explain about Balanced modulator to generate DSB-SC signal.
4. Explain the power distribution of DSB_FC AM wave.

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UNIT – II - Angle Modulation– SEC1304

II. Angle Modulation

Frequency Modulation

In amplitude modulation, the amplitude of the carrier signal varies. Whereas, in **Frequency Modulation (FM)**, the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.

Hence, in frequency modulation, the amplitude and the phase of the carrier signal remains constant. This can be better understood by observing the following figures.

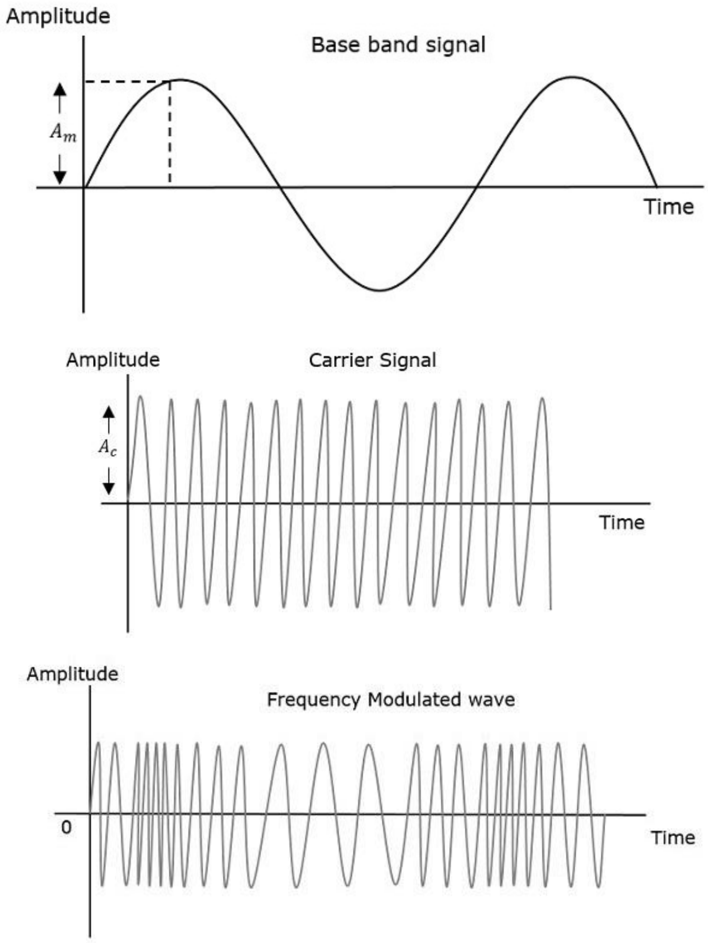


Fig 2.1 Frequency Modulation

The frequency of the modulated wave increases, when the amplitude of the modulating or message signal increases. Similarly, the frequency of the modulated wave decreases, when the amplitude of the modulating signal decreases. Note that, the frequency of the modulated wave remains constant and it is equal to the frequency of the carrier signal, when the amplitude of the modulating signal is zero.

Mathematical Representation

The equation for instantaneous frequency *fi* in FM modulation is

$$f_i = f_c + k_f m(t)$$

Where,

fc is the carrier frequency
kt is the frequency sensitivity
m(t) is the message signal
We know the relationship between angular frequency ωi and angle θi(t) as

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

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

$$\Rightarrow \theta_i(t) = 2\pi \int f_i dt$$

Substitute, f_i value in the above equation.

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

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

Substitute, $\theta_i(t)$ value in the standard equation of angle modulated wave.

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int m(t)\, dt\right)$$

This is the equation of FM wave.

If the modulating signal is m(t)=Am cos(2πfmt), then the equation of FM wave will be

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

Modulation Index of FM

The ratio of frequency deviation to the modulating frequency is knwn as the modulation index of FM.

$$\beta = \textit{modulation index} = \frac{\Delta f}{f_m} = \frac{k_f A_m}{f_m}$$

Frequency Deviation

The difference between FM modulated frequency (instantaneous frequency) and normal carrier frequency is termed as **Frequency Deviation**. It is denoted by Δf, which is equal to the product of k_f and A_m.

Deviation Ratio

Accordingly the FM deviation ratio can be defined as: the ratio of the maximum carrier frequency deviation to the highest audio modulating frequency.

m=Max frequency deviation/Max modulation frequency

Carson's Rule for Bandwidth of FM

This rule states that the bandwidth of an FM system is double the sum of the maximum frequency deviation and the highest modulating frequency *fm*. Thus, if B is the bandwidth of the system; then according to Carson's rule:

$$B=2(f_d + f_m)$$

Comparison of AM and FM

Table 2.1 Comparison of AM and FM

AM (Amplitude Modulation)	FM(Frequency Modulation)
The amplitude of the carrier wave is modified in order to send the data or information.	The frequency of the carrier wave is modified in order to send the data or information.
It works in a frequency range of 535 to 1705 Kilohertz (KHz).	It works in a frequency range of 88 to 108 Megahertz (MHz).
It has two sidebands.	It has an infinite number of sidebands.
In this method, the frequency and phase remain the same.	The amplitude and phase remain the same.
Its modulation index varies from 0 to 1.	Its modulation index is always greater than one.
It can transmit over long distances, have a large range.	It cannot transmit over long distances, have a smaller range.
AM based signal transmission consumes power than an equivalent FM based signal transmission.	FM based signal transmission consumes more power than an equivalent AM based signal transmission.
It is more susceptible to noise, has poor sound quality.	It is less susceptible to noise, has better sound quality.
It is more prone to signal distortion and degradation.	It is less prone to signal distortion and degradation.
In AM, if two or more signals are received at the same frequency, both are demodulated which causes interference.	In FM, if two or more signals are received at the same frequency, the receiver captures the stronger signal and eliminates the weaker one.
It has simple circuit design.	It has complex circuit design.
It is a less costly method.	It is more costly than AM.
It requires low bandwidth in the range of 10 kHz.	It requires high bandwidth in the range of 200 kHz.
It operates in the medium frequency (MF) and high frequency (HF).	It operates in the upper VHF and UHF range where noise effects are less.
Wastage of power is more as a major part of the power carried by the carrier wave does not contain the information.	No wastage of power as all transmitted power is carried by the information signal.
The received signal is of low quality.	The received signal is of high quality.

Narrow Band and Wideband FM Comparison

Narrowband FM

Following are the features of Narrowband FM.

- This frequency modulation has a small bandwidth when compared to wideband FM.
- The modulation index ββ is small, i.e., less than 1.
- Its spectrum consists of the carrier, the upper sideband and the lower sideband.
- This is used in mobile communications such as police wireless, ambulances, taxicabs, etc.

Wideband FM

Following are the features of Wideband FM.

- This frequency modulation has infinite bandwidth.
- The modulation index ββ is large, i.e., higher than 1.
- Its spectrum consists of a carrier and infinite number of sidebands, which are located around it.
- This is used in entertainment, broadcasting applications such as FM radio, TV, etc.

Generation of FM Using Varactor Diode Modulator (Direct Method)

- The varactor diode FM modulator has been shown below in figure .

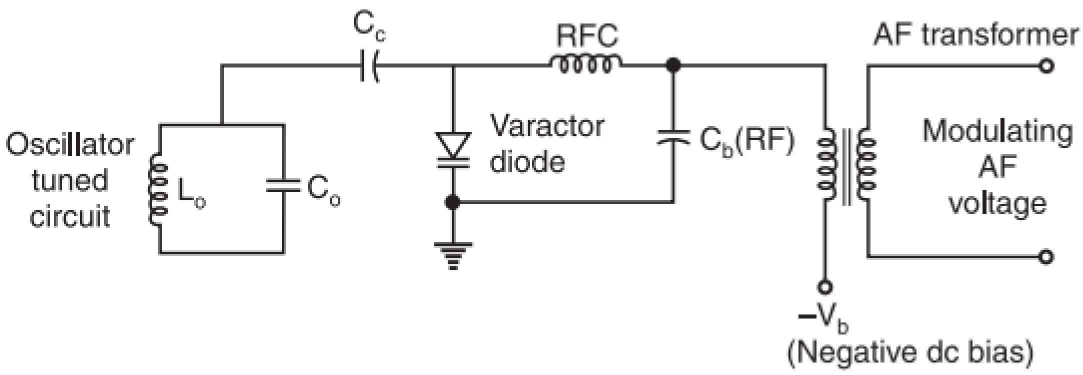


Fig 2.2 Varactor Diode Modulator

- A varactor diode is a semiconductor diode whose junction capacitance varies linearly with the applied bias and the varactor diode must be reverse biased.

  Working Operation

- The varactor diode is reverse biased by the negative dc source –Vb.

- The modulating AF voltage appears in series with the negative supply voltage. Hence, the voltage applied across the varactor diode varies in proportion with the modulating voltage.
- This will vary the junction capacitance of the varactor diode.
- The varactor diode appears in parallel with the oscillator tuned circuit.
- Hence the oscillator frequency will change with change in varactor dioide capacitance and FM wave is produced.
- The RFC will connect the dc and modulating signal to the varactor diode but it offers a very high impedance at high oscillator frequency. Therefore, the oscillator circuit is isolated from the dc bias and modulating signal.

Indirect Method of FM Generation

In the direct methods of generation of FM, LC oscillators are to be used. The crystal oscillator cannot be used. The LC oscillators are not stable enough for the communication or broadcast purpose. Thus, the direct methods cannot be used for the broadcast applications.

The alternative method is to use the indirect method called as the Armstrong method of FM generation.

In this method, the FM is obtained through phase modulation. A crystal oscillator can be used hence the frequency stability is very high and this method is widely used in practice.

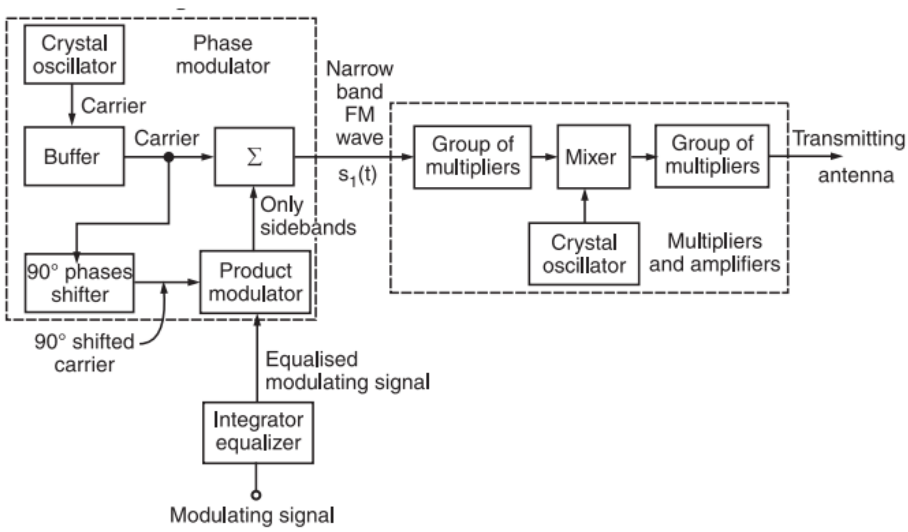


Fig 2.3 Armstrong Method for FM Generation

The Armstrong method uses the phase modulator to generate a frequency modulated wave.

Working Principle

The working operation of this system can be divided into two parts as follows:

Part I: Generate a narrow band FM wave using a phase modulator.

Part II: Use the frequency multipliers and mixer to obtain the required values of frequency deviation, carrier and modulation index.

Part I: Generate a narrow band FM using Phase Modulator

As discussed carrier, we can generate FM using a phase modulator.

The modulating signal x(t) is passed through an integrator before applying it to the phase modulator as shown in figure 1.

Let the narrow band FM wave produced at the output of the phase modulator be represented by s1(t) i.e.,

$$s_1(t) = V_{c1}\cos\left[\,2\pi f_1\, t + \phi_1(t)\,\right]$$

where Vc1 is the amplitude and f1 is the frequency of the carrier produced by the crystal oscillator.

The phase angle Φ1(t) of s1(t) is related to x(t) as follows:

$$\phi_1(t) = 2\pi k_1 \int_0^t x(t)\, dt$$

where k1 represents the frequency sensitivity of the modulator.

If Φ1(t) is very small then,

$$\cos\left[\phi_1(t)\right] \cong 1 \quad \text{and} \quad \sin\left[\phi_1(t)\right] \cong \phi_1(t)$$

Hence, the approximate expression for s1(t) can be obtained as follows:

$$\begin{aligned} s_1(t) &= V_{c1}\cos\left[2\pi f_1\, t + \phi_1(t)\right] \\ &= V_{c1}\left[\cos(2\pi f_1\, t)\cos\phi_1(t) - \sin(2\pi f_1\, t)\sin\phi_1(t)\right] \end{aligned}$$

After approximation, we get,

$$s_1(t) = V_{c1}\left[\cos(2\pi f_1\, t) \times 1 - \phi_1(t)\sin(2\pi f_1\, t)\right.$$
$$s_1(t) = V_{c1}\cos(2\pi f_1\, t) - V_{c1}\phi_1(t)\sin(2\pi f_1\, t)$$

Substituting,

$$\phi_1(t) = 2\pi k_1 \int_0^t x(t)\, dt, \quad \text{we obtain}$$

$$s_1(t) = V_{c1}\cos(2\pi f_1\, t) - 2\pi k_1 V_{c1}\sin(2\pi f_1\, t)\int_0^t x(t)\, dt$$

This expression represents a narrow band FM. Thus, at the output of the phase modulator, we obtain a narrow band FM wave.

Part II: Implementation of the Phase Modulator

Figure.2.4 shows the block diagram of phase modulator circuit.

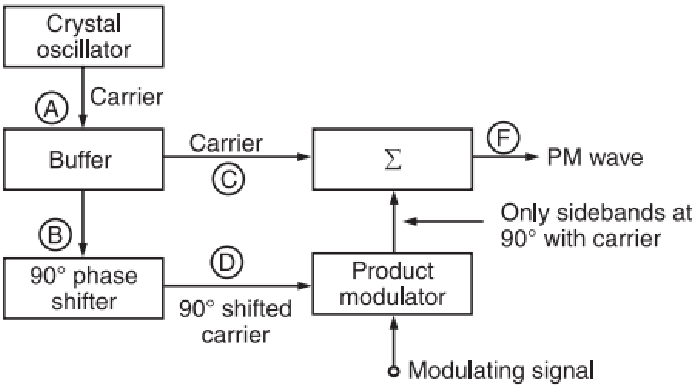


Fig 2.4 Phase Modulator Circuit

Working Principle

The crystal oscillator produces a stable unmodulated carrier which is applied to the 90° phase shifter as well as the combining network through a buffer.

The 90° phase shifter produces a 90° phase shifted carrier. It is applied to the balanced modulator along with the modulating signal.

Thus, the carrier used for modulation is 90° shifted with respect to the original carrier.

At the output of the product modulator, we get DSB SC signal i.e., AM signal without carrier.

This signal consists of only two sidebands with their resultant in phase with the 90° shifted carrier.

The two sidebands and the original carrier without any phase shift are applied to a combining network (∑). At the output of the combining network, we get the resultant of vector addition of the carrier and two sidebands as shown in figure 2.5.

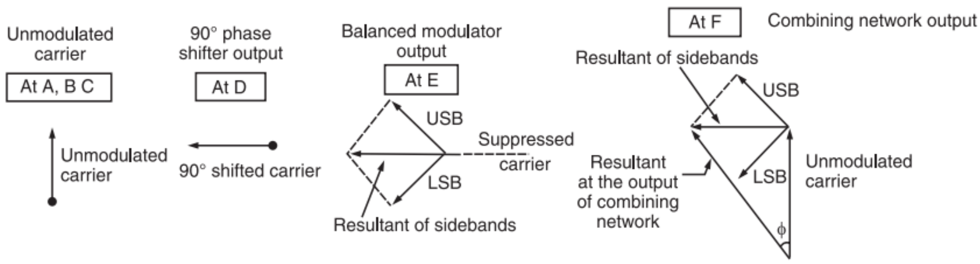


Fig 2.5 Phasor explaining the generation of PM

Now, as the modulation index is increased, the amplitude of sidebands will also increase. Hence, the amplitude of their resultant increases. This will increase the angle Φ made by the resultant with unmodulated carrier.

The angle Φ decreases with reduction in modulation index as shown in figure2.6.

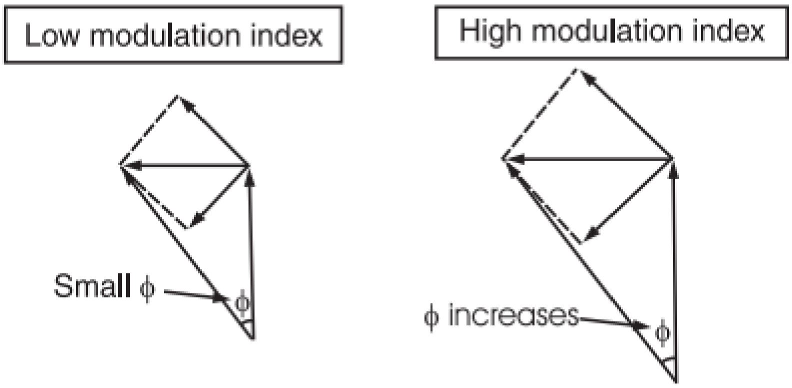


Fig 2.6 Effect of modulation index on frequency

Thus, the resultant at the output of the combining network is phase modulated. Hence, the block diagram operates as a phase modulator.

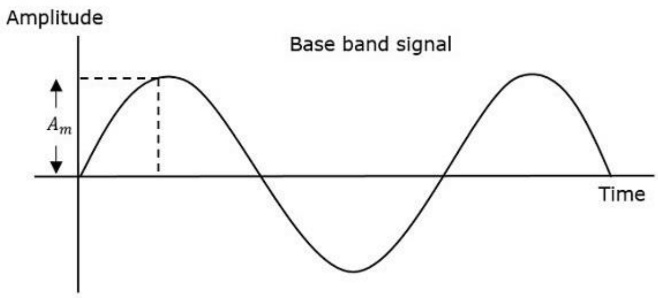
Part III: Use of Frequency Multipliers Mixer and Amplifier

The FM signal produced at the output of phase modulator has a low carrier frequency and low modulation index. They are increased to an adequately high value with the help of frequency multipliers and mixer.

Phase Modulation

In frequency modulation, the frequency of the carrier varies. Whereas, in Phase Modulation (PM), the phase of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.

So, in phase modulation, the amplitude and the frequency of the carrier signal remains constant. This can be better understood by observing the following figures.



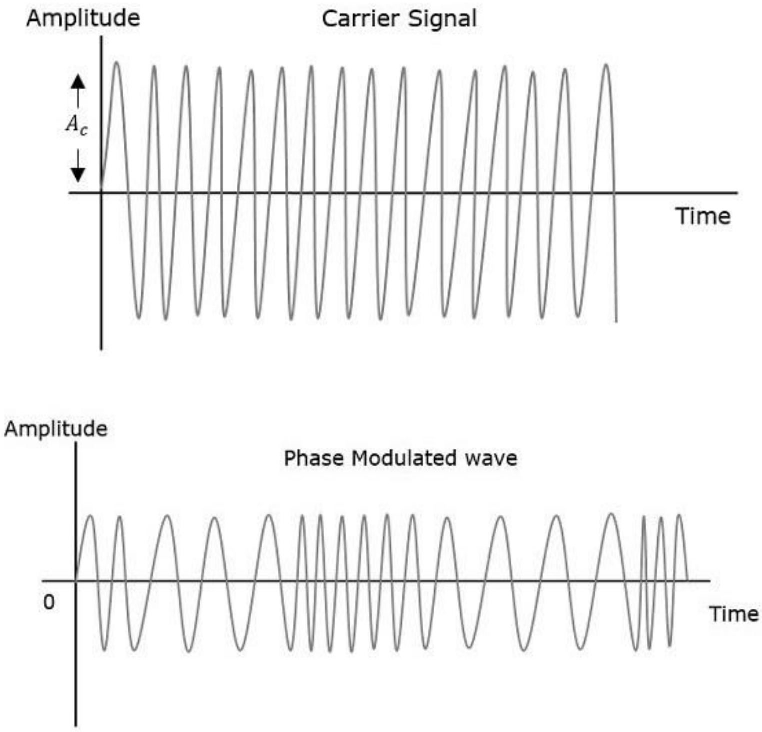


Fig 2.7 Phase Modulation

The phase of the modulated wave has got infinite points, where the phase shift in a wave can take place. The instantaneous amplitude of the modulating signal changes the phase of the carrier signal. When the amplitude is positive, the phase changes in one direction and if the amplitude is negative, the phase changes in the opposite direction.

Mathematical Representation

The equation for instantaneous phase ϕi in phase modulation is

$$\phi_i = k_p m(t)$$

Where,

- k_p is the phase sensitivity
- $m(t)$ is the message signal

The standard equation of angle modulated wave is

$$s(t) = A_c \cos(2\pi f_c t + \phi_i)$$

Substitute, ϕ_i value in the above equation.

$$s(t) = A_c \cos(2\pi f_c t + k_p m(t))$$

This is the **equation of PM wave**.

If the modulating signal, $m(t) = A_m \cos(2\pi f_m t)$, then the equation of PM wave will be

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

Where,

- β = *modulation index* = $\Delta\phi = k_p A_m$
- $\Delta\phi$ is phase deviation

Phase modulation is used in mobile communication systems, while frequency modulation is used mainly for FM broadcasting.

Principle of Slope Detection

According to the principle of the slope detector, the received FM signal is applied to an LC circuit whose output is an amplitude and frequency-modulated signal. This signal is then passed to an AM detector, which uses a detector diode, D, as shown in Figure (a) to recover the modulating signal, Vo.

The circuit diagram of a slope detector is shown in Figure 2.7. This circuit is also known as a single-tuned slope detector.

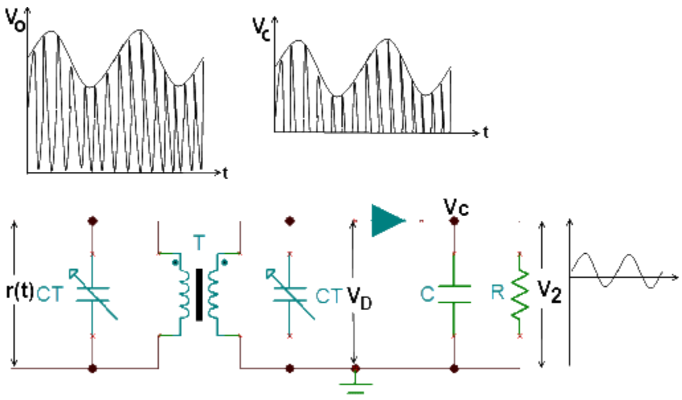


Fig 2.7 FM Slope Detector Circuit

The transformer, T, shown in Figure (a), passes the received signal to the diode D. The secondary winding of the transformer T used as the inductor, and a capacitor. CT is connected in parallel to constitute an LC resonating circuit. The secondary winding is tuned to a frequency slightly less than the resonating frequency of the LC resonating circuit the resonating frequency of the resonating circuit. fo is greater than the central frequency of the input signal fc.

Symbolically. fo > fc as shown in below Figure 2.8.

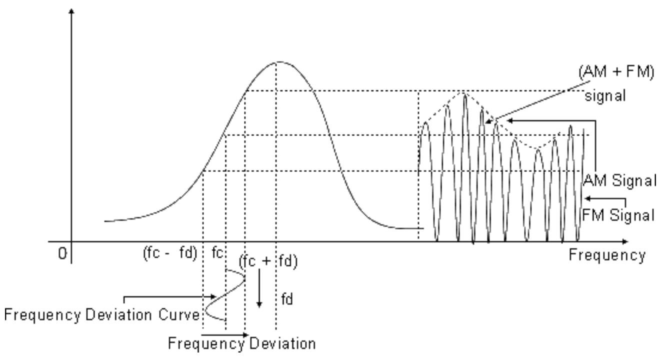


Fig 2.8 Slope Detector Waveform

If the maximum frequency deviation in the input FM signal is ±df. the operating frequency range of the voltage versus frequency curve of Figure (b) will be (fc -fd) to (fc + fd), as clearly shown in Figure (b). This range covers the linear region of the curve. The frequency variation is converted into the corresponding voltage variation, and the voltage available at the anode of the diode D carries both the amplitude variation and the frequency variation in direct proportion to the modulating signal. This is marked as Vo in Figure 2.7.

The detector diode D rectifies the secondary voltage VD, which is marked as r(t) in Figure 2.7. The rectified voltage is used to charge the capacitor C up to the peak values.

The capacitor discharges through the resistance R to develop the modulating, voltage Vo. This modulating voltage is the voltage em, as marked in Figure 2.7. As a result, the slope-detector circuit demodulates the received signal and recovers the original modulating signal.

Balance Slope Detector

A balanced slope detector is an improved version of the slope detector. The drawback of harmonic distortion is removed in this detector by using two slope detectors instead of one as in a single-tuned slope detector.

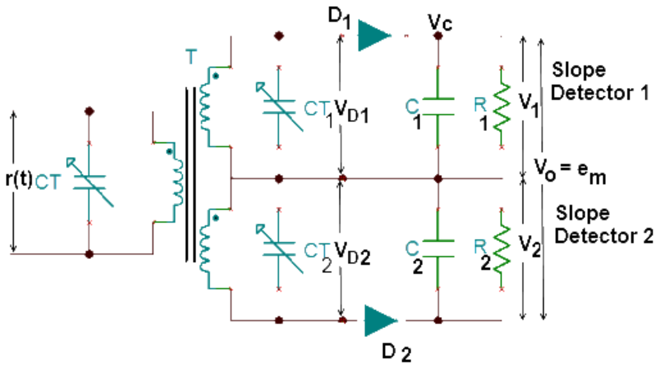


Fig 2.9 FM Balanced Slope Detector Circuit

Circuit Description

The circuit diagram, shown in Figure 2.9, has two slope detectors marked slope detector 1 and slope detector 2. Both the slope detractors are called balanced because they have identical components as follows:

Slope detector 1: It consists of a detector diode D1, filter capacitor C1, load resistor R1, and variable capacitor CT1. The variable capacitor CT1 is called the tuning capacitor because it is adjusted to tune the upper winding of the secondary Winding of the input transformer T.

Slope detector 2: It consists of a detector diode D2, that is identical to D1. It also has filter capacitor C2, load resistor R2, and variable capacitor CT2. The tuning capacitor CT2 tunes the lower winding of the secondary winding of the input transformer T.

The two slope detectors are balanced because C1 = C2, R1 = R2, and D1 is identical to D2. The upper and lower windings of the secondary windings of the center-tap transformer T are also identical.

The primary winding of the input transformer T is tuned to the central frequency of the carrier signal fc by using the tuning. capacitor CT. The secondary windings are tuned to different frequencies so that the circuit is staggered tuned. The outputs of the detector diodes D1 and D2, are filtered by C1, R1 and C2, R2, respectively. The voltage V1. which is developed across R1, and voltage V2, which is developed across R2 are added together between the points E and F to get the final output voltage Vo, which is the modulating voltage em.

Circuit Operation

The operation of the circuit can be explained by considering the two slope detectors separately.

Slope Detectorn 1

The resonating frequency of slope detector 1 is set to *f1*, by adjusting CT1 so that it is greater than *fc* by an amount of Δ*f*. As a result, slope detector 1 is tuned to *fc*, given as:

$f1 = fc + \Delta f$

The signal coupled to the upper winding of the secondary winding of the transformer T has a central frequency *fc*. If the maximum frequency deviation of the incoming FM signal, r(t), is ±*fd*, the operating range of slope detector 1 is between *fc* and (*fc* + *fd*), shown in figure (b). Which illustrates the frequency response curve of both slope detectors.

When the incoming signal frequency deviation between *fc* and (*fc* + *fd*), diode D1 is forward biased because voltage VD1 increases according to the frequency response curve of slope detector 1. The diode rectifies this amplitude and frequency modulated signal. Capacitors C1 and resistor R1 then filter the rectified voltage. The voltage so developed across R1 is the positive half of the modulating signal. This can explained by the frequency-deviation curve applied at *fc* on the frequency-axis of Figure (b). The positive half of the frequency-deviation curve lies in the response curve of slope detector 1. Therefore, the frequency deviations that are greater than are converted into corresponding amplitude by slope detector 1.

Slope Detector 2

Slope detector 2 is tuned to f2, by adjusting the tuning capacitor C2 such that:

$f2 = fc - \Delta f$

The input FM signal whose frequency deviation lies between fc and (fc — df) is converted into corresponding amplitude variations by slope detector 2 because this part of the frequency-deviation curve lies toward the frequency-response curve of slope detector 2. The voltage VD2, developed across the lower winding of the secondary winding of the transformer T is amplitude-and frequency-modulated, which corresponds to the frequency deviation from fc to (fc — df). This voltage, VD2 is rectified by diode D2, because it is forward-biased and the capacitor C2, filters this rectified voltage. The filtered voltage is developed across R2 and, as a result, the output of the balanced slope detector is the negative half of the modulating signal corresponding to the frequency deviations lower than fc.

Combined Response of Slope Detectors 1 and 2

The output of the balanced slope detector is the combined output of the individual slope detectors. Slope detector 1 provides the positive half of the modulating signal across the Output terminals E and F. When D1 is forward-biased, the diode D, is also slightly forward-biased because the voltage developed across the lower winding is very small. This is because slope detector 2 is tuned to (fc — Δf), while the incoming signal lies between fc and (fc + Δf)

The conduction of diode D2, results in a small current that flows though R2 from F to D. The direction of the current due to D1 is from E to D through R1

Therefore, when a positive voltage is developed across R1 a very small negative voltage also develops across R2. The sum of these two voltages appears between output terminals E and F.

Thus, the final output is sligtly reduced by a negative voltage developed across R2, because: -

$Vo = V1 - V2$

A similar action takes place when the incoming FM signal contains the frequency deviations between fc and (fc - df). The diode D2 is forward-biased because appreciable positive voltage VD2 appears across D2. This is because slope detector 2 is tuned towards (fc - df). This frequency-deviation range lies in the negative half of the modulating signal. As a result, the voltage developed across R2 after filtering the rectified output of D2 is the negative half of the modulating signal.

During the conduction of D2, a small voltage also appears across the upper winding of the secondary winding of transformer T. This voltage is small because the upper winding is tuned to (fc — Δf), while the incoming voltage lies in the frequency range fc to (fc — Δf). Due to this voltage, diode D2, also supplies a positive voltage across R1. This positive voltage slightly reduces the negative voltage appearing across R2, when summed up between the output terminals E and F. The net voltage is still a negative half cycle of the modulating signal.

The positive and negative halves of he modulating signal are available across the output terminals of the balanced slope detector marked as shown in Figure (b). The overall response of both slope detectors is shown in Figure (b), and the operating range at the curve is marked between the points K and L. This operating range of the overall response curve is a straight line, and therefore, the operation is linear. This removes the; nonlinear behavior of a single-tuned slope detector, and the higher harmonics are not generated in a balanced slope detector.

The overall response curve takes the shape of the letter S, as shown in the Figure 2.10. This is called S-response of the FM detectors.

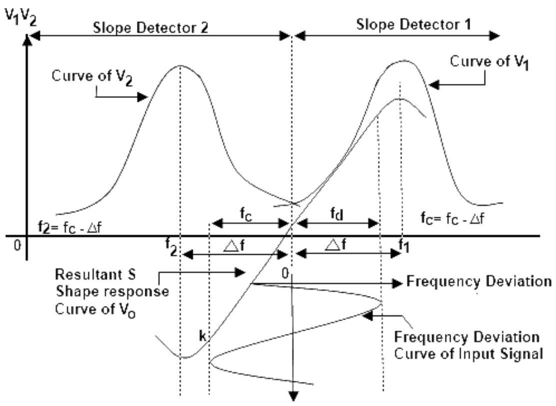
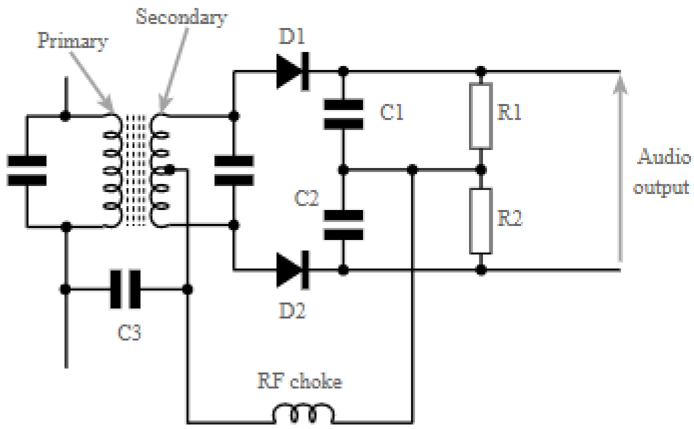


Fig 2.10 FM Balanced Slope Detector Frequency Response Curve

Foster Seely Discriminator

The Foster Seeley detector or as it is sometimes described the Foster Seeley discriminator is quite similar to the ratio detector at a first look. It has an RF transformer and a pair of diodes, but there is no third winding - instead a choke is used.



FM Foster Seeley discriminator / detector circuit

Fig 2.11 Foster Seeley Discriminator Circuit

In many respects the Foster Seeley FM demodulator resembles the circuit of a full wave bridge rectifier - the format that uses a centre tapped transformer, but additional components are added to give it a frequency sensitive aspect.

The basic operation of the circuit can be explained by looking at the instances when the instantaneous input equals the carrier frequency, the two halves of the tuned transformer circuit produce the same rectified voltage and the output is zero. If the frequency of the input changes, the balance between the two halves of the transformer secondary changes, and the result is a voltage proportional to the frequency deviation of the carrier.

Looking in more detail at the circuit, the Foster-Seeley circuit operates using a phase difference between signals. To obtain the different phased signals a connection is made to the primary side of the transformer using a capacitor, and this is taken to the centre tap of the transformer. This gives a signal that is 90° out of phase.

When an un-modulated carrier is applied at the centre frequency, both diodes conduct, to produce equal and opposite voltages across their respective load resistors. These voltages cancel each one another out at the output so that no voltage is present. As the carrier moves off to one side of the centre frequency the balance condition is destroyed, and one diode conducts more than the other. This results in the voltage across one of the resistors being larger than the other, and a resulting voltage at the output corresponding to the modulation on the incoming signal.

The choke is required in the circuit to ensure that no RF signals appear at the output. The capacitors C1 and C2 provide a similar filtering function.

Both the ratio detector and Foster-Seeley detectors are expensive to manufacture. Any wound components like the RF transformers are expensive to manufacture when compared with integrated circuits produced in vast numbers. As a result the Foster Seeley discriminator as well as the ratio detector circuits are rarely used in modern radio receivers as FM demodulators.

Foster Seeley circuit for frequency control

Prior to the introduction of very stable local oscillators within superhet radios - the universal format for radios receiving FM, local oscillators had a tendency to drift. Drift was a major factor in domestic radio receivers, although it was present in all radios.

When receiving FM signals the drift meant that the incoming FM signal might drift away from being at the centre of the FM detector slope onto the non-linear portions. This meant that the signal would become distorted.

To overcome this, radio receivers would incorporate a facility known as automatic frequency control was implemented. Using this, the DC offset from the FM demodulator is used to tune the receiver local oscillator to bring it back on frequency.

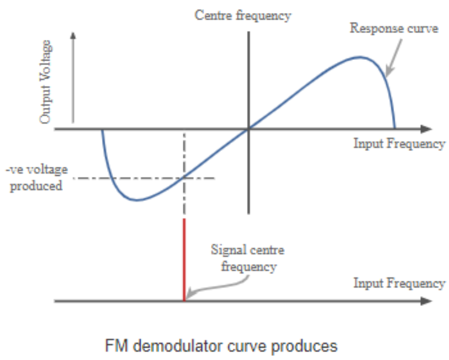


Fig 2.12 FM Demodulator Curve

A DC offset is produced when the centre frequency of the carrier is not on the centre of the demodulator curve. By filtering off the audio, only a DC component remains. Typically a long time constant RC combination is used to achieve this. The time constant of this RC network can be quite long as the drift of the oscillator occurs gradually over a period of seconds, and it must also be longer than that of the lowest frequency of the audio.

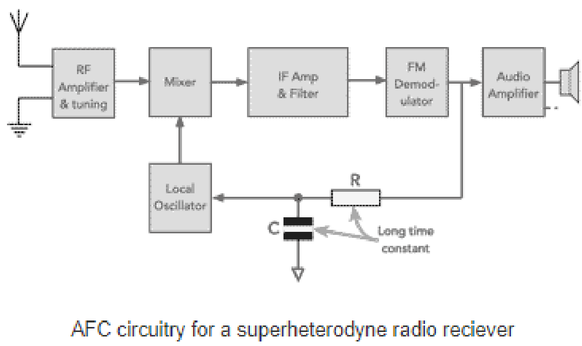


Fig 2.13 AFC Circuit

The filtered voltage is applied to a varactor diode within the local oscillator such that it causes the local oscillator to remain on tune for the FM signal being received. In this way the receiver can operate so that the signal being received is demodulated within the linear region of the FM demodulator.

Essentially the effect of the AFC circuitry is to create a form of negative feedback loop that seeks to keep the centre of the FM signal at the centre of the FM demodulation S curve. It is essentially a frequency locked loop.

Most radios used for FM reception that have free running local oscillators incorporate an automatic frequency control, AFC circuit. It uses only a few components and it provides for a significant improvement in the performance of the receiver, enabling the FM signal to be demodulated with minimum distortion despite the drift of the local oscillator signal.

Prior to the widespread introduction of frequency synthesizers, AFC was not always used in radios such as walkie talkies and handhelds radios aimed at for two way radio communications applications as they tended to use crystal controlled oscillators and these did not drift to any major degree. Hence there was less requirement for an AFC.

Ratio Detector

In the Foster-Seeley discriminator, changes in the magnitude of the input signal will give rise to amplitude changes in the resulting output voltage. This makes prior limiting necessary. It is possible to modify the discriminator circuit to provide limiting, so that the amplitude limiter may be dispensed with. A circuit so modified is called a Ratio Detector Circuit. As we now, the sum Vao + Vbo remains constant, although the difference varies because of changes in input frequency. This assumption is not completely true. Deviation from this ideal does not result in undue distortion in the Ratio Detector Circuit, although some distortion is undoubtedly introduced. It follows that any variations in the magnitude of this sum voltage can be considered spurious here. Their suppression will lead to a discriminator which is unaffected by the amplitude of the incoming signal. It will therefore not react to noise amplitude or spurious amplitude modulation.

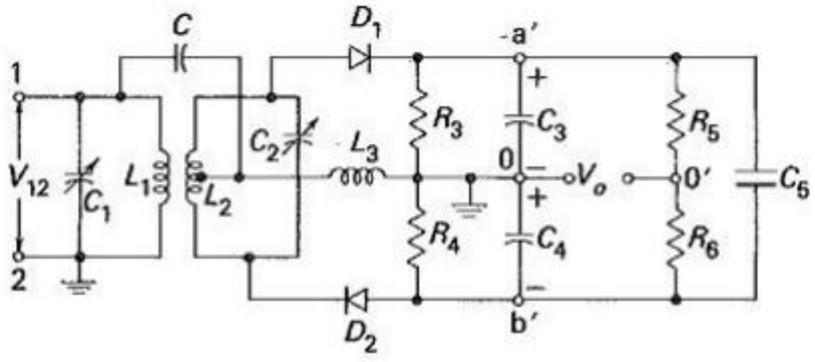


Fig 2.14 Ratio Detector Circuit

Operation:

With diode D2 reversed, o is now positive with respect to b', so that Va′b′ is now a sum voltage, rather than the difference it was in the discriminator. It is now possible to connect a large capacitor between a' and b' to keep this sum voltage constant. Once C5 has been connected, it is obvious that Va′b′ is no longer the output voltage; thus the output voltage is now taken between o and o′. It is now necessary to ground one of these two points, and o happens to be the more convenient, as will be seen when dealing with practical Ratio Detector Circuit. Bearing in mind that in practice R5 = R6, Vo is calculated as follows:

$$V_o = V_{b'o'} - V_{b'o} = \frac{V_{a'b'}}{2} - V_{b'o} = \frac{V_{a'o} + V_{b'o}}{2} - V_{b'o}$$
$$= \frac{V_{a'o} - V_{b'o}}{2}$$

Equation shows that the ratio detector output voltage is equal to half the difference between the output voltages from the individual diodes. Thus (as in the phase discriminator) the output voltage is proportional to the difference between the individual output voltages. The Ratio Detector Circuit therefore behaves identically to the discriminator for input frequency changes. The S curve of Figure 6-40 applies equally to both circuits.

Amplitude limiting by the ratio detector:

It is thus established that the ratio detector behaves in the same way as the phase discriminator when input frequency varies (but input voltage remains constant). The next step is to explain how the Ratio Detector Circuit reacts to amplitude changes. If the input voltage V12 is constant and has been so for some time, C5 has been able to charge up to the potential existing between a' and b'. Since this is a dc voltage if V12 is constant, there will be no current either flowing in to charge the capacitor or flowing out to discharge it. In other words, the input impedance of C5 is infinite. The total load impedance for the two diodes is therefore the sum of R3 and R4, since these are in practice much smaller than R5 and R6. If V12 tries to increase, C5 will tend to oppose any rise in Vo. The way in which it does this is not, however, merely to have a fairly long time constant, although this is certainly part of the operation. As soon as the input voltage tries to rise, extra diode current flows, but this excess current flows into the capacitor C5, charging it. The voltage Va′b′ remains constant at first because it is not possible for the voltage across a capacitor to change instantaneously. The situation now is that the current in the diodes load has risen, but the voltage across the load has not changed. The conclusion is that the load impedance has decreased. The secondary of the ratio detector transformer is more heavily damped, the Q falls, and so does the gain of the amplifier driving the Ratio Detector Circuit. This neatly counteracts the initial rise in input voltage. Should the input voltage fall, the diode current will fall, but the load voltage will not, at first, because of the presence of the capacitor. The effect is that of an increased diode load impedance; the diode current has fallen, but the load voltage has remained constant. Accordingly, damping is reduced, and the gain of the driving amplifier rises, this time counteracting an initial fall in the input voltage. The ratio detector provides what is known as diode variable damping. We have here a system of varying the gain of an amplifier by changing the damping of its tuned circuit. This maintains a constant output voltage despite changes in the amplitude of the input.

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2. Anokh Singh, “Principles of Communication Engineering”, S. Chand & Company Ltd, 2006

3. Taub and Schilling, “Principles of Communication Systems”, Third Edition, TMH, New Delhi, 2008.

4. Louis Frenzel, "Communication Electronics”, McGraw-Hill Companies,1999.

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Question Bank

Part A

1. What do you mean by narrowband and wideband FM?
2. Give the frequency spectrum of narrowband FM?
3. Why Armstrong method is superior to reactance modulator.
4. Define frequency deviation in FM?
5. State Carson’s rule of FM bandwidth?
6. Differentiate between narrow band and wideband FM.?
7. What are the advantages of FM.?
8. Define PM.
9. What is meant by indirect FM generation?
10. Define modulation index of FM and PM.

Part B

1. Explain the indirect method of generation of FM wave and any one method of demodulating an FM wave.
2. Derive the expression for the frequency modulated signal. Explain what is meant by narrow band FM and wide band FM?
3. Explain the FM detection using balanced slope detector.
4. Explain the Foster Seeley discriminator.

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DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

UNIT – III- Pulse Modulation and Multiplexing– SEC1304

III. Pulse Modulation and Multiplexing

Pulse Modulation

Pulse modulation is a technique in which the signal is transmitted with the information by pulses. This is divided into Analog Pulse Modulation and Digital Pulse Modulation.

Analog pulse modulation is classified as

Pulse Amplitude Modulation (PAM)

Pulse Width Modulation (PWM)

Pulse Position Modulation (PPM)

Digital modulation is classified as

Pulse Code Modulation

Delta Modulation

Pulse Amplitude Modulation

Pulse amplitude modulation is a technique in which the amplitude of each pulse is controlled by the instantaneous amplitude of the modulation signal. It is a modulation system in which the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the signal at the instant of sampling. This technique transmits the data by encoding in the amplitude of a series of signal pulses.

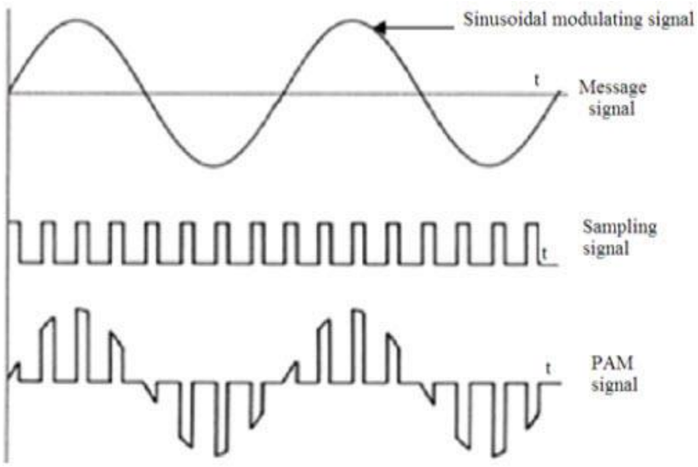

Fig 3.1 Pulse Amplitude Modulation Signal

There are two types of sampling techniques for transmitting a signal using PAM. They are:

Flat Top PAM

Natural PAM

Flat Top PAM

The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. The amplitude of the signal cannot be changed with respect to the analog signal to be sampled. The tops of the amplitude remain flat.

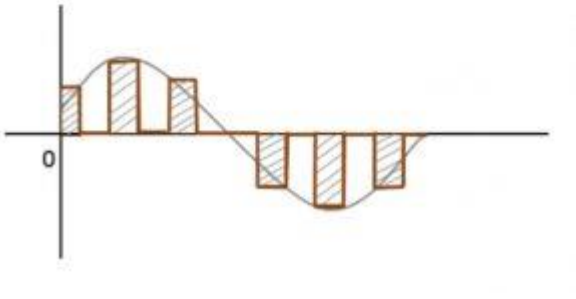

Fig 3.2 Flat Top Pulse Amplitude Modulation

Natural PAM

The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. Then follows the amplitude of the pulse for the rest of the half-cycle.

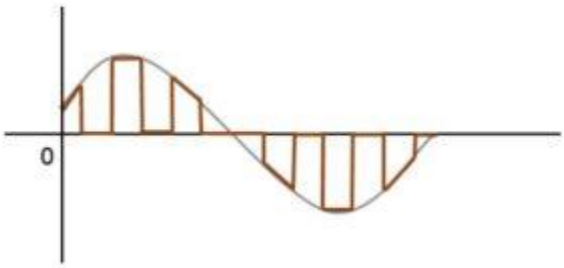

Fig 3.3 Natural Pulse Amplitude Modulation

Pulse Width Modulation (PWM)

In PWM, the width of the modulated pulses varies in proportion with the amplitude of modulating signal. The waveforms of PWM is shown in fig.

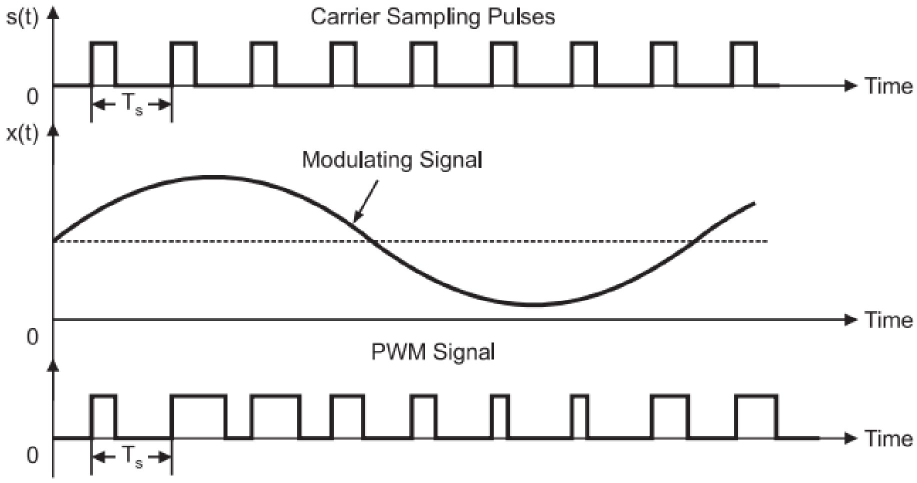

Fig 3.4 PWM Signal

Generation of PWM Signal

The block diagram of a PWM signal generator is shown in fig.2 below. This circuit can also be used for the generation of PPM signal.

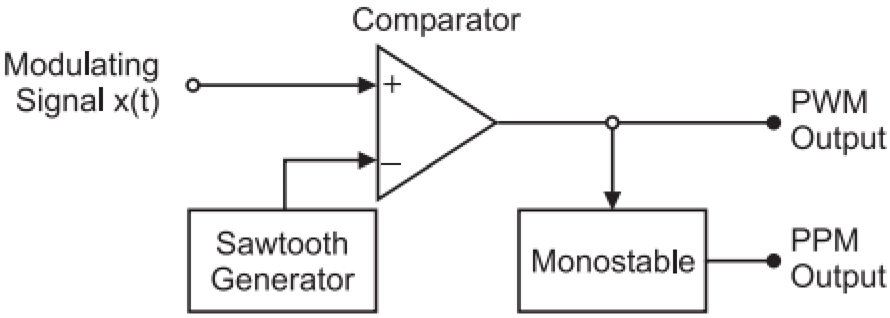

Fig 3.5 PWM and PPM Generator

A sawtooth generator generates a sawtooth signal of frequency fs, and this sawtooth signal in this case is used as a sampling signal.It is applied to the inverting terminal of a comparator.The modulating signal x (t) is applied to the non-inverting terminal of the same comparator.The comparator output will remain high as long as the instantaneous amplitude of x (t) is higher than that of the ramp signal.This gives rise to a PWM signal at the comparator output as shown in fig.

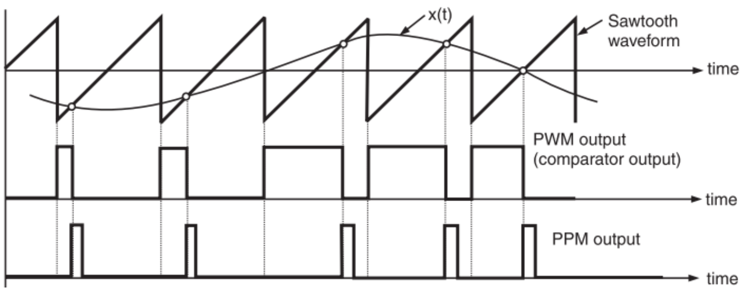

Fig 3.6 Fig Waveforms

Here, it may be noted that the leading edges of the PWM waveform coincide with the falling edges of the ramp signal. Thus, the leading edges of PWM signal are always generated at fixed time instants.

However, the occurance of its trailing edges will be dependent on the instantaneous amplitude of x(t). Therefore, this PWM signal is said to be trail edge modulated PWM.

Detection of PWM Signal

The circuit for the detection of PWM signal is shown in fig.

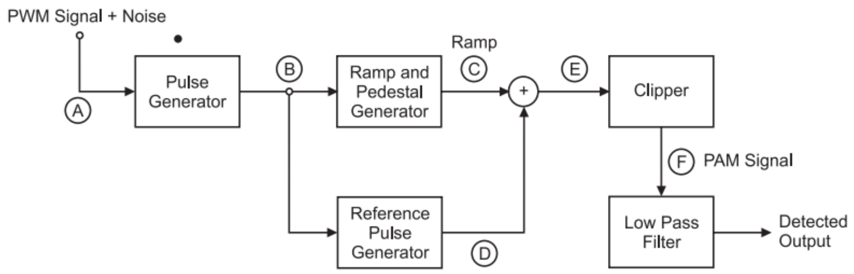

Fig 3.7 PWM Detection Circuit

The working operation of the circuit may be explained as under:

The PWM signal received at the input of the detection circuit is contaminated with noise. This signal is applied to pulse generator circuit which regenerates the PWM signal.

Thus, some of the noise is removed and the pulses are squared up.

The regenerated pulses are applied to a reference pulse generator. It produces a train of constant amplitude, constant width pulses.

These pulses are synchronized to the leading edges of the regenerated PWM pulses but delayed by a fixed interval.

The regenerated PWM pulses are also applied to a ramp generator. At the output of it, we get a constant slope ramp for the duration of the pulse. The height of the ramp is thus proportional to the width of the PWM pulses.

At the end of the pulse, a sample and hold amplifier retains the final ramp voltage until it is reset at the end of the pulse.

The constant amplitude pulses at the output of reference pulse gtenerator are then added to the ramp signal.

The output of the adder is then clipped off at a thereshold level to generate a PAM signal at the output of the clipper.

A low pass filter is used to recover the original modulating signal back from the PAM signal. The waveforms for this circuit have been shown in fig.

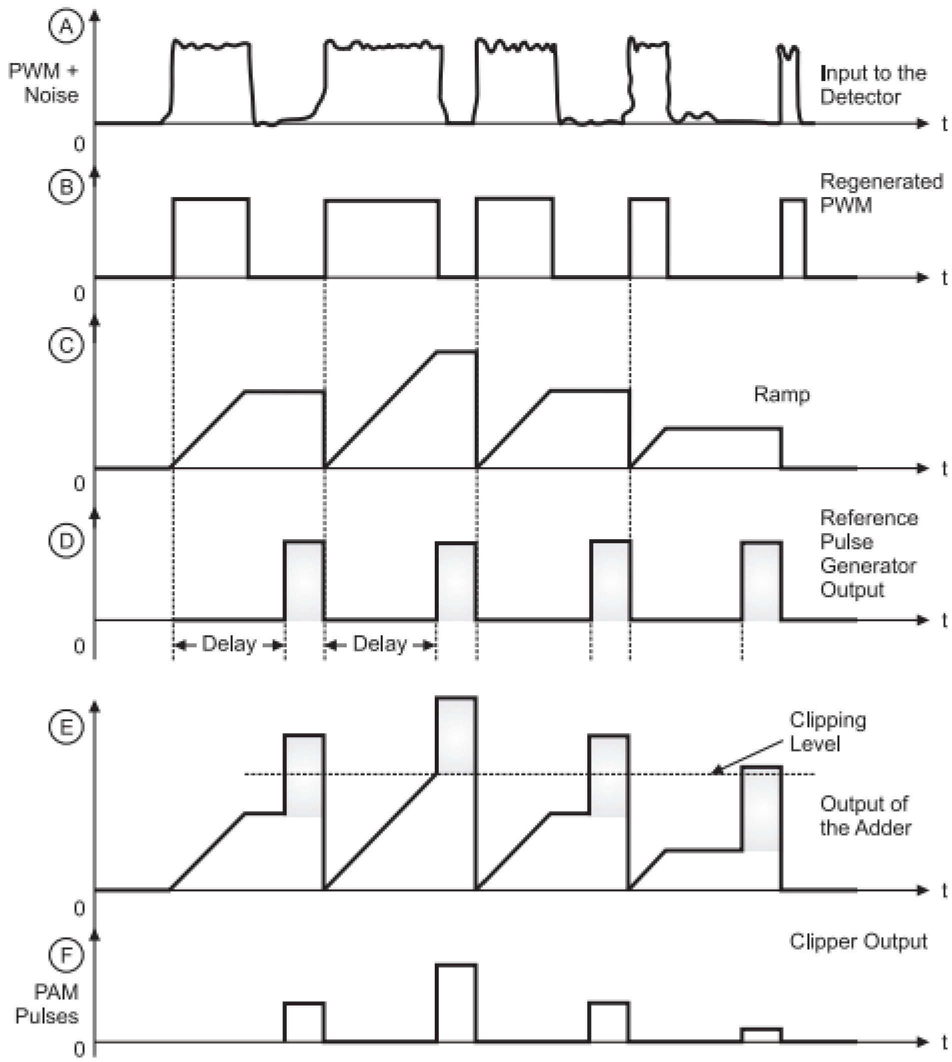

Fig 3.9 Waveforms for PWM detection circuit

Advantages of PWM

- Less effect of noise i.e., very good noise immunity.
- Synchronization between the transmitter and receiver is not essential (Which is essential in PPM).
- It is possible to reconstruct the PWM signal from a noise, contaminated PWM, as discussed in the detection circuit. Thus, it is possible to separate out signal from noise (which is not possible in PAM).

Disadvantages of PWM

- Due to the variable pulse width, the pulses have variable power contents. Hence, the transmission must be powerful enough to handle the maximum width, pulse, though the average power transmitted can be as low as 50% of this maximum power.
- In order to avoid any waveform distortion, the bandwidth required for the PWM communication is large as compared to bandwidth of PAM.

Multiplexing is used in the cases where the signals of lower bandwidth and the transmitting media is having higher bandwidth. In this case, the possibility of sending a number of signals is more. In this the signals are combined into one and are sent over a link which has greater bandwidth of media than the communicating nodes.

Time Division Multiplexing (TDM)

This happens when data transmission rate of media is greater than that of the source, and each signal is allotted a definite amount of time. These slots are so small that all transmissions appear to be parallel. In frequency division multiplexing all the signals operate at the same time with different frequencies, but in time division multiplexing all the signals operate with same frequency at different times.

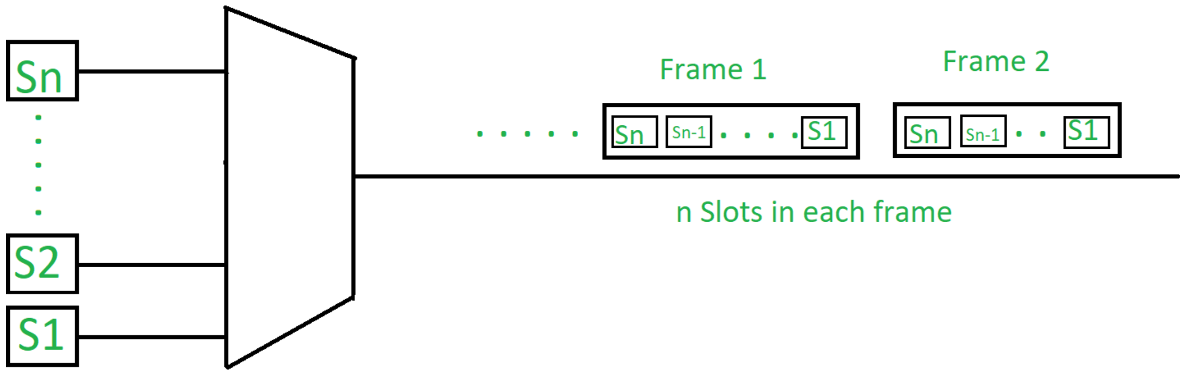

Fig 3.10 Fig Time Division Multiplexing

Synchronous TDM

The time slots are pre-assigned and fixed. This slot is even given if the source is not ready with data at this time. In this case the slot is transmitted empty. It is used for multiplexing digitized voice stream.

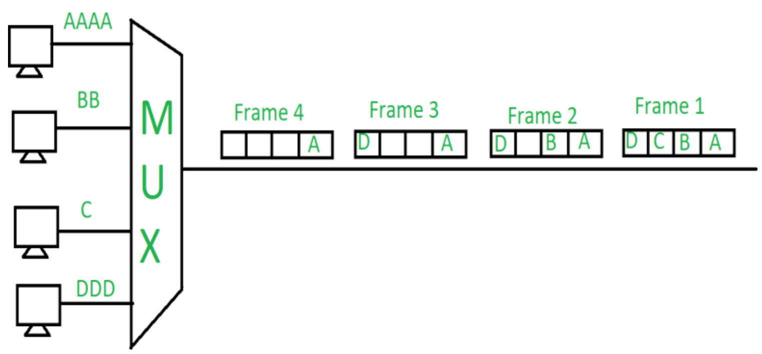

Fig 3.11 Synchronous Time Division Multiplexing

Asynchronous (or statistical) TDM

The slots are allocated dynamically depending on the speed of source or their ready state. It dynamically allocates the time slots according to different input channel's needs, thus saving the channel capacity.

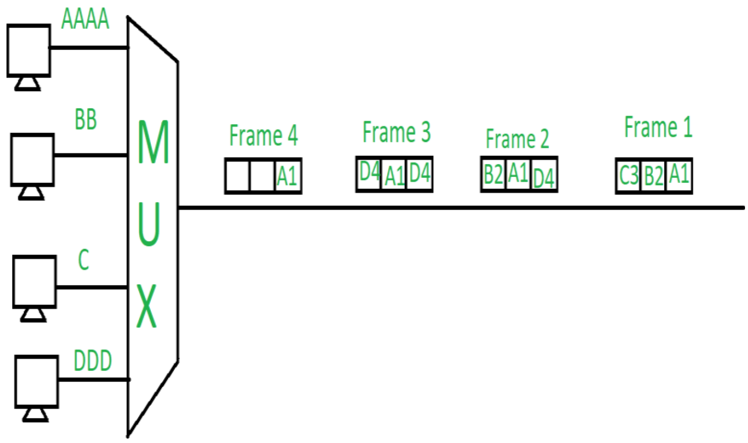

Fig 3.12 Asynchronous Time Division Multiplexing

Frequency Division Multiplexing (FDM)

In this a number of signals are transmitted at the same time, and each source transfers its signals in the allotted frequency range. There is a suitable frequency gap between the 2 adjacent signals to avoid over-lapping. Since the signals are transmitted in allotted time so this decreases the probability of collision. The frequency spectrum is divided into several logical channels, in which every user feels that they possess a particular bandwidth. A number of signals are sent simultaneously on the same time allocating separate frequency band or channel to each signal. It is used in radio and TV transmission. Therefore, to avoid interference between two successive channels Guard bands are used.

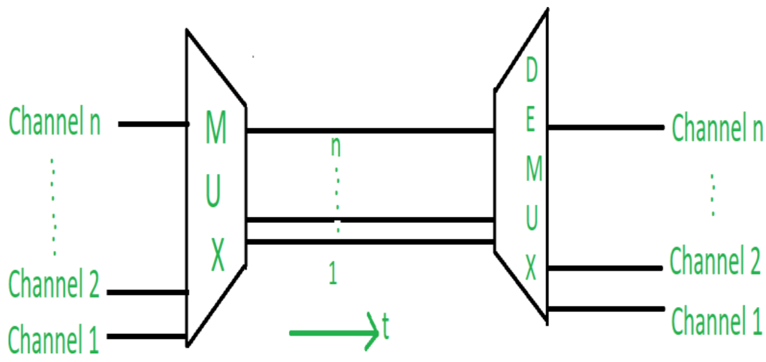

Fig 3.13 Frequency division Multiplexing

Pulse Code Modulation

A signal is Pulse Code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a **Pulse Code Modulation (PCM)** will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.

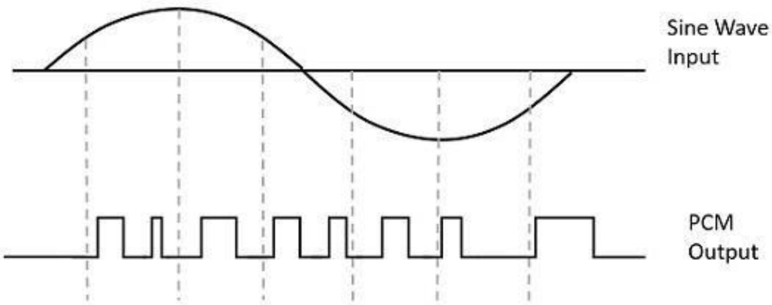

Fig 3.14 Output of Pulse Code Modulation

Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant. In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of Sampling, Quantizing and Encoding, which are performed in the analog-to-digital converter section. The low pass filter prior to sampling prevents aliasing of the message signal.

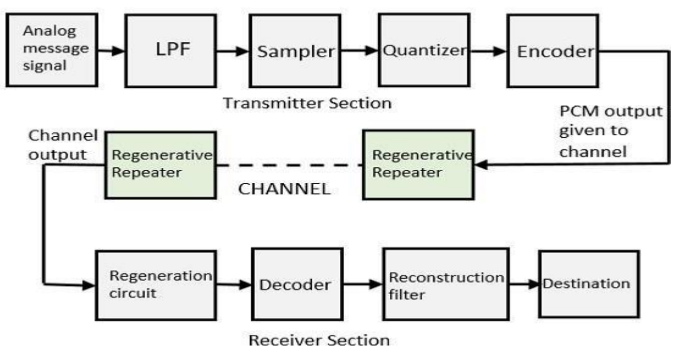

Fig 3.15 Block Diagram of PCM

The basic operations in the receiver section are regeneration of impaired signals, decoding, and reconstruction of the quantized pulse train. The following figure is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.
Elements of PCM

Low Pass Filter (LPF)

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the circuit which uses the technique that helps to collect the sample data at instantaneous values of the message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component W of the messagesignal, in accordance with the sampling theorem.

Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level bya binary code. The sampling done here is the sample-and-hold process. These three sections will act as an analog to the digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal.

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the demodulator.

Reconstruction Filter

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low pass filter is employed, called as the reconstruction filter to get back the original signal. Hence, the Pulse Code Modulator circuit digitizes the analog signal given, codes it, and samples it. It then transmits in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

Quantization

The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as Quantization.

Quantizing an Analog Signal

The analog-to-digital converters perform this type of function to create a series of digital values out of the given analog signal. The following figure represents an analog signal. This signal to get converted into digital, has to undergo sampling and quantizing.

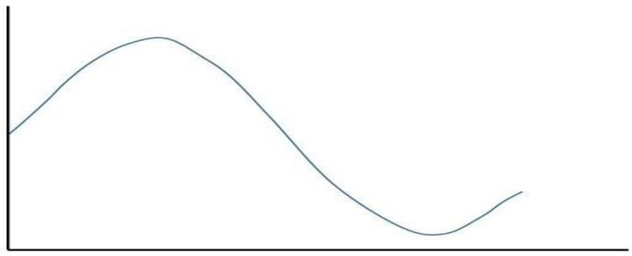

Fig 3.16 Analog Signal

The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal. The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the brown one represents the quantized signal.

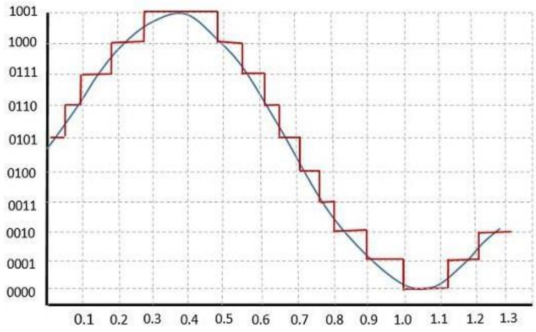

Fig 3.17 Quantizing an Analog Signal

Quantization

Both sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels. The spacing between the two adjacent representation levels is called a quantum or step-size. The below figure shows the resultant quantized signal which is the digital form for the given analog signal.

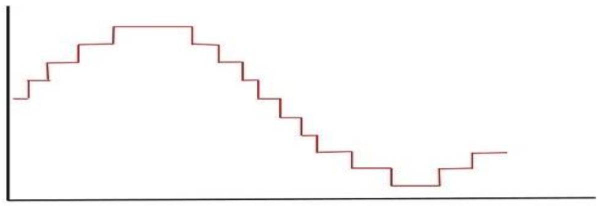

Fig 3.18 Stair Case Signal

Resultant Quantized Signal: This is also called as Stair-case waveform, in accordance with its shape.

Types of Quantization

There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

The type of quantization in which the quantization levels are uniformly spaced is termed as a Uniform Quantization. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a Non-uniform Quantization. There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

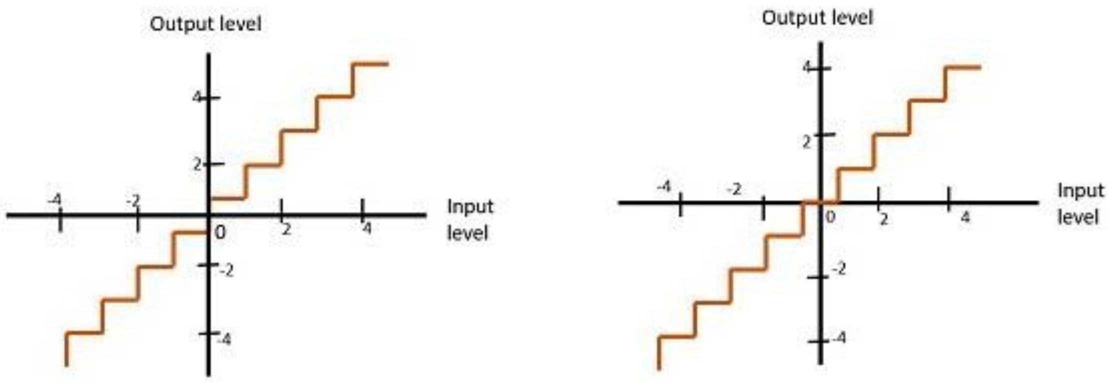

Fig 3.19 Mid-rise **TreadQuantization Types**

Figure 3.19 shows the mid-rise type and the mid-tread type of uniform quantization.

The Mid-Rise type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number. The Mid-tread type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number. Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

Quantization Error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values. The difference between an input value and its quantized value is called a Quantization Error. A Quantizer is a logarithmic function that performs Quantization rounding off the value. An analog-to-digital converter (ADC) works as a quantizer.

Quantization Noise

It is a type of quantization error, which usually occurs in analog audio signal, while quantizingit to digital. For example, in music, the signals keep changing continuously, where a regularityis not found in errors. Such errors create a wideband noise called as Quantization Noise.

Companding in PCM

The word Companding is a combination of Compressing and Expanding, which means that it does both. This is a non-linear technique used in PCM which compresses the data at the transmitter and expands the same data at the receiver. The effects of noise and crosstalk are reduced by using this technique.

There are two types of Companding techniques. They are

A-law Companding Technique

- Uniform quantization is achieved at A = 1, where the characteristic curve is linear andno compression is done.
- A-law has mid-rise at the origin. Hence, it contains a non-zero value.
- A-law companding is used for PCM telephone systems.

µ-law Companding Technique

- Uniform quantization is achieved at µ = 0, where the characteristic curve is linear andno compression is done.
- µ-law has mid-tread at the origin. Hence, it contains a zero value.
- µ-law companding is used for speech and music signals.
- µ-law is used in North America and Japan.

For the samples that are highly correlated, when encoded by PCM technique, leave redundant information behind. To process this redundant information and to have a better output, it is a wise decision to take a predicted sampled value, assumed from its previous output and summarize them withthe quantized values. Such a process is called as **Differential PCM** DPCM technique.

DPCM Transmitter

The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits. Following is the block diagram of DPCM transmitter.

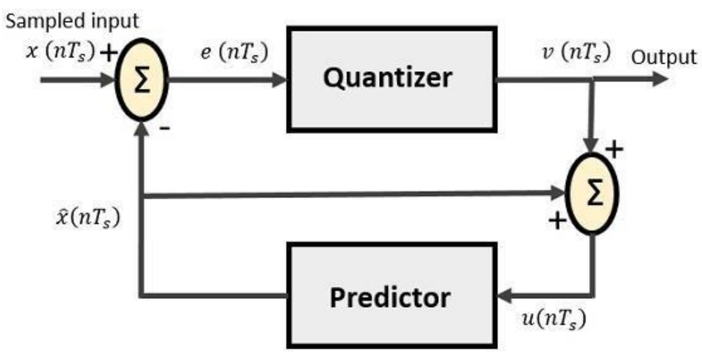

Fig 3.20 DPCM Transmitter

The signals at each point are named as −

- x(nTs) is the sampled input
- x^(nTs) is the predicted sample
- e(nTs) is the difference of sampled input and predicted output, often called asprediction error
- v(nTs) is the quantized output
- u(nTs) is the predictor input which is actually the summer output of the predictor outputand the quantizer output

The predictor produces the assumed samples from the previous outputs of the transmitter circuit. The input to this predictor is the quantized versions of the input signal x(nTs).

Quantizer Output is represented as

$$v(nT_s) = Q[e(nT_s)]$$

$$= e(nT_s)+q(nT_s)$$

Where **q (nT_s)** is the quantization error

Predictor input is the sum of quantizer output and predictor output,
u(nTs)=x^(nTs)+v(nTs)

u(nTs)=x^(nTs)+e(nTs)+q(nTs)u(nTs) = x(nTs)+q(nTs)

The same predictor circuit is used in the decoder to reconstruct the original input.

DPCM Receiver

The block diagram of DPCM Receiver consists of a decoder, a predictor, and a summer circuit.Following is the diagram of DPCM Receiver.

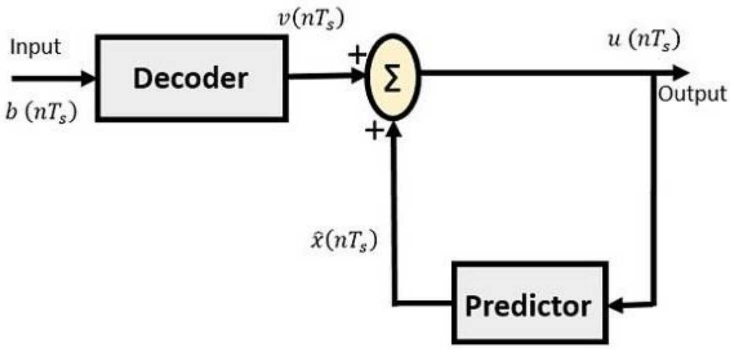

Fig 3.21 DPCM Receiver

The notation of the signals is the same as the previous ones. In the absence of noise, the encoded receiver input will be the same as the encoded transmitter output. As mentioned before, the predictor assumes a value, based on the previous outputs. The input given to the decoder is processed and that output is summed up with the output of the predictor, to obtaina better output.

The sampling rate of a signal should be higher than the Nyquist rate, to achieve better sampling. If this sampling interval in Differential PCM is reduced considerably, the sample to-sample amplitude difference is very small, as if the difference is **1-bit quantization**, then the step-size will be very small i.e., **Δ** delta.

Delta Modulation

The type of modulation, where the sampling rate is much higher and in which the step size after quantization is of a smaller value **Δ**, such a modulation is termed as **delta modulation**.

Features of Delta Modulation

Following are some of the features of delta modulation.

An over-sampled input is taken to make full use of the signal correlation.

The quantization design is simple.

The input sequence is much higher than the Nyquist rate.

The quality is moderate.

The design of the modulator and the demodulator is simple.

The stair-case approximation of output waveform.

The step-size is very small, i.e., **Δ** delta.

The bit rate can be decided by the user.

This involves simpler implementation.

Delta Modulation is a simplified form of DPCM technique, also viewed as **1-bit DPCM scheme**. As the sampling interval is reduced, the signal correlation will be higher.

Delta Modulator

The Delta Modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.

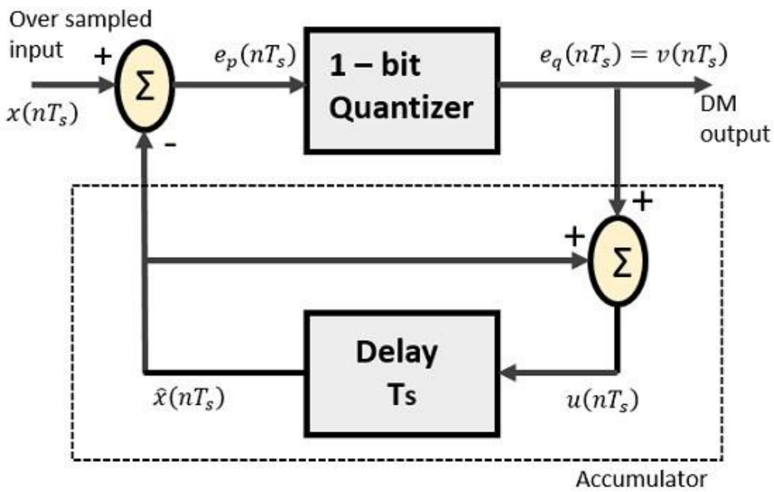

Fig 3.22 Delta Modulator

The predictor circuit in DPCM is replaced by a simple delay circuit in DM.From the above diagram, we have the notations as −

- x(nTs) = over sampled input
- ep(nTs) = summer output and quantizer input
- eq(nTs) = quantizer output = v(nTs)
- x^(nTs) = output of delay circuit
- u(nTs) = input of delay circuit

Using these notations, now we shall try to figure out the process of delta modulation.

$ep(nTs)=x(nTs)-x^{\wedge}(nTs)$ ---------------------------equation 1

$=x(nTs)-u([n-1]Ts)$

$=x(nTs)-[x^{\wedge}[[n-1]Ts]+v[[n-1]Ts]]$ ------------------------------------equation 2

Further,

$v(nTs)=eq(nTs)=S.sig.[ep(nTs)]$ --------------------------equation 3

$u(nTs)=x^{\wedge}(nTs)+eq(nTs)$Where,

- $x^{\wedge}(nTs)$ = the previous value of the delay circuit
- $eq(nTs)$ = quantizer output = $v(nTs)$Hence,

$u(nTs)=u([n-1]\ Ts)+v(nTs)$--------------------------- equation 4

Which means,

The present input of the delay unit

=The previous output of the delay unit + the present quantizer outputAssuming zero

condition of Accumulation,

$$u(nTs) = S\sum_{j=1}^{n} sig[ep(jTs)]$$

Accumulated version of DM output = $\sum_{j=1}^{n} v(jTs)$--------------------------------equation 5

Now, note that

$x^{\wedge}(nTs) = u([n-1]\ Ts)$

$$= \sum_{j=1}^{n-1} v(jTs)$$ --- equation 6

Delay unit output is an Accumulator output lagging by one sample. From equations 5 & 6, we get a possible structure for the demodulator. A Stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (**Δ**). The output quality of the waveform is moderate.

Delta Demodulator

The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

Following is the diagram for delta demodulator.

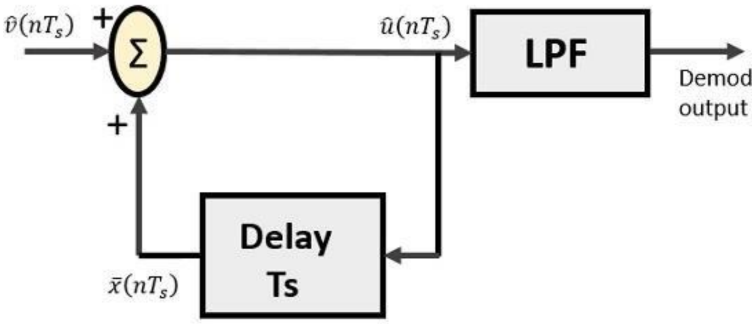

Fig 3.23 Delta Demodulator

From the above diagram, we have the notations as −

- v^(nTs) is the input sample
- u^(nTs) is the summer output
- x¯(nTs) is the delayed output

A binary sequence will be given as an input to the demodulator. The stair-case approximated output is given to the LPF. Low pass filter is used for many reasons, but the prominent reason is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Advantages of DM Over DPCM

1-bit quantizer

Very easy design of the modulator and the demodulator however, there exists some

noise in DM.

Slope Over load distortion (when **Δ** is small)

Granular noise (when **Δ** is large)

TEXT / REFERENCE BOOKS

1. George Kennedy, "Electronic Communication Systems", Third Edition, TMH, New Delhi,1991

2. Anokh Singh, "Principles of Communication Engineering", S. Chand & Company Ltd, 2006

3. Taub and Schilling, “Principles of Communication Systems”, Third Edition, TMH, New Delhi, 2008.

4. Louis Frenzel, "Communication Electronics”, McGraw-Hill Companies,1999.

5. Bruce Carlson, “Communication Systems”, McGraw Hill, 2010

Question Bank

Part A

1. Define Pulse Code Modulation?
2. What is line coding?
3. What are the disadvantages of PCM?
4. What are the disadvantages of delta modulation?
5. What is Guard band?
6. Define Aliasing.
7. Define Nyquist rate.
8. What is quantization?
9. Mention the types of digital modulation techniques.

Part B

1. Explain the detection of PWM signal.
2. Explain Frequency division multiplexing and Time division multiplexing
3. Explain Pulse code modulation.
4. Explain the generation of Binary and Quadrature phase shift keying.

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DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

UNIT – IV- Transmitters and Receivers– SEC1304

IV. Transmitters And Receivers

AM transmitter - High level modulator

Below figure's show the block diagram of high-level and low-level transmitters. The basic difference between the two transmitters is the power amplification of the carrier and modulating signals.

Figure (a) shows the block diagram of high-level AM transmitter.

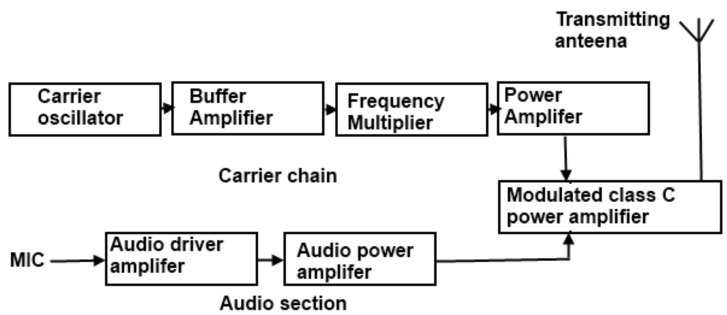

Fig 4.1 Block Diagram of High Level AM Transmitter

Figure (a) is drawn for audio transmission. In high-level transmission, the powers of the carrier and modulating signals are amplified before applying them to the modulator stage, as shown in figure (a). In low-level modulation, the powers of the two input signals of the modulator stage are not amplified. The required transmitting power is obtained from the last stage of the transmitter, the class C power amplifier.

The various sections of the figure (a) are:

- Carrier oscillator
- Buffer amplifier
- Frequency multiplier
- Power amplifier
- Audio chain
- Modulated class C power amplifier

Carrier oscillator

The carrier oscillator generates the carrier signal, which lies in the RF range. The frequency of the carrier is always very high. Because it is very difficult to generate high frequencies with good frequency stability, the carrier oscillator generates a sub multiple with the required carrier frequency. This sub multiple frequency is multiplied by the frequency multiplier stage to get the required carrier frequency. Further, a crystal oscillator can be used in this stage to generate a low frequency carrier with the best frequency stability. The frequency multiplier stage then increases the frequency of the carrier to its required value.

Buffer Amplifier

The purpose of the buffer amplifier is two fold. It first matches the output impedance of the carrier oscillator with the input impedance of the frequency multiplier, the next stage of the carrier oscillator. It then isolates the carrier oscillator and frequency multiplier.

This is required so that the multiplier does not draw a large current from the carrier oscillator. If this occurs, the frequency of the carrier oscillator will not remain stable.

Frequency Multiplier

The sub-multiple frequency of the carrier signal, generated by the carrier oscillator , is now applied to the frequency multiplier through the buffer amplifier. This stage is also known as harmonic generator. The frequency multiplier generates higher harmonics of carrier oscillator frequency. The frequency multiplier is a tuned circuit that can be tuned to the requisite carrier frequency that is to be transmitted.

Power Amplifier

The power of the carrier signal is then amplified in the power amplifier stage. This is the basic requirement of a high-level transmitter. A class C power amplifier gives high power current pulses of the carrier signal at its output.

Audio Chain

The audio signal to be transmitted is obtained from the microphone, as shown in figure (a). The audio driver amplifier amplifies the voltage of this signal. This amplification is necessary to drive the audio power amplifier. Next, a class A or a class B power amplifier amplifies the power of the audio signal.

AM transmitter – Low level modulator

This is the output stage of the transmitter. The modulating audio signal and the carrier signal, after power amplification, are applied to this modulating stage. The modulation takes place at this stage. The class C amplifier also amplifies the power of the AM signal to the reacquired transmitting power. This signal is finally passed to the antenna, which radiates the signal into space of transmission.

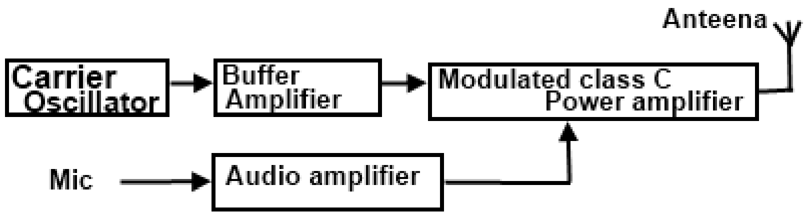

Fig 4.2 Block Diagram of Low Level AM Transmitter

The low-level AM transmitter shown in the figure (b) is similar to a high-level transmitter, except that the powers of the carrier and audio signals are not amplified. These two signals are directly applied to the modulated class C power amplifier.

Modulation takes place at the stage, and the power of the modulated signal is amplified to the required transmitting power level. The transmitting antenna then transmits the signal.

Coupling of Output Stage and Antenna

The output stage of the modulated class C power amplifier feeds the signal to the transmitting antenna. To transfer maximum power from the output stage to the antenna it is necessary that the impedance of the two sections match. For this , a matching network is required. The matching between the two should be perfect at all transmitting frequencies. As the matching is required at different frequencies, inductors and capacitors offering different impedance at different frequencies are used in the matching networks.

The matching network must be constructed using these passive components. This is shown in below Figure **Fig 4.3**.

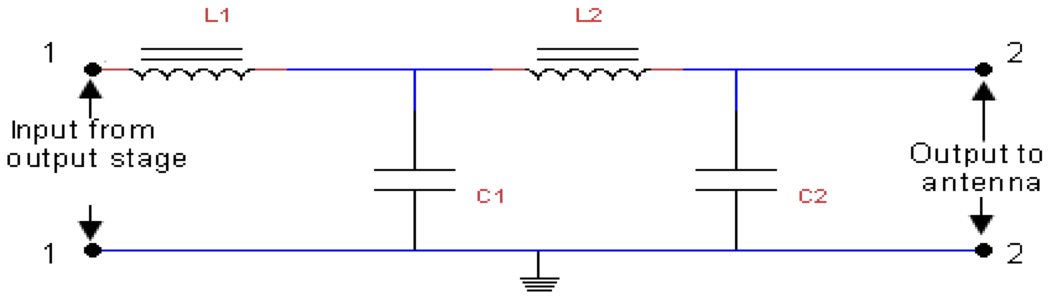

Fig 4.3 Double Pi Matching Network

The matching network used for coupling the output stage of the transmitter and the antenna is called double π-network. This network is shown in figure (c). It consists of two inductors , L_1 and L_2 and two capacitors, C_1 and C_2. The values of these components are chosen such that the input impedance of the network between 1 and 1'. Shown in figure (c) is matched with the output impedance of the output stage of the transmitter. Further, the output impedance of the network is matched with the impedance of the antenna.

The double π matching network also filters unwanted frequency components appearing at the output of the last stage of the transmitter. The output of the modulated class C power amplifier may contain higher harmonics, such as second and third harmonics, that are highly undesirable. The frequency response of the matching network is set such that these unwanted higher harmonics are totally suppressed, and only the desired signal is coupled to the antenna.

preemphasis concept

At the transmitter, the modulating signal is passed through a simple network which amplifies the high frequency, components more than the low-frequency components. The simplest form of such a circuit is a simple high pass filter of the type shown in fig (a). Specification dictate a time constant of 75 microseconds (µs) where t = RC. Any combination of resistor and capacitor (or resistor and inductor) giving this time constant will be satisfactory. Such a circuit has a cutoff frequency fco of 2122 Hz. This means that frequencies higher than 2122 Hz will he linearly enhanced. The output amplitude increases with frequency at a rate of 6 dB per octave. The pre-emphasis curve is shown in Fig (b). This pre-emphasis circuit increases the energy content of the higher-frequency signals so that they will tend to become stronger than the high frequency noise components. This improves the signal to noise ratio and increases intelligibility and fidelity.

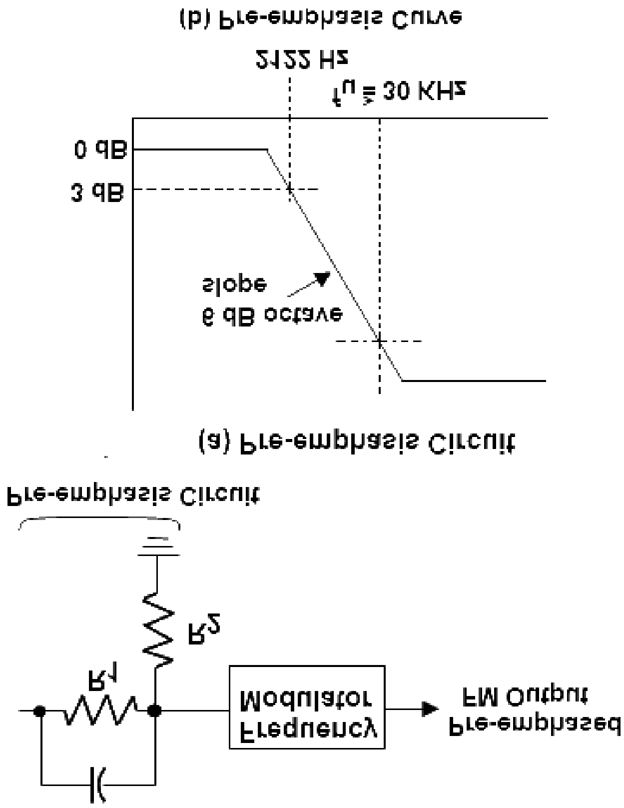

Fig 4.4 Pre-emphasis Circuit & Curve

The pre-emphasis circuit also has an upper break frequency fu where the signal enhancement flattens out.
See Fig (b). This upper break frequency is computed with the expression.
fu = R1 +(R2/2πR1R1C)
It is usually set at some very high value beyond the audio range. An fu of greater than 30KHz is typical.

FM stereo broad cast transmitter

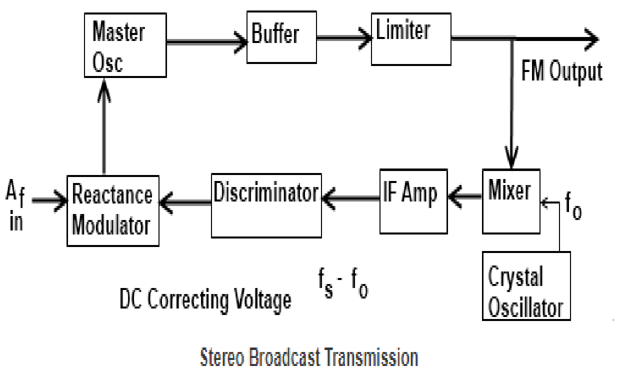

Fig 4.5 Stereo Broadcast Transmiiter

A small portion of the signal is taken from the limiter output and fed to mixer in which this signal is mixed with a signal from the crystal oscillator. The difference signal which is usually about one- tenth- to one-twentieth of the master oscillator frequency is amplified and applied to a phase discriminator. The output of the phase discriminator is a DC signal which is applied to the reactance modulator as a correcting voltage to counteract any drift in the average frequency of the master oscillator.

A block diagram of a typical frequency stabilizing system used is shown in Figure. It uses a basic frequency, Standard, a crystal oscillator, and the carrier frequency of the FM signal is compared with it. We know that reactance modulator works across the tank circuit of a LC oscillator, whose output is isolated by a buffer stage. The output of buffer is fed to an amplitude limiter and subsequently to the class C power amplifiers (not shown).

AM Receivers-super heterodyne receiver

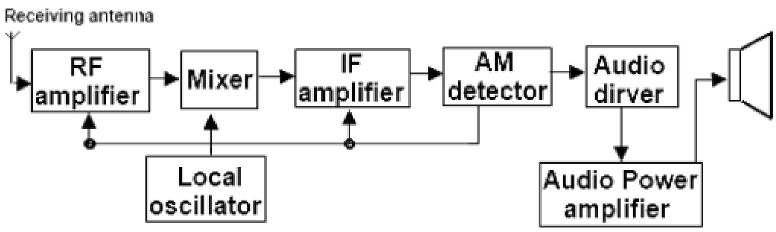

Fig 4.6 Superheterodyne AM Receiver

In superheterodyne radio receivers, the incoming radio signals arc intercepted by the antenna arid converted into the corresponding currents and voltages. In the receiver, the incoming signal frequency is mixed with a locally generated frequency. The output of the mixer consists of the sum and difference of the two frequencies. The mixing of the two frequencies is termed *heterodyning*. Out of the two resultant components of the mixer, the sum component is rejected and the difference component is selected. The value of the difference frequency component varies with the incoming frequencies, if the frequency of the local oscillator is kept constant. It is possible to keep the frequency of the difference components constant by varying the frequency of the local oscillator according to the incoming signal frequency. In this case, the process is called Superheterodyne and the receiver is known as a superheterodyne radio receiver.

AGC

An **Automatic Gain Control (AGC) circuit** is a circuit that is designed to maintain a constant output signal level after amplification, despite variations in signals at the input of the amplifier or system. This is achieved by providing more amplification to weak signals and less amplification to strong signals thus maintaining a constant signal amplitude level at the output.

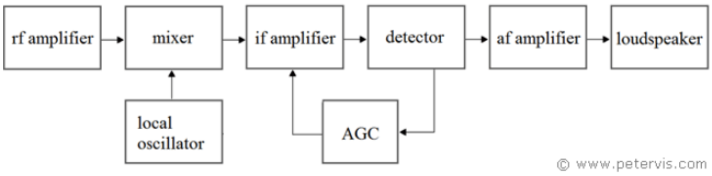

Fig 4.7 Automatic Gain Control

Deemphasis concept

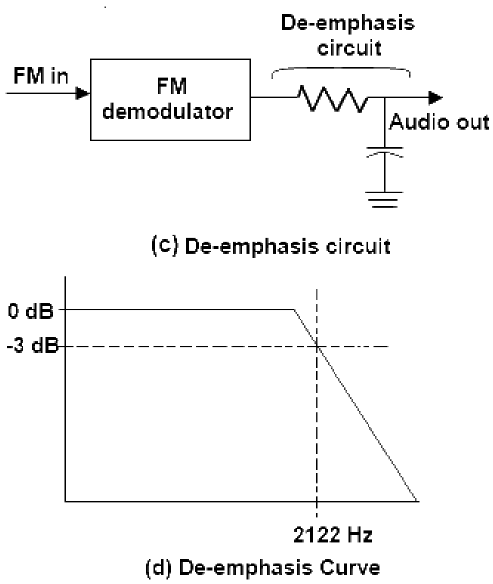

Fig 4.8 De-emphasis Circuit Curve

To return the frequency response to its normal level, a de-emphasis circuit is used at the receiver. This is a simple low-pass filter with a constant of 75 πs. See figure (c). It features a cutoff of 2122 Hz and causes signals above this frequency to be attenuated at the rate of 6bB per octave. The response curve is shown in Fig (d). As a result, the pre-emphasis at the transmitter is exactly offset by the de-emphasis circuit in the receiver, providing a normal frequency response. The combined effect of pre-emphasis and de-emphasis is to increase the high-frequency components during transmission so that they will be stronger and not masked by noise.

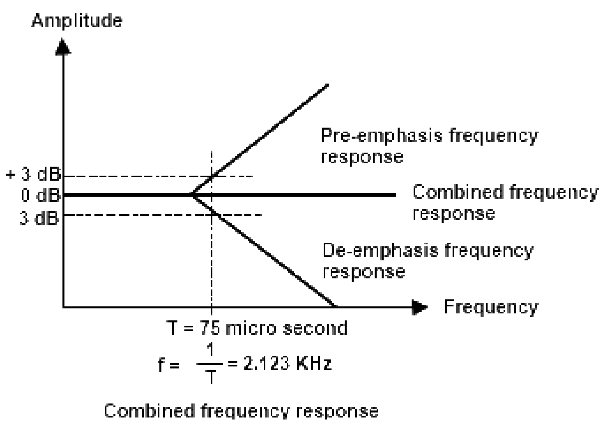

Fig 4.9 Combined Frequency Response

AFC.

In radio equipment, Automatic Frequency Control (AFC), also called Automatic Fine Tuning (AFT), is a method or circuit to automatically keep a resonant circuit tuned to the frequency of an incoming radio signal. It is primarily used in radio receivers to keep the receiver tuned to the frequency of the desired station.

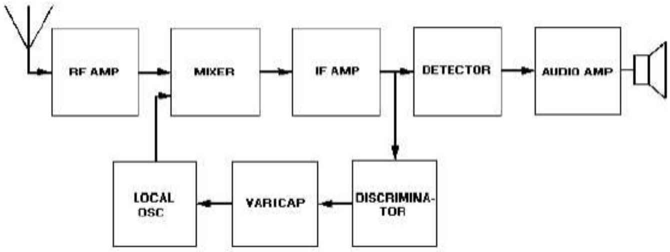

Fig 4.10 Automatic Frequency Control

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1. George Kennedy, “Electronic Communication Systems”, Third Edition, TMH, New Delhi,1991

2. Anokh Singh, “Principles of Communication Engineering”, S. Chand & Company Ltd, 2006

3. Taub and Schilling, “Principles of Communication Systems”, Third Edition, TMH, New Delhi, 2008.

4. Louis Frenzel, "Communication Electronics”, McGraw-Hill Companies,1999.

5. Bruce Carlson, “Communication Systems”, McGraw Hill, 2010

Question Bank

Part A

1. Define super heterodyne principle.
2. Define AGC.
3. Define AFC.
4. Mention the use of Pre-emphasis circuit?
5. Mention the use of De-emphasis circuit?

Part B

1. Explain Super hetrodyne AM receiver.
2. Explain the low level and high level AM transmitters.
3. Compose short notes on Automatic gain control and Automatic frequency control.

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DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

UNIT – V- Broad Band Communication System– SEC1304

V. Broad Band Communication System

Facsimile system

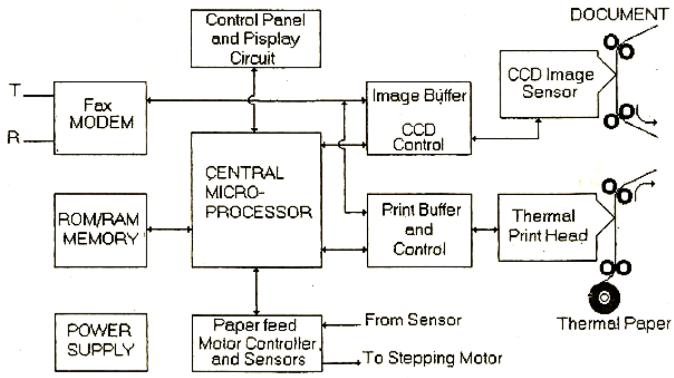


Fig 5.1 Block Diagram of Fax-Machine

In spite of the fact, that Fax-machine is small in appearance yet it incorporates a large amount of electronics and mechanical components into its operation. A simplified block diagram of the Fax-machine is shown in the figure.

Central Microprocessor

It is the heart of all Fax-Machines. The microprocessor manages all the operations of the machine and coordinates the flow of data into and out of the system. The microprocessor consists of the following memories:

a. ROM: It stores the program used to run the machine.

b. RAM: It stores results of calculations, variables, system status flags, or any other information that will change regularly during normal operations.

Fax-Modem

It generates and detects signals necessary for signal interfacing between the machine and telephone lines. When transmitting, the modem accepts scanned image data from an image memory buffer converts it into an analog form and then feed it to the telephone network. When receiving, an analog signal is received from the telephone network converts it into digital form and stored it in the image memory buffer for processing and printing.

Control Panel

It serves two functions;

a. Allows the user to enter operating parameters, such as, date, time, print resolutions, baud rate, and desired destination telephone number.

b. It displays the system status and operating parameters.

Image Buffer and Control

When a Fax-machine receives data, the modem coverts it into digital form and the microprocessor after processing it stores the data in the image buffer memory. Then the microprocessor activates the printing circuit and the data is loaded into the print buffer where it is processed into signals for the fax-machine's print mechanism.

CCD (Charge Coupled Device) Image Sensor

For the transmission of a data, the document is fed into the machine. The microprocessor activates the document feeder and scanning circuits. The scanning circuit scan one line at a time using CCD contact image sensors. The line sensor delivers data to the scan controller which processes it and then stored the image data in the image buffer memory. The microprocessor interprets this data and then transfers it to the modem. The modem then transmits the data on the telephone line.

Thermal Printer

The printer consists of thermal heads and the technique used is thermal line printing, i.e., the image is printed one line at a time onto thermal paper. The thermal printing is now being replaced by electrostatic printing, the technique used in laser printers. This technique allows single sheet paper to be used. Thus, allowing plain paper Fax-machines.

Power Supply

The power supply operates on 220/110 Vac, 50/60 Hz, single phase. The power needed varies depending on the function performed by the Fax-machine. The primary function and power requirements are: standby, transmission, reception, and copying.

Optical fiber Communication systems

An optical fiber can be understood as a dielectric waveguide, which operates at optical frequencies. The device or a tube, if bent or if terminated to radiate energy, is called a **waveguide**, in general. Following image depicts a bunch of fiber optic cables.

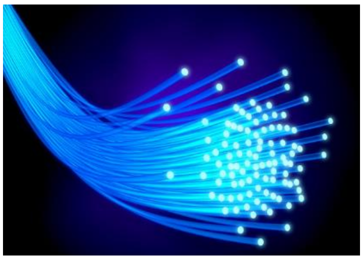


Fig 5.2 Fibre Optic Cable

The electromagnetic energy travels through it in the form of light. The light propagation, along a waveguide can be described in terms of a set of guided electromagnetic waves, called as **modes** of the waveguide.

Working Principle

A fundamental optical parameter one should have an idea about, while studying fiber optics is **Refractive index**. By definition, "The ratio of the speed of light in a vacuum to that in matter is the index of refraction ***n*** of the material." It is represented as −

$$n = \frac{c}{v}$$

Where,

c = the speed of light in free space = 3×10^8 *m/s*

v = the speed of light in di-electric or non-conducting material

Generally, for a travelling light ray, **reflection** takes place when $n_2 < n_1$. The bent of light ray at the interface is the result of difference in the speed of light in two materials that have different refractive indices. The relationship between these angles at the interface can be termed as **Snell's law**. It is represented as −

$$n_1 \sin\phi_1 = n_2 \sin\phi_2$$

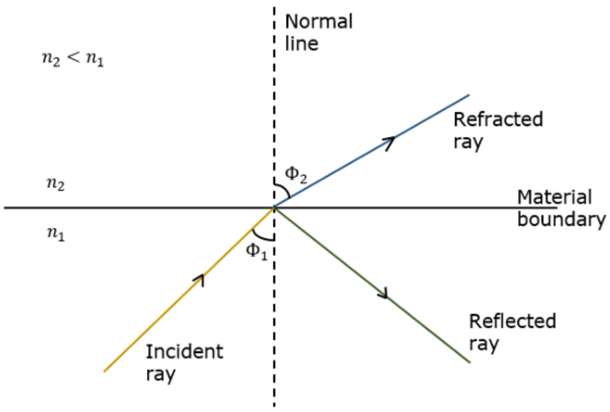
Where,

ϕ_1 is the angle of incidence

ϕ_2 is the refracted angle

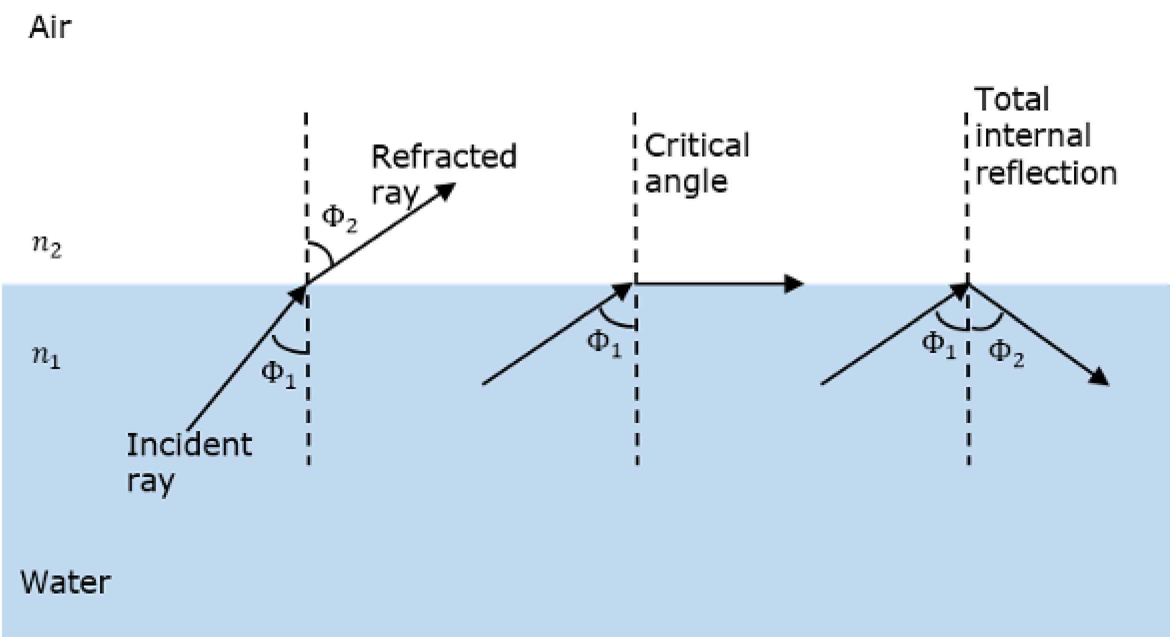
n_1 and n_2 are the refractive indices of two materials

For an optically dense material, if the reflection takes place within the same material, then such a phenomenon is called as **internal reflection**. The incident angle and refracted angle are shown in the following figure.



If the angle of incidence ϕ_1 is much larger, then the refracted angle ϕ_2 at a point becomes $\Pi/2$. Further refraction is not possible beyond this point. Hence, such a point is called as **Critical angle ϕ_c**. When the incident angle ϕ_1 is greater than the critical angle, the condition for **total internal reflection** is satisfied.

The following figure shows these terms clearly.



A light ray, if passed into a glass, at such condition, it is totally reflected back into the glass with no light escaping from the surface of the glass.

Parts of a Fiber

The most commonly used optical fiber is **single solid di-electric cylinder** of radius **a** and index of refraction n_1. The following figure explains the parts of an optical fiber.

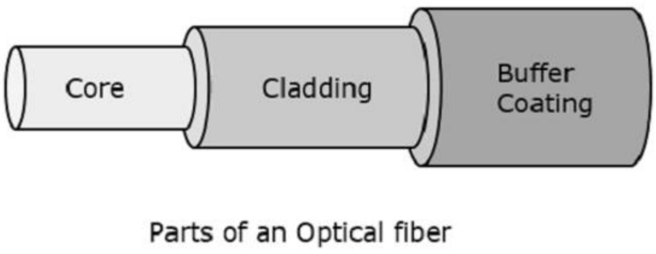


Fig 5.3 Parts of an Optical Fibre Cable

This cylinder is known as the **Core** of the fiber. A solid di-electric material surrounds the core, which is called as **Cladding**. Cladding has a refractive index n_2 which is less than n_1.

Cladding helps in −

- Reducing scattering losses.
- Adds mechanical strength to the fiber.
- Protects the core from absorbing unwanted surface contaminants.

Types of Optical Fibers

Depending upon the material composition of the core, there are two types of fibers used commonly. They are −

- **Step-index fiber** − The refractive index of the core is uniform throughout and undergoes an abrupt change (or step) at the cladding boundary.

- **Graded-index fiber** − The core refractive index is made to vary as a function of the radial distance from the center of the fiber.

Both of these are further divided into −

- **Single-mode fiber** − These are excited with laser.
- **Multi-mode fiber** − These are excited with LED.

Optical Fiber Communications

The communication system of fiber optics is well understood by studying the parts and sections of it. The major elements of an optical fiber communication system are shown in the following figure.

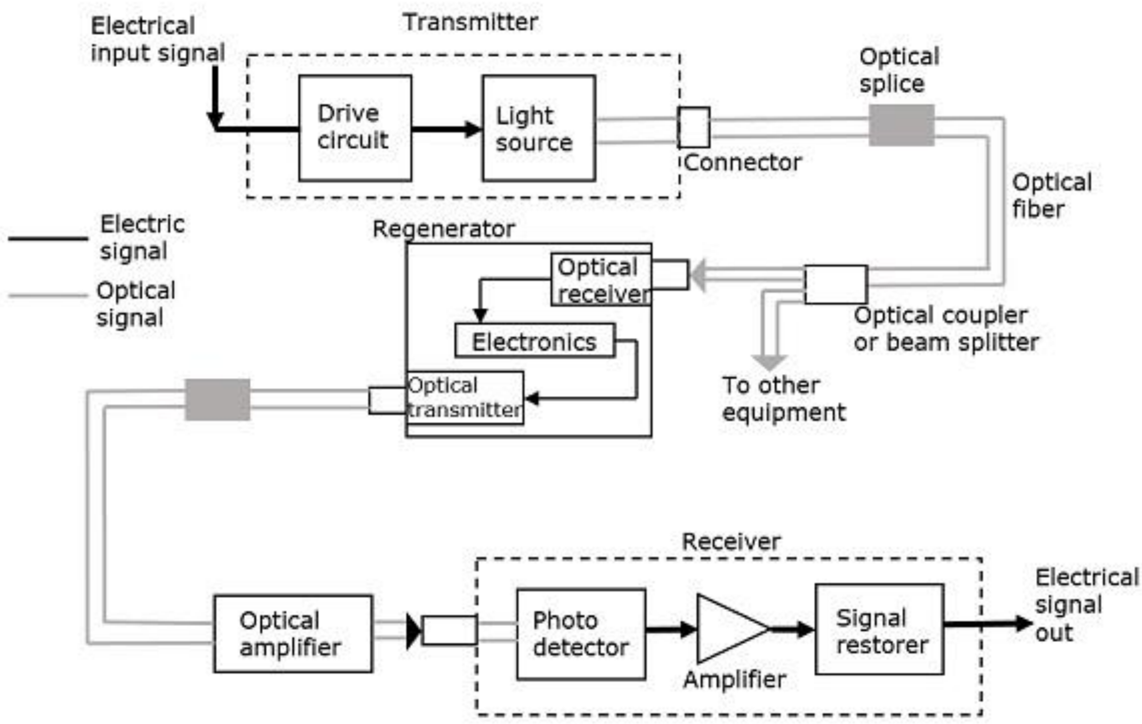


Fig 5.4 Block Diagram of Optical Fibre Communication System

The basic components are light signal transmitter, the optical fiber, and the photo detecting receiver. The additional elements such as fiber and cable splicers and connectors, regenerators, beam splitters, and optical amplifiers are employed to improve the performance of the communication system.

Functional Advantages

The functional advantages of optical fibers are −

- The transmission bandwidth of the fiber optic cables is higher than the metal cables.
- The amount of data transmission is higher in fiber optic cables.

- The power loss is very low and hence helpful in long-distance transmissions.
- Fiber optic cables provide high security and cannot be tapped.
- Fiber optic cables are the most secure way for data transmission.
- Fiber optic cables are immune to electromagnetic interference.
- These are not affected by electrical noise.

Physical Advantages

The physical advantages of fiber optic cables are −

- The capacity of these cables is much higher than copper wire cables.
- Though the capacity is higher, the size of the cable doesn't increase like it does in copper wire cabling system.
- The space occupied by these cables is much less.
- The weight of these FOC cables is much lighter than the copper ones.
- Since these cables are di-electric, no spark hazards are present.
- These cables are more corrosion resistant than copper cables, as they are bent easily and are flexible.
- The raw material for the manufacture of fiber optic cables is glass, which is cheaper than copper.
- Fiber optic cables last longer than copper cables.

Disadvantages

Although fiber optics offer many advantages, they have the following drawbacks −

- Though fiber optic cables last longer, the installation cost is high.
- The number of repeaters are to be increased with distance.
- They are fragile if not enclosed in a plastic sheath. Hence, more protection is needed than copper ones.

Applications of Fiber Optics

The optical fibers have many applications. Some of them are as follows −

- Used in telephone systems
- Used in sub-marine cable networks
- Used in data link for computer networks, CATV Systems
- Used in CCTV surveillance cameras
- Used for connecting fire, police, and other emergency services.
- Used in hospitals, schools, and traffic management systems.
- They have many industrial uses and also used for in heavy duty constructions.

Mobile telephone communication

A mobile phone is an electronic device used for mobile telecommunications over a cellular network of specialized base stations known as cell sites. A cell phone offers full Duplex Communication and transfer the link when the user moves from one cell to another. As the phone user moves from one cell area to another, the system automatically commands the mobile phone and a cell site with a stronger signal, to switch on to a new frequency in order to keep the link.

Mobile phone is primarily designed for Voice communication. In addition to the standard voice function, new generation mobile phones support many additional services, and accessories, such as SMS for text messaging, email, packet switching for access to the Internet, gaming, Bluetooth, camera with video recorder and MMS for sending and receiving photos and video, MP3 player, radio and GPS.

The cellular system is the division of an area into small cells. This allows extensive frequency reuse across that area, so that many people can use cell phones simultaneously. Cellular networks has a number of advantages like increased capacity, reduced power usage, larger coverage area, reduced interference from other signals etc.

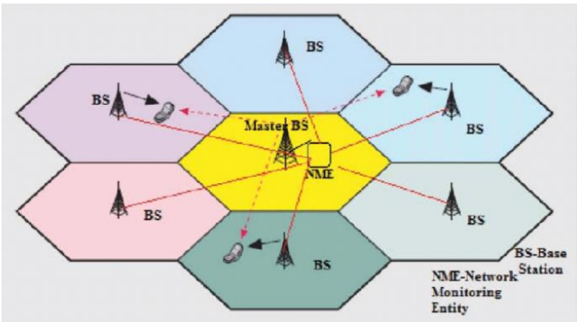
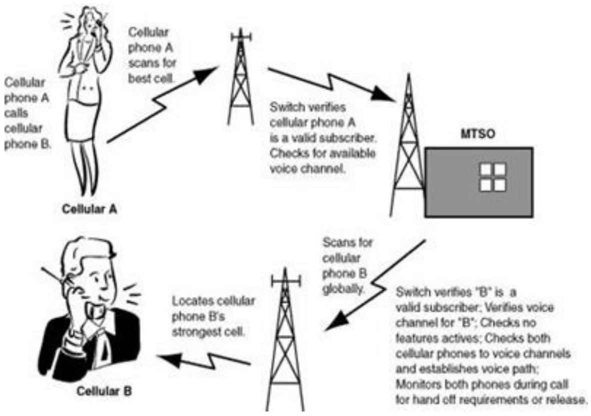


Fig 5.4 Geographical Area Divided into Hexagonal shapes called Cells

- Mobile phones have special codes associated with them. These include:
- Electronic Serial Number (ESN) -Unique 32-bit number programmed in the phone
- Mobile Identification Number (MIN) – 10 digit number derived from the phone's number.
- System Identification Code (SID) – unique 5 digit number that is assigned to each carrier by the FCC.

- When we switch on the mobile phone, it tries for an SID on the Control channel. The Control channel is a special frequency that the phone and base station use to talk to one another. If the Mobile phone finds difficulty to get link with the control channel, it displays a "no service" message.

- If the Mobile phone gets the SID, it compares the SID with the SID programmed in the phone. If both SID match, the phone identifies that the cell it is communicating is the part of its home system.
- The phone also transmits a registration request along with the SID and the MTSO keeps track of your phone's location in a database. MTSO knows in which cell you are when it wants to ring the phone.
- The MTSO then gets the signal, it tries to find the phone. The MTSO looks in its database to find the cell in which the phone is present. The MTSO then picks a frequency pair to take the call.
- The MTSO communicates with the Mobile phone over the control channel to tell it what frequencies to use. Once the Mobile phone and the tower switch on those frequencies, the call is connected.
- When the Mobile phone move toward the edge of the cell, the cell's base station will note that the signal strength is diminishing. At the same time, the base station in the cell in which the phone is moving will be able to see the phone's signal strength increasing.
- The two base stations coordinate themselves through the MTSO. At some point, the Mobile phone gets a signal on a control channel and directs it to change frequencies. This will switch the phone to the new cell



Frequency Reuse

Frequency reusing is the concept of using the same radio frequencies within a given area, that are separated by considerable distance, with minimal interference, to establish communication.

Frequency reuse offers the following benefits −

- Allows communications within cell on a given frequency
- Limits escaping power to adjacent cells
- Allows re-use of frequencies in nearby cells
- Uses same frequency for multiple conversations
- 10 to 50 frequencies per cell

For example, when N cells are using the same number of frequencies and K be the total number of frequencies used in systems. Then each cell frequency is calculated by using the formulae K/N.

In Advanced Mobile Phone Services (AMPS) when K = 395 and N = 7, then frequencies per cell on an average will be 395/7 = 56. Here, cell frequency is 56.

Satellite communication system

In general terms, a satellite is a smaller object that revolves around a larger object in space. For example, moon is a natural satellite of earth.

We know that Communication refers to the exchange (sharing) of information between two or more entities, through any medium or channel. In other words, it is nothing but sending, receiving and processing of information.

If the communication takes place between any two earth stations through a satellite, then it is called as satellite communication. In this communication, electromagnetic waves are used as carrier signals. These signals carry the information such as voice, audio, video or any other data between ground and space and vice-versa.

Soviet Union had launched the world's first artificial satellite named, Sputnik 1 in 1957. Nearly after 18 years, India also launched the artificial satellite named, Aryabhata in 1975.

Need of Satellite Communication

The following two kinds of propagation are used earlier for communication up to some distance.

- Ground wave propagation − Ground wave propagation is suitable for frequencies up to 30MHz. This method of communication makes use of the troposphere conditions of the earth.
- Sky wave propagation − The suitable bandwidth for this type of communication is broadly between 30–40 MHz and it makes use of the ionosphere properties of the earth.

The maximum hop or the station distance is limited to 1500KM only in both ground wave propagation and sky wave propagation. Satellite communication overcomes this limitation. In this method, satellites provide communication for long distances, which is well beyond the line of sight.

Since the satellites locate at certain height above earth, the communication takes place between any two earth stations easily via satellite. So, it overcomes the limitation of communication between two earth stations due to earth’s curvature.

How a Satellite Works

A satellite is a body that moves around another body in a particular path. A communication satellite is nothing but a microwave repeater station in space. It is helpful in telecommunications, radio and television along with internet applications.

A repeater is a circuit, which increases the strength of the received signal and then transmits it. But, this repeater works as a transponder. That means, it changes the frequency band of the transmitted signal from the received one.

The frequency with which, the signal is sent into the space is called as Uplink frequency. Similarly, the frequency with which, the signal is sent by the transponder is called as Downlink frequency. The following figure illustrates this concept clearly.

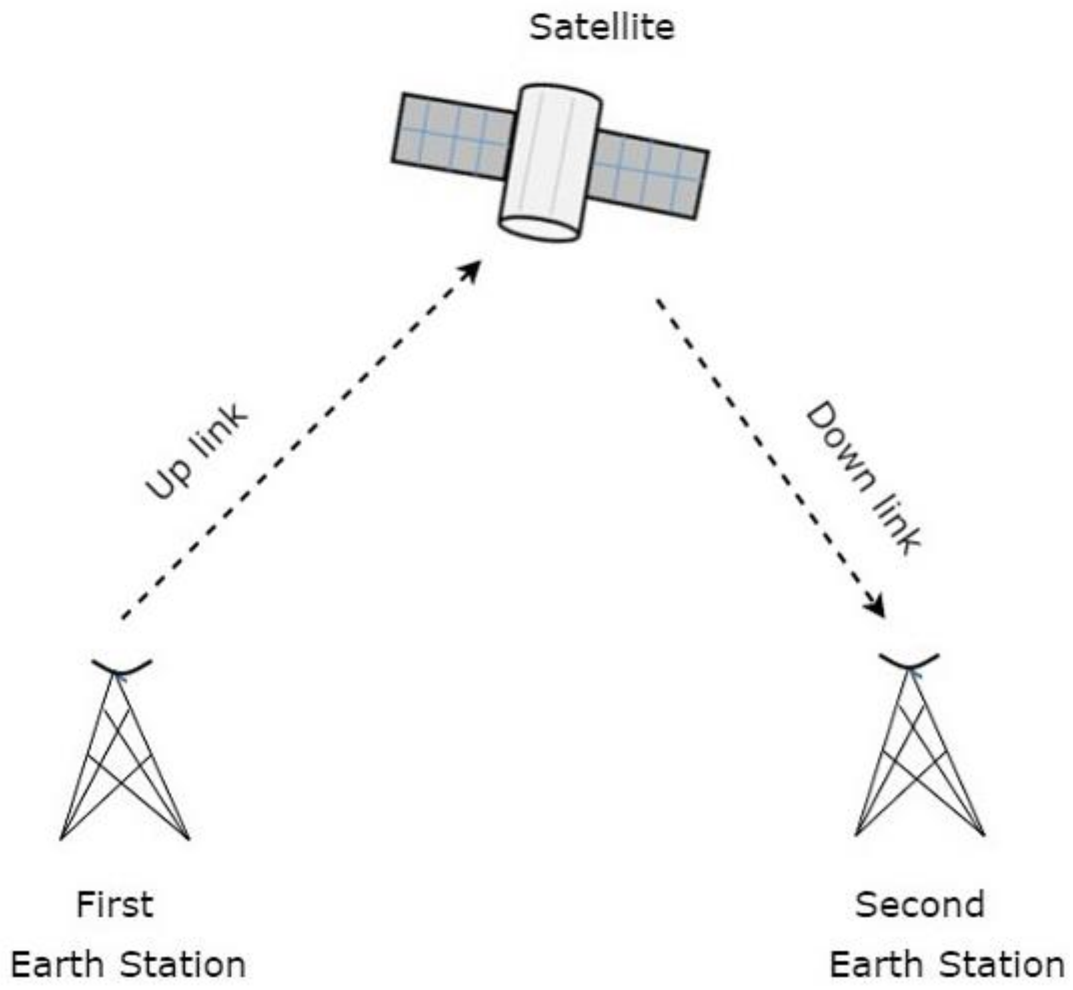


Fig 5.5 Satellite Communication

The transmission of signal from first earth station to satellite through a channel is called as uplink. Similarly, the transmission of signal from satellite to second earth station through a channel is called as downlink.

Uplink frequency is the frequency at which, the first earth station is communicating with satellite. The satellite transponder converts this signal into another frequency and sends it down to the second earth station. This frequency is called as Downlink frequency. In similar way, second earth station can also communicate with the first one.

The process of satellite communication begins at an earth station. Here, an installation is designed to transmit and receive signals from a satellite in an orbit around the earth. Earth stations send the information to satellites in the form of high powered, high frequency (GHz range) signals.

The satellites receive and retransmit the signals back to earth where they are received by other earth stations in the coverage area of the satellite. Satellite's footprint is the area which receives a signal of useful strength from the satellite.

Pros and Cons of Satellite Communication

In this section, let us have a look at the advantages and disadvantages of satellite communication.

Following are the advantages of using satellite communication:

- Area of coverage is more than that of terrestrial systems
- Each and every corner of the earth can be covered
- Transmission cost is independent of coverage area
- More bandwidth and broadcasting possibilites

Following are the disadvantages of using satellite communication −

- Launching of satellites into orbits is a costly process.
- Propagation delay of satellite systems is more than that of conventional terrestrial systems.
- Difficult to provide repairing activities if any problem occurs in a satellite system.
- Free space loss is more
- There can be congestion of frequencies.

Applications of Satellite Communication

Satellite communication plays a vital role in our daily life. Following are the applications of satellite communication −

- Radio broadcasting and voice communications
- TV broadcasting such as Direct To Home (DTH)
- Internet applications such as providing Internet connection for data transfer, GPS applications, Internet surfing, etc.
- Military applications and navigations
- Remote sensing applications
- Weather condition monitoring & Forecasting

Electronic Mail

Electronic Mail (e-mail) is one of most widely used services of Internet. This service allows an Internet user to send a message in formatted manner (mail) to the other Internet user in any part of world. Message in mail not only contain text, but it also contains images, audio and videos data. The person who is sending mail is called sender and person who receives mail is called recipient. It is just like postal mail service.

Components of E-Mail System

The basic components of an email system are User Agent (UA), Message Transfer Agent (MTA), Mail Box, and Spool file. These are explained as following below.

User Agent (UA)

The UA is normally a program which is used to send and receive mail. Sometimes, it is called as mail reader. It accepts variety of commands for composing, receiving and replying to messages as well as for manipulation of the mailboxes.

Message Transfer Agent (MTA)

MTA is actually responsible for transfer of mail from one system to another. To send a mail, a system must have client MTA and system MTA. It transfer mail to mailboxes of recipients if they are connected in the same machine. It delivers mail to peer MTA if destination mailbox is in another machine. The delivery from one MTA to another MTA is done by Simple Mail Transfer Protocol.

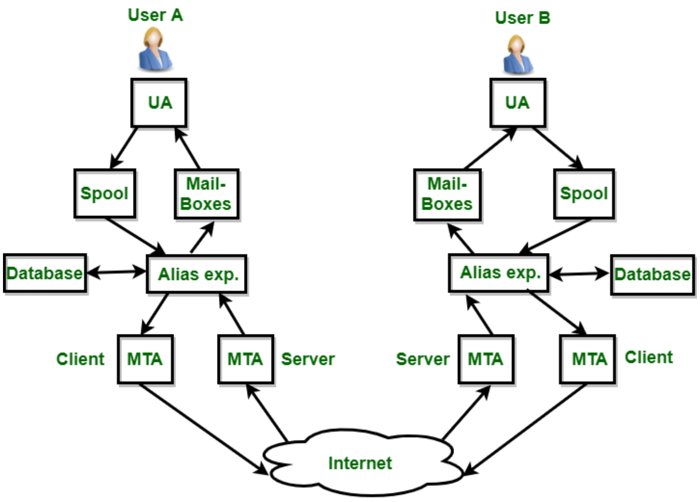


Fig 5.6 Electronic Mail

Mailbox

It is a file on local hard drive to collect mails. Delivered mails are present in this file. The user can read it delete it according to his/her requirement. To use e-mail system each user must have a mailbox. Access to mailbox is only to owner of mailbox.

Spool file

This file contains mails that are to be sent. User agent appends outgoing mails in this file using SMTP. MTA extracts pending mail from spool file for their delivery. E-mail allows one name, an alias, to represent several different e-mail addresses. It is known as mailing list, Whenever user have to sent a message, system checks recipients's name against alias database. If mailing list is present for defined alias, separate messages, one for each entry in the list, must be prepared and handed to MTA. If for defined alias, there is no such mailing list is present, name itself becomes naming address and a single message is delivered to mail transfer entity.

Services provided by E-mail system:

Composition
The composition refer to process that creates messages and answers. For composition any kind of text editor can be used.

Transfer
Transfer means sending procedure of mail i.e. from the sender to recipient.

Reporting
Reporting refers to confirmation for delivery of mail. It help user to check whether their mail is delivered, lost or rejected.

Displaying
It refers to present mail in form that is understand by the user.

Disposition
This step concern with recipient that what will recipient do after receiving mail i.e save mail, delete before reading or delete after reading.

Power Line Carrier Communication

The figure 1 shows a basic PLCC network used in power substations. The Power line carrier Communication (PLCC) uses the existing power infrastructure for the transmission of data from sending to receiving end. It works in full duplex mode. PLCC system consists of three parts:

The terminal assemblies include the receivers transmitters and protective relays.

The coupling equipment is the combination of line tuner, coupling capacitor and the wave or line trap.

The 50/60 Hz power transmission line serves as path for relaying data in the PLCC bandwidth.

Coupling Capacitor

It forms the physical coupling link between transmission line and the terminal assemblies for the relaying of carrier signals. Its function is to provide high impedance to power frequency and low impedance to carrier signal frequencies. They are usually made up of paper or liquid dielectric system for high voltage application. The ratings of coupling capacitors range from 0.004-0.01µF at 34 kV to 0.0023-0.005µF at 765kV (source: IEEE).

Drain Coil

As shown in the figure 1 the purpose of drain coil is to provide high impedance for carrier frequency and low impedance for power frequency.

Line Tuner

It is connected in series with the coupling capacitor to form a resonant circuit or carrier signal frequency high pass filter or band pass filter. Its function is to match the impedance of the PLC terminal with the power line in order to impress the carrier frequency over the power line. In addition it also provides isolation from power frequency and transient overvoltage protection.

Line Trap or Wave Trap

It is a parallel L-C tank filter or band-stop filter connected in series with the transmission line. It presents high impedance to carrier signal frequencies and very low impedance to the power frequency. It consists of

1. *Main coil*

   An inductor that is connected directly to the high voltage power line carries power frequency.
2. *Tuning device*

   It may be a capacitor or a combination of capacitor, inductor and resistor, connected across the main coil in order to tune the line trap to the desired blocking frequency.
3. *Protective device*

   It is usually a gap type surge arrester used to protect the line trap from damage due to transient over-voltages.

The line trap or wave trap prevents unwanted loss of carrier signal power and also prevents carrier signal transmission to adjacent power lines. Line traps or wave traps are available for narrow-band and wide-band carrier frequency blocking applications.

Power Line Channel Characteristics

- *Characteristic Impedance*

  The characteristics impedance of transmission line is given by :

  Where, L is the inductance per unit length in Henry(H).
  C is the capacitance per unit length in Farad(F).
  It varies in the range of 300-800 Ω for power line communication.
- *Attenuation*

  It is measured in decibels(db). Attenuation losses can be due to the impedance mismatching, resistive losses, coupling losses and various other losses that occur in the line trap, line tuner, power line etc.

- *Noise*

  The signal-to-noise ratio(SNR) must be high at the receiving end, other wise the carrier frequency shows erratic patterns at the receiving end. The noise level limit the attenuation that PLCC channels can tolerate.
- *Bandwidth*

The wider bandwidth means faster the channel, but it also leads to the accentuation of noise. For relaying purpose, AM channel bandwidth is around 1000Hz to 1500Hz and for FSK bandwidth it is 500Hz to 600Hz (source: IEEE).

Applications of PLCC in Power Systems

- *Protective Relaying*

  For the purpose of carrier aided protection, PLCC channels use modulation schemes namely the Amplitude modulation(AM) for blocking schemes and Frequency Shift keying(FSK) for unblocking, permissive and direct-trip schemes.

- *Telemetry*

  It is used to monitor electrical quantities like voltage, current, power etc. at remote locations. The analog data is converted in binary which is used to shift the FSK frequency HIGH and LOW and then transmitted over narrow band SSB channel.
- *Telephony*

  Voice messages are sent over SSB narrow band mode with bandwidth ~3khz.

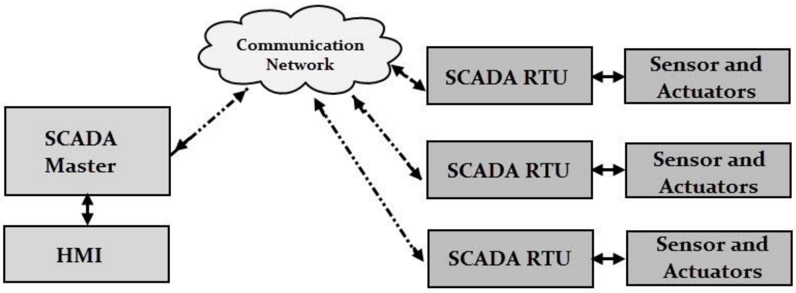
- *Home Automation and Home Networking*

  It is classified as low voltage power line communication. Using low voltage electrical network at home to control appliances by sending or receiving data through power line. It is used as narrow-band PLCC for home automation and metering purposes, and broadband PLCC for internet.

SCADA

As the name defines, SCADA system provides supervisory control, monitoring and management of various industrial automation systems (such as manufacturing and process control automation systems) by acquiring and analyzing the data from remote devices.

It gathers the real-time data from various remote locations or plants, presents the data on various HMIs, records and logs the data on SCADA database management.



It provides the centralized monitoring and control system by integrating data acquisition and control systems with a telemetry system (in other words with data transmission systems) and HMI software.

SCADA systems are designed to gather field information such as sensor output, status of various machines and set limits of various process variables, and to transfer it to the central computer thorough wireless communication systems.

At the monitoring side, the received information is displayed graphically or textually to the operator, thereby it allows the operator to monitor, analyze and control an entire system from a central control location.

Depending on the control system implemented, the control of any individual system can be performed manually by the operator commands or can be automatic using various closed loop control strategies.

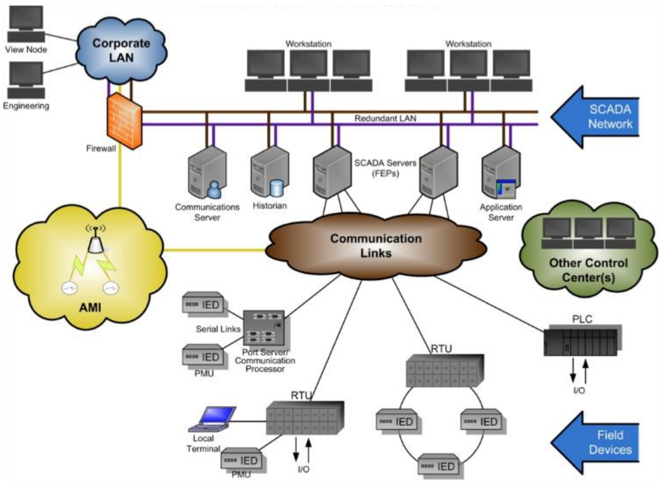


Fig 5.7 **Generic SCADA system**

SCADA system combines both hardware and software components. SCADA hardware includes MTU (master terminal unit) which is placed in a central location, communication equipment such as telephone line, radio, cable, or satellite, and one or more RTUs (remote terminal units) or PLCs which are placed at geographically distributed field sites.

These RTUs or PLCs are connected with various sensors and actuators and are responsible for gathering the data and controlling the field parameters. The Master Terminal Unit (MTU) collects and processes the data from RTU or PLC inputs and outputs, while the PLC or RTU controls field devices or a local process.

SCADA software performs the functionalities of a SCADA such as what and when to acquire and control, storing and accessing of acquired data, calculating parameters acceptable range (set limit checking), responding to parameter violations (beyond the range), providing HMI, reporting and accounting, generating alarms, etc.

There are different SCADA vendors, some of those include Siemens, ABB, Honeywell, Rockwell, Schneider Electric, Technomatix and Tibbo Systems.

Architecture of SCADA

A typical architecture of a SCADA system is shown in below figure which describes the general configuration and major components of the SCADA. In this, control area or master station house main server which acts as MTU and communication routers.

Also, the control area includes engineering workstations, HMI stations, data servers, and data historians, which are all connected by local area network (LAN). The master station is responsible for monitoring and controlling various remote stations.

Architecture of SCADA

Remote stations are equipped with one or more RTUs/ PLCs which performs local monitoring through sensors (such as voltage, current, temperature and pressure) and local controlling through actuators (such as pumps, relays, and valves). The field site devices are connected through WAN network to perform remote diagnostics.

The communication or transferring of information between field sites and control center is carried through proprietary communication protocols over serial communications using telemetry techniques such as cable, telephone line and RF.

Components of SCADA or Block Diagram of SCADA:

The major components of SCADA include

Master Terminal Unit (MTU)
It is the heart of the SCADA system, which can be a dedicated computer, a Programmable Logic Controller (PLC), or a network server that communicates with remote field side RTUs.

It initiates all communication, collects the data, stores the data in database, provides interfaces to operators and sends the information to other systems.

It allows the users to perform controlling functions on field devices such as breakers, switches and other actuators depending on the gathered data. It continuously communicates with other devices in master station so as to facilitate data logging, alarm processing, trending and reporting, graphical interface and security system.

Remote Terminal Units (RTUs)
RTUs gathers the information from various field sites in which they are employed. Each RTU is connected with various sensors and actuators that manage local process or field equipments.

It collects the information from various sensors and sends the information to the MTU. Also, it receives the control commands from MTU and correspondingly controls the various actuators.

Remote Terminal Units

Many RTUs store the data in their database and waits for a request from the MTU to send or transmit the data. In sophisticated systems, PLCs are used as RTUs which directly transfers the field data and controls the parameters without a request from the MTU. It uses a local area network to communication with various field intelligent devices.

Communication Equipment/Network
It provides the link between RTUs (in the field) to MTU (in the control center). The communication can be wired or wireless or through internet which provides bidirectional and uninterrupted communication between RTU and MTU.

SCADA systems can be connected using various communication mediums including twisted pair cables, coaxial metal cables, fiber optic cables, satellites, high frequency radio, telephone lines, and microwave radio.

The topology of the SCADA system network depends on the type of system or application it is intended for. Mostly redundant topology is recommended for critical control applications.

SCADA Software
It is an important aspect of every SCADA system which presents the information to the user and also allows the user to intervene in the process control. Many SCADA systems use commercial proprietary software upon which SCADA system is developed.

This software comprises a computer operating system which controls the central host computer hardware, communication network management, and graphical generation tool for HMI, database management and report generation tools.

Functions of SCADA

1. Data Acquisition
In SCADA systems, MTU performs the periodic acquisition of data from RTUs. As discussed above that the RTU can respond in either a request form the MTU or continuously transferring the data when changes of state of a parameter takes place or when limits of the parameter exceeded, even without a request from the MTU.

The data acquisition process includes internal scanning of RTU internal database, periodic RTU polling by MTU, transmission of data by RTU to MTU, scaling of data into engineering units and updating a previous value or state in the database.

2. Human Machine Interface (HMI)
SCADA products display the information on multiple screens, which combines both text and synoptic diagrams. It provides the provision for human operators to continuously monitor the operations and to intervene when necessary.

SCADA HMI software consists of library of graphical symbols to which tag names are associated for a particular device or parameter such as ON/OFF status of switch, level information on tank, etc.).

Display selection on HMI is organized mostly in a tree structure, where index pages allow human operator to select various displays using a cursor, keyboard, trackball, or touch-screen positioning techniques.

3. Supervisory Control
It is the process of controlling the equipment operations from remote locations. In SCADA systems, the MTU in the master station sends the control instructions such as set points and discrete control commands to the RTU at remote station. At the remote stations, RTU receives the commands and accordingly controls the appropriate actuator.

The supervisory control includes selection of the remote station, choosing the device to be controlled and executing the desired command such as close or trip. Most of the systems employ check-before-operate method for correct selection and operation of the equipment in the remote place.

4. Trending
All SCADA products provide trending facilities which display the gathered (real-time) or saved (historian) data on various charts. The parameters to be trended on a specific chart can be defined online or it can be predefined.

These charts are able to display one or more parameter using one or more plots. It provides the automatic scrolling of data with enhanced zoom features. Historian trending is possible with archived databases.

5. Alarm Processing
It involves in alerting the operator to unscheduled events by informing place of occurrence, time of occurrence, device ID and nature of the event.

Alarms are logically programmed on the master control station by comparing the received data with appropriate limits. It is possible to handle alarms on multiple priority level. Alarms can be suppressed either by individual or as a complete group.

6. Information Storage and Reports
SCADA stores the gathered data on either disks or permanent storage devices. The logging of data is performed on a cyclic basis, which means the time span of a rotating historical file is limited (which can be 40 days or 12 months).

Once the period is completed or the log is full, it archives the data to permanent storage device and then the information older than the file time span is discarded. This allows the user to retrieve and analyze the data whenever it is needed.

SCADA provides the report generation using SQL type queries. The historical file provides the source of information for generating various reports. SCADA also facilitates to print and archive reports.

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2. Anokh Singh, “Principles of Communication Engineering”, S. Chand & Company Ltd, 2006

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4. Louis Frenzel, "Communication Electronics”, McGraw-Hill Companies,1999.

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Question Bank

Part A

1. Define refractive index of optical fibre.
2. Define the frequency reuse factor in cellular communication.
3. Mention the use of Mobile Switching Centre.
4. What is SID?
5. Mention the advantages of PLCC.
6. What is the use of spool in electronic mail communication?
7. What is MTU in SCADA?
8. Mention the use of Optical splice in optical fibre communication.

Part B

1. Explain Facsimile system.
2. Explain optical fiber communication system.
3. Explain the concept of cellular communication.
4. Explain the Satellite communication system.