ELECTRONICS & TELECOMMUNICATION ENIGINEERING

SEMESTER : V

Subject: Audio, Video & TV Engineering

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<u>Sound Waves</u>

<u>Wave Motion</u>:-

A disturbance caused by repeated periodic wave motion.

These particles vibrate about their mean positions but the disturbance travel forward.

A wave is characterized by Amplitude, Freq, Phase, Wave length and velocity.

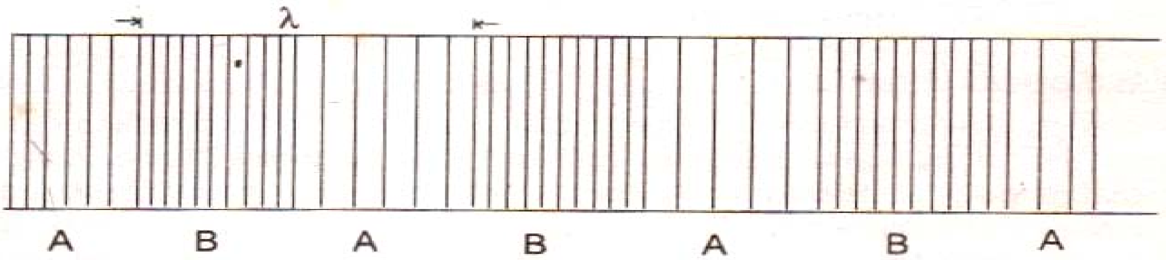
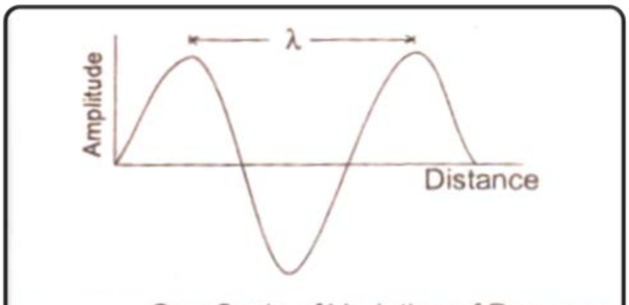


Fig. 1.1. Longitudinal wave, A Rarefaction B Compression

When a disturbance occurs, the particles of air form alternate regions of compression and rare fraction. Fig (1)

In the compression region the distance between the particles of the medium is less than normal while in rare fraction region those distances are more than normal.

If we plot a graph showing the relation between the variation of pressure above and below the normal atmospheric pressure and the distance from the source of disturbance, we get the familiar sinusoidal wave (Fig b). One compression and one rare fraction together constitute one wave.



One Cycle of Variation of Pressure

<u>Amplitude</u> : Amplitude of the wave is the max intensity of compression (Or rare fraction).

Frequency: The no. of successive compressions and rare fractions occurring in one second is the frequency of the wave. (Hz).

Wave Length: The distance between two successive regions of compression (Or rare fraction) is known as wave length.

Velocity: The speed with which the region of compression and rare fraction travels is the speed (Or velocity) of the wave.

Time period: The time taken to complete one fully cycle is known as time period (i.e. T) of the wave.

$T = \frac{1}{f}$

The distance travelled in a time T seconds is one wave length

$\therefore$ Velocity $= \frac{\text{Distance}}{\text{Time}} = \frac{\lambda}{T} = \mathbf{f\lambda}$

Sound Waves:

_ **A** Change of air pressure causes sound waves.

_ The variation of air pressure reaches the ear drum and is converted into electrical signals.

_ The nerve pulses carry these signals to the brain and a sensation known as sound is produced.

_ Ear is a very sensitive transducer and can convert even very minute changes in air pressure.

_ Sound wave travel as longitudinal waves and have all the features of waves i.e. amplitude, velocity, frequency, wave length and phase.

= Sound waves need a medium to travel. At O'C, the velocity of sound waves in air is 330 m/s

Newton's formula for velocity of sound is

In air $v=\sqrt{(E/p)}$ ……………(1)

Where ω = Velocity if sound waves.

E= Coeff. of elasticity of medium

Φ= density of medium

In gases, the velocity of sound waves is directly proportional to square root of absolute temp. of gas.

$(v1/V2)=\sqrt{(T1/T2)}$ …………..(2)

Where V1 is the velocity at absolute temp T1 and V2 is the velocity at absolute temp. T2.

Eqn (2) can be shown that velocity of sound in air increases by 0.608 m/s for every 1°C rise in temp.

Audible Frequency Range:

The audible freq range for human being is 16Hz to 20 KHz. Human ear cannot respond to freqs below 16 Hz or above 20 KHz.

The freqs less than 16Hz are known as infra-sonic freqs. While those above 20KHz are known as ultra-sonic freqs.

The exact audible frequency range varies from person to person. The upper limit decreases with age.

Some animals can hear sound waves much above the 20 KHz.

Characteristics of Sound:

The important characteristics of sound are pitch, loudness (Or intensity), quality (or timbre).

Pitch: Pitch determines the shrillness of sound.

It varies almost directly with freq.

A sound may have a high pitch (eg Cry of a Child) or low pitch (a Soft voice)

- The voice of an old man has low freq & low pitch.
- The sound produced by a mosquito has high pitch and high freq.
- Humming of bees also produces a high pitch sound.
- Pitch also depends on the relative motion between the listener and the source of sound. When the listener is approaching the source, the pitch of sound appears to be rising. When listener is recoding (going way) from the source, the pitch appear to decrease.

Intensity and Loudness:

Intensity of sound waves, at a point is the average rate of flow of wave energy through a unit cross sectional area at right angle to the direction of propagation waves. Intensity is expressed in watts per square metre and is proportional to square of amplitude.

Loudness of sound is the degree of sensation of sound produced in the ear. Loudness depends on intensity of sound waves and the response or sensitiveness of the ear for waves of that freq.

The intensity of sound depends on many factors such as :-

1) The intensity is directly proportional to the density of the medium in which the sound originates.

2) α the square of the amplitude of the wave.

3) α the size of the vibrating body. A body having larger size produces louder sound.

4) α 1/ Square of the distance from the source.

Intensity is a physical quantity, it is not dependent on the response of ear and can be measured by a physical apparatus.

Loudness depends on response of ear also.

Intensity of sound waves is usually expressed in dbs $=10 \log 10 \frac{P1}{P2}$

$= 20 \log 10 \frac{A1}{A2}$

Where P1 & P2 are the powers in watts and A1 and A2 are the amplitudes.

The amplitude of a sound wave is also expressed in units of pressure is Newton/m² or Pascal (Pa).

The minimum intensity of sound which a human ear can respond is known as threshold of hearing and is equal to 10^{-12} watts/ m² or 20 x 10^{-6} N/ m². This is taken as 0 dB. All other intensities are represented in dB with this threshold as reference.

Reverberation:

The sound (music as well as noise) produced in a room suffers multiple reflections from the walls, ceiling and floor. Thus, a listener hears direct sound as well as reflected sound. Because of this reason the listens continues to hear sound for sometime even after the original sound has ceased. This phenomenon is called reverberation and is due to following reasons:

1) The intensity of sound decreases exponentially and therefore takes some time to reach zero value.

2) The sound persists in the room due to reflections.

3) The original sound and reflected sound are out of phase. Therefore, intensity of sound may increase (if original and reflected sound are in phase) or decrease (if original and reflected sound are in phase opposition).

The time taken by a continuous sound to decrease to one-millionth of its original value (10^{-6}) (if decrease by 60 dB) is called reverberation time.

Acoustics:

A room is said to acoustically good if every syllable or musical note reaches every point in the room with a sufficient loudness and then this sound dies away quickly so that next syllable or musical note can be heard.

The factors affecting the acoustics are

a) Reverberation (Should not be too large)

b) Satisfactory Loudness (essential for proper hearing)

c) Freedom from Echoes. (If direct & reflected sound reach a listener with a time interval of about 1/7 second echo may result.

d) Absence of Resonance. (When window panes, wood portions etc start vibrating, un-wanted noise may result. The noises may also be in resonance with some audio frequencies.

e) Freedom from extraneous Noise.

MICROPHONES

In all audio systems sound energy has to be transmitted to distant places. This can be done by converting variations of sound pressure into electrical impulse. A microphone is a transducer which converts variations of sound pressure into electrical signals of same freq. The amplitudes of electrical signals are proportional to intensity of sound waves. It is necessary to amplify these electrical signals using an audio amplifier before these signals can be transmitted through wires (as in PA Systems)or by wireless means (as in radio & TV).

Quality of Microphones :- (Charterstics of Microphones)

a) **Sensitivity:** It is the output in milli volt for a sound pressure of 0.1 Pa (or 0.5 x 10^{-4} watts/ m^2) at 1000 Hz. Many time sensitivity is expressed in dB below IV.

b) **Frequency Response:** Audio freq range extends from 16 Hz to 20 KHz. However an audio system is considered to be very good if it has a flat response from 40 Hz to 15 KHz. The freq response of a microphone is described in terms of its bond width of frequencies in the O/P with in ±1 dB of the O/P at 1000 Hz.

c) **Output impedance:** **The O/P** impedance of a microphone is expressed in Ohms. The importance of this parameter arise because of the necessity to transfer maximum power to the transmission Circuit. As per max. power transfer theorem, the power transferred to the load (tx^n system in this case) is max, when load resistance is equal to source resistor . To ensure max. power transfer, resistance of load and source can be matched by the process of impedance matching. For this purpose the microphone is connected to a two winding transformer whose O/P is fed to the tx^n system. The transformer changes the impedance level. Let N1 & N2 be the no of turns in primary and secondary windings and Z1 & Z 2 be the impedances as seen from primary & secondary terminals. From the transformer theory, we have

$$\frac{Z1}{Z2} = \left(\frac{N2}{N1}\right)^2$$

d) **Distortion:-** A microphone may suffer from non-linear distortion, freq distortion and phase distortion.

_ **Non-linear distortion means** that the relative amplitude of diff. frequencies in the electrical O/P is not same as in the sound wave. For high fidelity sound systems, non-linear distortion should be less than 1%.

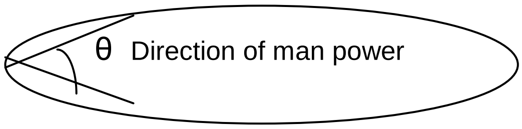
_ Freq. distortion means that some frequencies not present in the sound signal appear in the O/P. This is due to natural resonant freq of the movable parts of the microphone.

_ Phase distortion means that the inter-phase relationship between different freq. components in the O/P is not same as in the i/p. This generally occurs when more than one microphone are used in system.

e) **<u>Directivity:-</u>** The response or sensitivity of a microphone is not the same for sounds coming from different directions. This effect is known as directivity. A microphone may be unidirectional (i.e. it can pick up sound only from one direction), bidirectional (i.e. it can pick up sounds from two directions) or Omni directional (i.e. it can pick up sounds from all directions). However for an Omni directional microphone the response for sounds coming from different directions is not same.

Directivity in terms of angle θ which is the angle between half power points.

Half Power Point



Half Power Point

Directivity is also given by 20 log D

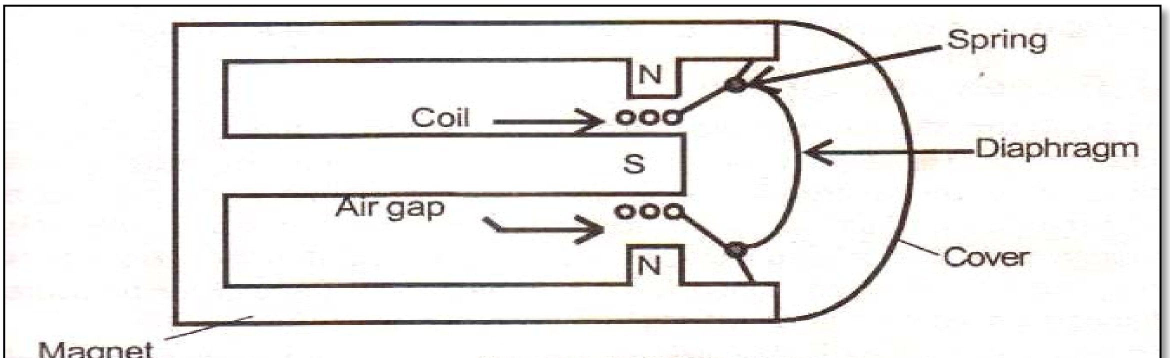
Where D=<u>Actual O/P in the direction of max. O/P</u>______________

O/P in that direction for an Omni directional microphone.

<u>Moving Coil Microphone:</u>

a) **<u>Principle</u>**: A moving coil microphone works on the principle of induced emf (Faraday's Law of electromagnetic induction). The variations of sound pressure cause the motion of a coil in a magnetic field. Thus an emf is induced in the moving coil. It is known as a dynamic microphone.

b) **Construction:-** Fig shows the construction.



Moving Coil Microphone

It consists if a permanent magnet, generally POT type with a central south pole and peripheral north pole. The magnet is so shaped as to give a uniform field in the air gap. A diaphragm made of non-magnetic material is fixed to the body by springs. A coil wound on card board cylinder is attached to the diaphragm and is free to move in the air gap as the diaphragm vibrates due to sound wave. A protective cover saves the delicate diaphragm and other parts from mechanical damage.

c) **Working:-** **W**hen sound waves strike the diaphragm it moves forward and backward. The coil also moves along with diaphragm and an emf is induced in the coil. The magnitude of emf is given by

e = B ℓϖ

Where e=emf, V, B= flux density wb/ m^2 or T

ℓ = Length of conductor / Coil, m

ϖ = Velocity of coil, m/sec.

Since the emf is proportional to velocity of coil, it is designated as velocity microphone. The motion of coil depends on pressure of sound waves. Therefore, it is a pressure microphone.

d) **Features:-**

1) Its sensitivity is about 90 db below IV when sound pressure is O.1 Pa. The voltage induced is about 30 mV.

2) The O/P impedance is low, around 30 to 40 ohms. If it is to feed a transmission line of impedance 250 Ohms, an intervening step up transformer of turn ratio about 3 :ics needed.

3) Its freq response is nearby flat in the 40 Hz to 8 Khz range.

4) It is an Omni directional microphone.

5) Its signal to noise ratio is about 25 dB.

6) Its distortion is less than 5%.

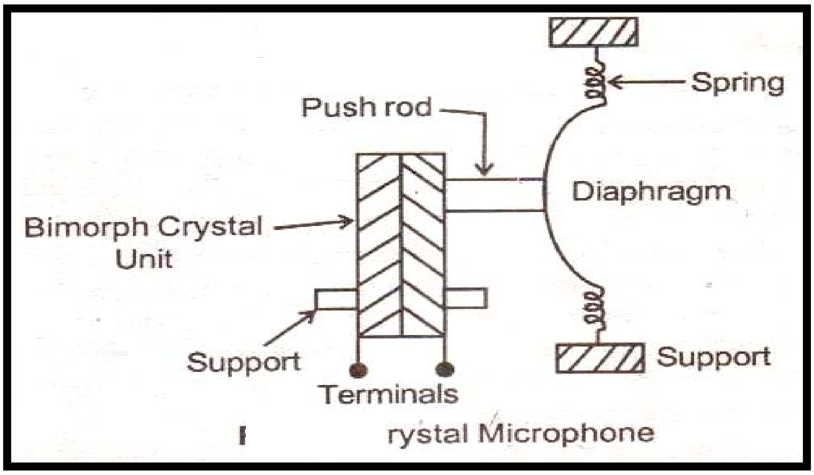
e) **Advantages:-** All its features are good. Overall performance is satisfactory. It is mechanically robust and is not affected by weather. It is cheaper than ribbon and condenser microphones but is more costly as compared to crystal and carbon microphones.

f) **Dis-advantages:-** It has to be kept within 25 cm distance from the source of sound. Its weight is considerable because of heavy magnet.

g) **Applications:-** P A Systems and broad cast studies.

Piezo Electric Microphone or Crystal Microphone:-

a) **Principle:-** Its principle is based on piezocrystal effect. As per this effect, the application of pressure across the faces of a crystal, produces a voltage across the faces. The crystals which exhibit this property are Rochelle salt, quartz, ceramic etc.
Ceramic is the most commonly used material.



b) **Construction:-** The crystal is in the form of bimorph crystal unit (two platter put together form the unit). Metal foils are attached to the crystal surfaces to serve as terminals. A diaphragm, generally made of aluminum is held between supports through springs. The diaphragm is connected to the bimorph through a push rod.

c) **Working:-** When sound waves impinge on the diaphragm it vibrates. These vibrations are picked up by the bimorph element and a voltage is developed across its faces. The voltage developed is proportional to pressure of sound wave. This voltage is generally fed to an amplifier.

d) **Features:-**

1) Its sensitivity is good about 50 mV (i.e. 26 dB below IV) for a sound pressure of 0.1 Pa.

2) Almost flat freq. response between 80 Hz to 8000 Hz.

3) The noise generated within this microphone is lows. Hence it has high signal to noise ratio (about 40 dB).

4) Its O/P impedance is very high, about 1 M Ω or so.

5) Its distortion is low, less than 1%.

6) It is Omni directional.

e) **Advantages:-** Good sensitivity, high SNR, low distortion, small size, Omni directional characteristics, low cost.

f) **Disadvantages:** Less rugged than moving coil, cannot withstand high temp, very low freq sound waves are attenuated. It is affected by moisture.

g) **Application:** Home recording, mobile communication systems, amateur communication.

Special Microphones:-

a) **Cardiod Microphone:-** It consists of a moving coil and ribbon microphone in series. It has a directivity pattern of the shape of heart.
It is suitable for court scenes in dramas.

b) **Tie Clip Microphone::-** It is an electrets Microphone. It can be clipped to the tie or collar. It is very light weight, very small size and very cheap.

c) **Wireless Microphone (Cordless Microphone):** It uses a small VHF transmitter and therefore does not need a wire. Used in sports meets.

d) **Lovaller Microphone:** It is a small moving coil microphone and can be suspended on the chest by cord around neck. It is suitable when the person using it has to move around.

e) **<u>Differential (Or Noise cancelling) Microphone:</u>**- It uses two similar microphones mounted a small distance apart in the same case. The two microphones are connected together in opposite phase. Noise Originating anywhere around it will affect both almost equally. Since they are connected in opposite phase, the noise in two microphone will cancel out. They are used in noisy environment i.e. sports meet, factories etc.

LOUD SPEAKER

Loud speaker is an essential component of all audio systems. It converts electrical audio freq signals into sound waves of the same frequency. Thus its work is opposite to that of microphone.

The i/p to a loud speaker may come from the audio system directly (as in 2 in 1 & PA systems) or from the antenna (as in radio & TV).

Features of Loud Speaker:-

A Loud speaker must convert the electrical signals into sound waves as faithfully as possible without any distortion and into the whole of AF range.

a) **Sensitivity:** Sensitivity is the i/p signal required to give a sound pressure of 0.1 Pa at a distance of one meter from the loud speaker.

Sometimes the word efficiency (ratio of o/p sound power to i/p AF electrical power) is used to describe the ability of the speaker to convert electrical signals into sound waves.

b) **Frequency response: The** frequency response is the range of frequencies having an amplitude ±1dB. Ideally the frequency response should flat over the whole audio range i.e. $16H_z$ to 20 .

However the mass of the diaphragm of the speaker attenuates high frequencies. The speaker will always have a natural resonant frequency. The i/p signal at this frequency is strengthened giving rise to non-linearity.

c) **Distortion: Distortion** may be in amplitude, frequency and phase. Causes of distortion are non –linear behavior of magnetic field, mass of diaphragm, natural frequency of the mechanical parts etc.

d) **Directivity:** It is defined as the ratio of sound intensity in the direction maximum response to the sound intensity which would be exist if the speaker is omnidirectional.

e) **Input Impedance**: It is expressed in Ohm. For maximum power o/p from the speaker, its i/p impedance should be equal to the source impedance so that power impedance matching is obtained.

Moving Coil (Cone Type) Loud Speaker:

a) **Principle**: Its principle is same as that of a motor. It is known as direct radiating loud speaker. A coil known as Voice coil, is placed in the magnetic field of permanent magnet. When electric current passes through the coil, a force acts on the coil causing the coil to move (Vibrate). A paper diaphragm vibrates, produces pressure vibrations in air and the result is the sound wave. The force acting on the voice coil causes vibrations of the diaphragm resulting in compressions and rare fractions in the air. Thus electrical signals are converted into sound waves of the same frequencies.

b) Construction: The main parts of this loud speaker are magnet, voice coil, and conical diaphragm.

c)

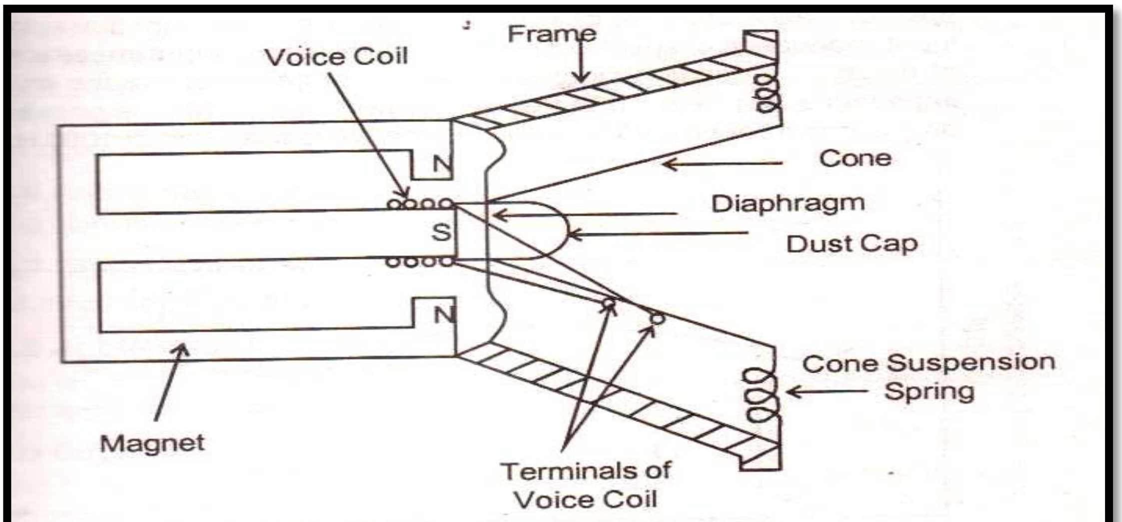


Fig. 6.1. Moving Coil Loudspeaker

(Magnet) T_1 & T_2 =Terminals of Voce Coil.

This loud speaker uses a pot type permanent magnet having central south pole and peripheral north pole. To give a strong magnetic field the magnet is made of special alloy Alnico (10% aluminum, 18% nickel, 12% Cobalt, 6% Copper and 54% Iron). This material has very high retentively and can thus retain magnetism almost indefinite. The magnet gives a strong magnetic field in the air gap. A voice coil is suspended by a suitable suspension in the air gap and is free to move in the air gap when a force action on it. The coil is attached to a conical diaphragm made of special paper. The cone has corrugated construction. Terminals of the voice coil are fixed on the cone surface.

d) Working: When electric current flows through the voice coil, interaction between the field of permanent magnet and the magnetic field of current occurs. Due to this interaction a force acts on the voice coil causing its to and fro motion. Since the conical diaphragm is attached to the voice coil, the diaphragm also vibrates causing compression and rare fractions in air. Thus the electrical signals are converted into sound waves of the same freq. The cone helps in spreading the sound over a large area and thus sound can be heard at a great distance.

e) Features:

1) The efficiency of this loud speaker is rather poor. Only about 5-10% of the signal power is converted into sound.

2) The SNR is about 30%

3) It gives a nearly flat frequency response in the $200H_z$ to $500H_z$ range.

4) The non-linear distortion can be as high as 10%.

5) It is Omni-directional. However the directivity can be modified by using baffles and enclosures so that most of the sound waves go to the area infront of it.

6) The impedance of the voice coil is the i/p impedance of the loud speaker. The impedance is function of frequency because of the inductance of the voice coil. For transfer of maximum power the i/p impedance of speaker should be equal to the o/p impedance of the system feeding signals to the loud speaker. Since the i/p impedance is a function of frequency, perfect matching is possible only at one frequency. The i/p impedance is specified at $1000H_z$.

f) Equivalent Circuit:

Equivalent ckt of a cone type loud speaker is shown below. Source is the source of signals. Vc and Lc denotes the resistance and self inductance of the voice coil, Cm is the mechanical compliance (stiffness) and Lm is the inductance which represents mass of the moving system (Mass Opposes, motion inductance opposes the flow of current. Therefore the effect of mass is always represented by inductance in the equivalent ckt). RL is the load (i.e. resistance of air to the change in pressure).

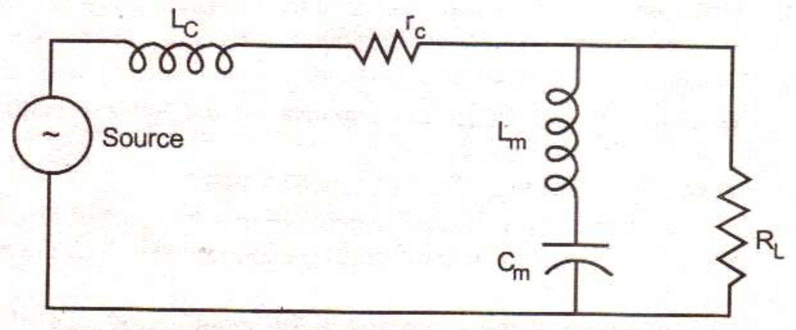


Fig. 6.3. Equivalent Circuit of Moving Coil Loudspeaker

At low frequency, the reactance of Lm (i.e. wLm) is low and shunts the load resistance. Therefore, the o/p at low frequencies is reduced. At high frequencies the reactance of series inductance (i.e. wLc) is high and causes attenuation of high freq components of signal. Effect of Cm is small & negligible.

g) Advantages: Small size, low cost, satisfactory freq response.

h) Disadvantages : Poor efficiency, Vey low & high freqs are attenuated.

i) Capacity Available: Direct radiating cone type loud speaker are available in a no of sizes up to about 25 w.

j) Applications: Radio receivers, TV receivers, cassette players, record players, used in all audio appliances designed for small groups of people.

Baffles and Enclosures:

Loud Speakers of radios, TV and other music systems are enclosed in a cabinets. The function of the cabinet is not only to provide a support the speaker but also to help in proper radiation of sound from the speaker.

Consider the cone type direct radiating speaker, when the cone moves forward, there is compression of air in the front and rarefaction of air at the back. When the cone moves backward, there is compression of air at the back and rarefaction in the front. Thus the sound waves on the front and back are always opposite to each other i.e. have a phase different of 180°. It is possible that these two set of waves tend to cancel each other when the air from back leaks to the front.

The function of the enclosure is to avoid this cancellation and helps in radiation of sound in a such a way that good sound reception is obtained.

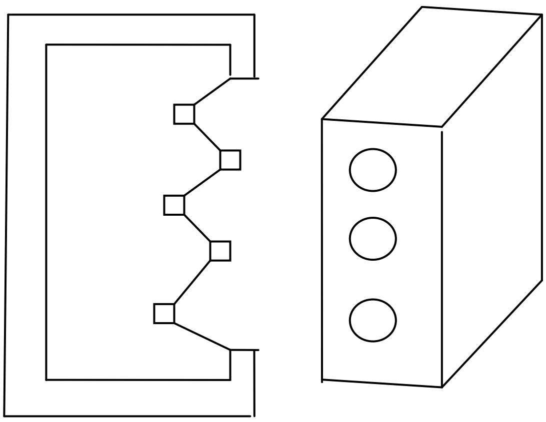
A baffle is a flat wooden board used for mounting the loudspeaker. The enclosure is a closed or partially controlled cabinet.

The above effect of cancellation of sound waves is seen only for low freq sound waves. For such waves the wave length λge and hence the distance from back to is lar the front almost180° out of phase at high frequencies, λ is small. Therefore compressions and arefractions are spaced very closely and occur in quick succession. Therefore, for high freq audio waves there is no cancellation and baffle is not required.

Baffles can classified as finite, infinite, enclosure and bass reflex baffles.

Column Loud Speaker:

When a no of cone type direction radiating speakers are mounted in the same enclosure, it is known as a column speaker.



It is a very suitable configuration for halls and auditoriums. By using this configuration the sound waves can be confined to a narrow beam directed towards the audience.

Many times three separate columns are used one long column for low AFs, a short column for high AFs and medium length column for medium AFs.

By this arrangement the entire freq range of the speech, music etc can be faithfully reproduced giving a very good sound effect.

<u>Woofers and Tweeters</u>:

The AF range from $20H_z$ to 20 KH_z can not be satisfactorily converted by a single speaker. The practical high fidelity range i.e $40H_z$ to 15 KH_z $is\ a\ bit$ lesser than the complete audio spectrum but a single loud speaker cannot give good freq response even for this practical range. The sound waves from the back of the speaker have a phase different of 180° with respect to those in the front. These two set of waves tend to cancel each other thus attenuating the low freq sound. The mass of the diaphragm and cone behave as an inductance and causes attenuation of high freq signals.

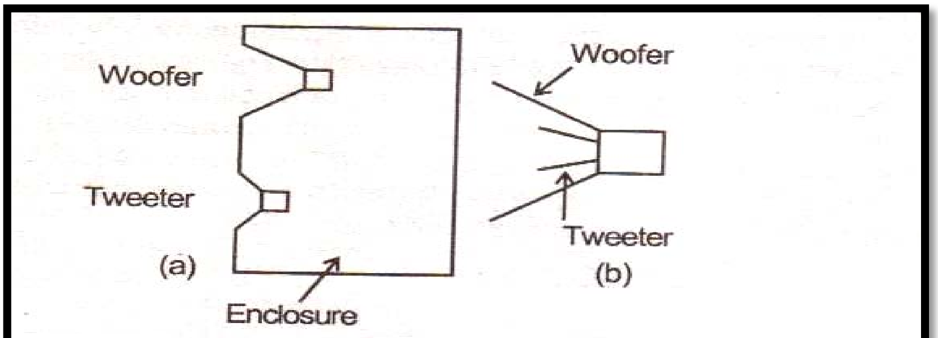
The only practical approach to the solution of the above problem is to have three separate speakers for low, medium & high frequency. The low freq loudspeaker covers $20H_z$ to $50020H_z$ and is known as woofer .

The medium freq load speaker covers $500H_z$ and is known as squawker.

The high freq loud speaker covers the frequencies above $5000H_z$ and is known as tweeter.

Some times the squawker is omitted. In this case woofer covers $20H_z$ to $1000H_z$ range and tweeter from $1000H_z$.

These speakers can be mounted vertically in the same enclosure freq (G). It is also possible to have a dual loud speaker as shown (b). A woofer has a heavy and large diameter cone while the tweeter has a light and small diameter cone.



(*a*) Woofer and Tweeter in a Common Enclosure
(*b*) Dual Cone Loudspeaker

Crossover Ckts:

When a combination of tweeter and woofer is used to give better sound representation. It is essential that the i/p signal be split into two parts i.e. one for freqs less than $1000H_z$ and the other for freqs more than $1000H_z$. Such a Ckt which separates the signal into two parts is known as a cross over Ckt. Such a Ckt splits the signal into two parts at freq known as cross over freq (which is $1000H_z$ mostly). A cross over Ckt is a filter Ckt having a cut-off freq equal to cross over freq.

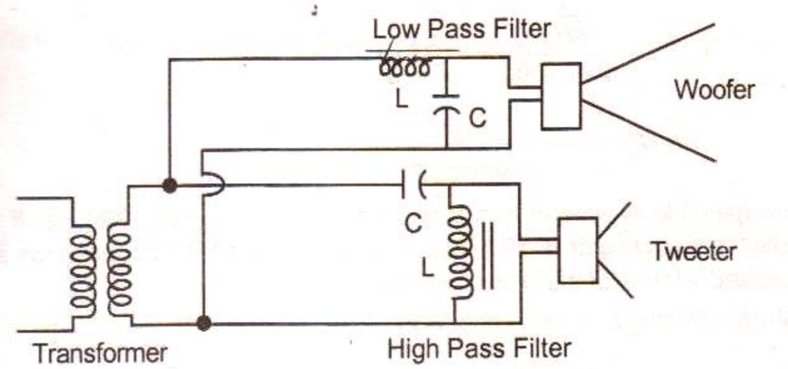


Fig. 6.14. Cross over Circuit for Loudspeakers

Fig shows a cross-over Ckt. The audio freq signal is fed to a transformer for impedance matching. The Ckt feeding the woofer is a LPF having a series inductance and a shunt capacitor. The cut-off freq of this Ckt is $1000 H_z$ to pass through freely but freqs higher than $1000 H_z$ are attenuated. The O/P of this LPF is fed to a woofer. The Ckt feeding tweeter is a HPF having a series capacitance and shunt inductance. If passes all freqs more than $1000 H_z$ but freqs lower than this value are attenuated.

The value of Land C can be found as

L= $\frac{\sqrt{2R}}{2\pi f c}$ C=$\frac{1}{2\sqrt{2}\pi f c R}$

L= inductance, H, C =Capacitance F

R= Loud Speaker resistance Ohm.

Fc= Cross-over freq. H_z .

It is also possible to have a cross-over Ckt for splitting the signal into three parts.

One part ($<500 H_z$) and fed to woofer.

2nd part ($500 H_z < f < 5000 H_z$) and fed to squawker & 3rd part ($>5000 H_z$) and fed to tweeters.

This Ckt will have a LPF having crossover freq or cut off freq of $500 H_z$ to $5000 H_z$ feeds the squawker.

A HPF having a cut off freq of $5000 H_z$feeds the tweeter.

HIGH FIDELITY AUDIO SYSTEMS

The word fidelity means faithfulness. A high fidelity (Hi-Fi) audio system reproduces the original audio programmes with great degree of faithfulness. However the degree of reproduction by any system cannot be 100% exact. So we can achieve high fidelity only to a certain degree.

Features of Hi-Fi Audio System:

A Hi-Fi Audio System has the following features

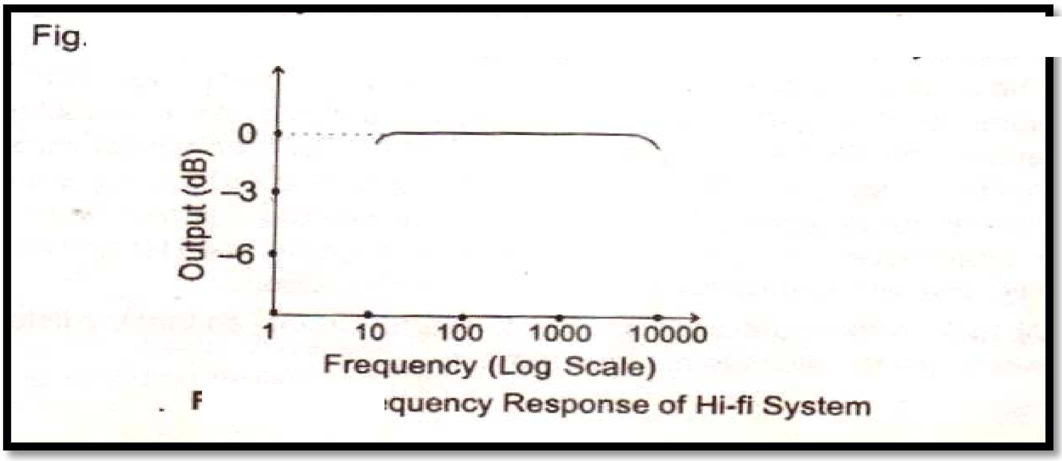
1) SNR should be at least 50dB.

2) Non-linear distortion should be less than 1%.

3) Freq response curve should be perfectly horizontal (within +/- 1 dB) over the audio freq range of 40 Hz to 15 KHz.

4) **The** intensity of sound w.r.t. threshold of heaving should be more than 80 dB.

Even if an audio system is hi-fi, the listeners may not be able to enjoy it fully if the room has external noise. Therefore the conditions of environment in the vicinity of listeners also affect the degree of reception. Most hi-fi systems are stereo (solid sound) systems**.**



Signal to Noise Ratio:

Almost every component, electronic, electrical, mechanical, generates noise. Since an audio system has electronic, electrical and mechanical components some degree of noise is unavoidable, the sources of electrical and electronic noise are : A chum, thermal noise in resistors, diodes, transistors etc, noise due to sparks, noise due to non-uniformity of magnetism, noise due to unwanted coupling between diff. ckts etc. The sources of mechanical noise are vibration, transport mechanism of recording and play back systems etc. All these sources of noise should be eliminated as far as possible.

Proper shielding, earthling filtering (electrical as well as mechanical) etc are necessary Hi-Fi systems should use stabilized power supply, directive microphones and noise reduction techniques like pre-emphasis and de-emphasis should be employed. Sometimes micro phones may pick up noise from environment. Therefore special noise free environment is necessary for recording.

$$S/N = \frac{\text{Signal power}}{\text{Noise power}}$$

$10\ \log_{10}(\frac{S}{N})$ dB

Non-linear Distortion:

Non-linear distortion is due to non-linear characteristics of transistors. In all hi-fi systems non-linear distortions is kept less than 1% by using –Ve feedback in amplifier Ckts. AC bias is very commonly used for magnetic recording. Push pull Ckts help in elimination of core saturation.

Frequency Response:

Frequency response is affected by inductances and capacitances in the Ckt. The series capacitance should be as low as possible and shunt reactance should be as high possible. A low series capacitance (Coupling capacitance) causes the response at low frequency to be poor. A high shunt capacitance causes high frequency response to be poor. Therefore a proper value for these parameters can help in achieving good frequency response. A rule of thumb is to keep the reactance of coupling capacitor, at 40 Hz less than 10% of load resistance and audio freq, more than ten times the load resistance.

The mechanical elements are contribute to the system inductance and capacitance. The mass of the moving system behaves as an inductance and affects the low frequency response. Therefore all mechanical parts, in hi-fi systems should have low mass. The compliance of moving system behaves as capacitance and affects the high frequency response.Moreover the nature frequencies of the system should be kept below 40Hz and above 15 KHz. This will ensure that transient response is good.

At hi-fi systems use bass and treble control ckts so that the listener can adjust these values in high requirement.

Intensity of Sound:

Hi-fi systems are designed to handle very high intensity sounds. The term dynamic range is used to specify this characteristic. Dynamic range is the dB ratio of the sound with highest and lowest intensities to which the system can respond. A hi-fi system should have a dynamic range of 60 dB or more. Many times the pop music has an intensity level of more than 100 dB over the threshold of hearing.

Good Environment:

A proper noise free environment is necessary if a hi-fi system is to give a pleasant hearing. The acoustic design of the room should also be good. In recording studios these aspects are especially kept in view. External noise can be kept low by using sound insulation materials on walls etc.

Equalization:

Equalization means changing the freq spectrum of the i/p. An equalizing ckt then is a ckt which alters the freq spectrum of i/p signal. The simplest equalizing ckt is tone control ckt.

Tone Control:

Every audio equipment has a tone control knob. However in relatively cheap equipments tone control is achieved by a losser type of control. Fig (a) shows a simple RC Ckt for losser type of tone control coupling the nth and (n+1)th stages.

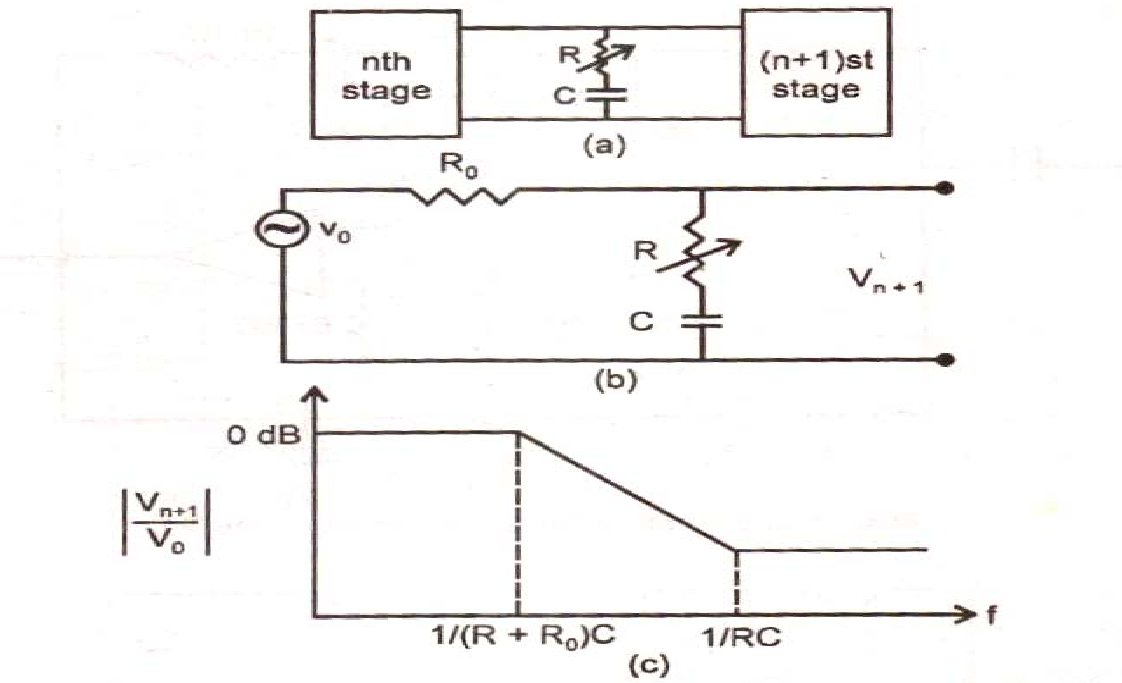


Fig. 11.2. Losser type tone control (*a*) Circuit (*b*) equivalent circuit (*c*) Bode Magnitude Plot

To analyze the response of this Ckt we need o/p impedance of nth stage and i/p impedance of $(n+1)^{th}$ stage Assuming the i/p impedance of $(n+1)^{th}$ stage as infinite (∞) and representing the nth stage as a voltage source Vo in series with source resistance Ro, we can draw the equivalent Ckt as shown in Fig.

The transfer function of this Ckt is

` $V_{n+1}(S)/V_0(S) = (R+1/sc)/(R_0+R+1/sc)$

STEREO SOUND SYSTEMS

The word stereo means solid. A stereo sound system means solid sound system. The word solid represents three dimensional spaces. Thus a stereo sound system is one which produces three dimensional effects. When a programme is being recorded (or played live) different sources of sound (say music) are located at different positions around the singer. When such a programme is played back it would appear original only if sound reaches to the listener's ears from more than one direction.

Monophonic system has one amplifier to amplify the signals from one or more microphones on the stage. Finally these loud speakers are alike and would give the same sound effect.

Stereophony (or stereo sound system) has at least two channels. Each channel has separate microphone, amplifier, loud speaker etc. These two channels known as left and right channel are recorded and played back separately and independently.

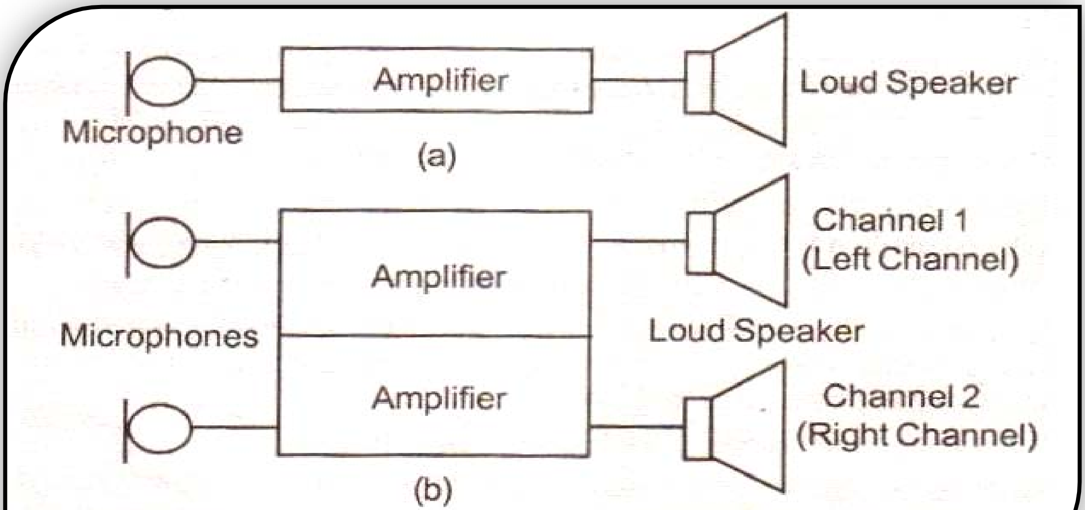


Fig 12.1 (*a*) Monophonic System (*b*) Stereo phonic Sysytem

The concept of stereophony evolved due to the reason that the sounds reaching our two ears have minute differences and our ears have the capacity to judge the direction of sound very minutely.

It is necessary that the left and right channel microphones and loud speakers are kept at a distance of about 3 meters. If left and right microphones and loud speakers are very near to each other, the stereophonic effect is not proper because the two channels do not remain isolated. If earphones are used in place of loudspeakers, the stereophonic effect is better appreciated. The left and right amplifiers are separated but may in the same cabinet.

Stereo Controls:-

A stereo system needs more controls than a monophonic system because a balance between left and right channels has to be obtained. The main controls are bass and treble control, quasi stereo switch, loudness control, blend control, master gain control and balance control.

(Bass means low frequency tones while treble means high freq tones, Bass gives depth in the sound while treble gives sharpness in the sound.)

Bass and treble controls are necessary on both the channels.

A quasi stereo switch is to switch on the same channel (either left or right) to both the speakers. It effectively converts a stereo system into a mono system.

Loudness Control:

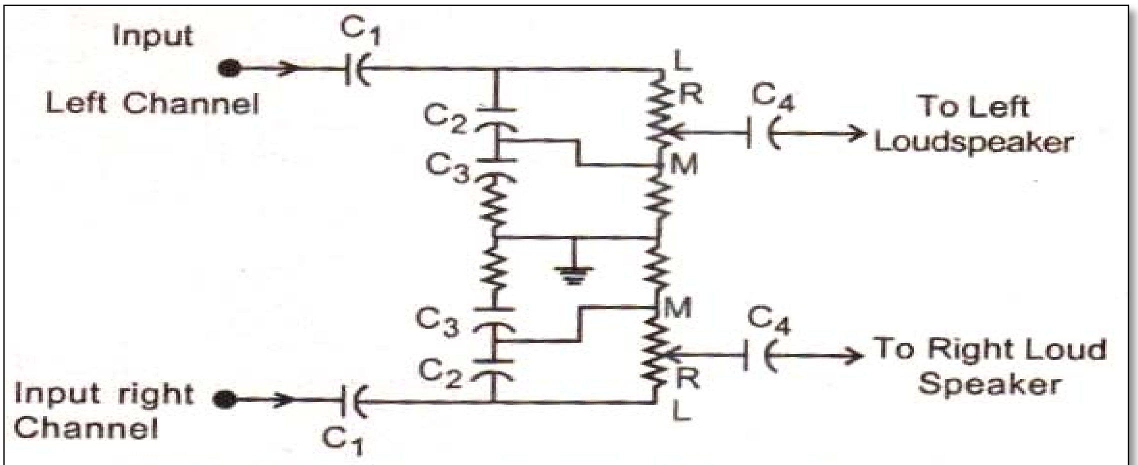


Fig 12.3. Loudness Control in Stereo Systems

C1 and C4 are coupling capacitors. The loudness control consists of potentiometer R and capacitance **C2 for each** channel. Loudness of sound is provided bass. Treble has almost no role to play in loudness to bass and little boost to treble. The general level of boost is +12dB for bass and +3 dB for treble.

For low singnal ilps, potentio meter R is set at L so that the signal does not suffer any attenuation. For high signal ilps, R is set towards M. This causes the R and **C2** combination effectively in series with the i/p signal causing attenuation of the signal. With proper values of resistances and capacitances the ckt gives +12dB boost at 50 Hz and +3 dB at 10 KHz.

<u>Blend Control</u>:

The function of blend control is to dilute any channel (left or right). Initially the blend control resistances of both channels are set at zero. This gives balanced o/p from the left and right channels. To dilute the effect of any channel, the resistance in that channel is increased.

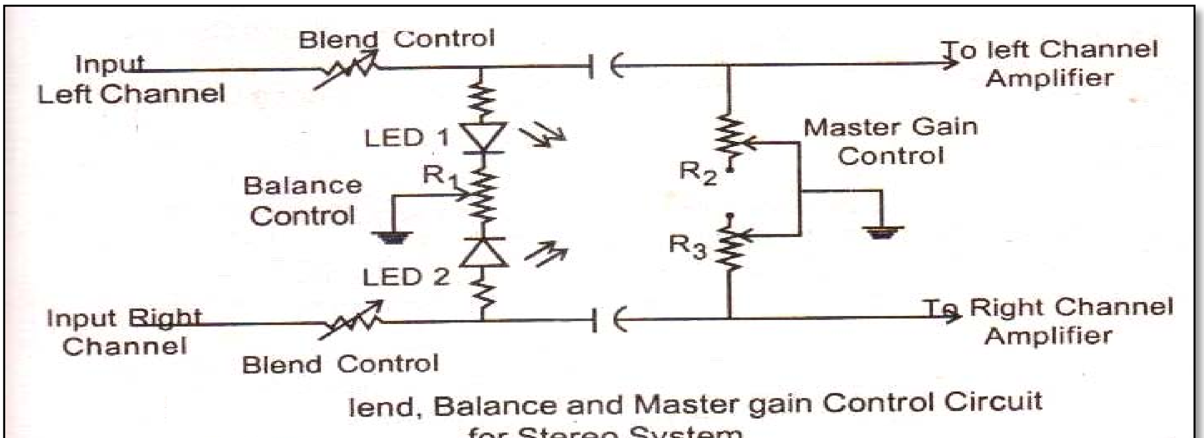
<u>Balance Control</u>:

The left and right channel amplifiers of a stereo system are always a matched paid having similar circuitry and component.

It is expected that gains of these two amplifiers should be equal so that for equal i/ps to the two channels, the o/ps are also equal.

However due to small variations in characteristics of components, this exact matching is not obtained in practice.

The function of balance control is to account for these small variations in the characteristics of component.



lend, Balance and Master gain Control Circuit
for Stereo System

With potentiometer R1 exactly at centre, the currents through LED 1 and LED 2 should be equal for left and right channel i/ps. For such a case the brightness of the two LEDs should be equal. If this is not so, there is some mismatching.

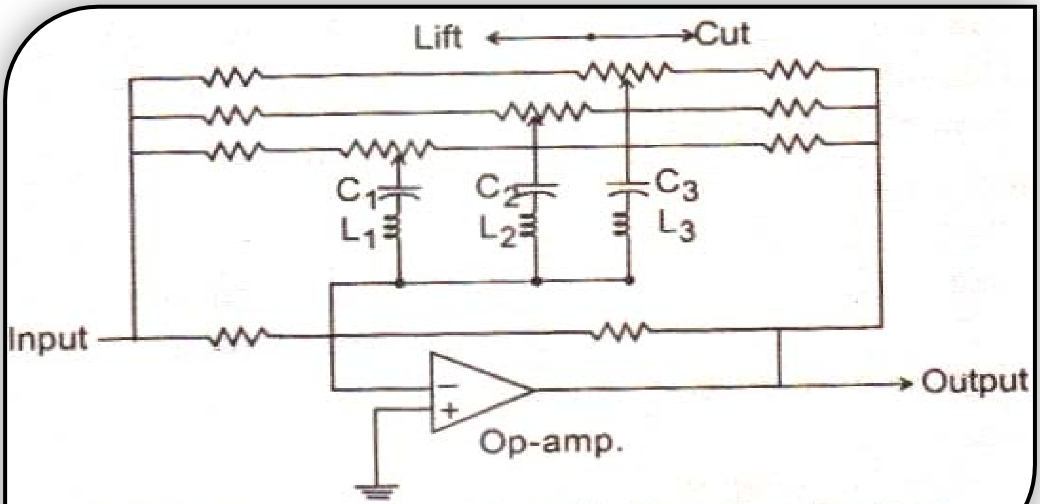
To achieve balance equal mono phonic signals are fed to both the channels and potentiometers R1 is adjusted so that the two LEDs show equal brightness.

Master Gain Control:

Master gain control adjusts the overall volume of sound without disturbing the balance between two channels. First the balance control is done by adjusting R1 as detailed above. The resistances R2 and R3 are ganged together. Thus they are increased or decreased together. By increasing or decreasing R2, R3, combination, the overall volume can be adjusted.

Graphic Equaliser System:

Most stereo system use graphic equalizer system to compensate for dips and peaks in the frequency response. This system consists of series LC tuned ckts in the feedback path. The system may have three or four band graphic equalizer. Most stereo system has three bands at 300 Hz, 1000Hz and 10 KHz respectively. The resonant frequency of the three L.C ckts. are equal to the above three frequencies. . Operation amplifier is frequently used in present day stereo systems.



Fi(ee Band Graphic Equaliser System

TELEVISION

HISTORY :

Isolation of selenium by the Swedish scientist, Berz Clius in 1817 and the discovery of Light sensitive properties of selenium by May in 1873 revealed the possibility of converting light from pictures into electrical signal.

In 1892, Elster and Geitel devised a photo electric cell based on this property.

Inventors could see the Selenium Cell as a counter part of the carbon microphone which altered its electrical resistance accordingly to the sound pressures impinging on it.

Like sound converted into electrical signals by the microphone, the selenium cell could possibly send picture signals over wire by transforming them into electrical counterparts.

Although the idea was simple, it was not very easy to implement. Transmission of picture required simultaneous sensing of the light intensity at various elements in the picture by thousands of selenium cells at the sending end transmitting the signal over an equal no of wires to an equal no of reproducing devices.

Paul Nipkow, a German experimenter, invented the scanning disc in 1884.

It was Nipkow's concept to change the picture into electrical bits at the sending end by means of spirally arranged holes in his rotating disc in front of the picture. The sequential transmission of these bits over a single wire and reconstruction by a similar scanning technique at the receiving end, serves as the basic principle of present day TV.

The first demonstration of actual television was made in 1925 – 27 by J.L. Baird in London and by C.F. Jenkins in Washington both working independently of each other by using the technique of mechanical scanning, employing the Nipkow rotating disc.

A Colour sequential system developed by Columbia Broadcasting Service (CBS) was adopted in USA in 1950, for colour TV Transmission. This system employed a rapidly rotating disc containing filters of the primary color red, green & blue in sequence, interposed disc between the lens and the camera tube at the taxation and also a similar disc between the receiving picture tube and the viewer, both discs rotating at the same synchronized speed.

The Radio Corporation of America (RCA) system compatible with the monochrome system was experimentally demonstrated early in 1949, but was accepted by the Federal Communications Committee – USA (FCC) for colour transmission in 1953, after being standardized as National TV Sub-committee –USA (NTSC) system.

Modified systems, Viz Sequential colour and memory (SECAM) in France and Phase Alternation Line (PAL) in Germany were introduced in later to overcome problems of phase errors in Transmission paths in the NTSC system.

<u>CCIR – B System :</u>CCIR stands for **Committee Consultative for International Radio.** It has not adopted any uniform TV standard.

As such we have CCIR-B standard (used in India, Germany etc), CCIR-L standard (used in France etc) and CCIR-M standard (used in USA).

Evidently TV receivers designed for one standard cannot be used for any other standard.

In this technique of interlaced scanning, since 312 ½ lines are scanned in 1/50 second, the band width requirement reduces to half i.e. to about 5 MHz and total BW of about 6.5 MHz .

<u>PERISTENCE OF VISION:</u>

Retina has the characteristic of retaining image for a short time. This is due to the reason that retina senses the brightness by a photo chemical process which has its own lag. The sensation in the eye, due to a single short flash, is a function of intensity and duration of the flash. This sensation continues for about 20 ms.

A flash of low intensity and long duration causes the same sensation as a flash of high intensity and short duration.

When flashes occurs intermittently the sensation of the eye continuous for 20 ms after each flash. If the flashes are repeated at a fast rate, these flashes appear continuous. This phenomenon is called persistence of vision, and is used in cinema and TV.

In both cinema & TV the repetition rate is fast enough to create an illusion of continuously. If the flash repetition is low, the pictures appear as separate flickers. The flash reputation rate above which the separate flickers appear as continuous is called critical flicker frequency.

Evidently the critical flicker frequency depends on the level of brightness and the colour spectrum of the flash. Human eye has max sensitivity for yellow-green light. Therefore flashes of these colours cause max persistence of vision.

In cinema the no of frames per second is 24 and each frame is illuminated twice during this internal. Due to persistence of vision the eye cannot distinguish the discontinuity.

In TV the repletion rate is kept the same as mains frequency (50Hz). In India and the countries who use 50Hz power frequency the repletion rate is 50.

In USA and some other countries the repletion rate is 60 Hz.

The 60 Hz rate is better and causes greater continuity of the picture.

Scanning :

For converting a scene or picture into electrical signal, the electron beam has to explore the picture point by point. This process is called scanning. Scanning is done at a fast rate thus creating an illusion of continuity due to persistence of vision.

Aspect Ratio:

All TV systems use rectangular frames. The ratio width / height of the frame is called aspect ratio and equals 4/3. The reasons for this value of aspect ratio are

a) The eyes can view with greater comfort and ease when width of the frame is more than the height.

b) The region of max. resolution at the centre of the retina has greater area along the width than along height. This region of max. resolution at the centre of retina is called fovea.

c) A large width of frame means a more efficient use of the area of the fovea (FOVEA).

d) Aspect ratio of 4/3 pleasing aesthetically and causes less fatigue of the eye.

e) The motion picture have a width / height ratio of 4/3. Therefore film programmes can be directly televised.

The High definition TV systems are designed to give better quality of viewing. For these systems aspect ratio is kept as 5/3.

Interlaced Scanning :-

From consideration of flicker, it has been found that 50 picture frames per second is the minimum requirement in TV scanning.

For a 625 line system the horizontal line scanning frequency should be 625 x 50 = 31250 Hz, with the line period of 32 μs . Period for blanking of the fly back retrace = $12\,\frac{\mu s}{2} = 6\ \mu s$

For a desired resolution of 546/2 =273 alternation in the horizontal line, this least to a very high band width requirement will be =10 MHz

To reduce the band width requirement, while still maintaining effective vertical picture scan rate of 50Hz is to employ interlaced scanning.

In interlaced scanning, the picture is divided into two fields each field containing 312 ½ interlaced line.

The first set of 312 ½ odd number lines in 625 lines called the first field or the odd field are first scanned sequentially and the remaining 312 ½ even number lines, called second field or even field are then traced interlaced between the lines of the first set.

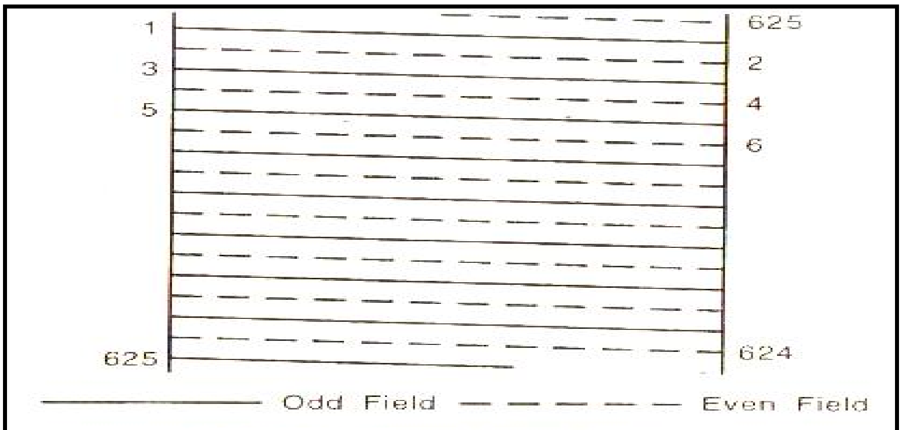
This is done by operating the vertical field scans at 50 Hz so that the two successive interlaced scans, each at 25 Hz rate make up the complete picture frame.

This keeps the line scanning speed down, as only 312 ½ lines are scanned in 1/50 second, thus keeping down the bandwidth requirement.

Thought the picture is scanned 25 times per second, the area of the screen is covered in an interlaced fashion at twice the rate Viz 50 times per second.

The fly back from the bottom to top is not instantaneous and takes a finite time equal to several time periods. Up to 20 lines are allowed for vertical fly back after each of the two fields that make a complete picture.

This means that out of 625 lines only (625-40=) 485 lines actually bear the picture information these are called the active lines.



F nterlaced Scanning

Video Bandwidth and Horizontal Resolution:

The horizontal resolution of TV system is the ability of the scanning system to resolve the horizontal details i.e. Changes in brightness levels of elements along a horizontal scanning line.

The horizontal resolution in a scanning system depends upon the rate at which the scanning spot is able to change brightness level as it passes through a horizontal line across the vertical lines of resolution as shown

In a 625 – line system, there are effectively about 410 lines of vertical resolution. The horizontal resolution should be of the same order.

Because of the aspect ratio 4:3, the number of vertical lines for equivalent horizontal resolution will be 410 X 4/3 N 546 black and white alternate lines, which mean (546 x ½ =273 cycles of black & white alternations of elementary areas.

Which mean (546 x ½) =273 cycles of black and white alternations of elementary areas.

For 625- line system, the horizontal scan or line frequency f_H is given by

f_H= No of lines per picture x picture scan rate =625 x 25 = 15625 Hz.

As each picture line is scanned 25 times in one second.

The total line period is thus.

TH $= \frac{1}{fH} = \frac{2}{15625}$ Sec = 64 μs

Out of this period, 12 μs are used for the blanking of the fly back retrace.

Thus the 546 black & white alternation i.e. 273 cycles of complete square waves are scanned along a horizontal raster line during the forwarding scan time of (64-12) =52 μs.

The period of thus square wave is 52/273 = 0.2 μs giving the highest fundamental frequency of 5MHz= [$\frac{1}{0.2 \; x \; 10^{-6}}$] which is adequate as the highest video frequency in the signal.

The highest fundamental video frequency in a scanning is thus given by

f max =$\frac{Active\ lines\ x\ Kell\ factor\ x\ Aspect\ ratio}{2\ x\ Line\ forward\ scan\ period}$

where f_{tH} is the horizontal – line forward scan period.

KELL FACTOR (RESOLUTION FACTOR) :-

The process of fly back of beam from bottom to the top is not instantaneous. Twenty lines are allowed for this fly back after each of the two fields. Thus the actual number of lines which contain picture information is (625 – 2 x 20) =585 lines. These 585 lines are called active lines.

The extent to which the picture details can be resolved in vertical direction is known as vertical resolution.

Due to the fact that the beam has a finite size and that the beam alignment does not coincide exactly with the elementary resolution line, the actual vertical resolution is less than the number of active lines available for scanning.

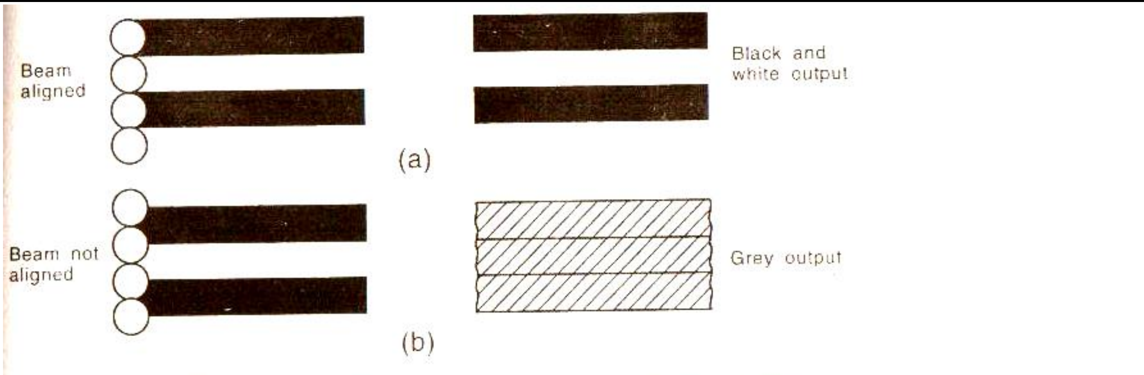


Fig. fect of non-alignment of beam on vertical resolution

Thus the finite beam size and misalignment cause degradation in vertical resolution.

Statistical analysis and subjective tests have indicated that about 30 percent lines get merged with other elements due to beam spot falling in two consecutive lines. This means only about 70% of the active lines are effective for vertical resolution.

This factor which indicates the reduction in vertical resolution is called Kell factor or Resolution factor and varies from 0.65 to 0.85. The Kell factor is generally assumed to be 0.7.

Taking Kell factor into account, the vertical resolution is (625 -40) x 0.7 = 485 x 0.7 =409.5 lines.

The horizontal resolution should not exceed this value multiplied by aspect ratio (i.e. 4/3).

Composite Video Signals:

Video signal is meant the electrical signal corresponding to the picture information at the o/p of TV camera which scans the picture. To this video signal, we add

(i) horizontal blanking pulses,

(ii) horizontal synchronizing pulses,

(iii) vertical blanking pulses,

(iv) vertical synchronizing pulses and

(v) Equalizing pulses.

The resulting video signal is referred to as the composite video signal.

The horizontal sync pulses are needed at the end of the horizontal scan when horizontal fly back is desired.

Similarly vertical sync pulses are needed at the end of the vertical scan.

The sync pubes occupy amplitude level corresponding to blacker than black level in the video signal.

Different TV standards are used in different countries. The CCIR-B standard is used in India, Pakistan, Australia and various other countries. In CCIR system –B uses 625 lines interlaced scanning with field frequency of 50Hz (picture freq. of 25 Hz). The Horizontal Lines freq. is 625 x 25 =15,625 Hz. The aspect ratio is 4:3.

Blanking pulses: - The picture tube is made incorporative during the horizontal & vert. retrace intervals by means of blanking pubes. These blanking pubes are thus included in the composite video signal and they modulate the transmitter carrier. At the receiving end, after the detector stage, these blanking pubes are separated out from the composite video signal and used for blanking the picture tube.

The horizontal blanking pulse at the black level (75% black) extending for 0.19 H Where H is horizontal trace period.

Vertical blanking pulse of duration 0.065 V (about 20 H) in each field at the end of each field.

Horizontal Sync. Pulse & Blanking pulse standard.

f_H =15625

H=1/f_H = 64 μs

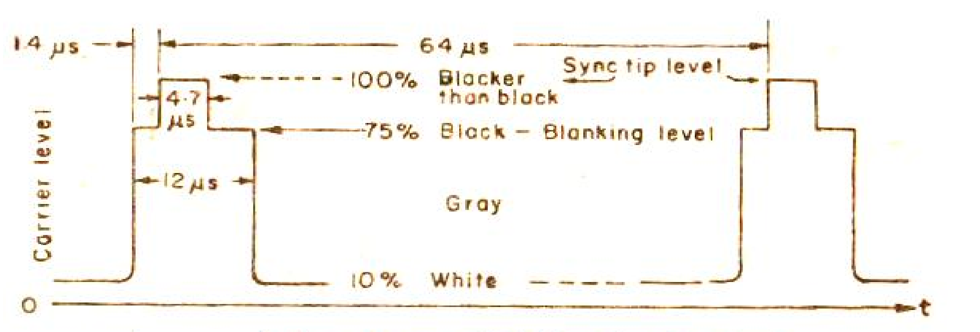
Line Blanking period (LB). It is kept 0.19 H = 12 μs

Line Sync Pulse:- It is sent from the transmitter to maintain the horizontal scanning rate at the TV receiver in synchronism with that at transmitter.

As per CCIR-B system standard, its length is HS =0.075H= 4.7 μs

Front Porch : The horz. Sync. Pulse starts about 0.025H (=1.5) later than the blanking pube. This period of 0.025 H, called the front porch. It permits every sync. Pulse to build up in the positive direction starting from fixed level of the blanking pulse.

Back Porch: The sync. Pulse ends about 0.09H before the end of the blanking pulse. This period, called the back porch it permits the line retrace to complete itself and all oscillations in the deflection circuit current to die down before the next forward deflection begin.



Horizontal sync and blanking signal standards

Vertical Sync. & Blanking pulse Standard: A vertical sync. Wave form is inserted in the composite video signal at the end of each field of 312 ½ lines.

CCIR-B (Standardized by) : International Radio Consultative Committee.

Monochrome (Black & White) Television Receiver:-

It consists of RF tuner, IF circuits, video detector, video amplifier, AGC, synchronization and deflection circuits, audio section, low voltage power supply, high voltage power supply, monochrome picture tube etc.

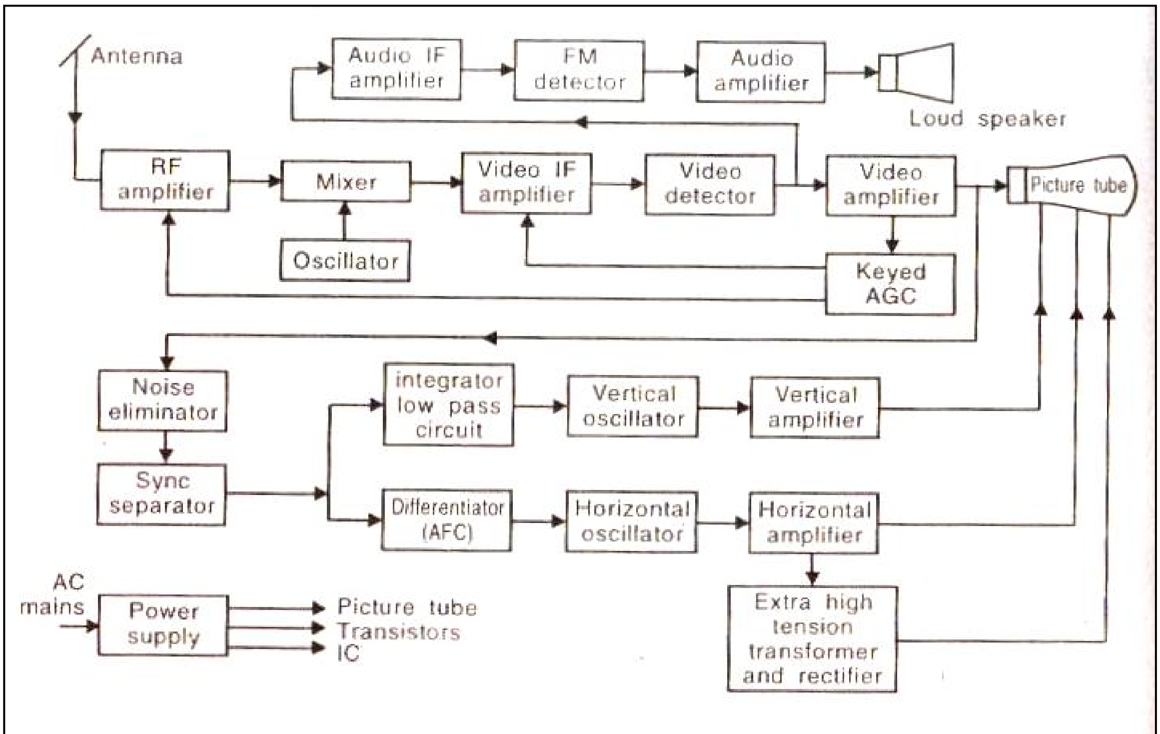
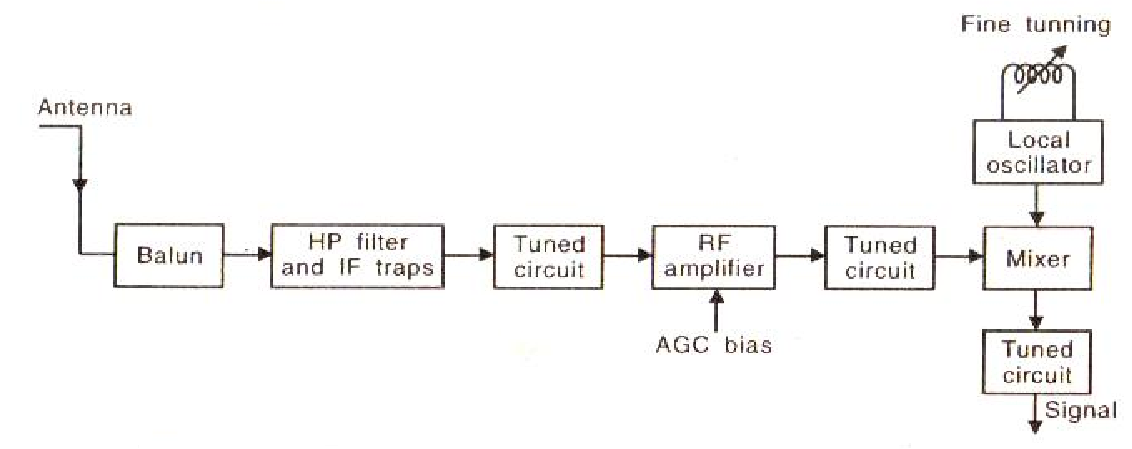


Fig. 13.34. Block diagram of B & W television

RF Tuner:

The RF tuner is known as front end of TV receiver because the signal from antenna (or cable in case of cable TV) is first of all received in RF tuner. The Functions of RF tuner are: It receives the TV signal and selects the required channel.

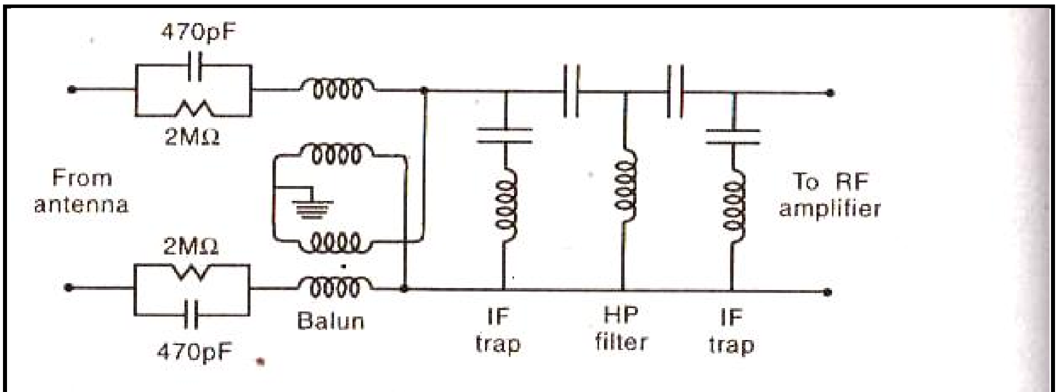
1. It provides impedance matching to maximize the power delivered to TV set. Impedance matching also improves the signal to noise ratio.
2. It converts RF signals into IF signal by mixing it with local oscillator frequency and feeds the IF signal to video IF amplifier.
3. It prevents interfering signals from entering TV set.
4. It provides isolation between signals as received from antenna and local oscillator signal.
5. It rejects the image frequencies through the use of RF selective circuits.



Block diagram of RF Tuner of TV Receiver

The functions of different components of RF tuner are as under:

a) **Balun:** It is a radio frequency transformer for matching the impedance of antenna feeder to that of RF tuner. It gives a 4:1 impedance change thus giving impedance matching between feeder (characteristic impedance 300 ohm) and RF input impedance (75 ohm). It has a ferrite core with four tightly coupled bifilar windings in the form of two quarter wave lines of 150 ohm each. At one end these two windings are in series giving an impedance of 300 ohms and at other end they are in parallel (with one terminal grounded) thus giving 75 ohm impedance. Two small capacitors about 470 pF each are connected, one in each series lead, to block the dc path from chassis to antenna and also prevent damage due to lightning. The capacitors have 2 MΩ resistors in parallel to discharge any static charge accumulated on capacitors.



.36 lun, IF traps and HP filter section of RF tuner in TV receiver

b) **HP Filter and IF Trap:** This function of this section is to let the frequencies beyond 40 MHz pass and block the frequencies in the range of 33 to 40 MHs. Evidently this section has a high pass filter having pass band about 40 MHz A high pass T section filter has capacitor in the series arms and inductance in the shunt arm. The IF trap consists of two trap circuits (L & C in series) on either side of high pass section as shown in in Figure. It is necessary to have IF traps because it is difficult to remove them it they reach the mixer and IF stages. The IF traps at this stage help in rejecting the IF in the range of 33-40 MHz

c) RF Amplifier: The signals from HP filter and IF traps enter the RF amplifier, whose function is to amplify weak signals and improve, signal to noise ratio. The equivalent noise voltage at the input of the RF amplifier sets limit to the minimum signal strengths at the input to RF amplifier. The noise voltage appears as a showy background of black and white spots moving randomly on the TV screen. The RF amplifier also acts as a buffer between local oscillator and antenna terminals so as to minimize radiations from the local oscillator.

RF amplifier should have a bandwidth to pass the selection TV channel. Double tuned filter circuit is used to provide a flat top response and couple the RF amplifier to the mixer. A dip of 1 dB between the peaks is allowed as shown in figure.

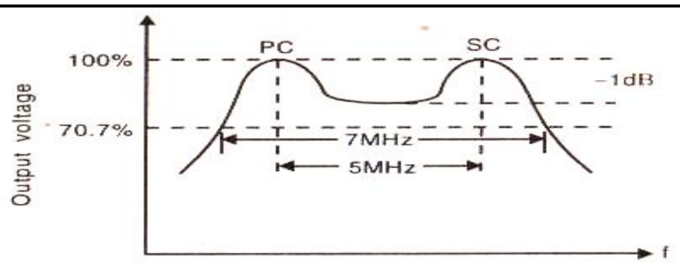


Fig sponse of RF Amplifier

d) Mixer : The mixer changes the RF signals from different channels into a common IF signal by heterodyning the RF signal with the frequency of local oscillator. Thus the mixer and local oscillator can be considered as a frequency changer. The RF signal has two carriers viz, picture carrier and sound carrier. The output of mixer consists of two Ifs viz. picture IF equal to 38.9 MHz and sound IF equal to 33.4 MHz. The local oscillator frequency for channel 3 is 55.25+38.9=94.15 MHz and for channel 6, the local oscillator frequency is 182.25+38.9=221.15 MHz.

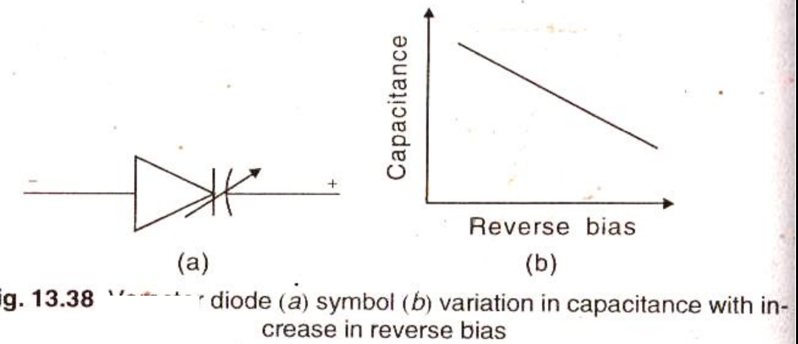
e) Local Oscillator. : It provides the local frequency so that it can be heterodyned with the signal obtained from RF amplifier in the mixer section. Whenever a different channel is selected for viewing, the frequency of local oscillator is changed so that the intermediate frequency is the same. The local oscillator is provided with fine tuning so that its frequency can be adjusted accurately. The frequency of the local oscillator must be stable and free from drift. Use of high quality components, compensation techniques, stabilized power supply and special capacitors (having zero temperature coefficients) help in keeping the local oscillator frequency stable.

f) Electronic Tuning and Channel Selection: TV sets manufactured many years ago used mechanical tuner. Such tuners had the disadvantages of difficulty in tuning, wear, high cost and inability to tune precisely. Present days TV sets all use electronic tuners which do not have any moving part. An electronic tuner uses varactor diode as a variable reactive element.

When a reverse bias is applied to p-n junction the thickness of depletion layer increases. An increase of reverse bias increases the thickness of depletion layer. This increase of thickness with increase in reverse bias can be considered as a capacitive effect and the incremental capacitance Ct defined as

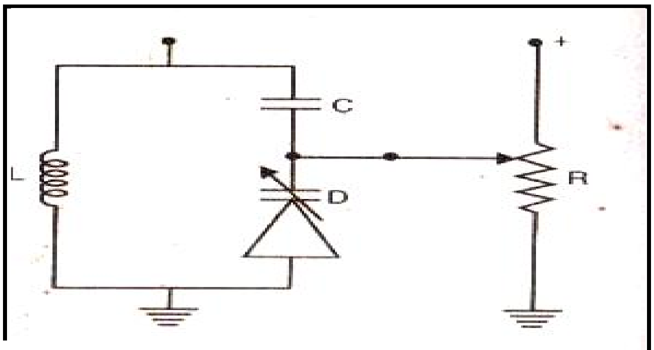
$$Cr = \frac{dQ}{dVx}$$

Where dQ is the increase in charge due to voltage increase dV.



ig. 13.38 [illegible] diode (*a*) symbol (*b*) variation in capacitance with increase in reverse bias

This variable capacitance is used for tuning in electronic tuners. **Fig-** (a) Shows the symbol of varactor diode and **Fig-** (b) shows the variation of capacitance with increase in reverse bias.(Capacitance is inversely proportional to thickness. As reverse bias is increased, thickness of depletion layer increases and capacitance decreases).



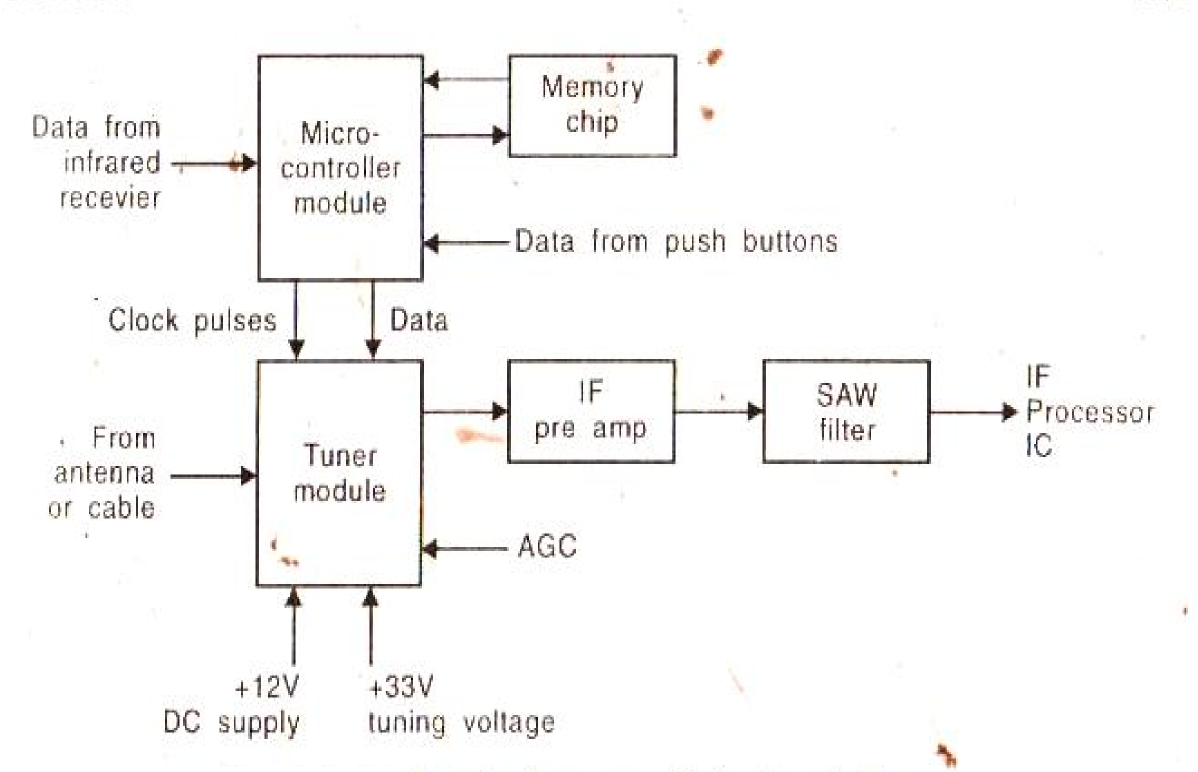
. Tuning Circuit using Varactor diode

Figure Shows a varacter used in a tuned circuit. When the tap position on the potential divider R is changed, the reverse bias applied to varactor diode D changes and therefore, its capacitance changes. The result is a new value of total capacitance and hence a new resonant frequency.

A varacter tuner can be either continuous type (commonly used in small portable TV receiver) or push button press in which a channel can be selected through the use of push button.

Advanced varactor tuners use a voltage synthesizer. Synthesizer converts each channel voltage into a binary code. This code is stored in a 1C memory. The selection of channel is done by the unit reading this code. Such tuners can be used with infrared remote control systems.

Figure Shows the block diagram of an advanced automatic varactor tuner. The system micro-controller (Micro controller is a microprocessor used for control purposes) receives data from infra red receiver. In addition to this data, micro-controller receives information from channel selector push buttons. Such a tuner can tune all **VHF, UHF** and cable channels**.**



Block diagram of electronic tuner

The microcontroller and its associated circuit provide the control signal to the tuner module. The signal from selected channel is amplified, heterodyned with the frequency of local oscillator to produce the output in IF range.

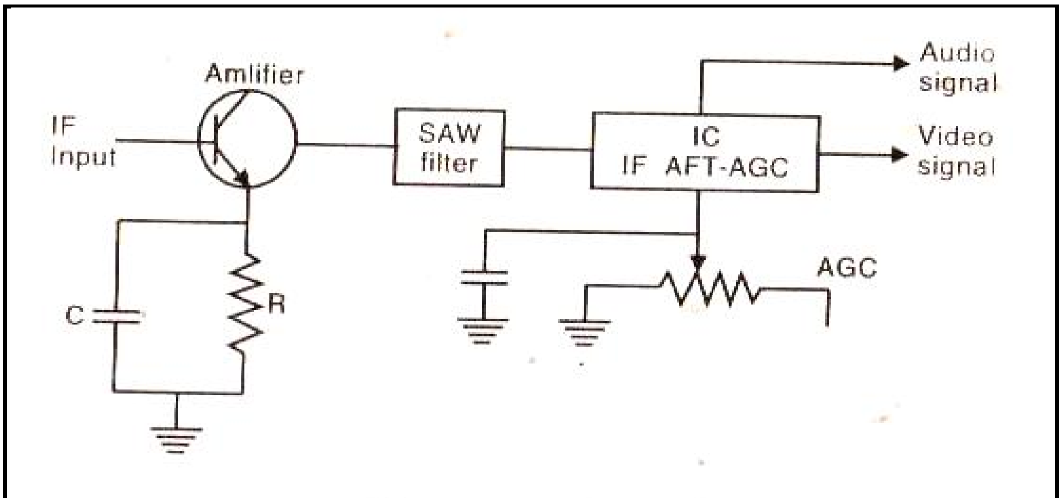
Varactor tuner has a higher noise level than the one using passive LC components. Therefore, a varactor tuner definitely needs RF amplifier to improve the signal to noise ratio.

The circuit of Fig-7 uses a SAW (surface acoustic wave) filter which is a special type of filter. This filter does not require LC resonant circuit. AGC is automatic gain control to maintain constant output even though the RF signal may vary in strength. AGC operates by reducing the gain of IF and RF amplifiers if the signal strength increases.

IF CIRCUITS:

The intermediates frequency in CCIR-B system used in India and many other countries is 38.9 MHz for video signal and 33.4 MHz for audio signal. This value of IF is a compromise between many conflicting requirements. It is easier to have high and stable gain from IF amplifier if IF is low. However; IF should be high enough to provide desired bandwidth. Moreover a high IF is desirable from the considerations of spurious response and ease of filtering at video detector. Thus the choice of 38.9 MHz and 33.4 MHz as IF for video and audio respectively is a compromise between the above opposing requirements.

The IF circuit may consist of three or four transistor stages or one 1C. In the present day TV sets the IF input amplifier is located between the RF tuner, SAW filter and IF AFT-AGC IC as shown in **Fig.8.**



Fi circuit

The RF and IF AGC Circuit are developed inside the IC. Todays IF circuits contain SAW filter (surface acoustic wave filter) with two pairs of transducer electrodes. One is the input and the other is the output transducer.

A SAW filter comprises piezo-electric substrate on which two inter digital transducers are deposited. The piezo-electric property generates surface acoustic wave when an electric signal is applied to input transducer. This wave travels along the surface of substrate. The output transducer converts the surface wave back to electric signal.

If the finger structure of output transducer is the same as that of input transducer, a larger voltage is generated across output transducer. The amplitude response can be changed by varying the substrate material, finger spacing, finger width and omission of some fingers. This filter provides frequency selective filtering action and has replaced LC filters in all modern receivers. It is a very suitable solid state device.

Video Detector:

The two basic functions of video detector are (1) to detect amplitude modulated carrier IF to produce video signal (and filter out carrier wave) (2) to separate inter carrier sound IF and feed into sound section.

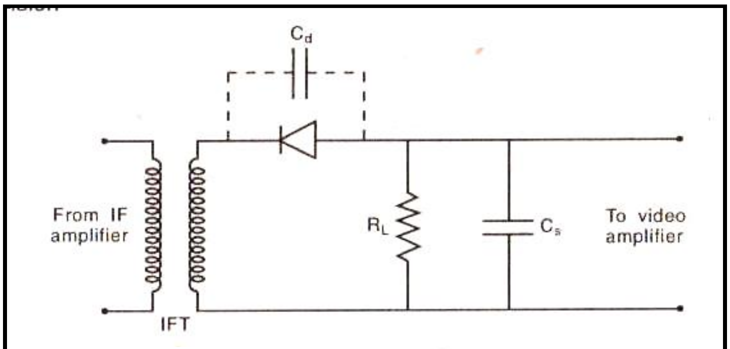


Fig. 13.42. Video Detector

A video detector is basically a diode. Thus it demodulates the video IF by the rectification process. The IF component in the rectified output signal is eliminated by low pass filter. **Fig-9** illustrates the principle of video detector. The diode blocks the positive portion of the signal but allows the negative portion. The IFT (intermediate frequency transformer) couples the IF amplifier to the video detector. To have high detection efficiency and to avoid distortion due to clipping, the load resistance RL should be much higher than rf , i.e. forward resistance of diode and filter capacitor Cs should be much higher than diode capacitance Cd.

The polarity of video detector can be positive or negative depending on the connection (Fig-9 the polarity is negative). The actual polarity depends on the number of stages due to phase reversal at each stage. A wrong polarity would produce reversal of blacks and whites in the picture thus showing a negative picture (similar to negative of a photograph).

Modern television (especially the colour TV) receivers use a synchronous type detector contained in an LSI (Large Scale Integration) circuit. The synchronous type detector gives better signal to noise ratio and lesser distortion. IC 1001 is one such integrated circuit containing IF/colour detection, horizontal and vertical circuits.

Video Amplifier:

A video amplifier is needed because the output of video detector is only a few volts and it is necessary to strengthen the video signal before feeding it to the picture tube. Video amplifier increases the strength of video signal to about 50 V. The adjustment of gain of this amplifier provides contrast control.

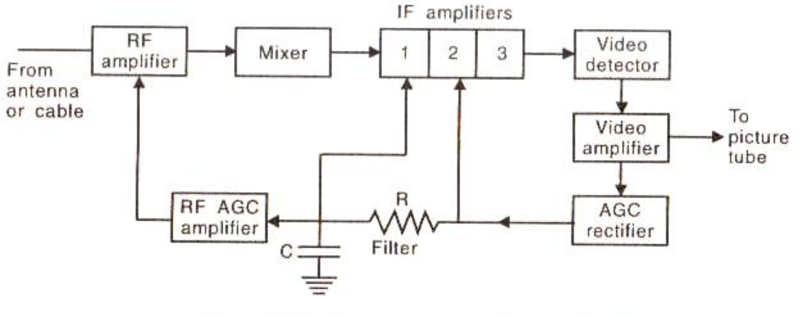
The older TV sets have emitter coupled transistor amplifier as a video amplifier. It is generally direct coupled to the video detector stage. Negative feedback is used to stabilize the gain. Modern TV sets use an IC as video pre amplifier. IC TBA 890 is one such IC. It is a two stage differential amplifier and may be preceded by a driver stage to provide impedance matching. It provides about 70 dB gain. The output from this IC drives video output transistor through a contrast control circuit. This IC also feeds sync. Separator, AFC circuit and AGC circuit.

AGC (Automatic Gain Control):

The output of video amplifier tends to change because the strength of signal from the receiver antenna changes from time to time. AGC has the following functions in a TV receiver.

1. To provide more amplifier gain for weak signals.
2. To decrease the gain for strong signals so that distortion is reduced and the reception remains nearly constant.
3. To keep the contrast the same when we switch from one channel to another.
4. To reduce flutter, in the TV reception, due to passing aeroplanes etc.
5. To maintain constant sound level.

Thus AGC provides a fixed level of video signal at the output of video detector despite the widely varying levels of RF signal supplied by antenna or cable. The signal from antenna/cable may vary from less than 100 mV to about 5000 mV. The signal strength depends on transmitter power, distance of receiver set from transmitter, cable losses etc. With AGC, the picture contrast remains nearly the same.



3.43. Feedback loop of AGC circuit of TV Receiver

Figure shows a block diagram illustrating the working of AGC. The rectifier produces a dc control voltage proportional to the peak value of the received signal. To eliminate ac signal variation and smoothen the dc control voltage RC filter is used. This dc bias is connected to the IF amplifier and RF amplifier. A stronger signal increases the AGC bias and reduces the gain thus maintaining the picture contrast constant.

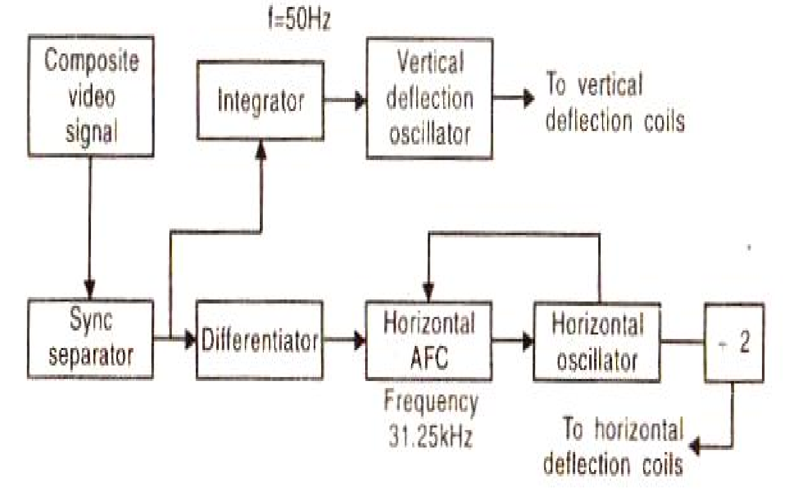
Modern TV sets use keyed AGC. In this system the AGC rectifier conducts only during horizontal sync. Pulse periods, using the fly back pulses derived from the output of horizontal deflection circuit of the receiver. Thus the AGC rectifier conducts only for a short duration when keying pulse is applied. In keyed AGC, the dc control bias is a true representation of the incoming signal and noise effects are eliminated.

Synchronizing and Deflection Circuits:

The video signal contains horizontal and vertical deflection, and equalizing pulses. These pulses are used in a television receiver to ensure that scanning at the receiver is in exact synchronism with the scanning at the transmitter. The details of these pulses are:

1. The H Sync. Pulses are narrow having 4.7 μs pulse pulse width. They are repeated at 15625 Hz line scanning frequency.
2. The V sync. pulses are much wider. These pulses are repeated at 50 Hz field scanning rate.
3. The equalizing pulses at 31250 Hz are repeated at half the line intervals. A group of five equalizing pulses occur just before and after each V sync pulse, to made the vertical synchronization the same in even and odd fields for good interlacing.

The TV receiver has two separate scanning circuits. One of these circuits deflects the electron beam in horizontal direction and the other in vertical direction. Each scanning circuit has an oscillator and a power output stage. The sync pulses as obtained from the composite video signal are used to control the vertical and horizontal oscillators so that the picture tube is scanned in synchronism with the scanning of the picture at the transmitter.



3.44. Block Diagram of Sync. Separator and Deflection Circuits

Figure shows the block diagram of sync separator and deflection circuits. The composite video has sync. in the negative direction. In this case the sync separator is a transistor (PNP) and the negative polarity can drive the base of common emitter PNP transistor into conduction. Only the sync pulses cause flow of current in the transistor. The output of transistor is total sync i.e., H, V and equalizing pulses. The sync separator operates as inverting amplifier and therefore, the output sync pulses are positive.

The difference in the pulse duration of H and V pulses is utilized to separate these two pulse from the output of sync separator. The H pulses have a width of 4.7 μs and frequency of 15625 Hz. Thus these pulses are high frequency pulses. The V pulses have width of 160 μs and a frequency of 50 Hz and are thus low frequency pulses. Thus it is possible to separate H and V pulses using RC circuits. A low pass filter allows 50 Hz V pulses to through it and thus its output is V pulses. Th low pass filter is also called integrator and has large time constant. A high pass filter allows 15.625 kHz H pulses to pass through it. It has a very small time constant and is also known as differentiator.

The output of differentiator i.e. the H pulses drives the horizontal AFC (Automatic Frequency Control) which controls the frequency of a 31.25 kHZ crystal oscillator. This frequency is divided by 2 and the output fed to H deflection coils. The output of integrator controls the frequency of V deflection oscillator which is 50 Hz. The output of this oscillator feeds the V deflection coils.

Modern TV receivers use IC for performing many complex amplification and sync separator functions. This IC performs the following functions:

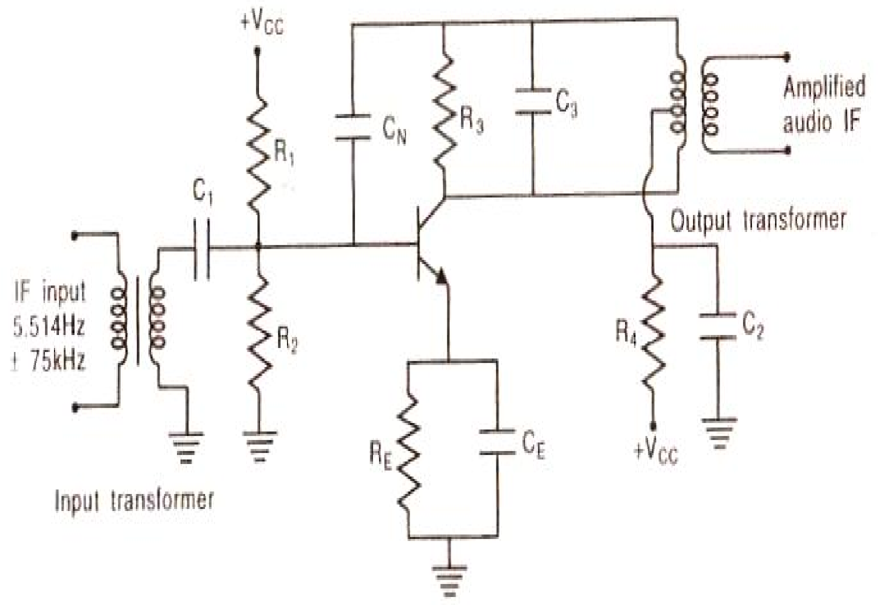
1. Video pre-amplifier
2. Gated AGC detector to supply AGC voltages to VIF (Video Intermediate Frequency) and tuner circuits.
3. Noise cancellation in AGC and sync separation.
4. Sync separator
5. V-0pulse separator
6. H pulse separator.

Audio Section:

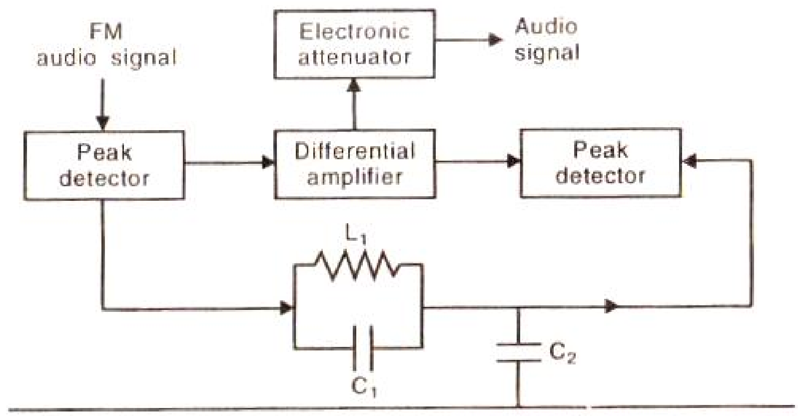
Modern TV receiver use inter carrier sound system invented by RB Dome in 1947. In this system the combined video and audio signals are first amplified in video IF amplifier. The output from IF stage is picture IF (38.9 MHz) amplitude modulated with composite video signal and sound IF (33.4 MHz) frequency modulated with audio signal. The two intermediate frequencies along with their side bands have a bandwidth of 6.75 MHz. This is fed to video detector. The sound take off points is after the video detector. The amplitude of inter carrier audio signal at the output of video detector is quite low and is amplified by two stage pre-amplifier and then fed to FM detector. Each IF audio pre-amplifier is a tuned amplifier with a centre frequency of 5.5 MHz and a bandwidth of about 150 kHz to provide full gain to FM side bands. Thransformer coupling is used at the input and output of pre-amplifier to provide impedance matching.

Figure shows the circuit of CE transistor pre-amplifier for sound IF. C1 is coupling capacitor. R1 and R2 are for biasing. RE and CE form the bypass circuit CN is neutralizing capacitor for providing negative feedback. R4 and C2 form the decoupling circuit.

A number of methods for FM demodulation are available. The most commonly used is differential peak detection. In this method two peak detectors use differential amplifier configuration with emitter follower at the inputs. The emitter followers provide high input impedance. The output circuits of the two detectors have identical RC networks of suitable time constants to provide peak detection of their input signals. The output signal appears as the differences between the two peak detected signals. This output is fed to electronic attenuator which does the job of volume control.



udio IF Amplifier



F ifferential Peak FM Detector

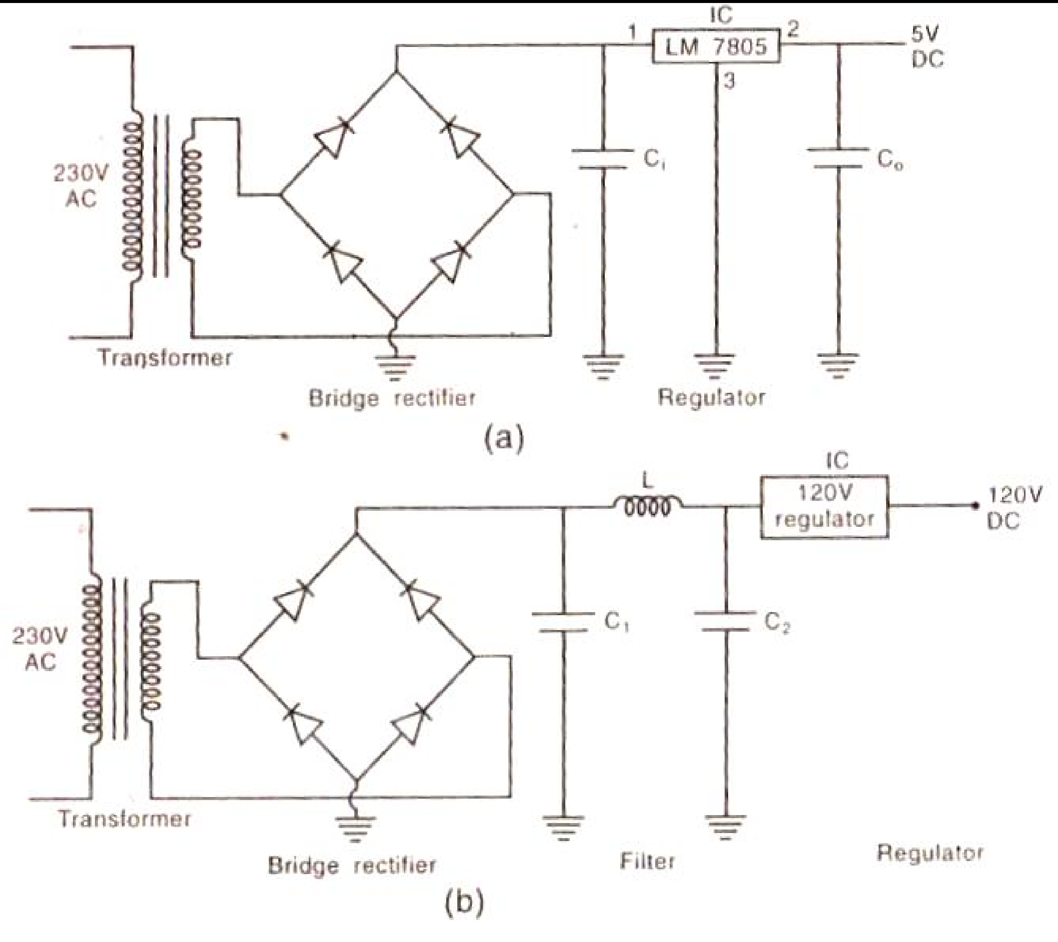
Figure shows a block diagram of differential peak FM detector.

The circuit formed by L1, C1 and C2 is a frequency selective network.

Modern TV Sets use IC to perform the above functions in the audio section, IC BELCA 3065 is one such IC used in black and white TV Sets. This IC consists of a regulated power supply, IF amplifier, limiter, FM detector, electronic attenuator and buffer amplifier and audio driver stage.

Low Voltage Power Supply:

A TV receiver requires low voltage power supplies around 5V, 12V, 120V etc for various ICs and other functions. These supplies are always obtained from mains 230V supply through a transformer, (for changing 230V to proper ac voltage), bridge rectifier (for converting ac to dc), filter (to remove ripples from rectifier output) and IC regulator to keep the dc output voltage constant. **Fig.47** shows the configurations generally used.



F ow voltage power supplies for TV receiver

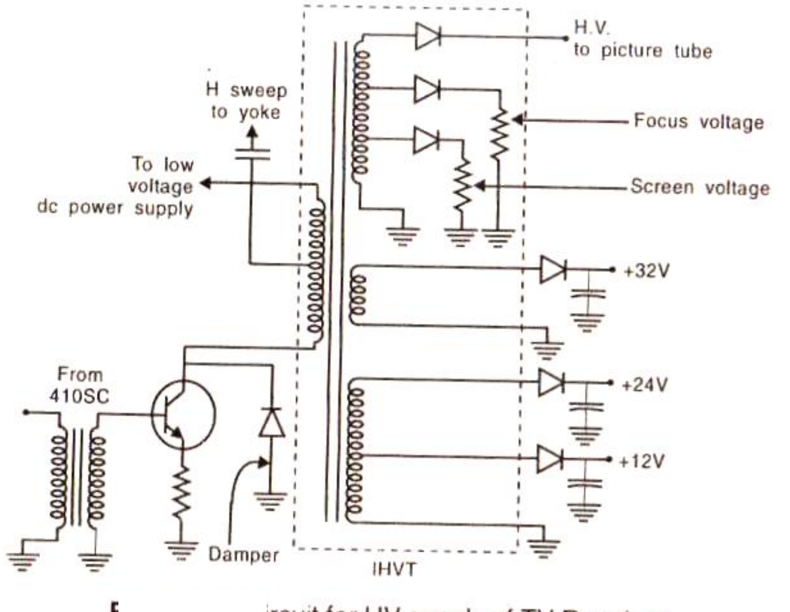
Figure (a) shows the circuit for a 5 V dc regulated supply. The two winding transformer steps down the 230 V ac voltages by about 60 (the turn ratio N1/N2 is about 60). The bridge rectifier converts the transformer secondary ac voltage to dc. LM 7805 is an IC regulator which gives a constant output voltage of 5V. It is basically a series regulator using zener diode and op amp with negative feedback from output. The capacitor C1(typical value $0.22\mu F$)prevents oscillations due to the inductive reactance of leads. The capacitor C0 improves frequency response.

Figure (b) shows the circuit for 120 V dc supply. The two winding step down transformer has turn ratio of 5:3. The bridge rectifier converts ac to dc. The bridge rectifier converts ac to dc. The filter removes ripples from the output of rectifier. The IC regulator maintains constant output voltage at its terminals.

Sometimes switched mode power supplies are also used. These power supplies operate at a high frequency and thus the transformers used in SMPS are low cost, light weight and efficient.

High Voltage Power Supply:

Picture tube of a TV receiver requires high voltage power supply. High voltage is necessary to light up the picture tube. Roughly 1 kV of high voltage is needed or each diagonal inch of CRT screen. Thus a 21" TV receiver requires about 21 kV dc supply for the picture tube.



F ircuit for HV supply of TV Receiver

Figure shows a circuit for H V supply. When the TV receiver is switched on, the regulated low voltage dc supply is applied to horizontal oscillator, horizontal driver and horizontal output transistor amplifier. The horizontal output transistor and 1 HVTC (integrated high voltage transformer) is shown in Figure. This transistor is a class C amplifier. When this transistor is on, power is supplied to horizontal deflection coil in yoke of picture tube and electron beam moves from left to right across the screen. During this time an increasing magnetic field is being developed around the primary winding and concentrated by the ferrite core of the transformer (since frequency is over

15 kHz, ferrite core is used). When horizontal output transistor turns off, the magnetic field collapses and emfs are induced in secondary winding of transformer. This emf is produced during horizontal retrace time and is known as fly back emf. The transformer is called fly back transformer. HV rectifier, focus control, screen control are all in the sealed high voltage transformer. Therefore, the transformer is called integrated high voltage transformer. Since this power supply is horizontal sweep driven, it is known as Scan Derived power supply.

Monochrome (Black and White) Picture Tube:

The picture tube has to convert the video signal (received from video amplifier) into a picture of the scene being telecast. Its construction is in much respect, similar to that of a cathode ray tube. The principle of operation of the two is also very similar.

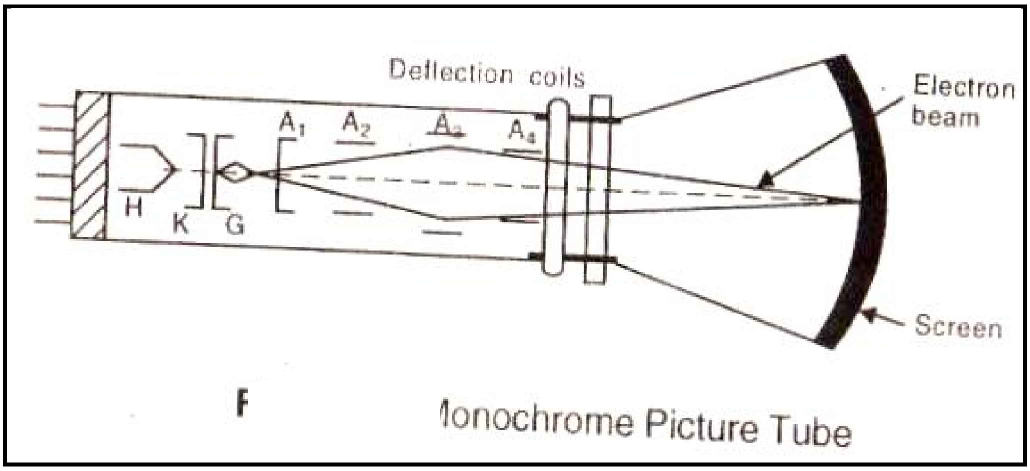


Figure shows the construction of a monochrome (black and white) picture tube. It consists of an electron gun and a number of other electrodes. The electron gun consists of an indirectly heated cathode (K), control grid (G) and focusing anodes A1,A2,A3,A4. Anode A1 is also called screen.

The video signal is applied to the cathode. The control grid G is at a variable DC negative potential of about -50V with respect to cathode. The potential of grid G acts as brightness control. The electrons coming out of grid G are accelerated by anode A1 is such that the electrons converge towards the axis at a point in between G and A1 as shown. Anodes A2 and A4 are at EHT potential of +16kV while anode A3 has a potential of about 400 V. The potential of A2 is kept very high so that even a small beam current can give the required light intensity.

A fine aluminum coating inside the fluorescent screen reflects light to illuminate the screen. Two pairs of deflection coils called yoke coils are mounted externally around the neck of the picture tube to cause horizontal and vertical deflection of the beam to produce the pattern of 625 illuminated horizontal lines called raster. Even when the video signal is not applied to the tube, raster still exists. When a video signal is applied, the raster gets converted to the picture.

Colour TV Signal

Colour Fundamentals: Light rays are electromagnetic rays whose properties are governed by their wavelength (or frequency). The spectrum of wavelengths visible to human eye range from 7000 A for red light to 4000 A for violet light. White light is a combination of all the seven colours in this range. The three colours red blue and green are known as primary colours because all the other colours can be obtained by proper mixing of these three colours. However none of these three primary colours can be obtained from the other two.

Colour Specifications: A colour is completely specified by the following three terms:

1) **Hue : It** is the actual colour seen by eye. Red, green, blue, yellow etc. represent different hues in the spectrum of colours. Hue is the result of the effect produced on the eye by wavelengths of that colour. By mixing two or more primary colours many hues can be produced.

2) **Luminance or Brightness:** It is the total amount of light intensity or light energy received by eye. It is expressed in lumens. It is known that certain colours appear brighter than other. This is because of brightness.

3) **Saturation:** It indicates the purity of the colour. In other words it represents the amount of other colours present in it. A pure green light is a saturated colour but becomes desaturated when white is mixed with it. It is seen that brightness is the properly of white light and colour light whereas saturation is an attribute of colour light.

Different colours have different wavelengths. Thus a colour has a frequency and amplitude. The frequency corresponds to hue and amplitude corresponds to brightness. Saturation is akin to signal to noise ratio.

Colour Signal :

The three primary colours blue, green and red produced by the three camera tubes (or a combined tricolor camera tube having three electron guns) are used to produce the colour signal. However, if these colours are transmitted directly, there will not be proper compatibility with monochrome television receivers. Therefore the three colour signals are combined to form luminance signal and chrominance signal.

1) <u>Luminance Signal</u>:

This signal is obtained by mixing the three colours red, green, blue in proportion of 30%, 59% and 11%. This mixed signal is called Luminance signal denoted as Y signal (Y does not stand for yellow). Thus,

Y=0.3R + 0.59G + 0.11B

Where Y is luminance, R is red, G is green and B is blue.

The above proportions have been selected keeping in view the sensitivity of eye to these colours.

Y has the maximum value of 1 V for peak white. Thus white includes all the three primary colours red, green and blue. For other colours the luminance signal Y is sum of luminance contribution of the primary components as follows:

Colour	**White**	**Yellow**	**Cyan**	**Green**	**Magenta**	**Red**	**Blue**	**Black**
	R+G+B	**R+B**	**G+B**	**G**	**R+B**	**R**	**B**	
Y	**1.0**	**0.89**	**0.7**	**0.59**	**0.41**	**0.3**	**0.11**	**0**

2) <u>Chrominance Signal</u>:

Chrominance signal is also called C-signal. It indicates the hue and saturation of colour. Y signal has to be transmitted for compatibility with the black and white receiver. Instead of transmitting all the three R, G, B signals (in addition to Y signal) two difference signals R-Y and B-Y are sent. It is not necessary to send G-Y signal as it can be obtained from R-Y and B-Y signals. We have,

Y=0.3R+0.59G+0.11B

This can be split as (0.3+0.59+0.11) Y

Therefore, G-Y=$\frac{-0.3(R-Y)-0.11\,(B-Y)}{0.59}$= -0.51 (R-Y)-0.186 (B-Y)

It is important to note that it is possible to transmit any two of the three difference signals (R-Y), (B-Y) and G-Y). However (G-Y) is not selected for transmission because G is generally the largest amplitude colour and, therefore, (G-Y) is the smallest. Therefore, (G-Y) is more vulnerable (exposed to danger) to noise interference than (R-Y) and (B-Y) signals.

When the scene being telecast is devoid of any colour i.e., when only luminance grey shades are transmitted, colour difference signals become zero. For peak white when R=G=B=1.

Y=0.3(1)+0.59(1)+0.11(1)=1

R-Y=1-1=0 and B-Y=1-1=0

On grey shades R, G, B signals from camera tubes are less than 1 but are equal and hence colour difference signals are zero. For colour scenes, R, G and B signals are not equal and hence finite colour difference signals exist. For every colour scene the Y signal has 0.3 of red, 0.59 of green and 0.11 of blue. However for scenes of different colours, the respective values of R, G, B are different. The luminance of a given picture area is unaffected by signals carrying colour information for that area (this is called constant luminance principle).

When colour scenes are telecast with unequal R,G, B signal voltages, the Y signal still represents the monochrome equivalent of the colour because the proportions 0.3, 0.59 and 0.11 of R,G,B represent the contribution of these three colours to the luminance signal.

As an example take desaturated purple colour. This colour is another shade of magenta. The hue is magenta (Purple) and is, thus, a mixture of red and blue. The word desaturated means that it has some white content also. The white light will develop all the three R,G, B Voltages. However, R and B will dominate. Let R=0.75, B=0.6 and G=0.2. White is due to equal quantities of the three primary colours. The actual amount of white is indicated by the smallest of these three. Thus white is due to 0.2R, 0.2B and 0.2G. The remaining i.e. 0.55R and 0.4B represent the magenta hue

a) Luminance signal Y=0.3R+0.59G+0.11B

Or Y=0.3R+(0.75)+0.59(0.2)+0.11(0.6)=0.409 V

b) Colour difference signals are

R-Y=0.75-0.409=+0.341V

B-Y=0.6-0.409=+0.191 V

c) At the colour receiver the signals y (R-Y) and (B-Y) are received after demodulation. Then, by matrixing R and B signals are retrieved. Thus

R=(R-Y)+Y=+0.341+0.409= 0.75 V

B=(B-Y)+Y=0.191+0.409=0.6v

d) (G-Y) matrix. The (G-Y) signal is not transmitted but is retrieved by matrixing as illustrated below

Y=0.3R+0.59G+0.11B

Or (0.3+0.59+0.11) Y=0.3R+0.59G+0.11B

Rearranging the terms

0.59 (G-Y) =0.3(R-Y) – 0.11 (B-Y)

Or (G-Y)= -0.51(R-Y)-0.186 (B-Y)

Since (R-Y)=0.341 and (B-Y)=0.191, we get

(G-Y)=-0.51(0.341) – 0.186(0.191)=-0.209

G=(G-Y+Y=-0.409=0.2

e) **Reception at black and white receiver. In** this case luminance signal is 0.409 V . Since peak white corresponds to a luminance of 1V, this desaturated purple will appear as dull grey in a black and white receiver.

<u>Generation of Y and Colour Difference Signals</u>

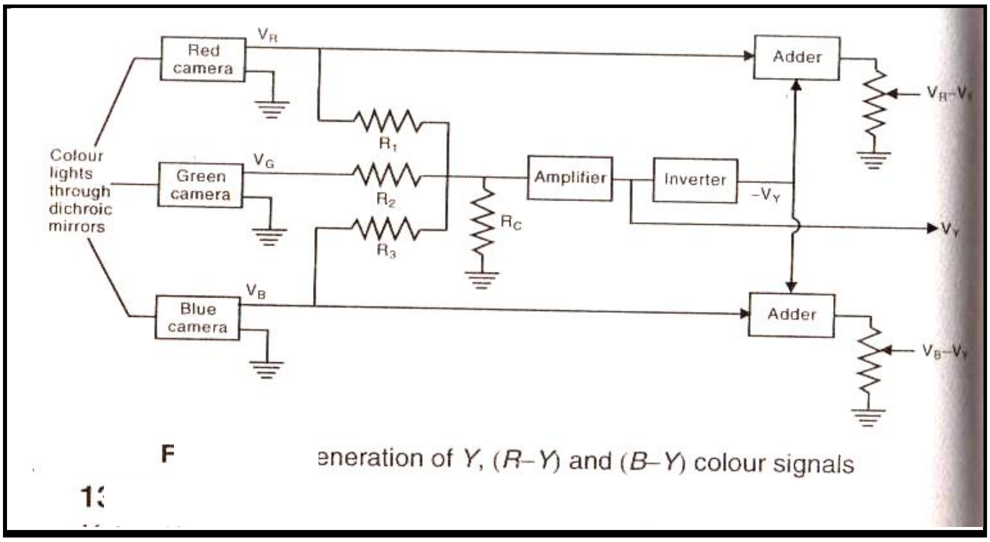


Figure shows a simple circuit which illustrates generation of y, (R-Y) and (B-Y) signal.

The three camera tubes give V_R V_G and V_B . These three are added in proportion of 30%, 59% and 11% by the resistances R1, R2 and R3 to give the Y signal. The values of R1, R2 and R3 are chosen so as to add VR, VG and VB in the above proportions. In order to avoid cross talk the resistance Vc is kept small. Therefore, the Y signal is amplified by an amplifier. Then Y signal is inverted (by using an inverter) to give – Y signal which is added to R and B signals to give (R-Y) and (B-Y) signals.

Bandwidth for Colour Signal Transmission:

Y signal is transmitted with full frequency bandwidth of 5 MHz for maximum horizontal details in monochrome. However, it must be remembered that human eye cannot differentiate between very fine colour details. Since the eye cannot recognize colours of objects below a certain finite size, it is not necessary to transmit any colour information which lies above 1.5 MHz. For very fine colour details produced by frequencies in the range of 1.5 MHz to 5 MHz every person is colour blind. Very very small pixel areas are interpreted as luminance information with only grey shades. Some complex colours human eye can perceive only large pixels which produce video frequencies below 0.5 MHz. The relatively less complex colours perceive pretty fine pixels which produce video frequencies between 0.5 MHz and 1.3 MHz. In view of this the maximum bandwidth necessary for colour signal transmission is $\pm$1.5 MHz i.e. 3 MHz.

Weighting factors:

The unweighted colour difference signals are (R-Y) and (B-Y). The magnitude and phase angle of C are thus give by

$CuW = [(R-Y)^2 + (B-Y)^2]0.5$

Where denote amplitude and phase angle of unweighted C signal.

When peak to peak amplitude of C signal is (for saturated colours) added to the Y signal to form the combined video signal, the peak to peak amplitude of (Y+C) can exceed the permissible range of video signal e.g. 100% saturated yellow colour yields Y+C between 0.89$\pm$0.9 i.e., between 1.79 and -0.01. This large amplitude may overload the video circuits and also cause over modulation at the transmitter. Instead of reducing the whole signal it is considered prudent to reduce (R-Y) signal by a factor 0.877 and (B-Y) signal by a factor of 0.493. No reduction is made for Y signal. Thus

(R-Y) weighted=0.877(R-Y) unweighted

(B-Y) weighted=0.493(B-Y) unweighted

The weighted (B-Y) and (R-Y) signals are called U and V signals respectively. Then

The weighting is done by the help of potentiometers at the outputs of (R-Y) and (B-Y) adders in Fig-50. At the colour TV receiver the values of (R-Y) and (B-Y) are increased to unweighted values by adjusting gains of colour difference signal amplifiers. This is necessary for proper reproduction of different hues.

Frequency Interleaving:

The standard TV channel is fully occupied by the Y signal. In addition the colour signal has also to be sent. The colour signal has two independent informations i.e., hue and saturation. The colour information has to be adjusted in the standard TV channel. This is done by a process called frequency interleaving.

The basis of frequency interleaving lies in the fact that the spectrum of composite video signal possesses basic periodicity of TV line frequency i.e. the spectrum consists of harmonics of line frequency. In other words the energy content of video signal is contained in individual energy bundles which occur at harmonics of line frequency. Thus

1fn=1 x 15625 = 15625 Hz

2fn=2 x 15625 =1562500 Hz and so on in PAL 625/50 scanning system.

In addition to above vertical scanning periodicity occurs at picture frequency of 25 Hz. Thus a group of harmonics occurs around 25 Hz, 50Hz, 75Hz and 100 Hz etc. Thus the video spectrum consists of energy clusters of these harmonics occurring near about the harmonics of line frequency. As the order of harmonic increases, the amplitude decreases. Thus a part of the bandwidth in monochrome signal remains unused due to the spacing between the bundles. This available space between the bundles is used for the colour signal. The colour information is located by modulating the colour difference signals with a carrier frequency called subcarrier. The carrier frequency is so chosen that its side bands frequencies lie exactly midway between the harmonics the line frequency. The frequency of subcarrier is an odd multiple of line frequency. The avoid cross talk the frequency of subcarrier is kept on the high side of channel bandwidth. It is 567 times of subcarrier is kept on the high side of channel bandwidth. It is 5676 times of one half line frequency in PAL system i.e. its value is 4.43 MHz.

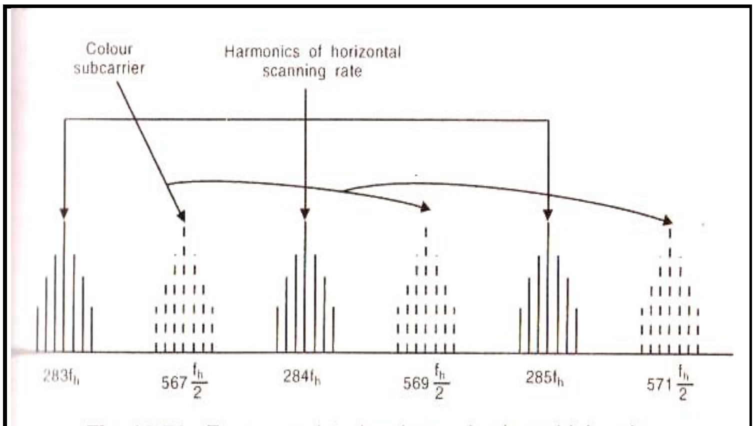
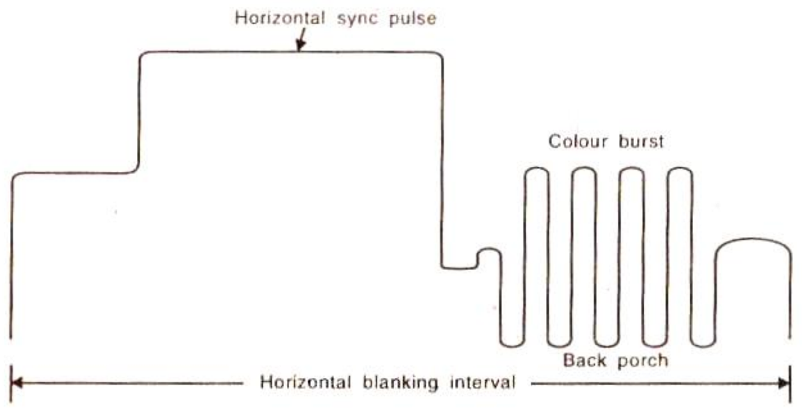


Fig. 13.51. Frequency interleaving and colour sideband

Colour Burst:

It is necessary to decrease the subcarrier interference in the picture. To achieve this colour subcarrier is completely suppressed in the balanced modulator which gives out only the chroma side bands. The ratio of side band power to carrier power increase with the depth of modulation. However, even with 100% modulation two third of total power is in the carrier and the remaining one third is the useful side band power. Therefore, suppressing the carrier eliminates the main source of interference. The transmitted signal does not contain the subcarrier frequency. However, it is necessary to generate it in the receiver with correct frequency and phase relationship so that the colour side bands can be properly detected. To ensure this a short train of about 10 cycles called colour burst is sent to the receiver along with the sync signals. The colour burst is located at the back porch of the horizontal blanking pedestal.



olour burst signal

The colour burst does not interfere with horizontal sync because it is lower in amplitude and follows the sync puses. Its location is shown in Figure. The colour burst is gated out at receiver and is used in conjunction with a phase comparator circuit to lock the local subcarrier oscillator frequency and phase with that at the transmitter.

Colour TV systems: Three colour television systems have been developed viz. NTSC system, PAL system and SECAM system.

1)NTSC : National Television Systems committee system was developed in USA and was introduced on 1.1.1954. This system is rather sensitive to phase errors introduced in the subcarrier, in transmission path etc. These phase errors give frequent changes in colours at the receiver (and, therefore, the system was also called Never The Same Colour).

This system is compatible with 525 monochrome TV system used in America. In this system advantage is taken of the fact that eye's resolution of colours along reddish blue-yellowish green axis on the colour circle is much less than those colours which lie around yellowish red-greenish blue axis. Therefore, only two primaries are used for small areas as red-orange and blue-green. This system uses two axis/ and Q. The I signal is oriented along red-orange and blue-green phasors while Q signal is along blue-magenta-yellow-green axis. The Q phasor is at right angles to I phasor. The Y, I and Q signal magnitudes are

Y= 0.30R+0.59G+0.11B

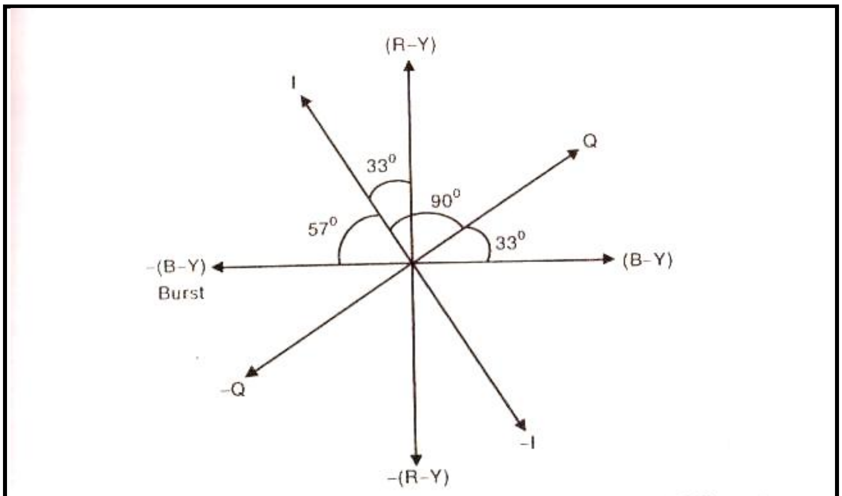
I = 0.74(R-Y) – 0.27 (B-Y)

= 0.60R – 0.28G – 0.32B

Q =0.48(R-Y) +0.41(B-Y)

= 0.21R – 0.52G+0.31B

I and Q signals are derived from colour difference signals(R-Y) and (B-Y) by using a suitable matrix. I signal lies in a region 33 degree counter clockwise to +(R-Y) where the eye has maximum colour resolution and is at 57 degree clockwise with respect to –(b-y) axis as shown in Figure.

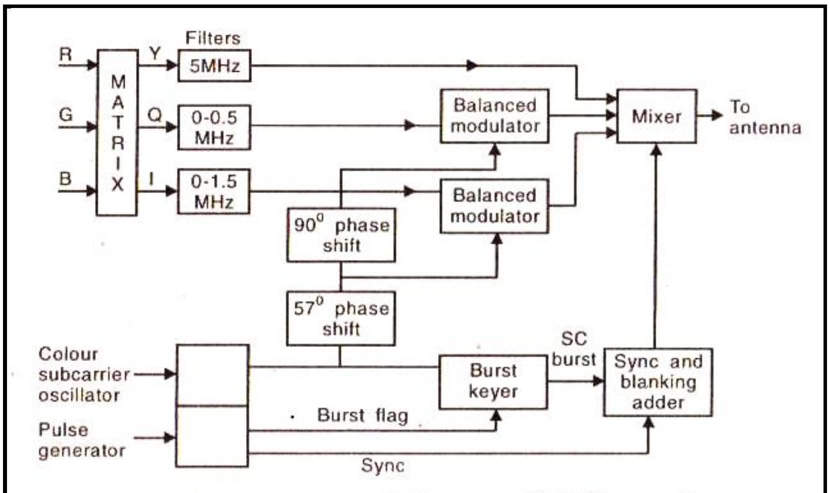


Fi_ nase relationship of colour signals in NTSC system

The Q signal is located 33 degree counter-clockwise to +(b-Y) axis. Thus I and Q signals are in quadrature to each other.

In NTSC system Q signal has a channel bandwidth of 1 MHz ($\pm$0.5 MHz). For I signal, the bandwidth extends to 0.5 MHz on the upper side and 1.5 MHz on the lower side (i.e. a total of 2 MHz for I signal). Thus a total bandwidth of 2 MHz is required for colour signal. It is important to note that if (R-Y) and (B-Y) signals are directly transmitted a bandwidth of 3 MHz is needed. Thus the use of I and Q signals means a saving of 1 MHz in channel bandwidth. The Y signal is allowed a bandwidth of 4.2 MHz. The colour subcarrier of 3.579545 MHz is Quadrature Amplitude Modulated by I and Q signals in two balanced modulators. The modulated I and Q signals with suppressed subs carrier are added to the sync signals and Y signal to give the colour plexed video signal M which can be written as

M=Y+ K Cos(wt+33 degree) +Q Sin ((wt+33 degree)

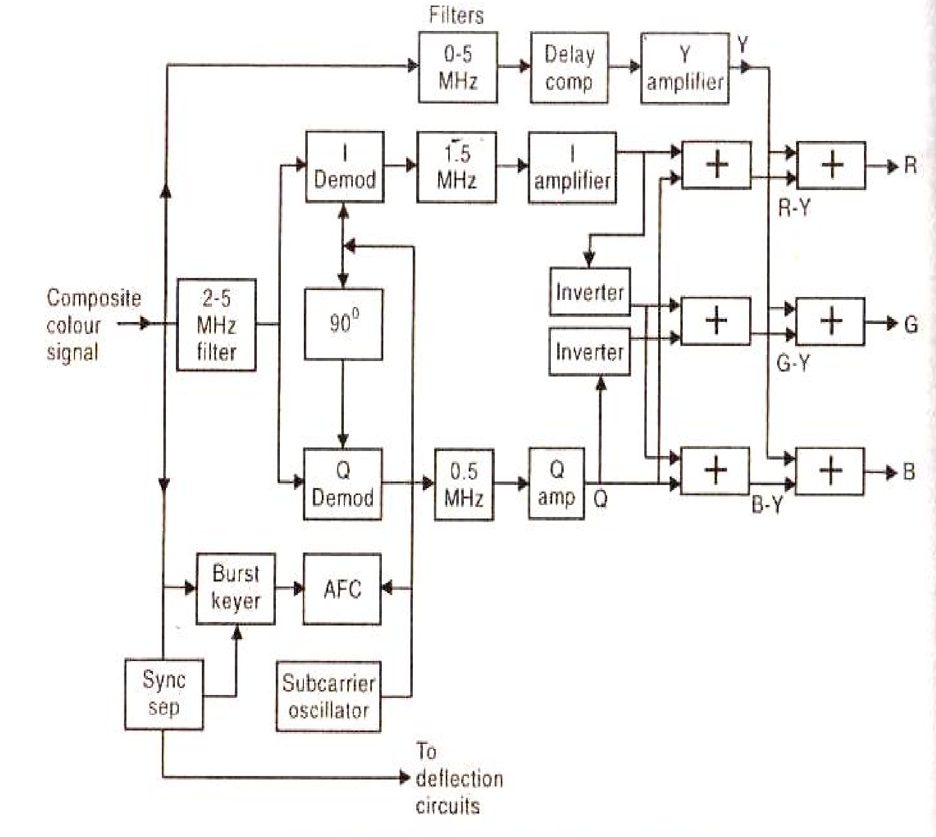


ock diagram of NTSC encoder

a) **NTSC Encoder**: Figure shows a simple block diagram of NTSC encoder.

The three camera tubes feed a suitable matrix for generating Y, Q and I signals. Since Q=0.21 R – 0.52G + 0.31B, the green camera output is inverted before mixing it with red and blue camera outputs. Moreover, I=0.60R-0.28G-0.32B. Therefore, it is necessary to invert green and blue camera outputs before mixing them with red camera output. The Y signal has a bandwidth of 4.2 MHz. The chrominance signals I and Q have reduced bandwidths of 1.3 MHz and 0.5 MHz respectively. The Y, I and Q signals produce the colour video signal. The colour subcarrier of 3.579545 MHz is Quadrature Amplitude Modulated by I and Q chroma signals in the two balanced modulators. The I modulator gets a subcarrier lagging 57 degree behind the colour burst reference signal sent from the subcarrier oscillator directly to the sync and blanking mixer-adder. The Q modulator gets its subcarrier with an additional 90 degree lag to provide quadrature amplitude modulation. The subcarrier reference signal from subcarrier oscillator is gated in by the burst gate flag pulses to feed subcarrier bursts of a minimum of 8 cycles to the sync and blanking adder. Thus the standard sync and blanking pulses are mixed with the burst and combined with Y, Q and I signals to give colour video signal which is fed to the antenna.

b) **NTSC Decoder (Receiver)**: Figure shows a block diagram of NTSC decoder (receiver).



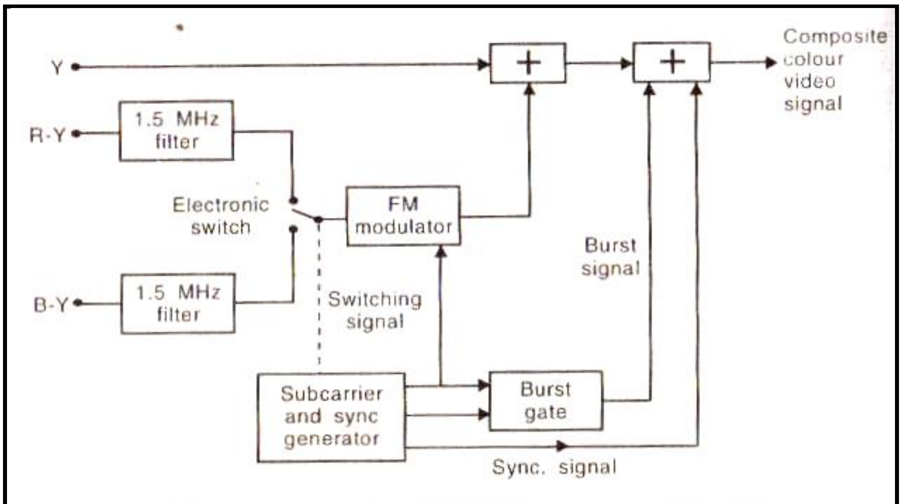
F TSC decoder (receiver)

In a television receiver the video IF amplifier strengthens the video signal. The video signal is detected by a detector. The output of detector is Y signal mixed with I and Q signals.

The Y signal goes to the picture tube through filter, delay compensator and Y amplifier of 4.2 MHz bandwidth (In some receivers Y amplifier bandwidth is limited to 3.2 MHz bandwidth (In some receivers Y amplifier bandwidth is limited to 3.2 MHz to reduce interference from the colour subcarrier in Y signal and the subcarrier in chroma section. However, in large screen receivers full bandwidth of 4.2 MHz is allowed). The I and Q signals are separated by two synchronous demodulators. A locally generated subcarrier of 90 degree phase shift is also used. The subcarrier oscillator is frequency and phase locked to the subcarrier burst signal separated from the detected signal by the sync separator and burst keyer with the help of AFC circuit. As shown in Fig-55 each demodulator has two inputs i.e. the chroma signal (which is to be demodulated) and a constant amplitude output from subcarrier oscillator. The I and Q synchronous demodulators convert the chroma signal (which is a vector quanity) into polar and rectangular components. The I and Q signals are fed to mixers to yield R-Y, G-Y and B-Y signals. These signals along with Y signals are fed to another set of mixers so as to give (R-Y) – (-Y) =R, (G-Y) – (-Y) =G and (B-Y) – (-Y)=B. Thus, we get R, G, B Signals.

c) **Limitations of NTSC System:** Many transmission path differences can occur in any system. These may arise due to differential phase distortion in studio equipments, along the transmission links, mistuned RF-IF circuits. The transmission path differences result in phase errors i.e. deviation in the phase between the subcarrier at the local receiver and the subcarrier at the transmission encoder. If the phase errors on the burst subcarrier and chroma signals are equal, their mutual phase relationship remains constant. However, due to non linearities in the signal path the phase relationships donot remain constant and incorrect hues are produced. Moreover, cross talk between demodulator outputs at the receiver causes colour distortion. In NTCS receiver a Hue or Tint control is generally provided to correct these phase efforts. Wherever a channel is changed an adjustment is needed.

d) **SECAM System:** **H**.de France proposed the SECAM (Sequential Colour a memoire) system in France in 1958 to avoid the problem of phase efforts. In SECAM system the two chrominance signals i.e. (R-Y) and (B-Y) are transmitted on the two chrominance signals i.e. (R-Y) and (B-Y) are transmitted on alternate lines in sequence. Though half the colour information is lost due to this alternate transmission, the eye cannot distinguish the colours in such minute detail and hence this loss of colour information is not a much consequence. SECAM system underwant progressive developments. SECAM III is the accepted standard.



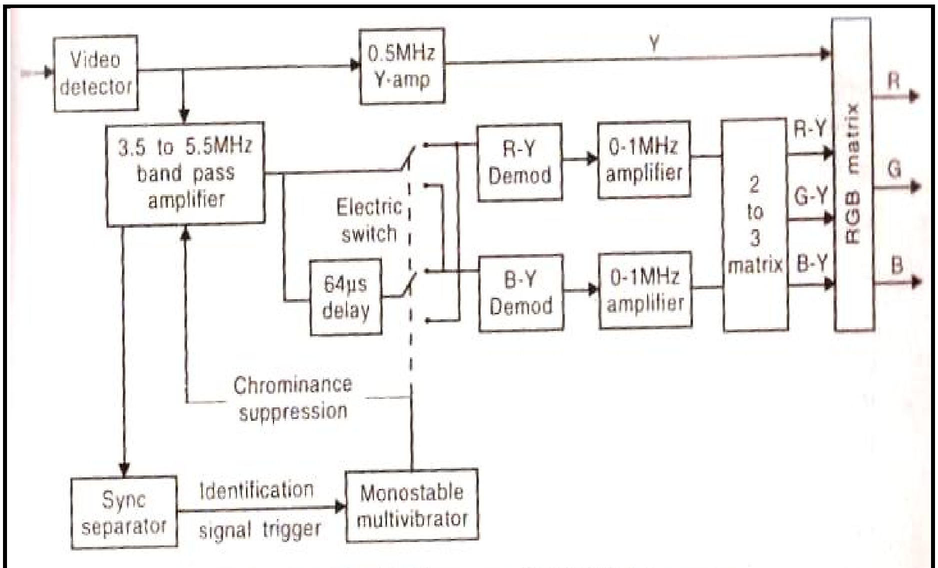
Fi ck diagram of SECAM encoder

e) **SECAM Encoder:** Figure Shows simplified block diagram of SEFCAM encoder. The colour difference signals (R-Y) and (B-Y) are band limited to 1.5 MHz by the filters. These two signals are applied to FM modulator alternately by using an electronic switch. This switch operates at line frequency rate. The colour subcarrier is, therefore, modulated alternately. This method makes the system less sensitive to pulse interference. The channel bandwidth is 8 MHz with 625 lines 50Hz scanning. The chrominance signals D_R and D_B have different weighting factors. i.e.

D_R = -1.9 (R-Y) and D_B=1.5 (B-Y)

The negative value of (R-Y) signal is necessary to give positive frequency deviations of the subcarrier to keep them away from upper end of video band. The modulated subcarrier is then mixed with luminance signal, sync and blanking signal and burst or switching pulse signal to yield composite colour video signal. The nominal subcarrier frequency is 4.4286 MHz. To suppress the dot pattern interference due to residual subcarriers two separate subcarriers fOR and fOB are used for $D_{\mathbf{R}}$ and $D_{\mathbf{B}}$ signals, i.e.

f_{OR} = 4.40625 MHz and f_{OB}= 4.250 MHz



ig. 13.57. Block diagram of SECAM decoder

(b) **SECAM Decoder**: Figure shows a simplified block diagram of SECAM decoder. The sequential colour difference signals are transformed to simultaneous signals. The chrominance amplifier passes through the FM band of about 1 MHz around the subcarrier. The chroma signal is passed through two paths i.e. one directly and one through 64 μs delay. This 64 μs delay ensures that each transmitted signal is used twice, one on the line on which it is transmitted and a second time on the succeeding line of that field. The electronic switch ensures that D_R signal whether coming directly or through 64 μs delay goes to D_R and D_B signals from successive lines are thus switched along with one line period delayed version. In this way, the two colour difference signals are effective simultaneously though they are from successive lines. The outputs of electronic switch go to FM demodulators and amplifiers and then to a 2 to 3 matrix. This 2 to 3 matrix converts 2 signals (R-Y) and (B-Y) to 3 signals viz (R-Y), (G-Y) and (B-Y). These three colour difference signals are then combined with Y signals to give R, G, B signals which are fed to picture tube.

(c) **Advantages and Disadvantages of SECAM system:**

SECAM system has many advantages. Due to FM, the SECAM receiver is free from phase distortion efforts. There is no possibility of cross talk between colour difference signals since they are not present simultaneously. The receiver does not require ATC and ACC (automatic colour control) circuits. Separate manual saturation control and hue control are also not required. Therefore, SECAM receiver is simpler and cheaper as compared to NTSC and PAL receivers.

The vertical resolution of SECAM system is interior because one line signal combines with the previous to produce colours. However, this is of minor significance because our perception for colours is not very good. In SECAM luminance is represented by a voltage but he and saturation are represented by deviations of subcarrier. Thus the colour is more saturated during fade to black. During fade out the pink colour changes to red. This is not so in NTSC and PAL systems.

PAL System:

PAL means phase alternation by line. It was proposed by Prof. Walter Bruch of Germany. In this system the (R-Y) and (B-Y) colour signals are weighted by 0.877 and 0.493 to give V and U chroma signals. Weighting reduces the chances of over-modulation of the transmitter carrier in the presence of highly saturated colours when picture has very light or very dark parts. They are both allowed a bandwidth of .3 MHz. The colour difference signals carry all the colour information and are transmitted as modulation of a subcarrier i.e. 4.43 MHz. At the receiver the process is reversed to recover R, G,B signals. The modulated picture subcarrier containing the colour information is accommodated in the same standard television channel (5 MHz) by a process known as frequency interleaving. The main features of PAL system can be summarized as under:

1) The signals (R-Y) and (B-Y) are weighted by factors 0.877 and 0.493 to give V and U signals. The V and U signals are modulated without giving any phase shift.
2) Both V and U signals arre given a bandwidth of 1.3 MHz on modulation.
3) The chroma signal is of vestigial side band type. The upper side band attenuation slope starts at 0.57 MHs (i.e. 5-4.43 = 0.57 MHs) but the lower side band extends to 1.3 MHz before the start of attenuation.
4) The colour subcarrier frequency is 4.43 MHz(4.43361875 MHz precise value). It is an odd multiple of one quarter of line frequency.
5) The phase of V signal is reversed on alternating lines. Phase errors which may be present on one line are cancelled by equal and opposite efforts in the next line. Thus the effect of colour shifts on human eye is cancelled.

6) Chroma signals are combined electronically by means of delay line and additional circuits in the PAL receiver. This feature exists in PAL-D which is the present version of PAL and is standard PAL now-a-days.

7) The PAL colour bursts are made to swing in phase 45 degree on either side of –U axis in synchronism with phase alternations of V subcarrier.

a) **PAL Encoder:**

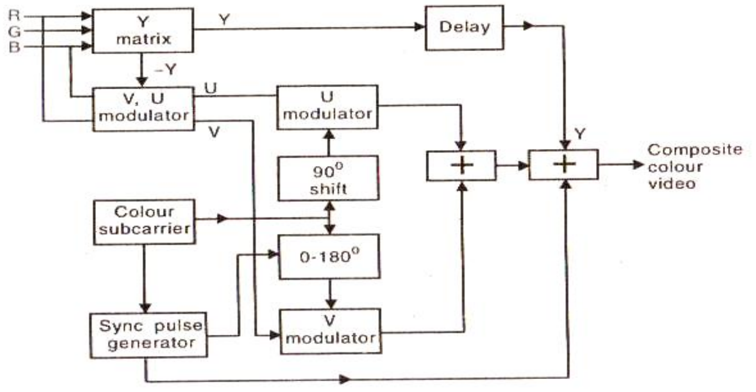


Fig. 1 : diagram of PAL encoder

Figure shows a simplified block diagram of PAL encoder. The Y matrix combines the R, G, B signals (from camera tubes) to form the Y signal.

$Y=0.30R +0.59G+0.11B$

Where Y is luminance, R is red, G is green and B is blue. The V, U matrix combines R, B, Y signals with =weights to give V and U signals. Weighting prevents over modulation on saturated colours

$U= 0.493 (B-Y)$

$V = 0.877 (R-Y)$

The collour subcarrier is modulated in quadrature by V and U signals band limited to 1.3 MHz. The subcarrier is fed to U moidulator through a 90 degree phase shift network. The V modulator gets the subcarrier after 0-180 degree phase shift on alternate lines. The two modulated colour signals are added and the resulting signal is further combined with Y, sync and blanking signals and swinging burst obtained from sync pulse generator. The output of the last adder stage is composite colour video signal.

b) **PAL Decoder:**

Figure shows a simplified block diagram of PAL decoder. In many respects it is similar to NTSC decoder. A special feature of PAL decoder is delay line. This delay line is an ultrasonic delay line fitted with two piezoelectric ceramic transducers one acting as input driven by chroma signal and the other gives out a chroma signal delayed by one line period.

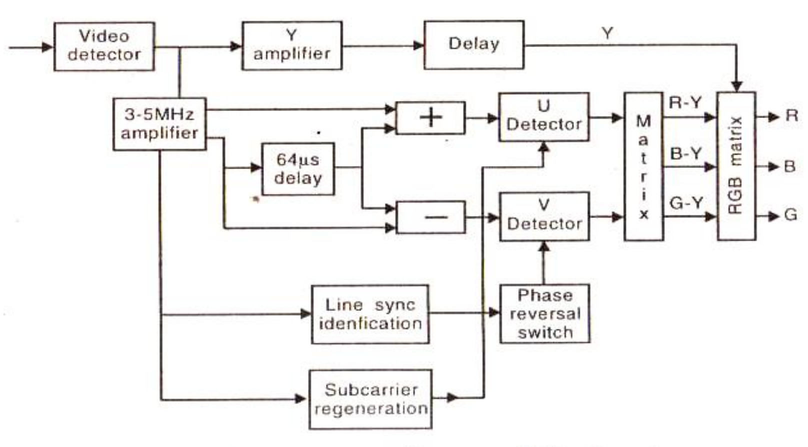


Fig. ‹ diagram of PAL decoder

The direct signal and the delayed signal are combined to separate the U and V signals. If one line has chroma signal U+jV, the next line has U-jV (because V signal is phase reversed every alternate line). When output of delay line ia added to the direct signal the result is (U+jV) +(U-jV) = 2U. Thus we get U signal. When output of delay line is subtracted from the direct line we get (U+jV) -(U-jV) = 2jV. Thus we get V signal. The U, V and Y signals are combined in the R, G, B matrix to yield R, G, B colour signals which are fed to colour picture tube.

Colour Television Receiver:

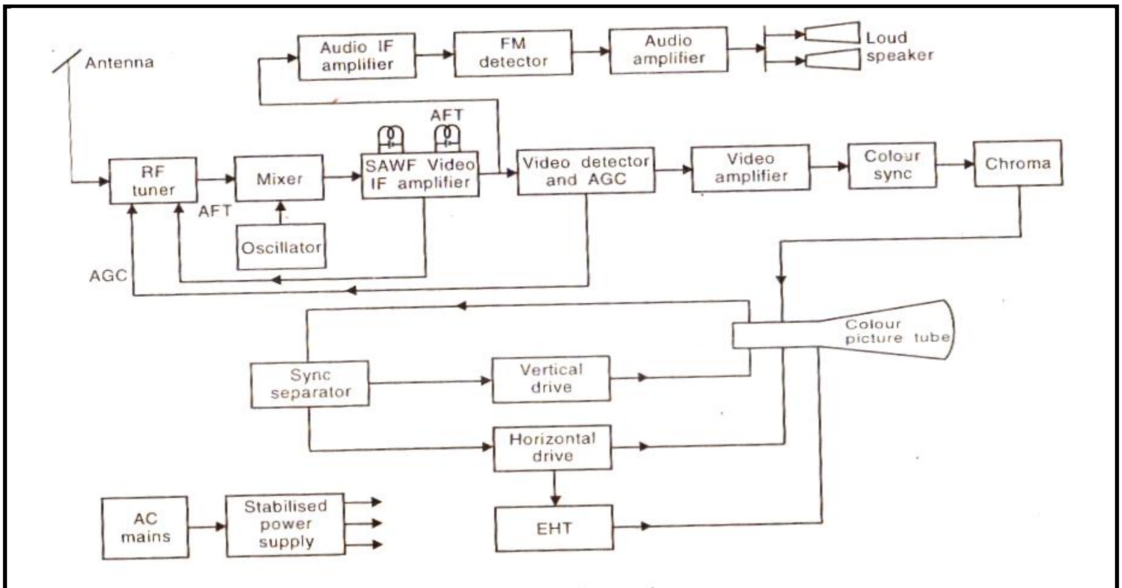


Fig. 13.60. Colour TV receiver

Figure shows a simple block diagram of a colour television receiver. It is very similar to that of a monochrome receiver but contains a colour picture tube and a chroma section.

It is seen that sound take off point in Figure is after the video IF amplifier but before the video detector. This is done to avoid interference between the sound IF and the chroma signal.

A colour television receiver is provided a very low noise RF tuner having flat frequency response over the whole bandwidth of the channel. The tuning must be stable to ensure that side bands do not suffer any degradation which would affe3ct picture quality. An automatic fine tuning (AFT) is necessary to ensure correct tuning.

Another feature of a colour television receiver is that a surface acoustic wave filter (SAWF) using a piezo electric device is used along with a preamplifier to give stable band pass response and high gain (80 dB).

The chroma section is for decoding of colour signal. It contains the necessary circuits to amplify and detect the chrominance © signal. The colour difference signals, after demodulation of C signal, are mixed with Y-signal so as to again produce red, blue and green video signals which modulate the three beams of tricolor picture tube. Thus chroma section contains the necessary circuitry to decode the modulated colour sub carrier C signal.

Three Tube Colour Camera:

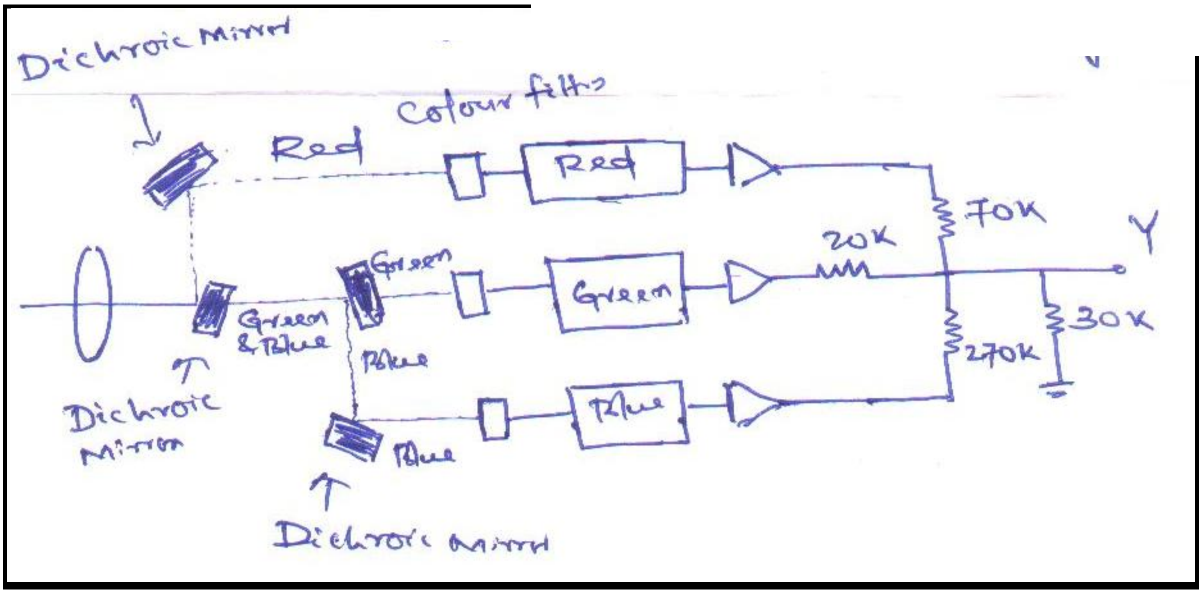
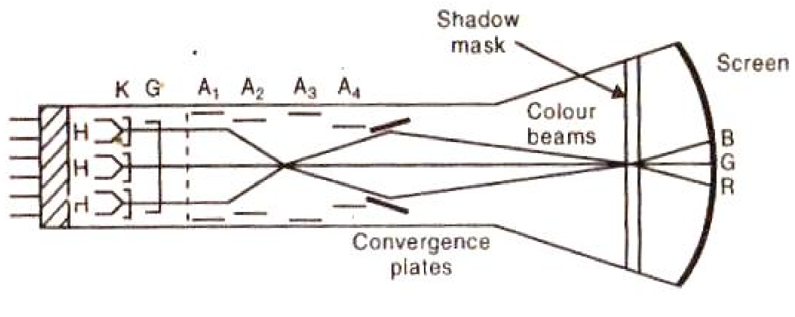


Figure shows a simple block diagram illustrating the principle of a colour TV camera.

The light coming from the scene is fed to special colour filters (known as dichroic mirror system) which split the light and allow only one colour to pass through it. One filter allows only blue light to pass through it, the second allows only green to pass through it while the third allows only red to pass through it.

Figure shows a complete block diagram of a colour camera tube illustrating the generation of colour signals and matrix for obtaining the colour brightness signal.

Colour Picture Tube:



F olour Picture Tube

Figure shows a simplified diagram of a colour picture tube. Many types of colour picture tubes have been developed over the years but the Trinitron tube shown in Fig-61 is one of the most commonly used one. It was developed by Sony Corporation of Japan in 1968.

It consists of a single electron gun with three separate horizontally mounted in line cathodes for the three red, blue, green colours. The control grid G is a single cup electrode with three apertures for the three coloured beams. The other electrodes A1, A2, A3, A4 are common for all the beams. The beams cross over to converge within the focus field before leaving the gun to cross again at the slots in the shadow mask. When these beams strike their respective targets on a phosphor screen coated with phosphors corresponding to red, green and blue colours the phosphor dots are excited. The three phosphors are arranged in triangular groups called triads.

It is necessary to ensure that a particular beam strikes only its own colour phosphor dot in the triad. For this purpose the three beams are made to converge a small distance behind the screen. At this point a thin perforated metal sheet called shadow mask is placed as shown. This shadow mask has several hundred thousand very fine holes so that one hole is available for every dot triad on the screen. The alignment is such that the red beam can illuminate red dot only, blue beam can illuminate blue dot only and green beam can illuminate green dot only.

The functions of anodes A_1, A_2, A_3, A_4 are essentially the same as in monochrome tube.

Some other special components of colour picture tube include purity magnetic rings (to ensure that each beam strikes its own phosphor dot), ring magnets (for convergence of beams), automatic degaussing system (to remove extraneous magnetic fields) etc.

Compatibility between Colour and Monochrome Systems:

A colour telecast can be viewed on a monochrome receiver. A monochrome telecast can also be viewed on a colour receiver. Thus compatibility exists between the two systems. Of course the reception in both the above cases will be in black and white only. The ensure compatibility the following steps are taken:

1. The monochrome luminance signal is telecast as a separate entity in the composite colour video signal. The monochrome set can, thus, function without any problem.
2. The colour signal is transmitted in such a method that a monochrome receiver ignores this transmission but a colour receiver can produce a proper colour picture.
3. The overall bandwidth and channel width is same for both transmissions.
4. The position of sound carrier in the channel is not affected. The colour scheme mentioned in section 13.26.3 consisting of Y signal and C signal satisfies the above required conditions for compatibility.

<u>Video Monitor:</u>

A video monitor is a display device in computers, closed circuit television systems etc. It is similar to a television receiver but is somewhat different also. These differences are as under:

1. **Direct Base band Signal:** A television receiver gets RF modulated signal from the antenna and converts it into video output (for picture tube) and audio output (for loudspeaker). On the other hand a video monitor is given the base band video signal at its input and converts it into video and audio outputs. Therefore it does not need the tuner and IF amplifier stages.

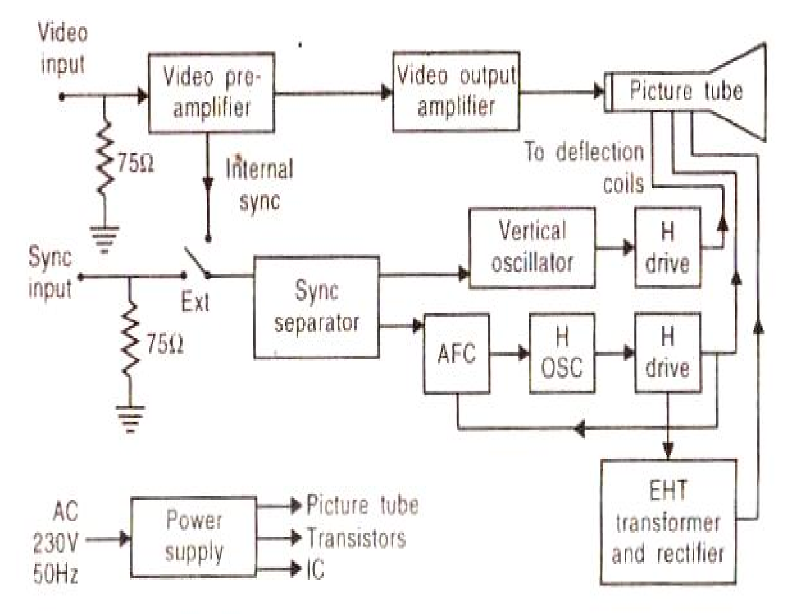
2. **S**ome video monitors do not have the audio section.

3. **Deflection Linearity**: A smaller angle of deflection gives better linearity, lesser geometrical distortion and lesser defocusing effects near the edges and corners. A grade video monitors use 90 degree deflection picture tube. B grade video monitors use 110 degree deflection picture tube. Television receivers use 110 degree or 114 degree deflection picture tube.

4. **Bandwidth and Resolution: T**elevision systems have a bandwidth of 7 MHz. Video monitors use 8 or 10 or even higher bandwidth. Television receivers have 625 lines per frame. In video monitors the resolution can be high because there is no limit on frequency. Computer video monitors have horizontal and vertical resolutions of 1024 and 768 respectively.

5. **Input Sensitivity: Input** signal for video monitors is generally 1 V $_{p-p}$.

6. Television receivers always have internal sync. However video monitors may have external sync also in addition to internal sync.



Fi... ck diagram of black and white video monitor

Figure shows a block diagram of a video monitor. The various terms are the same as in television receivers. Most video monitors have a 75 Ohm termination for the shielded cable which supplies the video signal. The monitor in Fig-62 does not have audio section.

Desk top computers use the cathode ray tube monitor shown in Figure. The laptop and notebook computers use flat panel display which may be a liquid crystal display (LCD) unit or electro luminescent (EL) display. To connect a monitor to a computer one needs a graphic adapter board (more commonly known as video card). Some graphic boards have video memory also.

Closed Circuit Television (CCTV):

CCTV system is very useful and has applications in education and training, industry, security, surveillance etc. These systems essentially consist of TV camera and video monitor along with associated circuitry. The TV camera pectoris's the scene and it is displayed on monitor. One camera may feed a number of monitors (placed at different locations) or a number of cameras may feed one or more monitors.

1. Synchronising System: CCTV systems do not have master synchronizing pulse generator. The mains supply frequency of 50 Hz with suitable wave shaping circuit is used to trigger the vertical oscillator. The horizontal oscillator is a crystal controlled astable multivibrator. The output of astable multivibrator is in the form of clock pulses which are used to establish the timing sequences necessary for synchronization. In this system of synchronization the line spacing between the two fields may vary in a random fashion. Therefore, it is known as random interface system.

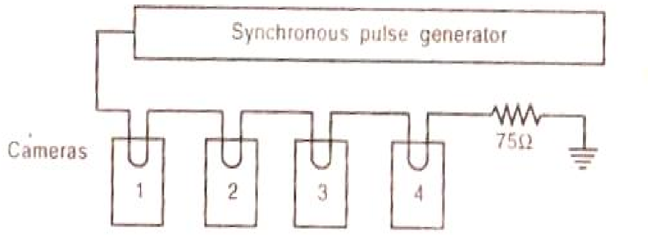
2. Self Contained Units and Remote Units: CCTV system may have self contained camera units or remote operation type camera.

Self contained camera unit has all the equipment (i.e. camera, video processing circuitry, synchronization and blanking pulses circuitry etc) at one place. The video signal from the unit is sent to monitors through cables.

Remote operation cameras have all the equipments except camera tube at a central place. The camera is separate and is installed at the actual site e.g. on the ceiling above the operation table in a hospital. The camera is connected to the remaining equipment through multi core cable. In this arrangement the camera can be located very near the actual scene to be shown and at a convenient place.

3 Common Synchronisation System:

A CCTV may have more than one camera to show different aspects of the scene e.g., demonstration of a surgical operation is done by using three cameras located at suitable positions around the body part to be operated on. All these cameras can have a common synchronization system. The synchronization pulses could be provided by a common synchronous pulse generator (SPG).



63. A ngle Synchronous Pulse Generator Feeding 4 Cameras in a CCTV System

Figure shows a common synchronous pulse generator feeding four cameras and terminated finally in 75 Ohm cable impedance.