CHAPTER 1

- Q1. Explain the characteristics of Microphones.
- A.1 There are many types of microphones available. Each has certain advantages and disadvantages. Hence the selection of microphones depends upon the certain characteristics as below:
 - (1) Output level; (2) Frequency response; (3) Output impedance and (4) Directivity.

(1) Output level: -

The output level of a microphone governs the amount of amplification that must be available for use with the microphone. The output level of microphone is usually given in dB preceded by a minus sign. The minus sign means that the output level is so many dB below the reference level of 1milliwatt for a specified sound pressure.

(2) Frequency response: -

The frequency response of a microphone is a rating of the fidelity of relative output voltage which results from sound waves of different frequencies. The simplest way to find a complete picture of the frequency response characteristics of a microphone is to plot a curve of its output voltage v/s input frequency. Since good microphones are relatively flat over their range, it is often considered sufficient to specify the range over which their output does not vary more than ± 1 -2 dB.

(3) Output impedance: -

A microphone, like any other component with electrical i/p of o/p, has a value of impedance. When a microphone is connected to an amplifier, a complete circuit is formed and electric current flows whenever a sound causes the microphone to generate an electrical voltage.

For most high quality microphones impedance is low, a few ohms ranging upto a hundred ohms or so. The importance of microphone impedance is not a matter of the precise value but of the ability of the microphone and the recorder to be matched together. High impedance microphones must be connected must be connected into a recorder with high impedance input, otherwise both the signal amplitude and the frequency range will be adversely affected.

(4) Directivity: -

Microphones do not respond equally to sound reaching them from all directions. Their frequency response characteristics also vary, depending on the angle at which the sound reaches them. The way in which a microphone responds to sounds coming from different directions is plotted on a circular graph which is known as a polar diagram. The centre of the circle is zero point and concentric circles indicate successively higher levels of response as they move outward. The top of the circle is the front and the bottom the back of microphones, and the straight lines radiating from the centre denote the corresponding angles.

There are few of the typical types of the directivity pattern available as the standard directivity patterns. These are shown below:



Fig 2.2 Omnidirectional pickup pattern

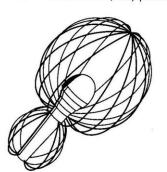


Fig. 2.4 Hyper or super-cardioid pickup pattern



Fig. 2.3 Cardioid pickup pattern

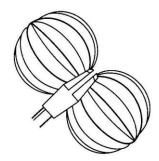


Fig. 2.5 Figure-of-eight pickup pattern

Q2. Explain the types and applications of microphones from below (Any 2): Carbon, Crystal, Electret, Tie-Clip and wireless.

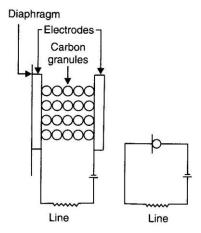
A.2 (1) Carbon Microphone: -

This type of microphone does not produce the e.m.f. but modulate the current obtained from the external supply. The essential components of this microphone are two electrodes and carbon granules which are loosely packed in between the electrodes/ one

electrode is fixed relative to the other which carries a diaphragm to respond to the pressure changes of the sound waves. Movements of this diaphragm vary the resistance of the

granules and so control the line current in accordance with the sound waves reaching the diaphragm. An increase in pressure produces a reduction in resistance and an increase in current. The elementary circuit is as shown below:

The average output level of carbon microphone is of the order of -30 dB and has a frequency response of approximately 60 - 70,000



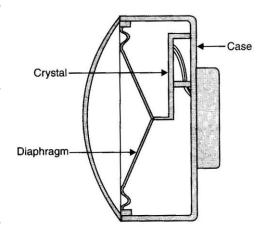
Hz. They are substantially no-directional. When the maximum output level is required from a microphone, the carbon microphone is often used. The frequency response characteristics of the carbon microphone are poor and cannot be used for high-fidelity work.

(2) Crystal Microphone: -

Certain crystals, such as a Rochelle salt and quartz possess the property of

generating small e.m.f.'s when subject to stress or strain. This effect is utilised in what is known as the Crystal microphone. The construction of a crystal microphone is as shown below:

A thin finger shaped slice of crystal is secured at one end by means of a compliant clamp, and the apex of a cone is



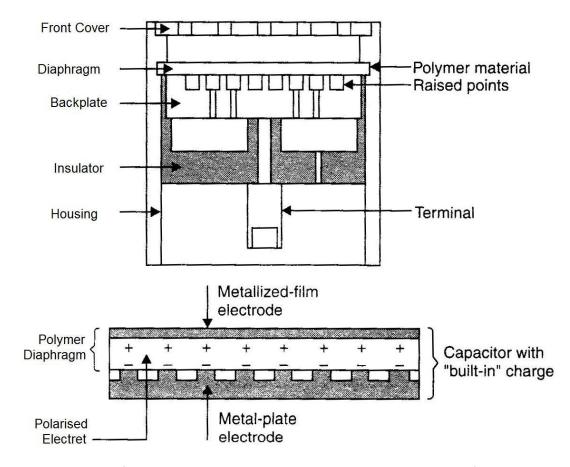
made to bear against the other. Sound pressure waves cause the cone to alternately press against and bend the crystal slice and release it. Thus, corresponding voltages are generated across the slice. A pair of contacts is fixed to opposite surfaces to take off the signal.

An improvement is obtained if the single slice of crystal is replaced by two slices cemented together. Then, when pressure is exerted, one slice is compressed while the other is stretched. This double crystal unit is termed as bimorph.

Another type of construction is the sound cell where several crystal elements are sealed together, this also being termed as multimorph. The crystal microphone is the type most widely used in lower cost installation. It has a relatively high output level and high impedance. The output level of this type of microphone is usually between -48 dB and -60 dB. Their output impedance is almost always more than 1,00,000 ohms. Good units have a frequency response between 50 - 10,000 Hz. They are non-directional and are not really durable when subjected to a high temperature and shock.

(3) Electret Microphone: -

In Electret microphone a permanent electrostatic charge is implanted into the metallised diaphragm, thereby eliminating the need of an external high-voltage source.



One way of doing this is to put the diaphragm between the plates of an air-spaced capacitor which is then charged upto a high-voltage. The sheet is ten heated and allowed to cool with the charge still maintained across it. When it is removed, it has a permanent static charge which is the equivalent of an applied voltage of around 100 V. This may be

considered on par with a permanent magnet which retains its magnetism after the

energising field has been removed.

Any electrostatic charge will slowly leak away due to the fact that there is no

material yet known that is a perfect insulator. The charge in the diaphragm will gradually

diminish and also the effectiveness of the microphone. The half-life of this type of

microphone is claimed to be 100- 1000 years. A built-in amplifier is still necessary, being

supplied by a small battery, as the output impedance here is of the order of conventional

capacitor microphone.

(4) Tie-clip Microphone: -

The Tie-clip microphone is a most widely-used portable microphone with the factors

of having small size and lightness.

Here again the Electret microphone gains the preference. These units can be made

very small and are inherently light. Although they need an internal amplifier, these can be

formed on a tiny chip of silicon in the form of an integrated circuit, and so add hardly

anything to the bulk or weight. A low voltage is needed to power an amplifier and hence a

small battery can power the device for a sufficiently long duration of operation. Battery life

extends from 5,000 to 10,000 hours depending upon use.

Chest cavity resonance is less of a problem with the tie-clip or lapel unit because the

microphone is not supported directly on the chest but stands off to some extent.

(5) Wireless Microphone: -

The wireless microphone is nothing but any microphone having a wireless module

attached to it. The wireless microphone comes in two basic types: One with the radio

transmitter built-in the case of the microphone and another with the cable attached to a

slim pocket unit about the size of a wallet which can be plugged into by any microphone. The

batteries are atleast 9V but the size limits the capacity.

The lavalier or Tie-clip microphone attached to the radio transmitter gives the

complete freedom to the speaker. There are fifteen frequencies allocated for the wireless

microphones.

The frequencies are in four groups:

Wide-band: 174.1, 174.5, 174.8 and 175.0 MHz

Narrow-band: 174.6, 174.675, 174.77, 174.885 and 175.020 MHz

Narrow-band (For deaf children): 173.4, 173.465, 173.545 and 173.64 MHz

- Q3. Explain the properties of Ideal and Basic Loudspeaker.
- A.3 Properties of Ideal Loudspeaker:
 - An electro-acoustic efficiency approaching 100%.
 - An acoustic output response that is independent of frequency over the entire audible range.
 - No interference with harmonic and inter-modulation distortion in its output.
 - Faithful reproduction of transients as well as steady input signals.
 - Capable of producing a non-directional radiation pattern,
 - Should be as small as in size as is possible considering the required acoustic output.

No single transducer has been designed that is capable of satisfying all the above requirements. Hence the characteristics of *Basic loudspeaker* are as follows:

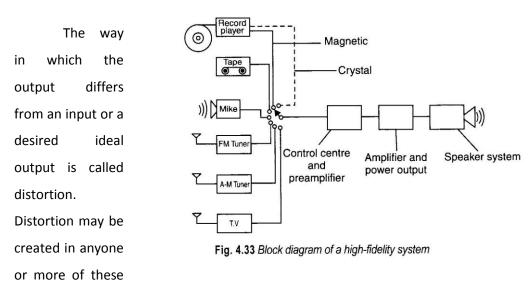
- The two problems faced by the early engineers were: amplifying the electrical signal enough to drive the loudspeaker and designing an off-the-head speaker.
- One of the first speakers that were designed was simply a horn attached to an ordinary telephone earpiece.
- This speaker had fairly large amplification but very poor frequency response.
- In another design were a vibrating reed was used instead of the thin disk of magnetic material. The free end of the reed was attached to a speaker cone, and the reed's vibration caused the whole cone to vibrate.
- Yet another type consisted of a balanced pivoted armature located between a pair of magnets. Electrical impulses through the magnet coils caused up-and-down vibrations.
- Q4. Explain the construction and working of Hi-Fi Systems.
- A.4 There are several concepts for High-Fidelity systems:
 - (a) 'Play it exactly as performed'; (b) 'play it so it sounds real'; (c) 'play it the way I like to hear it'.

There is no exact scientific operational definition of a hi-fi system as yet. Standards and specifications are dependent on various factors such as variations in human taste, room acoustics, system distortions, noise and comparative volume levels.

A commonly accepted concept of hi-fi system is that it should reproduce sound with high degree of similarity with the original sound that has travelled from the source and has undergone several conversions from electrical to acoustical.

Like photography modern hi-fi techniques encompasses controls for modification of the original sound to compensate for certain defects and make provisions to actually improve the effects according to an individual listener's tastes.

A complete hi-fi system may be divided into functional sections as shown in the block diagram below:



sections. If more than one section is causing distortion, the final output sound may reflect the sum of the distortions from all distorting sections. A section may be purposely designed to introduce distortion of a type which compensates for inherent distortion in another section.

The hi-fi system is somewhat like a chain, which is likely to be limited in overall performance by its weakest section; but the chain analogy breaks down in the case where distortion is used to compensate the distortion created in another section.

The speaker system is the weakest link in hi-fi system as there are two energy conversions taking place in form of electrical-to-mechanical-to-acoustic. Such energy conversions are known as transduction.

Input devices such as microphones also inherent such a weakness and involves exactly the reverse two conversions and hence weakens the system's response.

Depending upon the number of distortion and its introducing factors we may summarize the ideal system as follows:

- 1. Interpret, amplify, compensate and reproduce sound components of any and all frequencies in the audible range with good efficiency.
- 2. Add negligible frequency components not in the original sound.
- 3. Distribute the sound in such a way that its sources would appear to be located nearly the same as they were in the original and so that the quality of the sound would be independent of the location of the listener w.r.t. the speaker system.
- 4. Allow negligible unnatural delay of some frequency components relative to others.
- 5. Reproduce without resonance effects or hangover, the sudden large changes in sound volume level.
- Q5. Explain the Cross-Over networks. Differentiate between basic and filter system.
- A.5 In multi-speaker systems, in which specialised speakers are used for different frequency bands, it is necessary and desirable to segregate (separate) different bands of acoustic energy ensures optimum utilisation of audio power resulting in better overall performance of the system.

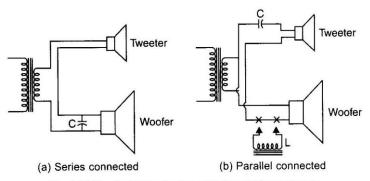


Fig. 4.36 Simple divider circuits employing single reactances

The simplest type of network consists merely of a single capacitor. The fact that the reactance of the capacitor is inversely proportional to frequency is employed to distribute the audio signal. The tweeter and woofer voice coil are connected in series and a capacitor is connected across the woofer. The value of the capacitor is made such that frequencies above the range of the woofer, the reactance of the C becomes so low that it shunts the woofer, which acts as a bypass capacitor.

Inductances can be used with capacitors to make the crossover network more complete. The inductance can be used such that the inductor, whose reactance increases with frequency, chokes the high frequency components out of the woofer, and the capacitor blocks low frequency components out of the tweeter.

Although the crossover range should not be too narrow, simple reactance circuits are ordinarily too broad in the changeover region. A combination of low pass filter and a high-pass filter is usually employed. With this type of circuit, much more rapid attenuation can be made near the crossover frequency than is possible with simple capacitor and inductor arrangements.

Attenuation of 12 dB per octave is considered proper in most applications. Gradual crossover arrangements attenuate at about 6 dB per octave. Filter with sharper cut-offs than this involves extensive power loss and additional sharpness is not necessary anyway.

The two methods of implementing cross-over networks are differentiated based on the following points:

Sr. No.	Basic Filter	High Order Filter				
1.	The filter is less bulky and easy	These types of filters are bulky and				
	to implement.	complicated to implement.				
2.	Less number of components	More number of components involved.				
	involved.	more number or components involved.				
3.	Substantially lower sharpness in	Substantially higher sharpness in				
	frequency cross over region.	frequency crossover region.				
4.	Wastage of power during	Power savings during crossover.				
	crossover.					
5.	Introduction of hiss due to	Sharper crossover introduces less				
	overlap.	interference and hiss.				

Q6. Explain the characteristics and planning of PA Systems.

A Public Address (PA) system consists basically of a microphone, amplifier and speaker(s) to facilitate the communication of intelligible speech to groups of people. Coverage of large groups may be the chief purpose of providing communication over outdoor areas or of providing sound reinforcement indoors with sufficient fidelity so that the sound system is unobtrusive.

The characteristics of PA systems may be considered as follows:

1. Speech reinforcement systems include more than one sound source for better coverage of the area.

A.6

- 2. The maximum tolerable delay between sounds arriving from two sources should not be more than 25 milliseconds in any case.
- 3. A cluster of speaker or speaker columns can be used to effectively cover the region.
- 4. The intensity of delayed sound should not be more than 10 dB greater than that of the first-arriving sound; otherwise separation is likely to become evident.

Hence proper planning will provide with efficient results. The planning of PA systems includes the following points:

- i. Calculation of the area to be served,
- ii. Estimation of the required audio power,
- iii. Choice of the types of speakers to be utilised, and
- iv. Estimating the approximate number of speakers.

Besides these factors the others are:

- ✓ The selection of amplifier plays a major role in the planning of PA system.
- ✓ The amplifier should be selected such that there will be a provision of adding few more speakers and microphone in the system if needed.
- ✓ The placement of the speakers also plays an important role in acoustics.
- ✓ The level fader must be included in the amplifier so that the choice of the sound source may be implemented.

Besides the selection of the amplifier, the selection of microphone is also important. The choice of microphone is chiefly determined by intended use. The basic conditions are:

- a) The pick-up element that is utilised,
- b) The pick-up pattern that is optimum for the location, and
- c) The conditions under which the microphones have to operate.

The next come the selection and implementation of speakers. The separation of the speakers determines the optimum usage of the amplifier. Hence the separation of the speakers will determine the maximum coverage with minimum power usage. And this will help to choose the proper amplifier for the faithful operation.

Q7. Explain the block diagram and working of Theatre Sound System.

A.7 The figure shown below depicts the theatre sound system. Here there are arrangements made such that the sound reproduction creates the environment similar to that present at the time of recording.

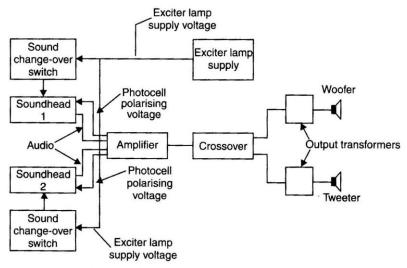


Fig. 27.9 Block diagram of a theatre sound system

As shown in the figure, there are two sound heads which are controlled by the sound change-over switch which is actually controlled by the exciter lamp supply. The reason behind this control is to mix the appropriate sound to reproduce the original sound from the film.

As we know that the sounds recorded on the films are speech signals as well as music signals (i.e. both high and low frequency components are present). Hence both Tweeters and Woofers are included in the speaker or o/p section. The two sound heads are further connected to the amplifier which amplifies the instantaneous audio signals coming from either sound head. The speakers are supplied with appropriate signals with the help of the crossover networks employed appropriately to devise the system.

The arrangement of the speakers is so made that the effect thus produce tend to the original sound coming from the actual source while recording is done. In order to achieve that, large woofers and multi-cellular horn type tweeters are also placed behind the screen so as to create an impact that the sound may actually be arising from the screen itself.

There are also few auxiliary tweeters placed near the screen on both sides such that it may be switched in and out as per the requirement of the sound track. This switching is basically done with the help of the control track which is available on the film.

There are few screens available in the market with built in speakers i.e. woofers and tweeters so that there is no need of separate sound arrangements. But their capabilities are

limited by its size. Horn type speakers are generally employed with low output in order to cover the theatre sufficiently with high intelligibility.

CHAPTER 4

Q1. Explain the application of solid state devices in automobiles.

A.1

Field of Applicatio	Signal Devices	Power Devices	Hybrid Subassembli es	Application s for IC's	Transducers and Sensors
Safety	Speed limits warning, obstacle warning.	Automati c braking, automatic headway control, Air bag trigger.	Automatic headlight dimmer, Obstacle detection.	Anti-skid braking, Failure monitoring, Drunk driver spotter.	Passive restraint sensor, Tire pressure monitor, Manifold pressure sensor, Engine temperature sensor.
Propulsion	Ignition trigger.	Breaker- less ignition, chopper for battery powered cars.	Voltage regulators, Electronic fuel Injection.	Triggers for power devices.	Torque measurement, Ignition timing Compression measurement.
Comfort and	Air conditioner	Remote control	Transistorised clock,	Automatic vehicle	Temperature control devices

convenienc	control,	for radio	Automatic	control.	(thermistors).
е	Headlight	of tape	temperature		
	time delayed	equipmen	control, and		
	action.	t, Cruise	Electronic		
		control.	speedometer.		
Entertainm -ent	Radio and tape player	Electronic tuning	Speed regulator for tape player drives.	AM/FM/MP X radio Tape players	
Economy of operation		Electric propulsio n	Sequential turn signals	Coolant temperature indicators, Electric fuel pump, Voltage regulator	Fuel flow measurement and control, Electric fuel pump.
Control	Fuel and coolant gauges, temperature indication, Electronic tachometer.	Voltage regulator, shaft and rotor position control.	Interval windshield wiper control.	Voltage regulator	Liquid gauges, Temperature control, Braking control.
Lighting	Automatic head lamp dimmer.	Instrume nt lighting.	Headlight and dome light delay.	Triggers for flashers	Instrumentatio n
Maintenan- ce	Brake fluid level warning, Tire pressure warning.	Ignition	Trouble indicators, Multiplexed power distribution.	Engine performanc e analyser, Voltage regulator.	Failure sensors.

- Q2. Explain Electronic Ignition Lock system.
- A.2 The Electronic Ignition Lock system is a technology used to prevent drunken driving. In modern cars, these systems are extensively used to ensure safe riding. This also ensures the car from theft.

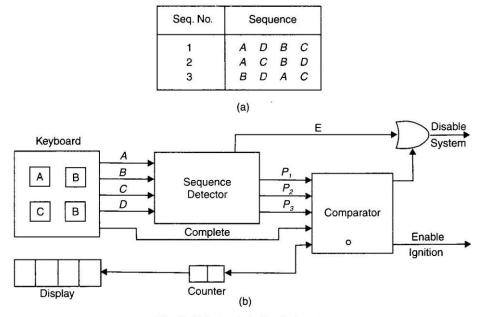


Fig. 49.4 Electronic ignition lock system

The above diagram depicts the working of this system. As show above there is an arrangement which involves a Keyboard, a display panel, a counter and the lock management system. In this system a driver is subjected to a sequence of characters, after he inserts the key, in a particular jumbled manner. This sequence flashed on the display panel, usually LCD. The driver is supposed to enter the correct sequence in a stipulated time. When this sequence is entered correctly, the computer repeats the process two more time and finally the Ignition starts.

If at all, the sequence entered is incorrect, the car starts sounding an alarm which shows the probable car theft process. The disabling system then disables the ignition process and thus the car is prevented from driving.

The arrangement is such that each character is allowed to be entered only once while in a particular sequence, this avoids the mistaken entry.

Hence the EIL serves to be a handful system to ensure safe driving and safety from theft.

Q3. Explain the Dashboard Computer system in detail.

A.3 Dashboard Computer systems include the Instrument panel displays, in the form of instrument clusters act as information sources for the driver. These clusters usually display parameters like speed, rpm, fuel level, coolant temperature, distance travelled, trip meter and various alarms as individual displays.

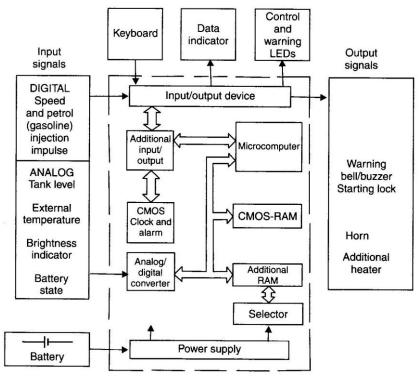


Fig. 49.7 Dashboard computer for motor vehicle

Newer car versions incorporate instrument clusters with microprocessors which can perform advanced functions. The processor carries out entire signal processing. Most of the processor based designs use application-specific standard processors (ASSP) or system-on-chip (SoC). In these ASSPs, all peripherals like digital and analog I/Os, LCD and stepper motor drivers, lamp drivers etc.; are integrated with the CPU core as a single IC.

EEPROMs are used to store information like the odometer data and other parameters for compensation of the temperature, fuel tank dimensions, etc.

Trip computers are usually present in the modern cars which are present as the separate panels. These are used to calculate data such as:

- Remaining fuel in the tank;
- The distance to be travelled (in case of the pre-determined distance entered);
- The average distance can be travelled with the remaining fuel etc.

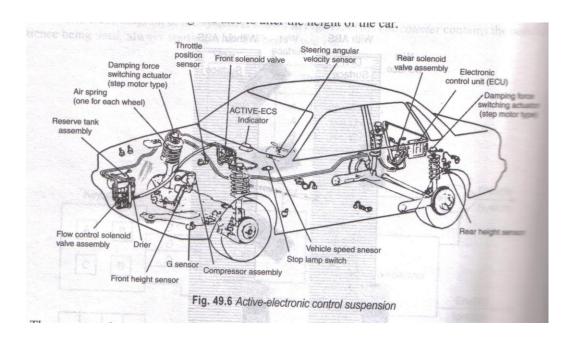
Different types of programs can be stored and can be used to derive many calculations out of the car such as:

- Speed of engine in RPM;
- Entertainment devices controls;
- Security systems monitoring;
- Monitoring of different types of sensors available in the car etc.

Hence a Dashboard Computer is a whole lot of handful device to make most out of the car using the electronics.

Q4. Explain Electronically Controlled Suspension (ECS) system in detail.

A.4



The name Electronically Controlled Suspension (ECS) signifies its operation. It has been designed to keep the car flat and levelled when riding over bumps, braking, accelerating and cornering. It achieves this with damper valves that can be independently addressed by the car's suspension computer to vary the damping effect, additional air springs for load sharing and also to alter the height of the car.

The processor has sensors which detect steering angle, gravity, road speed, the height of the vehicle at the front and rear, the accelerator's position and the rate of movement of various suspension components. There is some degree of driver over-ride over the sensors generally controlling the pump for the air springs and the damper valves.

Nose-dive under braking is not only uncomfortable, it is dangerous. By monitoring vehicle speed, longitudinal G forces and brake pedal application, the processor adjusts the

ride-height. The other case is that when the car is accelerating sharply, the car's rear suspensions are kept up and thus avoids the damping of rear side in case of front wheel drive. Also at the time of sliding down from the hill side, the car's front suspensions are lowered so as to maintain the car in a balanced horizontal position and give the level ride.

One of the most uncomfortable side-effects of a soft-riding suspension system is pitch and bounce. The car either bounces vertically over sympathetic bumps or pitches back and forth over bumps which are asynchronous with the car's suspension.

- Q5. Explain Car Safety Belt system.
- A.5 The Car Safety Belt system not only provides the safety of the driver, but also in some cases avoids the theft due to stringent norms. The figure below shows one of the Car Safety Belt systems:

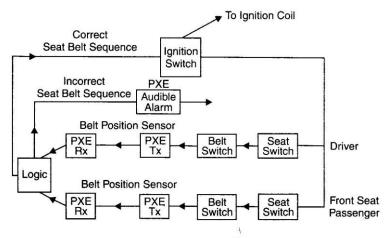


Fig. 49.9(b) Block diagram of the car safety belt system

As shown in the figure, there is a pressure switch which is placed under the seat of the driver which must be activated by sitting over it and to unlock the ignition system. Then the safety belt is to be tied in such a way that the ultrasonic sensors residing over it must be positioned correctly. There is a paired ultrasonic sensor at the windscreen which completes a circuit only when the belt is tied and also correctly positioned. Then the belt is to be buckled correctly to properly complete the sequence.

The 40 kHz signal is generated by the ultrasonic module residing over the belt. There is an audible alarm sounds whenever there is incorrect sequence is made while using the safety belt. The driver has to ensure the correct sequence again before starting the car engine. The system also holds equal when there is a front passenger riding in the front seat.

The system can be arranged so that if the belt is unfastened while the car is moving the ignition is not immediately affected. Instead the alarm is sounded and if, at the end of the specified time, the belt remains unfastened, the ignition will then be cut out. For very short operations such as parking or garaging the car, the logic arrangement can be adjusted to allow for car movement in first or reverse gear for a specified time without the driver being belted.

Q6. Explain Vehicle Proximity Detection system.

A.6 Over half-million automobile accidents each year results from improper overtaking.

This frightening statistics might be reduced substantially if this system is employed in each car or a vehicle moving on the road.

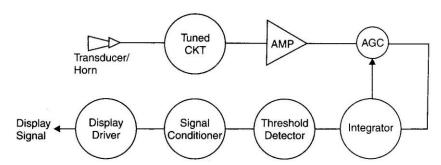


Fig. 49.13 In detection system, signals picked up by ultrasonic mike are used to activate a visual alarm on dashboard

The block diagram above depicts the working or this system. In operation, signals picked up by an ultrasonic transducer equipped with a horn are coupled through a tuned circuit to a high-gain, solid-state amplifier. An *AGC circuit* with a 20 dB dynamic range serves to suppress ambient highway noises, while a signal integrator and threshold detector, together, ensure a response only to target vehicles, rejecting shock and similar pulse-like signals. The signal is cleaned and fed to a solid-state lamp driver.

It was found that simple active systems could not discriminate between real target vehicles and such stationary objects as fences, sign-posts, tunnels, etc. Moreover it was also sensitive to rain, snow, dust, salt, shock, vibration and severe temperature changes.

In practice, the pick-up transducers may be mounted either in a special rear-view mirror package or within the vehicle's fender as part of the tail-and-turn-light assembly. The electronic control module may be placed wherever convenient.

Q7. Explain Car Navigation System.

A.7 The Car Navigation System is one of the revolutionary inventions of modern autotronics. This system locates, geographically, any vehicle in which it is mounted with the help of either satellites or any other concept such as GSM positioning.

Based on the technique employed in this system, the Car navigation system can be differentiated w.r.t. to the technique employed:

- 1. GPS based car navigation system;
- 2. GSM based car navigation system;
- 3. Combined GPS and GSM based;
- 4. Private network provided;

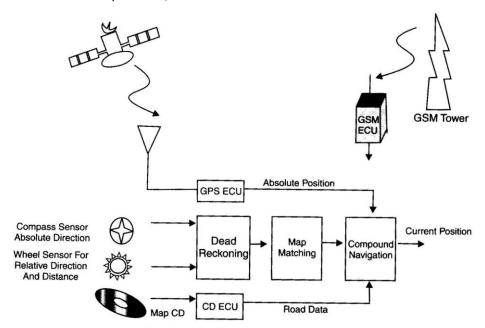


Fig. 49.16 Navigation system-block diagram

In GPS based car navigation system, the GPS module installed in the car acquires the location information from the satellite. The precision of the system depends upon the no. of satellites involved in the system. This system can also incorporate the maps of the local region which can be contained in a CD-ROM incorporated with the system. The location of the car is calculated with the help of triangulation method. This proves that the system needs more than two satellites for proper functioning.

In GSM based car navigation system, the GSM modem is used along with a Subscriber's Identity Module (SIM) which must be thoroughly registered to make use of the GSM services provided by any local service provider. The system works on the principle of Received Signal Strength concept. The system analysis the RSS level from the GSM modem and then decides the radius/periphery around which the vehicle may be present. Then it

checks for the RSS from surrounding antennae and hence it calculates out the probable location.

In combined GPS and GSM systems, the location is spotted with the help of both these techniques and hence this system is more accurate and faster. But both these system fails some time in some dense areas and hence there are compass and wheel sensors included in the car itself to guide the above two mechanism in case of environmental hindrances.

There are also some private companies who have their own technologies to fight with the navigation problems. These companies provide the system which works only in limited areas but are very efficient and accurate throughout those areas. Hence in all, the Car navigation system can help a driver to cruise through any unknown terrain to reach his destination already set. And these navigation systems help them to achieve their goal.

- Q8. Explain the Engine Management System.
- An Engine Management System is also called as an engine control unit (ECU), also known as power-train control module (PCM), or engine control module (ECM) is a type of electronic control unit that determines the amount of fuel, ignition timing and other parameters an internal combustion engine needs to keep running. It does this by reading values from multidimensional performance maps (so called LUTs), using input values (e.g. engine speed) calculated from signals coming from sensor devices monitoring the engine. Before ECU's, air/fuel mixture, ignition timing, and idle speed were directly controlled by mechanical and pneumatic sensors and actuators.
 - Control of fuel Mixture
 - Control of ignition timing
 - Control of idle speed
 - Control of variable valve timing
 - Electronic valve control

Programmable ECUs

A special category of ECUs are those which are programmable. These units do not have a fixed behaviour, but can be reprogrammed by the user.

Programmable ECUs are required where significant aftermarket modifications have been made to a vehicle's engine. Examples include adding or changing of a turbocharger,

adding or changing of an intercooler, changing of the exhaust system, and conversion to run on alternative fuel. As a consequence of these changes, the old ECU may not provide appropriate control for the new configuration. In these situations, a programmable ECU can be wired in. These can be programmed/mapped with a laptop connected using a serial or USB cable, while the engine is running.

The programmable ECU may control the amount of fuel to be injected into each cylinder. This varies depending on the engine's RPM and the position of the accelerator pedal (or the manifold air pressure).

Other parameters that are often mappable are:

- Ignition
- Rev. limit
- Water temperature correction
- Transient fuelling
- Low fuel pressure modifier
- Closed loop lambda

Some of the more advanced race ECUs includes functionality such as launch control, limiting the power of the engine in first gear to avoid burnouts. Other examples of advanced functions are:

- Wastegate control
- Banked injection
- Variable cam timing
- Gear control

A race ECU is often equipped with a data logger recording all sensors for later analysis using special software in a PC. This can be useful to track down engine stalls, misfires or other undesired behaviours during a race by downloading the log data and looking for anomalies after the event. The data logger usually has a capacity between 0.5 and 16megabytes. In order to communicate with the driver, a race ECU can often be connected to a "data stack", which a simple dash board is presenting the driver with the current RPM, speed and other basic engine data.