Audio visual covers a huge range of equipment and services. There are many common mistakes that are made in setting up a system as any professional AV integrator will confirm.

To narrow down the list to a manageable number, more information will be needed about the system being proposed, the location, the size of the room or rooms and the use to which it will be put.

The main things that need to be considered in setting up a system will include the following:

a. Sight lines and viewing angles
b. Positioning of speakers for effective stereo and surround sound handling
c. Control of equipment
d. Video and audio sources, both local and remote
e. Mechanical fixings and related concerns regarding room structure
f. Electrical supplies, cable routes for local and remote signals
g. System design to allow for expansion and future change
h. Commissioning and tuning of system to suit the users

There are other issues that need to be thought about and each one needs to be considered to avoid a poor performing system. A good system has always been designed with care and with experience. It is always a good idea to consult a professional who has the level of experience to avoid simple and common mistakes and can help to get the best out of any system.

We hear different sounds from different vibrating objects because of variations in:

- **Sound-wave frequency** - A higher wave frequency simply means that the air pressure fluctuates faster. We hear this as a higher pitch. When there are fewer fluctuations in a period of time, the pitch is lower.

- **Air-pressure level** - This is the wave's amplitude, which determines how loud the sound is. Sound waves with greater amplitudes move our ear drums more, and we register this sensation as a higher volume.

### A. AMPLIFIER

Generally, an amplifier or simply amp, is any device that changes, usually increases, the amplitude of a signal. The relationship of the input to the output of an amplifier—usually expressed as a function of the input frequency—is called the transfer function of the amplifier, and the magnitude of the transfer function is termed the gain.

As an amplifier's job is to take a weak audio signal and boost it to generate a signal that is powerful enough to drive a speaker. This is an accurate description when you consider the amplifier as a whole, but the process inside the amplifier is a little more complex.

In actuality, the amplifier generates a completely new output signal based on the input signal, these signals separate as two circuits. The output circuit is generated by the amplifier's power supply, which draws energy from a battery or power outlet. If the amplifier is powered by household alternating current, where the flow of charge changes directions, the power supply will convert it into direct current, where the charge always flows in the same direction. The power supply also...
smoothes out the current to generate an absolutely even, uninterrupted signal. The output circuit's load (the work it does) is moving the speaker cone.

The input circuit is the electrical audio signal recorded on tape or running in from a microphone. Its load is modifying the output circuit. It applies a varying resistance to the output circuit to re-create the voltage fluctuations of the original audio signal.

In most amplifiers, this load is too much work for the original audio signal. For this reason, the signal is first boosted by a pre-amplifier, which sends a stronger output signal to the power amplifier. The pre-amplifier works the same basic way as the amplifier: The input circuit applies varying resistance to an output circuit generated by the power supply. Some amplifier systems use several pre-amplifiers to gradually build up to a high-voltage output signal.

i. FIGURES OF MERIT

The quality of an amplifier can be characterized by a number of specifications, listed below.

a) Gain

The gain of an amplifier is the ratio of output to input power or amplitude, and is usually measured in decibels. Example: an audio amplifier with a gain given as 20 dB will have a voltage gain of ten (but a power gain of 100 would only occur in the unlikely event the input and output impedances were identical).

b) Bandwidth

The bandwidth of an amplifier is the range of frequencies for which the amplifier gives "satisfactory performance". The definition of "satisfactory performance" may be different for different applications. The gain of a good quality full-range audio amplifier will be essentially flat between 20 Hz to about 20 kHz (the range of normal human hearing).

c) Efficiency

Efficiency is a measure of how much of the power source is usefully applied to the amplifier's output.
d) Linearity

An ideal amplifier would be a totally linear device, but real amplifiers are only linear within limits. When the signal drive to the amplifier is increased, the output also increases until a point is reached where some part of the amplifier becomes saturated and cannot produce any more output; this is called clipping, and results in distortion. The problem of nonlinearity is most often solved with negative feedback. Linearization is an emergent field, and there are many techniques, such as feedforward, predistortion, post distortion, EER (Energy efficiency ratio) in order to avoid the undesired effects of the non-linearities.

e) Noise

This is a measure of how much noise is introduced in the amplification process. Noise is an undesirable but inevitable product of the electronic devices and components, also much noise results from intentional economies of manufacture and design time.

f) Video amplifiers

These deal with video signals and have varying bandwidths depending on whether the video signal is for SDTV, EDTV, HDTV 720p or 1080i/p etc., for acceptable TV images to be presented.

ii. Electronic Elements

The component at the heart of most amplifiers is the transistor. The main elements in a transistor are semiconductors, materials with varying ability to conduct electric current. Typically, a semiconductor is made of a poor conductor, such as silicon, that has had impurities (atoms of another material) added to it. The process of adding impurities is called doping. N-type semiconductors are characterized by extra electrons (which have a negative charge). P-type semiconductors have an abundance of extra holes (which have a positive charge).

An amplifier built around a basic bipolar-junction transistor. This sort of transistor consists of three semiconductor layers – in this case, a p-type semiconductor sandwiched between two n-type semiconductors. This structure is best represented as a bar, as shown in the diagram below (the actual design of modern transistors is a little different).
B. LOUDSPEAKER

A device that converts an electrical signal from an amplifier into sound. A loudspeaker driver is an electromechanical-acoustic device with two electrical input terminals, to which an electrical signal is applied, and a diaphragm which vibrates and radiates sound. An electromechanical motor mechanism exerts a force on the diaphragm to cause it to vibrate.

The two types of electrostatic transducers are used: the **piezoelectric transducer** and the **condenser transducer**. The piezoelectric transducer uses a piezoelectric crystal between the capacitor plates. The condenser transducer uses an air dielectric. One plate of the capacitor is a flexible membrane which serves as the diaphragm.

Diaphragm is alternately charged with a positive current and negative current, based on the varying electrical audio signal. When the diaphragm is positively charged, it fluctuates toward the front plate, and when it is negatively charged it fluctuates toward the rear plate. In this way, it precisely reproduces the recorded pattern of air fluctuations.
a) Driver types

Individual electrodynamic drivers provide optimal performance within a limited pitch range. Multiple drivers (e.g., subwoofers, woofers, mid-range drivers, and tweeters) are generally combined into a complete loudspeaker system to provide performance beyond that constraint.

i. Full-range drivers

A full-range driver is designed to have the widest frequency response possible. These drivers are small, typically 3 to 8 inches (7.6 to 20 cm) in diameter to permit reasonable high frequency response.

Full-range (or more accurately, wide-range) drivers are most commonly heard in public address systems, in televisions (although some models are suitable for hi-fi listening), small radios, intercoms, some computer speakers, many public address systems, etc. Full-range drivers often employ an additional cone called a whizzer: a small, light cone attached to the joint between the voice coil and the primary cone.

Limited-range drivers are typically found in computers, toys, and clock radios.

ii. Subwoofer

A subwoofer is a woofer driver used only for the lowest part of the audio spectrum: typically below 200 Hz for consumer systems, below 100 Hz for professional live sound, and below 80 Hz in THX-approved systems.

Many subwoofer systems include power amplifiers and electronic sub-filters, with additional controls relevant to low-frequency reproduction. These variants are known as "active subwoofers". In contrast, "passive" subwoofers require external amplification.

iii. Woofer

A woofer is a driver that reproduces low frequencies. The driver combines with the enclosure design to produce suitable low frequencies. Additionally, some loudspeakers use the woofer to handle middle frequencies, eliminating the mid-range driver.

iv. Mid-range driver

A mid-range speaker is a loudspeaker driver that reproduces middle frequencies. Mid-range driver diaphragms can be made of paper or composite materials, and can be direct radiation.

v. Tweeter

A tweeter is a high-frequency driver that reproduces the highest frequencies in a speaker system.
vi. **Coaxial drivers**

A coaxial driver is a loudspeaker driver with two or several combined concentric drivers. Coaxial drivers have been produced by many companies, such as Altec, Tannoy, Pioneer, KEF, BMS, Cabasse and Genelec.

b) **Loudspeaker System Design**

i. **Crossover**

![Crossover Diagram]

ii. **A Passive Crossover**

![Passive Crossover Diagram]

iii. **Enclosures**

An unusual three-way speaker system. The cabinet is narrow in order to reduce a diffraction effect called the "baffle step".

Most loudspeaker systems consist of drivers mounted in an enclosure, or Cabinet, to prevent sound waves emanating from the back of a driver from interfering destructively with those from the front.

Other enclosure types alter the rear sound radiation so it can add constructively to the output from the front of the cone. Designs that do this (including bass reflex, passive radiator, transmission line, etc.) are often used to extend the effective low-frequency response and increase low-frequency output of the driver.

iv. **Wiring Connections**

Two-way binding posts on a loudspeaker connected using banana plugs.
Most loudspeakers use two wiring points to connect to the source of the signal (for example, to the audio amplifier or receiver). This is usually done using binding posts or spring clips on the back of the enclosure. If the wires for the left and right speakers (in a stereo setup) are not connected "in phase" with each other (the + and − connections on the speaker and amplifier should be connected + to + and − to −), the loudspeakers will be out of polarity and may result destructive interference of the sound waves. This type of wiring error doesn't damage speakers, but isn't optimal.

v. Other Driver Designs
Other types of drivers which depart from the most commonly used direct radiating electrodynamic driver mounted in an enclosure include:

i. Horn loudspeakers
ii. Piezoelectric speakers
iii. Magnetostrictive speakers
iv. Electrostatic loudspeakers
v. Ribbon and planar magnetic loudspeakers
vi. Bending wave loudspeakers
vii. Flat panel loudspeakers
viii. Distributed mode loudspeakers
ix. Heil air motion transducers
x. Plasma arc speakers
xi. Digital speakers

C. MICROPHONES
Sound is an amazing thing. Microphones just convert a real sound wave into an electrical audio signal. They have a small, light material in them called the diaphragm. When the sound vibrations through the air reach the diaphragm, they cause the diaphragm to vibrate. This will cause an electrical current in the microphone to vary, it is sent out to mixer, preamplifier or amplifier for use.

All modern microphones are trying to accomplish the same thing as the original, but do it electronically rather than mechanically. A microphone wants to take varying pressure waves in the air and convert them into varying electrical signals.

Location of Microphone Diaphragm

a. Microphone Topology
There are many different types of microphones – dynamic, ribbon and condenser – each with unique characteristics. These names refer to the technology used to convert sound waves into electrical waves, some being more suitable than others for specific applications.
b. **Main Types of Microphones**
   
   i. **Carbon Microphones**
   
   The oldest and simplest microphone uses carbon dust. This is the technology used in the first [telephones](#) and is still used in some telephones today. The carbon dust has a thin metal or plastic diaphragm on one side. As sound waves hit the diaphragm, they compress the carbon dust, which changes its resistance. By running a current through the carbon, the changing resistance changes the amount of current that flows.

   ii. **Dynamic Microphones**
   
   A dynamic microphone takes advantage of [electromagnet](#) effects. When a magnet moves past a wire (or coil of wire), the magnet induces current to flow in the wire. In a dynamic microphone, the diaphragm moves either a magnet or a coil when sound waves hit the diaphragm, and the movement creates a small current.

   iii. **Ribbon Microphones**
   
   In a ribbon microphone, a thin ribbon is suspended in a magnetic field. Sound waves move the ribbon which changes the current flowing through it.

   iv. **Condenser Microphones**
   
   A condenser microphone is essentially a [capacitor](#), with one plate of the capacitor moving in response to sound waves. The movement changes the capacitance of the capacitor, and these changes are amplified to create a measurable signal. Condenser microphones usually need a small [battery](#) to provide a [voltage](#) across the capacitor. Other types of condenser microphones:
   - Large-Diaphragm Condenser Microphones
   - Phantom Power
   - Electret Microphones

   v. **Crystal Microphones**
   
   Certain crystals change their electrical properties as they change shape. By attaching a diaphragm to a crystal, the crystal will create a signal when sound waves hit the diaphragm.

   vi. **Wireless Microphones**
   
   Wireless microphones are one kind of specialty microphone. They contain an internal transmitter that sends signals over radio waves to a receiver.

   vii. **Lavalier Microphones**
   
   Lavalier microphones are another kind. They are usually wireless, and they are small microphones that can be clipped on a shirt.

   viii. **Bass Microphones**
   
   Bass microphones have a very large diaphragm that makes a very loud signal. They are usually used inside drums are rock concerts.

   ix. **Pressure Zone Microphones**
   
   Pressure zone microphones are a general purpose microphone that amplifies large sound sources like choirs or other large groups.
x. *Plaintalk* Microphones
This type of microphones use for Macintosh sound-in jacks. They have the same purpose and standard stereo mini-phone jacks.

c. *Pickup Pattern*
A pickup pattern is the way a microphone picks up a signal. It is based on what direction the sound is getting to the microphone.
- Omnidirectional is the kind of microphone picks up sound from all directions. These are used for group vocals and recordings.
- Unidirectional microphones pickup sound from only one direction. They are good for recording single voices. This makes them good for interviews in places that are loud. Because they can pickup from long distances, they are also great for surveillance.
- Bi-directional microphones get sound from two places. It is great for recording two voices at the same time.
- Carotid is the last type of pickup pattern. It is very unusual, because it picks up sound in a heart shaped pattern. These are actually a very commonly used microphone. They are great for talk shows, because the audience sound will not be picked up as much as the people on stage. This also makes it very good for live music performances.

a. **FEEDBACK CAUSE HOWLING SOUND**
A simple PA (public address) system consists of a microphone, an amplifier and one or more speakers. Whenever those three components connected, the potential for feedback will occur. Feedback occurs when the sound from the speakers makes it back into the microphone and is re-amplified and sent through the speakers again, like this:

![Feedback Diagram]

*Feedback occurs when the sound from the speakers makes it back into the microphone*

If you are setting up a sound system and want to avoid feedback, there are a few general rules that can help you avoid the problem:
- Make sure the speakers are in front of the microphone and pointing away from the microphone. If the speakers are behind the microphone, then feedback is nearly guaranteed.
- Use a unidirectional microphone.
- Place the microphone close to the person who is speaking/performing.
- If you have access to an equalizer, dampen the frequencies where feedback is occurring.
D. EQUALIZER

Equalizers can be used in many applications. In music and sound reproduction, equalizers can compensate for artifacts of the electrical-to-sound conversion or for unwanted characteristics of the acoustic environment such as sound reflections or absorption.

Equalization is also used to enhance the performance of systems that communicate or record digital signals (streams of bits). All communications and recording systems utilize a physical medium, such as wires; coaxial cables; radio, acoustic, or optical-fiber waveguides; or magnetic and optical recording media. These media cause distortion; that is, the output signal is different from the input signal.

Communications channel with an equalizer placed at the output. The equalizer recovers an accurate replica of the channel input signal.

i. Disadvantages of equalization

- Make a poorly designed sound reinforcement system work satisfactorily. Every well-designed sound reinforcement system is subject to the laws of physics described by the Potential Acoustic Gain equation.
- Improve intelligibility problems caused by reverberation, reflections, mechanical vibration, high background noise levels, or other problems caused by the location or physical design of the room. These problems are acoustical in nature and cannot be solved electronically. They must be resolved with acoustical solutions, such as sound absorbent panels and heavy drapes.
- Improve intelligibility problems caused by the talker being too far from the microphone.
- Improve the performance of sub-standard audio components in the sound reinforcement system.
- Eliminate distortion or noise problems caused by mismatched audio levels between system components.
- Improve echo return problems in teleconferencing systems.
ii. **Best Equalization Curve For A Speech Sound Reinforcement System**

This curve is based on human hearing and perception research.

![Equalization Curve](image)

iii. **Acoustic Feedback**

Acoustic feedback occurs when the amplified sound from any loudspeaker re-enters the sound system through any open microphone and is amplified again and again and again.

iv. **Feedback Problems Become Worse**

- Placing loudspeakers too close to microphones.
- Too many open microphones.
- Boosting tone controls indiscriminately.
- Room surfaces that are hard and reflective such as glass, marble, wood.

v. **Remedies On Feedback**

- Request that the talker speak louder into the microphone.
- Reduce the distance from the talker to the microphone. Each time this distance is halved, the sound system output will increase by 6dB.
- Reduce the number of open microphones. Each time this number is halved, the sound system output can be increased by 3dB.
- Move the loudspeaker farther away from the microphone. Each time this distance is doubled, the sound system output can be increased by 6dB.
- Move the loudspeaker closer to the listener. Each time this distance is halved, the sound system output will increase by 6dB.
- Use an equalizer/feedback reducer to cut the frequency bands in which the feedback occurs. The sound system output will typically increase 3 to 9dB.

**E. MIXER**

A device with two or more signal inputs and one common output and consists of two primary classes: linear (additive) and nonlinear (multiplicative) mixers.

Linear mixers are used to add or blend together two or more signals, nonlinear mixers mainly to shift the spectrum (center frequency) of one signal by the frequency of a second signal.
Linear mixing is the process of combining signals additively, such as the summing of audio signals in a recording studio. This operation can be accomplished passively by simply using a resistive summing network. Although this approach appears very economical, there is a loss in signal strength and an interaction of the signal amplitudes as the gains are adjusted.

A nonlinear mixer is in radio and television receivers. They are widely used in such applications as amplitude modulation (AM) and demodulation, frequency demodulation, phase detection, frequency multiplication, and single-sideband (SSB) generation.

The incoming information to a receiver has been transmitted and received at a frequency far too high to permit efficient amplification and processing. Therefore the signal is translated or frequency-shifted or heterodyned by a mixer to a lower frequency, known as the intermediate frequency (IF), where amplification and processing are performed efficiently by an IF processor, sometimes referred to as the IF strip. A second application of a nonlinear mixer is frequency synthesis, where a stable but not easily changed signal at a high frequency is made tunable by mixing it with an easily tunable signal at a low frequency, which, perhaps, can be varied in precise increments of any size. The utility of the method is limited by the ability to filter or separate one frequency term from another, thereby determining the minimum practical value of low frequency for the application.

A mixer is an integral part of an AM-radio integrated circuit which contains virtually all AM-radio functions except filters. A particular type of mixer, the quadrature detector, is included in the frequency-modulation (FM)-radio integrated circuit.

F. AV RECEIVER

AV receivers or audio-video receivers are one of the many consumer electronics components typically found within a home theatre system. Their primary purpose is to amplify sound from a multitude of possible audio sources as well as route video signals to your TV from various sources. The user may program and configure a unit to take inputs from devices such as DVD players, VCRs etc. and easily select which source he wants to route to his TV and have sound output for.

- **Usage**

  The term receiver originally referred to a component which included a tuner, a pre-amplifier and a power amplifier. These were generally called stereo receivers. The built in tuner in these devices gave them the name receivers.

  As home entertainment options expanded, so did the role of the receiver. The ability to handle a variety of digital audio signals was added. More amplifiers were added for surround sound playback. Video switching was added to simplify switching. Within the last few years, video processing has been added to many receivers.

  The term Audio Video Receiver (abbreviated AVR) or Home Theater Receiver is used to distinguish the simpler stereo receiver from the multi-channel audio video receiver.

**FEATURES**

- **Radio Reception**

  Receivers usually have a built in tuner for AM and FM radio reception. Satellite radio tuners are also found in many modern receivers, allowing reception with just an external antenna (and a satellite radio subscription, if necessary). Some models have HD Radio tuners.
Some models have Internet Radio and PC streaming access capabilities with an ethernet port.

b) Decoder
AV receivers usually provide one or more decoders for sources with more than two channels of audio information. This is most common with movie soundtracks. Movie soundtracks have been provided via a number of encoded formats. The first common format was Dolby Pro Logic. This format contained a center channel and surround channel. These channels were mixed into the left and right channels using a process called matrixing. Receivers were produced with Dolby Pro Logic decoders which could separate out these two additional channels. A somewhat less common surround sound decoder called DTS is standard on current AV receivers.

c) DSP effects
Most receivers’ offer specialized Digital Signal Processors (DSP) made for handling various presets and audio effects. Some may offer simple equalizers and balance adjustments to complex DSP audio field simulations such as "Hall", "Arena", "Opera", etc. that simulate the audio being played in the places through use of surround sound and echo effects.

d) Amplification
Stereo receivers have two channel of amplification, while AV receivers may have more than 2. The standard for AV receivers is five channels of amplification. These are usually referred to as 5.1 receivers. This provides for a left, right, center, left surround and right surround speaker to be powered by the receiver. 7.1 receivers are becoming more common and provide for two additional surround channels, left rear surround and right rear surround. The '.1' refers to the LFE (low frequency effects) channel the signal of which is usually sent to an amplified subwoofer unit.

e) AV Inputs / Outputs
There are a variety of possible connections on an AV receiver. Standard connectors include:

i. Analog audio (RCA Connector, or occasionally XLR connector)

ii. Digital audio (S/PDIF; TOSLINK or RCA terminated coaxial cable)
### iii. Composite video (RCA connector)

![Composite video connector](image)

### iv. S-Video

<table>
<thead>
<tr>
<th>Pin</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>GND</td>
</tr>
<tr>
<td>2</td>
<td>GND</td>
</tr>
<tr>
<td>3</td>
<td>Y</td>
</tr>
<tr>
<td>4</td>
<td>C</td>
</tr>
</tbody>
</table>

Looking at the female connector

### v. Component video

![Component video cable](image)

### vi. HDMI

![HDMI connection](image)

---

**G. CD: Material**

A CD can store up to 74 minutes of music, so the total amount of digital data that must be stored on a CD is:

\[
44,100 \text{ samples/channel/second} \times 2 \text{ bytes/sample} \times 2 \text{ channels} \times 74 \text{ minutes} \times 60 \text{ seconds/minute} = 783,216,000 \text{ bytes}
\]
To fit more than 783 megabytes (MB) onto a disc only 4.8 inches (12 cm) in diameter requires that the individual bytes be very small.

A CD is a fairly simple piece of plastic, about four one-hundredths (4/100) of an inch (1.2 mm) thick. Most of a CD consists of an injection-molded piece of clear polycarbonate plastic.

During manufacturing, this plastic is impressed with microscopic bumps arranged as a single, continuous, extremely long spiral track of data. Once the clear piece of polycarbonate is formed, a thin, reflective aluminum layer is sputtered onto the disc, covering the bumps. Then a thin acrylic layer is sprayed over the aluminum to protect it. The label is then printed onto the acrylic. A cross section of a complete CD looks like this:

CD: The Spiral
A CD has a single spiral track of data, circling from the inside of the disc to the outside. The fact that the spiral track starts at the center means that the CD can be smaller than 4.8 inches (12 cm).

CD is approximately 0.5 microns wide, with 1.6 microns separating one track from the next. (A micron is a millionth of a meter.)

CD: Bumps
The bumps that make up the track are each 0.5 microns wide, a minimum of 0.83 microns long and 125 nanometers high. (A nanometer is a billionth of a meter.) Looking through the polycarbonate layer at the bumps, they look something like this:

The incredibly small dimensions of the bumps make the spiral track on a CD extremely long. If the data can be lifted from the track and stretch it out into a straight line, it would be 0.5 microns wide and almost 3.5 miles (5 km) long.
**CD Player Components**

The CD player has the job of finding and reading the data stored as bumps on the CD. Considering how small the bumps are, the CD player is an exceptionally precise piece of equipment. The drive consists of three fundamental components:

- **Drive motor** – used to spins the disc
- **Laser and lens system** - used to focus in, on and read the bumps
- **Tracking Mechanism** – used to moves the laser assembly

**LCD projectors**

There is a wide variety of LCD projectors available. It can be LCD projectors on the basis of image clarity, resolution, zoom and many more. The audience size as well as the size of the venue where the presentation or event will be held is other factors, which decide the type of LCD projector you should go for. If you are unsure, the salespersons from the video equipment rental company can help you choose the right projector for your needs.

**Overhead projectors**

Overhead projectors are cheaper than LCD projectors and produce larger images on a white surface or any screen, allowing them to be viewed from long distances. Such projectors are basic but are reliable and used commonly. If you want to go for projectors that produce enlarged images, overhead projectors are the ideal options.

**TV and Video**

A very essential part of audio visual presentation equipment, television and video players come with a variety of features.

**Projection Screens**

Projection screens are used to display images from the projectors. They come in a variety of sizes. These screens are also available as tripods and wall screens. Larger screens are expensive and if you want such screens for one-time events, it is advisable to rent them.

**AV trolleys**

Audio visual trolleys are required to maneuver television, overhead projectors and other equipment.

**Computer based displays**

Computer-based displays look modern and professional. Images can be displayed with the help of a laptop or a desktop via a projector. You can also enhance such presentations with the help of videos and sounds.
References:

http://www.howstuffworks.com/amplifier1.htm


http://library.thinkquest.org/04oct/00451/microphones.htm

http://www.articlealley.com/article_1962646_10.html