Introduction to Electronics

A microphone is an electro-mechanical transducer that converts sound vibrations into an electrical signal that oscillates in proportion to the air pressure variations. Once the amplitude and frequency information content of the sound is transferred to an electrical signal, it can then be stored and manipulated as a sound recording. Most of the recording process involves the electrical analog of the original sound pressure waves since sounds are impossible to store directly, so an understanding of the function of electronic devices is fundamental to mastering the art of sound recording.

Even though most modern recording systems are digital, analog equipment is still required at the input and output of any recording system and much of the digital signal processing is based on models of the analog equivalents. Equipment including recorders, amplifiers, mixers, equalizers, compressors and other signal processing gear can be used to alter the electrical information in specific ways, giving an engineer the ability to dramatically alter the reproduction of the sounds that were recorded. While these devices may be complex, their operation is based on a few elementary concepts of analog electronics. Analog electronics refers to a system where the electrical signal amplitude varies continuously in direct proportion to the intensity and frequency of the mechanical vibrations that were transduced. Until recently, this was the only method available for processing audio data. Digital electronics now allow discrete measurements of the transduced signal to be created, stored, and manipulated by computers at a very high rate of speed, producing a fast but non-continuous numerical representation of the original sound. We will begin by exploring the more simple concepts of analog electronics and then extend these to the digital realm.

Electricity is the movement or separation of charge (negatively-charged electrons in most cases). Charge is a fundamental property of matter. Electrons are negatively charged and protons are positively charged. When atoms have an equal number of electrons and protons, they have no net charge. When there are more or fewer electrons than protons, the atom is said to be charged or ionized. Like charges repel each other while opposite charges attract. The forces of repulsion and attraction allow electricity to do work. The flow of charge can be through a resistive medium like air (lightning), or through a solid conductor like a wire. The path through which the charges move is called a **circuit**; to be useful, the charges need to move from the source (battery) through a loop (circuit) back to the source. **DC** (direct current) flows in only one direction, while **AC** (alternating current) flows back and forth in both directions. The rate at which an AC signal changes direction (once per cycle) determines its frequency, the rate at which a sinusoidal oscillation repeats.

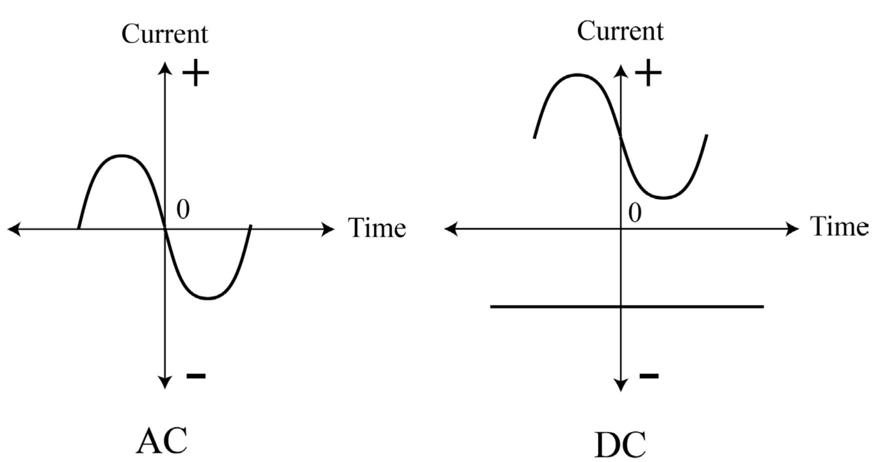


Figure 1: AC and DC electricity. AC flows in both direction. DC always flows in one direction, but it can vary in amplitude.

Some simple electrical principles include:

Voltage (V): electromotive force (EMF) pushing charges through a circuit (analogous to pressure in a liquid flow system). Audio signals are usually time-varying voltage signals.

Current (I): flow of charge through a circuit. (Amperes = coulombs/sec; the coulomb is a measure of electric charge: 1 coulomb= 6.414×10^{18} electrons).

Impedance (Z): opposition to flow of current (somewhat (inversely) analogous to the diameter of a pipe in a fluid flow system), measured in ohms (resistance (R), reactance (X): while resistance does not depend on the frequency of the signal, reactance does).

The term **signal** denotes a time-varying voltage or current that encodes information: a voltage or current that varies in proportion to a transduced air pressure, for example.

While we normally regard a wire simply as a conduit for electrical current flow, there is a very important phenomenon generated by current flow in a wire: namely the creation of a **magnetic field** that varies with the changing flow of current. Any time current flows, it sets up a magnetic field, and any time a wire moves through a magnetic field, a current flow is induced in the wire. This basic physical phenomenon provides the basis for many steps in the recording process. It also creates some basic problems by providing unwanted coupling of signals in some situations.

The simplest electronic components are the passive devices: resistors, capacitors, and inductors. Passive means they do not require external power to function, only the power contained in a signal itself. (Active components like transistors, vacuum tubes, and integrated circuits require external power as well as the energy of the signal itself.) Electronic devices are described by their current-voltage (I-V) relationships: as we vary the current through the device, what does the voltage across them do? (See Figure 2) (Although these circuit elements all exhibit slight deviations from the "ideal" behavior, the differences are usually small enough to ignore to a first approximation. These small deviations sometimes do contribute to a device's perceived sound, though.)

Passive devices are only capable of opposing the flow of current and therefore cannot amplify signals. The opposition to current flow occurs in two different forms: resistors dissipate power as heat while "ideal" capacitors and inductors temporarily store energy that can be returned to a circuit and do not dissipate power. The opposition to current flow in resistors is termed resistance while that of capacitors and inductors is called reactance, but both forms are included in the term impedance. Impedances are measured in ohms whether we are referring to resistance or reactance.

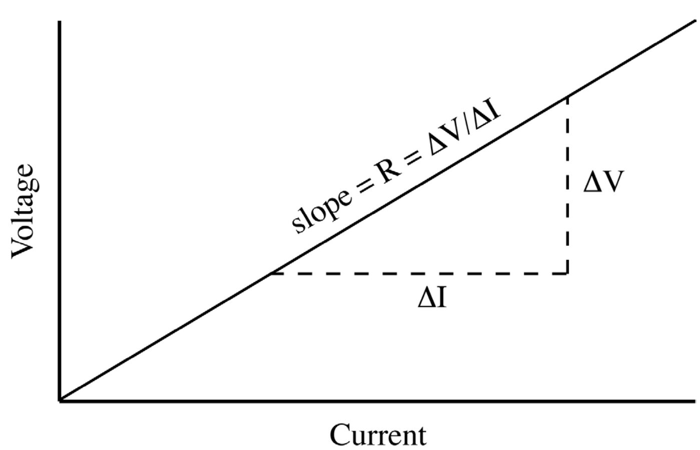
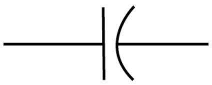


Figure 2: Current-voltage plot for a resistor. The slope of the line gives the resistance.

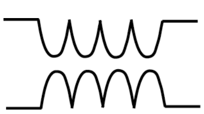
The passive electronic devices

Resistors: V = I x R (resistance (R) does not depend on time or frequency) Resistors are passive devices which have a constant impedance regardless of the frequency of the current flowing. They can be conceptualized as the diameter of a hose: they oppose the flow of current by an amount directly related to the resistance (like a constriction in a hose opposes the flow of liquid). Resistors resist current flow by dissipating energy as heat. In a resistor the current and voltage are in phase.



Capacitors: $V = \int I\, dt/C$ (or $I = C\, dv/dt$) : $X_c = 1/(2\pi fC)$ (reactance (X_c, X_L) involves the storage of energy and therefore does depend on time and/or frequency). A capacitor is simply two charged plates placed close together with a dielectric (non-conducting) material sandwiched between the plates. When a charge is applied to one plate, it repels like charges on the opposite plate until an equilibrium is established. For direct current, the capacitor charges up with a time constant that depends on the capacitance value and the impedance through which the current flows into the capacitor, which limits the available current. Once the capacitor is fully charged, no more current flows. This means that the capacitor is an effective block for direct current. For alternating current (like audio signals), the response is more complicated. The charge that develops on the capacitor depends on how fast the current is changing. It takes time for the charge to build up, and that time results in a frequency dependent delay (or phase shift) in the output signal. In series, a capacitor acts somewhat like a rubber diaphragm in a hose. Capacitors store energy in the electric field created by the separated charges on the plates. In capacitors, the current leads the voltage by 90^o: the current creates the charge accumulation that generates the electric field.

Inductors: $V = L\, dI/dt$: $X_L = 2\pi fL$ An inductor is simply a coil of wire, which can be wrapped around either air or metal cores. As current flows into an inductor, a magnetic field is created around the coil. When the current stops, the magnetic field collapses, generating an induced current flow in the coil. Low frequency currents flow easily into the inductor, but as the alternating current frequency increases the impedance of the inductor increases. Like the capacitor, the inductor introduces a phase shift. Inductors allow direct current to flow, but as the frequency of oscillation increases, so does the inductor's impedance. An inductor is conceptually similar to a water tank (resonant circuits with inductors are sometimes referred to as "tank" circuits). Inductors store energy in the magnetic field that builds around the device when current flows. In an inductor the voltage leads the current by 90^o: a rapid voltage change creates a current that grows slowly as it generates the magnetic field.



Transformers: (M = mutual inductance) $V_{out} = L\, dI/dt + M\, dI/dt$. A transformer consists of two separate coils with overlapping magnetic fields, so that current flowing in one circuit is inductively coupled to the other. Often, transformers consist of an iron core wrapped with two or more coils, which couple magnetically. Transformers are used to get voltage gain (at the expense of current reduction) and to step down power line voltages for power supplies. Transformers are also used to match impedances between devices and to provide ground isolation. Although transformers are bi-directional, the side that is actively driven is called the primary while the other side is the secondary.

Ohm's Law: Ohm's law is one of the simplest, yet most important principles of electronics. It states that:

$$V = I \times R$$

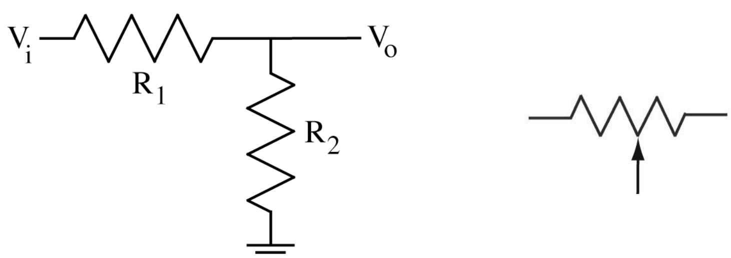
The voltage drop across a resistor is the current multiplied by the resistance. This holds true for the impedance of inductors and capacitors as well, if we take into account their frequency-dependent nature. **Power**, the amount of work done per unit time in a circuit is given by:

$$P \text{ (power)} = I \times V = V^2 / R = I^2 \times R$$

Where Ohm's law has been substituted for either I or V.

Analog circuits

Although most electronic devices are full of active components like op-amps and transistors, much of the actual "work" is done by simple arrangements of the passive elements: resistors, capacitors, and inductors. Understanding the simple circuits will allow an engineer to examine the schematic diagram of a new device and immediately gain knowledge about the operation of the device. It also makes troubleshooting possible. By examining some simple combinations of resistors and capacitors, we will see how equalizers function.



potentiometer

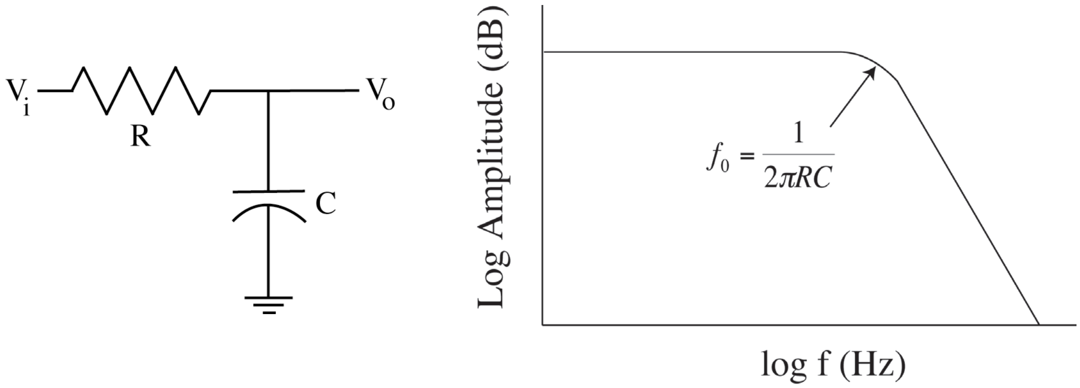
Voltage divider

Voltage Divider: The simplest functional circuit, but one of the most important, is the voltage divider:

$$V_{out}=V_{in}(R_2/(R_1+R_2))$$

The voltage divider is so-called because the input voltage divides linearly in proportion to the resistances of the circuit. If we measure the voltage drop across R_2, it is the input voltage times the ratio of R_2 to the total resistance of the circuit, R_1+R_2. A potentiometer or "volume control" is simply a continuously variable voltage divider, as shown above. By using capacitors and inductors along with resistors, the voltage divider can function as a frequency-selective circuit called a filter.

R-C circuits: Most filters in use today rely on resistors and capacitors. Inductors are less often used because they are large, expensive, hard to get large values, and susceptible to electromagnetic interference. Still, many classic equalizers were based on inductors and are considered desirable for the particular sound they produce. Inductance is sometimes simulated using capacitors and op-amps. The voltage divider circuit, when partly composed of frequency-selective passive elements (capacitors or inductors) will act as a filter. The frequency above (or below) which attenuation occurs depends on the value of resistance and capacitance (or inductance).



Low-pass filter

Above is a low-pass filter: low frequency signals pass through unattenuated. As the signal frequency increases, the capacitive reactance decreases. At the frequency at which the capacitive reactance just equals the resistance, the output signal is reduced by 1/√2 (-3dB). This is known as the corner frequency of the filter and is determined by:

$$f_0=1/(2\pi RC)$$

[It might appear that the amplitude of the output signal at the corner frequency should be half and therefore the output should be down by -6 dB *, but there is another consideration: the capacitor has an effect on the phase of the signal as well as its amplitude. As the frequency of the signal increases, the time response of the capacitor begins to shift the phase of a sine wave signal as it flows through the capacitor. We must use a vector description of the circuit, one that involves imaginary numbers to deal both with the amplitude and phase of the signal. When the vector description of the impedances are used, the magnitude part of the vector sum is:

$$Z_{total} = \sqrt{(R^2 + X_C^2)}$$

(* See discussion of dB below.)

So, substituting in the voltage divider equation, we get:

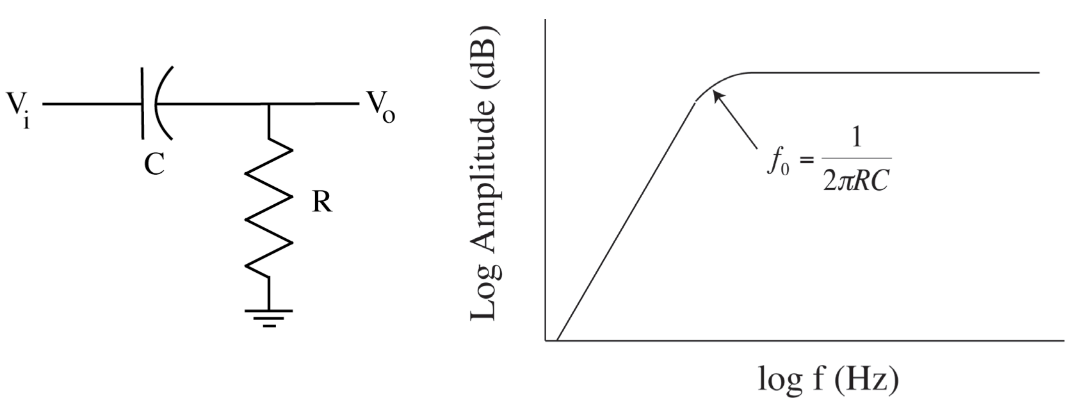
$$V_{out} = \left(\frac{R}{\sqrt{\left(R^2 + X_C^2\right)}} \right) V_{in}$$

Rearranging:

$$\frac{V_{out}}{V_{in}} = \frac{R}{\sqrt{(R^2 + X_C^2)}}$$

So at the frequency where X_c=R, the gain (V_{out}/V_{in}) is 1/√2 or -3dB.]

A similar analysis can be applied to the high-pass filter, shown below. Here, the capacitive reactance is high for low frequencies, but decreases as the frequency increases. This configuration is often used in audio input circuits to block unwanted DC components, which could overload amplifiers if not removed.



High-pass filter

By rearranging the components, we can make a hi-pass filter. In this case, the capacitor effectively blocks

DC and low frequency components. As the frequency of the signal increases, the impedance of the capacitor decreases and more of the signal is delivered to the output. Inductors can also be used to create filters. Since their impedance versus frequency behavior is the opposite of that of capacitors, substituting an inductor for a capacitor in the above circuits produces the opposite kind of filter.

Frequently, an intuitive understanding of the operation of circuit elements is as helpful as a complete engineering analysis. Most circuits can be understood on a superficial basis, since one is not trying to design a circuit, but simple appreciate what a circuit is doing to the signal in a general way: i.e. boosting high frequencies, mixing signals, buffering/impedance matching, etc.

Often, a circuit can be well understood just using Ohm's law and the R-L-C circuit theory introduced above. We, as device users, need not be concerned as much with engineering precision as with general circuit function.

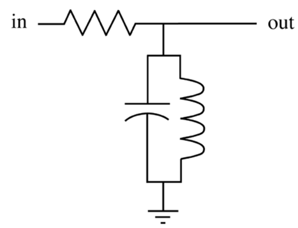
To recap, as far as capacitors and inductors are concerned, capacitors have high impedance for low frequencies and inductors have high impedance for high frequencies. Exactly what constitutes high and low frequency depends on the relative values of the other circuit elements' impedances (resistors and other capacitors and inductors.)

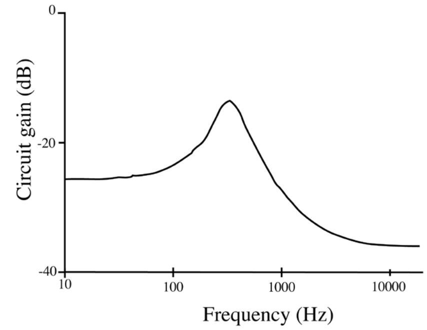
When inductors and capacitors are combined in a circuit, the interchange of energy back and forth between the two devices can lead to interesting behavior: **resonance**. At the frequency that results in the devices exhibiting equal reactance, the circuit displays extremes of impedance. If the devices are in series, the total impedance drops to zero. If the devices are in parallel, the total impedance increases to infinite. This behavior is caused as the energy is transfered back and forth between the devices. The magnetic field of the inductor collapses, inducing current flow that charges the capacitor. Then the electric field of the capacitor discharges and causes current flow that builds a magnetic field in the inductor. The back-and-forth continues until stray resistances dissipate the energy: in fact if there were no resistance and energy continued to enter the system, the circuit might burst into flames! An interesting consequence of resonance is that the resonant voltage swing can be larger than the original exciting signal voltage at the resonant frequency. This is not exactly amplification as it can only occur at the exact resonant frequency.

Resonant behavior occurs in mechanical and acoustical systems as well as in electric L-C circuits. A swinging pendulum resonates at a characteristic frequency as the **kinetic energy** (swinging mass) exchanges with **potential energy** (mass raised in height.) A weight on a spring behaves similarly. Sound waves resonate in a room as the compression of the air exchanges with deformation of structures that then vibrate and return energy to the air. It is the back-and-forth exchange of energy between the two forms that allows resonance in all these examples. In electric circuits, the energy is in fact not kinetic energy in the classical sense because no mass movement is involved, but rather the movement of an electric field. This distinction may be confusing since electric circuits exhibit behavior that looks a lot like the interchange of kinetic and potential energy in mechanical systems. The interchange between energy fields generated in inductors (magnetic) and capacitors (electric) allows resonance that is described by equations that are similar to those used to describe mechanically resonant systems.

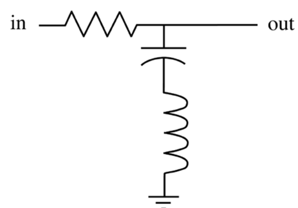
Resonant circuits

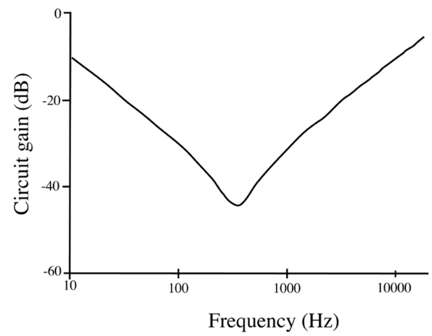
parallel
resonance





series
resonance

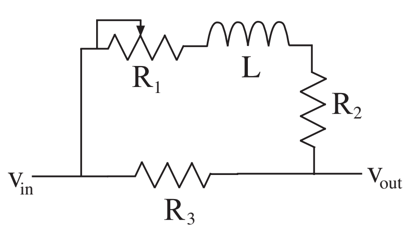




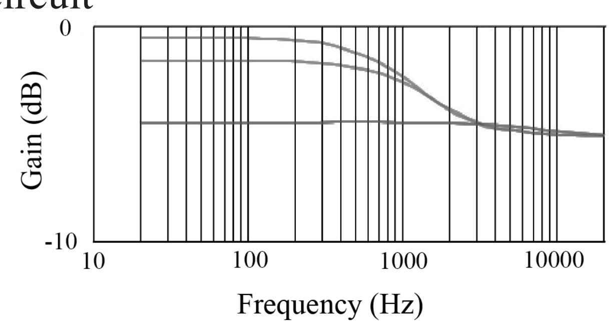
In the case of parallel resonance, the impedance of the L-C circuit is low at extremes of frequency since the inductor behaves as a wire at low frequencies and the capacitor operates like a short circuit at high frequencies. In the mid-frequency region where the L and C impedances are equal, the resonant part of the circuit exhibits a high impedance.

For the series resonant circuit, the capacitor is a virtual open circuit at low frequencies and the inductive reactance is high at high frequencies. In the mid-frequency range, the series L-C circuit exhibits a low impedance, shunting the signal to ground.

Analyzing a circuit



Low frequency boost
shelving filter
$R_1 > R_3$, $R_2 < R_3$
Load R = R_3



The circuit above is a low frequency shelving filter. The circuit can be analyzed using what we've learned about circuit elements and simple parallel (||) and series connections. The only frequency sensitive element is the inductor, which has low impedance at low frequencies. At low frequencies, X_L is low and the total circuit impedance is determined by how much the R_1/L/R_2 branch shunts (bypasses) R_3. Since R_1 is adjustable, it will determine how much the parallel pathway shunts R_3 with R_2 setting the minimum parallel impedance. At higher frequencies, X_L is very high and the total circuit impedance is determined by R_3 alone. This gives us a useful filter: a low shelf where every frequency below a corner frequency is increased an adjustable amount relative to the higher frequencies. It should be remembered, however, that no gain is available in such a circuit and we are actually attenuating the "boosted" frequencies less. This circuit would be followed by an amplifier to restore the gain lost in the passive network. Without considering the values of the components we cannot tell exactly where the line between low and high frequencies will occur, but we can determine the general function of the circuit by inspection, keeping in mind the basic operation of the passive electronic elements.

Audio Signals

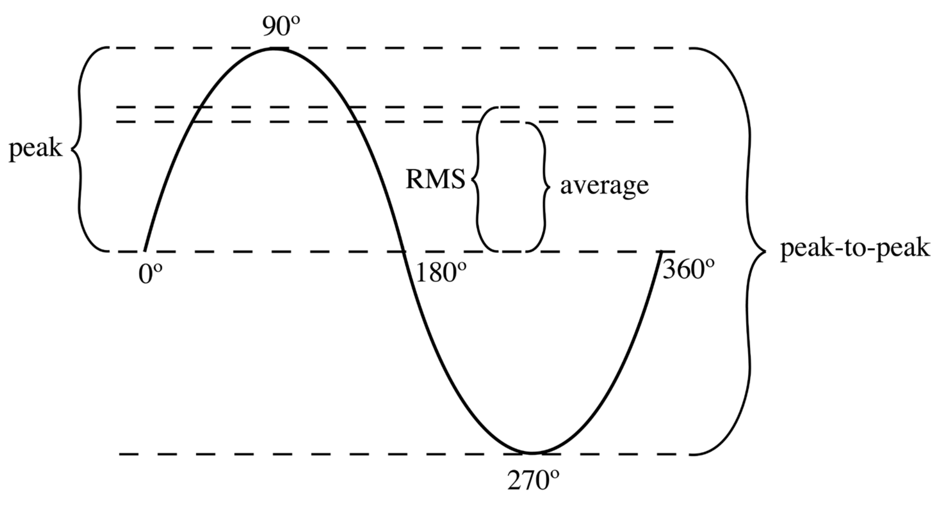
Audio signals are voltages (and/or currents) in electronic circuits that vary in time directly in proportion to a transduced sound pressure. Waves of air pressure are converted by a microphone or other transducer into electrical signals that can be manipulated and stored. In order to deal with these signals, we need some way of describing and measuring them. Most people are familiar with the **VU meter**, the indicator of signal amplitude so commonly found on audio equipment. The meter tells the observer how large the signal is relative to some reference level, thereby conveying information about the loudness of the sound that generated the electronic signal. But audio signals also convey information about pitch by varying the frequency of the electrical signal oscillation, and this is not directly reflected in the readings of a VU meter. There are many aspects of audio signal measurement that must be understood if one is to fully comprehend the operation of audio circuitry.

Of course, the obvious way of monitoring signals is to listen to them. But consider a modern console with 48 inputs and 24 outputs: it is simply not possible to listen to every individual signal at once, so other methods of signal monitoring have been devised to augment the auditory monitoring system. Since the main characteristics of signals are their amplitude and frequency, standard methods of describing these aspects have been adopted. Amplitude measurements are made in **decibels** (dB), a logarithmic measure of signal magnitude that describes the amplitude of a signal relative to a standard reference amplitude. Frequency measurements are more complicated, since most signals are composed of many frequency components added together. Further complicating things, most amplitude measurements assume a perfect sine-wave signal, which seldom occurs in real musical sounds. In order to understand the standard audio signal measurements, we will first examine sine waves, the simplest type of signal and the one commonly used for test and measurement.

Frequency measurement: The sine wave is an alternating voltage or current which swings symmetrically about 0 volts (or amps) or some other fixed DC value. One cycle consists of a positive and negative swing. The frequency is the number of such complete cycles per second. The measure of frequency is the Hertz (Hz), which is the same as cycles/second. The equation for the sine wave is:

$V=V_{max}\sin(x)$, where V_{max} is the maximum amplitude and x varies from 0° to 360° (0 to 2π radians).

The sine function is shown below, along with several of the conventional amplitude measurements.



(These relationships exist for pure sine waves only.)

RMS = 0.707 x Peak Voltage	Peak = 1.414 x RMS Voltage	Average = 0.637 x Peak Voltage
RMS = 1.11 x Average Voltage	Peak = 1.57 x Average Voltage	Average = 0.9 x RMS Voltage
RMS = 0.3535 x Peak-to-Peak Voltage	Peak-to-Peak = 2.828 x RMS Voltage	

In digital audio systems, frequency is often displayed in the form of a spectrum analysis, which plots amplitude versus frequency. This type of display can be helpful when it is available in conjunction with an equalizer, allowing the engineer to see where signal spectral components lie and raise or lower them. While this is tempting, it is eventually necessary to do this by ear and with experience, one will use the specrum display less often if at all.

Amplitude measurement:

The table below gives an idea of the range of loudness measurements encountered in everyday sounds.

Power (Watts)	Power Level (db) Re:10^{-12} Watt	Source
20-40 million	195	Saturn rocket
100,000	170	Jet afterburner
10,000	160	Jet engine
1000	150	
100	140	Propeller airplane
10	130	Orchestra, rock band
1	120	Small plane
0.1	110	Blaring radio
0.01	100	Auto on highway
0.001	90	Voice shouting
0.000 1	80	
0.000 01	70	Average conversation
0.000 001	60	
0.000 000 1	50	
0.000 000 01	40	Average house
0.000 000 001	30	Very soft whisper

The range of loudness perceived by the ear varies over 6 orders of magnitude, making linear measurements of loudness difficult. Therefore, a logarithmic scale of loudness has been adopted: the decibel (dB).

dB (power) = $10 \log(P_1/P_2)$, where P_1 is signal power measured in watts, P_2 is the reference power level

By substituting using Ohm's law V=IR, power = $V \bullet I = V^2/R = I^2R$;

$$10 \log([V_1^2/R]/[V_2^2/R]) = 10 \log(V_1^2/V_2^2) = 20 \log(V_1/V_2)$$

$$\text{dB (voltage, current, SPL)} = 20 \log(V_1/V_2) = 20 \log(V_{meas}/V_{ref})$$

Since these measures are a ratio of two values, standard (reference) values are used for comparison:

0 dBm———————1 milliwatt (power, not voltage measurement)
0 dBV———————1 Volt (dBA is similar only frequency-weighted to hearing)
0 dBu or (0dBv)——.775 Volts (= 1 mW in 600 ohm circuit)
0 VU ——————— variable reference, nominal operating level

Noise: Noise is often used as a convenient test signal since it contains all frequencies at once. Noise can be generated to contain different balances of frequencies based on the energies of the individual sinusoidal frequency components. When each cycle contains the same amount of energy, it is known as white noise. Thermal noise, also known as Johnson noise, is an example of white noise. When each octave contains the same energy, it is called pink noise. Pink noise is also known as 1/f noise, since the energy/Hz decreases with increasing frequency. Pink noise amplitude decreases 3 dB per octave or 10 dB per decade, thus it decreases 30 dB over the three-decade audible range of frequencies. White noise is audible much brighter sounding than pink noise. Pink noise is most commonly used in audio and acoustic measurements as it more closely relates to how the auditory system perceives loudness. Figure 3 shows a comparison of the amplitude spectra of white and pink noise.

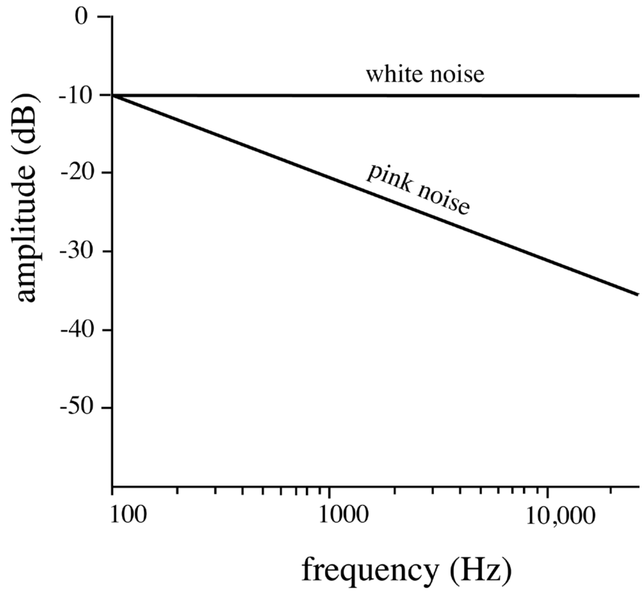


Figure 3: White and pink noise spectra.

Distortion: Distortion is any unwanted change in an audio signal. The most frequent types of distortion come from overloading circuits: that is, driving a circuit with a voltage too large for the power supply voltage of the circuit. When most solid-state circuits overload, they do so by voltage limiting and generate odd-order harmonics that sound harsh. Vacuum-tube circuits distort from current limiting instead of voltage limiting, generating even-order harmonics that have a warmer sound. Neither type of distortion is desirable in a high-quality audio circuit. Distortion may also be generated in transformers and by feedback circuitry in amplifiers. Measurements of device distortion are usually given as THD (either third harmonic distortion or total harmonic distortion), which gives a measure of the harmonic content of a signal added by a circuit or device. A newer measure is IM (inter-modulation) distortion, in which two non-harmonically related signals are mixed and the distortion products measured. This gives a picture of the dynamic distortion performance of a circuit.

Dynamic Range: Dynamic range is a measure of the ratio of the largest signal a circuit is capable of handling to the highest amplitude individual frequency component present in the system noise. Dynamic range may refer to a system with no signal present.

Signal-to-Noise ratio: The signal-to-noise ratio (SNR) is a measure of the signal level versus the amount of noise in a circuit. The higher the ratio, the better a piece of gear performs.

$$\text{SNR (dB)} = 20 \log (V_{signal}/V_{noise})$$

The signal measured is the nominal or average voltage level for the circuit. Since some amount of noise is present in every electronic circuit, it is important to amplify signals as much as possible without introducing overload distortion to keep the level as far as possible above the noise.

The distinction between dynamic range and signal-to-noise ratio is potentially confusing and the terms are sometimes used interchangeably. SNR refers the “expected” signal level (the nominal or average level, as 0 VU level) to the average level of the circuit’s inherent noise. (This will not include the headroom, the amplitude region above the nominal level that allows peaks to pass undistorted.) Dynamic range refers the maximum signal level possible to the highest amplitude component frequency of the system noise. The issue of how small a signal we can usefully reproduce is not clearly defined, as a very small signal may be perceived although swamped by the system noise. In this case, the dynamic range will be large, but the SNR for low-level signals will be quite low.

Gain structure: Describes the relative signal amplitude at various stages of a device’s circuitry. The amount of additional gain that may be applied before a signal reaches the voltage limit of the system is called headroom. Throughout a circuit, the amount of headroom may vary and the signal needs to stay below the voltage limit at every stage of circuitry. For instance, an equalizer in a mixer can raise the signal level significantly and may eventually cause overload distortion if the input signal to that circuit stage is too high. It is possible in a complex mixer to distort somewhere in the circuit without it being reflected in a meter reading, so understanding gain structure as part of signal flow is important to maintaining high quality audio. A gain structure problem might be using too little gain in the input stage and adding a lot of gain in subsequent stages, increasing noise in the process. The signal-to-noise ratio of the input signal is decreased by using insufficient gain at the input and later adding gain to restore the original level also amplifies the higher noise floor from the input stage.

Analog versus digital signal metering

The popularity of digital audio equipment has changed the way we think about signals. While continuous analog signals are simply time-varying voltages, digital signals are streams of numbers, each coding a voltage measured at a specific time. While both contain the same information, they behave differently. Amplitude measurements are quantized in digital systems, meaning that only specific amplitudes may be encoded. (*This does not mean the output signal can only produce the sampled values, a digital filter fills in the values between samples when the digital signal is converted back to analog*.) Further, the maximum is an absolute limit, while analog signals can continue to increase, although severely affected by the voltage (or current) limits of the circuits. The digital signal is not altered until it exceeds the limit, when it is completely corrupted. The analog signal begins to be altered as it approaches the limits of the circuit and becomes progressively more distorted.

The VU meter is calibrated in terms of its time response to simulate the auditory system’s perception of loudness. It displays the RMS signal level, so peaks may not be reflected in the VU reading and clipping is possible. To address that problem, many VU meters also have an LED to show when the level is approaching the limit. These LEDs usually flash when the signal is a few dB below the limit, but how close to that limit isn’t usually specified. In general use, these lights may flash occasionally but should not remain on or flash very frequently to avoid distortion.

The metering of digital systems has no headroom above 0 dBFS whereas analog systems have up to 24 dB available above 0 VU. This leads to confusion. The usual standard operating level is considered to be 0 VU in analog metering. In digital metering, 0 dBFS is the absolute maximum: therefore the standard operating level in digital metering is usually –12 dB FS (full scale), which allows peaks of up to 12 dB above the usual level before distortion. Some people allow more and some less, but there must be an allowance for peaks with digital metering. Figure 4 shows the two graphic display systems compared.

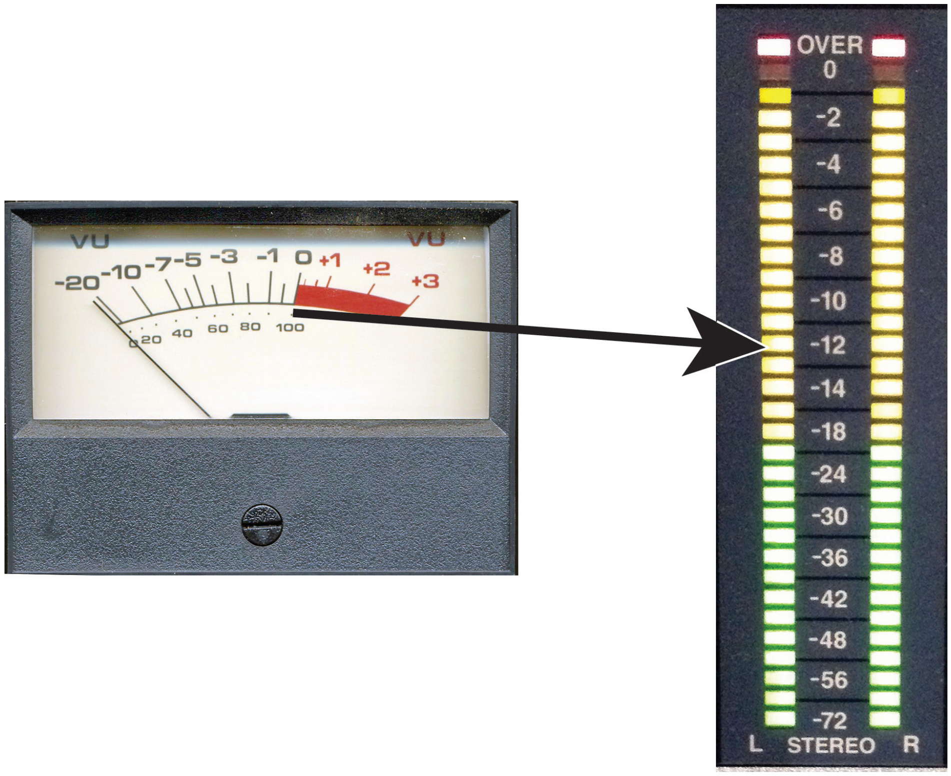


Figure 4: Analog VU and digital metering compared.

Many digital meters display both the average or RMS level and the peak level simultaneously. This dual display can be useful in determining how much dynamic range remains after mastering compression and limiting, for example. While it may be tempting to rely on meter readings, the ultimate decision about levels is determined by the ear.