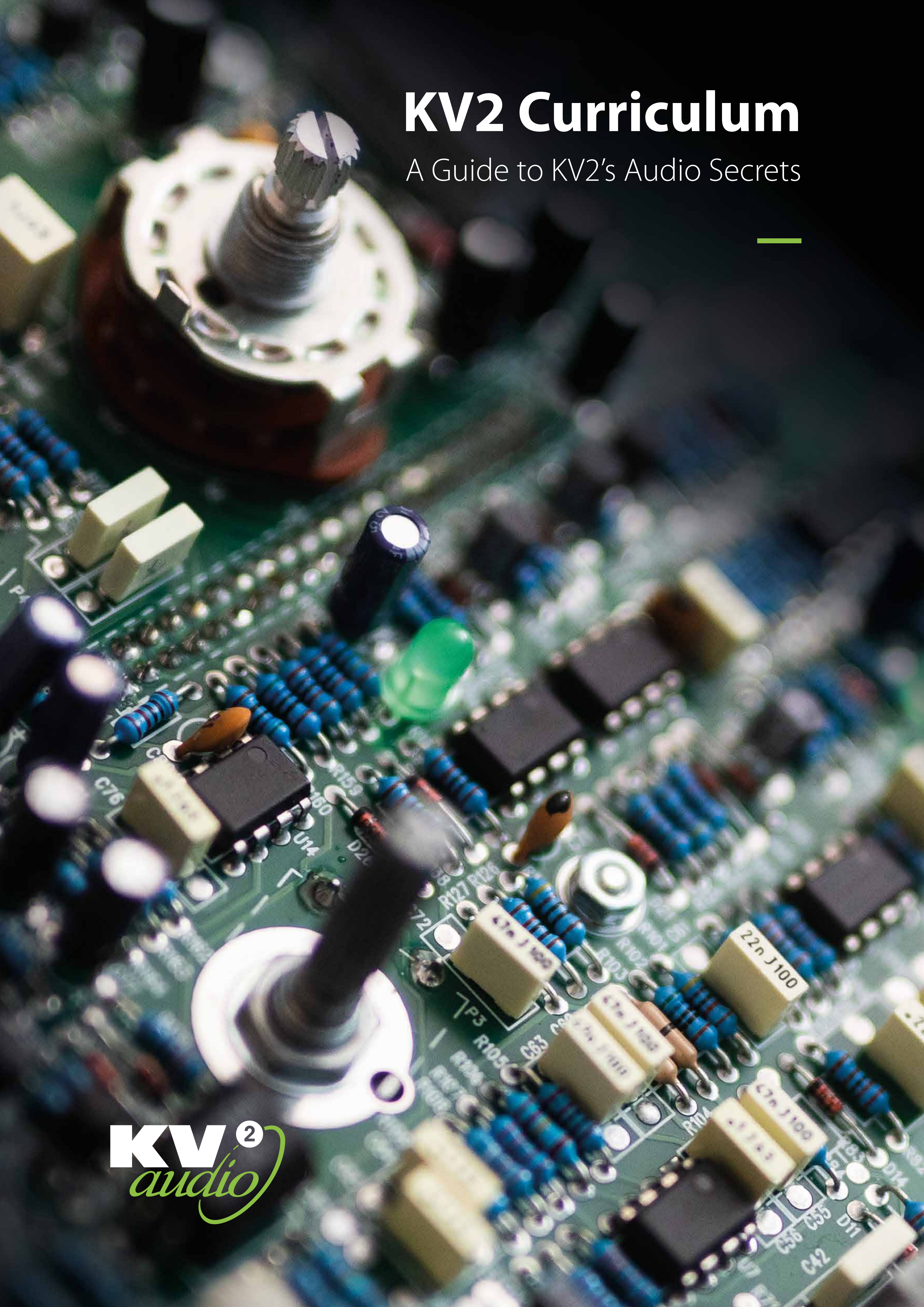
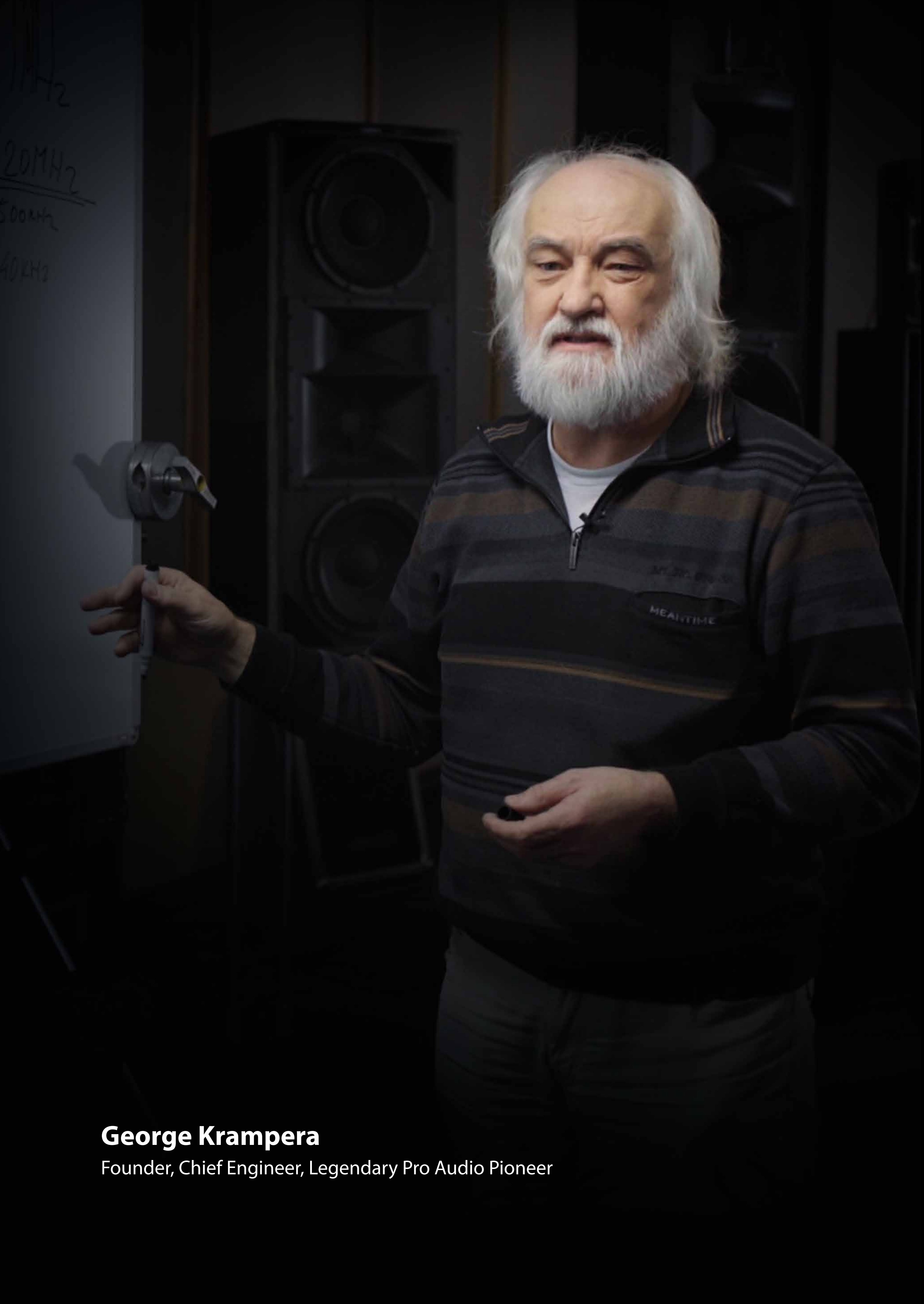
KV2 Curriculum

A Guide to KV2's Audio Secrets





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KV2 Curriculum

A Guide to KV2's Audio Secrets

SOUND BASICS
LIVE SOUND BASICS
TIME
SUPERIOR CIRCUITRY
DIGITAL
SUPERIOR DRIVERS
POINT SOURCE POWER

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Module 1
SOUND BASICS

There's an old Chinese saying that, loosely translated, reads *"A Journey of a Thousand Miles Begins with a Single Step".* The message of this adage is that any undertaking needs to be started. Perhaps if the phrase were modified, it would read something along the lines of *"if it needs to get done–get off the couch and start it".*

The direction of that first step is also important. Too often in the study of live sound, the information ecosystem is filled with incomplete, inaccurate, or even intentionally misleading information. Knowing what is factual and what's not can be difficult to determine, and all too often misinformation threatens to derail a serious effort to begin a career.
In this lesson and the ones to follow, we will explore many of the fundamentals of live sound and sound reinforcement.

Module 1.1
FREQUENCY AND AMPLITUDE

At its core, sound has two important features–frequency and amplitude–that determine how it is perceived:

FREQUENCY

When we refer to the frequency of a sound, we're talking about that sound's fundamental pitch and the sound's overtone series that give it its specific timbre. High-pitched sounds have a higher fundamental frequency value, and low-pitched sounds have a lower fundamental frequency value. An understanding of frequency is essential to any live sound engineer.

When viewed through an oscilloscope, all naturally occurring audio begins with the waveform ascending over the zero-volt line. This energy will reach its peak, and then go back down to the zero line. This part of the audio waveform–energy building, finding its peak, and then coming back down–is called the ***compression*** phase of an audio waveform. The term comes from the fact that air pressure (or the pressure of whatever medium the sound is passing through) increases in this part of an audio wave.

Because sound has a natural back-and-forth motion (since its energy comes from something vibrating), its energy will then drop below the zero line into negative territory. Here again, it will have a (negative) peak and return to the zero line. This is called the ***rarefaction*** phase of an audio waveform.

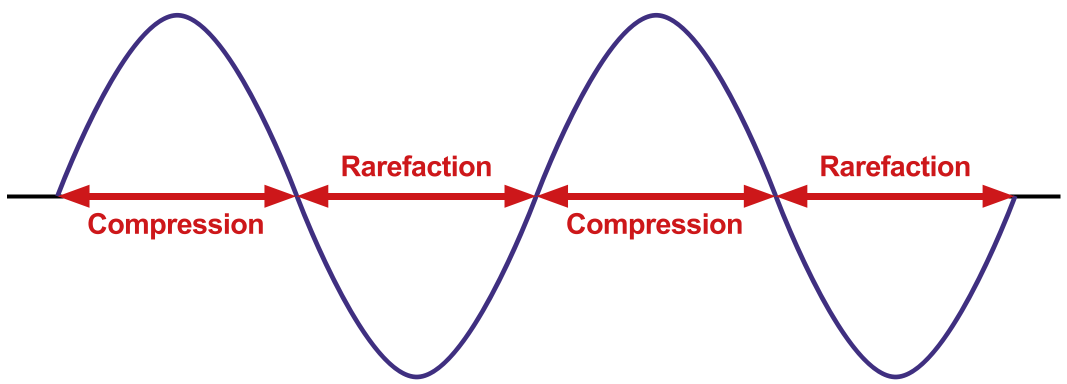


Figure 1.1: Compression and Rarefaction phases of a sine

Audio consists of a compression phase followed by a rarefaction phase (followed by another compression phase and another rarefaction phase, and so on until the sound ends). One period of compression followed by one period of rarefaction is called a ***cycle***.

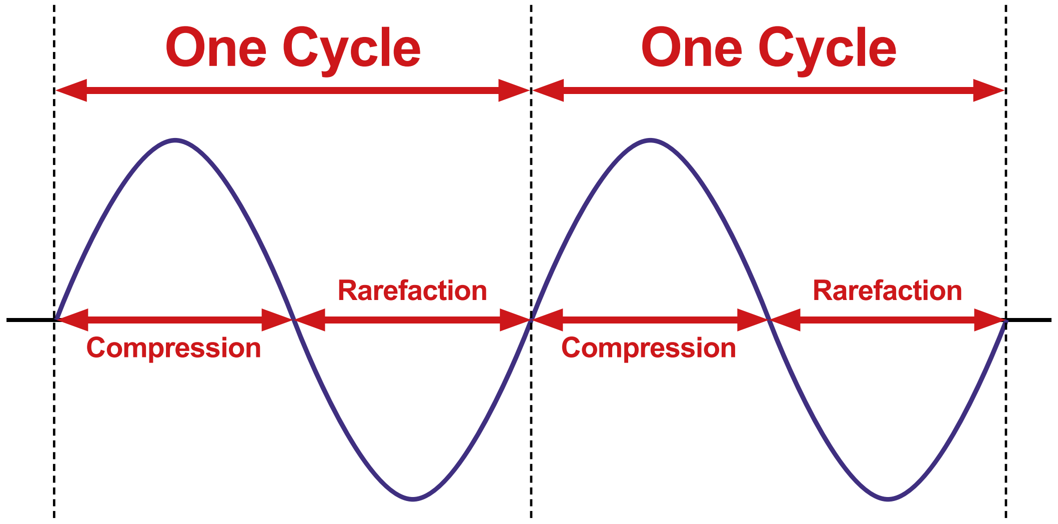


Figure 1.2: Two cycles of a sine wave.

And here we arrive at the definition: ***Frequency*** is the number of ***cycles*** per ***second***. The unit of measurement for frequency is Hertz (Hz). A lower-frequency sound has fewer cycles per second, and therefore a lower Hertz value. A higher-frequency sound has more cycles per second and a higher value.

NOTE: It's important to note that there's a difference between ***frequency*** and ***waveform***. Different waveforms (square waves, sine waves, and so on) can have identical frequencies. Different waveforms, however, exhibit different ***timbre***, which is defined by the ASA (Acoustical Society of America) as "that attribute of auditory sensation which enables a listener to judge that two non-identical sounds, similarly presented and having the same loudness and pitch, are dissimilar". This preservation of timbre is not only dependent on the frequency response of a recording or playback system, but also on its temporal (time) accuracy.

Different animals can hear different frequencies. That's how a dog whistle works–it makes a loud sound at a frequency that dogs can hear, but human beings cannot. A human being's frequency range is generally considered to be 20 Hz up to 20,000 Hz (or 20 ***kilohertz***, represented by the abbreviation ***kHz***).

NOTE: Although a hearing range of 20 Hz to 20 kHz is often quoted for human beings, it really doesn't apply to everyone. As we age, and depending on our listening habits, humans generally lose high-end frequency sensitivity over time. In general, our sensitivity to frequencies above 10 kHz begins to decrease significantly, beginning in our 30s.

AMPLITUDE

In the typical depiction of an audio waveform, the frequency is represented on the horizontal "X" axis (as the waveform oscillates over time), while the amplitude is represented on the vertical "Y" axis.

For example, the image below shows two sine waves of the same frequency, with the only difference being their amplitude – the top wave has a greater amplitude than the one below:

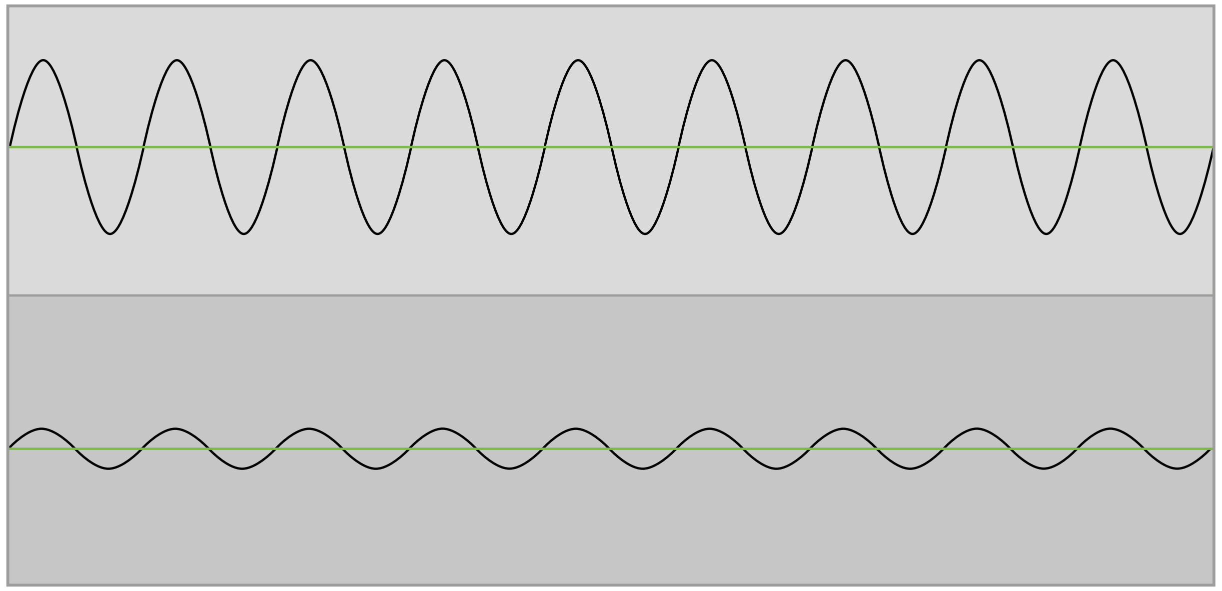


Figure 1.3: A sine wave with higher amplitude (top) and lower amplitude (bottom).

Amplitude can be affected in many ways–you can hit a drum harder or softer, blow more air through a tuba, or turn up the volume and send more voltage to a speaker. No matter how energy is supplied though, the result is the same: Waveforms with greater amplitude have stronger compression and rarefaction phases, moving more air, and can be heard over greater distances.

There are a few terms that could be used in place of amplitude, like ***volume***, ***gain***, or even ***voltage*** (when talking in terms of an audio signal going to a speaker). Be careful, though, of using the word ***loudness*** if what you're really talking about is ***amplitude*** (especially when talking to professionals!). Loudness deals with how sound is perceived, and the human ear is more sensitive to some frequencies than others. This means that two waveforms with identical shapes (for example, two sine waves) and amplitudes, but with different frequencies, could be perceived by people as having different levels of loudness.

NOTE: For those who want to learn more about the difference between amplitude and loudness, and the details of how we perceive sound at different frequencies, a great place to start is with a graph called the Fletcher-Munson curve[1].

Just as the unit of measurement for frequency is Hertz, amplitude has its unit of measurement, called a Decibel (dB). And just as humans have a frequency range of hearing, there is also a range of Decibels (or dynamic range) that we can withstand as well. The maximum human dynamic range is from 0 dB SPL (Sound Pressure Level), which is silent, up to 120~130 dB SPL, after which sound becomes painful. Again, this range, often called the threshold of hearing to the threshold of pain, varies from person to person.

[1] ***https://en.wikipedia.org/wiki/Equal-loudness_contour***

Module 1.2
SOUND AS A WAVE

It's commonly known that sounds are caused by vibrations, but to end the discussion there would be a mistake. To understand sound in the fundamental way needed for live sound reinforcement, we need to break down the process into individual elements that all play a role:

THE STARTING POINT: ENERGY

Sound essentially begins as a disruption, and the source of that disruption is energy. There are of course various kinds of energy, which fall into six general types:

- **Chemical** (for example, the burning of gasoline that powers a car)
- **Electrical** (energy created by the movement of electrons)
- **Radiant** (including visible and invisible light rays)
- **Mechanical** (determined by a physical object's movement or position)
- **Thermal** (the energy of vibrating molecules, expressed as heat)
- **Nuclear** (resulting from the fission or fusion of atoms)

Each of these forms of energy can come in the form of ***potential*** energy, such as a ball being held in the air, and ***kinetic***, which occurs as the ball drops to the ground. It is at this point that energy can be converted. For example, when a ball drops to the ground, a very small amount of that energy will be converted to thermal (heat) energy, some as elastic energy, while the remaining mechanical energy is manifested as sound.

The same logic can be extended to more complex situations: For example, an electrical signal can be converted to electromagnetic waves (that fall into the ***radiant*** energy family) which in turn are converted into the mechanical movements of a loudspeaker, again producing sound.

Cause and Effect

In some ways, light and sound are similar–for example, they both have frequencies and amplitudes–but when it comes to how they reach our senses, they are very different. We hear sound as something called a ***compression*** wave, but what exactly does that mean?

Sound is created as ***mechanical*** energy disrupts a medium (for example, air or water). When this happens, molecules are jolted into other molecules, who in turn ram into other molecules, and so on. This has the effect of crowding, or ***compressing*** the molecules, slightly increasing the air pressure.

Though no medium is completely static, the medium is certainly less dynamic than the sound source, and so the compression of the medium must be balanced out by a corresponding region of lower pressure (or ***rarefied***) medium.

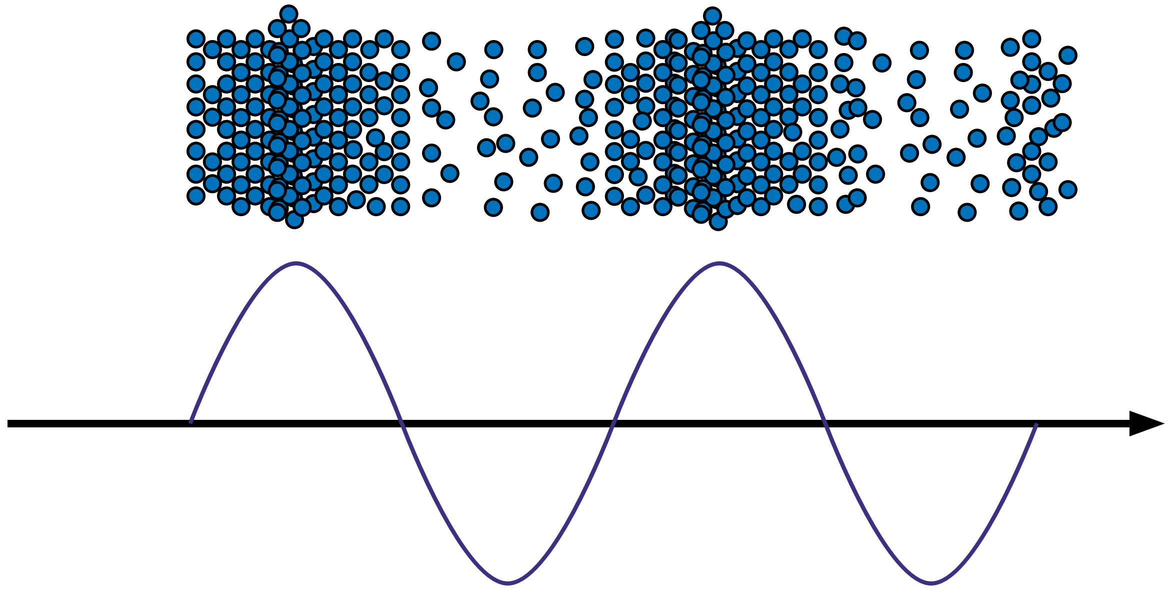


Figure 1.4: A compression wave, represented in molecular density (above), and a traditional audio waveform.

The traditional representation of a wave can be misleading, leading to the mistaken conclusion that molecules themselves are vibrating, as would be the case with thermal energy. That's not happening with the sounds you hear–what we perceive and represent as the crests and troughs of an audio waveform are the alternating states of compression and rarefaction of molecules in a medium. As the medium regains its equilibrium and as energy is dissipated, the perceived amplitude of the sound decreases to silence.

MEDIUM MATTERS

When it comes to light, photons are emitted by a luminous source until they directly reach our eyes. Light doesn't require a medium–in fact, it travels most efficiently without one, and even transparent media like air and glass will absorb a portion of light. Sound is nearly the opposite in this regard.

Audible sound in the real world is a manifestation of ***mechanical*** energy, expressed at the molecular level, and relies on a ***medium*** for sound to be transmitted.

The key to understanding how medium affects sound is based on the understanding that though we perceive light ***directly*** we hear sound ***indirectly***. How we perceive sound depends to a large degree on the ***medium*** of transmission.

When it comes to medium's effect on sound, there are a few general rules:

- Sound travels faster through denser media. For example, sound travels faster in solids than in liquids, and faster in liquids than in gasses.
- With a denser medium, less energy is lost as a result of the motion of particles, this results in sounds being audible over greater distances as the density of the medium increases.
- As the medium's density increases, more high-frequency content is absorbed and not transmitted.

This is perhaps best demonstrated by the famous "bell-in-vacuum" experiment, were air is pumped out of a jar in which a ringing bell is located. As the density of the medium decreases to a vacuum, the sound of the bell decreases until it's inaudible, despite the fact that it is still "ringing".

Blowing in the Wind

In addition to its density, the ***stability*** of a medium also has an effect upon its ability to transmit sound. Factors like wind and waves will disperse and redirect the motion of the medium's molecules, which will affect the distance and direction in which sounds will be heard.

Module 1.3
SOUND IN THE REAL WORLD

With a solid understanding of the fundamental aspect of frequency and amplitude, as well as to how sound is transmitted, a substantial discussion can be had on how sound manifests in real-world situations. In this final section we'll discuss two important factors: *Distance* and *phase*.

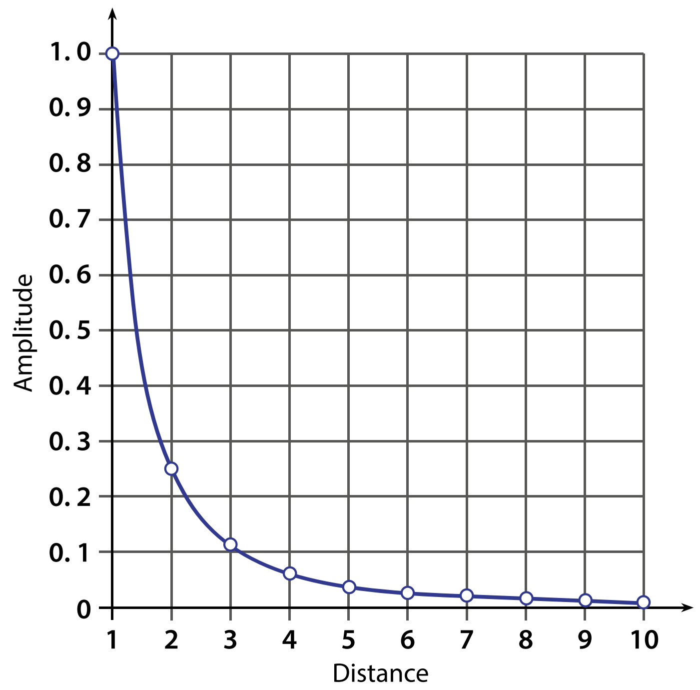
DISTANCE AND SOUND: THE INVERSE SQUARE LAW

Consider the image of a pebble dropping into a pond of still water. The initial energy of the impact radiates outward in all directions, as circular ripples. As each ripple concentrically radiates, the height of the wave decreases.

This makes logical sense–as the energy of the wave covers a larger circle (a greater area), it is "stretched" progressively thinner as the radius of the circle increases. There's a formula for calculating this decrease in amplitude (which applies to sound as well as waves in a pond), called the ***inverse square law***.

The inverse square law can be expressed in a number of ways, perhaps most simply that

Intensity=1/Distance^{2}. This can be applied using any standard of measurements–in this case we'll use meters.



If the intensity is 1I at one meter, then:

- The intensity is 1/4 I at 2 meters.
- The intensity is 1/9 I at 3 meters.
- The intensity is 1/16 I at 4 meters.

Figure 1.5: Inverse Square

The crux of the inverse square formula is this: A sound's amplitude doesn't decrease in linear proportion with distance. Instead, it drops most extremely at the outset of the sound.

PHASE AND POLARITY

To finish this lesson, let's discuss two aspects of sound that particularly impact the live sound industry–***phase*** and ***polarity***.

Phase

Earlier in this lesson, you learned about what a ***cycle*** is–phase refers to how far along a wave is in its cyclic journey.

It is perhaps easiest to think of phase in terms of a circle:

- A cycle begins at 0 degrees
- Halfway through the compression part of the cycle, the wave has gotten to 90 degrees
- At the end of the compression part of the cycle, the wave is halfway done at 180 degrees
- Halfway through the rarefaction part of the cycle, the wave has progressed to 270 degrees
- At the end of rarefaction, the cycle is complete, equaling 360 degrees, and the process starts again with another cycle

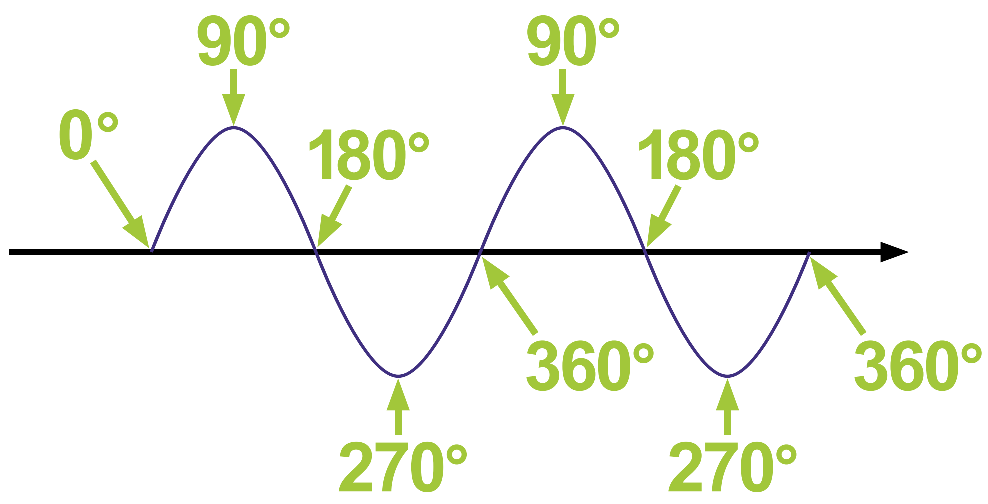


Figure 1.6: Two cycles, expressed in degrees of phase

In the real world, we don't hear sound from just one source, but many. That means that, prior to reaching our ears, different compression waves collide and combine. Phase plays a significant role in how we hear these multiple sounds.

Sometimes this combination can strengthen a sound. When two waves are ***phase-aligned*** (or "in phase"), their peaks and troughs strengthen each other. In the case of the image below, two sine waves of the same frequency are in phase on the top two tracks, and their result (on the bottom track), is the combined result–a similar sine wave with a greater amplitude:

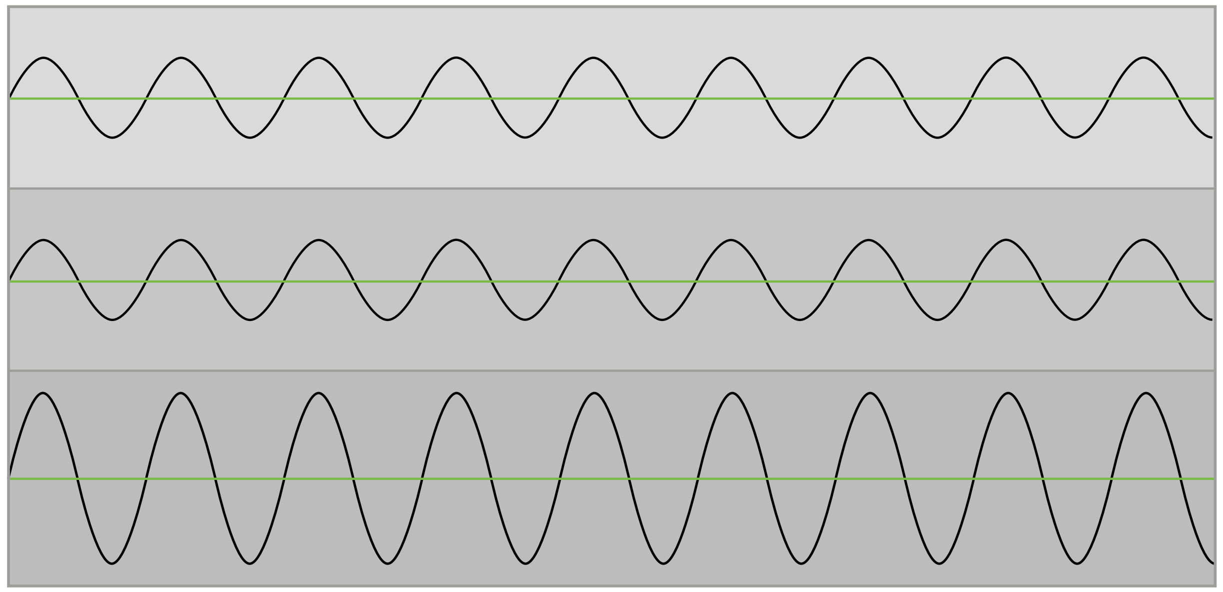


Figure 1.7: Two in-phase sine waves, and their resultant waveform (bottom track)

In actual practice, sounds from different sources are rarely aligned–even with "phase-aligned" systems, there is usually a degree of phase incoherence. This results in in a degree of cancellation, which at its most extreme (180 degrees out of phase) results in complete cancellation. The image below shows two sine waves, 180 degrees out of phase, with the combined result on the bottom track.

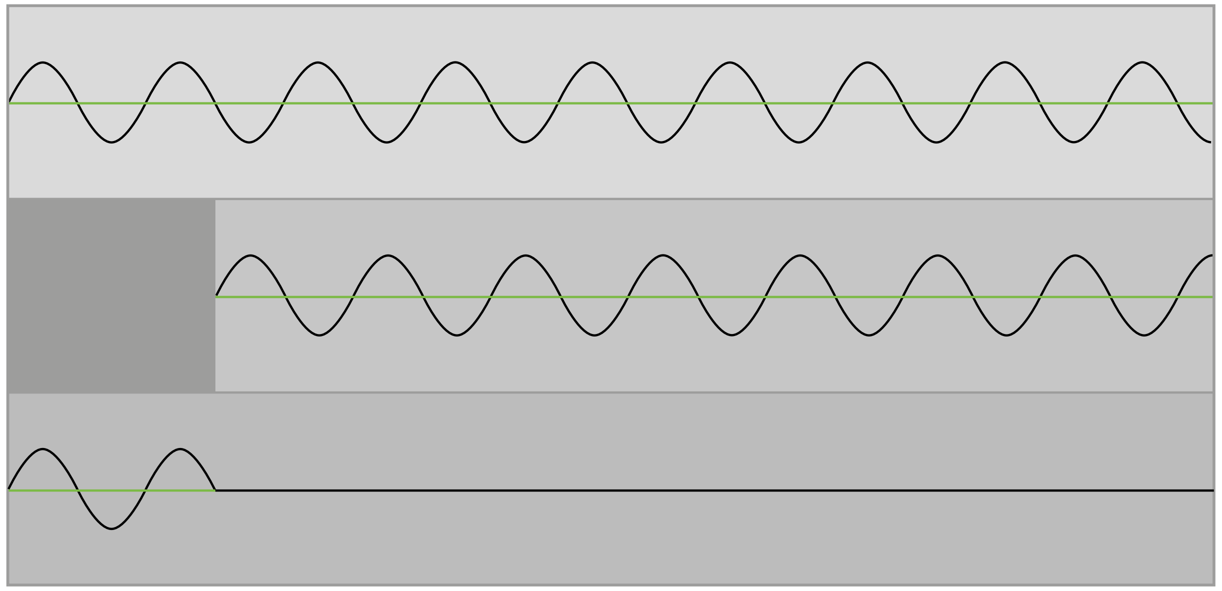


Figure 1.8: Two 180-degree out of phase sine waves, and their resultant waveform (bottom track)

Since we typically hear sounds from a variety of sources (and with a variety of waveforms), what we experience is an intricate combination of multiple compression waves, augmenting or diminishing each other depending on their phase relationships.

NOTE: In situations where similar signals are coming from multiple sources (for example, speaker arrays), even minute differences in phase alignment can have significant deleterious effect upon the timbre and clarity of the sound as it reaches the listener.

Polarity

Though the terms ***phase*** and ***polarity*** are often interchangeably used, they refer to two distinct things. ***Phase***, as we've discussed, refers to the progress of a waveform on its cyclic journey. ***Polarity***, on the other hand, refers to the sound pressure (or ***voltage***, in the case of a speaker).

The compression part of a waveform, with the wave ascending, represents an increase in the molecular density of the medium, and is represented electrically by a positive voltage. The rarefaction part of a waveform represents a decrease in the molecular density of the medium and is represented electrically by negative voltage.

Polarity can be inverted–either intentionally, or accidentally though incorrect wiring–so that positive values are changed to negative, and vice versa. This is commonly referred to as polarity inversion or "phase-flipping".

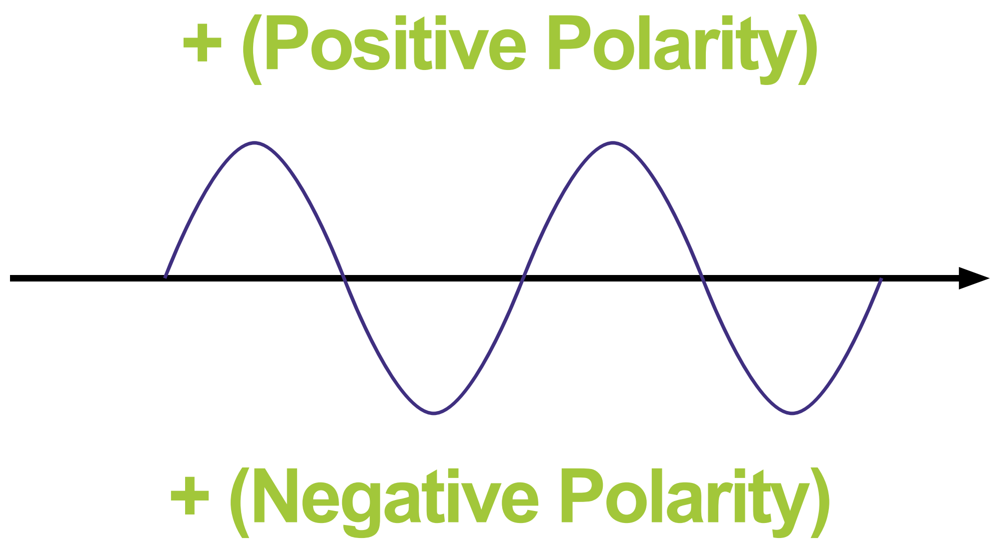


Figure 1.9: Positive and negative polarity in a waveform

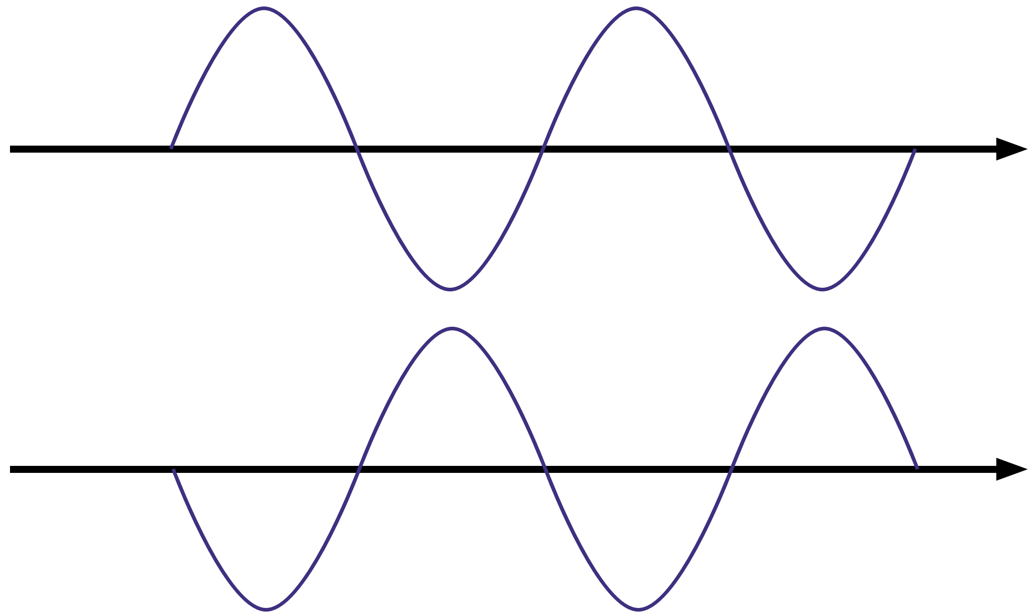


Figure 1.10: An original sine wave (top), and its inverted version (bottom)

NOTE: Though inverting polarity and being 180 degrees out of phase are technically different, the results are similar–combining a wave with its polarity-inverted mirror image will completely cancel each other out.

Given that sound reinforcement involves electrical wiring and (typically) identical sounds emanating from multiple sources, the issues of phase and polarity are critically important. Phase misalignment and inadvertent polarity inversions can radically alter the timbre and amplitude of sound and should be avoided.

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Module 2
LIVE SOUND BASICS

Live sound is a test of ability – regardless of the equipment being used, the job requires competence from each member of the technical staff. From riggers and system engineers, to the mixer ultimately responsible for the audio that the audience hears, a solid understanding of basic concepts is essential, in order to convey the artist's music to the audience.

In addition to mastering fundamental sonic concepts (discussed in the first lesson of this series), the live sound professional must also understand the operation of highly technical equipment like mixers, crossovers, equalisers and effect units. Armed with this knowledge, they can provide optimum system fidelity, capturing the real-time nuances of a performance. This applies to systems of all kinds – from small 2-way PAs on poles to huge multi-source systems at outdoor festivals – and everything in between.

Building upon concepts presented in this series' first lesson, this lesson deals with the core concepts of live sound.

Module 2.1
SIGNAL FLOW

For a live sound engineer, the most important skillset that must be learned is signal flow. Without this skill it is impossible to deliver a clean and controlled signal to the speakers. In fact, without a clear understanding of signal flow and how the elements of a PA system work together, you might not get any sound out of the speakers at all!

INPUTS

The first step in sound reproduction is the source. In the analog domain, this source involves the use of a microphone, a DI (Direct Injection) device, or Line Input.

Microphone Level Inputs

The term ***microphone*** level signal refers to the voltage generated by a microphone as it responds to sound. Just a few thousandths of a volt, this voltage is routed through a ***preamplifier***. This gives the sound engineer the first stage of ***gain control***, which increases the level of what the microphone "hears" before the signal is introduced into the next stage of amplification.

A microphone preamplifier often includes a ***Pad***, which attenuates very loud sources like drums, guitar amplifiers and brass to avoid distortion of the input signal. Other common preamplifier controls include ***phantom power***, (for condenser style microphones and active DI boxes), and ***High Pass Filters*** (which attenuate unwanted low-frequency information).

Line Level Inputs

A ***line level*** input is approximately one volt, or about 1,000 times as strong as a mic-level signal, and is used for replay devices, effect returns, and many other types of equipment. There are two standard line levels:

- **-10 dBV** generally used in consumer-level equipment (such as MP3 and DVD players)
- **+4 dBu** for professional equipment (including mixing desks and signal processing gear)

DI Boxes

DI - or ***Direct Injection*** – describes a device that will convert an unbalanced, high impedance, instrument or line level signal to a balanced, low impedance signal. Typically using a 1/4" or 6.3mm jack input and 3 pin XLR output, it can either be of a passive or active design, the latter requiring 48v phantom power, supplied by the mixing console through the XLR signal cable.

Input Connection Considerations

When connecting source signals to the mixing console, there are a few points to bear in mind:

- ***Cable Length*** can have a significant impact on the sound quality, due to capacitance. This results in signal loss, mainly in the high frequency domain.
- If long cable runs cause attenuation of the signal, it may necessitate the need to boost the input gain of a microphone preamp. This can raise not only the desired signal, but also the accompanying noise floor.
- This problem can be solved by using a Line Driver, such as the KV2 Audio SD8 or LD4, designed specifically to eliminate standing waves and signal impurities resulting in the delivery of a high quality audio signal no matter what the cable length.

MIXING CONSOLES

The nexus of the live sound system is the mixing console (or "mix desk"), where the live sound engineer combines and processes signals from various sources prior to output to the PA speakers. Mixing consoles come in two basic types – ***analog*** and ***digital***.

Analog Consoles

In the case of ***analog*** mixing desks, all audio mixing remains in the analog domain. As a general rule, analog consoles are outfitted with semi-parametric EQs on every channel, and outboard (external) processors are used for dynamics and effects processing (gates, compressors, reverbs, delays, and so on). Analog consoles come in all sizes and prices, with the leading analog consoles considered sonically superior to their newer digital counterparts.

Digital Consoles

Digital consoles have superseded their analog counterparts, due to several advantages that relate to their practical application. Digital mixing desks boast smaller size, higher channel count, internal effects and dynamic processing, and expanded routing capabilities. These features have made them the new standard in audio at all levels. However, even the top digital consoles are considered inferior to high-end analog consoles when it comes to audio quality.

NOTE: Digital audio formats (and their inherent sonic problems) are discussed in Lesson 5 of this series – ***The KV2 Difference: Digital***.

EFFECTS

Along with mixing various signals at the mixing desk, the live sound engineer employs a variety of effects to sculp disparate elements into a cohesive whole

Analog vs. Digital Effects

Like mixing consoles, effects processors come in two flavors; Analog and Digital. In recent years, analog effects are more rarely used than digital devices, apart from some very expensive units that are mainly intended for studio use and legacy units that pre-date digital (equalizers, compressor/limiters or gates, that populated effect racks before the advent of digital consoles). Also like mixing desks, well-designed analog processors are considered superior to their digital processor counterparts.

Figure 2.1: The first digital delay, released by Eventide in 1971

Dynamic vs. Time-based Effects

Effects – and how they are implemented – is differentiated by the type of effect. The two primary effect types are ***dynamic*** effects and ***time-based*** effects.

Dynamic effects include ***compressors***, ***expanders***, ***limiters***, ***gates*** and ***multi-band compressors.*** These effects affect the amplitude of the signal being processed to avoid overloading (or "clipping"), reducing noise, or reducing amplitude peaks (resulting in the overall output in being "louder" to the listener).

Time-based effects include any effects that utilize time delay to create an effect. This can include phasers, flangers, reverbs and digital delays.

Equalization

Equalization (EQ) is the process of adjusting the balance between frequency components within an electronic signal. Early equalizers were typically simple controls with bass and treble, and evolved to include ***Graphic Equalizers***, that comprised a bank of filters with set frequencies. This progressed further to the modern ***Parametric Equalizers***, which feature a series of multi-band filters that can control amplitude, frequency and bandwidth.

In music production, EQs are primarily used to shape the sound and correct tonal problems. In a live sound environment, they are used extensively to fine-tune a PA system to correct any deficiencies in the electronics and speakers and adjust the system's acoustic response in a venue.

Reverb

Reverberation – ***reverb*** for short - is a persistence of sound after the original sound is produced, created by the space within which the sound occurs.

In studio situations, natural reverb in music is achieved by capturing the ambient reverberation generated in the recording space by the musicians and their instruments during the recording. The distance between the microphone and the sound source determines how "wet" or "dry" the recorded sound becomes. One inherent problem with this method, is that the reverb is a part of the recorded performance and cannot be altered. This limitation has led to the creation of reverb effects units, allowing this ambience to be added manually by the mix engineer.

The first case of reverb being applied separately to the recording was by Bill Putnam Senior in 1947. He designed a concrete room that became known as an ***echo chamber***. A speaker was placed in the room and fed with an audio signal. This stimulated the natural room acoustics and was then picked up by a microphone and sent back to the mixer. Now the engineer had an isolated reverb signal to mix into the music as required.

Although a huge advance, this method provided no way of adjusting the reverb time and building empty rooms to create reverb was expensive (and space-consuming). Around this time, mechanical reverb systems were being developed, the first being a spring reverb, introduced in Hammond Organs in the 1930s and 1940s, and adopted by Fender and Moog in the 1960s. This was quickly followed by the famous EMT140 plate reverb, from EMT International GmbH.

In 1976 the first digital reverb was released – the EMT250, also from EMT International GmbH. Since then, digital reverbs have become ubiquitous, extremely sophisticated and are now a part of the internal digital processing chain in most digital live sound mixers. Ironically most digital reverbs have "chamber" and "plate" settings, an emulation of the earliest forms of analog reverb technology.

CROSSOVERS

Crossovers are devices that split an incoming signal into frequency bands appropriate to your speaker system. This may be a 2-Way (Lows and Highs), 3-Way (Lows, Mids, and Highs) or more splits to suit a particular speaker configuration.

Crossovers can be ***Active*** or ***Passive***, with the fundamental difference being where this split occurs. An active crossover splits the frequency band ***before*** the amplification stage, where a passive crossover splits the signal ***after*** the amplifier.

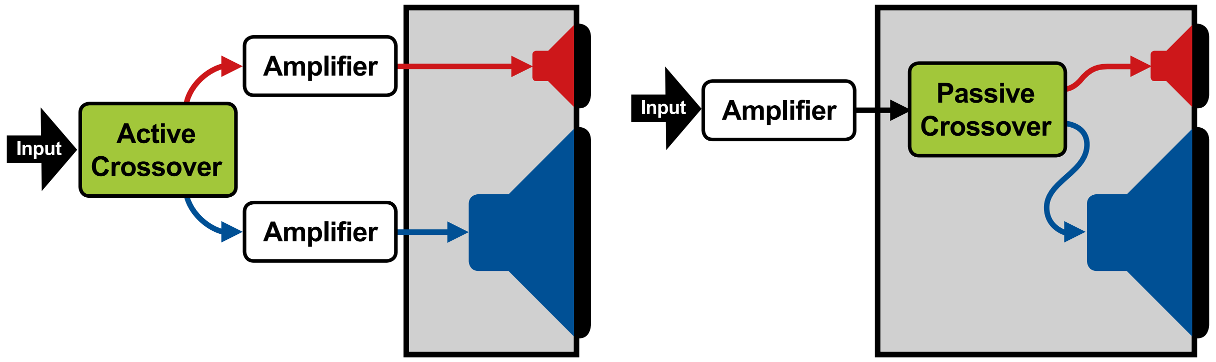

Figure 2.2: Active crossover (left) and passive crossover (right)

These two types involve different design concepts with each method having advantages and disadvantages. Active crossovers often provide more control over the signal (gain, crossover frequency, crossover slope, and so on), but require more amplification channels than passive crossovers. Passive crossovers have the advantage of only requiring a single amp channel prior to the crossover network but will generally need more power to share across the various bands.

AMPLIFICATION

Amplifiers are the penultimate stage in the audio signal path, providing the power to drive the speakers.

There are several types of amplifier topologies, including Class A, Class B, Class AB, Class D and several others. Currently, the most common types of amplifiers are based on Class D designs.

Whilst Class D amps have the advantage of being lightweight and very efficient, they are not linear because they do not amplify the input signal directly, like linear amplifiers do. Instead, they use pulse-width modulation (PWM) techniques to convert the input signal into a high-frequency signal that switches between a high and low voltage level.

The resulting output waveform is a series of pulses with varying width and duration, which are filtered to produce a high-power audio signal. This switching process introduces distortion and noise into the signal, which can result in non-linear behavior and can cause them to be susceptible to electromagnetic interference (EMI) and electromagnetic compatibility (EMC) issues.

In contrast, linear amplifiers amplify the input signal directly, without introducing almost any distortion or noise. The output waveform of a linear amplifier is a faithful reproduction of the input waveform, scaled to a higher voltage level. This linear behavior is desirable as fidelity and accuracy are a critical link in the audio chain.

Amplifier Considerations

With regard to amplifier usage, there are a few considerations for professionals to bear in mind:

- High quality amplifiers are specifically designed for the best quality reproduction of specific frequency ranges (Low, Mid, or High).

NOTE: The importance of amplifier electronics, and KV2's ultra-fast circuitry is discussed in Lesson 4 of this series – ***The KV2 Difference: Superior Circuitry***.

- Speaker cables feed the speaker system with a much higher voltage than is found at line level, requiring a different style of cable that can handle the voltages needed to drive a speaker level signal generated by an amplifier. Excessively long cable lengths (or cables that are too thin), can also cause significant loss of volume, high frequency loss, poor bass definition, and possible damage to the amplifier.

Module 2.2
SPEAKERS

Speakers are the crucial final stage in the audio signal path, converting electrical energy to mechanical speaker motion, generating sound. Speakers are arguably the single most important component in delivering quality sound to the listening audience.

SPEAKER USAGE

Speakers are typically used in a live sound system in two fundamental ways:

- **Main (FOH) Speakers** are speakers that provide audio for the audience. Speakers should be chosen that have the power to cover the audience area with no gaps in coverage. Different styles of content may require different types of systems - for example, spoken word and BGM (Background music) generally do not require extensive bass reinforcement, but EDM or other electronic music typically involve a large number of sub-woofer cabinets to provide enough low frequency energy.

- **Stage Monitors** are used to provide sound reinforcement or "foldback" to a performer on stage. If there are multiple performers, there can be a number of discrete monitor "sends", so that each performer is able to hear a mix of the sound that suits their individual needs.

CHOOSING THE RIGHT SPEAKER

There are two types of speakers in common use in professional audio:

- The Dynamic Loudspeaker is commonly seen in speaker cabinets that deal with frequencies from sub and low frequencies to high mid-range signals.
- Compression Drivers generally then cover the higher frequency band and use a horn flare, or waveguide, to direct the high frequency information in a particular direction with a specific coverage pattern.
- Whenever possible, it is preferable to "fly" the audio speakers. By increasing the angle of incidence at which the sound arrives at the listeners' ears, **flown speakers greatly improve coverage and intelligibility**. This type of installation clears the floor of the venue, improving sight lines to the stage and making the event more professional-looking.

Speaker selection should be based on several criteria:

- Size of venue
- Type of audio content
- Maximum required volume (SPL or Sound Pressure Level)
- Ground stacked or flown configuration
- Indoor or outdoor venue
- Budget

SPEAKER PLACEMENT

Speakers can either be flown or stacked:

Flown Speakers

Speakers can be suspended, or "flown", through the use of cables or chain motors. It is important that professional rigging equipment (that has been certified for use by the appropriate authority) is used, strictly observing the legal weight rating for the equipment (the speakers must never exceed this limit). The installation of flown speaker setups must be conducted by qualified riggers.

When possible, it is preferable to "fly" the audio speakers. By increasing the angle of incidence at which the sound arrives at the listeners' ears, flown speakers greatly improve coverage and intelligibility. This type of installation clears the floor of the venue, improving sight lines to the stage and making the event more professional-looking.

Ground Stacked Speakers

In smaller venues, or where there is not the required supporting structure, it is necessary to set up the PA system on the ground. This could be as simple as a speaker pole or stand or may require the stacking of several cabinets which are then strapped together to ensure that they are stable and won't collapse onto the audience.

A significant disadvantage of this type of setup is that the speakers are usually arranged in such as way that the output of the speakers is directed parallel to the ground and the top of the audience. This means that as distance from the speaker increases, there is a corresponding decrease in high frequency energy, as it is absorbed by the audience.

SPEAKER ARRAYS

As sound systems become more complex, they will naturally involve more amplification and greater numbers of speakers. There are a variety of configurations that can be used for these larger, multi-speaker setups.

Multi-source

A ***multi-source*** system involves a number of identical full range cabinets, distributed along horizontal (X) and vertical (Y) axes. This configuration is based on a belief that more speakers will equal more volume, allowing these systems to service larger audience areas. Although this was achieved in terms of increased volume or SPL, it was at a significant cost.

Despite multi-source systems producing greater SPL at their source, the ultimate sound that reaches the audience suffers from some substantial problems. Combining the signal from multiple drivers that are spaced apart introduces significant errors due to the different arrival times of the multiple signals. In a large multi-source system, this could be in excess of 800 individual speakers, each one creating sound that arrives at the listeners ears at a different time (with varying degrees of delay).

Line-source

A ***line-source*** or "***line-array***" system is also a multi-source configuration, but with speakers positioned in vertical Y axis only. This type of system is a significant improvement on the original multi-source concept, but still fails to address some fundamental issues associated with the use of multiple speakers. A line-source system still has the inherent problem of distortion along the Y axis and though some manufacturers claim that their designs can create a coherent wave front and controlled vertical dispersion, empirical evidence shows that this claim is far from true.

Point-source

The term ***point source*** is used to describe a speaker (or speaker system) wherein all of the audio emanates from a single source. This can include multiple loudspeakers that are mounted in the same enclosure, each reproducing a different part of the audible frequency range, with the total set of speakers behaving as a single source.

Until recently, audio technology had not reached the point where large audience areas could be covered by a single point source speaker system. However, with advances in amplifier and transducer design it is now possible to create a point-source system that is capable of covering very large audience areas, and with sonic quality that surpasses that of multi-source or line-source systems. With all sound being projected from a single coherent source, intelligibility, impulse response, and evenness of coverage surpasses any other system technology.

NOTE: Speaker arrays were introduced with multi-source and then line-source systems, but it is worth noting that point-source systems can employ arrayed cabinets without breaking the rules of point-source technology.

Distributed Systems

Commonly found in large-scale background music or theme park installations, a ***distributed system*** is one in which multiple speakers are powered by a single amplifier in parallel. All types of speaker design can be employed in a distributed system, which is typically required to provide coverage of an area that cannot be covered by a single speaker array. They typically use a 70 v or 100v amplification system to reduce signal loss over long distances.

Normally not employing high quality sound-reinforcement speakers, these systems are designed to distribute medium quality audio over large areas.

Delay Systems

In large venues, there can be a need to add additional speakers known as a ***delay system***. This is required to restore the clarity and volume of the main speaker system, once the original sound has deteriorated due to increased distance from the main speakers (due to the ***inverse square law***, discussed in the first lesson of this series). Delay systems are also employed in venues where the main system is unable to cover the entire audience area due to factors such as overhanging balconies, irregular shaped rooms, and so on.

In order to maintain coherence and intelligibility, A delay must be applied to these additional speakers. The time value (in milliseconds) is set so that the sound from the delay system's speakers arrives to the listeners ears at precisely the same time as the attenuated sound from the main system.

Delay systems allow the engineer to create a consistent sound field by augmenting the main system with additional speakers where required. This provides better overall coverage, but care must be taken to ensure that the results are as seamless as possible, with special attention paid to the placement of speakers and the application of delay times and gain control.

Speaker Array Issues

Multi-source and line-source have inherent weaknesses that stem from the fact that the listener is hearing multiple instances of the original sound - with each signal exhibiting minute differences in arrival time as it reaches the listener's ears. This results in phase alignment issues in a phenomenon called ***destructive interference***. Destructive interference is perceived as a "smearing" of the sonic image, reduction in clarity and intelligibility, and damage to the system's overall impulse response.

Though the use of multi-source or line-source configurations have been generally adopted to cover large areas, they carry serious issues that should not be overlooked;

- As the listener's distance from the PA increases, so does the damage to the signal, reducing its ability to accurately reproduce the original detail, dynamics and nuance in the sound.
- The problem of destructive interference is exacerbated as the listener moves, as the small delays between all of the speakers continuously change due to the changing distances between the listener and the sound sources. For this reason, it is very hard to create an even sound field when employing a multi-source or line-source system.
- Multi-source configurations used in an outdoor setting can be significantly affected by wind. As a result of wind moving the air molecules in a random and vigorous manner, the sound from disparate sources can be severely disturbed, and in extreme cases can cause a complete loss of high frequency content.
- Line array and Multi-source architectures are also inherently inefficient - more speakers result in more wasted energy (through destructive interference). The presumption that adding more speakers will result in a more efficient and louder PA system is mitigated by destructive interference.

It is important to note that point-source systems do not introduce these problems.

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Module 3
TIME

When we think about audio, it's common to talk about frequency (the number of waveform cycles per second, measured in Hertz), and amplitude (the amount of energy that comprises a sound, measured in Decibels). While these are certainly important parameters of sound – aspects that any quality speaker must deliver to the listening public – one critical aspect that is often overlooked is the time dimension.

In this lesson, we'll define the time dimension and how it impacts the listening experience. In following lessons we'll show how KV2 uniquely addresses this often-overlooked aspect of sound, delivering the highest-quality listening experience to the audience.

Module 3.1
THE TIME DIMENSION

One of the keys to KV2's pristine audio quality is the dedication to all aspects of audio reproduction, including the often-overlooked *time* aspect. But what exactly is meant by "the time dimension"?

AUDIO FUNDAMENTALS: FREQUENCY AND AMPLITUDE

In order to understand where time fits into the overall picture (and why it's so crucial), let's start with some review of concepts you learned in the first lesson of this series:

Frequency

When we refer to the frequency of a sound, we're talking about that sound's fundamental ***pitch*** and the sound's overtone series that give it its specific ***timbre***.

Audio consists of a compression phase followed by a rarefaction phase (followed by another compression phase and another or rarefaction phase, and so on until the sound ends). One period of compression followed by one period of rarefaction is called a ***cycle***.

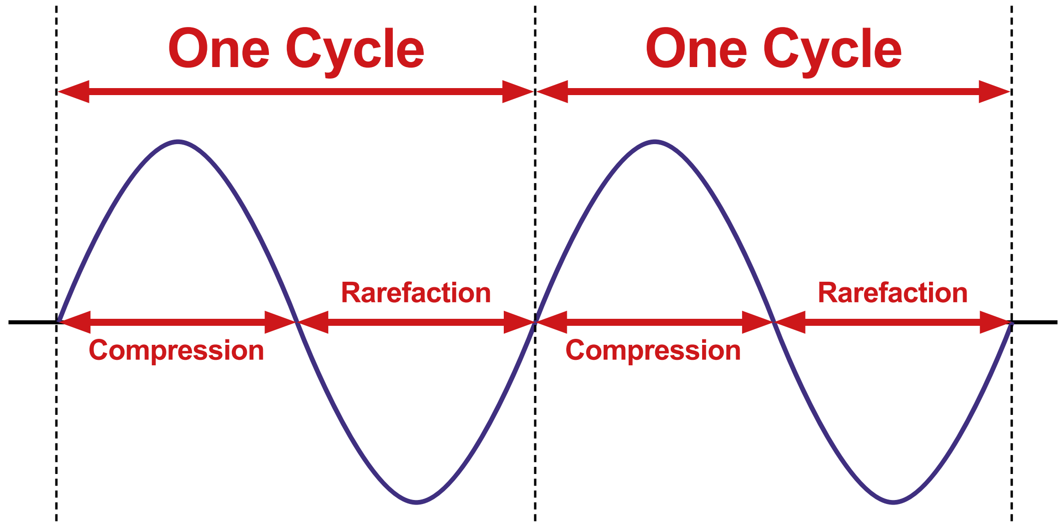

Figure 3.1: Compression and Rarefaction phases of a sine wave.

Frequency is the number of ***cycles*** per ***second***. The unit of measurement for frequency is Hertz (Hz). A lower-frequency sound has fewer cycles per second, and therefore a lower Hertz value. A higher-frequency sound has more cycles per second and a higher value.

Amplitude

In the typical depiction of an audio waveform, the frequency is represented on the horizontal "X" axis (as the waveform oscillates over time), while the amplitude is represented on the vertical "Y" axis. Waveforms with greater amplitude have stronger compression and rarefaction phases, moving more air, and can be heard over greater distances. Just as the unit of measurement for frequency is Hertz, amplitude has its unit of measurement, called a ***Decibel*** (dB).

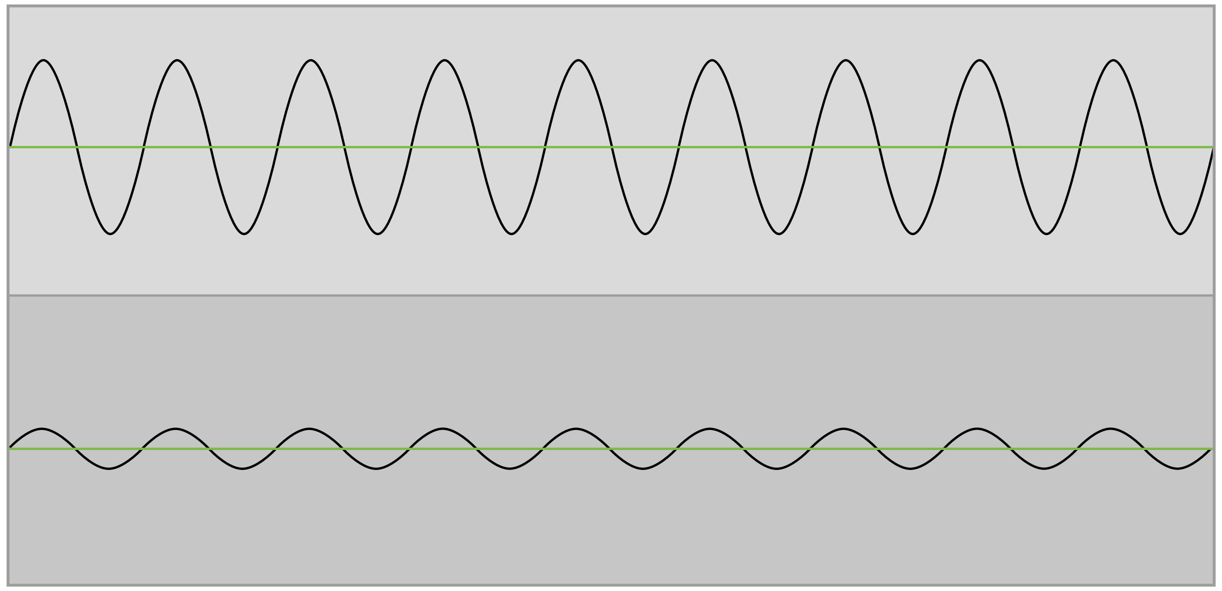

Figure 3.2: A sine wave with higher amplitude (top) and lower amplitude (bottom).

ADDING ANOTHER DIMENSION: TIME

There is another dimension to sound that is often overlooked, which is the dimension of ***time***. Since it's so underrepresented in discussions of audio recording and reproduction, it's important to understand exactly how the time (or ***temporal***) aspect of a sound contributes to the listening experience.

The Temporal Axis

As scientists will tell you, many of the visual representations of physical concepts are simply models, allowing our limited intellects to comprehend the world around us. This is certainly true of our typical depictions of audio on a two-axis graph. We traditionally think of amplitude as being the energy of the wave, measured on the "Y" axis of the graph, and the frequency of the audio represented on the "X" axis in terms of cycles per second. This horizontal axis represents time to be sure, but it's not the only (and arguably not even the most important) role that time plays in the sound that we hear.

There is another time-based dimension of sound, which we will refer to as ***temporal*** for the sake of clarity. The temporal aspect of the sound refers to the time at which the sound takes place. For example, you could have two completely identical sine waves, at exactly the same frequency and amplitude, but occurring at different times – that's the ***temporal*** difference.

Expanding on the typical representation of sound, factoring the temporal dimension of sound might be visually represented as shown below, showing that the amplitude and frequency of a sound exist independent of when that sound occurs, something that is certainly assumed, but rarely discussed as a critical aspect of sound. Furthermore, the temporal dimension of a sound doesn't simply apply to the start of the sound, but to every momentary detail of it, from start to finish.

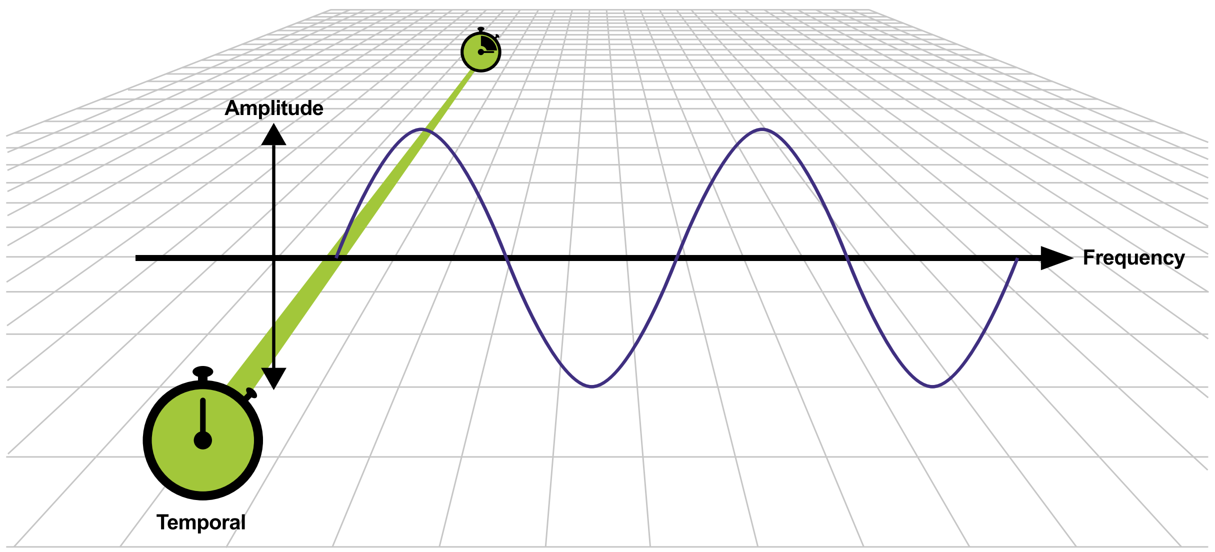

Figure 3.3: Adding a time (temporal) axis to the amplitude and frequency axis.

Just as we have a range of amplitudes that we can tolerate, and frequencies we can hear, we have temporal differences that we can perceive. The scale that we use when talking about temporal differences is microseconds (µs).

Temporal Resolution

The acuity with which we can discern very small differences in temporal location is commonly referred to as ***Temporal Resolution***. Given that this aspect of auditory perception is more subtle and less discussed than the more obvious amplitude and frequency aspects, our scientific understanding of our temporal senses is evolving, but the data are pointing in a specific direction.

In 2003, the Journal of the Acoustical Society of America published a white paper on the current thinking on temporal resolution and asserted that "that the limit of temporal resolution in humans is as little as 10–20 microseconds". Coincidentally, this lined up with the sampling period for the Red Book Audio format (44.1 kHz sample rate, with a Nyquist frequency (fn) of 22.05 kHz) which was roughly 23 microseconds (1/44100).

By 2007, the data pointed to a much greater degree of temporal discernment – roughly 5µs on average. In addition to the refinement of the testing criterion, this change in common thinking arose from the use of new ultrahigh-fidelity equipment.

One of the more important studies at this time was published in ***Technical Acoustics***, where ***Milind N. Kunchur*** (University of South Carolina Department of Physics and Astronomy) concluded "for transient stimuli, the auditory system's temporal acuity τ may be estimated to be in the 2–16 µs range, taking $t_0/\sqrt{N}$ with N=60–4000. Notice that the value of this τ has very little to do with the high-frequency audibility limit f_{max}." This conclusion, based on rigorous testing, is important – it not only suggests that humans have more accurate temporal perceptions than previously thought, but also specifically differentiates between temporal acuity and frequency acuity.

Temporal Resolution and Frequency

Because both the timing of a sound (temporal position) and its frequency are both related to time, it might be tempting to equate the two. However, our ability to discern temporal differences is largely unrelated to our ability to hear frequencies.

Here's one way to look at it: The upward limit of the human frequency range is generally considered to be 20kHz, or 20,000 cycles per second. The time that a full cycle at that high frequency takes is 50 microseconds (1/20,000), which by any measure (even outdated ones) is much greater than our temporal resolution. Looking at it from the other way, a frequency of a period of 5 microseconds would be in the range of 200kHz! Clearly, our ability to recognize the temporal information of audio events is largely independent of our ability to hear frequency.

Figure 3.4 shows the different aspects of human audio perception in another way - a single integrated graph:

1. The vertical scale shows the levels of ***amplitude***, from 0dB (the threshold of hearing) to 120dB (the threshold of pain)

2. On the horizontal time scale, the first number that you come to is 5µs, which is our ***temporal resolution***.

3. The next significant numbers are 50µs-50ms, which are the cycle periods for 20kHz-20Hz, the ***audible frequency spectrum***. Note that these numbers do not approach the temporal resolution value

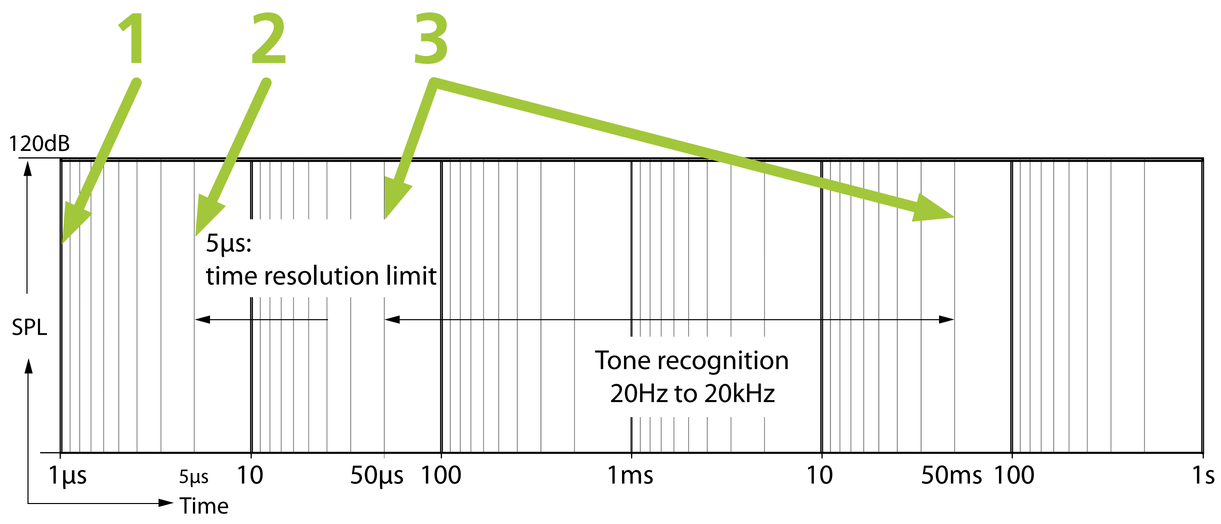

Figure 3.4: Another way to look at how we perceive audio.

NOTE: Just as our ability to hear higher frequencies decreases as we age, our temporal resolution will decrease as well, but to a lesser degree – in other words we retain more of our ability to discern timing even after we lose the ability to hear higher frequencies.

Module 3.2
THE IMPORTANCE OF TIME

In the previous lesson, we discussed what the temporal (time) dimension is, but what does it mean to the listener's experience?

TIME AND LOCALIZATION

The children's game of "Marco Polo" is a great example of how our brain processes audio input. One player closes their eyes and calls out "Marco!", and the other players respond by saying "Polo!". The first player then must locate one of the other players without opening their eyes, using only aural cues.

Of course, behind this simple entertainment, the players' brains are furiously interpreting audio data in a number of different ways and calculating different sounds' position with a high degree of accuracy. This is called localization.

A number of factors come into play when the brain calculates a sound's position. With regard to amplitude, for example, louder sounds are generally perceived as being closer than quieter ones. With regard to frequency, we can determine altitude and distance (with high frequency content diminishing as a sound becomes more distant as well as pitch shifting as a result of the Doppler effect).

Time also plays a major role in how we perceive position. Let's take a look at a few examples, starting with a simple one. If you look at figure 3.5, you'll see a simple example – a single sound source that is placed directly in front of the listener:

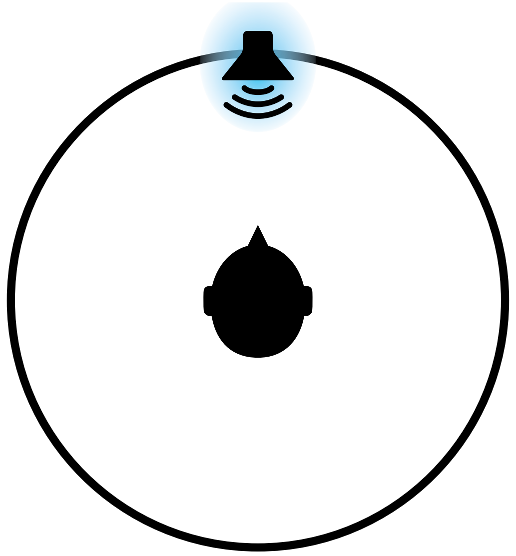

Figure 3.5: Sound directly in front of the listener.

With the sound emitting from a single position that is equidistant from each ear, the sound arrives at both ears simultaneously.

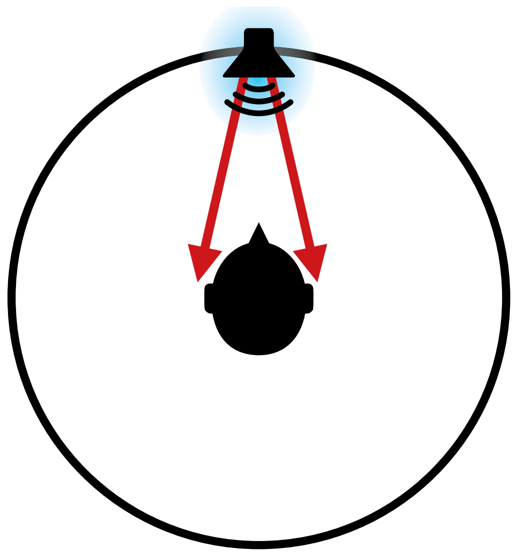

Figure 3.6: Sound from a single source arriving at each ear at the same time.

Moving the sound to the left of the listener, the sound will reach the closer left ear before reaching the right (there will of course be some effect of the size and shape of the head, as well as the position of the ears).

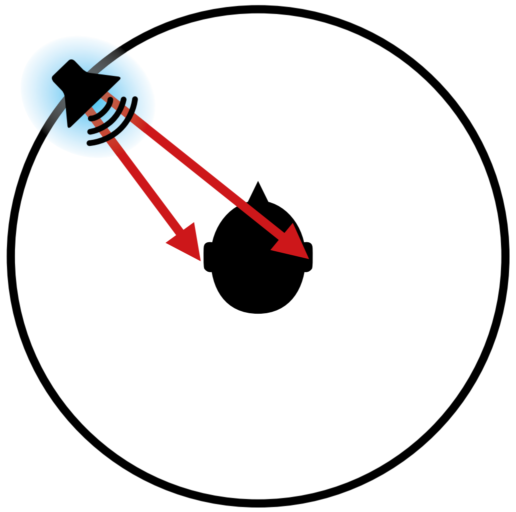

Figure 3.7: Sound from a single source arriving at the left ear before the right ear.

…and of course, the reverse is true when the sound source is moved to the right of the listener:

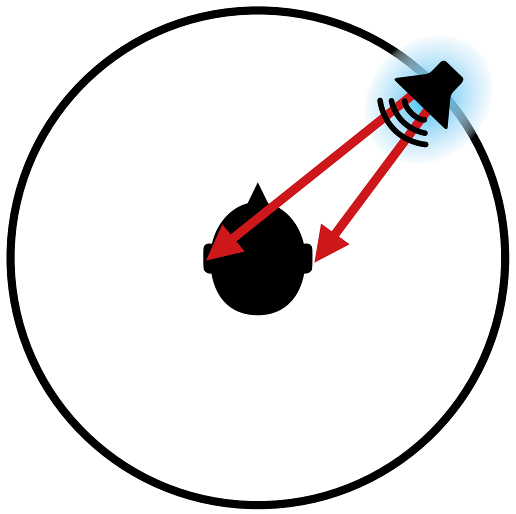

Figure 3.8: Sound from a single source arriving at the right ear before the left ear.

How great can this difference in timing be? If you consider that the speed of sound through air under normal conditions is 344 m/s (or 1,238 km/h or 770 mph) and the average distance between a human's ears is roughly 14 cm, then the maximum difference in time between the left and right ears is 411.402 µs. This would be the case where a sound was directly to the left or the right of the listener, as **shown in figure 3.9**. As a sound approaches being directly in front of the listener, the interaural difference is reduced to zero.

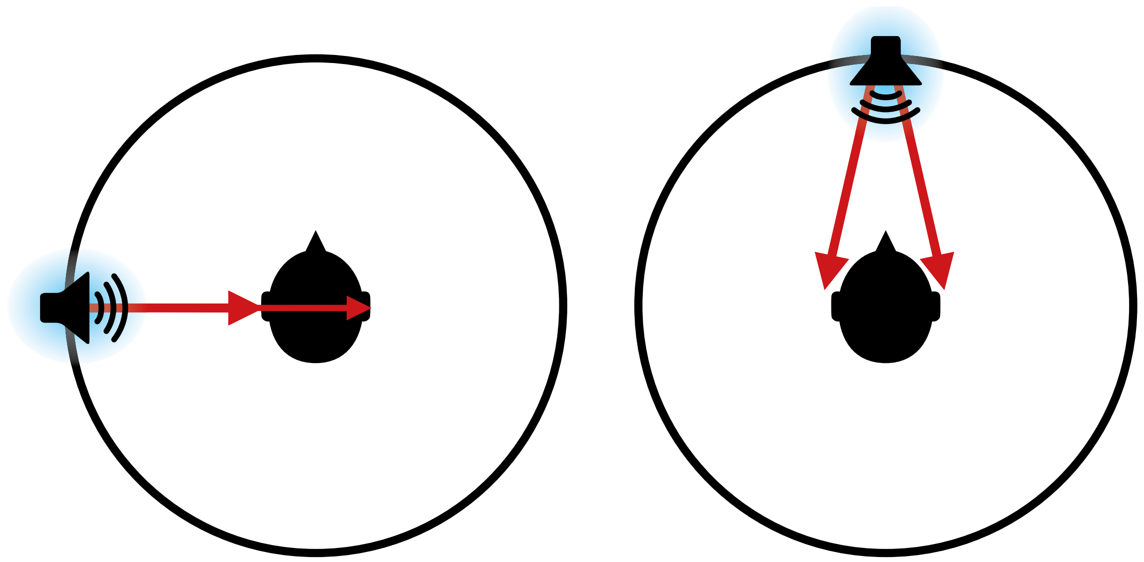

Figure 3.9: Sounds that are 90 degrees to the left or the right will exhibit the greatest interaural difference and sounds directly ahead of the listener will exhibit no interaural difference.

The human brain compares the difference in arrival times between the left and right ears and (with the help of other aural cues) can interpret the location of multiple audio sources with an impressive degree of accuracy – within 2 degrees!

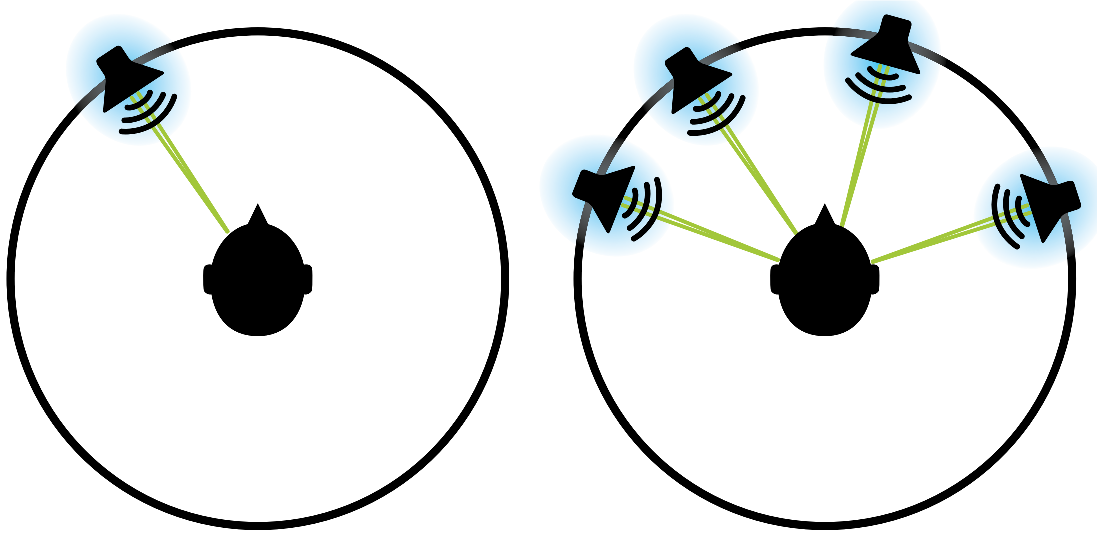

Figure 3.10: Sounds can be localized with an accuracy up to 2 degrees.

Our ability to discern position based on minute differences in the timing of the signal as it reaches our two ears has practical ramifications for sound reproduction. Inaccuracies in the reproduction of the timing of sounds will distort the spatial image experienced by the listener. Taken to extremes, these inaccuracies can cause flanging or chorusing effects, or even degrees of phase cancellation when dealing with multiple sound sources (for example, stereo speaker setups).

TIME AND SONIC DETAIL

"May I mambo dogface to the banana patch." **(Steve Martin)**

Our brains are conditioned by years of learning (and many more years of evolution) to expect certain things out of our environment. For example, in the sentence above, we can recognize the sentence structure, and the sentence is – though bizarre – grammatically correct. Similarly, our brains have evolved over time to "know" what overtones are associated with certain fundamental pitches, collectively creating ***timbre***. But what if (as is the case in the sentence above), some information is confused or incorrect? Well, at that point, our brains don't really know quite what to make of what we're hearing.

Time plays a critical role in the information we rely upon to understand our aural environment. Differences, not only in the timing of combined signals, but the timing of ***detailed nuances*** in the aggregate sonic input allow us to distinguish between overlapping and concurrent sounds (this detail, for example, allows us to discern between a violin and a trumpet playing the same note at the same time in an open field, while at the same time hearing the sound of the wind underneath). Much of this information is in the higher harmonic frequency ranges, (with shorter cycles), so the accurate timing of signals is especially important in hearing clearly the world around us.

The "Cut Bell" Experiments

One famous example of the importance of this detail (which will also be explored in the next lesson) was done in the 1950's by influential composer Pierre Schaeffer (1910-1995). Shaeffer is credited for leading the way as an avant-garde music composer, innovating with a style called ***musique concrete***, which in turn is one of the key origins of electronic music and sampling.

In the 1950s, Shaeffer hypothesized that our identification of instruments occurs primarily at the beginning of that sound – in the ***attack*** component of the ADSR (Attack, Decay, Sustain, Release) sonic envelope. He began by removing the attack of a bell (hence the name of the series of experiments), and had subjects attempt to identify the sound by the remaining decay, sustain, and release of the sound.

Here's what he learned: The subjects of his test could not easily identify that the sound that they were hearing was a bell without the crucial – and complex – information at the beginning of the sound. Shaeffer proceeded to cut the beginnings of many other instruments (the orchestral instruments of the day) and found similar results. Our ability to identify sounds primarily comes from the temporally and harmonically complex attack of a sound, and without that component being heard (or being heard accurately) we lose that ability to discern what we are hearing.

NOTE: These experiments were taken a step further by others splicing the attack of one instrument with the decay, sustain, and release of another, to create a new instrumental sound!

Here again, the temporal axis rises to often-overlooked prominence. Not only do we rely on our delicate temporal resolution to determine the source position of the sound, but we also depend on our ability to discern minute temporal variations of a sound wave as it plays to identify instruments and complex interactions of multiple sounds.

THE KV2 DIFFERENCE: TIME

If you look at many speaker manufacturer's product spec sheets, you'll quickly find information on frequency response and dynamic range. Those (important) parameters of sound are certainly well-represented in debates on the pros and cons of different speaker manufacture. What is underrepresented in the discussion is the temporal dimension.

There might be a reason behind this. Properly addressing the temporal dimensions of sound requires dedication, and a commitment to quality from the input of the speaker to its final output. This involves not only the mechanical aspects of speaker design and construction, but also the electronic components that drive them – there can be no weak link.

KV2 believes that this commitment is required to deliver on the promise of a premium speaker: To faithfully and completely reproduce the signal that is sent to it. KV2 further believes – and the science will prove this in the lessons that follow – that by addressing the temporal aspect of sound head-on, without compromise, that expensive problems that plague other speakers need not be solved ***because they don't occur in the first place***.

KV2 speakers will bring you sound as it was meant to be heard. And it's about time!

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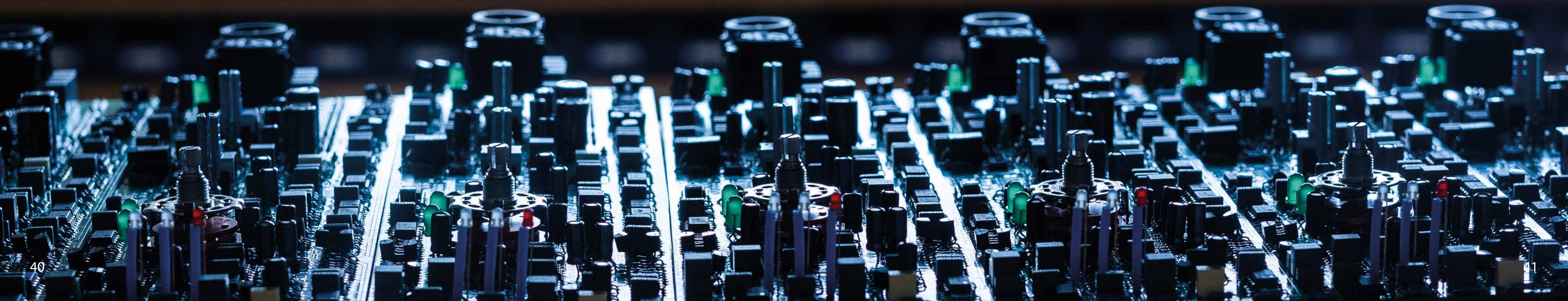
Module 4
SUPERIOR CIRCUITRY

Amplification is a critical component of any sound system, greatly impacting the power and clarity of the system's output. However, there are many fundamental aspects of amplification that go overlooked and underappreciated for the roles that they play. This lesson will explore in detail the amplification circuitry that drives the speaker.

In the previous lesson, we explored the temporal dimension of sound and how critical it is to our full enjoyment of the music that we hear. Slow, unresponsive circuitry creates sonic problems and a loss of clarity and definition. As part of their dedication to uncompromising specifications, KV2's use of ultra-fast circuitry and innovative amplification solves those problems and delivers unparalleled sonic quality.



Module 4.1
DISTORTION

In the audio world, the term "distortion" can mean many things, from the warmth added by a guitar amplifier's tubes to the destructive degradation that accompanies digital conversion. No matter the context, distortion can be fundamentally defined as a *change from the original signal*.

One rule that professionals are constantly mindful of is that distortion is a desirable change in sound ***only*** when it's ***intentional***. For example, adding saturation or a lush overdriven guitar - though certainly a change from the original signal - can enhance a mix when deliberately applied by an experienced mix engineer. On the other hand, when a signal is distorted ***unintentionally*** (including by means of inferior componentry, as will be discussed in this lesson), the result is an ***unwanted*** degradation in sound quality.

NOTE: With regard to speaker design, the goal is to transparently reproduce and amplify a signal without otherwise changing it – in other words, without introducing any distortion.

From a technical perspective, there are two kinds of distortion; ***Linear*** distortion is a simple change in amplitude or phase, with no new frequencies added. ***Non-Linear*** distortion results in a change in a signal's frequencies –and perhaps even the introduction of new frequencies - resulting in a change in sonic quality. This latter type – non-linear distortion – is what is generally referred to when the term "distortion" is used.

Distortion can come in a variety of forms with different causes and sonic effects. The two types which relate directly to amplification circuitry are ***harmonic distortion*** and ***non-harmonic*** distortion.

HARMONIC DISTORTION

The term "harmonics" refers to a specific behavior in nearly all the sounds that you hear. Put simply, a sound's harmonics (also referred to as "overtones" or "partials") are multiples of the lowest frequency of the sound (called the fundamental frequency). Together with the ***fundamental*** frequency, the sound's harmonic series gives it its tonal quality, or timbre.

A simple harmonic structure is **shown in Figure 4.1**:

The various harmonics have a specific relationship:

- The top row represents the 1st harmonic, also called the fundamental. This is the lowest frequency component of the sound.
- The second row shows a waveform that is twice the frequency of the fundamental, with a wavelength equaling 1/2 of the fundamental. This is called an "even harmonic".
- The waveform in the third row is three times the frequency of the fundamental, with a wavelength equaling 1/3rd of the fundamental. This is called an "odd harmonic".
- The fourth, fifth, and sixth rows show waveforms that are additional divisions of the fundamental (1/4th, 1/5th, and 1/6th respectively).As with the previous rows, even divisions (1/4th and 1/6th) are called "even harmonics", whereas odd divisions (1/5th) are "odd harmonics".

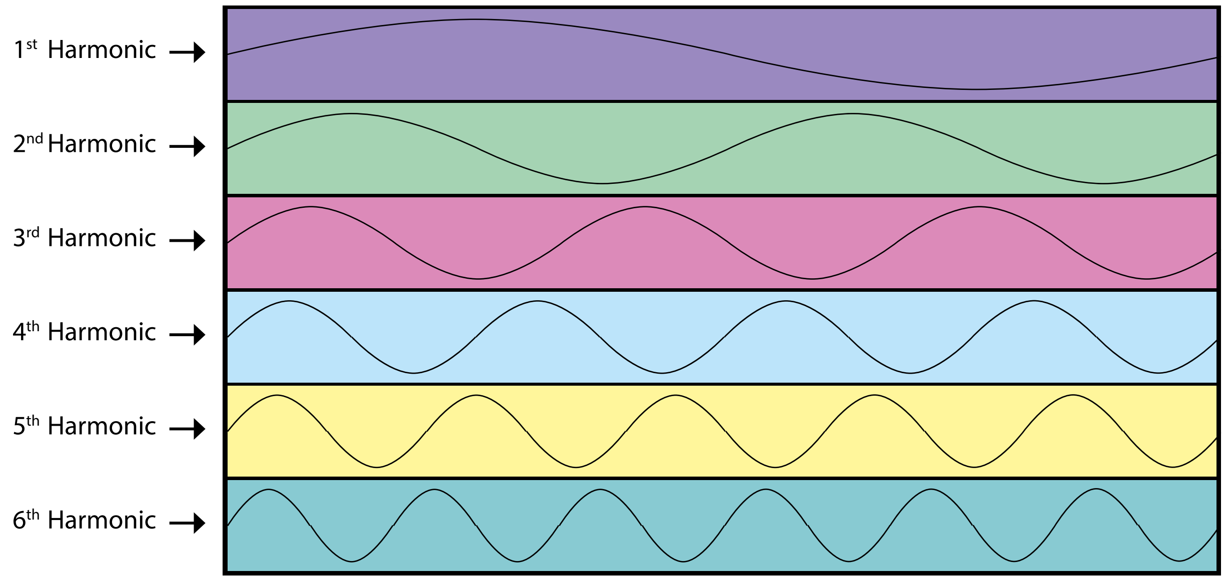


Figure 4.1: A simple harmonic series

All sounds are comprised of these sinusoidal harmonic elements, which together determine the timbre of that sound. For example, consider a tuba and a soprano saxophone; they both have harmonic content, but the relative strength of the individual harmonics is different, which in combination gives each instrument its distinctive tone. Sounds can be often differentiated by the strength of their even harmonics as compared to their odd harmonics.

NOTE: All the sounds that you hear have some degree of harmonic content, with one notable exception; a ***sine wave***, whose pure tone derives from the fact that a sine wave is simply a fundamental pitch with no even or odd harmonic content.

NOTE: Harmonics extend through the audible range into the inaudible, with the amplitude of harmonics usually progressively decreasing in amplitude.

When we use the term ***harmonic distortion***, we're referring to any process that alters the original harmonics of the incoming signal. Typically, this involves the boosting of existing harmonics but can occasionally involve the introduction of new harmonic tones.

Even Harmonic Distortion

Any process that alters the even harmonics (2nd, 4th, 6th, and so on)of a signal is collectively referred to as creating ***even harmonic distortion***. In production scenarios, boosting these harmonics tends to yield a more full, round sound. Tube amplification exhibits this kind of behavior and is sometimes specifically employed by musicians to "warm" the sound. For playback systems however, this kind of distortion – like all distortion – is unwanted.

From a technical perspective, ***even harmonic*** distortion results from disturbances or changes in one half-wavelength of the fundamental. Put another way, it results from changing one half of a cycle (compression or rarefaction) of the signal. In playback systems, even harmonic distortion can be created by high frequency compression drivers, due to high sound pressure within the device (2nd harmonic distortion is a function of acoustic pressure).

Odd Harmonic Distortion

Any process that alters the odd harmonics (3nd, 5th, 7th, and so on) is called ***odd harmonic distortion***. Increasing the amplitude of these harmonics tends to create a more aggressive, saturated sound. Recording to audio tape naturally boosts these odd harmonics, as do solid state amplifiers. In general, odd harmonic distortion results in a "grittier" (and often more nasal) change in the sound.

Odd harmonic distortion results from disturbances or changes in ***both*** half-wavelengths of the fundamental (in both the compression and rarefaction portions of each cycle). In playback systems, odd harmonic distortion is commonly caused by the non-linearity of amplifier circuitry and by overdriving an amp's output.

Total Harmonic Distortion

As the name suggests, ***Total Harmonic Distortion (THD)*** is the sum of all distortion at all harmonic levels. This is determined by comparing the relative amplitudes of the original harmonics from the source with the amplitudes of the corresponding harmonics at the output stage. Any discrepancy in these individual measurements is a harmonic distortion and contributes to this total.

Collectively, Total Harmonic Distortion relates to a system's reproduction of a signal and will alter the timbre of the output sound. The lowest possible THD is the goal for clear and accurate playback.

NON-HARMONIC DISTORTION

The term ***non-harmonic distortion*** refers to any alteration of a sound that creates frequencies that are ***not*** part of the original sound's harmonic series (i.e. that are not multiples of the sound's fundamental). Commonly referred to as ***Intermodulation Distortion (IMD)***, non-harmonic distortion is extraneous noise, and though it's occasionally used intentionally in production, it is completely undesirable in playback systems.

Intermodulation distortion occurs when the source signal is comprised of two or more frequencies in a non-linear system. The frequencies interact in a number of ways – for example, in a simple scenario (often used to test a system) two sine waves are generatedat two different frequencies (F_a and F_b). This initially results in what is called ***second-order*** intermodulation, and includes:

- $F_a + F_b = F_c$
- $F_a - F_b = F_d$
- $F_b - F_a = F_e$

The introduction of these three new frequencies (F_c, F_d, and F_e) would be concerning enough, but the distortion doesn't stop there. As the input to the system is increased, or when signals have harmonic content, multiple tones can also mix with each others' harmonics, creating 3rd-, 4th-, and 5th-order intermodulation and more. From just two sounds, dozens of new frequencies can be created. KV2 addresses this issue by using custom made output transformers combined with other unique circuitry.

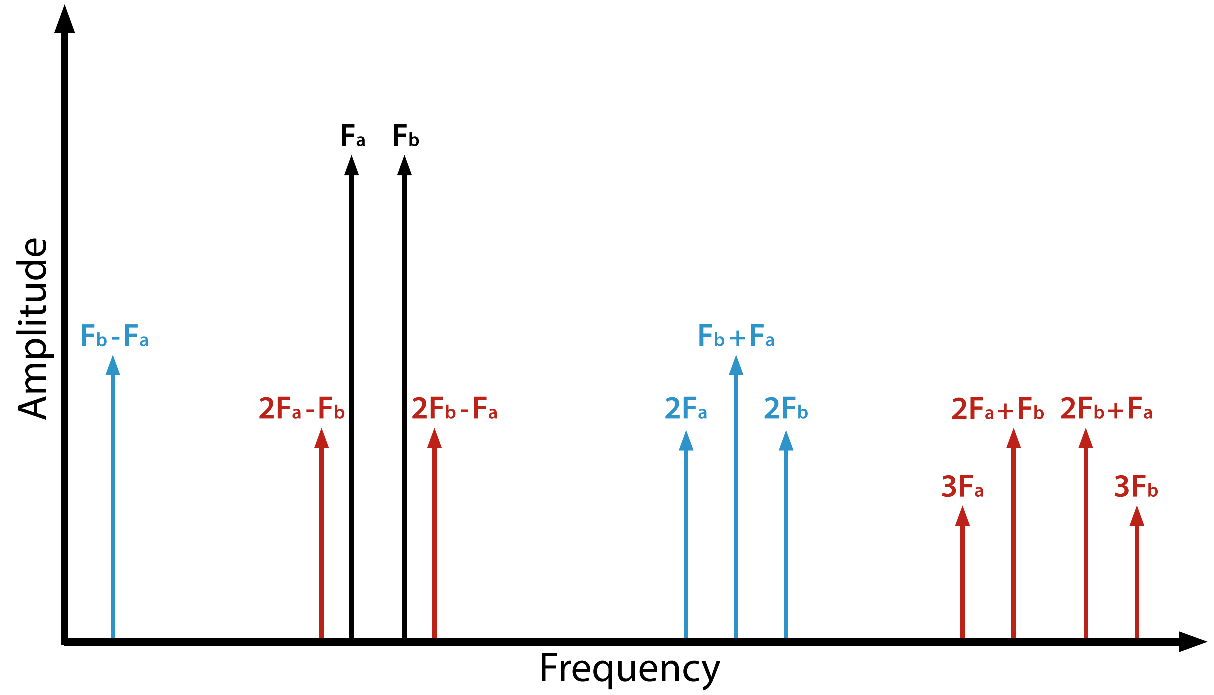


Figure 4.2: Two frequencies (F_a and F_b), with 2nd order IMD (blue) and 3rd order IMD (red).

NOTE: When a signal is amplified, IMD distortion increases in amplitude ***faster*** than the original signal. For example, for each 1dB of increase of the original signal, the 3rd-order distortion increases by 3dB! At some point, the amplitude of the distortion can meet – and then exceed – the amplitude of the original signal. This is called ***third-order intercept***. Thankfully this doesn't generally occur in the real world and is more of a theoretical method of determining a device's sonic quality.

In practical situations, Intermodulation distortion is often caused by the playback system due to the slow reaction of various components (including amplification and inadequate damping). much of this distortion naturally falls outside of the audible range, so a good portion is easily filtered out with no issues. However, some IMD – especially 3rd-order intermodulation – tends to fall very close to the fundamental frequency, well within the audible frequency range.This results inunwanted noise that is unrelated to the original signal,leading tounpredictable sonic behavior.

DISTORTION IN THE REAL WORLD

Though distortion might on occasion be a useful tone-shaping tool in the hands of a musician or producer, it is ***never*** desirable for high-fidelity audio playback. Measuring and recognizing distortion is essential to the live sound engineer.

Measuring Distortion

Total Harmonic Distortion (THD) and ***Total Harmonic Distortion plus Noise*** (THD+N) are the most common measurements for audio signals. They are identical, except for the ***+N***, which includes any noise that is created in addition to changed harmonics.

To measure THD+N, a signal (typically a sine wave) is introduced to the input of the device under test (DUT). Upon output, a notch filter is used to remove the input tone, and any signals that remain are reported as the THD+N measurement, representing the total harmonic distortion, plus any inherent noise of the DUT and other residual distortion.

Alternatively, Total Harmonic Distortion only (excluding noise) can be measured by comparing the levels of individual harmonics in an incoming source signal with those of the output signal. This is a measurementof harmonic distortion only, and excludes other distortion introduced by circuitry settling times (which will be discussed in the next section), loss of sonic information, power response and so on.

Distortion is usually expressed as a percentage of the output signal. Non-harmonic (IMD) and odd harmonic distortion becomes audible at 0.1% (equaling an amplitude of-60dB), with even harmonic distortion becoming audible at 1% (equaling an amplitude of -40db).

The Result of Distortion

Distortion can be introduced into a system in a number of manners, and from a variety of sources. No matter the origin of the distortion however, it impacts the listener's experience in a variety of ways, including:

- **Clarity:** The midrange and high frequency content of a sound gives us important information about the timbre and position of that sound. These frequency ranges necessarily have more complex waveforms with greater detail. It is precisely this detail that is most negatively impacted by the effects of distortion. This greatly reduces the clarity of reproduced sound in the very frequency ranges where we need it most.

- **Masking:** When distortion reaches audible percentages, any sonic information below the distortion level is completely lost in the output signal. Even with minimally audible IMD or harmonic distortion, this results in the complete removal of sonic information below -60dB and -40dB respectively. This results in the loss of detail and nuance in the signal presented to the audience.

- **Coverage:** Distortion reduces the clarity of a sound, but the damage doesn't stop there – these audible effects multiply as the listener's distance from the speaker increases. Conversely, lower-distortion (clearer and more accurate) signals will travel greater distances without a loss of clarity.

- **Tone:** Distortion can impact tonality in a number of ways. In particular, non-harmonic distortion often manifests as persistent high-frequency content, which is often mistaken for high frequencies in the original signal.

Since the majority of this distortion is happening at the amplification stage, any efforts made by the mix engineer to fix these problems will have minimal impact (as the problem is manifesting after the mixer's output). Some speaker manufacturers attempt to remedy this change in tone through the use of DSP processing, but this only makes the problem worse, as digital conversion and processing both introduce distortion.

NOTE: The problems introduced by traditional digital audio, and KV2's solution to the digital audio dilemma are discussed in Lesson 5 - ***The KV2 Difference: Digital Audio in this series.***

The best way to address these problems is to go straight to the source and improve the behavior of the components that could generate distortion in the first place. But how does circuitry create distortion and how can this problem be fixed?

Module 4.2
CIRCUITRY MATTERS

In the previous lesson, we discussed the often-overlooked temporal dimension of sound, and how critical it is to everything that we hear. This certainly applies to how sounds are reproduced, and the difference between mediocre and excellent audio reproduction often comes down to tiny increments of time.

A system's amplification plays an important temporal role, with the critical aspects of *slew rate* and *settling time* determining the clarity and accuracy of the final output.

NOTE: In an audio system, every link in the signal chain has the potential to add distortion, to varying degrees. This lesson focuses on the distortion introduced by electronic circuitry. The additional distortion that can be created by speaker transducers and horn design will be covered in Lesson 6 of this training series - ***The KV2 Difference: Superior Drivers.***

SLEW RATE

When an amplifier's circuitry reacts to an incoming signal, the change from low (or no) voltage to a higher voltage takes time. This is known as ***slew rate*** and is measured as the amount of time it takes for the amplifier's output to change from a low-threshold value (usually 10% of the incoming signal) to a high-threshold value (usually 90% of the incoming signal).

Figure 4.3 shows a simple example of slew rate, with the interval between the low (t10) and high (t90) thresholds highlighted in red. The slew rate (tr) is expressed in real-time units (usually microseconds):

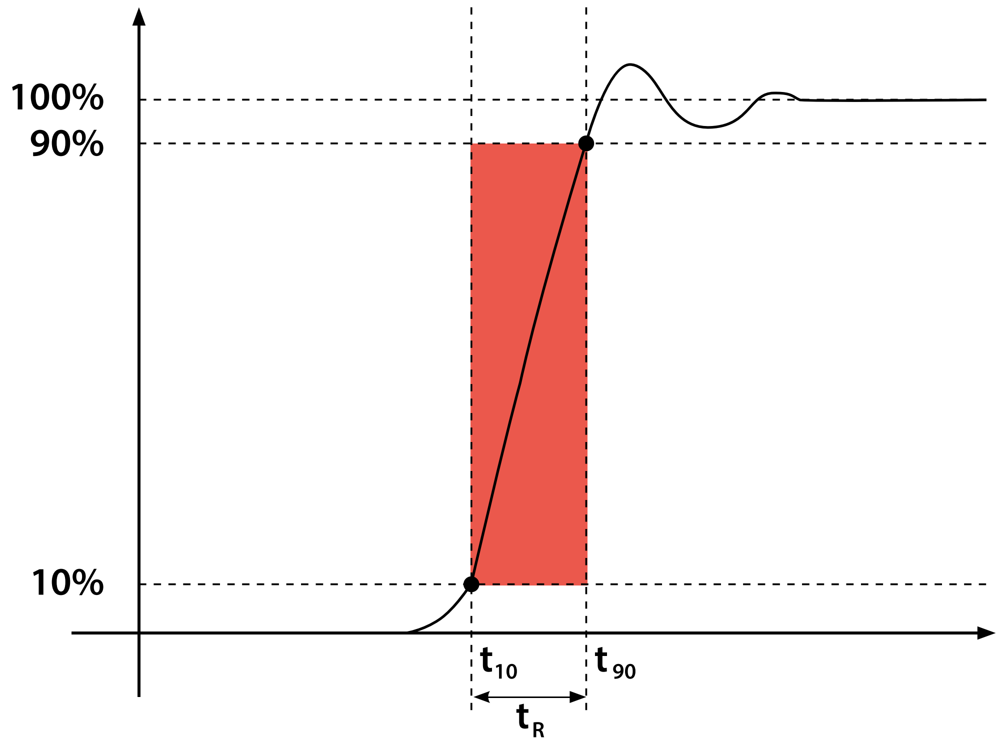


Figure 4.3: Slew rate (Tr) indicated in red.

A shorter slew rate indicates more responsive electronics, with more of the original sound's initial transient preserved. Additionally, a fast slew rate will provide the most accurate temporal placement of a sound, impacting the listener's sense of spatial location and stereo imagery.

Overshoot

Overshoot occurs when a circuit's output voltage exceeds 100% of the incoming signal, as **shown in figure 4.4**. This commonly occurs in band-limited systems (i.e. systems including a low-pass filter), and is generally unwanted – the less overshoot, the better. Because overshoot exceeds the incoming signal, it can create unexpected problems that manifest in the successive steps of the overall signal chain.

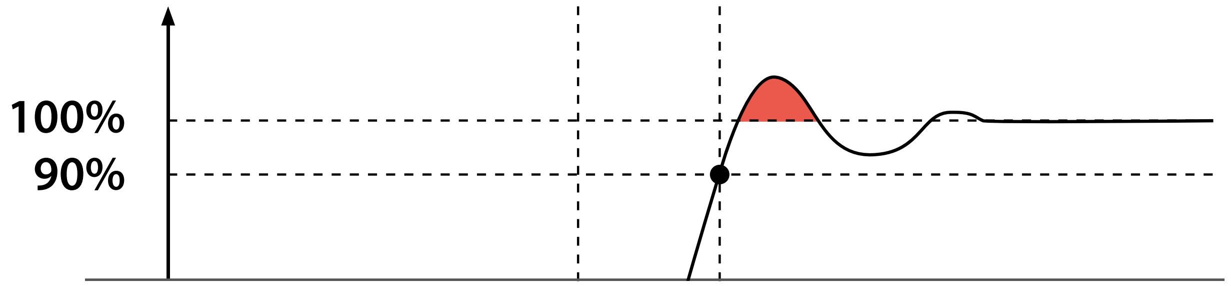


Figure 4.4: Overshoot indicated in red.

It's worth noting that while slew rate involves a reduction in sonic information (as the circuitry strives to keep up with the incoming signal), overshoot is the first point where we see amplification ***adding*** elements to the output signal that weren't present in the input.

Unfortunately, it's not the only way that circuitry can add unwanted audio to a signal…

SETTLING TIME

Once an amplifier's output has risen in response to an incoming signal (and overshot the mark to some extent), the amp's circuitry doesn't immediately attain the proper level. ***Settling time*** refers to the time required for the output level to match the same level as the original signal, within a certain range called the ***error band*** or ***tolerance***.

Figure 4.5 shows an example of settling time – after an amplifier's output rises from 10% of the incoming signal to the 90% level (slew rate) and then exceeds 100% (overshoot), the electronics will oscillate with decreasing amplitude for a period of time. Once this oscillation has calmed to the point where it doesn't exceed the error band (which is typically between 2% and 5% of the input signal amplitude), the settling is said to be completed. In KV2 amplifiers, this range is much more precise, measured at 1µs or .01% of the input signal.

Settling time is typically measured from the beginning of incoming signal to this final, relatively accurate state, with any fluctuations falling within this error band. Like slew rate, the goal for playback systems is to have the shortest settling time possible.

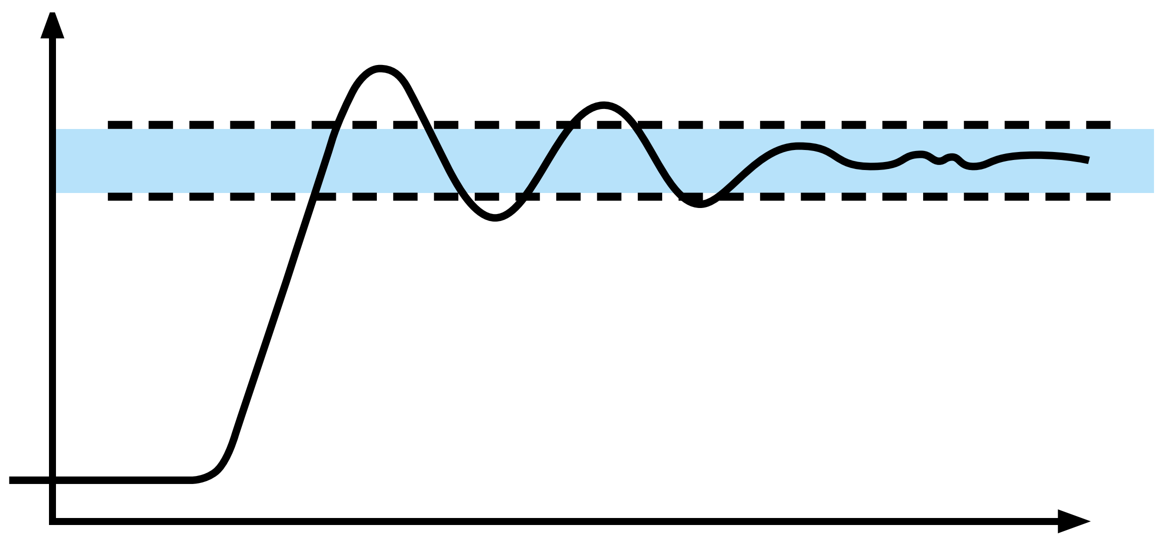


Figure 4.5: Settling time (error band highlighted in blue).

SLEW RATE, SETTLING TIME, AND DISTORTION

Both slew rate and settling time alter a signal, and so both contribute to distortion. Since the goal of a playback system is to minimize this distortion, the objective is to minimize the time taken for electronics to rise and then settle.

During the slew rate period, critical information is lost, as well as the accurate temporal placement of a sound. This can affect not only the initial transient of a sound, but also the overall stereo image of the total output, since the initial timing of a sound is critical to our ability to localize its position.

But slew rate, as important as it is, is only half of the story and if what happens after that initial rise isn't followed by a fast settling time, the definition of the sound is lost. Settling time, like overshoot involves the introduction of new audio frequencies not present in the original signal. Additionally, the distortion caused by settling time masks sonic information that was present in the original audio - lost information that cannot be recovered. In playback situations, this damage doesn't occur just once – because the level of signals is constantly changing, settling time is constantly recurring.

In the previous lesson, we discussed the outsized importance of the beginning of a sound (as determined by the famous "cut bell" experiments). Settling time clouds the beginning of a sound's waveform, interfering at exactly the point where human beings need the most accurate information. Higher-pitched sounds are more affected, with more cycles being impacted by the damaging effects of settling time. This particularly impacts sonic elements like breath, the high frequency detail of stringed instruments, and the upper harmonics of all sounds. As frequencies decrease, the effect of settling time, though still significant, becomes less audible.

Because settling time is frequency independent (meaning that it doesn't change depending upon the frequencies of the incoming signal), the non-harmonic noise created by settling time has no relation to the harmonic structure of the original audio. This distortion gives a bright, "fizzy" high end, and because this brittle high frequency content is being created by the amplifier, no amount of equalization done by the sound engineer can impact this unwanted sound.

The distortion caused by slow electronics is not commonly discussed by manufacturers, but avoiding the issue doesn't diminish its impact. For a live sound engineer, the best that can be done is to try to minimize the symptoms of long rise and settling times, but true audio fidelity can only be possible when the root problem is solved…

Module 4.3
KV2'S SUPER-FAST CIRCUITRY

Thus far, we've delved into the nature of distortion and the role that circuitry plays in how extensive (and destructive) that distortion will be. In this final section, we will explore the flaws in conventional thinking regarding amplification, and how KV2 breaks new ground in delivering pristine audio playback.

THE PROBLEM WITH TRADITONAL DISTORTION MEASUREMENT

In a perfect world, distortion is measured by simply comparing the output of a signal with the original input. In the studio, this is easily accomplished by inverting the phase of one of the signals and summing the two together. Given that identical signals in this scenario would phase-cancel each other completely (resulting in a "silent" output), any remaining audio would represent the difference from the original signal – in other words, the distortion (additive or subtractive change).

However, in the live sound world, this method is rarely employed. Previously in this lesson, we discussed the two conventional methods of measuring distortion – using a static tone or measuring harmonic discrepancies. To obtain these values, most of the audio industry uses digital high-precision FFT (fast Fourier transform) devices to measure a signal and then extract a variety of information, including frequency and time, before processing it internally to calculate the THD.

Being a digital device, an FFT measuring instrument relies on analysing and calculating based upon a sample or a number of samples taken from the incoming signal. As you will learn in Lesson 5 of this series – standard digital formats, even at high sample rates, omit critical sonic information, and worse – they introduce errors and distortion of their own. This makes arriving at an accurate measurement in this way absolutely impossible.

As a result nearly the entire audio electronics industry has accepted that they cannot measure the distortion that they are unmistakably hearing, but simply adopt a common practice of producing and publishing measurements and calculations seriously flawed and overrun by the distortion created within the measuring device itself!

Conveniently from a marketing perspective, this tends to result in lower distortion measurements, which of course looks attractive on a spec sheet, offering a small number to compare their products against other manufacturers. It is, however, such an incomplete measurement that it does little to show the true performance of a system and explains why we can all here distortion on a CD but on paper that distortion is almost non-existent.

In order to deliver faithful audio playback, KV2 first needed to revolutionize distortion measurement itself by finding a way of measuring, and comparing, audio at a time resolution equivalent to (or as close as possible to) human hearing.

CORRECT MEASUREMENT - KV2'S TRUE TOTAL DISTORTION

When it comes to distortion, KV2's attention to the temporal dimension is key. In Lesson 3 of this series (The KV2 Difference: Time), you learned that 5 microseconds is the commonly accepted figure for temporal resolution (the smallest perceptible difference in timing) for most humans. This presents a challenge – the traditional digital formats (that are used by the majority of FFT devices) fall far short of being able to capture this level of sonic detail.

The digital sample rate required to process a signal to this degree of accuracy would be well beyond any of the traditional digital formats (even 192kHz sample rate). Further, to achieve this bandwidth, the device's delay line would need to be capable of at least five times higher resolution than the temporal resolution threshold, meaning that accuracy down to 2 microseconds or less is required in order to not adversely affect the output.

However, this isn't a problem for a KV2 system – as you will learn in Lesson 5 of this series (The KV2 Difference: Digital) KV2 has revolutionized digital audio with ultra-high PDM (Pulse Density Modulation), featuring a sample rate of over 20 MHz! With a precisely delayed signal utilising this 20MHz format, an inverted copy of the output can be created and compared with the incoming signal.

This in turn allows for differences between the source signal and the output to be measured down to the smallest audible sonic detail – a measurement of ***True Total Distortion***. KV2 meticulously aligns the two signals, matches the amplitude, and inverts the phase of one of the signals. By measuring the result of the combined signals, ***True Total Distortion reveals*** all of the nonlinearity in a system, including settling time and any other changes or noises that are introduced through the electronics design and componentry. To accomplish this comparison, KV2 employs best-in-class hardware, capable of filtering the input from 1KHz to 100KHz (these specifications are based upon the thresholds of human hearing, which are discussed in detail in Lesson 3 of this series).

Combining the world's best reference measuring instruments, revolutionary digital technology, and a focus on the critical temporal dimension of sound (beyond just frequency response) KV2 has achieved a true visualization of the complete input compared with the complete output to the minutest detail, revealing everything that has been added or removed by the amplifier.

NOTE: In the case of a perfect amplifier (one completely without distortion, a practical impossibility), the measurement of True Total Distortion would yield distortion results of zero.

Testing many of today's common electronics and amplification systems using ***True Total Distortion*** as a measuring gauge (rather than traditional THD methods) reveal actual distortion figures above 10% and often approaching 30%! This compares with manufacturers' own FFT calculated THD figures of perhaps 0.05% or less – a difference of up to 400%.

These new measurements are revelatory, bringing into question some of the biggest decisions made and directions taken in audio electronics design over the last 40 years. This possibly explains the lack of appetite by some to pursue this more accurate approach to measurement. To KV2, however, this is precisely the information needed to design a system with the lowest possible distortion. In a very real sense, KV2 has transformed the way that we look at electronic performance by first revolutionizing the way we measure distortion itself!

Guidelines for measuring True Total Distortion

KV2's innovative technology, allowing for the true measurement of distortion, gives not only the most accurate results possible, but uses a simpler and more straightforward method than the convoluted calculations commonly used in the live audio industry. That said, there are some considerations to bear in mind:

- It is critical that the levels at the summing point are precisely the same.
- The measurement must be made with a pulse signal (i.e. pink noise or music). A periodic signal (commonly used for FFT measurements) should not be used, as it will not provide complete information.
- True Total Distortion can be measured with real music, pink noise or any signal representing a complex dynamic signal with changes of frequency and level. Pink noise is particularly suitable as a measuring signal as it contains all the audible frequencies and is very close to music in terms of dynamic changes and spectral distribution.

NOTE: Standard measurement software and hardware tools cannot achieve the detail required to measure True Total Distortion.

KV2'S HIGH-PERFORMANCE CIRCUITRY

After resolving the issue of proper measurement, KV2 got to the business of building the most transparent, lowest distortion system in the industry. Armed with their unique ability to quantify electronic performance, KV2 has incorporated circuitry with ultra-fast rise and settling times, ensuring that there is no perceptible damage done to the signal. Measuring with True Total Distortion, KV2's systems' distortion is measured at 0.1% - roughly 100 times more precise than standard amplifier circuitry.

This is a clear example of KV2's dedication to the important (but often overlooked) temporal dimension of sound. KV2's entire system – electronics included – is built with extremely accurate timing throughout. Tolerances in KV2 speakers are within 1 microsecond, ensuring that any distortion, though still technically present, is well below the level of human perception.

The benefits of fast circuitry

KV2's dedication to an exceptionally responsive system – including fast electronics – yields a variety of sonic benefits:

- **Clarity:** Slow settling time introduces very high levels of distortion, and the persistent high frequencies created have a profound effect on sound quality, resulting in a 'smeared' top-end and listener fatigue. With KV2's ultra-fast circuitry, the amplifier effectively "gets out of the way" of the original signal quickly, allowing critical high frequencies and transients to be reproduced faithfully.

- **Coverage:** Sound will naturally degrade over distance – something that is exacerbated when the originally output sound is distorted. By drastically reducing distortion, KV2 speakers can be clearly heard over greater distances.

- **Tonal Linearity:** With many amplifiers, as gain is increased sonic character drastically changes. This can cause massive problems in a live sound environment and is a direct result of the system's distortion. KV2 systems, having effectively removed amplifier distortion, will give almost no change in tonality, timbre, and dynamic response from very low levels to maximum amplification. This is also due to careful gain matching to ensure that every amplifier in a multi band system will clip at exactly the same time, delivering an extremely linear tonal and dynamic output at any volume.

- **Dynamic Range:** Fast circuitry can not only get loud quickly but can get quiet just as rapidly. This gives a true representation of the dynamic range of the incoming signal. Slower systems, on the other hand, tend to compress the dynamic range – not only by losing the impact of signal peaks, but also failing to reproduce lower-amplitude sonic nuance.

Module 4.4
KV2 SYSTEM AMPLIFICATION

Too often, amplification choices are based on cost or convenience, compromising on sound quality. With KV2 systems however, amplifiers are designed from the ground up, delivering premium performance within the overall system.

THE RIGHT CLASS FOR THE JOB

Amplifier topologies come in a number of classes, the primary ones being:

- **Class A:** Notable for its low overall distortion and (no crossover distortion). This is a relatively simple design, with a single device that conducts through the entire 360° cycle of the waveform signal.

- **Class B:** With this kind of amplification system, half of the components are conducting the first half (+180°) of the waveform, and the other half of the components are conducting the second half (-180°) of the waveform. The two halves of the device work together in a push/pull handoff between the various components. This type of amplifier, despite its efficiency with power, has an obvious disadvantage arising from the crossover (push/pull) distortion inherent in the design.

- **Class A/B:** This is more similar to a Class B amplifier than a Class A, in that there are two components that work together in a push/pull architecture (with the accompanying crossover distortion). However, with a Class A/B amplifier, the individual components conduct more than half of the incoming signal, so that the crossover distortion is minimized.

- **Class D:** This is the most efficient kind of amplifier, and the most lightweight as well. This efficiency is attained through a complex switching mechanism using a pulse width or pulse density modulator. However, frequency reproduction is limited by the frequency of the modulations, and sound quality can vary based on speaker load.

- **Classes G+H:** These amplifiers are a variation on the A/B class, with less crossover distortion than class B. These amplifiers use voltage rail switching and modulation to deliver impressive efficiency in a relatively small physical package. These classes are generally more expensive than their class A/B counterparts.

KV2 Amp Topology Structure

Manufacturers frequently compromise on sound quality in order to gain convenience and efficiency.

With KV2, every facet of the system's design has been aimed at superb audio reproduction, and their choices for amplification are no exception to that driving philosophy:

- To deliver quality in the high frequency range, KV2 uses mainly Class AB Push-Pull, transformer balanced output amplifiers. The lack of crossover distortion and fast rise/settling time make this the right choice for this critical sonic range.

- For mid-frequencies, KV2 again uses Class A/B amplification. When Class A/B amps are used, KV2 employs Mosfet output devices for fast recovery time and low crossover distortion. The amplifier's output transformer controls the output signal to reduce IM distortion.

- For low frequencies, what is needed most is high current, due to the damping factor needed for large speakers. Standard amplifiers cannot accomplish this, so KV2 has developed a new amplifier topology focused on delivering high current while achieving over 90% efficiency (to minimize cooling requirements and increase reliability). The design features a switching mode power supply with modulated voltage rails that maintains a low voltage across the output devices. This topology, called Class G, is capable of providing much higher current and better damping factor characteristics than standard Class H designs. The Class G amplifiers used by KV2 feature a linear output filter to remove the distortion caused by switching, while still providing high efficiency.

IMPEDANCE MATCHING

NOTE: Damping factor – the ability to stop a speaker from vibrating – is often overlooked but is at least as important as the power needed to put the speaker in motion in the first place. It is a measure of how effectively the amplifier can dampen or control the speaker's oscillations after it receives a signal. A high damping factor means that the amplifier can more effectively control the movement of the speaker, resulting in tighter and more accurate bass response. In contrast, a low damping factor means that the amplifier may not be able to control the speaker's movements as effectively, leading to a looser and less controlled bass response. Overall, a high damping factor is desirable in an amplifier, particularly for speakers with a low resonant frequency, as it helps to prevent unwanted resonances and ringing in the speaker's frequency response.

When we talk about impedance matching, we're referring to designing a signal source (in this case an amplifier's op-amps) to match the electrical load of a cable. While this seems fairly straightforward, the reason why this is important is a much deeper topic: Proper impedance matching can maximize power transfer and minimize internal signal reflection in the line – something that in itself can create distortion.

This distortion is created as a signal is reflected within a cable due to incorrect loading of the signal by the amplifier, creating destructive interference. Unfortunately, this is ignored by many manufacturers, despite the fact that this kind of distortion can be quite audible and threatens to undo any benefit of high-quality components elsewhere in the system.

To ensure that there are no weak links in the signal chain, KV2 has given users extensive impedance matching selection options and also incorporates Line Drivers in their hardware, as another way to address this common issue.

NO WEAK LINKS

One of the challenges in designing a system aimed at sonic excellence is that one compromise in any part of the system limits the benefits of all of the other components. In the case of amplification, when compromises are made to the circuitry driving the signal, this mitigates other quality components in the overall system.

KV2 is ever-mindful of this fact, and maintains the highest standards and commitment to faithful sound reproduction. This is applied to all components, including circuity and amplification, giving their speakers (which will be explored in lesson 6) the signal that they need to do their job – delivering pristine audio to the listening audience.

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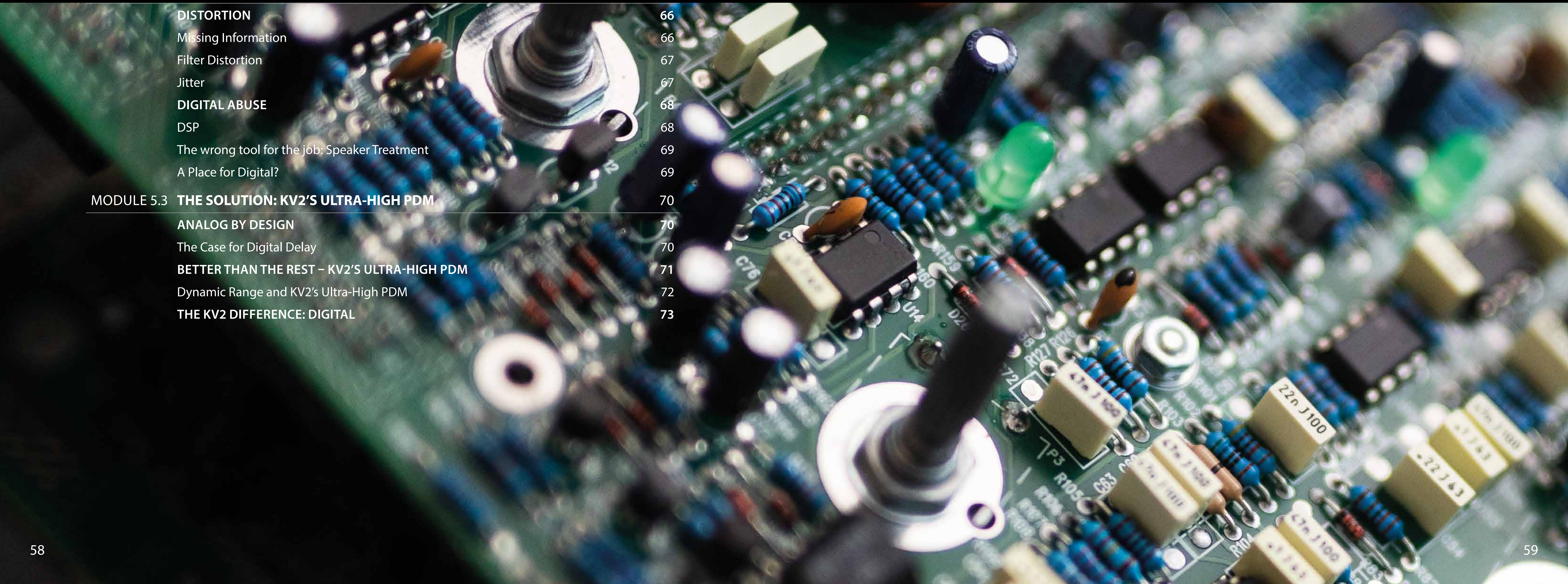
Module 5
DIGITAL

From its introduction in the early 1970s, digital audio has transformed the audio landscape. Of course, this transformation hasn't been without its obstacles, and there are both advantages and disadvantages to the more commonly used digital audio formats.

In this lesson, we'll discuss the different flavors of digital audio and how they are used in both studio recordings and live sound. You'll then learn how KV2 has addressed the inherent limitations of traditional digital audio and surmounted these problems with their own ultra-high Pulse Density Modulation (PDM), delivering on all the advantages of digital audio (when needed) with none of the disadvantages!



Module 5.1
DIGITAL BASICS

There's no doubt that digital audio is ubiquitous – in our daily lives, in recording studios, and even in live sound. But what is it, and just how – and why – is it used?

A BRIEF HISTORY OF DIGITAL AUDIO

Digital Audio in Recording Studios

Before digital audio came on the scene, audio was recorded on magnetic analog audio tape. In the recording studio, it was common to see 2-inch wide tape being used on massive machines that could record up to 24 tracks. Both the tape recorders and the tape itself were very expensive. Even worse, if the producer wanted to make an edit, a razor blade was used to physically cut the tape. Any mistake made when the tape was cut was permanent!

Figure 5.1: Analog tape editing tools!

Digital audio started small–brief chunks of recorded sound (like drum hits, often called drum samples) began to gain popularity in the mid-1980s. The next evolutionary step in digital audio was the move from recording single brief segments of sound to recording longer multiple audio signals. These early digital recorders have evolved into comprehensive production platforms, collectively known as ***Digital Audio Workstations***, or ***DAWs***. There are a number of DAWs currently on the scene, featuring different workflows for different styles of music or post-production scenarios.

Digital audio now plays a role in nearly all production studios at any level, though it's implemented and used to different extents in different kinds of facilities. The primary reasons for its popularity are ***cost*** and ***ease of use***. The reduction of equipment costs and the increased editing and processing power that came with the introduction of digital audio led to wide adoption among professional recording studios, despite significant concessions to sound quality and fidelity.

Digital Audio in Live Sound

Digital audio can also be found in the world of live sound and performs a number of roles when it's employed. For example, digital mixing consoles have found a place on the professional landscape, though it's worth mentioning that digital mixing is less common in this context than in the professional recording studio community. Though generally more expensive, digital live consoles boast the kind of small footprint and flexibility you'd expect from a digital device.

Digital audio isn't only used in consoles though. You'll also find digital audio used to align multiple speakers (either within a single enclosure or aligning multiple enclosures). Digital audio is used to facilitate other processes, like Equalization, that are aimed at compensating for problems encountered in the field or inherent speaker weaknesses. This invariably leads to other problems, which will be discussed in detail in the next section.

Digital Audio in "the Real World"

Beyond the inherent pros and cons of digital audio in the professional studio and live sound communities, digital audio has for many years been the dominant consumer format. The reason for this, unsurprisingly, is ***cost*** and ***convenience***. Digital audio is inexpensive to store and transport, and digital audio files can be distributed easily through a variety of means. Just as in the professional world, these advantages come at the cost of sonic clarity, and much of the audio that we hear in our daily lives has been badly damaged by digitization and data compression.

In the consumer's world, there are a number of digital audio formats that are in common use. All of these formats, from MP3 to Ogg Vorbis and everywhere in between, are based on either ***Pulse Code Modulation*** or ***Pulse Density Modulation…***

PULSE CODE MODULATION

Most of the audio that you hear in your daily life – probably ***all*** of it on most days - falls into the category of ***Pulse Code Modulation***, or ***PCM*** for short. Think of this format as a string of specific measurements played together in rapid succession, to create an audio signal. Each discrete measurement is called a ***sample***, so let's take a look at those first.

Samples and Sample Rates

The term ***sample*** is used in a number of ways in the audio world. When we use this term in the context of PCM digital audio, a sample can be defined as an instantaneous measurement of an audio signal. A sample measures one thing only, and that is the ***amplitude*** of the audio signal at a specific moment in time.

A single sample alone isn't enough information for us to record or reproduce a sound. What's needed is a number of samples, spaced evenly in time in order to re-create a sound properly. In **Figure 5.2, you can see** a sine wave with each sample marked. Individually, a sample represents only an amplitude value, but together they can represent a complete waveform.

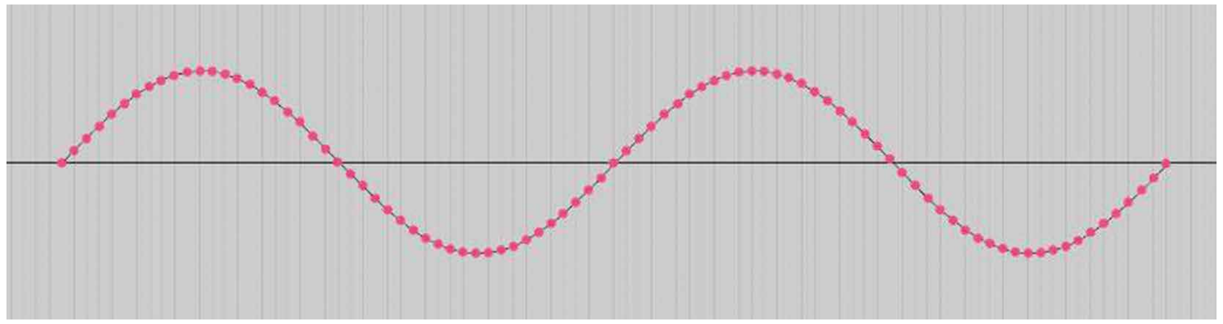


Figure 5.2: Sample points on a sine wave.

The number of samples used per second is called the *sample rate*. Sample rates are measured in hertz (Hz) and kilohertz (kHz). Some of the different sample rates that are popularly used are:

- 44.1 kHz (44,100 samples per second)
- 48 kHz (48,000 samples per second)
- 88.2 kHz (88,200 samples per second)
- 96 kHz (96,000 samples per second)
- 176.4 kHz (176,400 samples per second)
- 192 kHz (192,000 samples per second)

But what is the best sample rate to use? If you go on the Internet and search this topic, you'll find a wide range of opinions. Let's talk a little bit about the math behind sample rates to find the real answer.

Sample rate theory finds its roots with a mathematician named ***Harry Nyquist***, one of the fathers of digital audio and the inventor of a principal called the Nyquist theorem. Put simply, the ***Nyquist theorem*** states that in order for digital audio to record or reproduce a sound, the sample rate must be at least twice the highest frequency of that sound. In other words, you need at least one sample in each compression phase, and one sample in each rarefaction phase.

If you do the math, considering that the highest frequency that a human can hear is 20 kHz, and we need at least two samples per cycle, then a 44.1 kHz sample rate should be more than enough, right? Well, there's one more piece to this puzzle: When sound is recorded digitally, there is a small amount of distortion in the frequencies very near the Nyquist frequency (the frequency that is ½ of the sample rate, and the highest frequency that can be accurately recorded). In some cases, this high-frequency distortion can be audible–especially with sounds that have a good deal of high frequency content, like cymbals. For that reason, many users will choose higher sample rates in order to preserve the clarity of higher frequencies.

Bit Depths

Each sample comprises a number of bits – this is known as the digital audio's ***bit-depth***. These bits are the digits each sample uses to store its measurement of amplitude. There are a number of different bit-depths that are used in modern digital audio, with the most-commonly used being 16-bit and 24-bit.

Just as sample rates relate to frequency and time, bit-depths relate to amplitude and dynamic range. If you take a look at figure 5.3, you'll see how this works: In this example, the amplitude measurement of each sample is rounded to the nearest vertical step. This rounding will introduce measurement errors (called ***quantization error***), which will in turn introduce inaccuracies as the sound is recorded and result in a distorted reconstructed sound on playback.

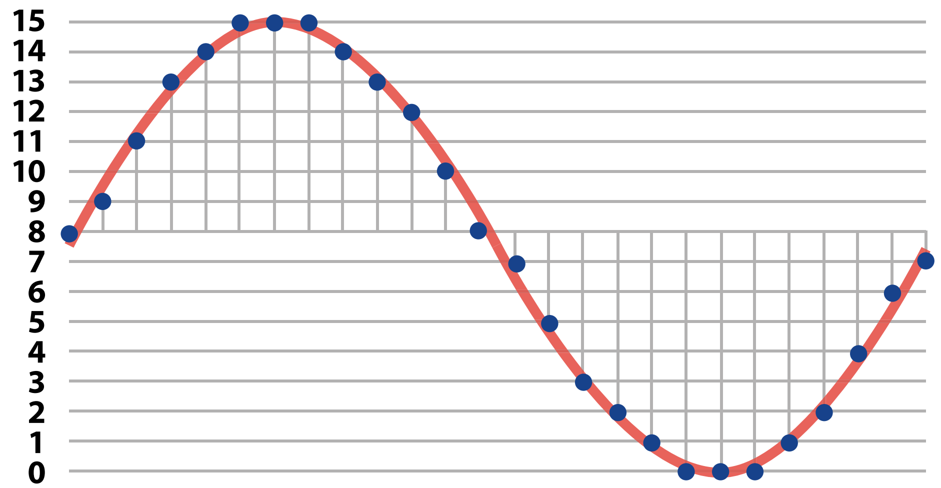


Figure 5.3: Amplitude measurements with 4-bit samples.

For simplicity's sake this example shows only 16 discreet measurement increments, which would correspond to 4-bit PCM digital audio (2 to the power of 4, commonly represented as 2^4). 16-bit PCM has 65,536 discrete values possible (2^{16}), and 24-bit has 16,777,216 discrete values (2^{24}).

Because of the nature of binary math, each bit added to each sample doubles the maximum values possible, and cuts quantization error in half in relation to the total audio signal. Regardless of the bit depth used however, there is always some degree of quantization error, and therefore some degree of distortion from the original audio signal.

There's more useful – and thankfully simple - math relating to bit-depth: Each bit of a sample will yield roughly 6 dB of dynamic range. If you consider 16-bit audio, that works out to a dynamic range of 96 dB (16x6=96). That's certainly a decent dynamic range, and significantly exceeds analog recording media like tape, but it doesn't quite take care of the full range of human hearing (the human dynamic range is 120 dB). 24-bit audio covers the audible dynamic range and more, with a dynamic range of 144 dB (24x6=144).

PCM Pros and Cons

The primary advantages of Pulse Code Modulation as a digital audio format are cost and ***ease of use*** in calculations for processing. For these reasons, PCM has been widely adopted ***despite*** significant concessions to the quality of the audio being recorded and reproduced. PCM's inherent sonic disadvantages, that negatively impact the listener experience, will be discussed later in this lesson.

PULSE DENSITY MODULATION

Although Pulse Code Modulation is the most commonly used digital audio format, it's not the only one: There's also ***Pulse Density Modulation***, or ***PDM***. This alternative format, though less well-known than Pulse Code Modulation, presents a solution for high quality digital audio by reimagining the ways in which samples and bits are used.

A Sense of Direction

Though the Pulse Density Modulation format does have a sample rate and a bit depth, the way that they are used is radically different from the more common Pulse Code Modulation. PDM audio features sample rates of 2.8224 MHz and higher - far greater than even the highest sample rates of PCM - and a bit depth of only 1 bit, so clearly the data are being used in a fundamentally different way than most other digital audio.

In the case of Pulse ***Density*** Modulation, instead of each sample being a discrete measurement of voltage, PDM is a very fast stream of bits indicating the shape of the waveform. A digital "1" is considered a "pulse", and a "0" is the absence of a pulse. Greater concentrations of pulses indicate positive voltage and greater concentrations of the absence of pulses indicates negative voltage. Silence is an alternating 1 and 0.

Though it's not a completely accurate analogy, you can think each of these 1s and 0s as being directional indicators, as opposed to being concrete values (as would be the case with PCM).

Pulses (digital 1s) generally point up and absences (digital 0s) generally point down. Since each "sample" is only 1-bit, there is plenty of bandwidth and processing power for extremely high sample rates, and therefore a high degree of temporal accuracy. Put into context, the highest sample rate for standard PCM is usually 192 kHz (192,000 samples per second), but the ***lowest*** standard PDM sample rate (for Super Audio CDs) is 2.8224 ***MHz*** - that's 2,822,400 samples per second! This gives Pulse Density Modulation the ability to capture and reproduce sound in more intricate detail.

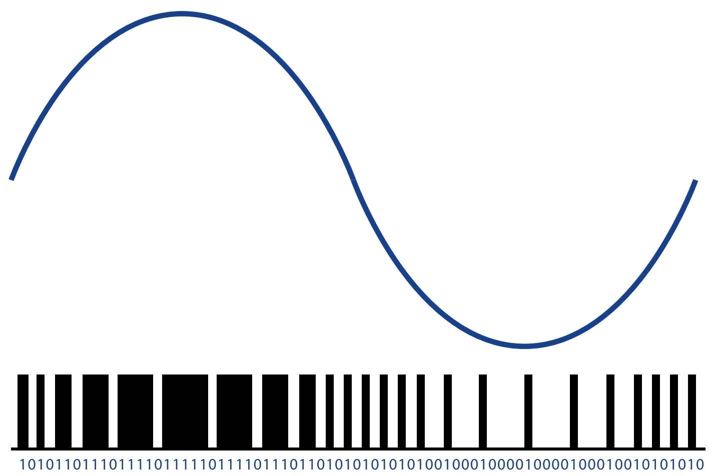


Figure 5.4: A Pulse Density Modulation DataStream and resultant waveform.

A Brief History of Pulse Density Modulation

The most notable example of Pulse Density Modulation in the commercial arena was Sony's ***Super Audio CD*** (SACD), which was based on technology called ***Direct Stream Digital*** (DSD). SACD featured a sample rate of 2.8224 MHz, and a bit-depth of 1 bit per sample. It was hailed as a high-fidelity format, delivering nuanced audio, especially with regard to transients. In listening tests, SACD has been found to be roughly equivalent to 192kHz PCM.

NOTE: One notable aspect of Pulse Density Modulation is that it is not free of audio artifacts, but the majority of these are well above the audible range and are eliminated with a simple low-pass filter.

Ultimately, PDM audio (along with the SACD format) did not thrive despite obvious sonic advantages. Pulse Density Modulation proved to be significantly more difficult to edit and mix than Pulse Code Modulation, which led to a lack of adoption by the production community. This had a knock-on effect with studio hardware and software manufacturers, and then on to consumer-level playback devices. However, the fundamental technology of PDM was adopted to an extent, as ***Delta-Sigma Modulation***, found in many professional audio interfaces. With Delta-Sigma Modulation, audio is initially captured as PDM, and then converted to PCM through a mechanism known as a "decimation filter".

There is one segment of the audio production community where PDM has found a natural home: audio mastering. In typical ***mastering*** studios, editing and extensive processing is rarely needed, and high-end mastering engineers sparingly use dedicated tools to tease out the highest quality from a musical track. DAWs like ***Pyramix***, that support Pulse Density Modulation, though not adopted in most recording studios, can be found in premium mastering facilities.

Module 5.2
THE PROBLEM(S) WITH DIGITAL

The twin worlds of studio production and live sound are vastly different, perhaps in no way more fundamental than this: Studio production is primarily engaged in recording audio which can be replayed at any time in the future, and live sound focuses on sound reinforcement in real time - recording is an optional and separate matter.

In a production studio, audio must be recorded (and stored) on some sort of media. There is no perfect medium – tape and digital storage both have their pros and cons, and a legitimate discussion can be had on which route to choose.

For the live sound engineer, the imperative is crystal clear: To faithfully receive and reproduce an audio signal without compromise. In this context, the digital vs. analog question is much more easily answered – *traditional digital conversion of an analog signal introduces errors and degrades signal quality, and should be avoided.*

DISTORTION

The term "distortion" can have many meanings in the audio world, from the warmth that comes from tape saturation and amplifier tubes to the unrecognizable destruction of a sound through extreme digital clipping. For the purposes of this discussion, distortion will be defined as any unwanted change from the original signal. Digital conversion introduces distortion in a number of ways, including ***missing information, filter*** distortion, and ***jitter*** chief among them.

Missing Information

One big drawback with the PCM audio format relates to the ***temporal*** dimension of sound that was discussed in Lesson 3 ("The KV2 Difference: Time"). Given that human beings have a temporal resolution as precise as 5 microseconds, commonly used sample rates such as 44.1 kHz or even 96 kHz capture only a fraction of the sonic detail that we can perceive. This results not only in a lack of important sonic detail, but also in an inaccurate stereo image compared to the original audio performance.

You'll _____ _____ the _____ _____ by _______ ____ part __ ___ story.

Imagine that you could only communicate by speaking just 1 out of every 3 words that you wanted to say, as shown in the sentence above. Of course, nobody could understand what you're talking about, since our brains need more information to work with. The same goes with sound, and with popular PCM sample rates (especially 44.1 kHz and 48 kHz) you're actually missing much ***more*** information than was stripped from the sentence! This inadequate detail translates to a lack of clarity and intelligibility. Higher (and much less commonly used) sample rates like 176.4 kHz and 192 kHz improve matters somewhat, as does Sony's Direct Stream Digital PDM, but even with these, important information is lost.

You'll never learn the whole truth by hearing only part of the story.

Less perceptible, but also worth mentioning, is information lost through quantization error, the rounding of amplitude levels which occurs during the Analog-to-Digital (or A/D) conversion process. This rounding error results in a subtly distorted waveform, damage further contributing to a lack of clarity and sonic accuracy.

Filter Distortion

As discussed earlier in this lesson, when sound is recorded to PCM digital audio, there is a low-pass filter that is applied prior to the A/D conversion process. The purpose of this filter – called the ***anti-aliasing filter*** - is to remove any frequencies above the Nyquist frequency (the frequency that is ½ of the sample rate, and the highest frequency that can be recorded). This filter creates a significant amount of distortion in the frequencies very near this Nyquist frequency. In some cases, particularly when using lower sample rates, this distortion can be audible - especially with sounds that have abundant high frequency content, like cymbals. For the same reason, the output of a DAC requires a low-pass analog filter, called a reconstruction filter - because the output signal must be bandlimited, to prevent imaging (meaning Fourier coefficients being reconstructed as spurious high-frequency ‚mirrors').

Jitter

Consider the most standard digital audio format, called "Red Book" audio: PCM audio with a sample rate of 44.1 kHz and a bit-depth of 16. With the Red Book format, there are 44,100 samples per second, and each sample represents 1/44,100th of a second. Even at this relatively low sample rate, time is being cut into very small chunks. This cutting of time is done by a small but important electronic component called a ***digital clock*** (sometimes called a ***sample clock***), found in every digital audio device.

In a perfect world, each of these slices of a second would be exactly evenly spaced. In reality however, there is no perfect clock, resulting in irregularities in the timing of samples. This irregularity is called ***jitter*** and is always present to some extent: Though the goal is to get jitter as low as possible, there will never be ***no*** jitter.

Jitter's inaccuracies manifest themselves in two parts of the process: When audio is being digitized (A/D) and when it is reconverted to an electrical signal being sent to a speaker (D/A). When audio is digitally recorded, due to jitter, the incoming audio signal is being sampled at the wrong time, with this erroneous data permanently and irrevocably stored to the hard drive medium.
To minimize these errors, studios invest heavily in the most accurate, low-jitter clocks that they can obtain, to minimize the deleterious effects of jitter.

Since audio is inaccurately digitized due to jitter, even if the timing of playback was perfect, there would still be significant damage done - but of course, there is no perfect clock for playback either. That means that jitter again introduces errors, with inaccurate timing in the playback of the digital audio. The extent of this additional damage depends on the quality of the clock in the playback device – the less expensive the hardware, the greater the extent of jitter's influence.

The effects of jitter are perceived in a number of ways. The distortion of the original waveform (during digitization and on playback) decreases the clarity of the audio, especially with high-frequency material, and has a noise-like quality. Also, because jitter is temporal in nature and relates to our temporal resolution (as discussed in Lesson 3), it can also affect the width and transparency of a stereo image.

DIGITAL ABUSE

By now it's clear that digital audio introduces degradation in sonic quality through a number of different flaws in the digitization and playback process. As previously stated, in a recording studio this might be considered a necessary and justifiable compromise. In a live situation however, it seems obvious that if this kind of inferior audio reproduction can be avoided, it certainly should be. So why are traditional PCM digital formats used by some pro speaker manufacturers?

DSP

All computers have Central Processing Units, which are more commonly known as CPUs. Digital Signal Processor – or ***DSP*** – chips are similar to CPUs in many ways, with one defining difference: A CPU is designed to accomplish a wide range of arithmetic tasks and also to manage hardware devices, but a DSP chip has a much simpler job, which is to modify the numbers in a digital data stream. The DSP chip is the workhorse of the digital audio world.

It's important to bear in mind that DSP chips, though commonly used in audio processing, weren't singularly designed for audio-only tasks. Early DSPs were developed in the 1970s and 1980s for the telecommunications industry, with the Bell Labs DSP-1 chip being used to separate information from background noise. Even now, DSP chips are used for a wide range of applications, including video processing, voice synthesis, radar, and more.

When used for digital audio processing, the DSP's goal is to alter a digital audio signal in as "real time" as possible. That being said, there will always be some degree of processing time required, resulting in a delay of the outputted signal, called ***latency***. In a recording studio, this latency can be managed, but in a live performance or other playback situation, latency can cause serious problems. In the case of digital delays for speaker alignment, latency can usually be worked around, but when used for other processes (like equalization, for example) DSP-induced latency will introduce a timing mismatch between the original signal and the processed output.

Also, due in part to its generic design, DSP processing will often introduce additional quantization errors due to ***truncation***. Truncation refers to the discarding of bits below the Least Significant Bit (LSB) following a mathematical operation. This amounts to small amounts of rounding error following each process. Individually, these errors are quite small, but when multiple processes are done in series, these errors can compound quickly, becoming significant and audible.

The wrong tool for the job: Speaker Processing

With regard to live sound, DSP processing is widely used. Much of this processing is used to compensate for weaknesses in speaker design and other product deficiencies. Many manufacturers rely on DSP processing to apply filters, equalization, delay, dynamic processes (like compressors and limiters), and more, in an effort to solve sonic problems in their systems.

However, when you consider that PCM-based DSP processing involves the following:

1. **Analog to Digital Conversion**
   - High frequency distortion due to anti-aliasing filters
   - Missing information due to insufficient sample rates
   - Errors in amplitude measurement due to quantization error
   - Errors in timing due to jitter
   - Latency due to digitization

2. **DSP Processing**
   - Latency due to processing time needed
   - Data errors created by quantization and truncation

3. **Digital to Analog Conversion**
   - Errors in timing due to jitter
   - Latency due to reconstruction filters

It becomes obvious that any attempt to solve sonic issues through DSP processing will invariably give rise to other problems, since the very process of PCM conversion degrades sound quality. With each DSP process more clarity is irrevocably lost, and each attempt at correction ultimately results in moving further from the fundamental goal of sonic transparency.

A Place for Digital?

In the recording studio, digital audio is an invaluable tool. In the "real" world, digital media of all kinds play a central role in mass distribution and consumption of audio and video. However, when it comes to realizing the goal of pristine audio quality, the traditional digital formats fall flat.

To make matters even worse, excessive DSP processing used by manufactures to treat specific problems can only further degrade the overall quality of the listening experience.

There is, however, a valid case to be made that not all DSP processes are equally destructive. For example, using digital audio processing for delay (which doesn't change the sonic quality of the sound, but instead only shifts the signal later in time) does no damage to sonic quality in and of itself. Even in this case though, ***the destructive degradation that accompanies traditional digital conversion still occurs.***

Nor are all digital formats equally destructive. Though harder to use in a production scenario, PDM (Pulse Density Modulation) preserves far more detail than its PCM (Pulse Code Modulation) counterpart. If the aim of preserving audio fidelity is paramount, the choice is clear – use PDM if at all possible. Here again, however, some loss in sonic quality is inevitable with the traditional PDM formats.

Is there no way that the power of digital audio can be harnessed without inheriting its sonic downside? There is – read on!

Module 5.3
THE SOLUTION:
KV2'S ULTRA-HIGH PDM

KV2 is committed to uncompromised quality in sound reproduction. How can digital audio factor into this exceptional ecosystem?

ANALOG BY DESIGN

Because every component of KV2's speaker design expresses a commitment to quality, there isn't a need to rely on DSP processing to compensate for the weaknesses typically found in other manufacturers' products. Put another way, there's no reason to apply digital processes to solve problems, because the problems are avoided in the first place. For this reason, the majority of signal flow in KV2 speakers can remain analog.

The Case for Digital Delay

There is one aspect of KV2's signal flow where digital audio can be appropriately used – delay. As discussed earlier in this lesson, delaying a signal can be done without the deleterious effects associated with other processes (effects like equalisation, pitch shifting, compression and so on). Additionally, the inherent latency introduced by the Analog>Digital>Analog conversion isn't an issue – it's factored into the total amount of delay applied.

In the KV2 ecosystem, digital delay is utilized in two ways: First, it is applied in speaker enclosures that comprise multiple speakers that must be meticulously aligned to preserve output phase and cohesion. KV2 also employs digital delay as a means to align multiple speakers that are placed at a distance from each other, in devices like KV2's ***SDD3***.

BETTER THAN THE REST – KV2'S ULTRA-HIGH PDM

Complete sonic fidelity is the KV2 standard, and traditional digital audio formats simply do not rise to the quality that human perception demands. To do the job right, KV2 has innovated and refined digital audio in their products to deliver all the benefit of digital, but with none of the compromises.

In their quest to redefine digital audio, KV2's first decision to make – PCM or PDM – was a relatively easy one. Their need was not for extensive editing or processing, but rather to preserve the detail of the incoming audio signal, especially with regard to the all-important (but often overlooked) temporal resolution dimension of the sound.

The next step was to carefully evaluate the prevalent PDM format – Sony's ***Direct Stream Digital*** (incorporated into their SACD specification). With a sample rate of 2.8224 MHz, it provides a sonic experience similar to PCM at a sample rate of 192 kHz in many listening tests. It was good, but not good enough, and the timing and amplitude of very quick transients wasn't perfectly captured and reproduced.

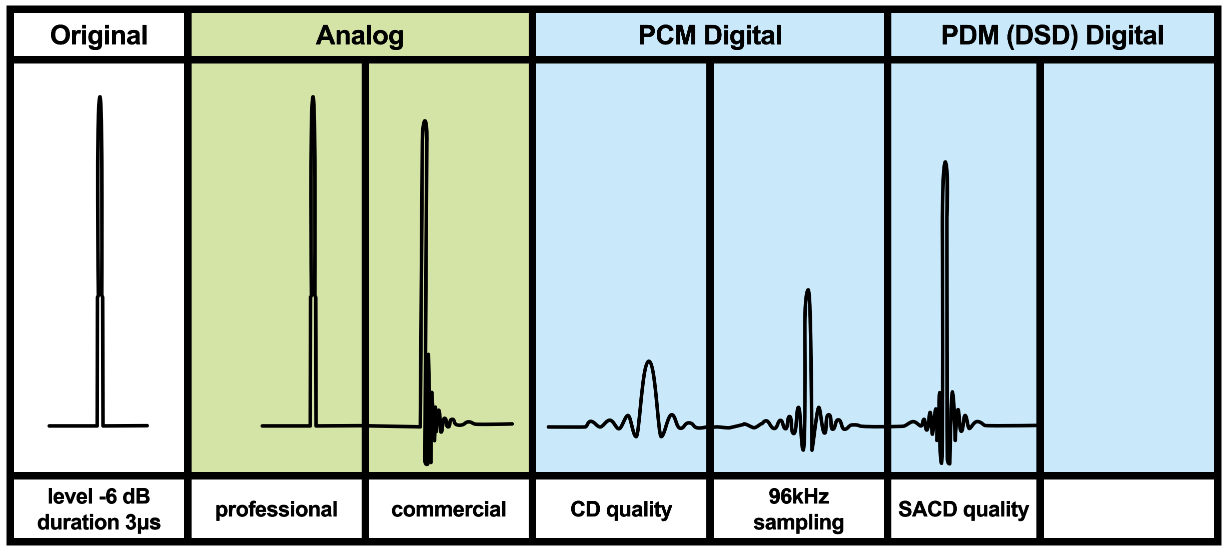


Figure 5.5: Original audio compared to common analog and digital formats.

In figure 5.5, you can see into KV2's testing process. An original audio signal of very brief duration was compared to the resultant reproduction using a variety of analog and digital systems. Unsurprisingly, the original analog signal is most faithfully reproduced with professional analog equipment, and with minimal damage done when using high-end commercial analog playback systems.

When digital conversion comes into play though, there are immediate and profound inaccuracies introduced. This is most obvious with CD-quality PCM audio (44.1 kHz sample rate, 16-bit depth), and though predictably improved with a higher sample rate of 96 kHz, there is still a significant loss of detail and information. The quality in terms of temporal response and accuracy improves markedly with SACD PDM audio (2.8224 MHz sample rate, 1-bit), but even with this high-end digital audio format, KV2's goal of sonic transparency was still elusive.

NOTE: The ripple on either side of the digital examples shows the result of slow conversion and subsequent filtering.

After extensive experimentation and meticulous measurement, a sample rate of almost 8 times that of SACD yielded the results KV2 were looking for. PDM audio with a sample rate of over 20 MHz (22 MHz to be precise) is more than ***7 times more accurate than 192 kHz PCM*** and exhibits no perceptible difference from the original analog signal. At the digital conversion (D/A) output stage, the KV2 specification is even higher – a 40 MHz sample rate – ensuring that both conversion and transmission is done with no audible artifacts.

Figure 5.6 shows KV2's ultra-high digital audio in relation to traditional digital formats – the closest analogue to analog:

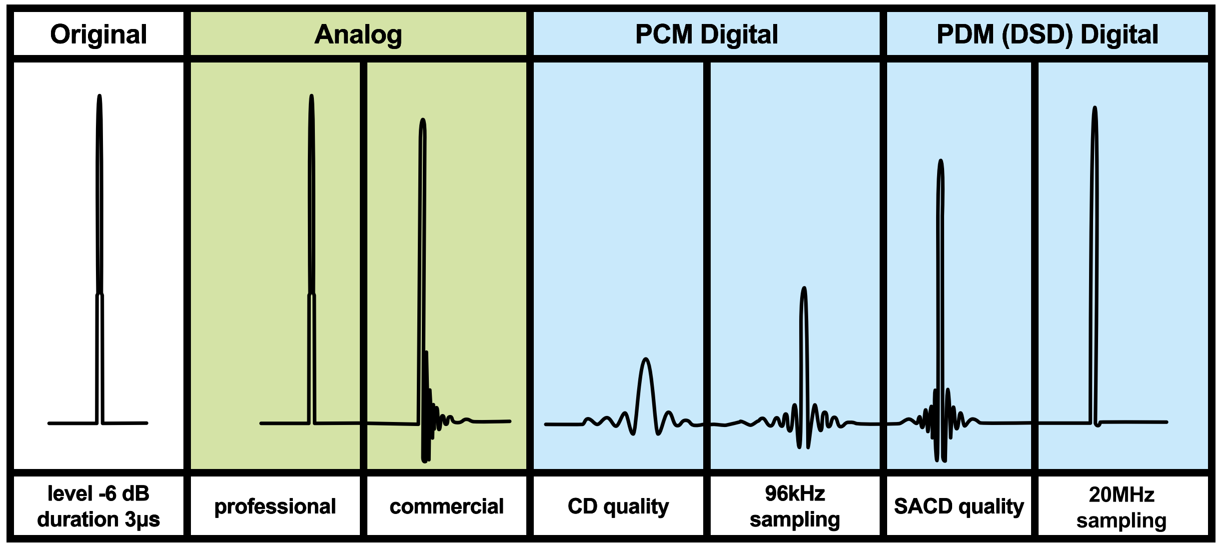


Figure 5.6: KV2's 20 MHz PDM audio (far right) compared to common analog and digital formats.

Dynamic Range and KV2's Ultra-High PDM

Many speakers boast impressive dynamic specifications, but a speaker that can support the entire 120+ dB dynamic range has a superior ability to reproduce audio ***if the problem of time is fixed***. With ultra-accurate digital conversion and precise components that are discussed in detail in other lessons, KV2 speakers can not only get loud quickly and accurately, but can ***also get quiet just as quickly***, giving the listener a true reproduction of transients - the punch and nuance that is actually being created on stage - preserving both high-voltage and low-voltage information.

If you refer again to figure 5.6, you'll see that the traditional PCM and PDM signals never reach the level of the original signal. This is due to the fact that, when using traditional formats, the electronics involved with the Analog to Digital (A/D) and Digital to Analog (D/A) conversion cannot react quickly enough to capture or recreate the original signal in such a small amount of time. KV2's ultra-high digital format solves this problem as well.

The importance of this highly responsive behavior becomes apparent if you consider complex signals comprising multiple instrumentalists and vocalists playing at different amplitudes. KV2's ultra-high PDM is able to keep all the detail of the original signal at all amplitudes, providing a clear and natural listening experience. Other systems, that apply destructive PCM DSP processes using traditional digital formats, are not agile enough to capture this information, muddying loud elements of the signal and masking quieter ones.

THE KV2 DIFFERENCE: DIGITAL

As audio professionals, we make thoughtful decisions about the equipment we use. We carefully choose the right microphone, cables, preamplification, mixing consoles, processing, amplification and speakers, knowing that any concessions that we make at any stage of the signal flow will degrade the final result. For the best listening experience, we know that there can be no weak link in our signal chain.

A speaker also has its own internal signal flow, with each component playing a critical role in the clarity of the sound heard by the audience. Speaker manufacturers, like their audio professional customers, must make decisions about the components that they use. Unfortunately, many manufacturers make choices – including the use of low-resolution digital audio and a reliance on DSP processing - that compromise sonic clarity.

KV2 is dedicated to precise audio quality, from input to output. As discussed in other lessons of the KV2 certification series, design tolerances of KV2 speakers are less than ***one microsecond***, avoiding the kinds of issues that plague other speakers ***before they can become problems in the first place***. For this reason, digital audio is not overly relied upon, and when it is required, KV2's ultra-high 22 MHz PDM provides the advantages of digital audio without sonic compromise.

KV2's "whole system" approach means no weak links in the signal chain – this includes KV2's innovative ultra-high PDM digital audio. KV2 has moved beyond conventional thinking and raised the bar of sound reinforcement.

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Module 6
SUPERIOR DRIVERS

Every component in a complex speaker system is important, but it's the drivers themselves that move the air and create the sound heard by the listener. Drivers are noteworthy in that they represent the stage where electrical energy is converted to mechanical, introducing new challenges to be met in the pursuit of pristine audio performance. This lesson will focus on the components of driver design, and how KV2 systems deliver sonic excellence.

Module 6.1
DRIVER FUNDAMENTALS

In order to judge quality in speaker design, we'll first need to understand the core function of a driver and the role of each of its components.

THE SCIENCE OF SPEAKERS

In the first lesson of this series, we identified sound as a form of mechanical energy, brought to the listener's ears from the sound source through a series of colliding molecules called a ***compression wave***. The challenge with mechanical energy is that it's not easily stored or manipulated in a way that suits the needs of sound reproduction. To work with sound, we need to convert–or ***transduce***–the details of sonic energy into a form that gives us the control that we need.

Dynamic microphones are an excellent example of that kind of transduction. When sound impacts the diaphragm of a dynamic microphone, an electrical current is generated through the movement of two magnets, which is then transmitted down the microphone cable to the destination (from there to be recorded or mixed with other signals). You can think of a speaker as a microphone in reverse: An electrical signal is sent to electromagnetic coils, which then in turn move a vibrating surface (similar to a microphone's diaphragm, but much larger), which in turn starts the compression wave process.

Once the electrical energy is converted to motion, the laws of physics take on greater significance, specifically Newton's first law: The law of inertia. An object at rest will stay at rest unless acted upon by a force (in the case of a speaker, a diaphragm is moved by electromagnetic coils) and will tend towards linear motion unless acted upon by another force. The oscillatory nature of sound requires that the speaker be as agile as possible, with motion starting, stopping, and changing direction many thousands of times per second.

DRIVER COMPONENTS

Drivers of all types share some common components:

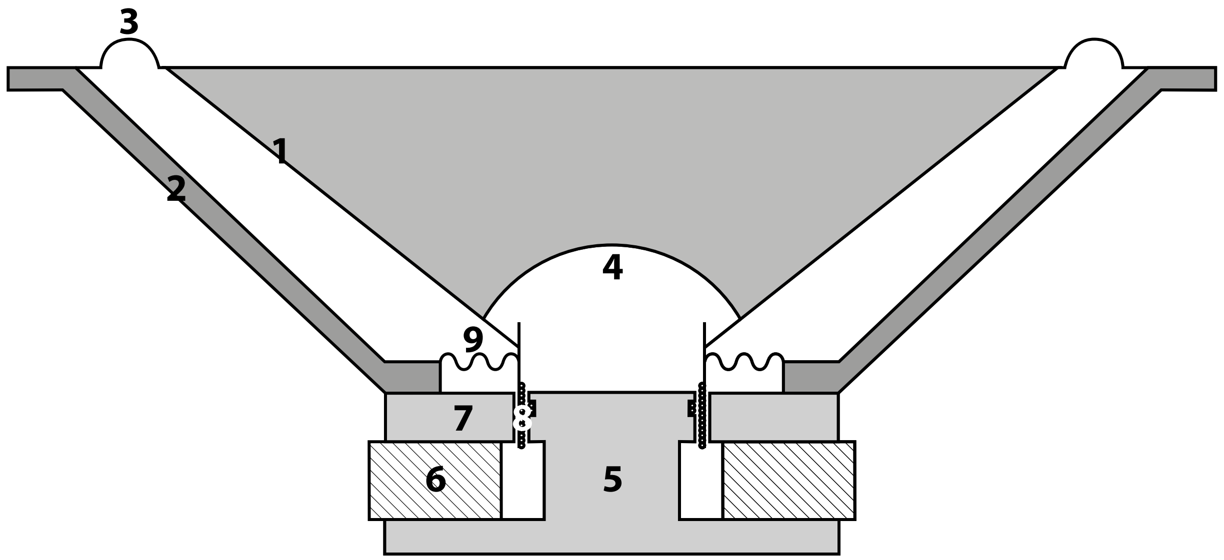

Figure 6.1: The cross-section of a loudspeaker

NOTE: The term driver is commonly used when discussing loudspeakers–sometimes referring to a speaker dedicated to a specific frequency range, sometimes referring to the vibrating membrane moving air molecules, and other times referring to the entire loudspeaker. For the purposes of this lesson the term driver will be synonymous with a loudspeaker.

1. The ***cone*** of a driver, (or ***diaphragm*** in the case of a compression driver), is a rigid surface, usually conically shaped, that directly moves air molecules to create sound. The ideal goals of cone and diaphragm construction are that they be as rigid as possible (to minimize unwanted motion), but also as low-mass as possible (in order to minimize inertia and wasted energy in the starting, stopping, and changing of its motion).

2. The ***basket*** is a protective frame and support structure for the drivers' cone, protecting the back of the cone from damage and ensuring that the cone is properly aligned with the rest of the system. Baskets are typically constructed of metal or plastic. In the case of metal baskets, this structure can also disperse heat from the moving parts of the speaker, increasing long-term efficiency and protecting critical components.

3. The cone/diaphragm is connected to the basket via a flexible ring called a surround. The purpose of the ***surround*** is simple, but important–to allow the cone to move freely back and forth, and to minimize side-to-side motion.

4. The delicate center section of the driver is protected by a small (usually dome-shaped) ***dust cover***. In addition to protecting sensitive central components, it also supports the diaphragm, and contributes to the diaphragm's sonic character.

Behind the conical diaphragm are various magnetic components, which translate incoming electrical signals to motion that moves the diaphragm:

5. The ***yoke*** is the metal back of the driver, contributing to the stability of the magnet's magnetic field by extending one pole of the magnet to the center of the speaker.

6. The permanent ***magnet*** is the component that provides the magnetic foundation of the driver and, along with the yoke and top pole plate, creates the fundamental magnetic field with which the electromagnetic voice coil will interact. The power of the magnet and the consistency of the field is central to the speaker's performance.

7. The ***top pole plate*** is a ring-shaped metal plate that extends the magnet's power so that it can interact with the voice coil (extending the opposite pole of the magnet from the yoke). The size and construction of this plate varies according to the overall construction of the driver.

The magnet, yoke, and top pole plate are arranged in such a way as to provide a circular gap between north and south magnetic poles. It is in this gap that magnetism and electromagnetism interact:

8. The ***voice coil*** (often referred to simply as the "coil"), is a tightly-wound corkscrew of conductive wire. At either end of this wire, a connection is made to the incoming electrical signal, and when electricity is applied, the coil becomes an electromagnet that interacts with the magnetic field of the yoke/magnet/top pole plate assembly. The accuracy of the winding is critical to the coil's field and therefore to the motion of the coil against the permanent magnetic field of the driver. The voice coil is attached to the diaphragm, and when the voice coil moves, so does the diaphragm.

9. The ***spider***, like the surround, plays a stabilizing role, limiting the motion of the voice coil. It is the spider's most critical role that the voice coil be able to move up and down, but not to move laterally in any way that might bring the voice coil in contact with the yoke/magnet/tip pole plate assembly. The component's name derives from early designs, with radiating leg-like structures that have since given way to a more substantial corrugated design.

Figure 6.2: Double silicone spider

SIZES AND TYPES

As a general rule, driver size is inversely related to suitability for a given frequency range. In other words, a large driver is best suited for low frequencies, and small drivers are best suited for higher frequencies. They are commonly broken into four types: ***HF, midrange***, ***LF,*** and ***subwoofer*** drivers.

- A ***HF (High Frequency) or compression*** driver's small size and low mass make this the most agile of the driver types, and best able to contend with high-frequency content. Generally, HF drivers can cover frequencies from around 2 kHz to 20 kHz.
- ***Midrange*** speakers are generally larger than HF drivers and are designed to cover middle ranges from 5~6 kHz down to 300~500 Hz.
- ***LF (Low Frequency)*** drivers are the larger still, designed to cover frequencies from 2 kHz down to 50 Hz.
- ***Subwoofers*** are the largest drivers in an enclosure, covering the lowest audible frequencies (and sometimes beyond). The traditional frequency range for a subwoofer is 100 Hz down to 20 Hz.

COMPONENT AND COAXIAL SYSTEMS

In a speaker enclosure comprising multiple drivers, crossovers are employed to isolate the frequency ranges being sent to each driver in the system, allowing it to perform at its best. Speaker enclosures that include two drivers (an HF and LF) are collectively called ***2-way***, and enclosures with three speakers (an HF, a midrange driver, and an LF) are called ***3-way*** speakers. This kind of layout is collectively referred to as ***component*** speaker design.

In the majority of cases, multiple drivers are arranged side-by side in a cabinet, as shown here in the KV2 EX10, an example of a component speaker:

Figure 6.3: The KV2 EX10 Speaker

A coaxial speaker is a different kind of a design where smaller drivers are placed in front of larger drivers–for example, a HF driver that is mounted in front of a midrange driver. Though component systems generally boast superior sound quality and power handling, coaxial speakers certainly have the advantage of more frequency coverage in a smaller form factor, making them preferable in space-critical situations.

Shown here is a KV2 EX15 3-way speaker, with the top driver being coaxial. This speaker includes both component and coaxial architecture.

Figure 6.4: The KV2 EX15 Speaker

COMPRESSION DRIVERS

Compression drivers are a combination of speaker design with an optimized delivery structure – a relatively small driver (in the far left of figure 6.5) is constructed in such a way that the energy of the diaphragm is funneled through a smaller throat. The sound is then focused through an acoustic horn, which functions as an acoustic impedance match between the diaphragm's sound source and the open air.

Compression drivers are generally much more efficient than direct-radiating speakers and are often used for midrange and HF applications where higher frequencies are involved.

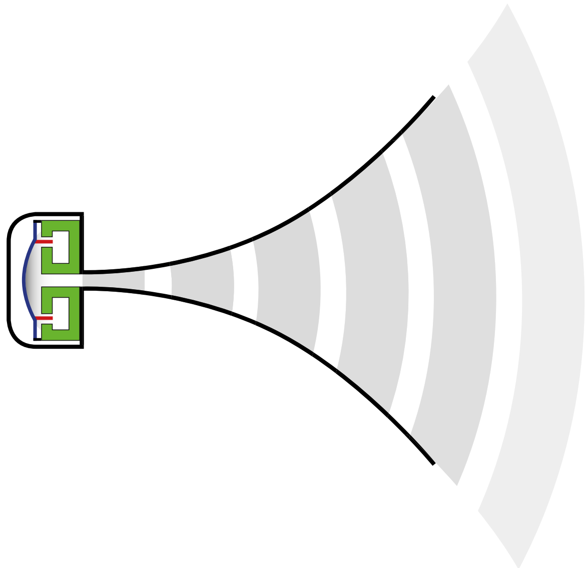

Figure 6.5: A Compression driver and horn

SPEAKER SPECS

The goal of a driver is to accurately translate the electrical signal sent to it into mechanical motion, with minimal sonic information lost (or added) in that translation. There are a number of ways to measure a speaker's features, performance, and quality:

- **Size:** This is generally considered to be outside diameter of the basket.
- **Impedance:** This refers to the resistance (measured in ohms) of the speaker. The impedance should be matched to the amplifier that is sending signal to the speaker. Impedance mismatching can cause components to overheat, shut down, or become damaged.
- **Sensitivity** is a measurement of a speaker's output with one watt of power input from an amplifier. Sensitivity is measured with a microphone connected to an SPL (Sound Pressure Level) meter placed one meter in front of the speaker, ideally in an anechoic environment. The resultant number is expressed in dB. Average sensitivity specifications range in the mid to high 80 dB range for small LF drivers to over 111 dB for compression drivers.
- **Frequency Response:** This measurement determines the highest and lowest frequency that can be reproduced by the driver, and also the ability of it to accurately reproduce sounds within that range. The goal of this specification is to illustrate the suitability of a driver in a component system and also to display any frequency biases exhibited by the driver. It's worth noting that this specification does not indicate the responsiveness of the speaker (how quickly the driver is able to respond in reproducing the frequencies measured).

DRIVERS AND RESONANCE

The ideal goal of a driver is to begin moving as soon as a signal is received and to ***stop*** as soon as the signal ends. However, as mentioned already in this lesson, Newtons first law (inertia) states that neither starting or stopping can occur instantaneously. The amount of motion from the driver after a signal stops is called ***resonance***.

Resonances caused by poor control over the diaphragm's mass reduce overall definition by masking lower-level signals and producing tones not related to the original signal. This can also have a negative effect on the ability of a speaker to reject feedback.

NOTE: Speaker resonance is similar in some ways to an amplifier circuitry's ***settling time*** (which was discussed in Lesson 4 of this series). Like settling time, speaker resonance involves the introduction of new audio frequencies not present in the original signal. Additionally, the distortion caused by resonance masks sonic information that was present in the original audio - lost information that cannot be recovered. In playback situations, this damage doesn't occur just once – because the level of signals is constantly changing, resonance is a constantly-recurring issue.

Figure 6.6 shows an original sine signal (red, top) with its sharply defined end and the same signal reproduced through a driver (blue, bottom), still oscillating after the signal stops:

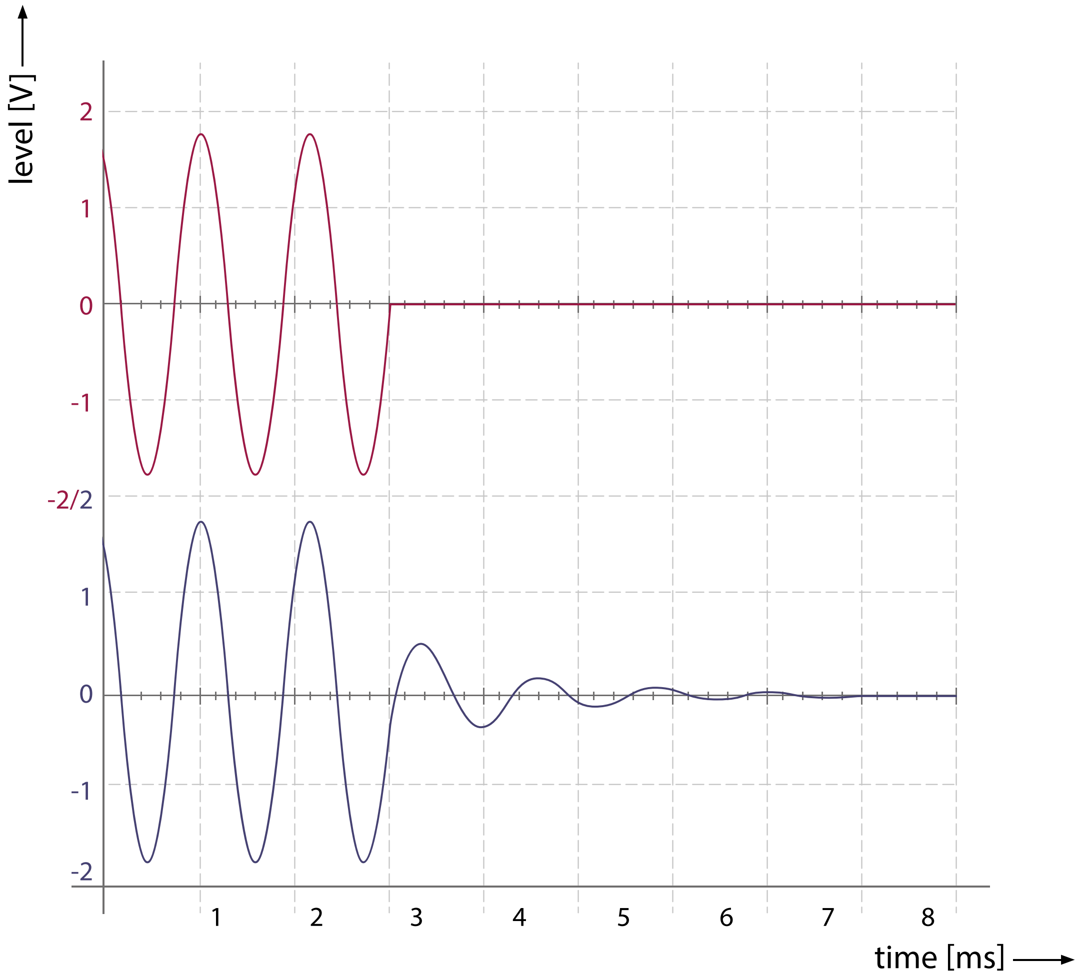

Figure 6.6: An original signal (top) and driver resonance.

The issue of resonance in particular is a differentiator between a typical driver and an excellent one. How does KV2 solve the issue of resonance and other design challenges to deliver the highest sonic quality?

Module 6.2
KV2 DRIVER DESIGN

When it comes to driver design, two of the biggest challenges are to reduce ***inductance*** (and the distortion that it causes), and to optimize the rigidity of the diaphragm (to ensure superior impulse response). Every KV2 driver is specifically designed to perform its role without compromise, with different kinds of drivers employing a number of innovative technologies to address the problem of inductance. Inductance can be described as the tendency of an electrical conductor to oppose a change in the electric current flowing through it.

NPVD COMPRESSION DRIVERS

The Roman philosopher Seneca is attributed with the saying "Luck is what happens when preparation meets opportunity." It was such meeting that gives us the pristine high frequencies of KV2 speakers.

As the story goes, the KV2 team were having lunch in Italy, discussing the right materials for their compression drivers. The material had to be light (low mass), but also extremely strong and rigid–two kinds of qualities rarely found in a single substance. At a nearby table, a group of Formula One racing engineers happened to overhear the conversation and offered up a suggestion: Nitrate coating. This innovative technique, used in high-performance race cars, had never been tried in the pro audio realm.

This chance meeting of two teams has given us KV2's ***Nitrate Particle Vapor Deposition*** (NPVD) process. By treating the diaphragm with Nitrate particles, the resonance and damping of the driver is greatly improved, further reducing distortion while simultaneously enhancing the frequency response of the driver.

The rigidity of NPVD compression drivers delivers true piston motion and clarity in the high frequency. Coupled with some of the largest Neodymium magnets available, KV2's compression drivers boast an astonishing flat frequency response up to 22kHz with only 0.03% distortion.

SHALLOW HORN DESIGN

The function of a horn in speaker design is to amplify and direct the sound created by the driver. They are effective and widely used, but they can cause audible distortion: The deep horns typically used in speaker designs create prolonged increased air pressure inside the horn which, in addition to creating heat at levels above 140dB, can create 2nd harmonic distortion. Further, as more power is put through the horns, more pressure is built up, which results in a loss of high frequency content.

KV2 has employed shallower horns for their speakers. Shallow horns dramatically reduce 2nd harmonic distortion by releasing the pressure on the sound wave sooner (this results in a somewhat wider horn). They do still operate by increasing pressure, and so will technically still increase a small amount of 2nd harmonic distortion, but KV2's shallower horn tracks this distortion with frequency, with lower frequencies exhibiting lower levels of distortion. This design results in cleaner mid and high-mid frequencies.

KV2'S TRANS-COIL SYSTEM

Thanks to the uncompromising standards of KV2's overall system design (including ultra-high resolution PDM digital audio and super-fast circuitry), KV2 drivers feature excellent impulse response–their ability to quickly respond to incoming signals. This contributes not only to their overall clarity, but also to their exceptional feedback rejection. But the ability to get a driver moving is only half of the story–the ability to ***stop*** motion when a signal stops is also crucial.

For their midrange drivers, KV2 takes control over the speaker's mass by using a ***trans-coil*** design (also called ***Active Impedance Control or AIC***). This employs a secondary electromagnetic coil that actively stops extraneous diaphragm motion, reducing inductance and resonance to nearly zero. As **shown in figure 6.7**, this secondary multi-turn coil is positioned in the magnetic gap.

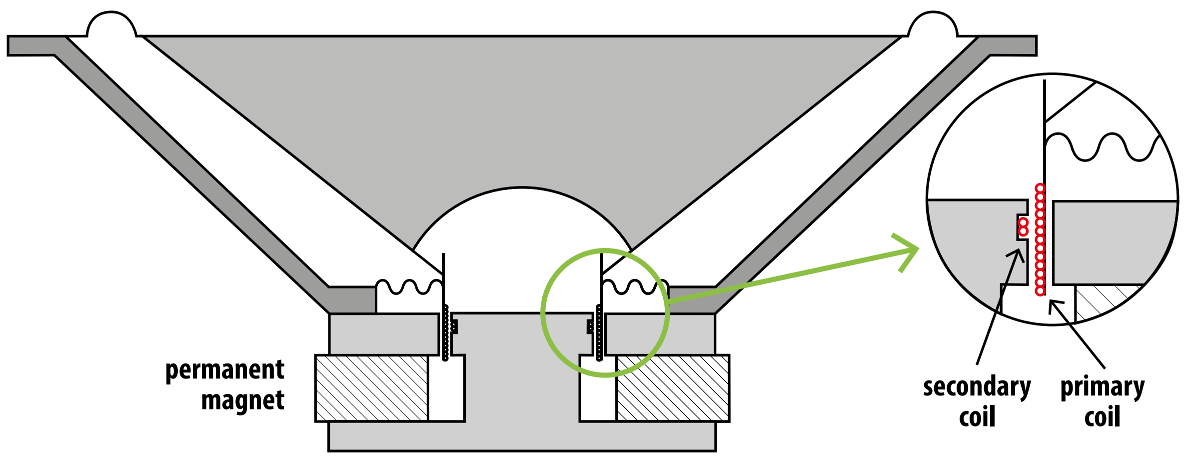

Figure 6.7: KV2's Trans-Coil System

The current sent to this secondary coil generates a magnetic field that is in opposition to the field of the primary (voice) coil. This is an effective 'braking' system that cancels voice coil inductance and greatly reduces unwanted signal modulation. Essentially, the trans-coil provides an active isometric counterforce to prevent the driver from creating distortion.

The dramatic clarifying effect of the trans-coil design is perhaps most apparent in the critical mid-range frequencies. Speakers used by other manufacturers can sometimes attain as low as 0.5% distortion, while KV2's trans-coil powered speakers reduce that distortion by over 10 times, down to 0.05%.

Trans-coil design also has the effect of making diaphragms more responsive, meaning that the dynamic performance of mid-range speakers matches the fastest drivers (compression drivers) in the system. This makes for a more cohesive, flat frequency response of the overall component system.

NOTE: It's important to note that the dramatic impact of trans-coil technology is to a great extent predicated on the high-quality of the overall KV2 system. The excellence of the KV2 sound relies on all links of the signal chain leading to the driver, allowing for the mass control of the secondary coil to have maximum effect.

Other speaker manufacturers have attempted to utilize KV2's innovative trans-coil approach, with little success. The reason for this lack of noticeable improvement is due to their overall slow-speed/ high-distortion design (including associated electronics and processing required). In systems where the quality of the signal coming to the driver is degraded, the more efficient diaphragm control only serves to reveal the other flaws in the system's signal chain.

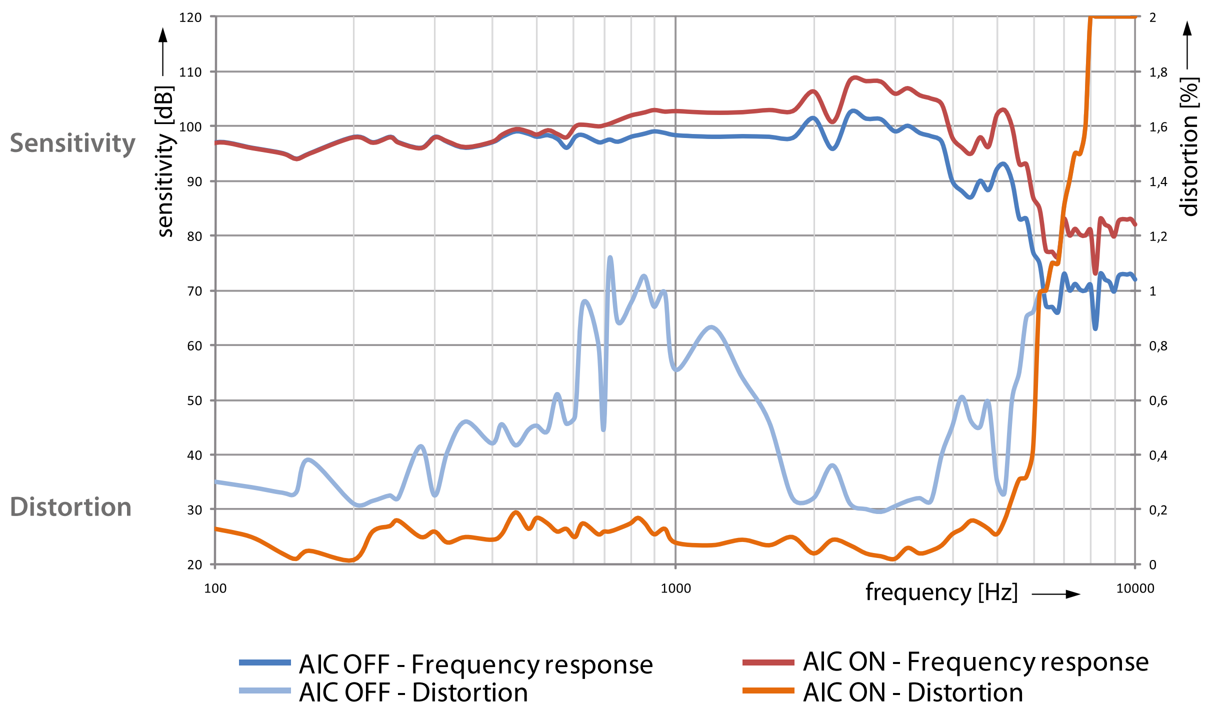

Figure 6.8: Frequency response comparison, showing the excellent frequency response and low distortion of trans-coil (AIC) technology.

PASSIVE IMPEDANCE REDUCTION

Due to the structure and size of a compression driver, using trans-coil (AIC) technology isn't practical. Instead of active impedance control, a copper ring is attached to the center pole piece to reduce inductance. This provides passive induction reduction similar to trans-coil's active impedance control through the copper ring's interaction with the voice coil's magnetic field.

The structure and function of a compression driver's copper ring and trans-coil in a mid-range speaker are similar, differing in degree: The copper ring passively reduces inductance, whereas trans-coil actively eliminates it.

Figure 6.9: A compression driver, showing the copper ring

FOR LARGER DRIVERS: DOUBLE SILICONE SPIDER

LF and subwoofer speakers present unique challenges: Due to the mass of the driver and the distance of the motion needed for low frequencies, technologies such as trans-coil or copper ring aren't applicable. Fortunately, KV2's founder, George Krampera has developed a technology that is as simple as it is effective in controlling the mass of these unique drivers.

In order to reduce inductance with larger speakers, a unique spider design is used to control the diaphragm's mass. This is achieved by gluing two conventional spiders with silicone. Because silicone is both flexible and returns quickly to its original shape, the diaphragm is allowed to move freely, and is quickly brought back to its resting position when motion stops. This allows for proper impulse response and control over speaker resonance.

NOTE: Though invented by George Krampera, this technology isn't solely used by KV2, and has been widely adopted by many speaker manufacturers.

NO WEAK LINKS: DRIVERS AND SYSTEM DESIGN

A speaker's transducers represent the culmination of a signal's journey, from the performers' microphone (also a transducer), through the mixing process, to the system's amplification and electronics, and finally to the voice coil and diaphragm. Throughout the lessons in this series, the idea of time–making that journey as swiftly and accurately as possible–has been a guiding principle of KV2's design philosophy.

KV2's speakers are fast and responsive because they ***have to be***, in order to keep up with the industry-leading speed of the other components of the KV2 system (as discussed in previous lessons in this series). Conversely, the electronics and amplification of KV2's system must be ultra-fast, because the KV2 drivers perform at such a high level as to reveal any problems anywhere else in the signal chain.

Put another way: KV2's electronics can perfectly control their drivers, ***because the drivers are designed to be able to be perfectly controlled***. KV2's dedication to every aspect of their system provides–with all components working at peak performance–the very best in premium sound reproduction.

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Module 7
POINT SOURCE POWER

In the first lesson of this series, we discussed the essential nature of sound as a compression wave emanating from a signal source. We then shifted our collective focus onto KV2's amplifier, processor and speaker design and philosophy, especially on their commitment to the often-overlooked temporal dimension of sound. You've learned that only through this dedication to swift and responsive components at every stage can true sonic transparency be achieved.

In this lesson, you'll learn about speaker arrays–why they're used, and the inherent problems associated with them. We'll then cover how, by offering the highest possible audio quality, KV2 provides superior performance without the need for these cumbersome and problematic arrays.

Module 7.1
SPEAKER ARRAYS

In lesson 2 of this series ("live Sound Basics"), we touched upon the two different ways of configuring multi-speaker arrays: ***multi-source*** and ***line-source***. Let's review these systems and dig a bit deeper.

MULTI-SOURCE

A ***multi-source*** system involves a number of identical full range cabinets, distributed along both horizontal (X) and vertical (Y) axes. This configuration is based on an idea that more speakers will equal more volume, allowing these systems to cover larger audience areas.

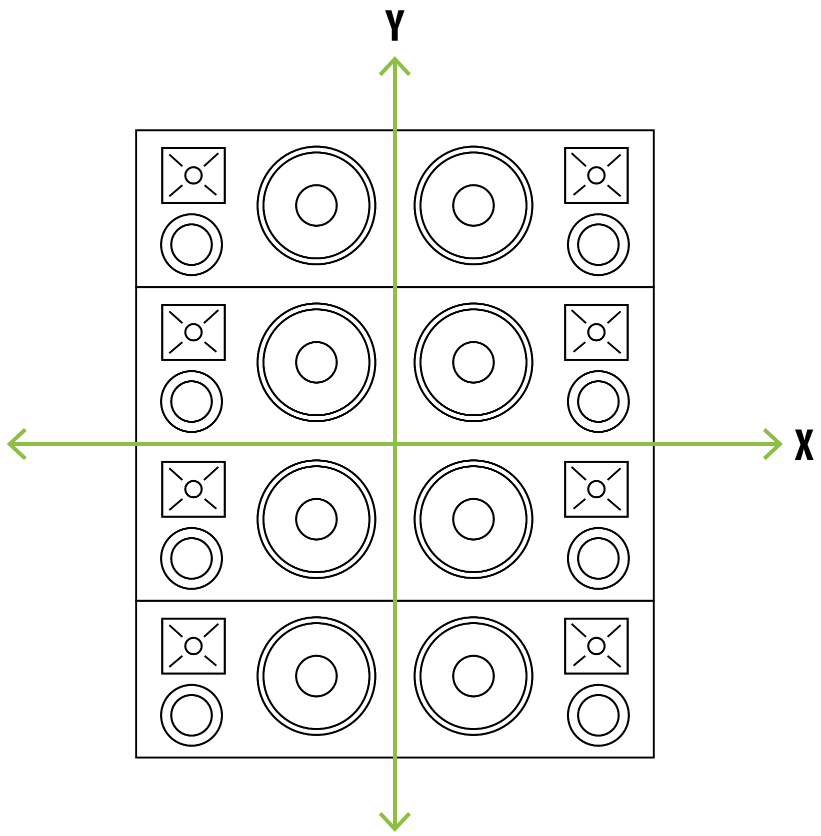

Figure 7.1: A Multi-Source Array Speaker Configuration

Although this goal is achieved in terms of increased volume or SPL, it comes at a significant cost. Despite multi-source systems producing greater SPL at their source, the ultimate sound that reaches the audience suffers from substantial problems. The combination of signals from multiple drivers that are spaced apart introduces significant errors due to the different arrival times of the multiple signals. In a large multi-source system, this could be in excess of 800 individual speakers, each one creating sound that arrives at the listeners ears with varying degrees of delay.

LINE-SOURCE

A line-source or "line-array" system is similar to a multi-source configuration, but with speakers positioned in one (usually the vertical) axis only.

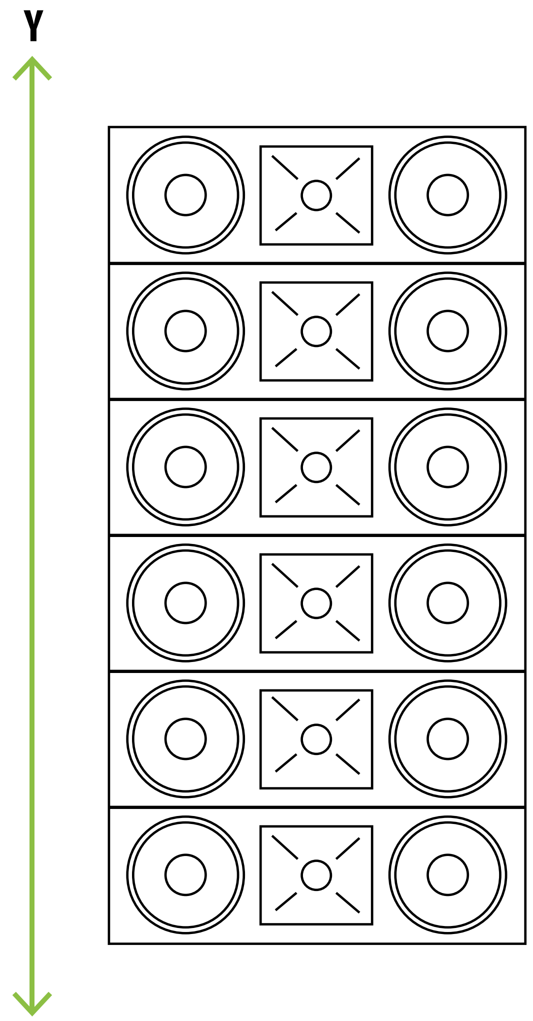

Figure 7.2: A Line-Source Array Speaker Configuration

This type of system marks a significant improvement over the original multi-source concept, but still fails to address fundamental issues associated with the use of multiple speakers. A line-source system still has an inherent problem of distortion along the Y axis, and though some manufacturers claim that their designs can create a coherent wavefront and controlled vertical dispersion, empirical evidence shows that this claim is far from true.

THE PROBLEM WITH ARRAYS

Multi-source and line-source arrays have inherent weaknesses that stem from the fact that the listener is hearing multiple instances of the original sound from multiple locations. This results in ***phase alignment*** and ***interference*** problems.

Consider this: a round pebble is dropped in still water. The ripples radiate outward in a circular pattern, with the height of the waves decreasing as their distance from the source increases. Sounds in the real world operate in much the same way, emanating from a single source, with the amplitude of the sound decreasing with distance according to the inverse square law (discussed in Lesson 1 of this series).

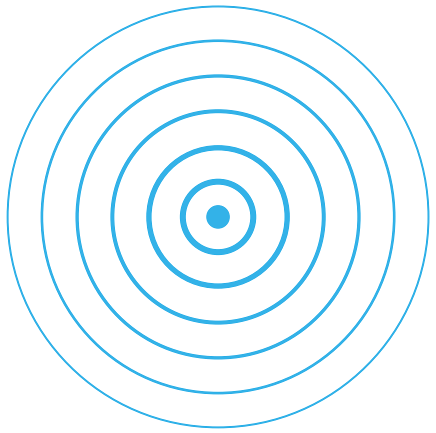

Figure 7.3: Waves radiating from a single source

When multiple speakers are used in an attempt to deliver more sonic power, they behave like multiple pebbles being dropped at the same time, each with their own ripples–ripples that collide with each other.

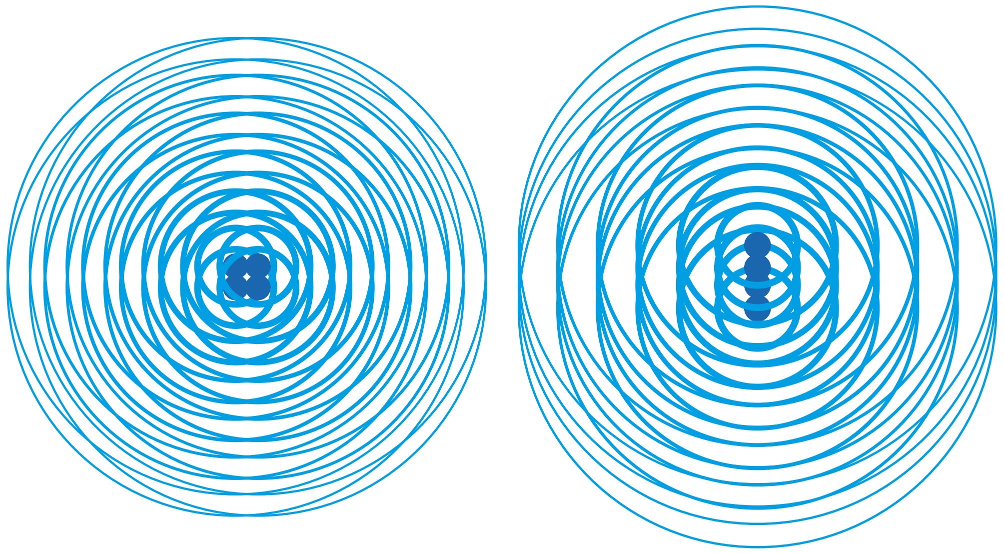

Figure 7.4: Waves radiating from a multi-source array (left) and a line-source array.

With a multi-source arrangement, with drivers arranged on both the x and y axis', the interference of the sound sources upon each other creates a complex web of sonic degradation. With line-source arrays, with drivers positioned in only one axis, the interference is reduced, but still very present.

Unlike our example of pebbles dropping into water, sound radiates in multiple directions–let's look at how different speaker positions are perceived by the listening audience:

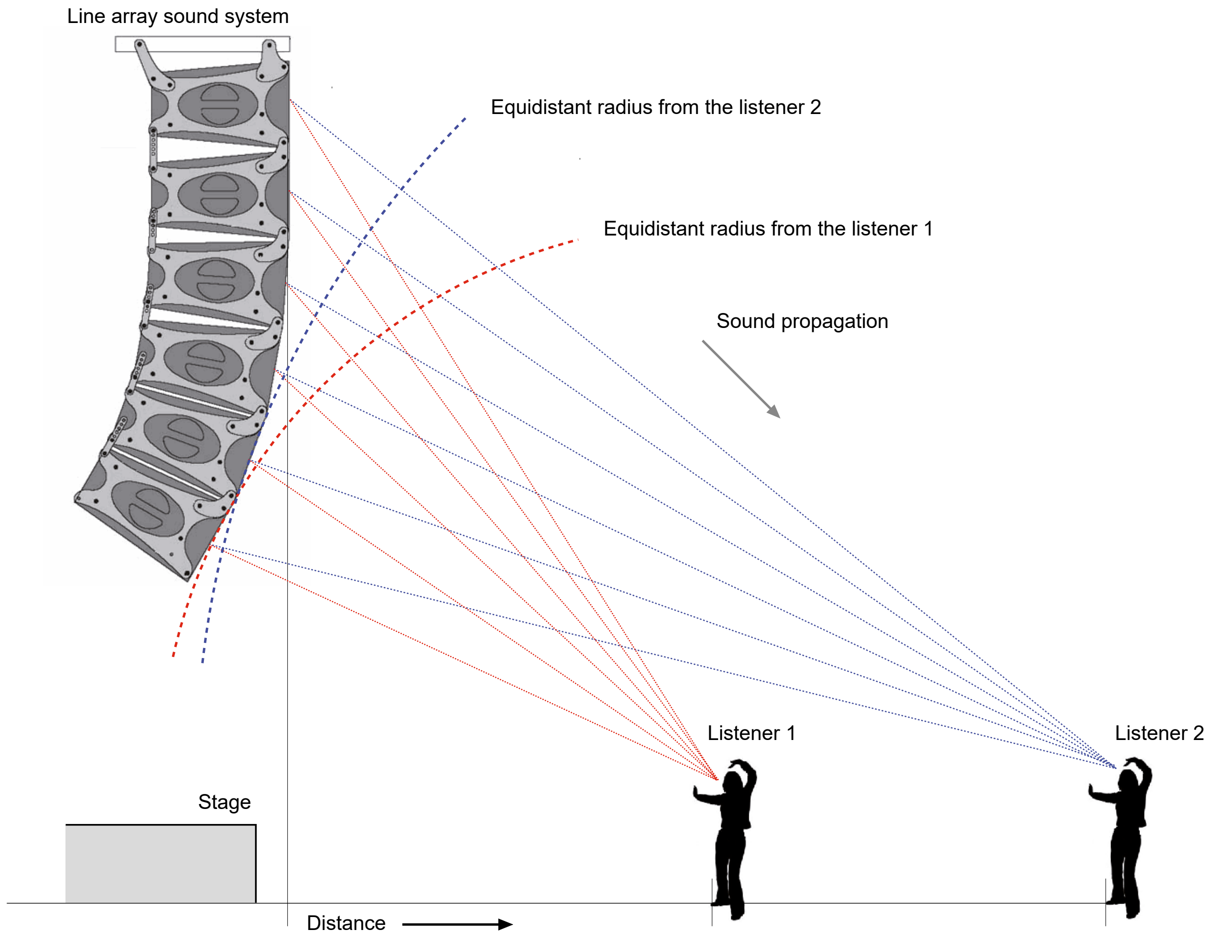

Figure 7.5: A Line-Array speaker configuration as perceived by two listeners

Figure 7.5 shows the output of a line-array as perceived by listeners in two different positions. This shows the insidious nature of the inherent line-array dilemma: A line array can be optimized for a single position only—any adjustments made for listener 1 would worsen the experience for listener 2, and further would only hold true if the listener stayed in one position.

For example, if a single pulse were sent out of a line array, a listener would hear the pulse from the closest speaker first, followed by the next closest speaker and so on. This has the effect of "smearing" the sound, especially in higher frequencies where accurate sound reproduction is most important in terms of timbre and intelligibility.

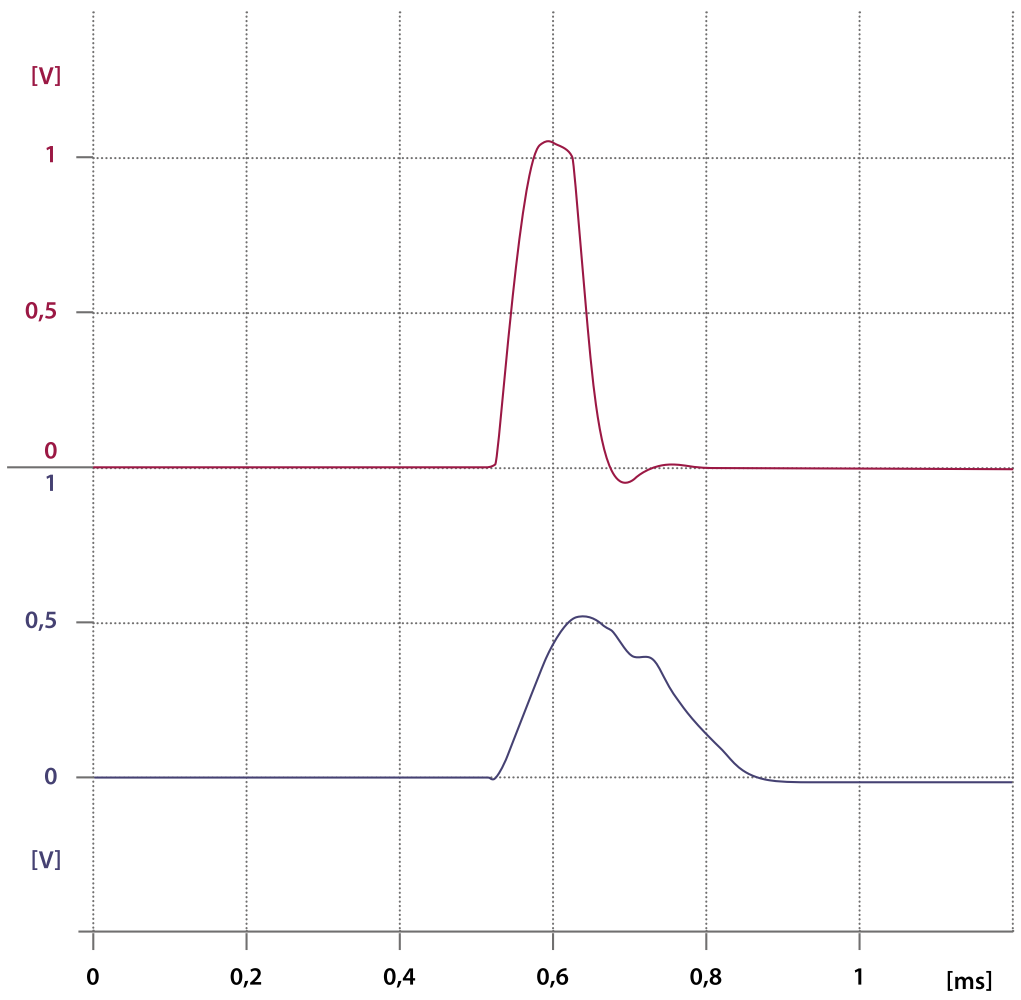

Figure 7.6: A short pulse as reproduced by a point-source system (top), and an array. This shows the "smearing" effect caused by multiple drivers at different distances from the listener.

To make matters even worse, errors in the timing and response of individual drivers multiply the incoherence of the arrayed system. Interference that is unavoidable due to "identical" sounds emanating from multiple locations is exacerbated by errors in timing from individual drivers due to slack tolerances regarding a sound's temporal position.

In short, in an effort to create a louder signal, the clarity of the sound is damaged by interference in three ways:

1. Identical signals emanating from multiple points will interfere with each other.

2. Inconsistent relationship between the array and the listener result in a "sweet spot" of minimized interference for some audience members at the expense of sonic quality for others.

3. Inconsistent performance from individual members of the array multiply the destructive effect of interference and decrease sonic fidelity.

Interference is perceived as a reduction in sonic clarity and intelligibility, and a decrease in the system's overall impulse response. In real-world situations, this has serious ramifications:

- Because an array results in the lengthening of a signal (as a result of a signal being played at different times from different speakers at different distances), a line array is more prone to feedback.

- As a listener's distance from the PA increases, so does the amount of interference between the different drivers, reducing the array's ability to accurately reproduce the original detail, dynamics and nuance in the sound.

- The problem of interference is obvious as the listener moves, as the small delays between all of the speakers continuously change due to the changing distances between the listener and the sound sources. For this reason, it is impossible to create an even sound field when employing a multi-source or line-source system. Though this problem is lessened with line array systems (as opposed to the 2-axis arrangement of a multi-source array), distortion is still apparent when moving laterally in the audience area.

- Array configurations used in an outdoor setting can be significantly affected by wind and changes in temperature. As a result of moving air molecules, sound from disparate sources can be severely disturbed, and in extreme cases can cause a complete loss of high frequency content.

- Line array and Multi-source architectures are inherently inefficient - more speakers result in more wasted energy (through interference). The presumption that adding more speakers will result in a proportionally louder PA system is mitigated by interference. This is most apparent in the high frequency range, where the interference results in a degree of cancellation so great that from 10 kHz upward, one point-source HF compression driver provides the same power as ten arrayed HF compression drivers.

- In an attempt to correct the problems caused by array systems, system designers and live sound engineers will often add large amounts of equalization and DSP processing. This not only reduces the systems overall headroom and reliability, but also tends to create other sonic problems and degradation.

NOTE: There has long been an assertion from some line-array manufacturers, that a line-array produces a coherent wavefront that means the listener only ever hears the box that is directly facing them. If it was true that a line-array produced a coherent wavefront and controlled vertical dispersion, then that would suggest that when you listen outdoors to the top 4 cabinets of a system that is flown 20 metres above you where there are no reflections from a ceiling or walls, you should not be able to hear ANYTHING. If the sound propagation was truly linear, then it should simply shoot straight over your head and then eventually dissipate. Of course, sound becomes less directional as the frequency lowers, but even so if you were just listening to the horns, the same theory should hold true. This would be the only acceptable result proving that line-array elements do not interact with each other, and we know that this is not the case.

Module 7.2
POINT-SOURCE

Sound emanating from a single point, as it does in the natural world, provides the highest quality listening experience. In this section, we'll explore ***why*** point-source speaker arrangements are superior, and how KV2's point-source solutions are the choice of discerning professionals.

POINT-SOURCE PRIMER

The advantage of a point-source speaker arrangement becomes apparent when you see it in the context of a listening scenario:

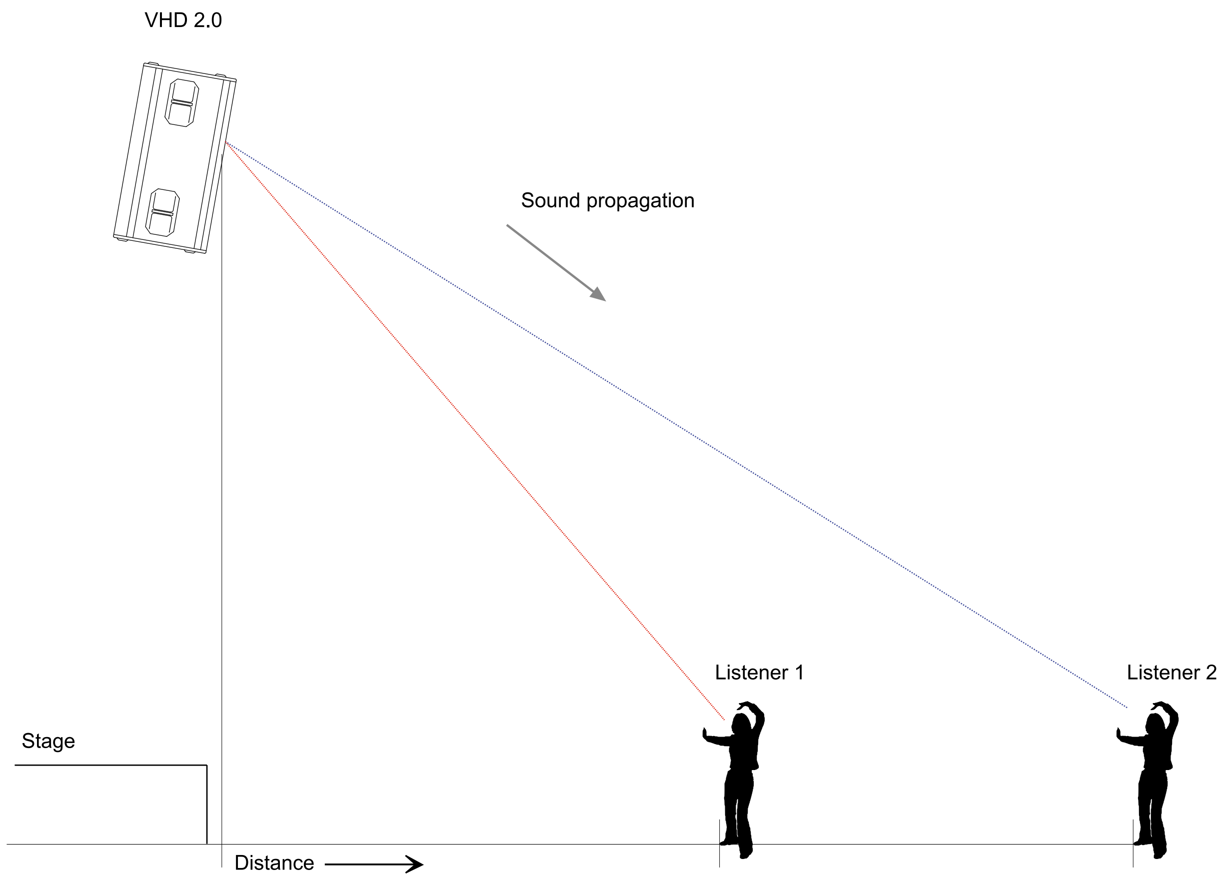

Figure 7.7: A point-source speaker configuration as perceived by two listeners

The difference between line arrays and point source is as obvious as it is profound: When sound emanates from a single point, the entire audience is in the sweet spot. While it is still true that amplitude decreases with distance (according to the inverse square law) and the normal rules of a transmission medium still apply, the fact that the signal isn't being muddled at its source allows for more of the listening audience to enjoy clear and defined sound.

This combined with Constant Power HF horns provides a naturally attenuated signal as you get closer to the system, delivering a linear sound throughout the listening area.

Impulse Response

In order to deliver high definition audio, it's important that the drivers used exhibit a short impulse time. This responsiveness is critical to intelligibility and sonic detail, but with an array system, impulse response is badly damaged due to variations in the timing of signals arriving to the listener. Combined with quality components, a point-source speaker can deliver that clarity without interference.

Feedback Avoidance

Feedback requires time–a late signal is picked up by a microphone, then played back late again, and so on. A speaker array exacerbates the probability of feedback, but because a signal sent to a point-source system starts and stops more quickly, the chance of feedback is greatly reduced.

Dynamics and Power

In previous lessons in this series, you learned about the importance of dynamic response, and how fast electronics and low-distortion transducers are critical in delivering accurate dynamics. With the interference of array systems, particularly in the high frequency ranges, large amounts of equalization are needed to be added to compensate for problems created by the array. This boost in high frequencies not only fails to properly address the underlying issue, but it also lowers the system's overall headroom.

This also requires more power–on average ten times the power–to drive high frequencies, compared to a single point source cabinet. This results in an overall inefficient system.

Coverage

With no interference from competing drivers, a point-source speaker arrangement significantly improves off-axis coverage. Further, the clarity of KV2's speakers eliminates the need for delay towers in many situations.

NOTE: A ***delay tower*** is set of speakers, typically set up every 50~80 meters, to supplement a main PA system, due to the loss of intelligibility over distance. Delay towers also help to address the destructive interference caused by air movement.

KV2's point-source approach is effective because all of the speaker components work together to provide excellence. Low-distortion drivers and horns, optimised acoustic designs and accurate fast electronics all combine to give, more than anything else, ***intelligibility over distance***. Because of the purity of the sound at its source, intelligibility is maintained as the sound goes out over distance. Line arrays will lose intelligibility over distance as a result of interference, necessitating delay towers.

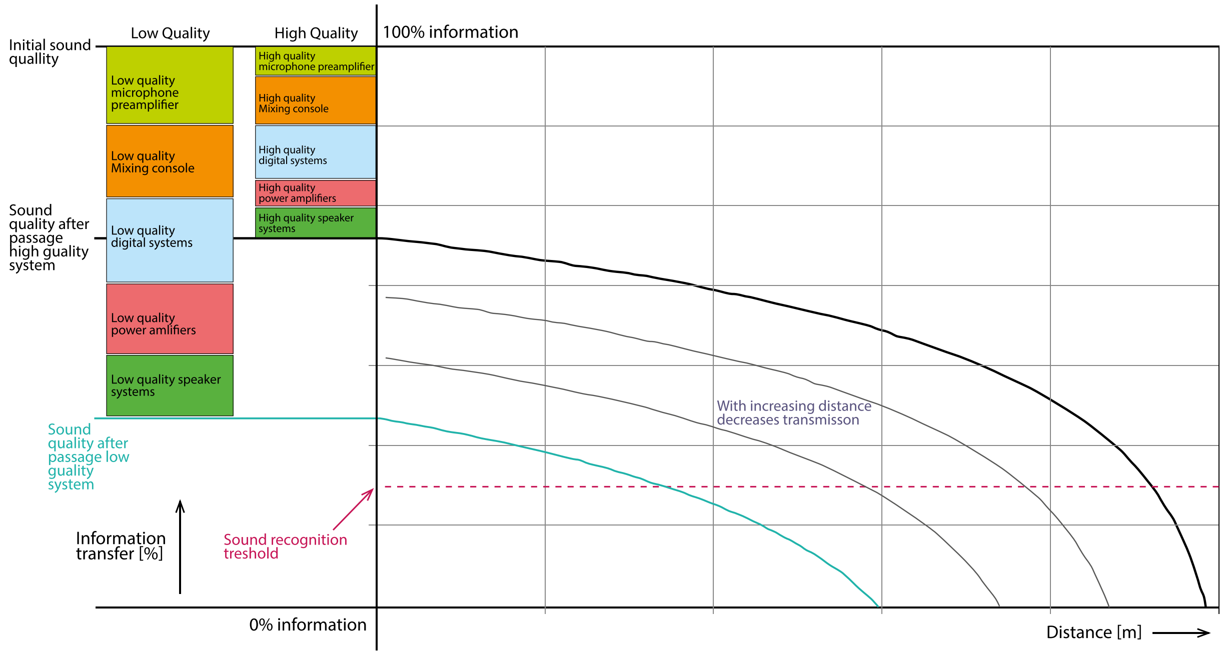

Figure 7.8: The effects of high-quality systems in terms of intelligibility over distance

POLAR PATTERNS

Viewing a speaker's polar pattern can be useful in judging its quality and efficiency. A polar pattern shows the radiant capabilities of a speaker or speaker array.

Think of a polar pattern as a birds-eye (in the case of a horizontal pattern) or a side view (in the case of a vertical pattern), with the coverage of that speaker plotted on a circular graph at various frequency levels (with 0º representing the direction in which the speaker is facing). Concentric circles in the graph indicate distance from the source, with decreasing amplitude levels as they radiate from the center point.

The ideal scenario is to have a wide area of coverage which is consistent throughout the speaker's frequency range. **Figure 7.9 shows** the polar pattern of a point-source speaker at 10 KHz, and you can see here the advantages of such a system–there is a large area of coverage, with a circular overall coverage shape.

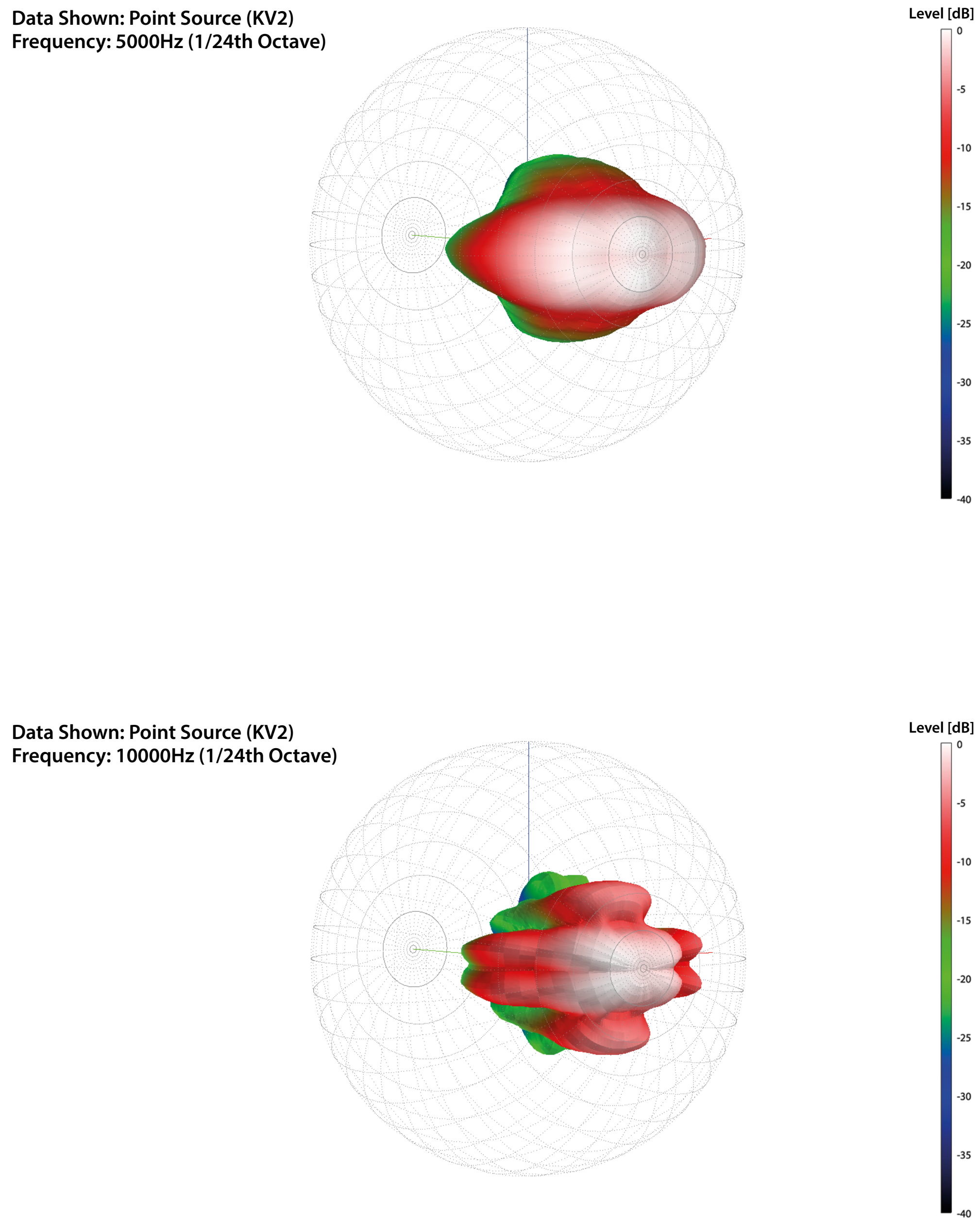

Figure 7.9: Point-source front and side polar patterns

In stark contrast, the polar graphs for these 2 popular line arrays shown at Fig's 7.10 & 7.11 are far more erratic.. Destructive and constructive interference causes sharp peaks and troughs, resulting in an inconsistent area of coverage.

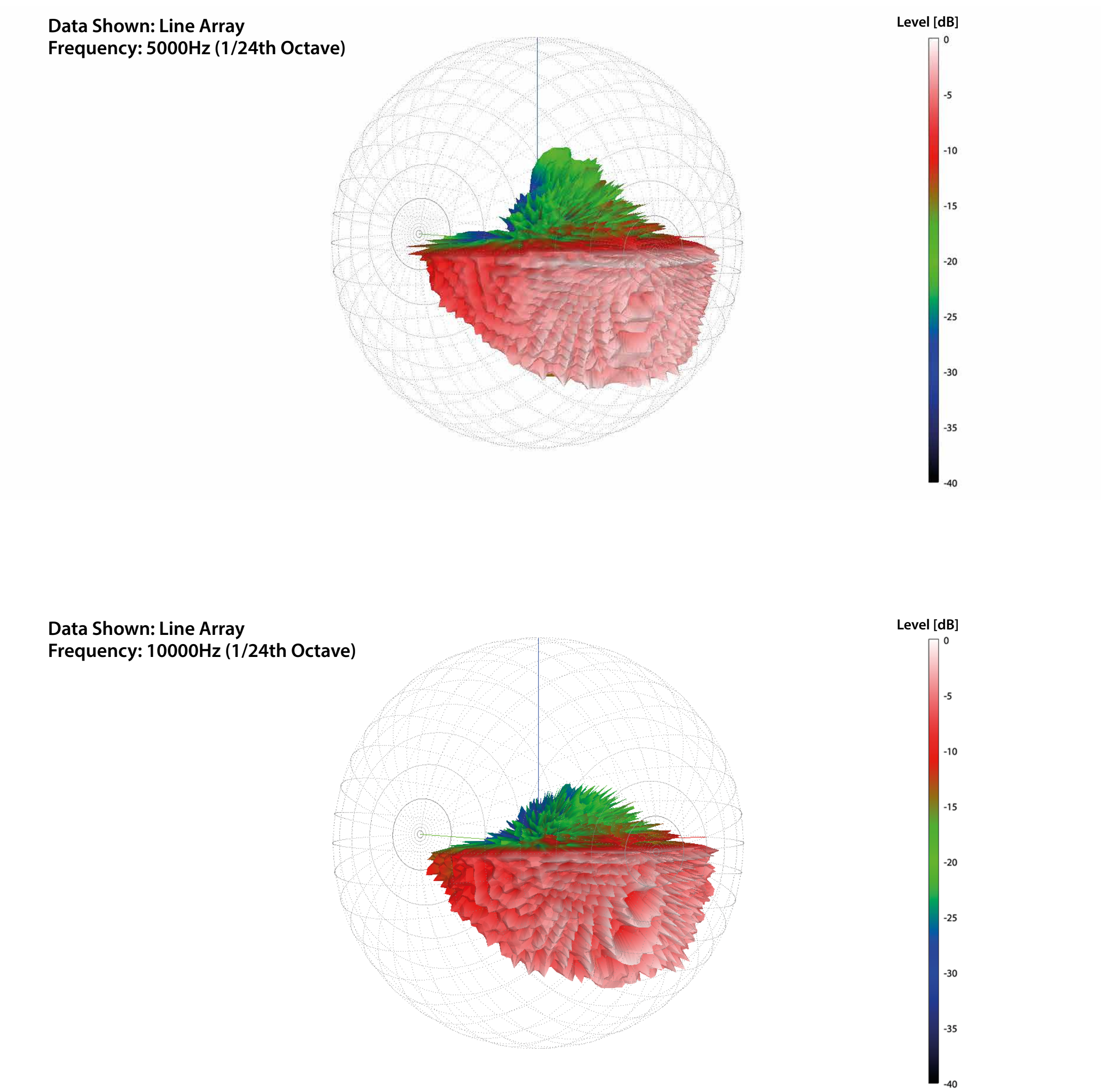

Figure 7.10: L'Acoustics K2 line-array polar patterns

In addition to not being a consistent area of coverage, it's also not a resilient one. The polar graphs **shown in Figure 7.10 and 7.11** represent systems that are easily destabilized by the normal movement of air in the listening area, or even the introduction of an audience. This has real-world implications for the live sound engineer who, after spending time preparing his system in an empty and still hall, has to try to recapture the same sound during a live performance when temperature change and subsequent air-movement is introduced into the venue.

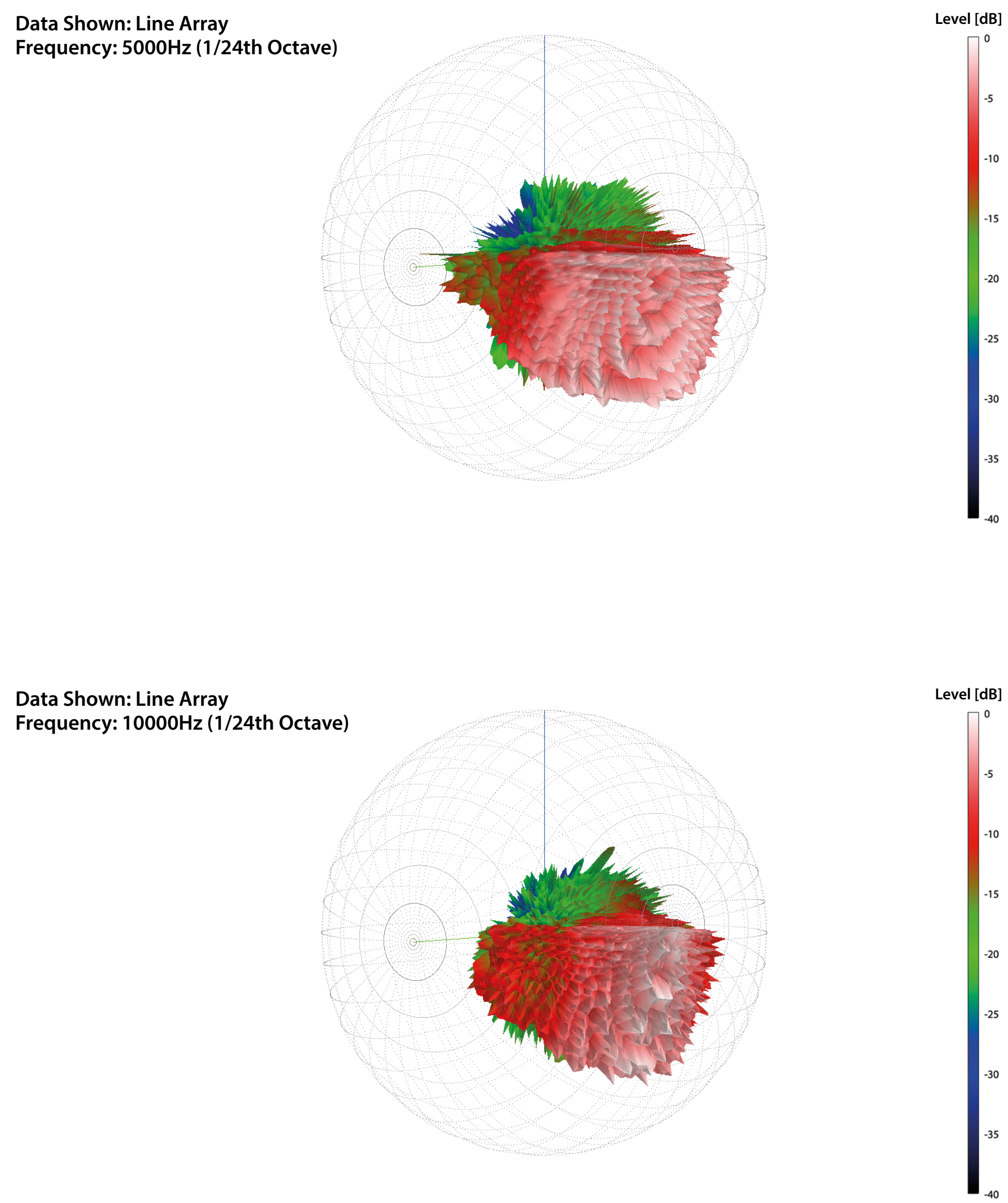

Figure 7.11: D&B GSL line-array polar patterns

QUALITY IS KING

Given that a point-source is inherently superior to a speaker array of any type in terms of consistency and intelligibility, the question that arises is "how can a single speaker match the coverage of multiple speakers?" The answer lies in the standards to which the point-source speaker is designed.

Many speakers utilize various gimmickry to compensate for weaknesses in design–low-resolution digital audio and destructive DSP processing chief among them–which serves to distort the output of the speaker as a whole. With this distortion built into the design of the speaker, even a point-source speaker arrangement requiring these treatments will provide minimal advantages if any.

On the other hand, if exacting specifications at every stage are designed to provide a speaker with uncompromising temporal accuracy, impulse response and low distortion, point source becomes a viable if not preferable option. With a transparent loudspeaker such as this–one that amplifies the incoming signal faithfully–point-source systems exhibit clear sonic superiority and more even coverage.

KV2's speakers are purpose-designed for point-source sound. Their fast componentry not only provides low distortion, but also helps to avoid the chance of feedback in the system–because KV2's speakers are so responsive as a result of fast electronics and low inductance, sound starts and stops nearly identically to the original source sound, allowing no time for feedback to occur.

Additionally, because KV2 speakers are designed with point-source delivery in mind, their systems are designed to provide constant power on every frequency in the audible range. With no other sound sources to cancel it, this provides greater coverage without sonic compromise, and higher quality at the source ensures the same high quality at any given distance.

Module 7.3
A CONVERSATION WITH KV2

The differences between average speakers in an array and a KV2 speaker may seem minor, but they're profound. Consider this: A performer expresses nuance at the speed of thought–the difference between articulations, the gestures of a conductor's baton–all of these occur with neurological velocity. The way that we ***hear*** audio is similarly agile, appreciating the emotion of a performance with the same sensitivity as the performers' ability to express it. KV2's top-of-the-line speakers combined with a point-source delivery, reveals artistic detail that the audience has never heard before, thanks to the single-minded dedication to delivering the best sound possible.

Before we leave this lesson, let's get to know the KV2 team, and hear what they have to say about the state of the live sound industry, KV2's philosophy, and thoughts on what to expect in the future!

MEET KV2

At the heart of KV2 is its team, representing the best in the business. Let's meet a few of them:

George Krampera

George Krampera grew up in the Czech Republic, surrounded by electronics and building his own radios, power amplifiers and PA systems for local bands from an early age. After he married and with a young family, George travelled to Canada, where he not only designed some of the finest pro audio solutions for Yorkville Sound, but also created his own Guitar amplifier brand ***Rexx Acoustics***, which outsold Marshall in Canada during its first year of production.

On returning to Europe, he continued his role as an innovator at RCF, creating the now-legendary ART, Monitor, and Event series, along with cutting edge component technologies at both RCF and B&C. A brief move to Mackie also saw him transform their MI box sales to a multi-million dollar success story all over the world. In his storied career, George has earned a reputation as one of the most significant and revolutionary transducer designers in the history of modern audio.

George's passion was always to create the perfect sound–a dream he finally realised by forming KV2 Audio. His focus since then has been to push industry boundaries, develop technologies that eliminate distortion (and loss of information), and deliver true dynamic range. Through KV2, George offers sound reproduction that is an absolute and honest representation of the original source and conveys true emotion to the listener.

George Krampera, Jr.

While growing up in Canada, ***George Krampera Jr.*** developed a keen interest in business (with his first management role at the young age of 13). As his family moved to Italy and then to the Czech Republic, his excellent language skills and intuitive administrative acumen led him to acting as a translator between international companies, as well as setting up his own businesses (in the timber, real estate, and renewable energy fields).

Following his family's famous tradition, George joined KV2, selling his other businesses to fund the construction of the KV2 factory and manufacturing infrastructure. Today he runs the global KV2 business, applying his unique business style, exceeding expectations, and building ironclad relationships within the professional audio community. With George Krampera Jr. at the helm, KV2 is well positioned to change the pro audio world.

Jonathan Reece

Jonathan Reece studied music and electronics before setting up an Installation, rental, and recording company in the early 1990's. The company grew significantly, looking after many of the larger UK leisure companies and becoming a prominent installer of high quality audio products including RCF. It was during this time that Jonathan was invited by Andy Austin Brown, at that time serving as RCF's UK sales Director, to meet George Krampera in the Czech Republic, as a reward for his business' high level of performance.

The two hit it off, and several years later formed KV2, with Jonathan selling his own company and becoming a founder member of KV2 Audio. In his 20 years with the company he has been involved in International and UK sales, product development, engineered testing, and even a period as CEO from 2006 to 2011. His knowledge of everything KV2 is a resource that the company has come to rely on and, like George he has an absolute passion for creating the perfect sound.

Andy Austin-Brown

Andy Austin-Brown (KV2's Technical Projects Director) began his musical journey at the age of 7 as a trumpet player, studying classical music and later joining Her Majesty's services in the UK and Europe. After his service, he played professionally in big bands and orchestras before moving into the pro audio community, where he worked at TOA Electronics and RCF UK–where he met George Krampera. From that first meeting, Andy has worked with George, and found a home at KV2.

In his current position at KV2, Andy supports customers with technical advice, system design consultancy, and presentations of KV2 systems to eager audiences worldwide. Andy listens and thinks like a musician, which combines with his deep technical skill to deliver the best of support for the KV2 customers.

KV2 Q&A

The following is a summary of a recent conversation among these key KV2 members:

Q: Through KV2, you've followed your vision to eliminate distortion and any loss of sonic information, largely as a result of slow components. The science behind what you're doing is solid, but why isn't anybody else talking about things like the importance of the temporal dimension of sound?

GK: I think it's all about marketing. They're always trying to sell something new, but is it better? Is it the best? Through marketing you can control people very easily. People are quite lazy, me included, and we tend to believe the marketing, especially if the technology that they're selling is cheaper and easier–people will go for it.

GK jr.: The managers of the companies today are not praised for a great sounding system–they are praised for a good profit achieved. These are the guys running the majority of the companies. Having a solution for a fundamental problem needs to happen in the hardware. And that with knowledge that any change or alteration has a huge effect on anything else in the whole signal chain. That means that it is a much more complex task and also that the solution is much more expensive as it can not be dealt with in the digital domain of things.

AAB: Competitors generally are focused on the Digital domain through manipulation and control, concentrating on SPL rather than quality, they therefore don't see the value in our theories, technologies or aims, preferring to market features rather than audio quality.

Q: From Yorkville sound to Rexx, to RCF, B&C, Mackie, and many other industry-leaders, you've been present during some of the most important innovations in audio technology. Is there any one technological breakthrough that stands out?

GK: I would say there are two main things. First, the trans-coil technology which improves the sound quality dramatically. Second, the shorter horns, which reduce the 2nd harmonic distortion–it might be a small improvement, but when you add those little things up, it makes a big difference.

GK jr.: I do not think that it is fair to say that there is one specific one that stands out more than another. I believe that the fact that a man [George Krampera] has led his whole life in one direction is the key here. By that I mean that to go and move from company to company, from continent to continent, with the goal of understanding and having hands-on experience in the details of each small part, and then to understand what can be done in each aspect–that leads to being able to have control over the situation as a whole.

Q: The components that you use in KV2's speakers, from amp topology to transducers, and virtually every component of the system has resulted in a speaker that can show all the detail coming from the stage. Of course, it can also reveal any other weaknesses in the signal chain. What are your recommendations for other parts of the overall live sound setup?

GK: If you want to shoot far, you have to take care of sound at the source. The higher the quality of the sound at the beginning, the further you can shoot. With KV2's speakers, it's possible to go around the corner of a building, or 130 meters from the PA and still hear clear sound. But to get this, you have to take care of how your amplifiers and other components are performing.

When it comes to the mixing board, and preamplifications, it's very important to have op amps running in Class A. Summing is very important as well, and analog is best–you can have digital channels but mix them in analog. Overall, you need components that are very high quality, high specification, low distortion, and high headroom. At every stage you need components that don't distort and that deliver very pure and clear signals, all the way from the microphone to the speaker.

GK jr.: Look and find the best there is. Accept limitations only when there is so far no other option. That way you can be sure that when a better component or part of the chain come up that you are still in the game of being the best.

JR: It is Imperative that you ensure that initial source material is of the highest quality, then just keep things simple. Accepting there may be a digital mixing desk involved, run it at the highest sampling rate you can afford, avoid unnecessary plug ins which can significantly reduce intelligibility over distance in exchange for a little sparkle at close range.

Digital solutions have been primarily created for convenience not quality, so be willing to go the extra mile, substituting an analog stage preamplifier or a line driver and copper cable for the length of Cat 6 the desk manufacturer has told you will be satisfactory.

Just remember, a KV2 system is a large open window on the sound. With each layer or covering you add to the window (additional parts added to the signal chain) you will lose a little of the view. But it will still be a great view and by applying conservatively, mindfully and intelligently, every time you look through that window, the performance soundscape you have created will make you smile!

Q: KV2 is a small company that doesn't seem to be interested in cutting the kinds of corners to offer products at the lowest price point, but rather is focused on high-end products with the accompanying price tag. How would you describe your target customer?

GK: We've done demonstration shows, and the clarity that people hear with a KV2 point-source is immediately heard by people when they compare it with a line array. They listen to multiple systems, do an A/B test and decide which sound better. I tell them "hey guys, it's your decision, I won't tell you what to do!"

GK jr.: A customers that cares about the atmosphere that he brings to an event. Sound is not something touchable. The customer that is for us is the one that wants to bring smiles to the crowd, not to interfere with the message that the artist has prepared for his crowd.

As for the price tag question then I must also say that for a full system (not price per one box) the price is actually lower then the leading brands and always will be. Purely from the fact that if you have fantastic control over all of your system the system can actually be much smaller and at the same time sound ***much*** better.

AAB: Operators who truly care about the sound quality first.

Q: Where do you see the live sound industry in the next 5 years–what are your predictions for the future?

GK: I think that point-source is going to come to the fore. It sounds better and it's a more compact solution. In the future, people will go for the quality.

GK jr.: I believe that the trend to have things cheaper and easier is going away. People are actually wanting higher quality and demanding it. I believe that this has now come to a point where people do really want to go back to a higher quality. And the fact that KV2 offers even a better level than ,back in the good old days' is a huge relief to them. Using the best technologies available , not necessarily the newest ones, is the key. That makes sure that the whole chain is as good as can possibly be.

JR: The focus has got to move more towards sustainability and genuine eco solutions, which would actually benefit KV2. Using less speakers and amplifiers to cover larger areas is already in KV2's DNA and a much greener solution than the majority of our competitors. Add in the long life expectancy of a KV2 system, reduced transport requirements and set up times and it becomes a highly relevant if not revolutionary alternative for the future of sound reinforcement. At that point the exceptional sound quality advances would be a bonus!

With Line array we have created our own industry problem akin to plastic waste with millions of these cabinets now decommissioned and piled up somewhere. The industry giants run by accountants and shareholders have ruled for years, insisting on re-investment into the 'latest systems' to retain 'preferred dealer status' or rider friendliness. That is simply not sustainable as either an economic or environmental model and dealers as well as end users deserve better. It has been our strap line for many years, but actually our ethical approach of trying to offer long lasting systems with the very best sound and the very best value will hopefully go full circle seeing KV2 Audio deservedly earn its place as 'The future of sound…. Made perfectly clear.'

Q: Finally, any words of encouragement for the next generation of live sound professions?

JK: If you want the best of both worlds, you need digitally-controlled analog. Anybody who can get into that business will take over the high-quality market!

GK jr.: When picking a system or even a part of it, the best recommendation is to trust your ears. Ask for an actual A/B comparison test. Sound is not something touchable and not something that can be set on a chart. It is an emotion. Human hearing is on a much higher level than what the measuring equipment of today is capable of, and much more complex. Ease models, SPL, and other specifications are not at all about what the system actually sounds like.

JR: Believe what you hear, not what you read or are told. The teaching in these modules have shown you that it is simply not possible to measure with existing technologies the temporal dimension that we can hear, or that a KV2 Audio system is capable of reproducing. Wherever possible do an A/B–in the same room, using the same high quality source. Simply switch the speakers for a competitor's alternative and you will be absolutely reaffirmed in your endorsement and deployment of KV2 products as the right choice for most applications.

AAB: Trust your ears immediately, not graphic displays and marketing word speak.

List of Figures

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