Technology

Super Live Audio Technology (SLA)

- A New Standard
- Definition and Distance
- Dynamic Range Vs Digital Sampling
- Electronic Integrity
- Speaker Design
- Sound System Design

The Future of Sound.
Made Perfectly Clear.

At KV2 Audio our vision is to constantly develop technologies that eliminate distortion and loss of information providing a true dynamic representation of the source.

Our aim is to create audio products that absorb you, place you within the performance and deliver a listening experience beyond expectations.

Contents

A New Standard

After years of research and development, KV2 Audio is pleased to announce a new standard in live sound reinforcement. Super Live Audio or as we refer to it 'SLA' has been developed through KV2's efforts to achieve the highest possible dynamic range and the lowest possible losses, caused through distortion or the altering of signal as it passes through the audio chain. Further to this, rather than develop technologies that try to compensate or fix problems in a system's design, KV2 focuses on building systems that are inherently superior from the start.

Our SLA standard reproduces high sound pressure levels in large spaces whilst delivering true dynamic range and source representation. There are a number of factors that KV2 have identified that make up SLA and the resulting benefits it provides to the listener. These factors include electronic integrity (settling time), digital sampling rates, pulse response, dynamic range and sound system design.

SLA
Super Live Audio

SUPER DIGITAL

- 20MHz A-D Extremly High Resolution
- Greater than 120dB dynamic range
- Very low non-Harmonic Distortion

SUPER ANALOG

- Very High Dynamic Range due to settling time < 1 µs
- Superior Large signal Frequency Response of 200 kHz

SUPER ACOUSTIC

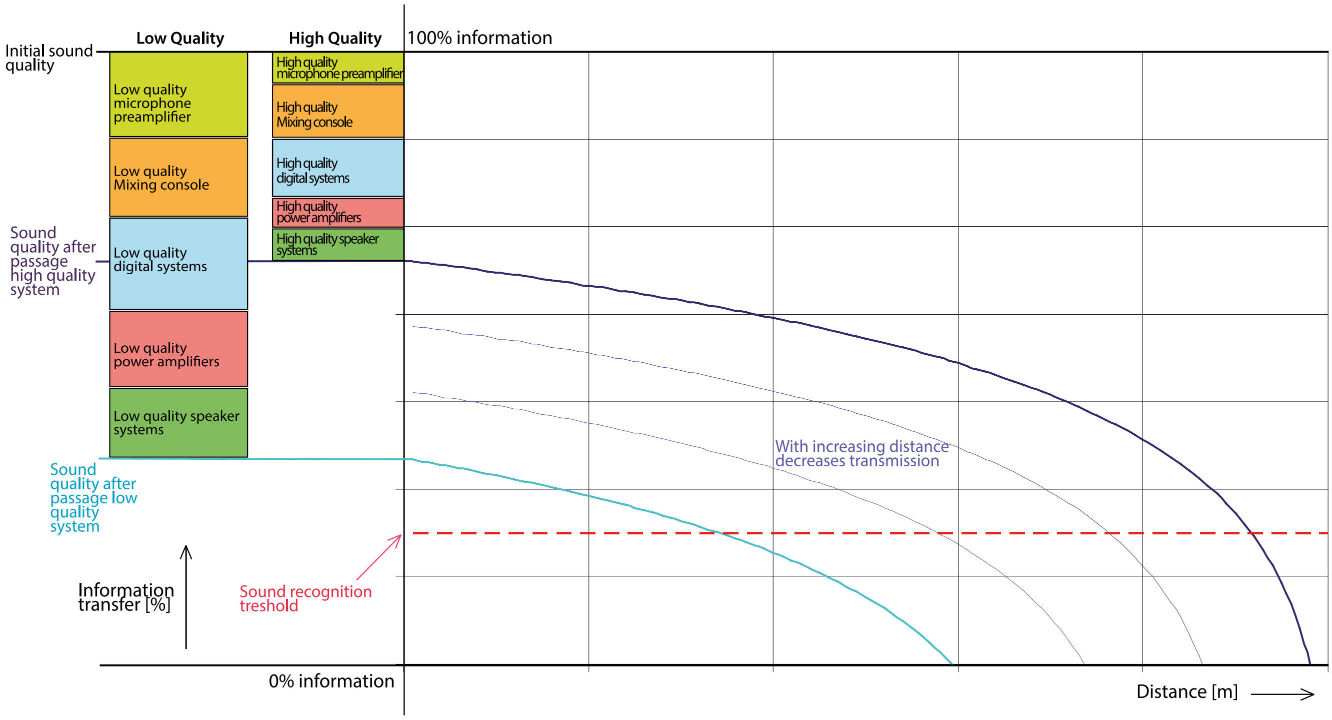
- True Point Source
- Zero inductance - Active Impedance Control
- Ultra Low Distortion, True Piston Motion Drivers

Definition and Distance

Fundamentally, the effect of a poor quality system comprising of inferior electronics, transducers and acoustic design is a lack of definition and detail, but equally important in a live audio situation is the distance in which a system can project clear defined audio. To maintain high-quality sound, especially at a long distance, it is vitally important that each part of the audio chain is of the utmost integrity.

The system must be capable of transferring an unchanged sound, including the ambience of a performance over distance at the required level to provide the greatest possible experience for the listener. As the area of coverage increases, the demand grows for system resolution. Sound pressure levels can reach values up to 140 dB, so the emphasis is on the dynamic range of the system. The system must not color or influence the sound quality; it must have minimum distortion and a maximum dynamic range. SLA technology has come about through KV2's advancements in achieving these goals.

Standard electro-acoustic devices have a limited dynamic range and invariably produce distortion not related to the original signal (Non Harmonic Distortion). This distortion becomes apparent in the high frequency range significantly masking the weaker parts of the signal. This masking has the effect of erasing a large proportion of the detailed information thus causing a significant reduction in clarity. The artificially changed signal makes it impossible to transmit the ambience or real atmosphere of the original sound to the listener, particularly over distance.



Effect of distance on the quality of sound transmission with different quality sound devices

Lets look at how KV2 achieves maximum dynamic range and resolution to deliver unchanged and uncolored audio reproduction at high SPL over large distances.

Dynamic Range and Resolution Vs Digital Sampling

Live audio is very dynamic, yet a dynamic range of 130db is almost impossible for any digital AD and DA converter at the current sampling rates used to replicate. Dynamic resolution is directly related to the bit and sampling rates used in the digital conversion process.

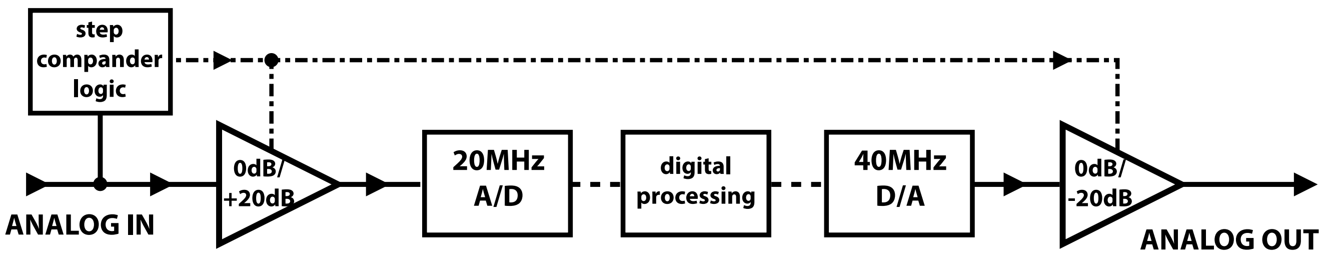
At the present time, commercial digital audio commonly uses multiple-bit digital standard sampling, (PCM, Pulse Code Modulation). Industry standards have determined that a 24-bit/96kHz sample rate (which corresponds to 2.8224 MHz 1bit PDM, Pulse Density Modulation, used in SACD), is adequate when professionally converting an audio signal consisting solely of harmonic signal components.

In practice however, the audio signal consists of many signals, therefore it is complex and its properties are actually closer to random signals. The spectrum of random signals is infinitely wide, so when converting analog signals to digital, the sampling rate must be as high as possible in order to maintain quality of the transferred signal in full resolution.

At the current sampling rates utilized in commercial audio system design it is evident that their resolution is compromised as the frequency range increases. Much of the detail and ambience captured in the sound is related to this area. A decrease of information due to sampling frequency affects the sound as a loss of brilliance and intelligibility while increasing high frequency sizzle and distortion. In addition, when using DSP, there is additional information loss due to reduced DSP power, precision and limited processing time.

By increasing sampling rates the amount of transferred information increases, but sampling rates cannot be unlimited, so digital audio can never reach the quality of a high definition analog system, however in some cases it is necessary to use digital audio in professional live audio. KV2 undertook a different approach to digital. They looked at an alternate conversion process developed by Sony and Philips called Direct Stream Digital or DSD. The Super Audio CD (SACD) is based on DSD and unlike pulse code modulation (PCM) technology used commonly in pro audio applications and normal CD recordings, DSD technology is based on a one-bit 'sigma-delta' converter that produces a stream of pulses. The amplitude of the analog waveform is represented by the density of pulses, and is called Pulse Density Modulation. The resulting digital bit stream is encoded at enormous 2,822,400 samples per second! (2.8224MHz)

Practical listening tests were undertaken by our engineers to determine the minimal sampling frequency required to eliminate any audible information loss. The result saw KV2 design a circuit based around DSD with a sampling frequency of an incredible 20MHz, when using 1 bit sigma-delta PDM modulation. KV2s new digital converter delivers resolution 7 times higher than SACD. A special step compander circuit adds a further 20db of dynamic range to utilize maximal range of the converter at low levels.

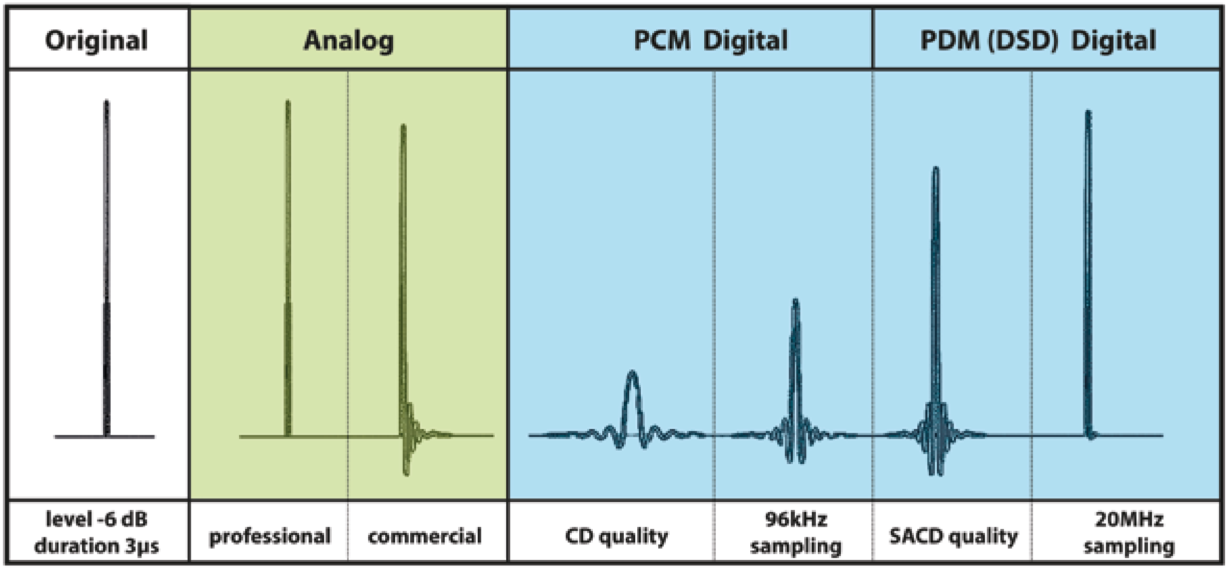


Step compander changes input and output level to increase dynamic range of the converter

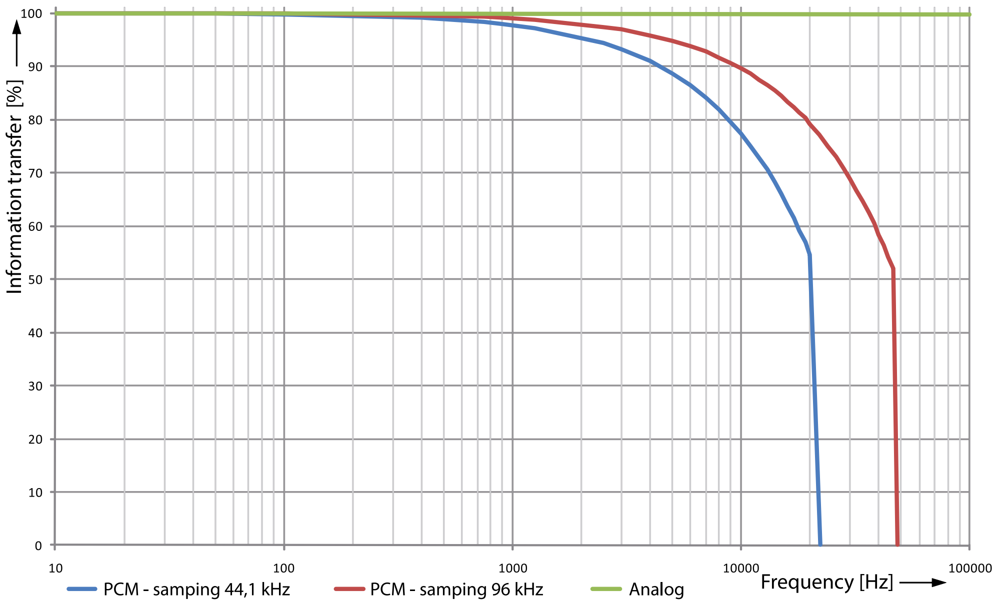
KV2 Audio utilizes hybrid signal processing, using the best of analog and digital technology to provide all necessary filtering, equalization and time alignment to their speaker systems. This best of both worlds approach provides unmatched dynamic range and audio reproduction.

Dynamic Range and Resolution Vs Digital Sampling

To maintain a high-resolution audio signal, it is also important for the system to maintain the shortest possible impulse response time. Impulse response time is effected by Settling time and circuit design in Analog and by Samplig frequency in Digital. In the Figure below it is evident that commonly used commercial systems, particularly digital, cannot pass the full resolution of the original signal. Impulse response time refers to the transmission system deficiencies, which is effectively the REAL distortion of the system. Systems with long impulse response are unable to transfer high dynamics and high definition signals. SLA systems incorporate hybrid signal processing at an industry leading sample rate of 20 MHz and settling time of 1microsecond, to ensure audio reproduction with the highest possible resolution and definition..



Impulse time response audio systems to input pulse, level -6dB, duration 3µs

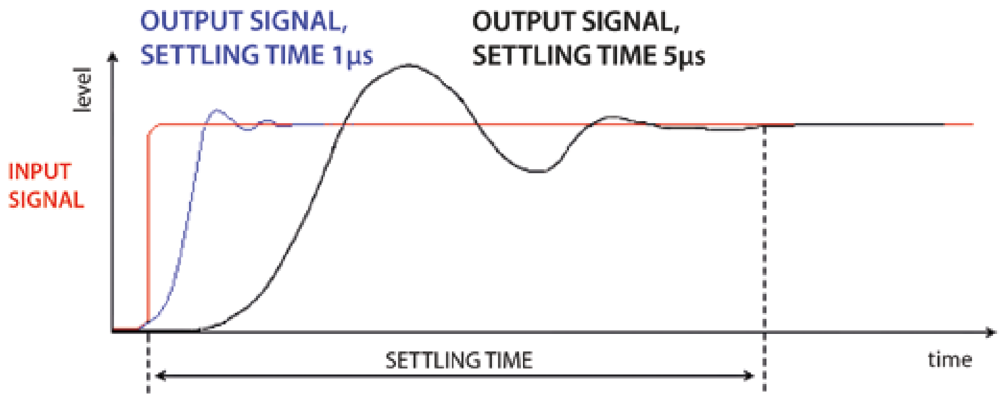


Information transfer of digital and analog sound systems

Electronic Integrity

Settling Time

The most common active electronic component used in audio engineering is an operational amplifier often referred to as an Op-amp. The settling time of an amplifier is defined as the time it takes the output to respond to a step change of input.



Definition of settling time

An Audio signal requires a fast settling time due to the signal constantly changing. Errors due to settling time produce noise level, non-harmonic distortion, (not related to original signal). This noise level increases with slower settling time, this noise level is defined as a ratio between system settling time and the highest transferred frequency. The settling time of the system feedback loop **must be less than 1µs (microsecond), to maintain high quality audio**. If this is not maintained, the system creates and introduces noises that are not related to the original sound. This is an element of distortion added to the signal that is rarely talked about, but its effects are much more significant and dramatic in damaging the audio quality, than any losses due to harmonic distortion, a figure commonly published by most manufacturers.

The Settling time of common electronics systems used in most commercial sound systems is around 10 µs, ten times longer than it should be. The distortion, created by slow settling times is not commonly discussed by many manufacturers as they fail to understand its significance, often overlooking it in providing the technical specifications of products. Moreover the noise this distortion adds is very often mistaken for original high frequency, especially in digital technologies where it can exhibit itself as a bright, "fizzy" high end.

Settling time is less of a problem on simple signals because it occurs with addition to the higher frequency of the original signal. However, when introducing a complex signal, due to its long settling time there is now a high level of distortion from the many different, prolonged scrambled elements of noise, which then create a cacophony, masking the weaker original signal nuances.

That system has a low resolution.

Settling time of Operational amplifiers therefore also has a large effect on a system's ability to trans fer information accurately, particularly in the higher frequencies. SLA circuitry only utilizes components with settling times of 1 microsecond or less to ensure a high definition low distortion signal path.

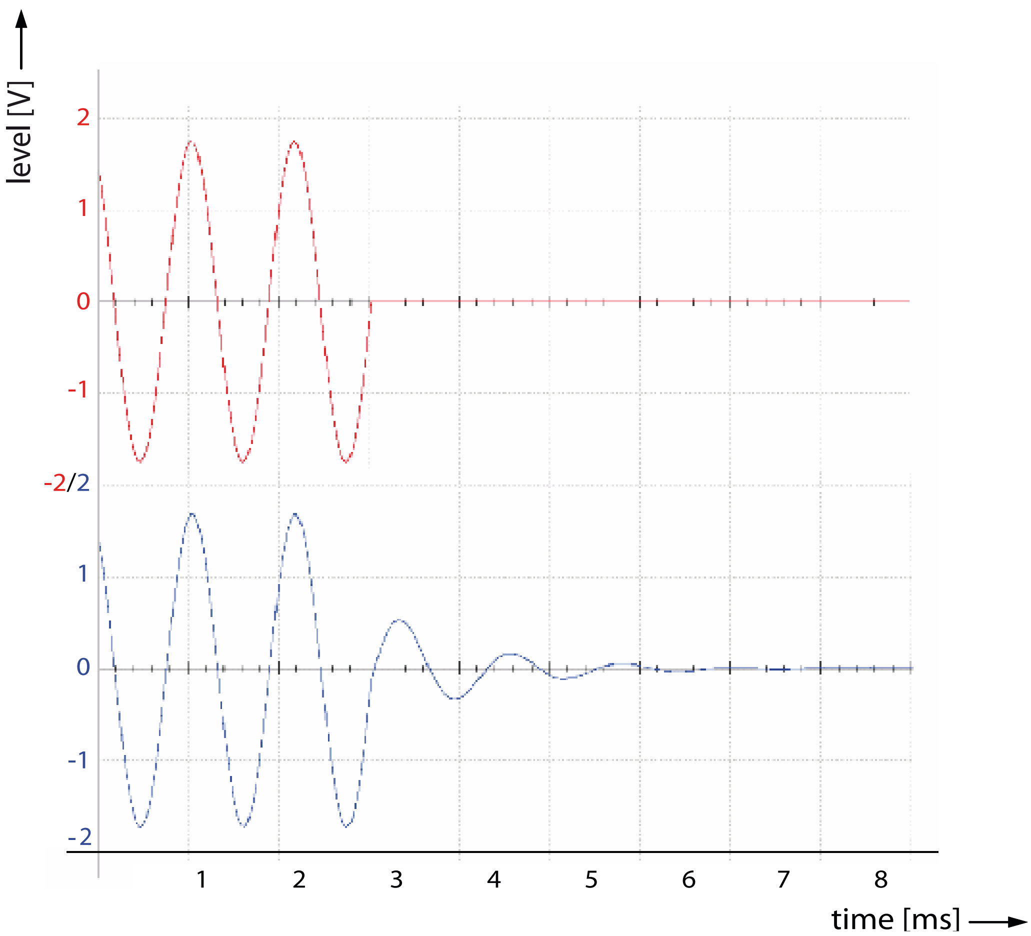
Electronic Integrity

Speaker Pulse Response

The main parameter of the speaker is pulse response. Parameters of the speakers are almost exclusively measured by a continuous sine wave signal. But continuous sine waves are not identical with regard to music signals. Music signals consist of many signal types, basic tones, harmonics and noises, (The best represented by pulse response). When a speaker is measured by continuous sinus, it has time to swing at higher frequencies, although the speaker has no control over diaphragm mass and motions.

One of the most important parameters in transducer design for Super Live Audio Systems are the removal of unwanted resonances. These resonances are usually caused by the mechanical design of the speaker and its failure to control the diagraph motions. Resonances reduce overall definition by masking smaller signals and producing tones not related to the original signal.

The figure below shows an original sine signal (red, top) with its sharply defined end and the same reproduced signal (blue, bottom), still oscillating after the signal stops due to poor control of speaker mass. Poor pulse response has a very negative effect on the ability of a speaker to reject feedback. SLA systems feature exceptional feedback rejection and this in part is due to their excellent pulse response.



Effect of speaker's resonance, original signal (red, top) and reproduced signal (blue, bottom)

Speaker Design

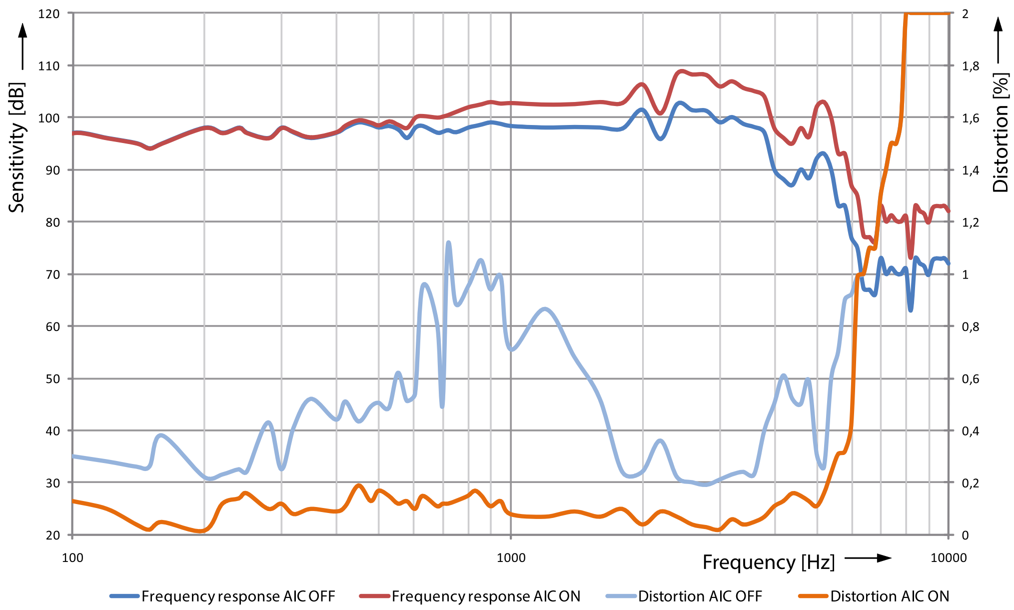
Active Impedance Control

Additionally, control over the speaker mass can be very positively impacted by using an active impedance control, (trans-coil) speaker system. This system utilizes a secondary stationary coil, which reduces inductance close to zero and dramatically improves pulse response. Inductance is the main reason for odd harmonic distortion.

Low Inductance = Low Non Harmonic Distortion

The active impedance control or AIC is an additional fixed, multi turn coil, positioned in the loudspeaker magnetic circuit gap. This coil is almost as long as the gap height and being wound around the pole piece to be very close to the voices coil. A current flowing into this coil generates a magnetic field that is in opposition to the field generated by the moving coil. This cancels out most of the voice coil inductance and reduces the flux modulation and inductance modulation. The A.IC device can be seen as an "active" shorted ring in the gap. The two A.IC terminals allow driving the additional coil in many different ways according to specific application needs.

Producing very high quality speakers for audio systems with minimal distortion has created the need for enhancement of the electronics for an SLA system design. In reality few manufacturers are able to utilize AIC Trans-coil technology, because upon testing they realize that it immediately shows up the fundamental flaws in their own electronics designs.
Even greater compromises in audio quality occur when utilizing digital processing to try and correct their acoustic design.



Frequency response and distortion curve with AIC ON and OFF

Sound System Designs - Multi Point Vs Point Source

Sound System Designs

There are two main types of sound system designs that have been prominent in the market consisting of single point source or multiple point source concepts. Multi point source arose from the requirements for very high output power. Multi point source arose from the requirements for very high output power. The idea satisfied that criteria, but with the increasing number of sound sources came an overall reduction in the quality of the sound. The two big disadvantages of multipoint source systems were the suppression of the high frequency output and the physically time-shifted outputs from the individual speakers. Adding a number of time-shifted outputs from individual speakers together causes poor system impulse response.

Multipoint two axes system **Multipoint one axes system** **One point system**

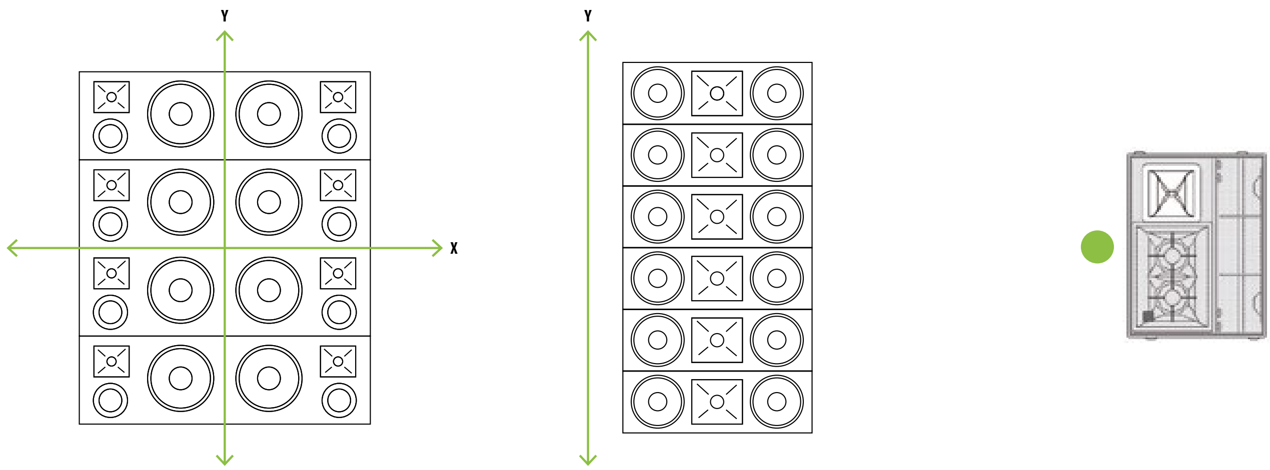


Illustration of two axes source, one axis source and one point sound source

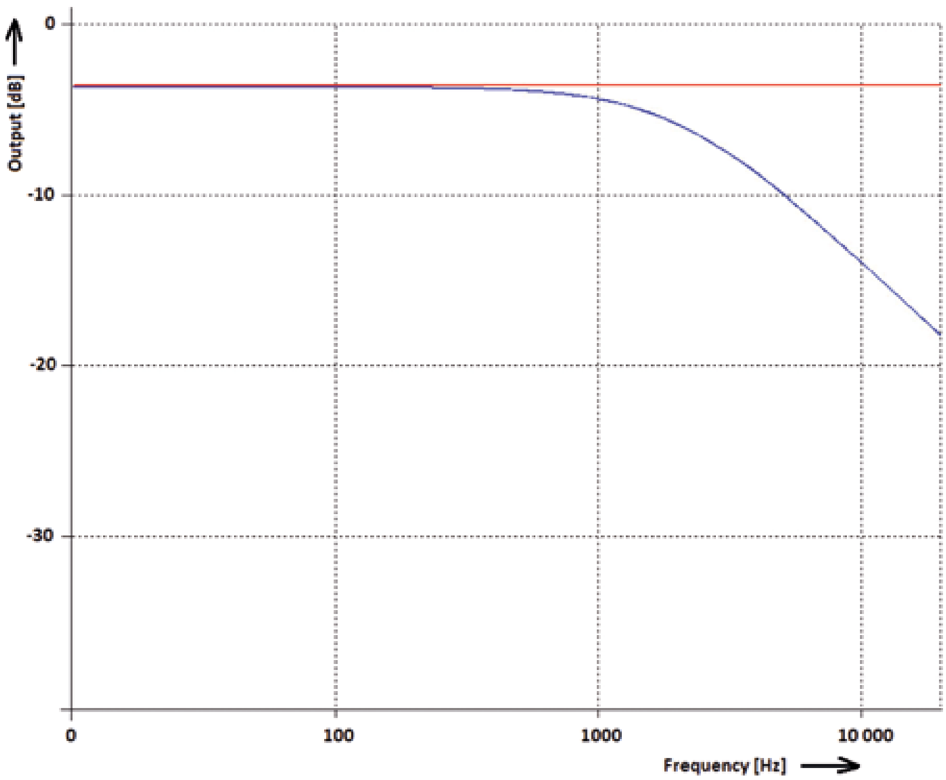
The first types of multipoint sources were simply a large pile of cabinets, stacked together like building blocks and intended to array on all axis'. A major improvement in the next generation of systems was the introduction of multipoint, one-axis systems that provided better frequency response and increased definition than previous multi axes systems. Unfortunately, whilst a step forward, the frequency response and impulse responses were still not ideal and the coverage was often inconsistent. A typical representation of the one axis multipoint source sound system used commonly today is a line array system.

Sound System Designs - Multi Point Vs Point Source

Single and Multi Point Source Frequency Response

Line array does reduce the effect of multipoint sources interfering with each other like the systems of twentyfive years ago, but it is still a long way from the superior results achievable with single point sources.
A single point source sound system offers the highest possible definition and the highest possible dynamic range available today. High intelligibility is a by-product of this, but is only guaranteed by maintaining this high definition and high dynamics through the use of fast and accurate electronics, with low distortion transducers. The frequency response graph below show that when using even a one axis multiple sound source like a line array (blue), the audible suppression of higher frequencies starts as low as 2kHz. Higher frequencies reduction is caused by mutual subtraction of individual sources of multi point source. This reduction is typically -16dB @15kHz, for flat frequency response is neccessary extra equalization. This reduction varies with position of the listener in space and air movement.

Conversely when using one single point sound source solution the frequency response is not affected in the same way and is actually very flat (red).



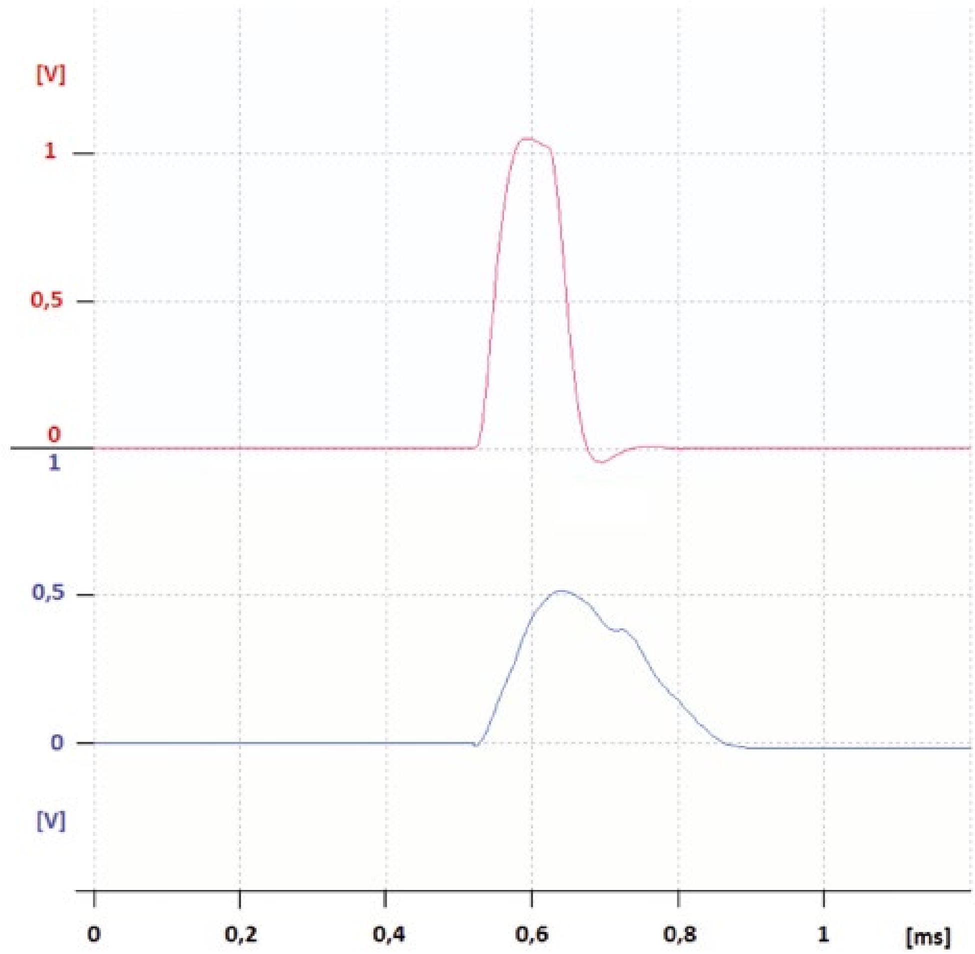
Frequency response of the one axis multipoint sound source (blue) and point source (red)

Sound System Designs - Multi Point Vs Point Source

Single and Multi Point Source Impulse Response

To maintain a high-resolution audio signal, it is vital that the system is able to exhibit a short impulse response time. This will create a sound signal like the original.

The figure below shows the comparable impulse responses of the one point source and one axis multipoint source (Line array). The Input pulse is 1V, pulse width 100 µs, period 10 msec. The damaged impulse response you see in the graph is the reason for the line array systems low resolution. Damaged impulse response is caused by mutual subtraction and addition of individual sources of multi point source system.



Impulse response of point source (red) and multipoint one axis source system (blue)

Sound System Designs - Multi Point Vs Point Source

Time-Shifts, Properties of Multi Point source

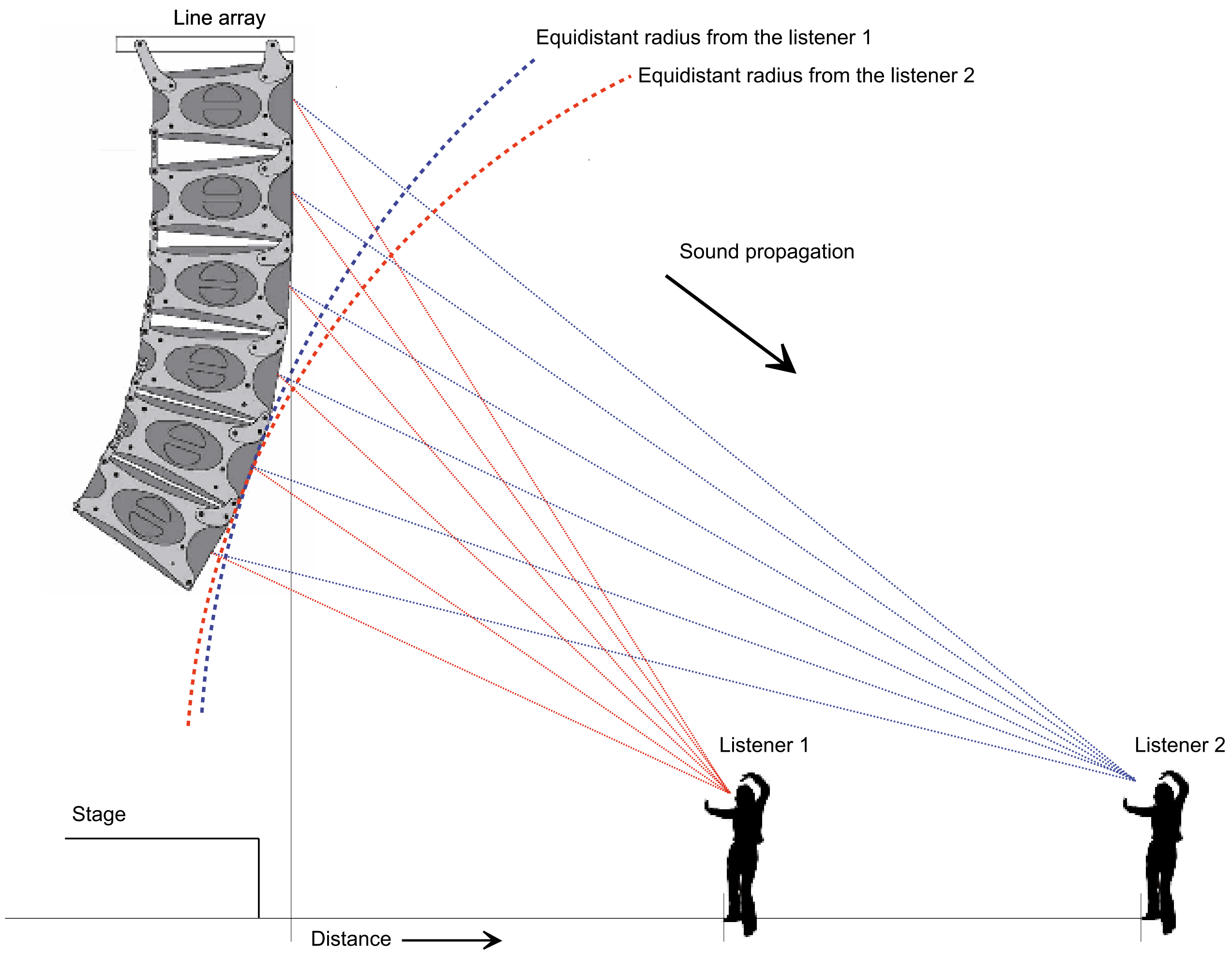


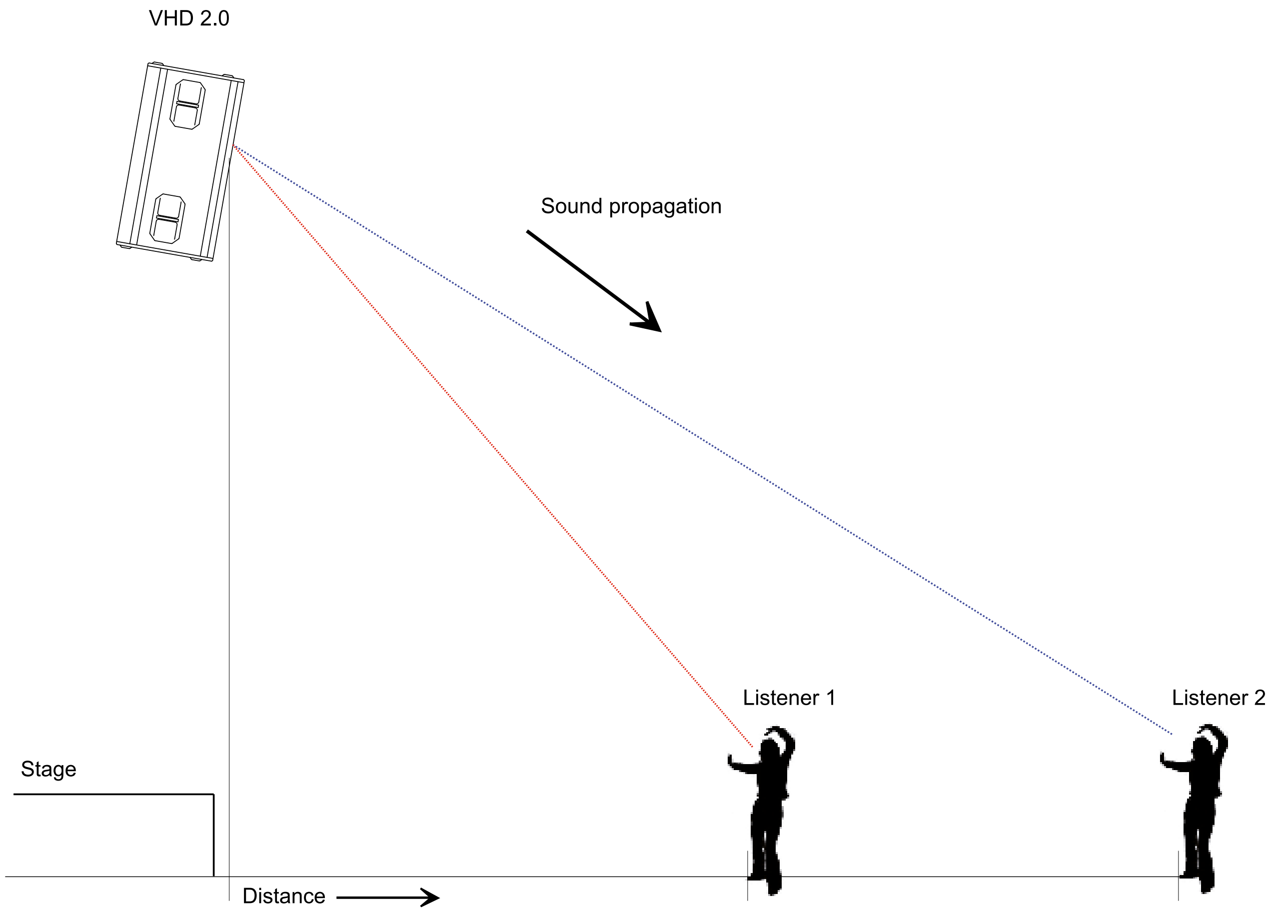
Illustration of the differences in distances to the listener from several Line array sources, each listener gets a blurred sound

The figure above shows that the physical characteristics and dimensions of a multi-point system determine the time shifts between several sound sources.

Transfer response and Pulse response will vary with the location of each individual listener. Time shifts for listener 1 are different to those for listener 2. Many manufacturers claim that time shifts are corrected using digital signal delays but this does not provide a solution because time shifts will infinitely vary with each new listener location. Even more critically, one factor over looked by the calculations and predictions of system engineers or line array prediction software is the RANDOM movement of the air in the listening area which causes huge changes in the transmission properties of multi-point systems regardless. This is the case when an audience arrives, after the system engineer has spent a whole day unnecessarily aligning the system to an empty but perfect theoretical environment – an environment that in the real world of random air movements and increasing temperatures will not exist.

Sound System Designs - Multi Point Vs Point Source

Time-Shifts, Properties of Multi Point source



When using one point sound source, the listener in any location gets only a pure (not blurred) sound

It is clear from the diagram above that the issues of interference and random air movements will simply not affect a one point sound source, like KV2 Audio VHD and ES, regardless of their complexity or intensity. This will give a more equal coverage across the listening area with each individual listener enjoying a pure sound.

The Future of Sound.
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