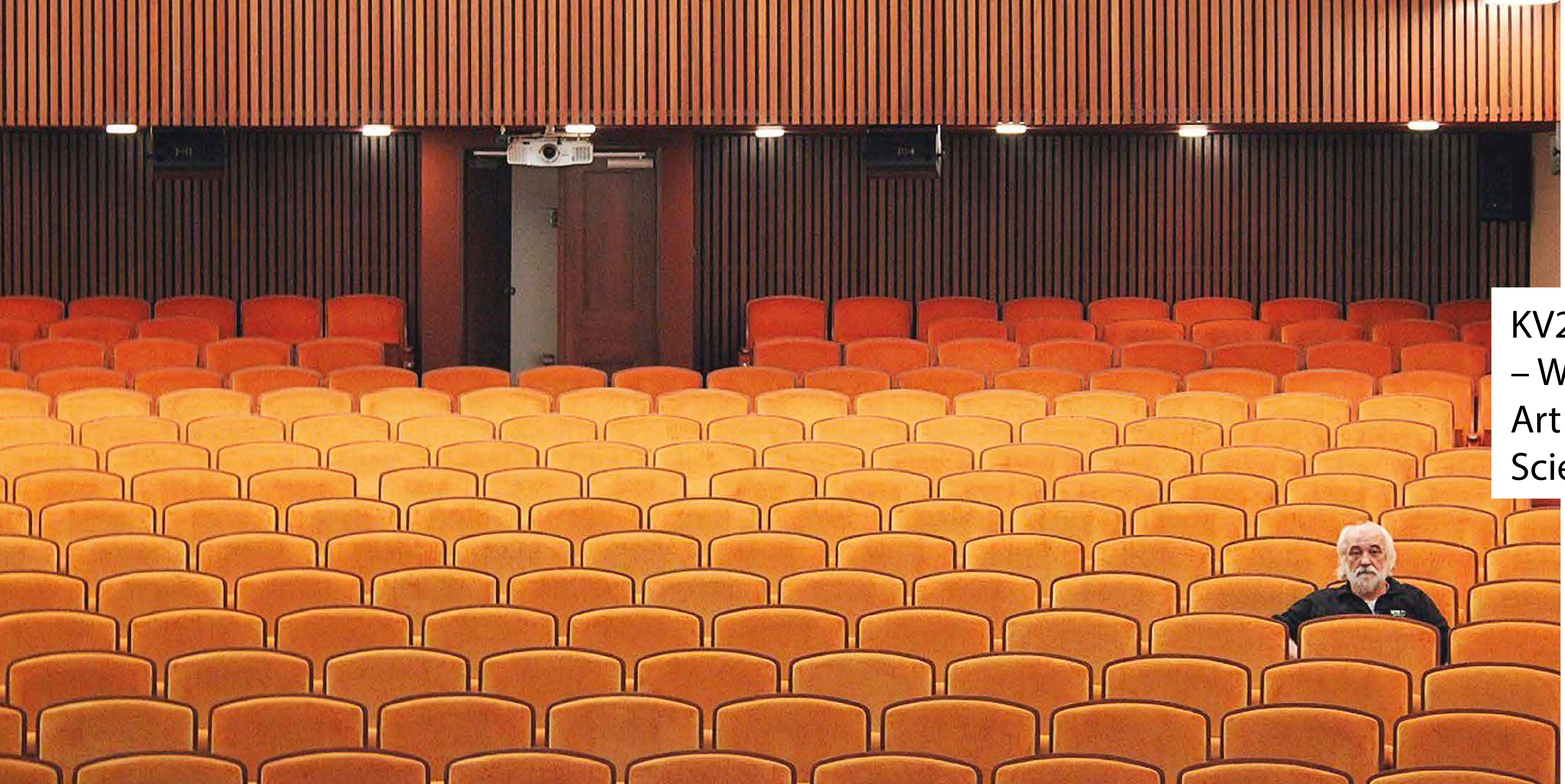
KV2 Audio - Where Art Meets Science



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“We have bucked industry trends and broken industry standards to find the best possible audio solutions both analog and digital. We don’t simply gauge our system’s performances on published specifications; we gauge it by the smiles on people’s faces.”

Just for a few minutes, forget everything else you have read about speaker system design and think about what truly defines clear, quality sound.

Today we live in a world that has compromised audio quality. Technological advancements now try to bend the rules of physics, focus on slick user interfaces and create virtual equipment in an effort to save space and money.

Digital Sound Processing is everywhere, manufacturers strive to control every aspect of sound reproduction but all the while it moves us further and further from the original source, its timbre and dynamics.

Capturing the emotion and ambience of a performance has become secondary. At KV2 the true reproduction of the original sound and its dynamics are the key elements in the development of our products. We have bucked industry trends and broken industry standards to find the best possible audio solutions both analog and digital. We don’t simply gauge our system’s performances on published specifications; we gauge it by the smiles on people’s faces.

As we move through the digital evolution the limitations of our imaginations have expanded into the virtual but the laws of nature can’t be denied. At KV2 we strive to bring art and science together, to reveal the true emotion in a performance.

Our products are designed not just to provide a solution for sound reproduction; they are built with the intention of providing the optimum listening experience and enjoyment for the audience. To deliver something special, something beyond expectation. KV2 builds plug and play systems that save you money, are fast and easy to set up, suitable for venues of all sizes and simply provide superior sound quality to any competing product.

With incredibly low distortion and extremely high definition, KV2’s point source systems give you the capability to cover more people with less equipment delivering real value in quality sound reinforcement.

KV2 today is the culmination of one man's life long search for perfect sound.

George Krampera

George Krampera is truly one of the audio pioneers of modern times. He has been building audio equipment for nearly fifty years, millions of people have experienced sound produced from equipment he has designed. Through his long and successful career, George's vision has remained unchanged, to eliminate distortion and loss of information in the signal path, thus providing sound reproduction that has true dynamic range and representation of the source.

Growing up in the Czech Republic, his father a notable technician, as a child George played with valves and other components, building his first radio before he was ten. His interest in music and sound grew and by the age of fourteen, George was building power amps and other equipment for local bands in Prague. George continued his electronic training after leaving school and worked as a technician in Prague repairing various pieces of equipment while retaining a strong connection with the local music scene.

KV2 Audio Research Team

The Russian invasion of Czech in 1968 brought tough times for the country and as George built his career, he became increasingly concerned about the future of the country and the safety of his family. In August of 1983 George loaded his young family into their car and left everything they owned behind to escape to Austria. Once safely across the border George was granted asylum status and a few months later they departed for Canada.

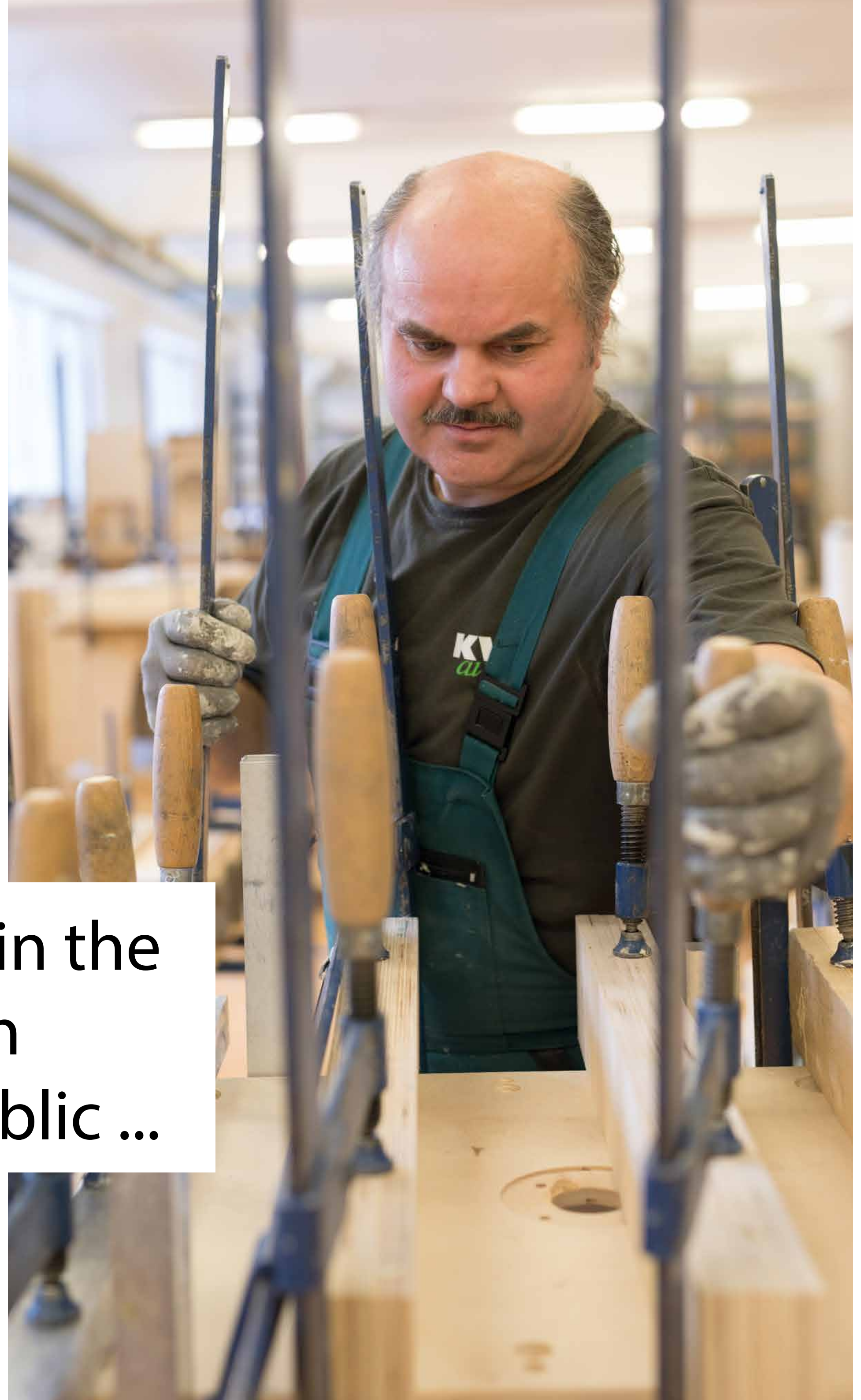
It wasn't long before George found himself drawn back to the music industry taking a job with Yorkville Sound. There George designed many of their guitar and keyboard amplifiers and a complete range of processed speaker systems. After leaving Yorkville, George started his first company Rexx, which built high quality solid-state guitar amplifiers. In their first year of production Rexx out sold Marshall in Canada, testimony to the quality and value of the product. After a change in ownership Rexx fell on hard times and George decided to head back to Europe taking up a job offer with Italian speaker manufacturer, RCF.

George's goal at RCF was to work with and improve transducers. This was the final part of the chain he felt he needed to master to achieve perfect sound reproduction. Heading a team of young engineers, George made a number of break throughs at RCF including the development of the silicone spider. He also designed the complete ART active speaker series, which was a hugely successful line for the company. George left RCF when it was sold to Mackie and continued his work in transducer development at B&C.

By the late nineties George had relocated back to his homeland of Czech forming his own pro audio design company, Class A. Here using B&C components he worked on a large format active system that could cover big crowds and distances with optimum quality and clarity. In partnership with one of his old team from RCF, Marcelo Vercelli, the pair started Fusion and took their new speaker system to a large US trade show where Greg Mackie was suitably impressed by the products and made them an offer to join Mackie.

George found himself back at RCF, which was then owned by Mackie and in his role there apart from putting Fusion into production, he designed a range of speakers for Mackie. Sales of these new boxes skyrocketed but George's real passion was for designing high quality equipment. George and his old partner Marcelo left Mackie to form their second company, which was clearly represented by its name, K for Krampera, V for Vercelli and 2 being their second venture. A third integral link came about with another of George's old RCF colleagues, Andrea Manzini, who became involved with a new transducer manufacturer called Eighteen Sound.

Working closely with Andrea, George co-designed components to match the products he wanted to develop. This strategic alliance along with his expertise in cabinet design and electronics, built the perfect platform for George to follow his vision, a vision realized today in the products of exceptional performance and quality built by KV2. George has now established a group of brilliant young engineers, who he has mentored and directed to continue his legacy and ensure his vision will continue well into the future.



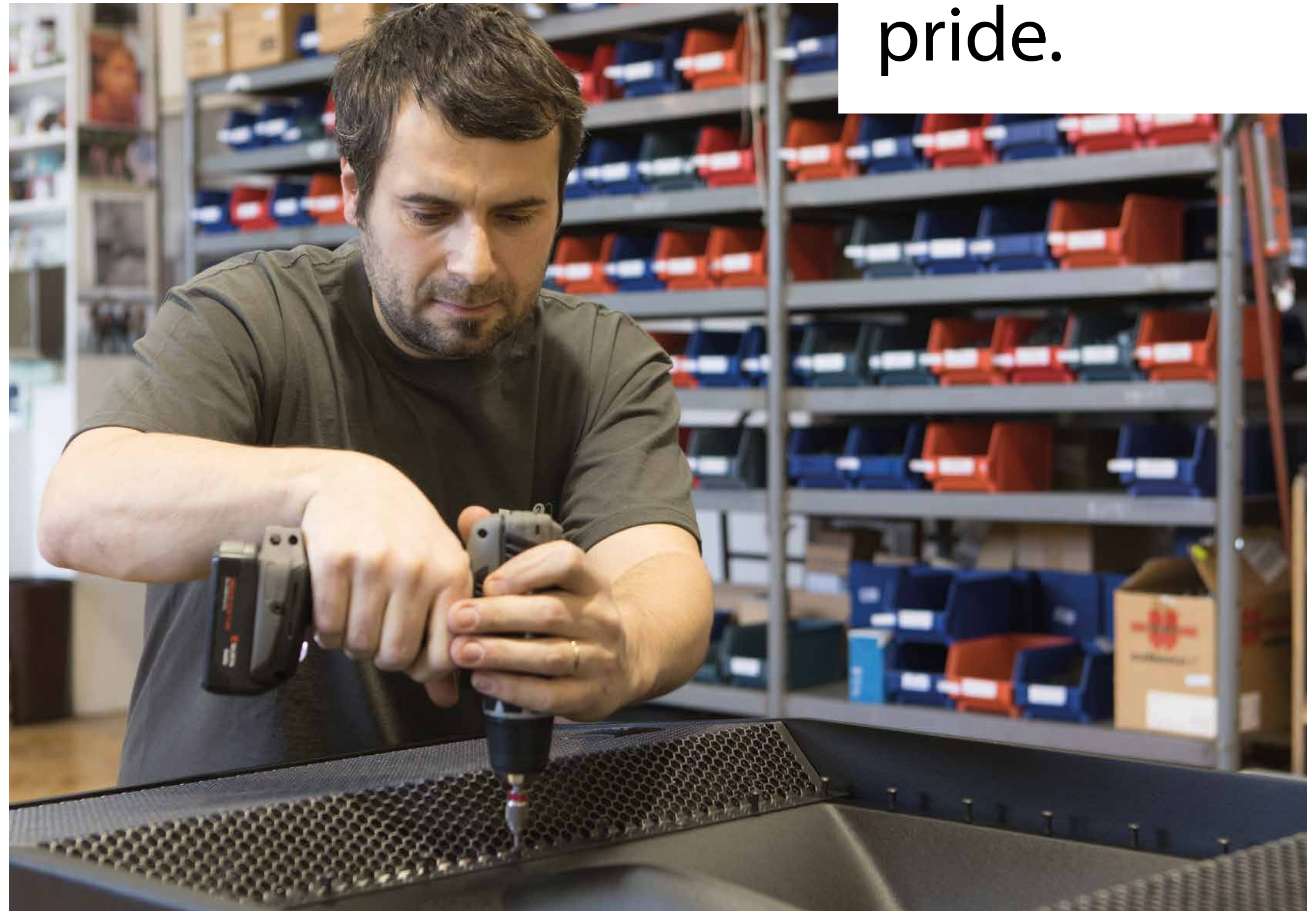
Built in the Czech Republic ...

Located in Southern Bohemia,

Located in Southern Bohemia, KV2 represents many of the unique aspects that are seeing Eastern European manufacturers prosper in these difficult economic times. It's about the people, the place and strong traditions in education, the arts and industry.

Prior to World War II, Czechoslovakia was one of the most industrialised countries on earth. The subsequent invasions of the country bankrupted its economy. Today the country is again prospering thanks to the resilience and hard work of its people. Unique in its make up, KV2 draws on it's Czech heritage. KV2's employee's have a strong connection with what they do and a genuine interest in the company, with a desire to build equipment to the highest standard for end user's around the world.

Just down the road from KV2's current location George's original R&D lab still stands. It was here George developed the original range of KV2 speakers and the journey began to create the best speaker solutions available. KV2's range of products grew and it was not long before manufacturing was moving from one building to the next to meet demand. KV2 recently purchased a factory that would handle their growth for many years to come.



... with passion and pride.

This 100,000 sq foot facility is fully renovated to meet KV2's needs. We have implemented state of the art manufacturing lines and a high quality paint shop. We built one of the largest anechoic chamber in the world for testing and expanded the current R&D lab, along with new offices and warehousing. But despite all this, we are still located in the small town of Milevsko, where it all began, employing local people and building on those strong Czech traditions.

The new R&D team is made up of outstanding technology graduates, specialists in acoustics, electronics and digital technologies. George has passed on to them years of experience and understanding of all things audio. This legacy is leading to ground breaking, innovative product development.

KV2 is not simply a large corporate company pumping out products at the lowest possible price point to maximise profitability. It is a small hands on manufacturer, focused on quality and most importantly the sound of its products, something that many companies seem to have overlooked in the race to develop the latest so-called 'advancement' in technology.

KV2 celebrated its twentieth birthday in 2022. It is still the same small vibrant company it was when it started. A company with a passion for building high quality pro audio products; products that deliver performance beyond expectation.

At KV2 We Believe Less Is More - But We Give You More For Less

Our Point Source Systems cover more people with clear detailed audio using considerably less equipment than our competitors. We embrace a 'less is more' philosophy while delivering you more for less, saving you time, money and increasing your return on investment. Transport costs and power requirements are also reduced making KV2 the green choice for today's carbon-conscious world.

Our systems are truly plug and play, no analysing software, external processors, or third party amplifiers required, making set up quick and simple, even for the inexperienced operator. Designed from the ground up and built with pride in the Czech Republic, KV2 Audio delivers unmatched quality and value for money.

The KV2 Difference

There is much more to creating high quality audio than simply building a good speaker system. Most manufacturers have access to the same components and acoustic designs so why is there such a disparity in the results achieved? The answer lies in taking a holistic approach to the whole audio chain. Let us explain the KV2 difference and why we developed our own standard.

Lower Transport Costs.

PLUG AND PLAY

True Plug & Play – All processing and amplification in one package.

LESS POWER MORE SOUND

A Greener Alternative – Reduce your carbon footprint with smaller transport requirements and considerably lower power consumption.

MADE IN CZ

High Quality European Built – Made with passion and pride in the Czech Republic.

A New Standard in Live Audio

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 acoustic system design.

SLA Super Live Audio

SUPER DIGITAL

- 20MHz Sampling - Extreme Resolution
- Greater Than 120dB Dynamic Range
- Very Low Non-Harmonic Distortion

SUPER ANALOG

- Super Fast Circuitry (1µs Settling Time)
- Ultimate Headroom - 200kHz Capability

SUPER ACOUSTIC

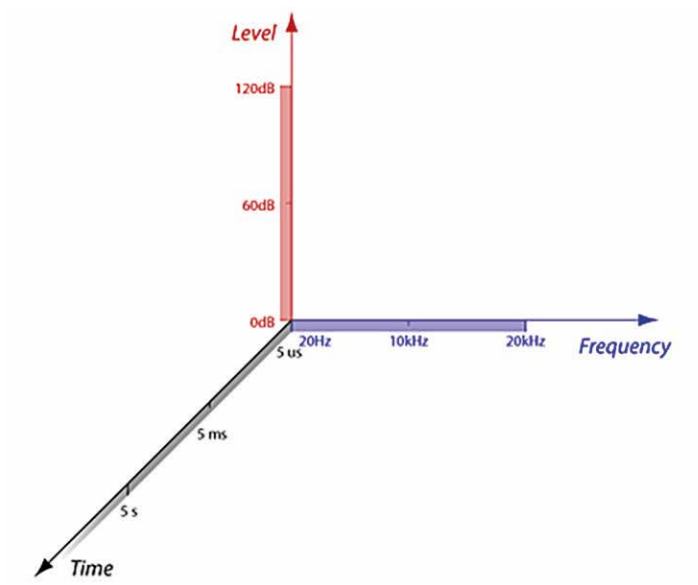
- True Point Source
- Active Impedance Control - Zero Inductance
- Ultra Low Distortion - True Piston Motion Drivers

Three Key Elements of Sound

To understand how the principles of SLA provide superior audio performance, we first need to look at the three main parameters that make up sound - frequency, level and time. While this may be common knowledge to most, we are often surprised how people focus on certain specifications like frequency response or sound pressure level, without truly understanding their relevance in respect to a system's overall performance. All three sound elements need to be properly replicated to achieve the optimum in sound reproduction and reinforcement.

By looking at the limitations of human hearing, we have a specification that if met by the systems performance will provide natural, uncolored sound to the audience. As shown in Figure A below, normal human hearing is from 0 to 120 dB+ of signal level and 20Hz to 20kHz in frequency range. What is often neglected is the importance of resolution in time.

Human hearing is able to recognise time definition, (the difference in incoming sounds), down to 10 microseconds and latest research has found that it may be as low as 5 microseconds. Much of a sound's spatial and directional information is directly related to the time component of the signal. For these reasons, extremely fast circuitry and high sampling rates are required to ensure total reproduction of the soundwaves arriving at micro second intervals to the microphone.



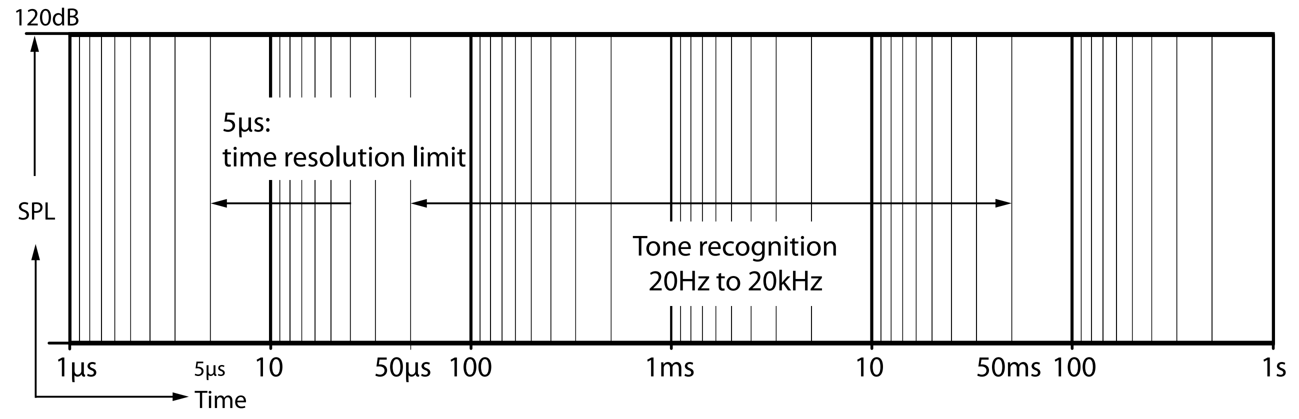


Figure A

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 quality of each component in the signal path will determine the amount of information loss.

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 and dynamic range. These factors will be determined by the quality and speed of the attached electronics, digital sampling rates, transducers and acoustic design, all of which are key elements of SLA.

Dynamic range is a system's ability to reproduce the softest signals to the very loudest. In this context, the different signals captured from multiple sources on stage may vary from the threshold of hearing to over 120dB and they should all be replicated accurately by the system relative to the engineers mix of those sources. It is therefore a requirement that when the system is operating at high SPL it has the ability to clearly transmit the low level intricate detail of the performance. For example, we should be able to hear the breath noise of a flute player through the volume of a drum kit.

Dynamic Range is not a pre-requisite of a system's SPL capability, high SPL does not directly equate to high dynamic range. In fact many systems are delivering high amounts of non-harmonic distortion when operating at high levels. While this may exhibit the system has high SPL capability, 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.

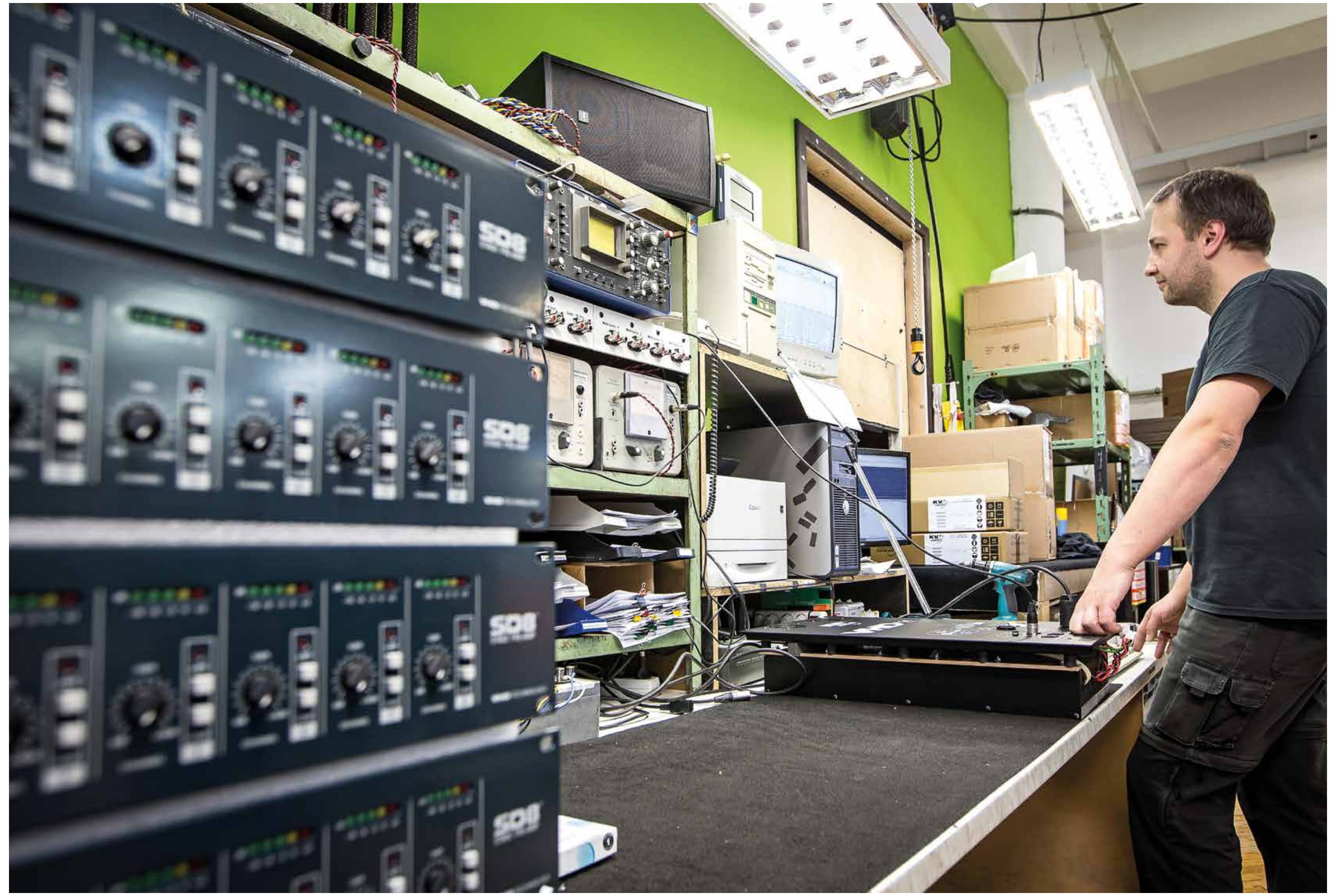
Most discussions relating to sound system design revolve around level and frequency response but fail to consider one of the most important factors – time; the speed at which the electronics and digital converters can process audio signals without loss or distortion.

Dynamic Range and Resolution in The Digital Domain

As previously stated, live music has the capability of producing a dynamic range in excess of 120dB. To reproduce this through an audio system with a suitable degree of headroom, a dynamic range capability of around 130dB is required. It is impossible for most digital AD and DA converters running industry standard PCM (Pulse Code Modulation) digital conversion of 24Bit/96kHz to replicate this level of dynamic range.

Secondly, while a 96kHz sampling rate has been deemed adequate when professionally converting an audio signal consisting solely of harmonic signal components, analog audio signals have complex harmonics and overtones and therefore should be regarded as 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 KV2 we undertook a different approach to digital to overcome the inherent problems in existing systems. We 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 this digital format and unlike PCM conversion, DSD technology is based on a 1 Bit Sigma-Delta converter that produces a stream of pulses. The amplitude of the analog waveform is represented by the density of pulses and this is called Pulse Density Modulation (PDM). The resulting digital bit stream is encoded at an 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 using a 1 Bit Sigma-Delta PDM converter. KV2s new digital converter delivers resolution seven times higher than the pro audio industry 24bit/96KHz standard. A special step compander circuit adds a further 20dB of dynamic range to utilise the maximum range of the converter at low levels.

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

Impulse Response – Electronic Integrity

Pulse response and the ability to capture and reproduce a sound's time component is the key to clarity, definition, spatial image and depth.

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 affected by the settling time and circuit design in analog electronics. The settling time of common electronics systems used in most commercial sound systems is around 10µs, 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.

Sampling frequency is the major determining factor of impulse response in the digital domain. In Figure B (below) it is evident that commonly used commercial systems, particularly digital, cannot pass the full resolution of the original signal. Impulse response time is affected in the digital domain by the sampling rate and in the analog signal path by the speed of the electronics (settling time) and control of the acoustic component motion (speaker movement). The change in the original signal caused through poor impulse response creates distortion. Systems with a long impulse response time are unable to transfer high dynamics and high definition signals. SLA systems incorporate hybrid signal processing at an industry leading sample rate of 20MHz and electronic settling time of 1 microsecond (1µs), to ensure audio reproduction with the highest possible resolution and definition.

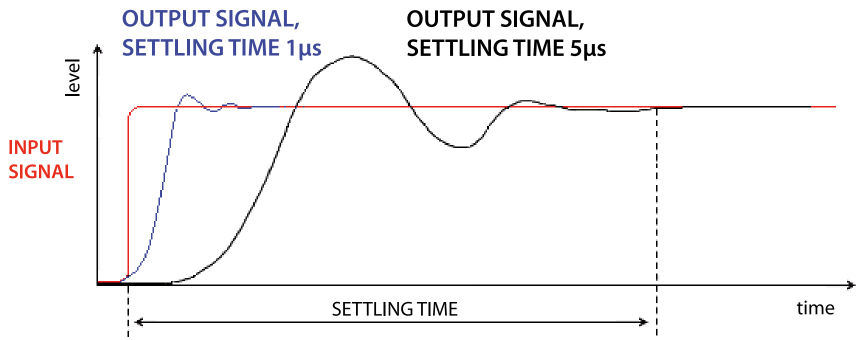


Figure B: Pulse response of an audio signal after transfer through various analog and digital systems.

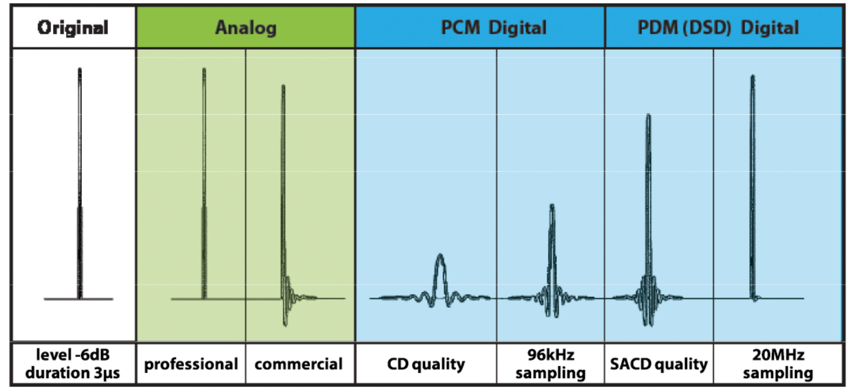


Figure B

Transducer Design

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

Every loudspeaker used in a KV2 Audio system is specifically designed. This leads to the development of components that become the ultimate solution for their given application, not just an off the shelf driver.
One of the most challenging projects undertaken by the team was the development of our new NVPD range of compression drivers. The idea came during an Italian lunch, where we discussed a new nitrate coating used in Formula One racing, offering extreme strength and rigidity. Extremely light, it is great for cars but had never been tried in pro audio.

By treating the diaphragm with a Nitrate Vapour Particle Deposition (NVPD) process, the dome's resonance and dampening characteristics are dramatically improved, lowering distortion even further and extending frequency response. By adding some of the largest Neodymium motors available today and our advanced phase plug design the result was a range of world beating high frequency units that produce distortion of less than 0.03% and measure flat up to 22kHz.

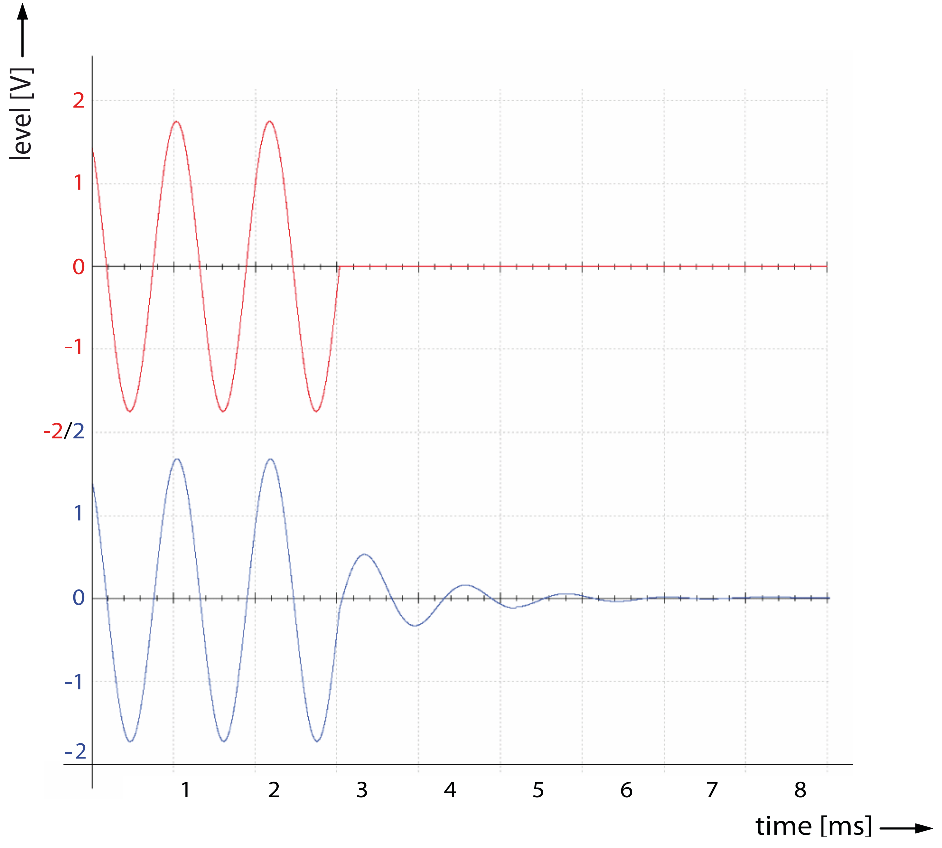


Figure C

Active Impedance Control

SLA systems feature exceptional feedback rejection and this in part is due to their excellent pulse response. 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. Odd harmonic distortion is far more audible than even harmonic distortion. Figure D shows the effects of AIC.

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 is wound around the pole piece to be very close to the primary voice 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 AIC device can be seen as an "active" shorted ring in the gap. The two AIC terminals allow driving the additional coil in many different ways according to specific application needs.

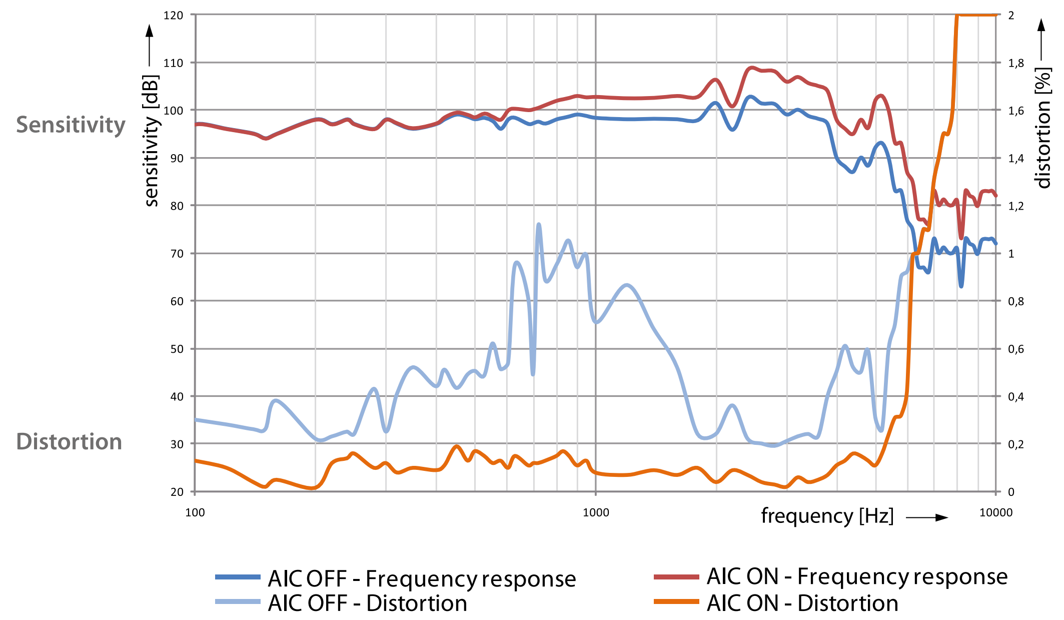
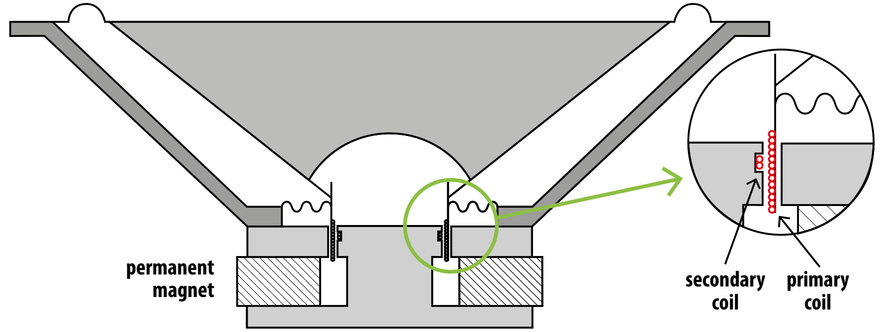


Figure D

Many Audio Manufacturers globally have tried to utilise KV2's trans-coil components with little success. They fail to realise that it is a combination of both transducer and electronics design that produce KV2's sound quality.



Transcoil driver

Amplification

KV2 Audio design amplifiers from the ground up for specific applications. This approach allows us to employ and refine the perfect types of power required for accurately reproducing highs, mids and bass frequencies. Low frequency devices have a unique set of requirements. Woofers are large, heavy and difficult to keep under control. On one hand you need lots of power, but besides cone size and weight, the single most important trait is the woofer's phase shift characteristics.

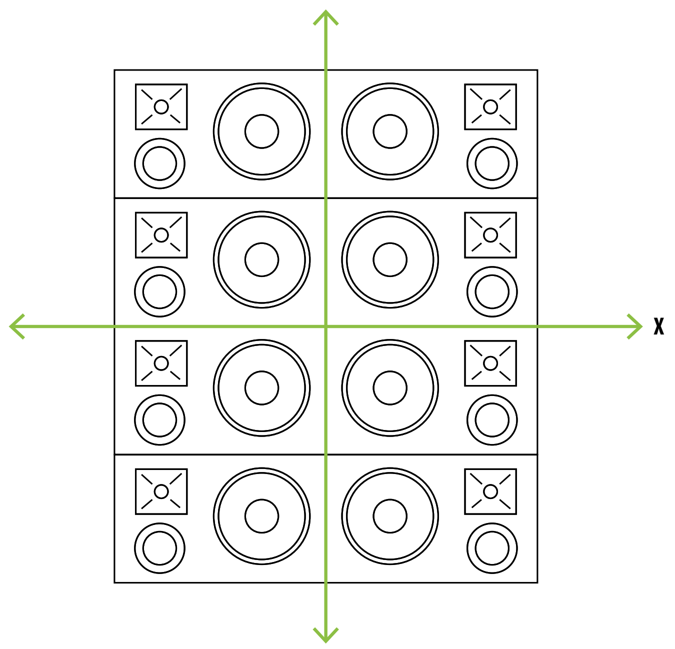
Simply put, phase shift is when current does not follow voltage as power flows through a voice coil. If you are sending 1,000 watts (100 volts and 10 amps coming out of the amplifier) under phase shift conditions, you may be required to produce double the amps at half the voltage in order to keep the woofer under control. A standard amplifier cannot accommodate this so we developed a new amplifier topology focused on developing high current but achieving over 90% efficiency to minimize cooling requirements and increase reliability.

The design features a switching voltage power supply that keeps the voltage across the output devices low, but capable of providing much higher current and better damping characteristics than standard Class H designs. For sound quality reasons in mid range and high frequency reproduction we use amplifier topologies based on Class A or Class AB. The warmth and clarity provided by this type of amplifier is ideal. Our design uses Mosfet output devices in a push-pull, transformer balanced amplifier featuring a fast recovery time. The amplifier's output transformer provides a vital technique for controlling the output signal of the amplifier under clipping by reducing the intermodulation distortion.

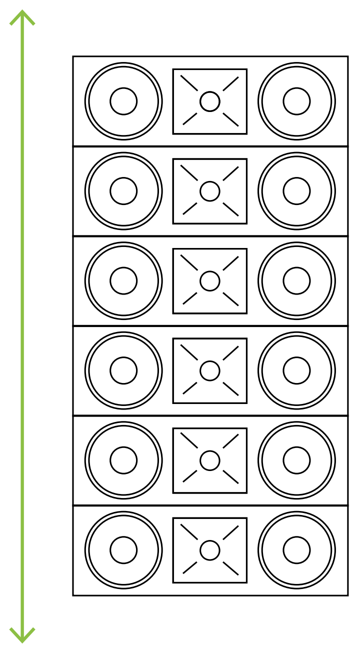
Sound System Designs - Multi Point Vs Point Source

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. 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.

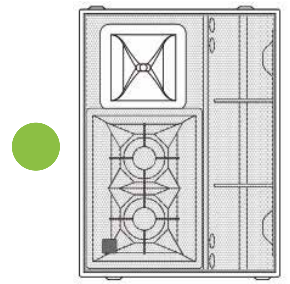
Multipoint two axes system



Multipoint one axes system



One point system

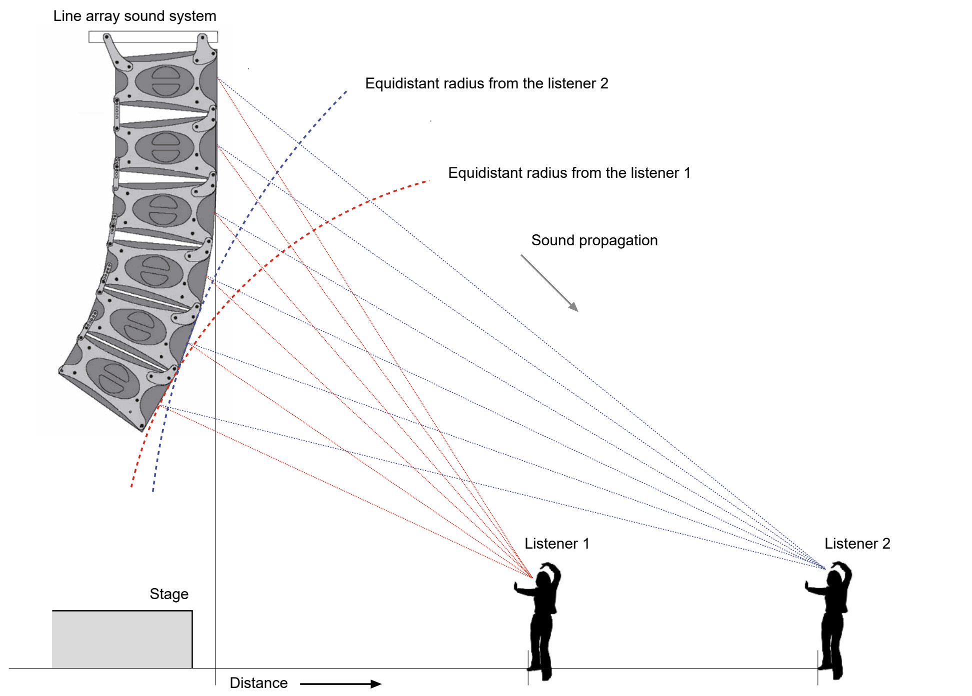


The first types of multipoint sources were simply a large pile of cabinets, stacked together like building blocks and intended to array on all axis. A major improvement in the next generation of systems was the introduction of multipoint, one-axis systems that provided better frequency response and increased 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. Line array does reduce the effect of multipoint sources interfering with each other like the systems of twenty-five 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 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.

A line array's natural frequency response before processing shows a continual roll off of high frequencies from 2 kHz upwards due to cancellation caused by the proximity of the numerous high frequency drivers. This requires large amounts of equalisation to be added to the top end to correct this phenomena. This huge boost in gain on the highs, lowers the system's overall headroom, on average a line array requires ten times the power to drive the top end compared to a single point source cabinet. Hence high power is not necessarily a requirement for large-scale coverage but quite often a result of a system's inefficiencies.

Time-Shifts, Properties of Multi Point source



Ver. Polarplots, 6 dB/div

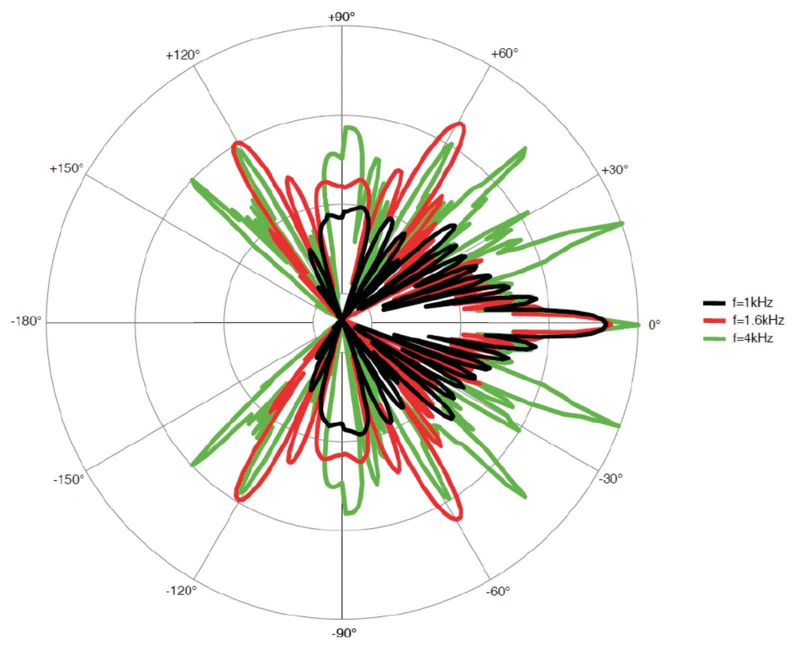
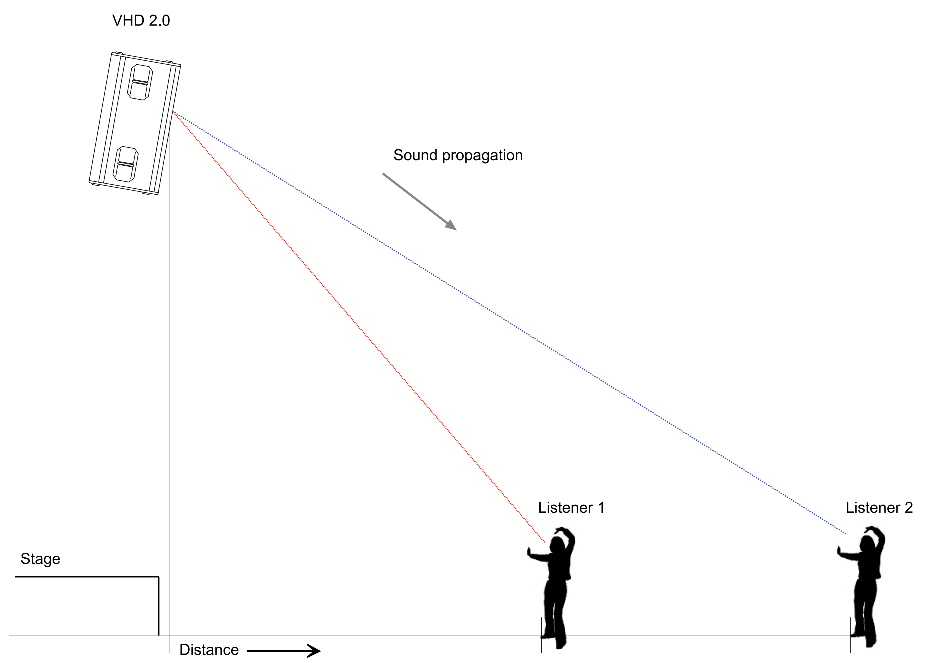
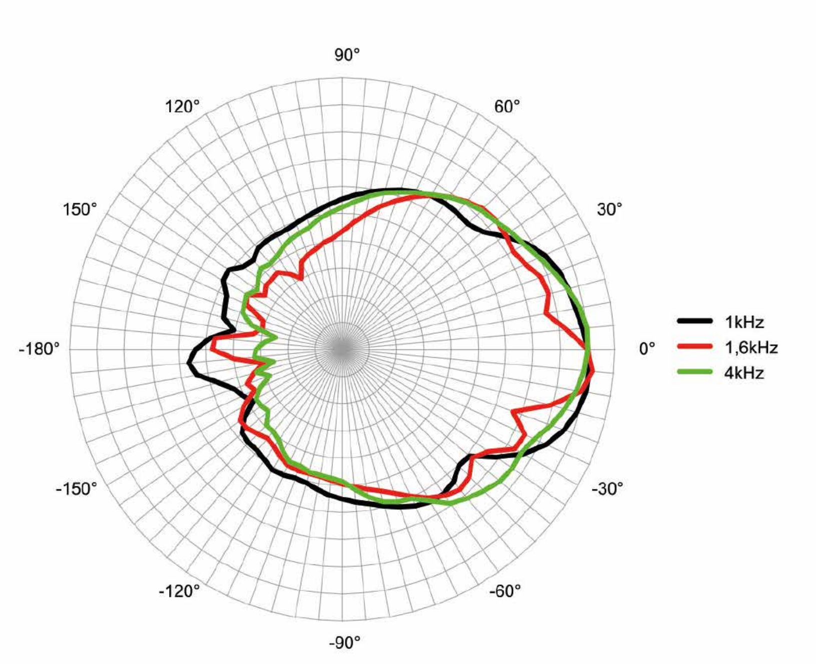


Illustration of the differences in distances to the listener from several Line array sources, each listener gets a blurred sound

Time-Shifts, Properties of Single Point source



Ver. Polarplots, 6 dB/div



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

Further to this when using multiple speaker cabinets in a line array the listener receives multiples of the original sound at slightly different times, smearing the time based information contained within. To maintain a high-resolution audio signal, it is vital that the system is able to exhibit a short impulse response time. The impulse response from a line array is damaged due to time shifts in the sound arriving to the listener. The diagrams show that the pulse response of a line array 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 these time shifts can be corrected using digital delays, however this does not provide a solution because time shifts will infinitely vary with each new listener position. Another myth relating to line arrays is the idea that all of the elements couple, to produce a controlled, directed, long throw soundfield.

As we can see by the polar patterns this is far from the case. The top pattern shows the smooth dispersion of a point source system compared to the erratic dispersion of a line array. As we can see from this polar pattern what actually occurs with a line array is a range of peaks and troughs, caused by destructive and constructive interference between the elements. Even more critically, one factor overlooked by system engineers or line array prediction software, is the random movement of the air in the listening area.

This causes huge changes in the transmission properties of multipoint systems. It occurs when an audience arrives, after the system engineer has spent the whole day aligning the system to an empty but theoretically perfect environment – an environment that in a real concert situation will never exist.

Watch our product videos on YouTube.

www.youtube.com/c/KV2Audiocom