KV2 Audio - Where Art Meets Science
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 strove to control every aspect of sound reproduction but all the while we move us further and further from the original source, its timbre and dynamics. Capturing the emotion and ambiance 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 performance on published specifications; we gauge it by the smiles on people’s faces.

As we move through the digital evolution the limitations of our imagination 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 achieving real value in quality sound reinforcement.

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

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 Tube Tech. There George designed many of their guitar and keyboard amplifiers and a complete range of processed speaker systems. After leaving Tube Tech, George started his first company, which sold high quality solid-state guitar amplifiers.

In their first year of production they sold out all inventory in Canada, testimony to the quality and value of the product. After a change in ownership of that company, George decided to head back to Europe to find a job 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.

KV2 Audio Research Team
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.George got it right, KV2’s employees 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 users 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 world’s finest products. 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.

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 groundbreaking, 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 nineteenth birthday in 2021. 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 precious 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 as 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.

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. Rather than developing 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.

A Super Live Audio System delivers:

- 20MHz Sampling - Extreme Resolution
- Greater Than 120dB Dynamic Range
- Very Low Non-Harmonic Distortion
- True Point Source
- Active Impedance Control - Zero Inductance
- Ultra Low Distortion – True Piston Motion Drivers

A Standard Above All Others

At KV2, we live and breathe sound quality. We believe in delivering the best possible audio experience for our customers.

Our systems are designed with the listener in mind, offering unparalleled audio quality and performance.

Lower Transport Costs.

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

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

High Quality European Built – Made with passion and pride in the Czech Republic.
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 must be by the systems performance will provide natural, acceptable sound to the audience. As shown in Figure A, 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 necessary to ensure total reproduction of the soundwaves arriving at microsecond intervals to the microphone.

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 is how audio signals travel through a space over distance at the required level to provide uncolored sound to the audience. 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 converters and transducers, 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 to the engineers mix of those sources. It is therefore 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 lost.

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.
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 is called Pulse Density Modulation (PDM). The resulting digital bit stream is encoded at an enormous 2,822,400 samples per second! (2.8224MHz)

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 as analog electronics. The settling time of common electronic systems used in most commercial sound systems is around 10μs, ten times longer than it should be. This distortion, caused by slow settling times, is not commonly discussed by many manufacturers as they fail to understand its significance, often misleading in providing the technical specification of products. Manufacturers who miss this distortion is a very simple mistake for high frequency, especially in digital technologies where it can extend about an eighth, “fizzy” high end.

Dynamic Range and Resolution in The Digital Domain

Impulse Response – Electronic Integrity

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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 spread 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. A complete system incorporates typical 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.
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 inability to control the diaphragm motions. Resonances reduce overall fidelity by introducing unwanted harmonics into the reproduction, and are 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 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’s 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 damping characteristics are dramatically improved, lowering distortion even further and extending frequency response. By adding one of the largest Neodymium motors available today and a unique phase plug design, the result was a range of world-leading high frequency units that produce distortion of less than 0.03% and measure flat up to 22kHz.

SLA systems feature exceptional feedback rejection and this in part is due to their excellent pulse response. Additionally, control over the speaker masses are 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 transducer magnetic circuit gap. This coil is almost as long as the gap height and is wound close to the pole to prevent it from affecting 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. The result is complete cancellation of the voice coil inductance and near-zero flux modulation and inductance modulation. The AIC device can serve as an “active” shorting ring in the gap. The two AIC terminals allow driving the additional coil in many different ways according to specific application needs.
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 source. If you are sending 1,000 watts (100 volts and 10 amps) under phase shift conditions, you may be required to produce double the amps or 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 valid technique for controlling the output signal of the amplifier under clipping by reducing the intermodulation distortion. Amplification

There are two main types of sound system designs that have been prominent in the market, consisting of single point source or multipoint source concepts. Multipoint sources arise 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 resulting disadvantages of multipoint source systems were the suppression of the high frequency output and the physical time shifted outputs from the individual speakers. Adding a number of time shifted outputs from individual speakers regardless causes a poor system impulse response. Sound System Designs - Multi Point Vs Point Source

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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.

When using one point sound source, the listener in any location gets only a pure (not blurred) sound. However, 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, targeted time delay. Time delays from a line array are detrimental 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.

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 is that any sound prediction software, is the random movement of the air in the listening area.

This causes huge changes in the transmission properties of multipoint systems. If a sound wave is emitted in a room with no audience, the air molecules will disperse the wave equally. However, when an audience is present, the sound waves are reflected and redirected by the audience, causing a change in the sound. 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.
Technical Talks
with George Krampera

Watch our product videos on YouTube.
www.youtube.com/c/KV2Audiocom