DESIGNERNOTEBOOK

True Dynamic Range

KV2 details a novel loudspeaker approach

By Marcelo Vercelli

When KV2 Audio opened its doors in November 2002, we were confident that our fledgling enterprise had the engineering/design experience to develop unique loudspeaker concepts.

Wait! Before you turn the page at the threat of yet another "Marketing 101" presentation from yet another manufacturer, let's get specific. After all, most anything can be looked at as "unique" – the definition of the term is often subject to the views of the presenter (in this case, that would be me), rather than the interests of the audience (that would be you).

This Designer Notebook was submitted by KV2. Live Sound makes every effort to eliminate any use of marketing inspired hyperbole.

The KV2 Audio ES1.0 flagship model.

Fortunately, our lead designer, George Krampera, came to the rescue by identifying something quite real and concrete, something that was always in front of us but had remained largely unnoticed. "If we want products with extreme clarity, definition and output, increase the audible dynamic range," he stated, and this would be the key to enhancing the "True Dynamic Range" (as George calls it) of a loudspeaker system.

Further, he defined True Dynamic Range (TDR) as the ratio between the maximum output of the loudspeaker versus the noise floor. Critical to the noise classification is inclusion of both harmonic distortion and intermodulation distortion.

The reasoning for this is relatively simple. As the audio signal becomes more complex, harmonic distortion and intermodulation distortion are summed together and also become more complex. This total distortion now looks like "a forest of trees," and acts like a random-noise floor. This reduces the dynamic range of the system accordingly.

Currently in our market, any loudspeaker that has been specified as having 1 percent Total Harmonic Distortion (THD) over a frequency range is generally considered "acceptable" within the pro audio industry. In terms of noise floor, 1 percent THD is equivalent to 40 dB of dynamic range (*Sound System Engineering*, Don and Carolyn Davis: 20 log10 [1 – 99%/100] = 40 dB).

Using 40 dB as a starting point seems pretty low as a dynamic range spec, but in studying many loudspeaker systems under normal live sound operation, we found that this total distortion reduces the dynamic range to an even-lower 30 to 26 dB (or less) when the amplifier enters clipping.

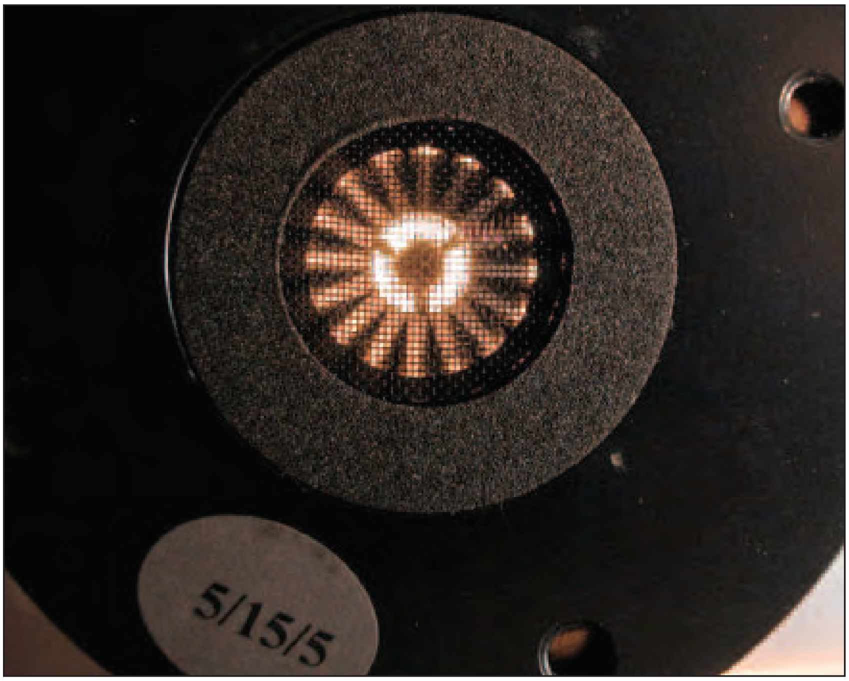
Further, what happens to the total noise floor in terms of composition and level is completely dependent on a complex series of parameters including the type of amplifier, the amplifier's protection circuitry, pre-amplifier signal processing, use of passive crossovers or equalization and the distortion characteristics of the loudspeaker components.

How an amplifier "clips" becomes a central focal point within the context of the loudspeaker because it is so prevalent in live sound. As an example, the dynamic peaks created by an acoustic guitar can easily clip a high-power amplifier. The average power required to reproduce the guitar may be only 10 watts, but with 20 dB peaks, you will clip a 1-kilowatt amplifier.

Clipping happens when the peak level of the output signal of an amplifier exceeds the output voltage capability of the amplifier. When this occurs, the output signal differs from the input signal with loss of peak information and creation of harmonic and intermodulation distortion.

Because of the very high dynamics and transients found in live sound applications, clipping is a common norm both inside amplifiers (large signal) and consoles/mixers (small signal). Our first question: "Is it audible?" The answer is a definite yes. If music is being played at 120 dB and the total noise floor is at 80 dB, it's audible. Push the system to 130 dB and witness the total noise floor rising to 110 dB – it's really audible.

When the total noise floor appears, the most audible impact is in the mid-high frequency range due to its harmonic content. The human ear is extremely sensitive around 4 kHz, which during clipping is clearly irritated by this distortion.



The high-frequency phase plug moderating between driver and horn.

AUDIBLE DIFFERENCES

How then to go about designing loudspeakers with increased TDR? Logically, we started by focusing on the loudspeaker under clipping conditions, for it is here that the largest amount of audible difference between loudspeakers can be determined.

Further, every aspect of a loudspeaker must be looked at: transducer design, acoustic design, horns, on-board power amplifiers, control electronics and integration schemes. This drove the development of the KV2 flagship ES Series of powered loudspeakers.

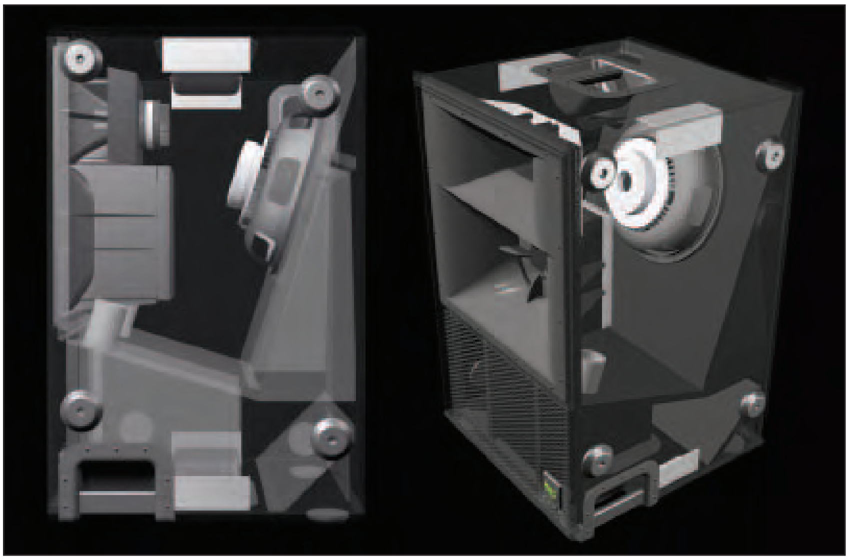
Developing transducer components requires scientific process and experience, and it is difficult to make quantum improvements in a technology that is mature. However, we've been fortunate to have the input and assistance of our colleagues at 18 Sound, a respected transducer developer/builder in Cavriago, Italy.

While in general we agreed to focus on new concepts of output, power handling and control, the fine-point focus emerged as the desire to reduce third-harmonic distortion. Our goal was based on the audibility limits of odd and even harmonics – generally speaking, humans can detect 1 percent to 2 percent of a signal's even harmonics, but we can detect 0.1 percent of its odd harmonics, particularly in the 4 kHz region.

While the exact "how" of this effort must remain undisclosed for proprietary concerns, the effort did result in mid-range and high-frequency transducer designs – compression driver and midrange cone – with third-harmonic distortion that we measured at the 0.1 percent mark. This was measured using standard AES guidelines of 10 percent rated power within the operating band-pass. (Some mid-range designs lower THD down to .03 percent when we implement a dual voice coil/magnetic structure design we refer to as Trans-Coil, which is currently used in our EX10 two-way product.)

An interesting focal point is the compression driver and midrange cone developed for the ES 1.0 mid-high three-way loudspeaker. The 1.75-inch-exit driver uses Nomex material for construction of the voice coil former that we found enhances the transfer of energy from the voice coil to the dome. Ventilation of the suspension system minimizes ring modes, and a neodymium-based magnetic circuit provides high force and control of the moving mass.

The key development, however, was the driver's phase plug. The design, which is patent pending, is


Using Superman's x-ray vision, we see the internal loading of the ES1.0.

based on a theory that combines both radial and concentric phase plug designs for the purpose of "averaging" the different amounts of output provided by the driver's dome. It also lowers the compression ratio of the driver, which lowers distortion and can smooth response.

Meanwhile, the 6-inch midrange cone, for reproducing frequencies from 500 Hz to 2.5 kHz, offers a new 44-millimeter neodymium motor structure focusing on control and lower distortion while maintaining output level. Third-harmonic distortion was lowered to a measured 0.15 percent using a standard non-Trans Coil magnetic motor structure. We also felt a real achievement came when this cone speaker yielded a sensitivity of 106 dB loaded on its wide-dispersion, 90-degree by 40-degree rotatable horn.

The horn design is based on constant-directivity geometry, with the midrange cone fixed to a large aluminum heat sink attached to a midrange "chamber." Further loading and dispersion is controlled through a 2.28-inch phase plug.

What about the low-frequency section of the ES1.0? Thanks for asking, and we'll get to it in a minute, because this section's performance ties directly into the on-board power amplifier design.

Unmasked, the "hybrid" power amplifier developed for the ES 1.0.

FAST RECOVERY TIME

Our preferred mid-high amplifier direction is based on classic Class-A or Class-AB topologies for reasons of midrange and high-frequency reproduction by this type of topology.

Addressing the design goal of lower distortion, we committed to fast recovery time, related to the voltage amplifier's bandwidth and speed capability. Extra care is also placed on designing circuits that show improved distortion characteristics under clipping conditions.

Specifically, the circuits employ metal oxide silicone field effect transistor (MOSFET) output devices in a push-pull, output transformer balanced, Class-AB amplifier. Push-pull was selected due to factors noticed during testing. We found no way to control the fundamental distortion created by the amplifier when it clips, but we did find that the output transformers could control the intermodulation distortion by-products.

As a result, when the amplifier clips, the noise floor is simplified, with only the harmonic distortion delivered to the loudspeaker.

Low-frequency transducers have a unique set of performance characteristics and requirements. Woofers are larger, heavier and difficult to keep under control. And they require lots of power. The important thing, however, is to understand the *type* of power needed.

Besides cone size and weight, a vital trait is the woofer's phase shift characteristics. Simply put, phase shift occurs when current does not follow voltage by 90 degrees as power flows through a voice coil.

For example, if you're sending 1000 watts (100 volts and 10 amps from the amplifier) under phase shift conditions, you may be required to produce double the amp's current output, at half the voltage, in order to keep the woofer under control.

It was this concept – an emphasis on low-frequency performance - that drove our topology development. First, there was focus on a high current design capable of controlling heavy, high-mass woofers while running "cold" to minimize cooling requirements and enhance reliability. To reduce the heat dissipated, the efficiency would need to be increased so that the voltage at the amplifier's output devices could be reduced.

What resulted is a distinctive amplifier class topology measured to be approximately 90 percent efficient within its operating bandwidth, with a practical operational limit of 1 kHz. (Above 1 kHz, losses decrease efficiency dramatically.) The amplifier uses an analog switch-mode voltage power supply that keeps the voltage across the output devices low and constant.

Further, we found that this amp provides much higher current and better damping than standard Class H designs. We also discovered that a surprising property of the switcher resides in how it interacts with the output stage and delivers higher current at lower voltage, thus adapting adroitly to woofer phase-shift conditions that can reach up to 45 degrees and exposing the very real limitations of standard designs by requiring an amplifier to deliver higher current at lower voltage.

For example, under phase-shift conditions, an amplifier producing 1000 watts (20 amps x 50 volts) could be asked to provide up to double the amps at half the voltage (40 amps by 25 volts) just to deliver the same power. This ability to adapt to the woofer's phase-shift conditions is primarily attributed to the amplifier's voltage power supply. It's a very high efficiency switch-mode supply constructed using low-inductance, high-voltage capacitors and huge coil inductors.

It's also unique in that it follows the input signal and keeps the voltage going across the output devices, low and constant, which significantly reduces loss. And, the amp is placed in series with the output capacitors and acts as a supplemental current source when voltage is not required and the switcher is off (phase shift condition).

MILDER RATIOS

We like to offer wide-dispersion loudspeakers because of a deep belief that most small-to-medium applications (up to 2,000 people, 120 feet or so of throw) can be satisfied using wide-dispersion, high-output devices employing a minimal amount of cabinets.

ES full-range loudspeakers, topping KV2 subs, doing their thing live.

It's generally believed that narrow-dispersion horns are superior due to their higher sensitivity than wide-dispersion horns. However, we disagree because distortion must also be taken into account. Shallow, wide-dispersion horns produce milder compression ratios. For mid-range designs, we avoid anything above a 4:1 ratio because it overloads the cone and inherently requires longer horn throats that can generate high distortion artifacts.

Finally, we've come to believe that the single most important characteristic to avoid is ring modes (or resonances). Every single element comprising a loudspeaker has resonance modes - the components, ports, horns and grilles. When something resonates, it either shows up as a peak in the response curve or as a hole by passively absorbing energy.

Our philosophy is to absolutely minimize the use of equalization filters that would normally be used to correct resonances. In fact, we never recommend use of more than a few dB of EQ anywhere in the system. When a frequency clips where an EQ filter is used, equalization is lost instantly because the level at that frequency is no longer lower or higher than that of other frequencies not equalized. All that can be heard is the fundamental response of the loudspeaker. If the fundamental response of the system isn't correct, any anomaly that's addressed with EQ becomes audibly apparent during clipping.

By going with an active rather than a passive electronics package, the system can be fine tuned and controlled to a much greater degree. Far less is left to the capabilities, or lack thereof, of non-professionals that may be operating the system. Crossovers, driver alignment, equalization, component and amplifier protection, and I/O flexibility are all optimized and easily manipulated.

Putting it all together, the TDR of our ES Series loudspeaker systems is 15 dB to 20 dB improved (frequency-dependent) when contrasted with comparable systems. ■

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