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Industry News & Developments

OEM DSP Active Line Level Crossovers from Danville Signal

By Vance Dickason

The advantages of active DSP crossovers have been apparent for a long time, so it should come as no surprise that today the majority (about 90%) of professional audio loudspeakers (studio monitors) are powered loudspeakers. However, there is a growing desire for powered monitors in consumer audio markets with companies such as Klipsch, Platin, and Enclave fielding WiSA-type wireless powered speakers.

The first DSP-powered loudspeaker systems that gained traction were mostly for large acoustic spaces (e.g., arenas and concert halls). DSP technology could be used to control coverage, frequency response, protection, and provide appropriate time delays, which in large venues can be significant. They didn't necessarily need to be small, embedded inside the speaker cabinet, or sound audiophile great to provide real value.

If we assume that most high-performance active loudspeakers need sufficiently large power amplifiers to meet the demands of the drivers and the enclosure, this greatly limits the options available. High-quality linear amplifiers are inefficient and generally produce way too much heat for an application that requires the amplifier to be mounted inside the speaker enclosure with the drivers (embedded), so the most practical option was to use Class D switching amplifiers.

Today's Class D amplifiers from the major suppliers have gotten significantly better. In fact, one of these manufacturers approached Danville Signal and asked if Danville would support their amplifiers because they felt their amplifiers did not have good enough DSP front ends from a sonic performance. That made Danville Signal smile, as that is where they live...high-performance DSP crossovers.

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		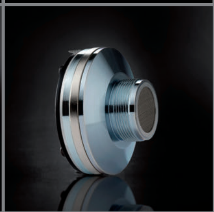	

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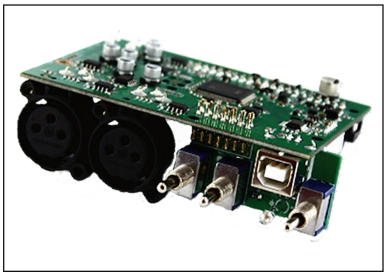

Photo 1: The Snowbird families of DSP Crossovers are priced competitively when compared against other DSP crossover solutions that rely on highly integrated, but lower performance parts.

powered loudspeaker applications (e.g., professional studio monitors and high-end consumer audio systems) and to be mounted inside the loudspeaker enclosure… embedded as it were.

Given that Danville Signal has a reputation for great sounding DSP crossovers, this obviously starts with incorporating great parts that make Danville's dspCrossover lineup. All of the Danville crossovers use an Analog Devices' floating-point SHARC DSP as the processor. SHARC DSPs are the gold standard for high-performance audio processing. The DAC and ADC functions are provided by AKM Velvet Sound converters operating at 192kHz. The Velvet Sound data converters are widely recognized as one of the best sounding DAC/ADCs in the audio industry.

Danville Signal offers three families of crossovers for powered speakers, GreyWolf, Bluebill, and Snowbird. This group of line level loudspeaker crossovers share many common features. Each crossover has one or two balanced audio inputs and up to four audio outputs. They also support a digital input option.

All the crossovers are dependent on a mating power amplifier for their power. Danville currently has support for amplifiers from Pascal, Hypex, ICEpower, and Powersoft. Each of these amplifiers has its own specific interfacing requirements. Danville has mating adapters that convert these requirements to the DSP board so that cabling is straightforward and control features (e.g., mute, standby, and fault protection) are addressed.

Each of the Danville Signal line level speaker crossovers has its own attributes:

Snowbird—The Snowbird families of DSP Crossovers (**Photo 1**) have high-performance expectations but are priced competitively when compared against other DSP crossover solutions that rely on highly integrated, but lower performance parts. The goal was to make the best sounding crossover among this class.

Greywolf—The Greywolf family (**Photo 2**) has higher performance goals than the Snowbird line. It offers state-of-the-art performance never before seen with COTS crossovers. The SHARC ADSP-21565 is the most powerful DSP crossover ever used in an embedded crossover design. A variety of DAC modules are available all operating at 192kHz with signal/noise > 123dB.

Bluebill—Not yet released, the Bluebill family has improved signal/noise (~123dB) over the Snowbird but uses the same SHARC DSP as the Snowbird. It uses SOA TI data converters for excellent sonic performance at a reasonable cost.

Greywolf, Bluebill, and Snowbird crossovers are footprint compatible as well. Manufacturers can incorporate any of the Danville dspCrossover line in the same loudspeaker without changing the housing design. This enables manufacturers to choose from very good to state-of-art, depending on their budget and performance requirements.

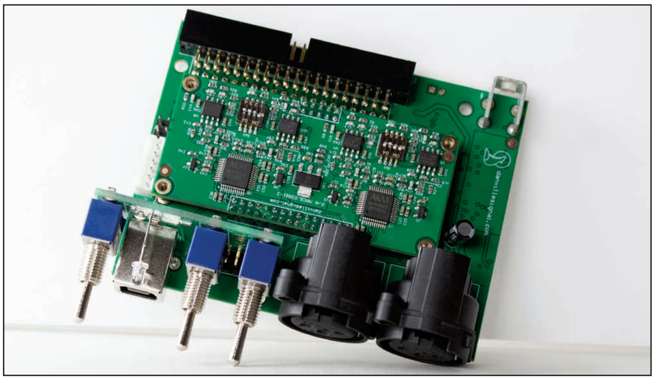

Photo 2: The Greywolf family offers state-of-the-art performance never before seen with COTS crossovers.

There are many different configurations. The basic feature list includes one or two analog audio input channels (balanced XLR) and two or four amplifier channel outputs, plus optional S/PDIF input and buffered S/PDIF retransmitted output. A USB Control Port is available for programming and field updates. The boards also feature toggle switches for user-defined controls.

For more information, visit www.Danvillesignal.com. ***VC***

Spotlight

DaS AuReality—3D Loudspeaker Technology

By Joe Hayes (VP Engineering, Vastigo, Ltd.)

DaS AuReality is a unique and fully patented technology that utilizes multiple speaker driver arrays, actively harnessing the non-resonant power available through prime numbers (**Photo 1**). Each array element, some in pairs, are driven by their own unique algorithm, based on the proprietary technology.

The purpose of DaS AuReality is twofold: First, is the introduction of the concept of "Diffusion at Source" (DaS), wherein each frequency has its own unique, time decorrelated polar (TDP), sound field (elliptical in polar shape). Conventional speakers have time coherent polar (TCP) sound field (circular in polar shape). Second, is to transform the sound field produced by time coherent, individual-driven piston radiators, into a dynamic, dual-state, constant power, acoustic sound field.

In this dualistic behavior, either the acoustic sound field is acting like a perfect point source (harmonically driven), or it acts like a rotating sound field, think Leslie Loudspeaker, whenever there is a sudden change in the signal (the step response). By combining these dual-properties, DaS AuReality acts as an acoustic redefining sound reinforcement/reproduction tool, capable of producing "true reality" like sound fields, in any listening room, at all scales of application.

DaS AuReality has evolved out of work known as "Acoustic3D" from an Australian-based company. DaS AuReality is now being commercialized by Vastigo, Ltd. (Vastigo.com), headquartered in Singapore. Vastigo is offering licensees for the DaS AuReality technology into its many market application segments, which include virtually all speaker applications. The technology is backed up by a comprehensive multi-generation and geographical market patent folio.

Photo 1: A DaS AuReality active digital tweeter

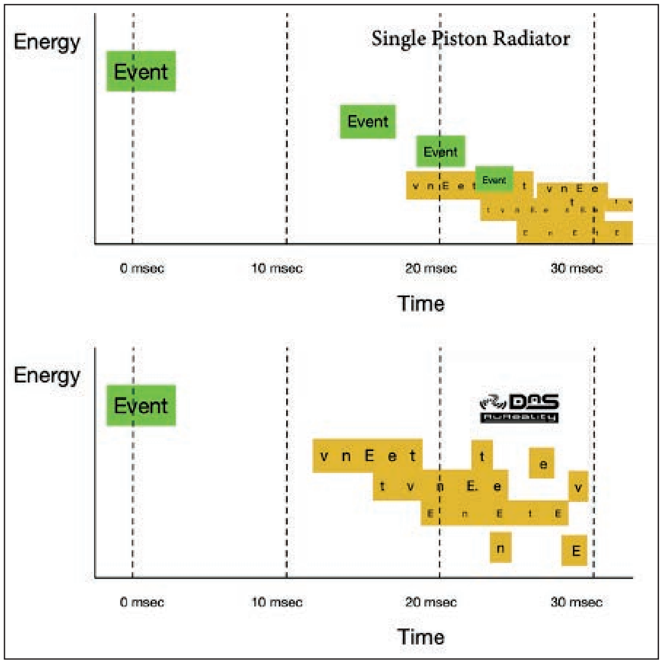

Figure 1: Non ideal (top) vs. ideal (bottom) early energy patterns of room acoustics.

Understanding the Relationship Between Diffusion and Time

To understand how DaS overcomes "listening room acoustic challenges," the reader will need to understand the relationship between "diffusion" and "time," when used in an acoustical context. A simple way to put it is: Diffusion is energy that has completely lost time information. Here, I will draw upon concepts used in auditorium acoustics and psychoacoustics. An "event" is a perfect descriptor for explaining the perception of a listening room acoustic. A new event is described in auditorium acoustics as a sudden change in the energy envelope by 10dB or more, indicating a sudden, but meaningful change in energy in the room.

This sudden change of energy will trigger the listener's perception system to localize the sound source and contextualize it in 3D space. In practice, this event could be someone saying a word, it could be a musical note, or a percussive strike on a drum! These events give a perfect opportunity for the human brain, via the mechanisms available in the anatomy of the inner ear, to hear the early reflection patterns of a listening room, from which it builds a perception of the sonic environment surrounding them.

In **Figure 1**, we see an auditorium acoustician's representative diagram of the early reflection behavior of the non-ideal listening room (Figure 1a), and then the more ideal, "optimized" listening room (Figure 1b). Let's use the word "Event" as the new acoustic sound source in this model. To the human brain, the word Event has meaning! It is a learned pattern in our psychology.

In the non-ideal listening room (Figure 1a), through a time coherent polar loudspeaker, the original word "Event" is reflected back to the listener at small time delays, and at ever reducing levels. The levels are reduced by the reflection process off the walls, floor, ceiling, and other surfaces where acoustical energy is absorbed from the wave front. The time delays are purely due to the difference in the pathways that the wave front travels, in finding its way back to the listener. The absorption of energy over time from the reflection process is quantized as reverb time!

In a non-ideal listening room, the word "Event" will be heard several times, subliminally, in the early reflection patterns. Assuming the walls are purely flat they will create specular reflections. This means the reflection energy is pretty much still the word "Event." It is just time delayed, and slightly reduced in energy, and perhaps missing a little bit of the top end.

Whenever the word "Event" can be clearly heard subliminally in the listening room acoustic, I will indicate it with a green background. In the top image, we can see five clearly repeated and audible "Event" of the listening room acoustic over the first 30ms. The first repeat arrives at around 15ms. All of this is made possible as the loudspeaker source is time coherent polar. Every direction has the same sound. The human ear and brain will process this clearly repeated pattern and allocates various perceived acoustic qualities, depending on the arrival time and energy of these early reflections. Almost always, the perceived quality will be a negative comparison to the ideal listening room.

What is well known in auditoria acoustics is the specification for the "holy grail" for listening room design: it is what is shown in the ideal room (Figure 1b). In this ideal listening room scenario, the only time you hear the word "Event" is directly from the source and never again. You want all the reflected energy to be diffuse, that is, to lose all of its timing information.

The DaS technology delivers this diffuse sound at the speaker source, making the room boundaries and contents irrelevant to the sound energy in the room. The diffuse sound hits the walls and other barriers and collapses into itself as decorrelated energy waves. The original sounds are heard crystal clear without compromise. In the context of the metaphor, instead of the reflected energy being the word "Event" again, can you somehow make the reflected word be "vnEet"—that is, move the order of the letters around (this is a metaphor as the timing is faster than the length of spoken vowels). Or better still, can you spread out the reflected word over time, so that it becomes twice as long in time (i.e., "v n E e t"? This is called a non-specular reflection. The reflection looks nothing, or very little, like the direct word "Event." I've indicated non-specular reflections in the diagram with amber backgrounds.

To take this concept even further, can you somehow make the other early reflections look nothing like either "Event" or the first reflection "v n E e t"? If you can do this, the human brain will not hear these subliminal repeats of the direct word, and will thus not allocate any negative listening room acoustic attributes to the sound quality.

Arguably, if you can do this well enough, the human brain may not even acknowledge that it's in a room without the presence of early specular reflections! The absence of subliminal repeats of these "events" deprive any opportunity for the human brain, via the mechanisms available in the anatomy of the inner ear, to assemble the early reflection patterns of a listening room to build a perception of the sonic environment surrounding them. Using DaS technology, we have just eliminated the "listening room effect" from the playback!

In the non-ideal listening room, specular reflections will eventually become non-specular reflections! This is just the nature of multiple generation reflected energy breaking down. In the non-ideal room (Figure 1a) eventually the amber background energy becomes dominant, but typically only after the acoustic energy in the room has significantly diminished. Think toward the tail end of a sound—this is generally too late to be useful!

In the ideal listening room scenario (Figure 1b), we have amazingly managed to get every reflection to be non-specular. That is, not a subliminal repeat of the direct or any other reflected sound! Without the DaS technology the only way to achieve the ideal listening room is extreme room treatment as shown in **Photo 2** [1]. The horizontal plane, defined by the listener's ears in the ideal mixing position, has been broken up using physical acoustical diffusors (white stalactite looking structures) on practically every horizontal surface in this the room. This design sets out to achieve the ideal room acoustic as defined by Figure 1b. It still misses a lot of surfaces including the complete floor and portions of the walls and ceiling.

The same listening result sought in Photo 2 can be achieved very simply through DaS, but far more efficiently, without any physical changes whatsoever to the listening room. I'll explain how! If we can get the sound field from the loudspeaker to leave, at every frequency, at every polar angle, at a different phase angle, we will effectively achieve exactly the same listening experience result as has been achieved in Photo 2. But, at considerably less effort and cost, and without any room surface treatment!

The DaS acoustical algorithm used in the device results in a loudspeaker with a constant power in all directions in the polar plane. The number sequences used have the

Photo 2: Myron E—Diffuser in a sound studio control room

unique property that the Fourier coefficients are equal in all directions. The Fourier transformation does not specify the exact location in time of the signal within the Fourier Window. The key difference to conventional loudspeakers is that the signal "timing/phase" is "variable" in the polar plane. Each frequency has its own unique "flapping" motion releasing variable polar delayed acoustic energy into the listening room, at different polar angles. The result is a completely de-correlated sound field in every polar direction (i.e., no two directions are copies of each other).

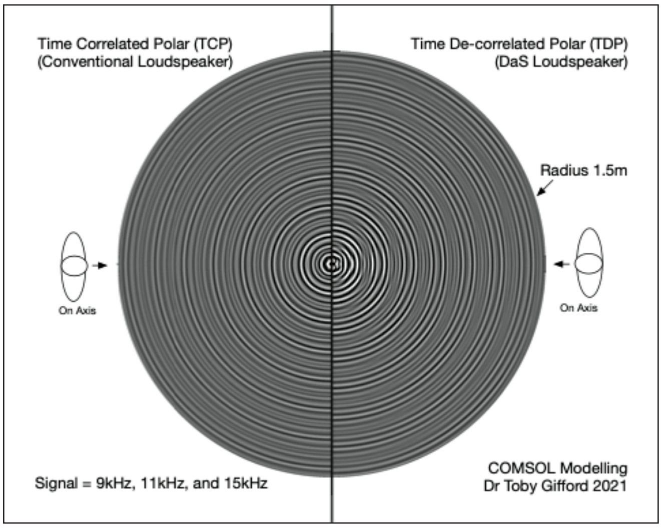

Figure 2: "Correlated" vs. "de-correlated" polar sound fields

Harmonically Driven Behavior (DaS)

DaS uses constant power with TDP sound fields. "Correlation" is the same as "specular" reflection wherein both the direct and reflected energy are substantially the same in shape. "Decorrelated" is the same as "non-specular" reflections wherein the reflected energy has nothing, or very little to do with, the direct energy.

We have used COMSOL to build simulations of DaS sound fields. **Figure 2** shows a conventional TCP loudspeaker on the left and a DaS TDP loudspeaker on the right. For the sake of simplicity, we used a harmonically driven source containing three frequencies of equal amplitude—9kHz, 11kHz, and 15kHz. As Dr. Toby Gifford very eloquently puts it, "looks like one half of the vinyl record got left in the sun!"

When a harmonically driven (constant signal or steady state response) audio signal occurs, acoustic energy is released at different "time/phases" in the polar plane. Each frequency has its own unique spatial shape.

As expected, in the left-hand image (TCP) of Figure 2, the sound field is exactly the same in each polar direction. In the right-hand image (DaS/TDP) of Figure 2, you can see both the bent elliptical nature of the three different frequencies, and most important, how each polar direction is completely different (decorrelated) in the sound field. It is physically impossible, even in a perfectly flat glass-walled listening room, for the reflected room energy from a DaS loudspeaker to be the same as the direct sound one would hear standing at one particular polar angle. Yes, DaS can turn a glass-clad room into an ideal acoustic environment, through the loudspeaker itself.

One huge benefit of the bending of time (TDP) is that echo-locating the source of the sound (the loudspeaker) becomes near impossible. “Out of box” sound becomes dominant, wherein the sound appears to have completely detached itself from the loudspeaker source.

Step Response Behavior (AuReality)

The ability to “create an ideal listening room environment” through DaS is, in itself, very exciting. But there is also the transformational ability of the technology, to make the step response driven sound field in the listening room to rotate, thus creating dominant intra-aural intensity differences (IID) cues for the listener. This causes DaS AuReality loudspeakers to have a presence that takes on an “amazing air” of reality!

A long time ago in my career, I worked on Guitar MIDI controllers. This work informed me about the time/frequency nature of musical information. I call this the “ergonomics” in room acoustics. You can think of this as the interaction between humans and acoustic spaces. These ergonomic observations—such as the 5ms threshold for a perceivable delay between a musician’s action and hearing a response—have influenced the design of DaS AuReality.

The original Yamaha DX7 synthesizer, allegedly the first equipped with MIDI, introduced to the design at the last minute before release (1983), suffered from an alleged internal MIDI latency around 10ms. This made it hard for virtuoso musicians to push the instrument via MIDI in certain performance situations.

The most applicable knowledge I learned from doing that guitar synth work, was that of the ergonomics of the musical performance itself. Things such as the number of cycles of a frequency that you would find, for example, in a short bass note. A very staccato bass note may have only 5 cycles of the fundamental between turning on and turning off. The fastest guitar player we could find back then could not even play 20 notes per second. Twenty notes per second also happens to be the equivalent to a 50ms interval between notes, the threshold in auditorium acoustics at which an early arrival time for the first correlated reflection starts to be heard as individual echoes (or individual notes)!

A short bass note, with a duration of 50ms, still has plenty of opportunity to set up a series of correlated specular reflections in a listening room. Fifty milliseconds is 18 meters when travelling at the speed of sound. That is quite a few laps of a small room. This series of correlated specular reflections is true for both the recording environment, caught in the audio, and in the reproduction environment, exhibited during playback. This also raises the additional question, is there a way to make these recorded reflection events in the audio become dominant in the listening room acoustic? The answer is yes, through AuReality.

Part of the mechanism for setting up the DaS decorrelation sound field is the use of a prime number of drivers. This would normally cause issues with the transient nature of sounds, the “attack”! Our engineers have developed a way to make the acoustical response to sudden changes in the audio signal (step response) to act like real acoustic reflections, in the now diffused listening room. It does this by harnessing the behavior of wave packet transients.

When a sudden energy is applied to the driver array, the energy between the individual elements of the array creates a wave packet transient output. This wave packet transient has very unique properties that are useful to psychoacoustic perception. The acoustical output is symmetrical about the y-axis, thus it is acausal (and therefore linear phase)! The shape of the wave packet does not change over distance (it is not dispersive). Consequently, DaS loudspeakers do not suffer from tonal distortional over distance. The sound simply gets softer, until you can no longer hear it, but remains a totally intelligible unit at that point.

This wave packet transient creates very strong interaural intensity differences (IID), all be it for a very short period of time (a sonic atom: Denis Gabor [2]). To the listener, suddenly the spatial cues caught in the audio recording start to stand out as a strong acoustical phenomena in the diffused listening room. We, the listeners, are picking up the benefits of DaS (decorrelation). We are also clearly picking up the audio spatial cues, through listener IDDs, in the recording’s “early reflection patterns—space.”

This acoustical stack of benefits, DaS AuReality (wave packet rotational sound fields), and each harmonically driven frequency having a slightly different spatial shape/presence, gives the sound field an uncanny resemblance to those of real life sound fields.

Application in the Audio Spectrum

To understand market segment applications, it is worth discussing how each technology embodiment might be realized. Smartphones are an exciting market segment. The application of Das AuReality may be realized using an array of the USound micro piezo drivers. Or, to that matter, any MEMS or microspeaker drivers [3].

Hi-fi tweeters are healthy, low-hanging fruit for immediate DaS AuReality product realization! Through the advent of the micro speaker drivers for laptop and pill speaker use, there are now a proliferation of small, high quality, low cost, dynamic loudspeaker drivers available.

Table 1 shows how the diffusion bandwidth ratio and the required amp channels change with DaS array size. The practical challenge is to maximize the DaS effect without

DaS Array Size (Drivers in array/ Prime numbers only)	Diffusion Bandwidth Ratio	Amp Channels Required
1	0	1 [i]
3	5	2
5	9	3
7	13	4
11	21	6
13	25	7
17	33	9
19	37	10
23	45	12
29	57	15

Table 1: DaS Array size vs. benefits

[i] Equivalent to a conventional speaker with one driver

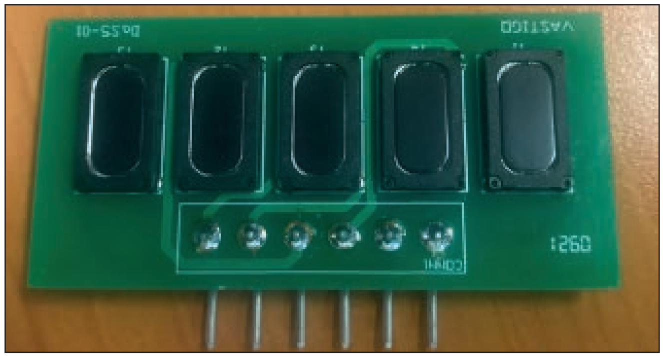

Photo 3: The DaS5 tweeter 3.5W RMS array (0.5W RMS drivers)

creating too much complexity and cost in the solution. The choice of seven drivers presents as a very good, practical option. It has a bandwidth ratio of 13. With seven drivers, we can build a DaS AuReality solution with a diffusion bandwidth from 1kHz to 13kHz. This seven-driver solution requires four channels of amplification.

The digital DaS AuReality tweeter shown in Photo 1 uses an array of seven off, 34mm x 11mm drivers. The four channels of amplification are currently achieved through using 2× Texas Instrument TAS5825M chips. These chips are sufficient for holding our proprietary DSP, as well as basic crossover and EQ settings. While each TAS5825M is capable of producing 2×50W (4Ω) we effectively have 8Ω drivers (2× 4Ω 3W RMS in series), and only need 12W RMS per channels to drive 6W RMS of driver capacity. The tweeter in Photo 1 is the equivalent of a 21W RMS device.

In practice, the tweeter shown in Photo 1 works from 1,500Hz, and is capable of more than "holding its own" in a high-end bookshelf speaker scenario. It is suitable in multiple applications, from soundbars to Hi-Fi equipment, and studio monitors.

We also see an application where multiple rows (columns) of these drivers are aggregated to very high-power levels, for concert sound (PA) applications.

Photo 3 shows an array of 16mm × 9mm 0.5W RMS drivers, suitable for pill and or small speakers. This requires three channels at 2W RMS amplification, with the array being five driver elements.

We have thus far concentrated mainly on applying DaS to the higher frequencies where specular reflections have the most deleterious consequences on sound perception. We have also applied DaS to a low frequency band, 90Hz to 450Hz (Array size = 3 in Table 1) and achieved the same "realistic" sound and presence. In this case, the drivers need to be no wider that the half wavelength of 450Hz, 38cm.

To complete the scenario of applications, let's look in more detail at concert PA sound (**Figure 3**). The Texas Instrument TAS5825M chips can be used in bridged mode to achieve 100W RMS into 4Ω. Using 25W RMS drivers per element (perhaps a column of eight high 34mm × 11mm 3W RMS drivers), one could build a concert PA array with 29 driver elements per row, requiring 15 channels of amplification (8× TAS5825M chips), totaling 1,500W RMS. The solution would be 29 drivers wide, and 8 rows high, and would fit inside a 400mm2 enclosure. As it is PCB-based, this concert PA driver is not too complicated, not too expensive, and uses contemporary available complete-off-the-shelf (COTS) componentry. It is like building with Lego blocks!

Additional beneficiaries of the DaS AuReality technology, may well be the touring musicians who, instead of regularly performing in an acoustically challenging venue, can now completely remove that "venue acoustics influence" from the performance sound. Likewise, wouldn't we all want to start hearing intelligible announcements at airports and train stations! In addition, for the hearing impaired, due to the removal of specular reflections, we have found that using DaS enables them to listen to the TV, even at moderate levels, despite hearing loss or needing to wear hearing aids.

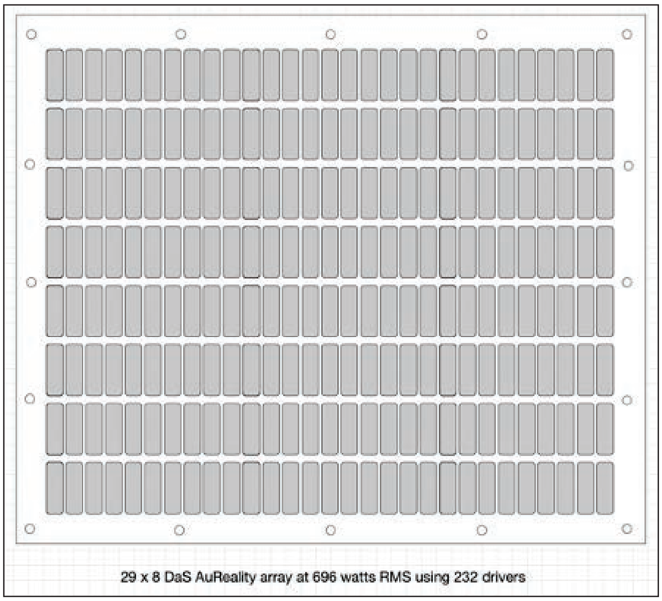

Figure 3: A prospective 232 driver concert PA DaS AuReality solution

Conclusion

The ability to easily and effectively eliminate the listening room acoustical specular reflections, creates an "uncannily real listening environment," where the performer's every nuance becomes audible. We believe this is truly "Sound beyond Space and Time." We also predict that for serious music lovers and audiophiles, once they have heard a "DaS AuReality enabled" loudspeaker, it will become their "must-have device" in their sound system! For information regarding technology licensing or partnering enquires, contact Olavs Ritenis, VP Global Business Development, Vastigo, Ltd. at +65 9339 4536 (Singapore) or email business@vastigo.com. ***VC***

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Focus

An Introduction to Smart Speakers for Speaker Engineers

By Mike Klasco (Menlo Scientific, Ltd.)

The cliché everything old is new again are lyrics from *All That Jazz*, a musical from the mid-1970s. Even further back, when I was a kid in the 1950s, we had table radios that often included a clock. Clock radios were truly a classic early example of a consumer electronics multi-tool pocket knife.

Jump forward a half century and the digital personal assistant, MP3 player, and flip-phone merged into the smartphone. Today, if you ask nicely, the smart speaker will tell you the time, and play your requested music. Perhaps even inadvertently order a pizza for you when you tell your voice command TV to change the channel or pass along your conversations back to "big brother's" cloud data bank.

Good, bad, or ugly, product designers must go with the flow and learn new tricks. Meanwhile the guys with the grey hair on your engineering team might wistfully remember when a speaker designer only had to worry about the woofer, put a cap in series with the tweeter, and stick it all in a fake walnut vinyl-wrap particle board box. Of course, there always were the pricey and more critical higher-end audio components, but most products on the market consisted of a receiver, a turntable, and a cassette deck along with the passive speakers. Each product design group could focus on their own turf.

Yet even the transition from designing soundbars to smart speakers has challenges. From the hardware side, the voice assistant smart speaker must be compact, yet be packed with a power supply, all sorts of signal processing and memory, an amplifier, an Internet interface, mic(s) and a loudspeaker—all stuffed in an enclosure the size of a thermos. Smart speaker technology is all over the place (and therefore so is this article!).

The software/firmware side that performs the tasks or services for an individual, based on commands or questions is provided in the form of the Intelligent Virtual Assistant (IVA). Amazon Alexa, Alphabet Google Assistant, Apple Siri, and Microsoft's Cortana. For Asian markets, there is DuerOS, Ali Genie, and Dingdong used by Chinese players such as Baidu, Alibaba, Anker, and Xiaomi with local language support in the main software. Regardless of the smart eco-system, voice-enabled smart speakers are the primary interface to the smart home. With the wake word and a simple sentence, you have the power to interact with the Internet.

To prime the pump, the early market leaders such as Amazon and Google saw voice assistants as a pathway for long-term future revenue and a funnel for acquisition of "big data." These devices were positioned at intensely competitive pricing, especially the entry-level models. The smart speaker market is estimated in the range of $7 billion and expected to grow at a compound annual growth rate (CAGR) of 17% or more through 2025. North America absorbs a major share with Asia growing at an even higher CAGR. Smart speakers strive to understand the natural connected speech. Even without voice command, the ability to use your smartphone to remotely take care of stuff you forgot to do (e.g., turn off the lights or lock the front door) even if you are not home is invaluable.

The IVA voice command industry became real when Amazon launched the Alexa Echo in 2014 and the platform was used by many companies. In 2016, Google followed with Google Assistant IVA for Google Home speakers. Google Assistant is supported by the data generated from its web search engine. Typically, IVAs gather usage data from their associated eco-system devices, a process that has created some controversy regarding privacy. The user's inquiry, whether for a delivery order from the local pizza joint or whatever, might not have all the meta-data passed to the intended recipient, but even more disconcerting are the security failures from "bad actors"—the cyber-creeps who hack electronic door locks and security systems.

When Amazon Echo was launched in 2014, fidelity was an afterthought. The inflection point toward quality audio in smart speakers came when Sonos entered the market in 2017. Also that year Apple introduced the HomePod as a self-contained quality sound system as well as the gateway to Apple's Music service through Siri and a hub for controlling lights and the rest of the automated home. I thought the HomePod was quite decent but the market did not. More recently, the Amazon Echo Studio was the first smart speaker to attempt to deliver Dolby Atmos, and from what I hear, sound is back in the spotlight.

Speakers, Speakers and More Speakers

At the core of *Voice Coil* magazine is the knowledge that the world revolves around speakers. Let's focus on a number of speakers that have some advantages for smart speaker appliances. Drivers for smart speakers are not inherently unique, but a single full-range driver, very shallow, high efficiency, and wide dispersion would be a good fit and we will discuss a half-dozen innovative designs.

Dinaburg Technology

The defining characteristic is a concentric passive ring radiator compliantly held in place by surrounds on both the inner and outer periphery. There are a number of positive aspects beyond the obvious benefits of a conventional vent-substitute design. The design techniques enable compactness, lower distortion, extended frequency range, higher efficiency, and more consistent beam width (dispersion).

The passive radiator's diaphragm (compared to a simple vent) blocks midrange sound energy that is otherwise emitted from a port and the passive ring configuration also provides for tighter constructive coupling to the active speaker (and to the room) compared to an open bass reflex port or a non-concentric passive radiator.

From the measured data, it can be seen that the Dinaburg topology results in more output than what would be predicted by the more basic speaker box modeling simulations. Actually, these simulations assume the on-axis response in the range of where the passive ring is, providing a larger effective radiating area and tighter coupling of the bass to the room. This is in the bottom end response with the added benefit of avoiding beaming in the midrange that would have resulted from an active speaker of the same overall size as the outer diameter of the ring radiator. On the other end, the passive ring radiator maintains the speaker's pattern control down to a bit lower inflection point than the active driver has if just mounted on a baffle, (from acoustic suspension, vented or with non-concentric passive).

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Premium Sound Solutions

Premium Sound Solutions (PSS), which was first conceived 50 years ago at Phillips, offers its patented and popular Coscone shallow speaker technology in sizes 2" and up. Essentially moving the magnet position forward and wrapping the diaphragm over the magnetic structure enables about 50% depth reduction. The shallow cone's integrity is due to the unusual contoured rib structure. The stable and linear excursion provides the extra margin from acoustic echo cancelation processing (full-duplex) and is offered in Neodymium and Ferrite.

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Fibona Acoustics

Fibona's Enclosure Magnet Coaxial Transducer (EMCT) technology is reminiscent to many smartphone microspeakers. Fibona's speaker is a full-range shallow coaxial neodymium magnet design with an integrated enclosure. The Fibonacci spiral was the inspiration for achieving diaphragm integrity with reduced depth. The company states "Nature uses this number sequence from the blooming and structure of something as simple as a flower, to the galaxy and stars in the sky." It would be interesting to see a Comsol simulation and I suspect that may show that there is something here of value for transducer engineers.

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Resonado Labs

Another driver technology that has been receiving attention is Resonado Labs' Flat Core Speaker (FCS) technology. FCS technology was introduced to provide the form-factor advantages of racetrack drivers with the acoustic performance of conventional circular drivers. The motor structure of FCS is the key differentiator of the technology as the flat voice coil is able to run along the entire length of the diaphragm and apply uniform force. This enables a larger bandwidth of pistonic behavior for a high-aspect ratio, low-depth driver superior to that of a conventional racetrack driver.

For smart speakers, Resonado Labs has introduced FCS Dual Core adding a second motor structure underneath one flat diaphragm. This enables a larger cone and greater surface area to push more air for the reproduction of lower frequencies. Resonado Labs licenses FCS technology to OEMs/ODMs and is currently licensing partners with Asian OEM/ODMs Zylux Acoustic and SoundLab.

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Trulli Thin Driver

The Trulli TD38S 2" driver is a combination of high thermal power handling, while maintaining a small footprint and shallow depth. This is achieved by repositioning and expanding the voice coil to the juncture of the flat square diaphragm periphery and the juncture of the surround. This 2" speaker boasts a 1.3" voice coil and the square diaphragm confers over 20% more piston area than a round diaphragm. This configuration enables more spider corrugations with huge gains in excursion.

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Mayht

If you have watched a compact Devialet woofer's mechanism go through its dramatic gyrations to produce bass, let's just say that evokes a similar perception of Mayht's operation. Highly compact and powerful, the developer promises increased bass performance. Motor, suspension, and dual membrane architecture moving in opposing directions is quite unique. Mayht feels this is the most efficient way to increase air displacement capability and prevent mechanical resonance of the enclosure, without increasing depth by having to mount two drivers back-to-back. Each driver uses "Negative Compliance," in order to avoid the effect of air pressure working against the movement of the membranes. Mayht developed a distributed suspension because at maximum excursion, the membranes almost touch each other so there is not any space for a conventional secondary suspension.

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Tectonic Audio Labs

NXT was formed 20 years ago to research and commercialize its bending wave flat panel Distributed Mode Loudspeaker (DML) concept. More recently, this has evolved to Balanced Mode Radiators (BMRs), a hybrid technology that blends DML technology with that of traditional pistonic-action loudspeakers. BMR variants have both square and round shapes and combine the low-frequency performance of a traditional loudspeaker, but with a wider directivity and shallower profile.

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How About Smart Speaker Electronics?

There are many compelling reasons for amplifier selection, and most will funnel the designer to select the smart amp choices from Texas Instruments, NXP, Maxim, Qualcomm, and Cirrus Logic. Perhaps your team is experienced using one vendor's parameter setting software for speaker protection or other key component parts from a specific vendor, and we may even want to consider the sound quality. But aside from the amplifier protecting the speaker and not degrading the signal, what else could a designer hope for?

The amplifier development that I see having the most impact on speaker design this year is not even a transducer but a processing technique that will change how most of us design not just integrated speaker systems, but the drivers themselves. This is Dr. Klippel's KCS in chip form from Nuvoton.

Nuvoton

For my recent projects, this is "just what the doctor ordered." This is essentially a dynamic predistortion circuit that is calibrated to the speaker and the enclosure. You might consider smart amps the caterpillar and KCS as the butterfly emerging as the complete form. A smart amp, typically designed with the smartphone as its intended home, is predominately a feed-forward protection circuit, specifically tuned for the limits of the speaker, both displacement and thermal. I have consulted for a couple of the leading smart amp chip vendors over the years and we consistently ran up against Dr. Klippel's patents.

For functionality, aside from keeping the speakers from damage, we wanted a drop distortion. For applications where there is full duplex with acoustic echo cancelers, the lower bass distortion provides significant margin before echoes—and improved wake-word barge-in function.

Another intriguing aspect is what Dr. Klippel has defined as "green speaker design." Given enough of a materials budget you can design an underhung voice coil with a huge magnetic structure and achieve high linearity—but at a cost, both in materials and weight. Or you can design a less extravagant design and use dynamic predistortion to keep your driver on its best linear behavior. This is what Klippel's KCS promises and as I found in a recent project, it actually delivers. I cannot talk about the current product development that so impressed me (it won't be shipping until after this article is published) but let's just say I am a believer. KCS changes what a system designer can demand from smart speakers, Bluetooth speakers, soundbars, conferencing room voice only products, and studio monitors.

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www.nuvoton.com/news/news/products-technology/TSNuvotonNews-000390

Voice Assistance—Privacy and Security

For speaker designers, Privacy and Security are something new on our plate, for sure! Not a topic for a loudspeaker article, but still it is an issue in a smart speaker product requirement document (PRD). Privacy is not just about keeping personal conversations from ending up somewhere public or targeted ads showing up in odd places. However, even more unnerving is the idea of a bad guy hacking your electronic key and unlocking your (not-so) smart front door or turning on your security camera to look around for themselves.

Syntiant

To mitigate these issues consider the edge on-board capabilities to avoid trips to the cloud for simple voice commands (turn on the lights, turn off the hot tub, etc.). Edge (onboard) computing ICs include Syntiant's voice always-on (VAO), which enables up to 60 command words. One reason for a brand (i.e., Uber, Bose, etc.) to consider bypassing Alexa in their products is to hang onto their big data, another is that Cloud computing is not free and can add up to significant costs for device manufacturers.

Google Dialog Flow pricing has just increased as you read this. Text is $0.007 per request and for audio input/output—speech recognition, speech-to-text (STT), speech synthesis, text-to-speech (TTS) telephony—is $0.06 per minute. Amazon Lex charges $0.004 per API call, which amounts to $14.60 per annum for a voice-enabled device with 10 voice interactions per day. This unbounded operating cost can be prohibitive for device builders. Edge processing can tap into freely available on-device compute resources to significantly reduce or eliminate cloud and connectivity expenses. But a few dozen commands cannot avoid the requirement of access to the Internet.

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Native Voice

The work-around from Native Voice sidesteps Alexa with custom wake words that designers can integrate into their architecture to bypass Alexis and Google Assistant paths to the cloud. I perceive this as similar to a VPN personal network, enabling brands direct access to the customer base. Essentially an on-demand voice services library that is using your voice. With about 50 direct connections to multiple voice services (e.g., Alexa and Siri), as well as the brands that are developing their own (e.g., “Hey Spotify” and “Hey Uber”). Native Voice is working with leading brands in fitness, retail, and smart home technology to create a large collection of voice services and partnering with audio hardware device manufacturers to integrate their SDK to minimize or by-pass the “Big Brother” toll-keepers.

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Linkplay

Another key provider is Linkplay and its integrated mobile app audio software and hardware solutions for lossless audio formats, Wi-Fi, wireless multi-room and multichannel audio streaming, and Alexa Voice Service (AVS) integration. Components include a Wi-Fi module, device cloud software, and global streaming content.

A single-chip solution and customization is available giving brands the ability to build out and launch innovative and differentiating products. There are more than 200 connected products powered by Linkplay in the home audio market in various wireless soundbars, speakers, and audio receivers, which are now optimized to integrate in devices across multiple verticals, including smart home devices such as robust security to and from the device through authentication and encryption.

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The next game-changer for smart speakers will be completely wireless operation, specifically the power cord. As we have touched upon here, there is a lot going on with smart speakers, and all that drains battery power—but that is another story for another time. ***VC***

Test Bench

Three New 25mm High-Frequency Transducers from BlieSMa

By Vance Dickason

Test Bench has previously featured two 34mm dome tweeters from BlieSMa: the T34A aluminum-magnesium alloy dome tweeter in the June 2018 issue of *Voice Coil*, and the T34B Beryllium Dome tweeter in the March 2019 issue of *Voice Coil*. BlieSMa, which is located in Blieskastel, Germany, was founded January 16, 2018 by Stanislav Malikov. Over the last 20 years, Malikov has worked at Ultrasound Technologies; at industry legend Morel as a QC manager and a transducer engineer; and at Accuton, the very well-known high-end driver OEM, as a production engineer for more than 8 years. For the last 3.5 years BlieSMa has been producing some very high-quality OEM high-frequency transducers and has now introduced four more 25mm domes to its catalog. Three of the four are being featured in this month's Test Bench article. All three tweeters appear to be built on the same platform, but optimized for each diaphragm material. The versions being tested this month are the T25S silk dome, the T25B Beryllium dome, and the T25D diamond dome tweeter. These will be explicated in that order, starting with the T25S silk dome model.

In terms of features, all three share pretty much the same feature set:

- 25mm dome diaphragm—both the Beryllium and Diamond domes have a first break-up mode in the ultrasonic range
- Extremely low moving mass—Mms=0.16g (silk), 0.10g (Beryllium), and 0.15g (diamond)—for better transient response and higher output
- Fully saturated neodymium motor with a copper sleeve shorting ring for low nonlinear and modulation distortion
- 2mm linear excursion and a large pole vent for low distortion low-frequency operation
- Narrow surround for less "soft dome" coloration
- Flush-mounted surround and a rear-mounted magnet system for flat frequency response and wide off-axis response
- No Ferrofluid for improved dynamics
- Underhung voice coil with a Titanium former
- Flexible and lightweight tinsel leads from Denmark
- Thick powder-coated mounting flange
- Aluminum rear chamber with wool damping material
- Gold-plated terminals
- Extremely wide frequency range 2.2kHz to 22kHz (silk), 2.2kHz to 40kHz (Beryllium and diamond)

BlieSMa T25S-6 Silk Dome

I began testing the T25S-6 silk dome version (**Photo 1**), using the LinearX LMS analyzer to produce the 300-point impedance sweep illustrated in **Figure 1**. The T25S impedance resonance occurs at a moderately low 894.5Hz (factory spec is 940Hz). With a 5.24Ω DCR (Re) (factory spec is 5.2Ω), with the minimum impedance for this tweeter measuring 5.47Ω at 5.28kHz.

After completing the impedance testing, I recess mounted the T25S-6 tweeter in an enclosure that had a baffle area of 12" × 6". Then I measured the device under test (DUT) using the LoudSoft FINE R+D 192kHz analyzer and the GRAS 46BE microphone (courtesy of LoudSoft and GRAS Sound & Vibration) both on- and off-axis from 200Hz to 40kHz at 2.0V/0.5m, normalized to 2.83V/1m (one of the really outstanding tricks FINE R+D can do), using the cosine windowed FFT method. All of these SPL measurements also included a 1/6 octave smoothing.

Figure 2 shows the T25S-6 on-axis response to be a flat ±2.75dB from 1kHz to 35kHz with zero diaphragm

Photo 1: BlieSMa T25S-6 silk dome transducer

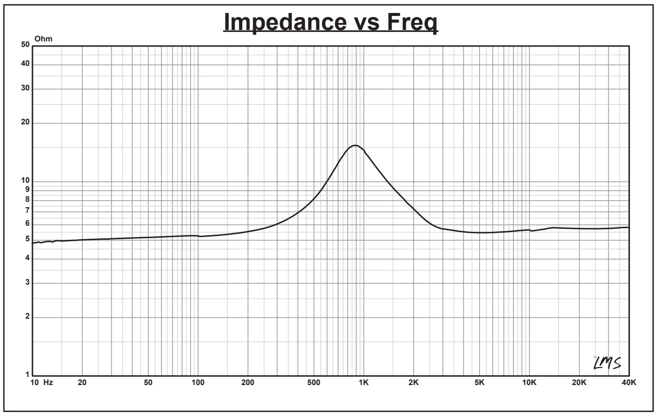

Figure 1: BlieSMa T25S-6 impedance plot

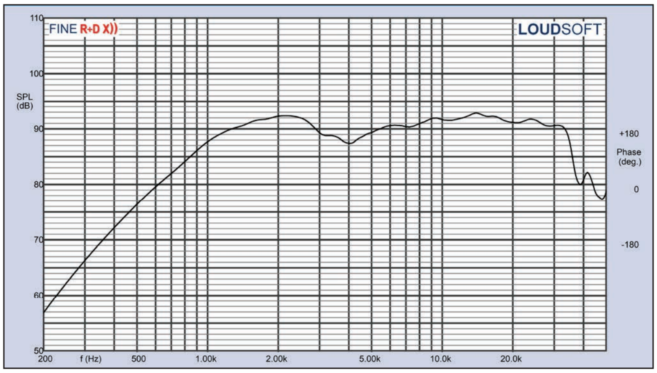

Figure 2: BlieSMa T25S-6 on-axis frequency response

break-up modes. **Figure 3** gives the on- and off-axis response of the T25S-6 high frequency device. The off-axis curves normalized to the on-axis response are shown in

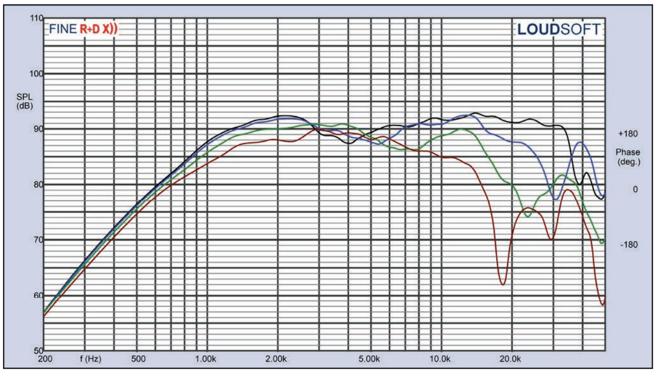

Figure 3: BlieSMa T25S-6 horizontal on- and off-axis frequency response (0° = black; 15° = blue; 30° = green; 45° = purple)

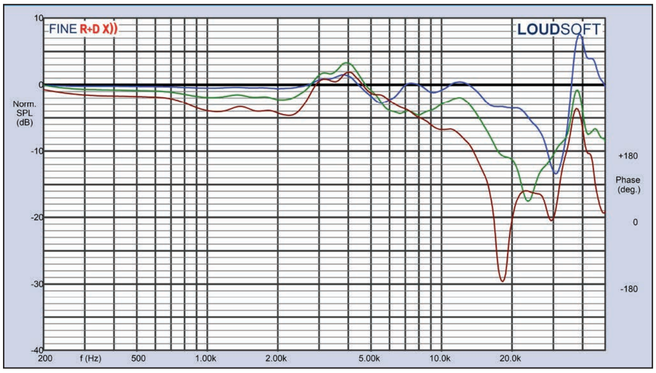

Figure 4: BlieSMa T25S-6 normalized on- and off-axis frequency response (0° = black; 15° = blue; 30° = green; 45° = purple)

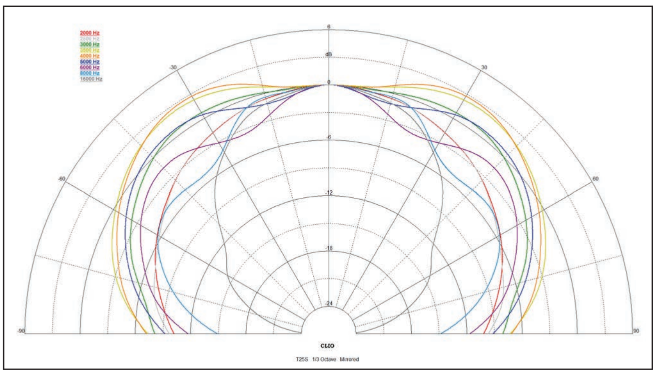

Figure 5: BlieSMa T25S-6 180° horizontal plane CLIO polar plot (in 10° increments)

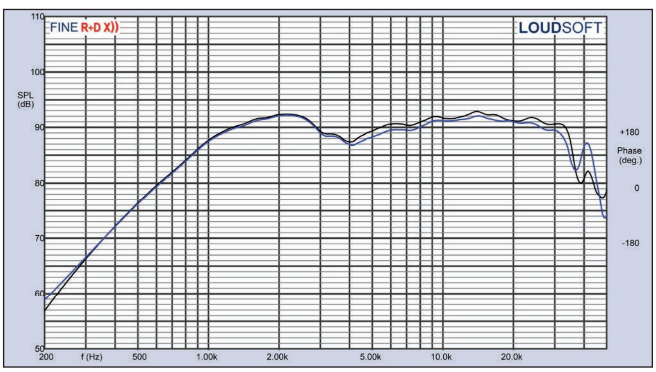

Figure 6: BlieSMa T25S-6 two-sample SPL comparison

Figure 4. And, the CLIO 180° polar plot (measured in 10° increments) is shown in **Figure 5**. **Figure 6** shows the two-sample SPL comparison, indicating the two samples were closely matched to within 1.25dB or less throughout most of its operating range.

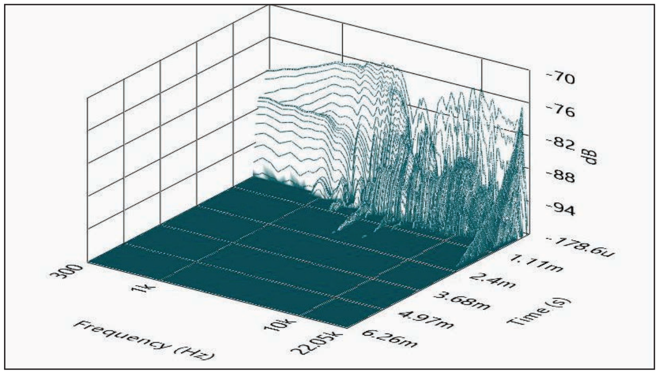

Figure 7: BlieSMa T25S-6 SoundCheck CSD waterfall plot

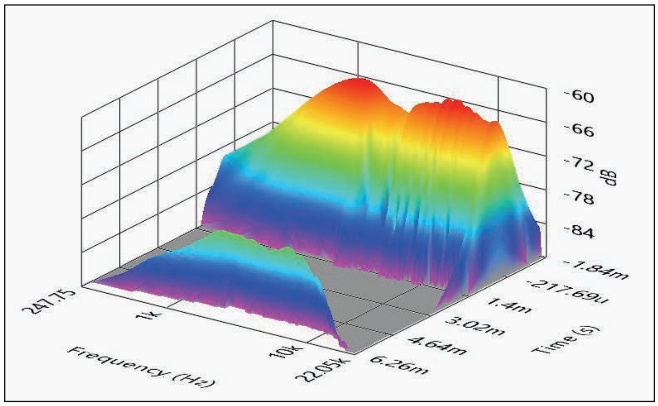

Figure 8: BlieSMa T25S-6 SoundCheck STFT surface intensity plot

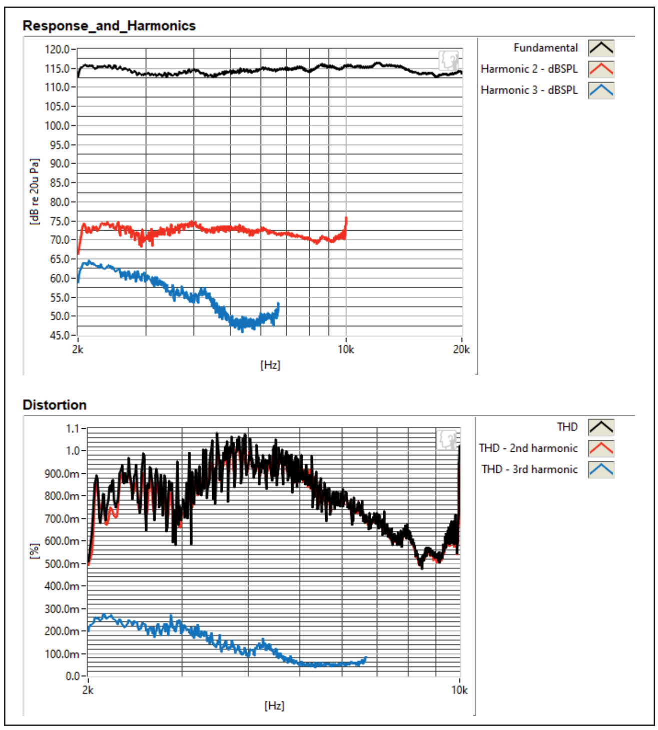

Figure 9: BlieSMa T25S-6 SoundCheck distortion plots

The next test procedure was to initiate the Listen, Inc. AudioConnect analyzer along with the Listen SCM 1/4" microphone (provided courtesy of Listen. Inc.). I used the SoundCheck 18 software to measure the impulse response with the tweeter recess mounted on the test baffle. Importing the impulse response into the Listen SoundMap software resulted in the cumulative spectral decay (CSD) plot (commonly referred to as a "waterfall" plot) given in **Figure 7**. I used the same data to produce the Short Time Fourier Transform (STFT) displayed as a surface plot shown in **Figure 8**.

For the last objective test, I set the 1m SPL to 94dB (2.54V for the silk dome) using a pink noise stimulus (SoundCheck has a built-in generator and SLM utilities for this purpose), and measured the second (red curve) and third (blue curve) harmonic distortion at 10cm, depicted in **Figure 9**.

BlieSMa T25B-6 Beryllium Dome

Moving on to the Beryllium version, testing commenced for the BlieSMa T25B-6 Beryllium dome depicted in **Photo 2**, and also started using the LinearX LMS analyzer to produce the 300-point impedance sweep illustrated in **Figure 10**. The T25B impedance resonance occurs at a moderately low 894.5Hz (factory spec is 1050Hz). With a 5.25Ω DCR (Re) (factory spec is 5.2Ω), with the minimum impedance for this tweeter measuring 5.47Ω at 10.3kHz.

After finalizing the impedance testing, I recess mounted the T25B-6 Beryllium dome tweeter in an enclosure that

Photo 2: BlieSMa T25B-6 beryllium dome version

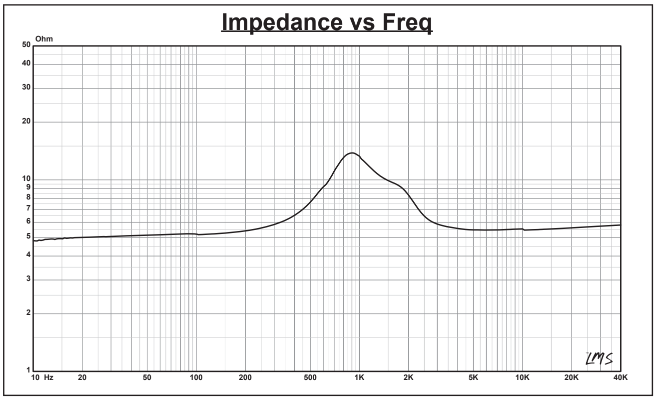

Figure 10: BlieSMa T25B-6 impedance plot

had a baffle area of 12" × 6" then measured the DUT again using the LoudSoft FINE R+D 192kHz analyzer and the GRAS 46BE microphone both on- and off-axis from 200Hz to 40kHz at 2.0V/0.5m normalized to 2.83V/1m using the cosine windowed FFT method. All of these SPL measurements had a 1/6 octave smoothing applied.

Figure 11 shows the BlieSMa T25B-6 on-axis response to be a flat ±2.5dB from 1.3kHz to 42kHz (±2.0dB from 2.4kHz to 28kHz), with the Beryllium diaphragm break-up mode centered on 48kHz. **Figure 12** gives the on- and off-axis response of the T25B-6 high-frequency device, with the off-axis curves normalized to the on-axis response in **Figure 13**. The CLIO 180° polar plot (measured in 10° increments) is shown in **Figure 14**. **Figure 15** shows the two-sample SPL comparison, indicating the two samples

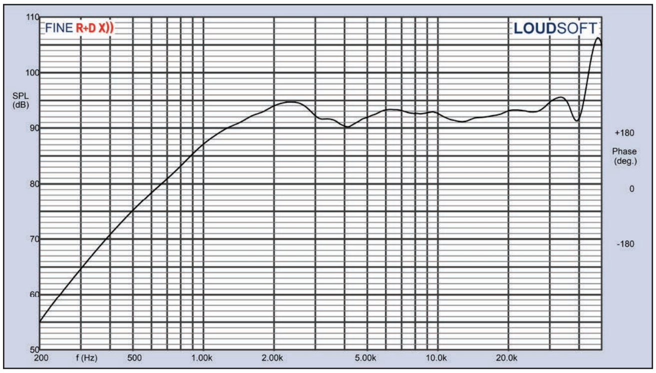

Figure 11: BlieSMa T25B-6 on-axis frequency response

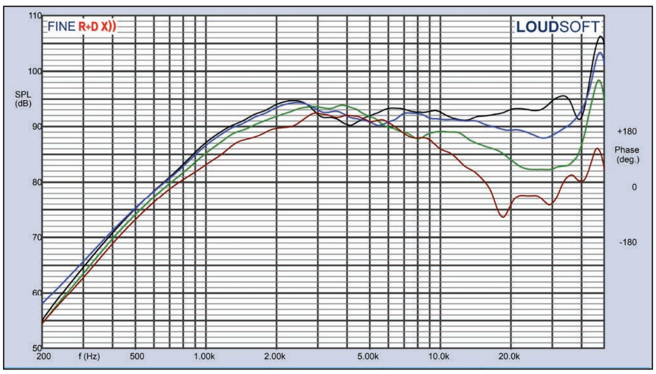

Figure 12- BlieSMa T25B-6 horizontal on- and off-axis frequency response (0° = black; 15° = blue; 30° = green; 45° = purple)

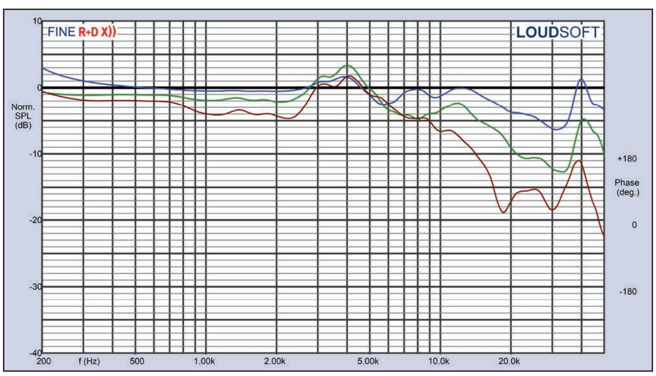

Figure 13: BlieSMa T25B-6 normalized on- and off-axis frequency response (0° = black; 15° = blue; 30° = green; 45° = purple)

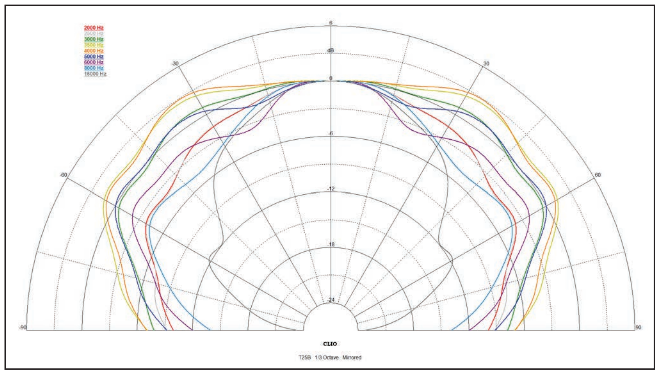

Figure 14: BlieSMa T25B-6 180° horizontal plane CLIO polar plot (in 10° increments)

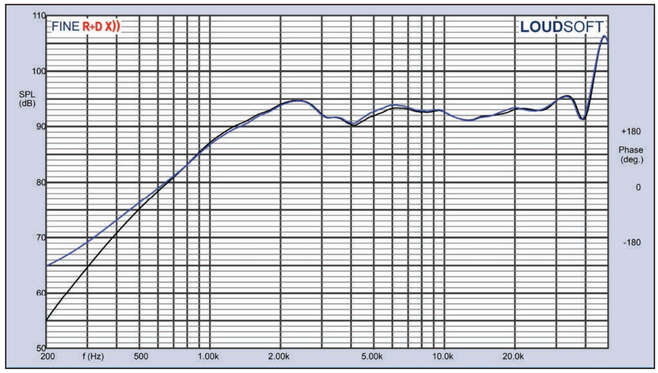

Figure 15: BlieSMa T25B-6 two-sample SPL comparison

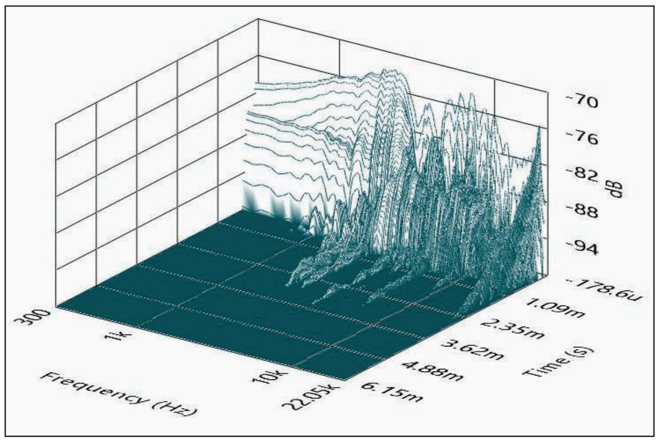

Figure 16: BlieSMa T25B-6 SoundCheck CSD waterfall plot

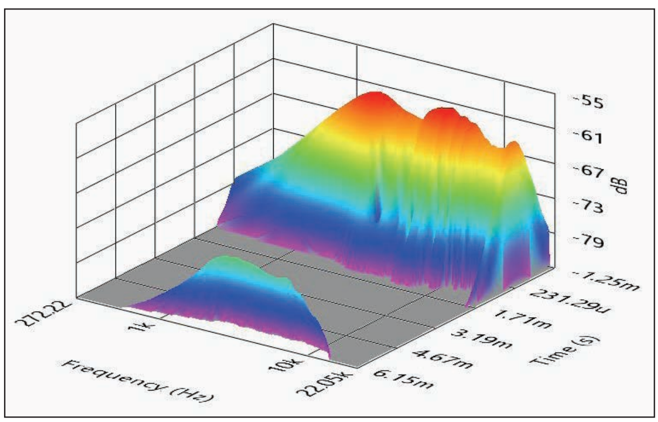

Figure 17: BlieSMa T25B-6 SoundCheck STFT surface intensity plot

were closely matched to within 0.9dB throughout most of its operating range.

For the next test procedure, I again used the Listen AudioConnect analyzer along with the Listen SCM 1/4" microphone (provided courtesy of Listen. Inc.). I used SoundCheck 18 to measure the impulse response with the tweeter recess mounted on the test baffle. Importing the impulse response into the Listen SoundMap software resulted in the CSD waterfall plot shown in **Figure 16**. I then used the same data to produce the STFT displayed as a surface plot in **Figure 17**.

For the last SoundCheck test, I set the 1m SPL to 94dB (2.82V for the Beryllium dome) using a pink noise stimulus, and measured the second (red curve) and third (blue curve) harmonic distortion at 10cm, depicted in **Figure 18**.

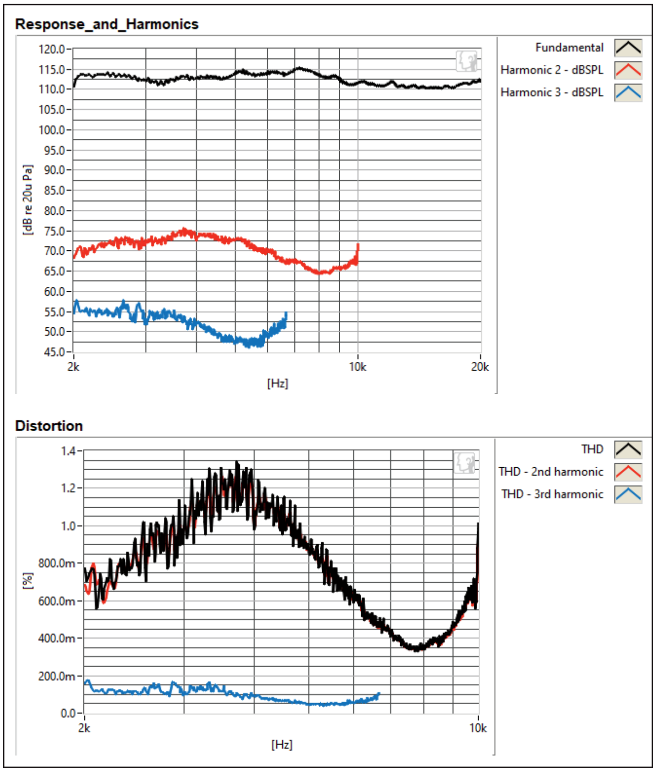

Figure 18: BlieSMa T25B-6 SoundCheck distortion plots

BlieSMa T25D-6 Diamond Dome

The last device I tested was the diamond dome BlieSMa tweeter, which is still a rather exotic tweeter diaphragm material. Testing for the T25D-6 diamond dome version seen in **Photo 3**, commenced one last time using the LinearX LMS analyzer to produce the 300-point impedance sweep illustrated in **Figure 19**. The BlieSMa T25D-6 impedance resonance occurs at 1.12kHz (factory spec is 870Hz). With a 5.18Ω DCR (Re) (factory spec is 5.2Ω), with the minimum impedance for this tweeter measuring 5.58Ω at 10.28kHz.

Following the impedance testing, I recess mounted the T25D-6 diamond diaphragm tweeter in an enclosure that had a baffle area of 12" × 6" then I measured the DUT using the LoudSoft FINE R+D 192kHz analyzer and the GRAS 46BE microphone both on- and off-axis from

200Hz to 40kHz at 2.0V/0.5m, normalized to 2.83V/1m using the cosine windowed FFT method. All of these SPL measurements also had a 1/6 octave smoothing applied.

Figure 20 shows the BlieSMa T25D-6 tweeter's on-axis response to be a flat ±2.5dB from 1.2kHz to 37kHz (±2dB from 2.8kHz to 37kHz), with the diaphragm break-up mode located above 50kHz, even higher than the Beryllium diaphragm. **Figure 21** gives the on- and off-axis response of the T25D-6 BlieSMa high-frequency device. **Figure 22** shows the off-axis curves normalized to the on-axis response. **Figure 23** gives the CLIO 180° polar plot (measured in 10° increments). The two-sample SPL comparison is illustrated in **Figure 24**, indicating the two samples were closely matched to ≤ 0.25dB throughout this BlieSMa tweeter's entire operating range.

For the final set of test procedures, I again configured

Photo 3: The BlieSMa T25D-6 diamond dome tweeter

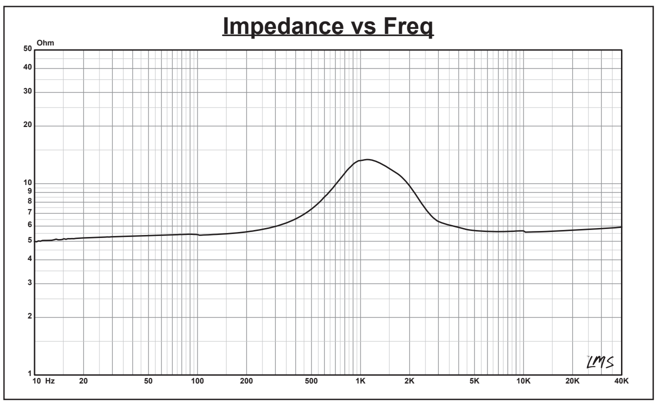

Figure 19: BlieSMa T25D-6 impedance plot

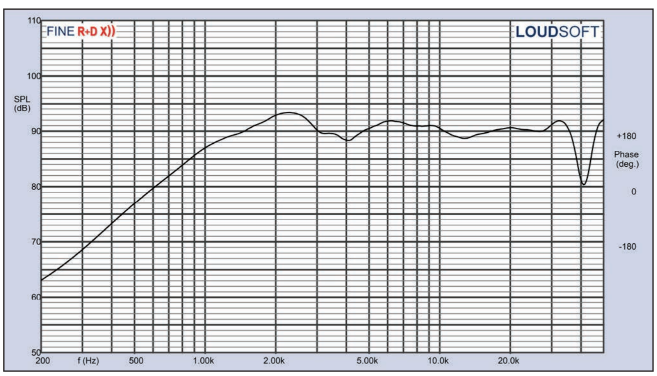

Figure 20: BlieSMa T25D-6 on-axis frequency response

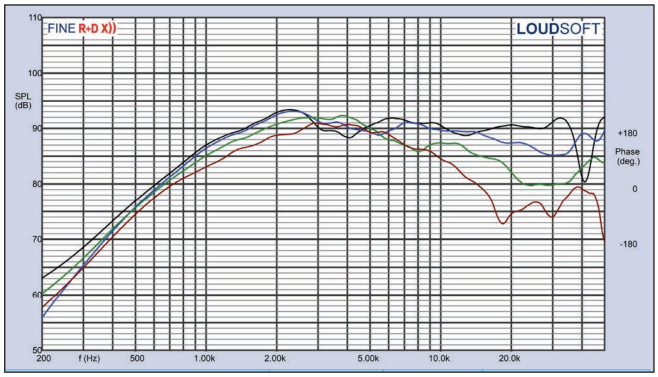

Figure 21: BlieSMa T25D-6 horizontal on- and off-axis frequency response (0° = black; 15° = blue; 30° = green; 45° = purple)

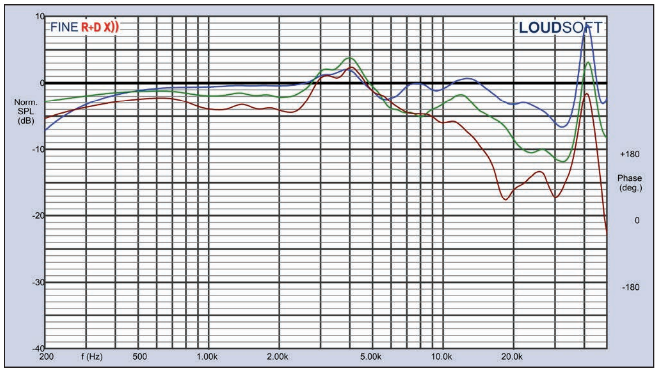

Figure 22: BlieSMa T25D-6 normalized on- and off-axis frequency response (0° = black; 15° = blue; 30° = green; 45° = purple)

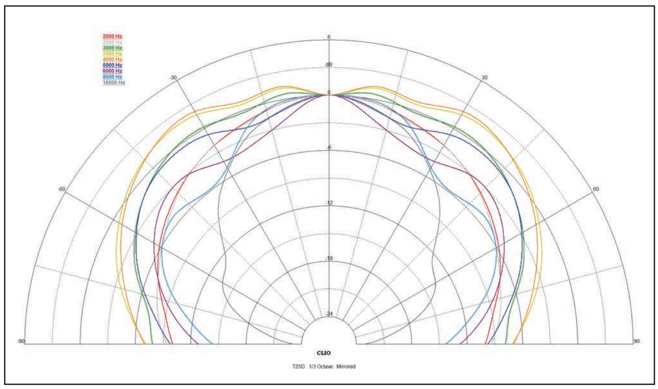

Figure 23: BlieSMa T25D-6 180° horizontal plane CLIO polar plot (in 10° increments)

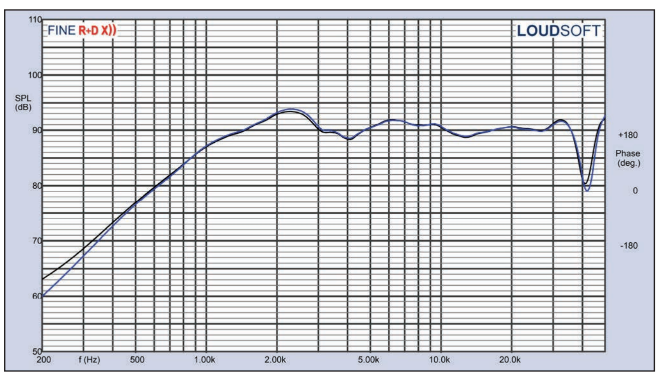

Figure 24: BlieSMa T25D-6 two-sample SPL comparison

the Listen AudioConnect analyzer (along with the Listen SCM 1/4” microphone all provided courtesy of Listen, Inc.) I then used SoundCheck 18 to measure the impulse response with the tweeter recess mounted on the test baffle. Importing the impulse response into the Listen Inc. SoundMap software resulted in the CSD waterfall plot given in **Figure 25**. **Figure 26** used the same data to produce the STFT displayed as a surface plot.

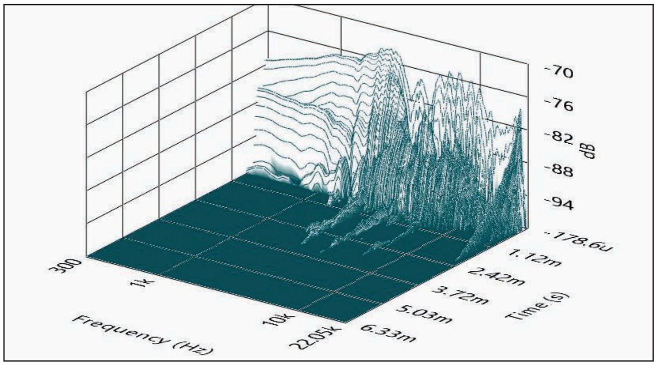

Figure 25: BlieSMa T25D-6 SoundCheck CSD waterfall plot

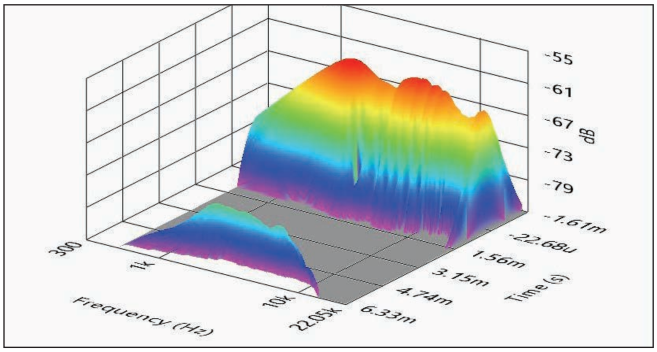

Figure 26: BlieSMa T25D-6 SoundCheck STFT surface intensity plot

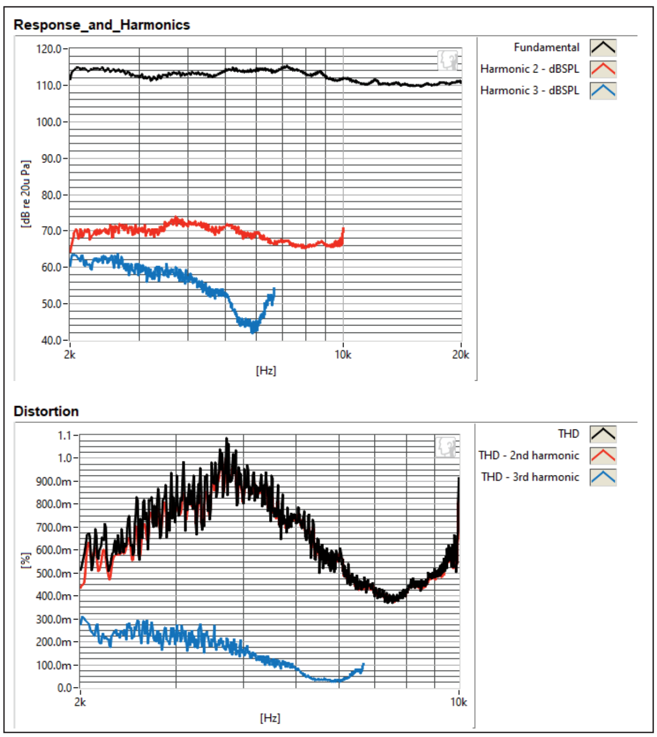

Figure 27: BlieSMa T25D-6 SoundCheck distortion plots

For the last objective test, I set the 1m SPL to 94dB (3.90V) using a pink noise stimulus, and measured the second (red curve) and third (blue curve) harmonic distortion at 10cm, illustrated in **Figure 27**.

After the first BlieSMa product (the T34A-4) was released to the OEM driver market, it was obvious that Malikov and his company, BlieSMA, are skilled practitioners. These three new 25mm dome tweeters also have the same fit, finish, and overall outstanding build quality and performance befitting OEM transducer products intended for the high-end loudspeaker market. For more information, visit www.bliesma.de. ***VC***

Submit Samples to Test Bench

Test Bench is an open forum for OEM driver manufacturers in the loudspeaker industry and all OEMs are invited to submit samples to *Voice Coil* for inclusion in the monthly Test Bench column.

Send samples in pairs and addressed to:

Vance Dickason Consulting
333 S. State St., #152
Lake Oswego, OR 97034
(503-557-0427) | vdconsult@comcast.net

All samples must include any published data on the product, patent information, or any special information necessary to explain the functioning of the transducer. This should include details regarding the various materials used to construct the transducer. For woofers and midrange drivers, please include the voice coil height, gap height, RMS power handling, and physically measured Mmd (complete cone assembly including the cone, surround, spider, and voice coil with 50% of the spider, surround and lead wires removed).

Industry Watch

By Vance Dickason

New Klippel dB-Lab Software Release

In July 2021 Klippel released its new major software update. The main software platform dB-Lab 212 for both, QC 7 and R&D, now provides a shared sensor management. The Klippel Multi-Scanning Workbench is now fully released bringing the near-field holographic sound field scanning technology that provides directivity, sound power, and room correction to a much smaller form factor. For comprehensive measurement of distortion using multi-tone stimuli, the MTON module has also now been released. Simulation tools are complemented by a new linear simulation module LSIM, dedicated to speaker and enclosure design.

Klippel QC software has also been upgraded with a new automation control interface and multichannel support for any Windows or ASIO audio interface, as well as wave-file-based open-loop testing. It includes more flexible options for testing and synchronization and for smart or stand-alone audio devices. Many small but useful tools and updates in existing modules round up this new major release.

Celestion Releases New CDX1-1412 Driver

Celestion has introduced the new CDX1-1412, a 1” exit, neodymium magnet high-frequency compression driver with a compact design that’s well suited for smaller two-way cabinet designs and other portable loudspeaker applications. Designed and developed at Celestion’s headquarters in Ipswich, England, the CDX1-1412 has a 1.4” diameter edgewound copper-clad aluminum voice coil and the previously noted 1” exit size. It is stated to have 70W power handling and delivers a 107dB sensitivity performance over a frequency range of 1,500Hz to 20,000Hz, with a recommended crossover frequency of 2,000Hz.

It also incorporates a single-piece Polymide film diaphragm and surround, and is fabricated using a rigid engineering thermoplastic with a standard 2× M5 bolt fitting. Acoustic foam is utilized to minimize internal air cavity resonances, dampening unwanted reflections from the inside of the cover.

ALTI Update—September, 2021

The Audio & Loudspeaker Technologies International (ALTI) will be celebrating its 60th anniversary at ALTI-EXPO 2021. Details will be coming soon in the ALTI Newsletter, which is free. Visit the ALTI website or subscribe to the newsletter at: https://altiassoc.us16.list-manage.com.

Technical presentations at ALTI will include sessions from JBL, Mvoid, NTi, Composite Sound, Klippel, and more! Also look for the unveiling of Incubator participant Dinaburg Technologies’ ground-breaking technology with technical presentations and working models.

New to ALTI-EXPO, Yunsheng USA will be exhibiting and John Ebert will present the culmination of his five-part White Paper “The Five Biggest Events Since the Tariff War,” which provides insight into the magnet industry and factors that will affect manufacturing, sourcing, and costs.

The Speaker Builders Workshop will also be back with Klippel presenting its new dB-Lab 212 for the Klippel Analyzer System to aid in the modeling and design of a project speaker. Eminence will discuss how the specs are put to work, and will help the class in assembling their speakers. Parts Express will talk about enclosure design and tuning a passive radiator. The speaker will be mounted into the cabinet and the radiator tuned. Then, Audio Precision will test the performance of each participant’s system. Participants can keep their completed systems.

ALTI will also provide more time for attendees to network with each other and exhibitors. Added to the CEO’s reception on Saturday, breakfast and lunch on both days, and the annual banquet, it’s all about face time!

ALTI also offers FREE All Access Passes for qualified college students as a service to the future of our industry and to provide valuable networking among ALTI member employers and their future employees.

ALTI members can get a FREE Exhibits Plus Pass at the ALTI website, and member pricing is available for All Access Passes that includes the breakout sessions, breakfast and lunch both days, and the annual ALTI Banquet. If you are not a member, you can still take advantage of ALTI-EXPO. Request a Promo-Code from ALTI-EXPO Exhibitors for a FREE Exhibits Plus Pass, or save $100 on an All Access Pass! Visit: https://altiassoc.org/alti-expo-2021 for information. See you in Orlando, FL, on October 24–25! ***VC***

HiVi

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