Focus

Enhancing Low-Frequency Performance

Part 2 — Even More Bass

By Mike Klasco (Menlo Scientific, Ltd.)

Last month the innovations we surveyed ranged from smart amplifier speaker overload protection with distortion canceling, amplifier control of the loudspeaker's Thiele-Small (T-S) parameters through motional feedback, and feedback network integrated circuits optimized for sensor servo control of speaker drivers. Here in Part 2, we continue our survey of bass magic to eke out more bass with sophisticated techniques.

Servo Feedback

Our first topic is further discussion of servo negative feedback. Aside from feedforward techniques (e.g., KCS), where the speaker system characteristics are modeled, a more complex solution is negative feedback where a speaker voice coil or diaphragm position sensor is used to reduce distortion and extend bass response. Just about every amplifier today uses negative feedback where the output of a system is fed back to reduce the deviation of the output from the input. By returning an out-of-phase output signal back to the input, errors can be significantly canceled.

Hybrid topologies, such as feedforward correction in conjunction with feedback to clean up the residual distortion, are common in active noise cancellation (ANC) headphones to acoustically stealth nuclear submarines, or even Klippel Sound Control (KCS) designs. The sensor is located between a stationary component of the speaker and an appropriate moving part to provide an error correction reference signal back to the amplifier.

Loudspeakers have an order of magnitude or more distortion than the amplifiers with which they are used, especially at the bottom-end and at large signal conditions. We could get off track wrestling with the non-linearities of the motor, the spider, and the surround but instead of fixing each less-than-perfect aspect to achieve a linear transfer function, we could take a more elegant route—negative feedback correction.

The ultimate objective is to get the entire system within the feedback loop, while the biggest challenge is including the speaker in the feedback loop. Anyone who thinks feedback correction (servo-control) is a no-brainer to instantly fix a poor speaker is in for more than a few surprises.

Servo control definitely offers performance advantages not readily achieved by tweaking physical construction, but there are many new design rules with which to wrestle. Nevertheless, finding a robust and stable sensor and the right positioning location can be a traumatic experience.

Speaking of trauma, in 1968, Infinity, when it was still a start-up, launched its Servo-Statik 1 and created quite a splash in the audiophile world. The (Cerwin-Vega) 18" woofer uniquely boasted a sensing coil coincident with the main voice coil. The signal induced into the sensing coil was fed back to the bass amplifier to provide the "servo" correction. While dramatic, it was not a completely stable design and eventually it was discontinued in 1972.

Yet perhaps the greatest stumbling blocks from making feedback speakers practical has been two-fold. One was that amplifier feedback loops are picky. To keep the "network" stable, an integrated solution is needed for a fighting chance of a reliable design. While today essentially all subwoofers have the amp built-in and these "systems" are optimized with all the variables predetermined by the design team. The other factor is finding a feedback sensor that is stable and more accurate than the speaker itself.

One of my first clients was Velodyne in the early 1980s (Velodyne subs are back in the market, by the way). We used a piezo sensor on the voice coil collar and eventually were able to have a stable network with high gain and distortion specs comparable to a good amplifier.

Feedback sensing can be game changing well beyond extended bass. Two aspects are not obvious but significant—better Acoustic Echo Cancellation (AEC) and Barge In. For a full-duplex operation, where a second talker can contribute without having to force their way into the call, there are various signal processing algorithms. But complicating life, these algorithms are surprisingly sensitive to speaker distortion. DSP guys look for bottom-end distortion levels in even modest applications (e.g., speakerphones) for unrealistic distortion performance not easily found in real-world speakers. With sensor feedback distortion cancellation, AEC can do its job better with more margin.

"Barge In" is a term that originated with half-duplex telephony where participants struggle to interrupt and engage with the incumbent talker. These days "barge-in" additionally refers to the voice assistant's (smart speaker) ability to respond quickly and accurately to voice commands while ignoring its own audio.

This is easier said than done because the loudspeaker is typically closer to the microphone than the person giving the voice command. Existing Barge In algorithms use DSP to subtract or cancel out the music so only voice is captured. This method only works if the music captured by the microphone perfectly matches the music delivered to the loudspeaker. In reality, the loudspeaker is off from the input signal. Because of this, the cancellation of the music is marginal. With the feedback sensor signal and the feedback loop to force the loudspeaker to produce the subtraction signal more completely, it nulls the loudspeaker's output with essentially the mic's pickup of the voice command.

subVo

Founded four years ago, subVo is bringing two new absolute position diaphragm tracking technologies, both contactless with no long-term degradation over time. One tracking technology is optimized for larger speakers and the other is optimized for microspeakers. Using AI algorithms, it is possible to adapt an anti-distortion model using the diaphragm tracking sensor. The tracking sensor enables the speaker to provide almost perfect frequency response, maximum excursion, and reduced distortion.

The sensor feeds back to the amplifier/DSP an output feedback voltage signal directly proportional to the diaphragm position. The speaker compensation algorithm controls the feedback loop, enabling each speaker to self-calibrate. All speakers have slightly different specs as they roll off the production line and with self-calibration this disparity is eliminated enabling higher production yield.

Beyond enabling more consistent quality without sorting and binning critical components, servo-correction tracks the exact position of the cone at any given moment, enabling lower distortion and extending bass, all in smaller enclosures. Speakers can be driven to their limits without damage and with low distortion. Compensation for degradation of the loudspeaker (surround/spider wear and tear, and neo magnet sensitivity losses) over time, temperature, humidity and power compression. For more information contact: Joe@subvo.com or visit www.subvo.com

Virtual Bass

Virtual bass (missing fundamental bass) is less bass without losing the oomph. When pure tones are produced together, they combine to produce additional tones whose frequencies are the sum and the difference of the two original tones. If the original frequencies are 32Hz and 48Hz, the resultant frequencies will be 80Hz and 16Hz. You might think you hear this, but it won't show up on a spectrum analyzer (at least not at the perceived frequencies). This psychoacoustic phenomenon improves the perceived bandwidth of the low-frequency audio content.

Discovered by Tartini in 1714, the technique was used to produce tones in the 32Hz and 64Hz octaves, using smaller organ pipes. Three hundred years ago, church construction lacked the structural integrity for huge pipes. Today, virtual bass has relevance for audio engineers struggling to achieve richer sound from compact form factors. Bottom line, virtual bass processing enhances the bass perception by tricking the human auditory system to perceive low bass from their higher harmonics.

The goal of the virtual bass is to extend the perceived low-frequency bandwidth utilizing this psychoacoustic missing fundamental phenomenon. There have been a couple of these processors—way back there was the SRS True-Bass, which did not have good acceptance. (SRS was folded into DTS, which went through another acquisition and this process has faded from the scene.)

Waves is famous for its plug-ins used for professional post production and you can hear their work on music and dialog tracks. Waves' first signal processing effort for consumer audio was Maxxbass, which is a real-time analyzer of the existing bass of the program, then creating harmonics around an octave higher than the original. Launched back in 2006 with their design win in the Altec InMotion docking station for the iPod with more than 1 million sold. While not for studio monitors, it is very interesting for a wide range of challenging situations—from entry-level no-sub soundbars, smart speakers, landscape garden audio, in-wall and ceiling speakers, and also high-intensity club sound systems and live concerts.

How realistic the processing is depends on the application. If used to extend a smartphone from 700Hz down to 350Hz, this won't have a good outcome, but in a soundbar without a subwoofer, extending the bass response from 90Hz down to 50Hz, you will have magic.

Virtual bass processing has other useful attributes for contrasting applications—such as when the sound system's bass capabilities are too powerful. Neighbors down the road from amphitheaters or in multiple family dwellings don't want to hear your bass. Virtual Bass means that only the harmonics are used to evoke what you think is real bass. The virtual bass processing has no real deep bass and therefore tends not to travel as far or through walls as well as the real thing. For more information visit: www.nuvoton.com/products/smart-home-audio/audio-enhancement/audio-enhancement-maxxaudio

I briefly mentioned that MaxxBass, by processing the harmonics of the bass (the missing fundamental approach), reduces the speaker's cone excursion for the psychoacoustical processed "restored" bass, as there is no real energy in this band. MaxxBass processed signals travel through walls following the STL band rating for the faked bass—so the leakage at 100Hz is far less than the leakage at 50Hz. This is very handy when the bass is high passed almost an octave higher than what is (sort of) heard. So for keeping your garden audio system or your in-walls out of your neighbor's ears, or the hotel bar's sound system out of the hotel rooms, it can be a viable solution.

Cardioid Arrays

But what about when only the real thing will do—like for professional performances?

In recent years, active cardioid speaker arrays have often been employed to address undesired rear low-end radiation. These require additional cancellation loudspeakers, amplifiers, and signal processors. These arrays are called "Cardioid" because the sensitivity pattern is "heart-shaped" and good at rejecting sounds from other directions. What about passive cardioid techniques? While new for speakers, Shure first used cardioid techniques in 1939 in its first moving coil dynamic microphone: the Unidyne Model 55. In speakers, the passive cardioid design can minimize rear bottom end radiation, at least over a narrow frequency band.

One example is Fulcrum's passive cardioid speakers, which produce high sound pressure in the forward direction and consistent attenuation in the rearward direction without additional amplifiers, drivers, and signal processors. Another application (suggested to me by our editor Vance Dickason) is avoiding the Allison rear wall bound response anomaly effect).

The "subcardioid" behavior of Fulcrum Acoustic's patented loudspeakers is produced by an ingenious acoustical circuit, which balances the position of the low-frequency driver, the enclosure depth and volume, and a unique arrangement of acoustical elements including rear-mounted ports and calibrated resistive elements.

Passive cardioid technology achieves impressive directional control mechanically within the enclosure in less than half the space required for active cardioid array systems. This allows for cardioid performance in applications that might not otherwise have the budget or space to accommodate an active cardioid array. A single-amplified, full-range passive cardioid speaker delivers system cost benefits over traditional bi-amplified active cardioid speaker arrays by enabling up to eight speakers to be driven by a single amplifier channel.

For those interested in more information on this topic: Fulcrum Acoustic (www.fulcrum-acoustic.com/audio-technology-insights-resources/passive-cardioid-technology); Kii Audio (www.kii.audio); or Musikelectronic Geithain GmbH (www.me-geithain.de).

Dinaburg Technology

Another brilliant twist on a long-established acoustic design technique is from Dinaburg Technology. Its "stabilizer" is essentially a passive ring radiator—but there is a half-dozen not so obvious second- and third-order benefits that speaker engineers should consider in the designs that we will discuss.

While computer modeling using bass box electrical filter analogy software is common, the most sophisticated teams use COMSOL and ANSYS. COMSOL Multi-Physics takes advanced design to another level and I have seen active noise cancelling, using fluidics modeling, extend to a soccer ball (from a ping pong ball) using their techniques. Mikhail Dinaburg has also based his work on BOYLE's physical law:

PV= CONSTANT

The absolute pressure exerted by a given mass of an ideal gas is inversely proportional to the volume it occupies if the temperature and the amount of gas remain unchanged within a closed system

The theory of "Diffuser (diaphragm) Active Volume Stabilization" focuses on a small volume of air around the diffuser (diaphragm—usually cone shaped). This volume is equal to the active area of the diaphragm (S), multiplied by the amplitude of oscillations (H), or $V=S \times H$.

Sound is created in a narrow strip of air around the diffuser, in a height equal to the amplitude of the diaphragm oscillations. The state of this air volume will affect whether the diffuser oscillates in strict accordance with the sound signal of the amplifier or differ from it. The diaphragm is affected by the braking force when moving inside the box and the same force holds it, when moving outside the box—forward. These forces change the speed of the diffuser and introduce distortion, and so any speaker diaphragm by its nature produces sound with distortion.

Maintaining a constant volume of air around the diffuser is key to minimizing distortion. In order to prevent changes in the active volume of the diffuser, it is necessary to install a stabilizer (e.g., a passive ring radiator) positioned concentrically to the active diaphragm. In some ways, it is slightly similar to a horn or waveguide, with the speaker diaphragm's coupling to the transmission medium (air) is improved.

After the first functional prototypes were made, it was found that when held in hand outside the enclosure, it creates low distortion with a linear transfer function. You can see how, under the action of the active part of the diffuser, the stabilizing part oscillates and both parts work together.

This use of the back wave makes better use of the bottom-end energy than a separate passive radiator, and sealed box designs waste through the shearing action of box fill fibers as heat. In a sealed design, this requires heavier cabinets to avoid enclosure barreling (all panels ballooning).

And yet this energy can be used to create sound without distortion and significantly increase the efficiency of the speaker, not just with the benefits of bass reflex and vent substitute designs, but some other little-considered aspects.

The active part radiates energy, not only forward and backward, but also in the plane of the diffuser, around the circumference to the sides. Of course, as the wavelengths approach the piston dimension, the waveform oozes outward with a lower Q than the upper range (same as with horn mouth). Put a small mouth horn on a large flat baffle and you will find some interesting polar pattern interactions.

While analogies to jet engines are not directly parallel, the early "pencil" jets provided very low efficiency, and today's high bypass jets use sort of an early fan jet core with a much larger diameter, drastically improving the coupling of the force as the back-end transition. Pardon my DC analogy to an AC phenomenon.

Back to the Dinaburg passive ring radiator. The stabilizing part (passive ring) of the diffuser (the active cone) performs the following functions:

1. Maintains a constant volume of air around the diffuser and prevents sound distortion (from poor coupling occurring).
2. Creates a balance between the active and stabilizing parts.
3. Collects unused energy in the plane of the diffuser (which would end up as a wasted side lobe) and maintains on-axis response for more direct sound into the listening area. As with established bass reflex designs the energy reflected from the back wall of the cabinet and turns it into useful sound energy, in addition to the sound that the active part creates.
4. Extends the bass range of the sound zones of the active part of the diffuser with minimal distortion.

Position 1. The active part and the stabilizing part of the diffuser, creating sound together.

Position 2. When around the active part of the diffuser, the stabilizing part of the diffuser is independently suspended. Under the action of the signal, the active part began to move forward on H1. The sound will be created when the air in front of the active part is completely compressed. But around the diffuser, there are no obstacles in diameter, and part of the compressed air will go to the side and carry with it energy, which will cause the stabilizing part to move backward, thereby relieving the load from the active part, which causes the speed of the diffuser to slow down forward. If the speed of the diffuser changes, there are distortion.

Position 3. This is the return of the active part of the diffuser to its original position. The total volume of air around the two parts of the diffuser is:

V1+V2=V PV=CONSTANT

If the volume of air around the diffuser does not change in the height of the amplitude of the oscillations, then there are no additional forces that distort the sound.

Visit: http://dinaburgtech.com/technical.html for more information.

Super Compliant Enclosure Fill Virtual Back Volume

No this is not virtual bass such as MaxxBass, but n-bass (a real metamaterial), which is a box fill that is super compliant/super porous and creates about twice the volume compliance than the air it displaces. Similar to a sponge, the new material adsorbs air in the back volume when the speaker membrane moves inward and releases the air again when the membrane moves outward, making it easier for the membrane to vibrate at full amplitude.

Developed a decade ago, n-bass has been basically unobtainable aside from its use in Apple iPhones. The technology was transferred from the NXP speaker group, which was acquired by Knowles and then developed this unique material further.

Knowles then sold its microspeaker group, which is now known as Sound Solutions International (SSI). SSI was then acquired by Foxconn and n-bass was spun off as SSI New Materials.

What is relevant in all that information is that n-bass is now more widely available. Applications that could benefit from smaller enclosures or more bass range from soundbars, in-walls, ceiling speakers, and garden speakers to concert sound. For more information contact, SSI New Materials (Zhenjiang) Co., Ltd. (www.n-bass.com).

In Summary

Regarding all of these technologies that have been discussed, the most exciting news is that now there are real-world tools for moving what is possible (and practical) forward. Each of these innovators is welcoming collaboration with the audio industry. ***VC***