Sound Design + Interactive Experiences, p. 38

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From the Editor's Desk

We've Got Range

audioXpress is all about *range*—range, in the sense of content diversity. Our international team publishes insightful articles on a wide variety of topics such as speaker design, sound analysis, glass audio projects, and hi-fi product reviews. This month we deliver once again.

First, I suggest you check out the interview with innovative sound designer Andrew Spitz (p. 38). Much of his current work focuses on "interactive experiences."

Interested in speaker design? We've got you covered. Consider starting with the first part of Richard Honeycutt's series, "Speaker Design School." Then turn to page 8 for the second part of the series "Ribbon and Planar Magnetic Loudspeakers." Finally, flip to page 14 for the conclusion to Ton Giesberts's series on his active loudspeaker system project.

Ready for a comprehensive transducer test? You can always rely on Vance Dickason's expert analysis. This month he presents the results of tests he ran on a Morel TiCW 634Nd (p. 30).

Once you've had your fill of speaker articles, read what Gary Galo thinks about remastered editions of David Hancock's 1967 recording of the Dallas Symphony's presentation of Rachmaninoff's *Symphonic Dances* (p. 24). What would you choose: vinyl, digital, or both?

If you're a glass audio enthusiast, head to page 35 to refresh your understanding of power supplies for hollow-state electronics. Transformers and rectifiers are covered in detail.

Lastly, note that the long-awaited amp test page referenced in Richard Honeycutt's *audioXpress* November 2012 article—"Differences in Amp Sound: What's the Truth?"—is now available at www.edcsound.com/amptest.htm.

Regards,

C. J. Abate
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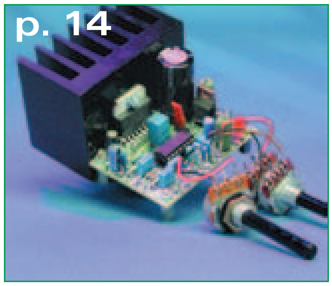

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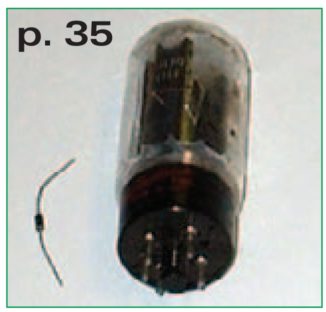

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By Mike Klasco and Steve Tatarunis (United States)

Ribbon and Planar Magnetic Loudspeakers (Part 2)

High Performance and Wide Dynamic Range

More sound system designers recognize the benefits of ribbons

While the pure ribbon was once considered the loudspeaker family's "weak sister," there has been quite the turnaround. In the 1990s, several firms started to use ribbon speakers for high output professional applications. Their motivation was to achieve the elusive combination of audiophile quality, wide dynamic range, high acoustic output, and extended frequency response. Now, there are ribbons that have finally met these performance goals, and they are being successfully installed in applications ranging from studio monitors to commercial venues.

In 1978, Genelec was one of the first companies to use the studio monitor ribbon in its S30 active studio monitor's top-end (see **Photo 1**). Following Genelec's lead, SLS, Stage Accompany (SA), Griffin (using Stage Accompany drivers), and the Samson Rubicon all introduced monitors that boasted ribbon tweeters. The highly respected ADAM and a few others used a variant on the ribbon planar, the Air Motion Transformer, essentially a non-tensioned pleated diaphragm of similar construction to a ribbon planar.

PROFESSIONAL RIBBON APPLICATIONS

So what about professional application for ribbons? Aside from the studio monitor tweeters used in Genelec products and a few others, the first really ambitious high-output ribbon came out in the late 1980s. The European-based Stage Accompany based its version on a scale-up of a Philips home speaker design. Stage Accompany has recently returned to the U.S., and it is focusing on touring sound applications and cinema sound systems.

SLS Loudspeakers took high-power/high-sound pressure level (SPL) ribbon driver technology even further, in terms of power handling, and developed its own range of high-performance ribbon drivers based on high-energy neodymium magnets and high-temperature tolerance Kapton/aluminum diaphragms. These drivers are incorporated in its line array systems for various professional commercial and cinema applications (see **Photo 2**).

For large ventures (e.g., permanent installation and touring) multiple ribbon elements may be arrayed for additional output and control over directivity. Throughout pro audio, the line array technique is popular because their excellent pattern control is an asset in many difficult installations.

We briefly mentioned in the first article in this series (*audioXpress*, November 2012) that with ribbons the diaphragm moves in way similar to any cone speaker, but with a significant subtle difference that makes ribbons superior for line array configurations. The ribbon's advantage is that the diaphragm is directly driven by the conductors, which are in intimate contact with the film diaphragm. Ribbons have been called regular phase (by Foster Electric, a large loudspeaker OEM) or isoplanar because the entire diaphragm is being driven, and the radiating surface is moving as one. The ribbon diaphragm itself is typically tensioned and clamped to keep it aligned in the gap between the magnets. A ribbon's surface is immersed in a uniform magnetic field and moves in a uniform manner across its surface in response to electrical input. So by their very nature, properly designed ribbons are able to maintain uniform phase response across their surfaces, unlike traditional transducers. This results in a more uniform cylindrical wave, regardless of frequency.

Photo 1: This is the front view of a Genelec S30 speaker. (Photo courtesy of Genelec)

Conventional, non-arrayed or non-line source speakers approximate a point source. A point source generates a spherical sound field in which sound pressure drops by 6 dB as a listener's distance from the source doubles. However, a line source (e.g., an array of long ribbon drivers) generates a cylindrical wave in its nearfield. This nearfield can extend large distances, depending

Photo 2: The SLS Ribbon Studio Monitor uses a line array system. (Photo courtesy of SLS Loudspeakers)

on frequency and line source length, to hundreds of feet. A distinct characteristic of this nearfield used in line array applications is that its sound pressure drops only by 3 dB as a listener's distance from the source doubles. Although an in-depth discussion of point source versus the planar line source radiation pattern is beyond the scope of this article, suffice it to say the ribbon works more like a line source than the cruder approximation resulting from cone speakers mounted in a row (see **Photo 3**).

Aside from the faster drop off of level in long throw applications, a point source radiators' array will produce an interference pattern with some comb filtering interference affects. Any transducer's radiation pattern is heavily influenced by the relationship between the wavelengths of the sound produced in relationship to the transducer's active portion's size (e.g., effective radiating area).

For commercial sound applications, the line source ribbon's additional control over directivity is quite useful, both for where the sound goes (i.e., to the audience) and where it does not go (e.g., a room's acoustically reflective floor and ceiling). That enables a much smoother and more consistent coverage and a superior ability to deliver direct sound to the audience even in reverberant environments. Even for home theaters, a single rectangular ribbon tweeter's typical dispersion pattern is one of the few ways to gracefully meet THX's tight vertical pattern control guidelines, without resorting to stacking multiple round tweeters. One of the many advantages of the control offered by a cylindrical wavefront is that listeners near the speakers do not have to suffer high SPLs in order to ensure audience members further back perceive adequate SPLs. Thus, a line source system is a superior tool where intelligibility is important or the venue is acoustically difficult (e.g., a longer throw is required with multiple glass surfaces, reflective or domed ceilings, or in corridor applications such as those found in museums).

With cone speakers, the vibrational energy from the voice coil has to pass via transconductance up through the voice coil bobbin and into the cone apex, where the sound will move from the cone's center outward and radiate into the air. Cones are uniform (piston like) up to the midrange. The effective radiating area versus frequency is never uniform on a cone speaker and the phase and group delay is not consistent over the operating frequency range. Specifically, the effective radiating area becomes smaller through decoupling as the frequency goes up. But with ribbons, the entire diaphragm moves simultaneously at all frequencies. Incidentally, this is antithesis of the concept of random phase transducers (e.g., NXT or "distributed mode") and the cone speaker falls somewhere in between. It is generally conceded that ribbons tend to sound better than cones, while the distributed mode panels sound worse.

However, in certain commercial sound applications a uniform phase line array's excellent tight pattern control is not what is needed, and the

Photo 3: The ribbon line array increases throw distances and produces tighter vertical control at lower frequencies. (Photo courtesy of SLS Loudspeakers)

distributed mode nature of NXT panel radiation (i.e., the "anti-ribbon") can be an advantage. The wide coverage and random phase of a well-executed NXT design (e.g., the Armstrong i-Ceilings panel) is an advantage in the struggle to achieve wide uniform coverage with low ceilings as well as higher acoustic levels before feedback. The distributed mode panel's random phrase nature helps prevent feedback, effectively randomizing the signal's phase. The Armstrong panels actually sound quite decent.

HIGHER POWER HANDLING & INCREASING EFFICIENCY

In the past, ribbons fell short on power handling, sensitivity, and low-end response. These limitations were due to the ferrite magnets' low energy and the Mylar film diaphragms' low-temperature tolerance. Using neodymium magnets with their higher magnetic energy density, the sensitivity issues and limitations on magnetic gap width were lessened. Superior performance from advanced mechanical and thermal films enabled higher temperature operation and further increased sensitivity by providing the possibility of thinner and lighter diaphragms.

Designers of ribbons using ferrite magnets have always been confronted with response anomalies. This is in part because the sound radiation coming from the diaphragm, at least in the case of the more linear and efficient push-pull designs, must pass between the bar magnets behind and in front of the diaphragm. The depth required for the ferrite magnets results in resonant cavities with tuned peaks, typically between 5 kHz and 10 kHz. Neodymium has a high magnetic density, so the magnets are not as thick, the air column in front of the diaphragm is shallower, any resonance is minimized, and it is typically toward the top of or beyond the audio band.

Assuming a generous gap between the diaphragm and the magnets, neodymium structures yield 0.6 to 0.8 tesla (6,000 to 8,000 gauss), whereas a ferrite magnet provides less than half that. The resulting higher electromagnetic damping eliminates the usual "camel hump" response of an underdamped speaker. Neodymium's higher energy enables wider gaps without the flux dropping below useable intensities, so higher diaphragm excursions are possible without hitting the magnets. As the excursion is increased, the diaphragm surface area can be reduced to a more reasonable size, which also improves dispersion.

Compared with cone and compression drivers, the structure is open on a ribbon diaphragm. Though air is not a good conductor of heat, the conductors' surface area enables heat transfer from the diaphragm so operating temperatures tend to be lower than a typical cone speaker's voice coil for a given sound level. Temperature tolerance is different because it is dependent on the diaphragm's materials and fabrication. Older Hi-Fi planar film ribbon designs used Mylar polyester that was laminated with an adhesive layer to an aluminum foil conductor pattern. Especially for smaller drivers, Mylar ribbons have limited power handling and get soft above 175°F. Higher temperature materials were needed beyond Mylar, and these were introduced in the 1990s.

A CLOSER LOOK AT NEW DIAPHRAGM MATERIALS

DuPont Mylar (e.g., generically polyethylene terephthalate or PET polyester) has traditionally been used for both ribbons and electrostatic planar diaphragms, not to mention Coke bottles! A single-sided aluminum conductor layer is usually laminated to the film, but some applications use conductive layers on both sides of the film. The conductor side is then silk-screened using a chemical-resistant coating that prints the pattern, with the uncoated paths of the surface being the conductor paths that will be etched away. This roll or sheets of silk-screened laminate is then chemically etched and die-cut into individual diaphragms. The lead-out wire is attached to the completed diaphragm, and the diaphragm is then assembled and tensioned into the magnetic structure frame. It is a difficult process, and each manufacturer has its own trade secrets for etching aluminum, lamination between the conductor and film, tensioning, and lead-wire attachment.

Mylar yields good sound quality but cannot tolerate conductor heating much beyond 80°C because that is Mylar's glass transition point (Tg), and at that temperature, it becomes soft. Because most ribbon diaphragms are tensioned, under high-power input the conductive voice coil heats up the substrate, the diaphragm tension drops because of the Mylar's softening, and the film becomes rippled, leading to distortion or buzzing.

Teijin DuPont Film's Teonex (polyethylene-naphthalate or PEN) high-performance polyesters are now commonly used in ribbons and can handle almost double the Mylar PET temperature before softening. Teonex's higher tensile strength (which increases the tension you can use) and higher Young's modulus (correlates to the speed of sound in the material) is better than in Mylar, yet the specific gravity is lower. The cost is just above that of Mylar. Sound quality is excellent, as reflected in both frequency response extension, and lower linear and nonlinear distortion.

Ribbon Diaphragm Material—Temperature Ratings

Material	Temperature Rating
Mylar	80°C
Teonex	160°C
PEEK	220°C
Kapton	400°C

Table 1: Temperature tolerance varies for different diaphragm materials.

For the highest power handling, a polyimide (PI) is the ultimate solution, such as Kapton from DuPont, or similar materials such as Uplex and Apical. Their high-performance polyimides can tolerate temperature excursions to 800°F, but it is likely the adhesive layer will handle only half of that. Some of the more advanced ribbons use a cast polyimide over the aluminum conductor, avoiding the adhesives layer's failure. Even with adhesiveless polyimide diaphragms, at elevated power levels, the aluminum work hardens and can crack. Selection of the aluminum alloy for the conductor is still another consideration.

While the ribbon diaphragm is essentially a flex circuit board, it is also something less clinical. It is the diaphragm and the largest factor in the sound quality. Most high-temperature polyimides have less than ideal sound quality characteristics as their Young's modulus is not as high as it ideally might be, and the tan delta (the internal loss damping factor) is also not ideal. If ultimate thermal tolerance can be lowered, then PEEK film should be considered. This is a high-temperature film which is positioned above PEN and PI but below Kapton (see **Table 1**). There are fillers that might be considered for enhanced tan delta (for even lower diaphragm distortion) and PEEK can be heat bonded without adhesive to the aluminum conductor layer.

How else are ribbon speakers increasing power handling? Stage Accompany has offered a forced-air cooling option on its high-power ribbons. SLS has integrated a heatsink to pull heat off the aluminum conductors on the diaphragm, and DuPont has introduced thermally conductive black heat emissive Kapton to improve the film substrate's ability to dump the heat.

Ribbons continue to have their place in upscale stereo and home theater products and are advancing into mainstream pro-sound and cinema applications as their clean sound quality, ideal radiation characteristics for columns and line arrays, low weight, and increasing robustness, sensitivity, and acoustic output capacity attract more sound system designers. *aX*

By Ton Giesberts (The Netherlands)

Active Loudspeaker System (Part 2)

A subwoofer for use with multimedia loudspeaker systems

As with most small enclosures, the bass response of the active two-way loudspeakers described last month has its limits. The compact subwoofer will be a welcome addition for most enthusiasts. With its adjustable filter, this subwoofer will also work well with other small (satellite) speakers (see **Photo 1**).

As a rough guide, the loudspeakers' bass responses tend to improve in proportion to their dimensions. This is not completely true since there are a few tricks to produce a reasonable bass response from a small enclosure. But, these tricks may reduce the loudspeaker's power-handling capacity, efficiency, and sound-pressure level.

Of course, what everybody wants is a loudspeaker with a volume of a few liters, using a small woofer, which is still capable of reproducing perfect-sounding bass guitar or pipe organ sounds down to 30 or 40 Hz. Unfortunately, this is a physical impossibility. Such a bass response requires a larger air displacement, which requires larger speaker diaphragms and enclosures.

Last month's active two-way system article (*audioXpress*, November 2012) had a 13-cm woofer in a closed box with a volume of about 4 liters. This is a small-sized enclosure and the response graph indicated that not much should be expected of the response below 100 Hz. I decided not to use artificial (i.e., electronic) means to boost the low-frequency response because I already had a much better solution: a separate subwoofer placed underneath the table (or desk), inconspicuously adding another octave to the two miniature loudspeakers' response.

Photo 1: This compact subwoofer will also work well with other small speakers.

DUAL-BASS REFLEX

So, how did the subwoofer's design come about? I tried to get the subwoofer to complement the two-way loudspeakers discussed in last month's article. This meant my design had to be good quality, without becoming too large or costly. Since I intended to place the subwoofer underneath a desk, I based the dimensions on a medium-sized PC tower case, and I arrived at a 50-liter volume.

Next, I looked for speakers that would fit in such an enclosure and would provide an acceptable bass response. For the speakers to fit in a PC tower-sized enclosure, they couldn't be large. They also needed magnetic shielding, so they wouldn't cause problems when placed near a monitor or TV.

After a search, I found Visatron's W170SC, a robust 17-cm woofer with a 50-W load-handling capacity, a 36-Hz resonant frequency, and a large 20-mm peak diaphragm displacement. The W170SC is most at home in a bass reflex enclosure with a 25-liter volume and is capable of reproducing frequencies down to about 45 Hz. Since a single 17-cm speaker only produces a limited amount of air displacement, I used two of them in a "double" subwoofer using an enclosure with a 50-liter volume.

Just like the two-way loudspeakers described last month, the subwoofer is an "active" design, with an electronic filter and its own power amplifier. To make this design more useful, the cut-off frequency is adjustable between about 75 and 145 Hz. This enables the subwoofer to be combined with virtually any compact (satellite) speakers. The power amplifier is the same integrated dual-bridge amplifier used in the two-way loudspeaker design. Since the WS170SC has an 4-W impedance, the amplifier has a 20-W power output per woofer.

I found the subwoofer's frequency response and sound pressure level perfectly complemented the small two-way loudspeakers' response.

MIXER & LOW-PASS FILTER

The design similarities to the active two-way loudspeaker should be obvious from the circuit diagram since the same ICs have been used (see **Figure 1**).

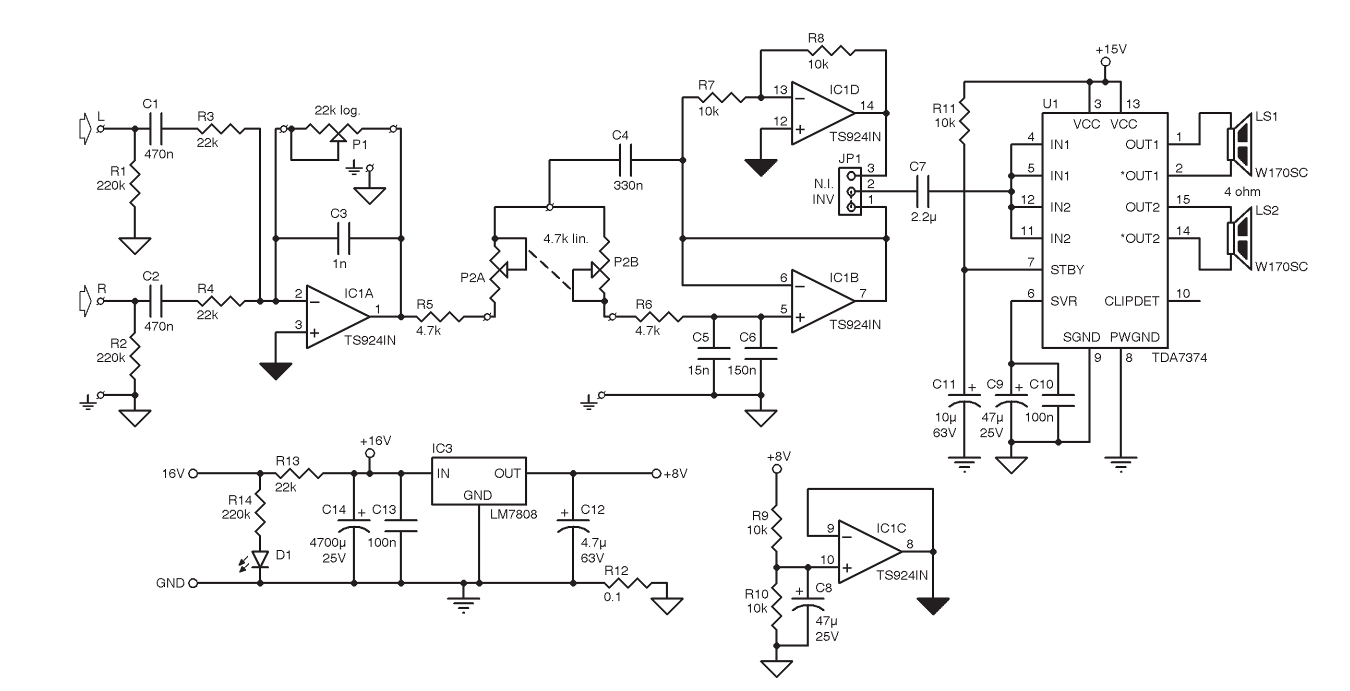

Figure 1: The circuit is very similar to the active two-way system.

Again, I've used a TS924IN quad rail-to-rail op-amp (IC1) for the input stage and filter. Voltage regulator IC3 supplies a stabilized 8 V to this op-amp. R12 was added to separate the signal ground and the supply ground when a single power supply is used for several channels. The fourth op-amp in the TS924IN (IC1c) creates a stable virtual ground that is at exactly half the supply voltage.

The stereo output from the pre-amplifier is fed to the left (L) and right (R) inputs. Since the subwoofer is monophonic (most are) the left and right channels are first combined using a simple mixer (IC1a). (Stereo information below 100 Hz is virtually nonexistent.) This has been implemented in its inverting form so the gain can be turned down to zero with P1. The mixer can accept signals at practically any level and be connected to another power amplifier's output. C3 suppresses high-frequency interference and restricts the bandwidth to about 7 kHz.

The crossover filter consists of a second-order Butterworth low-pass filter. This filter is built around IC1b and its cut-off frequency adjusts using stereo potentiometer P2. When both P2a and P2b are set to 0 W (effectively a short-circuit), the cut-off frequency is about 145 Hz. When the resistance is turned to its maximum, the cut-off frequency moves to about 75 Hz. For use in conjunction with the active two-way speakers a 100-Hz cut-off frequency is ideal, but P2 covers a wider range so the subwoofer can also be used with other (satellite) speakers. I found that the subwoofer could be adapted for use with various satellite speakers by using trial and error to adjust the volume and frequency controls to their best settings.

I've also added an inverter (IC1d) to the filter's output, which is selected when jumper JP1 is in the upper position. Depending on the relative positioning of the subwoofer and satellites, a (subtle) improvement can sometimes be heard when the subwoofer's phase is inverted. I should point out that the IC1d's output is actually in phase with the input signal, since the filter round IC1a is also inverting. To use the phase-switching facility after encasing the electronics, I recommend replacing JP1 with a changeover switch.

POWER AMPLIFIER

Because of my positive experience with the active two-way system, I decided to use the same TDA7374B integrated power amplifier. The small number of external components required and the good internal protection circuits are among this IC's advantages. Another excellent feature

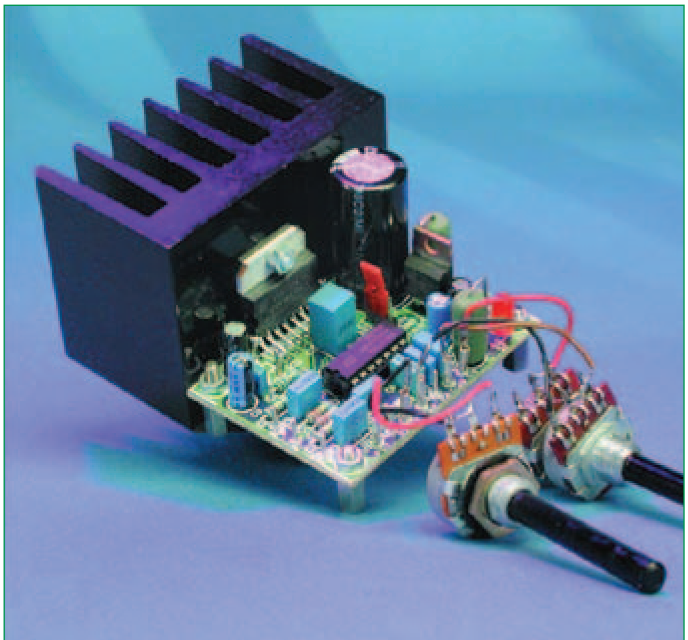

Photo 2: The completed PCB for the subwoofer is attached to its heatsink.

a) b)

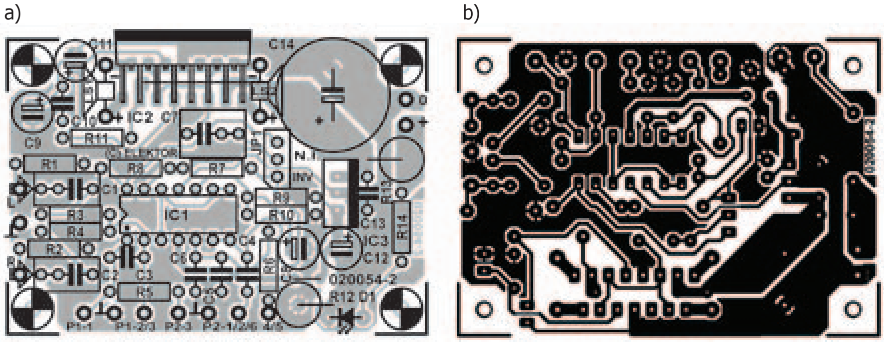

Figure 2: All space is utilized on this PCB. Note the two wire links.

PARTS & COMPONENTS

Capacitors
- C1, C2 = 470 nF
- C3 = 1 nF, lead pitch 5 mm
- C4 = 330 nF, lead pitch 5 mm
- C5 = 15 nF, lead pitch 5 mm
- C6 = 150 nF, lead pitch 5 mm
- C7 = 2 µF2 MKT, lead pitch 5 or 7.5 mm
- C8, C9 = 47 µF, 25-V radial
- C10, C13 = 100 nF, lead pitch 5 mm
- C11 = 10 µF, 63-V radial
- C12 = 4 µF7, 63-V radial
- C14 = 4,700 µF, 25-V radial lead pitch 7.5 mm max., 17-mm diameter

Miscellaneous
- Bass reflex tube (two), 72-mm diameter, 145-mm length (Conrad Electronics)
- Heatsink (e.g., SK100 Fischer, height approximately 50 mm)
- JP1 = three-way pin header with jumper
- LS1, LS2 = Visaton W170SC 4-Ω (Conrad Electronics)
- PCB
- Sealant tape (Monacor/Monarch)
- Wadding material (BAF)
- Wood: 12-mm MDF (see Figure 3)

Resistors
- R1, R2 = 220 kΩ
- R3, R4 = 22 kΩ
- R5, R6 = 4 kΩ7
- R7–R11 = 10 kΩ
- R12, R13 = 0.1 Ω, 5 W
- R14 = 2 kΩ7
- P1 = 22-kΩ logarithmic, mono
- P2 = 4-kΩ7 linear, stereo

Semiconductors
- D1 = LED, red, high-efficiency, 5 mm
- 1C1 = TS9241N ST (Farnell)
- 1C2 = TDA7374B ST (RS Components)
- 1C3 = 7808

of this IC is that it contains two amplifiers, so each woofer can have a separate power amplifier.

Since the woofers' impedance is 4 W, the TDA7374B delivers a bit more power than it did with the two-way loudspeakers (i.e., the power output is 2 × 20 W with a 17-V supply voltage). The selected heatsink should be capable of handling this. When a continuous signal is played at full power, you should, in theory, use a heatsink with a 1.5-KW thermal resistance. But with normal music, a heatsink rated at 2.5-KW is sufficient in practice. Fischer's SK100 was used in the prototype shown in **Photo 2**.

PRINTED CIRCUIT BOARD

All the subwoofer's electronics fit on a small 44-mm × 63-mm PCB. That is a bit smaller than the two-way loudspeaker's PCB! **Figure 2** shows the layout and parts numbers.

Again, I placed the input pins on the PCB's left-hand side, the supply pins on the top-right, and the speaker pins on either side of the power amplifier IC. The connections for the potentiometers are at the front of the PCB. Take great care when connecting stereo potentiometer P2. The numbers next to the pins correspond to those in the circuit diagram. As long as the connections between the PCB and potentiometers are less than about 4 cm, you can use standard connecting wire. If you have a longer connection, you must use shielded audio cable. With that in mind, there are earth pins included near P1 and P2 for connection to the shield.

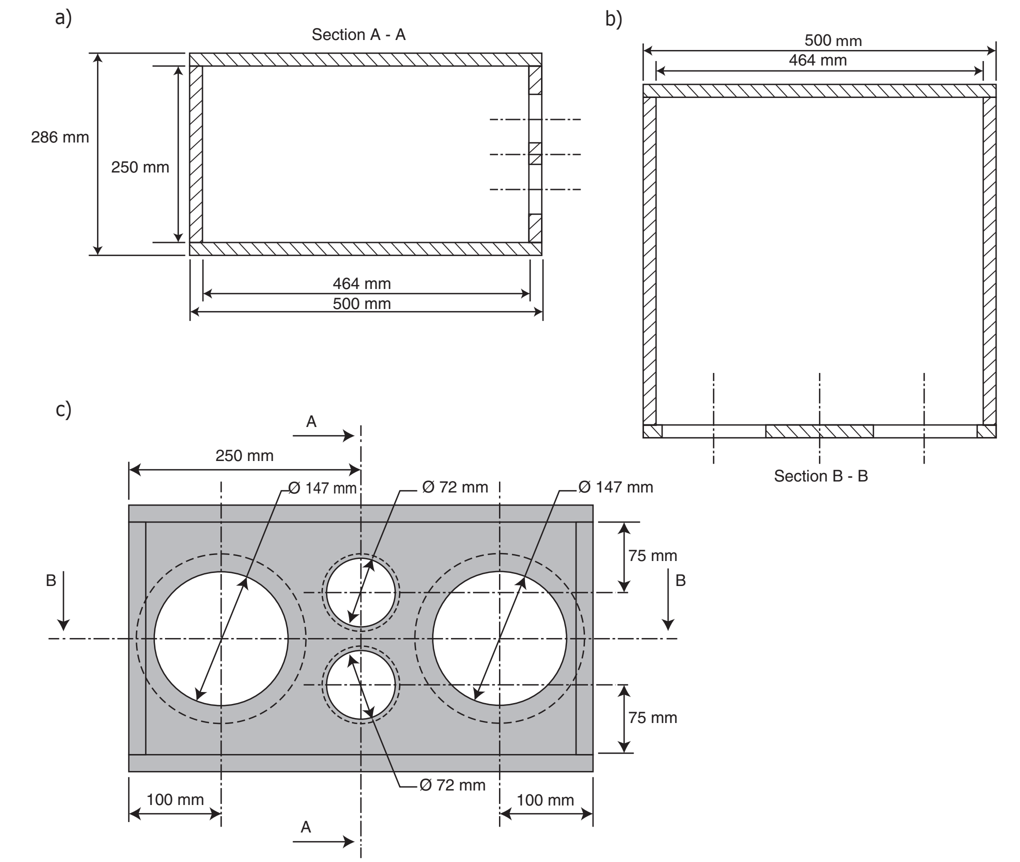

Figure 3: These design drawings for the enclosure are complete with all the measurements, assuming that 18-mm thick MDF board is used.

Start with the wire links when populating the PCB. That way, you won't forget them later. There are two of these: one near R12 and the other underneath the IC2 pins. The last link could also be soldered on the PCB's underside, but in either case you should use isolated wire. The power amplifier IC has purposely been placed at the PCB's edge, making it easier to mount the heatsink. I recommend using isolating material between the heatsink and the IC's tab.

The rest of the construction shouldn't present any difficulties, and as long as you keep to the parts list and the layout in **Figure 2**, there is no reason why the circuit shouldn't work the first time. **Photo 2** shows a correctly populated PCB, including the SK100 heatsink.

The electronics can be mounted in a separate case or inside the loudspeaker enclosure. Choose whichever suits you best. Use a standard transformer/bridge/electrolytic circuit for the power supply. A transformer rated at 12 V/50 VA, a 10-A bridge rectifier, and a 10,000-µF/25-V smoothing capacitor should be sufficient.

STURDY ENCLOSURE

The ideal enclosure for two W170SC woofers is a bass reflex enclosure with a 50-liter volume. According to the calculations, the enclosure should be tuned to just below 50 Hz. The required bass reflect port can be perfectly implemented, using two standard Conrad reflex tubes, with a 72-mm diameter and a 145-mm length. These tubes don't need to be cut to size and they can be mounted straight into the enclosure. In theory, this provides tuning at exactly 46 Hz.

Figure 3 shows the subwoofer enclosure's complete design. This is a straightforward rectangular enclosure, which even a less experienced carpenter should be able to assemble. This will certainly be the case if you can find a merchant to cut the panels to size.

Because this enclosure is substantially larger than that of the satellites and needs to support much heavier speakers, I used 18-mm thick MDF board for its construction. This also prevents unwanted resonances in the panels. In my prototype, I glued two small 250-mm support beams between the two largest panels, although this isn't a necessity.

Carefully cut the openings for the speakers and the bass reflex tubes. A special tape (e.g., speaker foam insulation) should be used to obtain an airtight seal between the speakers and the enclosure. The speakers should be screwed down tightly, either with selftapping screws or (better still) using nuts and bolts (see **Photo 3**). The bass reflex tubes should be fixed with wood glue.

Only a limited amount of acoustic damping is required in the enclosure. It is sufficient to cover all sides with a layer of polyester wadding. There is no need to completely fill it. The input connectors can be placed anywhere on the enclosure. Their position will depend on


Photo 3: The woofers must be tightly screwed down to avoid unwanted resonance.

SPECIFICATIONS (using a 17-V supply voltage)

Input impedance .... 2 × 20 kΩ
Sensitivity (20 W/4 Ω, P1 max., P2 = 0) .... 250 mV (L = R)
Distortion + noise (65 Hz, B =22 kHz, P1 max., P2 = 0) .... 0.01% (1 W/ 8 Ω)
.... 0.032% (1 W/ 4 Ω)
Bandwidth (P2 = 0) .... 18.5 Hz to 160 kHz (relative to 65 Hz)
Bandwidth (P2 = max.) .... 16.5 to 86 Hz (relative to 44 Hz)
Output power .... 2 × 20 W (4 Ω)
Quiescent current .... 0.16 A

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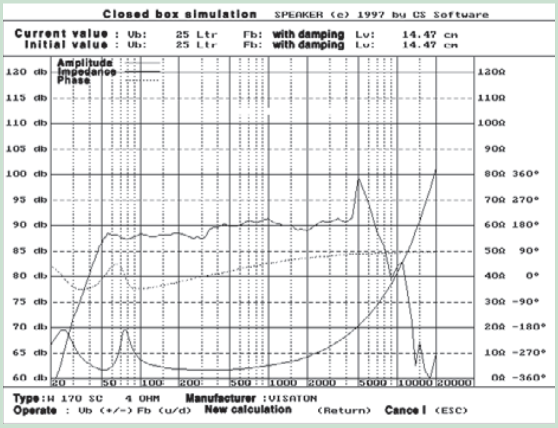

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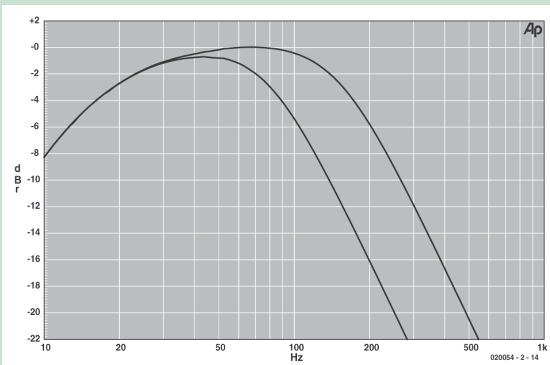

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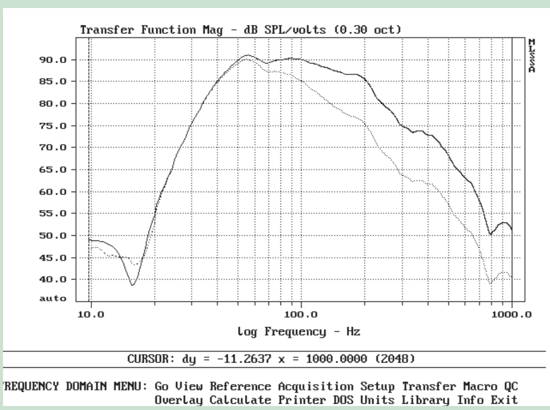

The first graph (a) shows the computer simulation of a W170SC in a 25-liter bass reflex enclosure. With the bass reflex port tuned to 46 Hz, the graph is almost flat up to 300 Hz. There are some fluctuations above this, but those frequencies are sufficiently suppressed by the active filter and will therefore not affect the response. The second graph (b) shows the frequency response of the power amplifier with P2 in its two extreme positions.
The exact cut-off frequency and the response depends on the capacitors' tolerance and potentiometer P2. When used in conjunction with the active two-way system, the cut-off frequency should be around 100 Hz. P2's range adequately covers this. The third graph (c) shows the measured frequency response of the complete system. This, too, has been measured with P2 in its extreme positions. The advantage of the smaller bandwidth is that the lower cut-off frequency, is on average, a few Hertz lower.

the enclosure's orientation. Ensure the polarity is clearly marked next to the connectors (or use loudspeaker terminals).

POSITIONING & SOUND

The positioning of a loudspeaker system consisting of a subwoofer and two satellites is fairly flexible. For optimal stereo reproduction, the loudspeakers and your usual listening position should form an equilateral triangle. This is just a standard rule for stereo reproduction. The subwoofer's best position should be centered between the satellites. However, since these low frequencies have virtually no directional information, you'll find that it doesn't matter much in practice. Anywhere in the satellites' vicinity seems to provide good results.

I received a pleasant surprise when I listened to the two-way loudspeakers in combination with the subwoofer. You don't have to take my word for it. Once you've heard the small loudspeakers in combination with the subwoofer you won't want to part with it again. Even with the volume control at a relatively low setting you will get a much richer sound. The small loudspeakers' inherently puny response disappears and the sound provides a lot more "body." The subwoofer produces a well-defined bass and has power to spare.

Whether it is worth experimenting with phase switch JP1 depends on the loudspeakers' positioning. When the subwoofer and satellites are all in one line, the sound is clearly better in phase rather than out of phase. If the woofer is nearer to or further from this line, experiment with JP1 to obtain the best results. You shouldn't expect a large difference though, since the change really is subtle.

HALF A SUBWOOFER?

A nice feature of this subwoofer design is that it can be made as a smaller "trial" version. When you want to limit the size and/or cost and are happy with a slightly less punchy bass response, you can simply use half the design. So you'll only need one W170SC, one bass reflex tube, and a 25-liter enclosure. Should you change your mind at a later date, you can always build a second identical enclosure, making the end result exactly the same as the dual version just described. *aX*

By Richard Honeycutt (United States)

Speaker Design School

Session 1A: Meet the Speakers

Providing the foundation for future DIY audio design projects

Audio enthusiasts often progress through several stages of our "aud-diction." First, we like music, and we use audio equipment to listen to it. Next, we become interested in the technical art and science behind our audio equipment. Then we may progress to building some of our own equipment, using kits or how-to articles. Finally, our passion leads us to design the equipment we then build. This article and the others in this series are designed to abet your step into this last stage. I will discuss various kinds of speakers and systems, look at some theory, and summarize the design process. There will be enough detail for some readers to successfully embark on their own design projects. Other readers will want more specifics, and will be directed to books for further study.

EARLY SPEAKER TYPES

Beginning with the Bell telephone receiver, transducers to convert AC electricity to sound have been ubiquitous. Not long after the telephone's invention, Thomas Edison patented a loud-speaking telephone, of which other versions were later introduced, to enable more than one person to hear the receiver at the same time (see **Photo 1**). Expanding beyond their use for point-to-point communication, telephone receivers were used to listen to radio programs, both in the receivers' original hand-held form and as headphones.

Loud-speaking telephones were eventually called "loudspeakers," or just "speakers," and the Kellogg-Rice cone speaker replaced them in radio use soon after its invention in 1924 (see **Photo 2**). Since then, much research has focused on the application of direct-radiator speakers and their enclosures. They have also reigned supreme in the home audio market.

In the late 1920s, the explosion of talkie cinemas opened a new commercial market for loudspeakers. These, however, needed to produce much more sound than even a loud-speaking telephone could generate. Physicists at Western Electric, RCA, and General Electric developed horn-loaded speakers with electroacoustic conversion efficiencies close to 10 times those of direct radiator speakers (see **Photo 3**). These efficiencies were needed because amplifiers of the time were limited to small output power levels. That changed with the invention of the push-pull power amplifier and negative feedback by Bell Labs's H. S. Black in the early 1930s. Horn speakers owned the cinema market for many years. They are still found in some of the best cinema loudspeaker products, and they began to find their way into the high-end home audio market in both the U.S. and Europe via the Klipsch and Lowther brands.

The invention of the electrostatic loudspeaker (ESL) in the late 1920s marked the beginning of an era in which audiophiles had their choice among speakers using relatively heavy conical paper diaphragms and those with thin metal diaphragms, or later, metalized plastic. The ESL can be very thin, and has its own set of advantages and disadvantages.

MODERN UPDATES

The Kellogg-Rice cone (direct radiator) speaker, the horn speaker, and the ESL act as "piston sources," at least at low frequencies, with radiation from all parts of the cone, horn mouth, or ESL panel pretty much in phase. **Photo 4** shows an example of a two-way direct radiator speaker and **Photo 5** shows

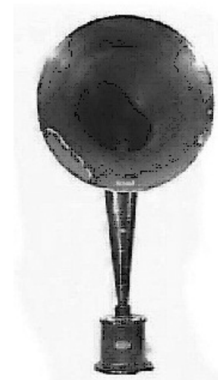

Photo 1: This loud-speaking telephone is an early vocal-range horn speaker. (Photo courtesy of www.porticus.org/bell/images)

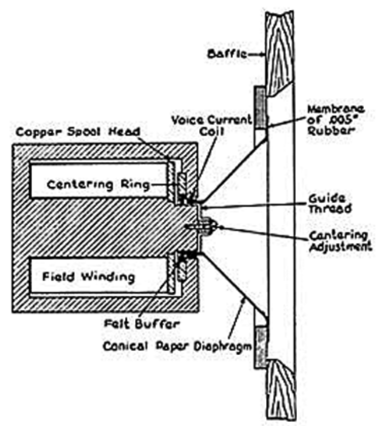

Photo 2: This patent drawing details the design of the Kellogg-Rice cone speakers. (Photo courtesy of http://tecfoundation.com/hof/07techof.html#kellogg)

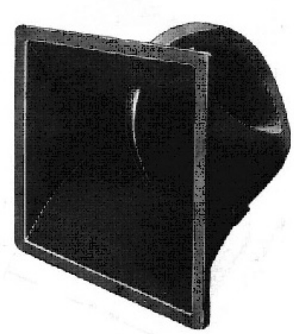

Photo 3: A Western-Electric "Ram's horn" full-range horn increased loudspeaker efficiency.

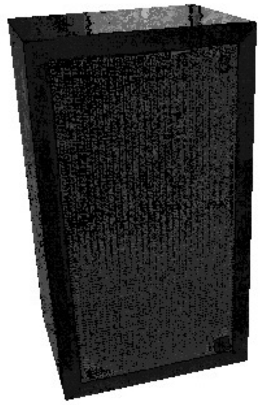

Photo 4: This is an Acoustic Research AR-1A two-way direct radiator speaker.

a three-box cluster, that contains a direct-radiator and horn speakers. In the mid-20th century, several companies experimented with using flat panels as diaphragms for direct radiator speakers. The Poly-Planar speaker, which used a thin, square polystyrene diaphragm, was one example. These were easily mounted on or in walls because of their thin profiles, but they were not noted for their fidelity. In the 1990s, New Transducers, of England (now NXT) introduced a new concept—flat-panel speakers specifically designed to radiate according to the natural vibrational panel modes. Since these modes produced non-correlated radiation areas, these speakers were dubbed "distributed-mode loudspeakers" (DMLs). These speakers produce an almost perfect hemispherical radiation pattern at all frequencies within their passband, and since they are carefully designed to use (rather than vainly attempt to eliminate) the panels' natural modal behavior, they can have quite flat frequency responses.

About the same time DMLs were commercialized, an old 1930s concept called "line array" was rediscovered. It is ideally a long, narrow sound source that radiates—within a certain range of listener distances—in a cylindrical pattern typically 120° to 150° wide horizontally, and vertically with a height equal to the array's length. This vertical radiation pattern is well defined only for frequencies above that at which the array is about two wavelengths long. At lower frequencies, a line array becomes omnidirectional (see **Photo 6**).

Today, a loudspeaker designer has many options from which to choose when planning a speaker project. Let's use a broad-brush approach to compare the various technologies available. For the sake of convenience, I'll take these technologies in the order of their invention, skipping over the loud-speaking telephone as it is only of historical interest.

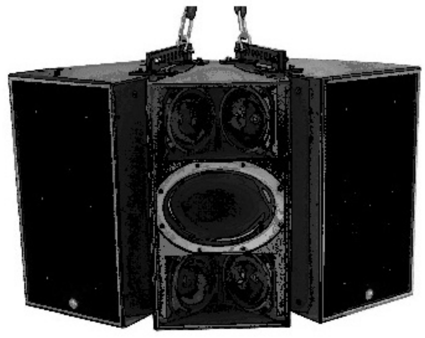

Photo 5: The Renkus-Heinz ST4-3(T) is a three-box cluster that contains a direct-radiator and horn speakers. (Photo courtesy of Renkus-Heinz)

CONE SPEAKERS

The cone speaker can provide flat frequency response and low distortion. Like most speakers, it radiates omnidirectionally at low frequencies, developing a conical radiation pattern at frequencies with wavelengths less than the circumference of the speaker. If a cone speaker acted as a true rigid piston, its directionality would continue to increase with frequency, but actually, the cone "breaks up" into modal patterns at higher frequencies, eventually halting the increase in directionality with increasing

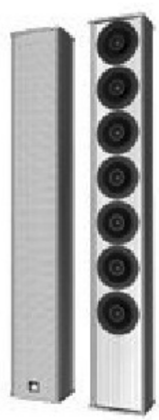

Photo 6: A Renkus-Heinz IC-7 line array provides a long, narrow sound source. (Photo courtesy of Renkus-Heinz)

frequency. This modal radiation also causes irregularities in frequency response. **Figures 1–3** show different cone breakup modes.

Cone breakup involves two sets of resonant modes. Those at the lower frequencies occur when the cone's circumference is an integer multiple of the sound's wavelength in the cone. These are called "radial modes" or "bell modes." When the cone is vibrating in the lowest of these modes, a 90° pie-shaped segment of the cone may be seen to move toward the front while the neighboring segments are moving toward the back. (Certainly this is an oversimplification. The actual cone shape when executing this mode is that of two cycles of a sine wave wrapped around the circumference.) The radial modes' effect upon low-frequency driver response is generally negligible, except that they may interact with other modes in undesirable ways. **Figure 1** illustrates radial modes.

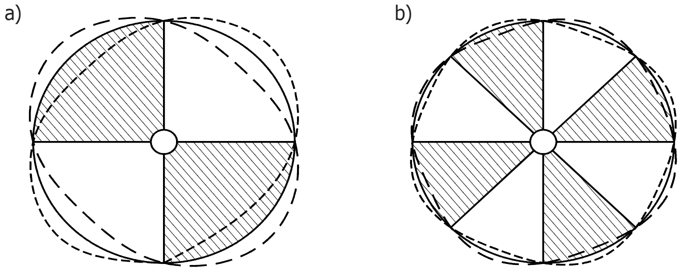

Figure 1: The first radial mode (a) and second radial mode (b) are shown here.

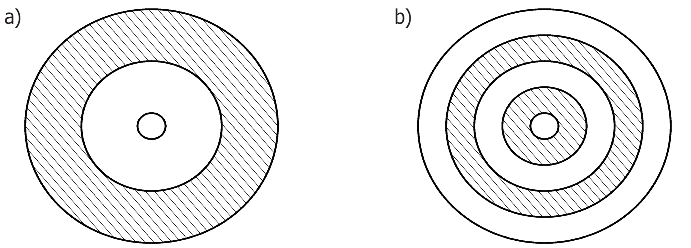

Figure 2: First concentric mode (a) and second concentric mode (b) occur when the cone's radius is an integer number of half wavelengths.

In addition, at higher frequencies, "concentric" modes occur when the cone's radius is an integer number of half-wavelengths. The action of these modes may be compared to a transmission line in which the initial wave travels from the voice coil joint toward the cone's periphery. Then, it is incompletely absorbed due to an impedance mismatch where the cone is terminated at the surround, and is reflected back toward the voice coil. Where the incident and reflected waves cross, there will be constructive or destructive interference, and where the dimensions of the path are appropriate, standing waves will be set up. Concentric modes can cause varying degrees of raggedness in the frequency response, depending on the cone material's internal damping and the surround's ability (via proper damping and mechanical impedance) to absorb the incident waves. **Figure 2** shows two concentric modes.

When both radial and concentric modes occur, the effect upon frequency response and polar response (directionality versus frequency) can be great, and its exact nature depends not only on the frequency and severity of the modes, but upon which modes were initiated first. This is one reason their mathematical analysis is actually impossible (see **Figure 3**).

Figure 4 shows the frequency response of a 20-cm speaker with a cone without a curved flare or corrugations. Notice the uniform response up to about 800 Hz, then the ragged response at higher frequencies, caused by cone breakup (modal behavior). The cone flare, presence or absence of corrugations in the cone, and the choice of cone material affect the degree of raggedness in the frequency response.

In general, cone speakers are inefficient. Typically, a cone speaker requires from 25 W to more than 100 W of electrical power to produce one acoustical watt of sound. In a future article, I will

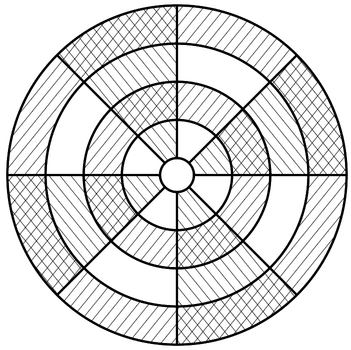

Figure 3: Occurrence of both radial and concentric modes can greatly affect frequency and polar response.

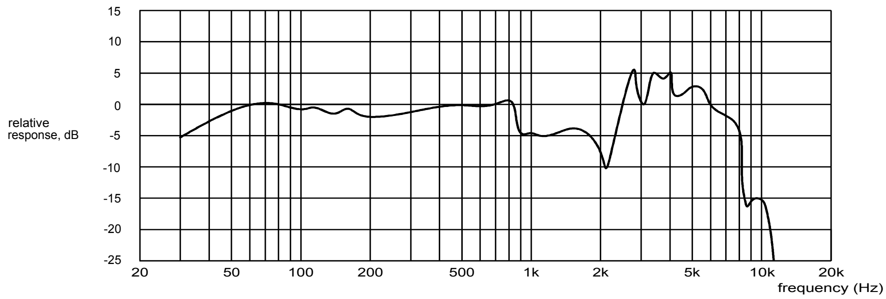

Figure 4: This graph shows the frequency response of a 20-cm speaker with a straight-sided cone.

examine the limiting factors on a cone speaker's efficiency.

Cone speakers radiate from both sides of the cone. The radiation from the rear is exactly out of phase with the radiation from the front. At low frequencies, where the radiation is omnidirectional, the front and rear radiation cancel each other except very close to the speaker, so cone speakers are usually enclosed in cabinets to keep the rear wave from interacting with the front wave. These cabinets can be simple closed boxes with or without absorptive linings, or they can be more sophisticated designs such as vented cabinets (also called "ported," "bass reflex," etc.) A given driver (the speaker without the box) will work optimally for either a sealed or a vented design, but usually not both. Speakers in sealed boxes provide a gradual low-frequency 6 dB/octave cutoff. (If the system's low-frequency limit is 40 Hz, it will put out 6 dB less sound with the same electrical power at 20 Hz, which is an octave below 40 Hz.) As the frequency is lowered, the cone excursion (motion) required to produce a given output level increases. For example, a 30-cm speaker in a 0.011-m^3 cabinet requires about 2-cm peak-to-peak cone excursion to produce one acoustical watt output at 50 Hz, and the required excursion increases as the inverse square of frequency. Smaller cones require greater excursion in inverse proportion to cone area. (One acoustical watt corresponds to a sound pressure level of 112 dB at 1 m from the speaker in anechoic conditions.)

A vented cabinet uses the speaker's back wave to drive an opening called a "vent" or "port" at frequencies close to the system's low-frequency limit. The back wave is out of phase with the front wave, but the vent is a driven mechanical oscillator. Driven oscillators vibrate exactly out of phase with the driving force. Thus, the vent output is in phase with the front wave. This action enables a vented cabinet to be designed for reduced cone excursion, higher efficiency, lower minimum output frequency for the same size box, or a limited combination of the above. Vented enclosures' variations are referred to as "alignments," and they are named according to the standard electrical filters to which their performance corresponds.

Vented cabinets' sound outputs decrease at 12 dB/octave below the minimum frequency, so they do not produce as much sound below their so-called minimum frequency as sealed boxes. But, depending on the design, that minimum frequency may be lower than for a sealed box of similar size.

In the next article, I'll continue this discussion by examining other speaker types. *aX* recording review By Gary Galo (United States)

The Legendary Rachmaninoff *Symphonic Dances*

Differing audio nuances discerned through a single recording

Sergei Rachmaninoff: *Symphonic Dances*, Op. 45 and Vocalise, Op. 34 Dallas Symphony Orchestra
Conductor: Donald Johanos Recorded: Feb. 20, 1967 McFarlin Auditorium, Southern Methodist University, Dallas, TX Recording Director and Musical Supervisor: Thomas Mowrey Recording Engineer: David B. Hancock 45-rpm, 2-disc vinyl release: Analogue Productions APC 34145S SACD Release: Analogue Productions CAPC 34145SA
www.analogueproductions.com

Donald Johanos's 1967 recording of Rachmaninoff's *Symphonic Dances*, Op. 45 for Vox/Turnabout has retained something of a cult status among audiophile recordings (see **Photo 1**). *Symphonic Dances*, Op. 45 was written in 1940 while Rachmaninoff was living in Huntington, Long Island, NY. Strangely, this recording's original program notes, written by the late Dallas Symphony Orchestra violinist Dorothea Kelley and reprinted in all the reissues discussed in this article, state it was "the first composition he wrote after moving to the United States, and the last that he was ever to write."[1] While the last statement is true, the first is not.

Rachmaninoff settled in the United States in November 1918. Predating *Symphonic Dances* are his *Symphony No. 3* (1935–36, rev. 1938), *Rhapsody on a Theme of Paganini* (1934), *Variations on a Theme of Corelli* (1931), and *the Fourth Piano Concerto* (1926, rev. 1941). Though hardly his best or most popular work, *Symphonic Dances* has earned a place in the orchestral repertoire. Currently, ArkivMusic lists more than 70 recordings of the work on its website (www.arkivmusic.com).

Photo 1: This is the original LP cover for Rachmaninoff's *Symphonic Dances*, recorded by engineer David B. Hancock, and performed by the Dallas Symphony Orchestra conducted by Donald Johanos. Although never intended as an "audiophile" recording, the demonstration-quality sonics captured by Hancock have earned this recording cult status in high-end audio circles. It has received several audiophile reissues over the past 15 years.

THE CONDUCTOR

The late conductor Donald Johanos, who died in 2007 at the age of 79, led the Dallas Symphony Orchestra from 1962 until 1970. American born, in Cedar Rapids, IA, Johanos studied with Eugene Ormandy, the Philadelphia Orchestra's long-time conductor who gave the world premiere of *Symphonic Dances* in 1941. Johanos was a champion of 20th-century music, and his small recorded output includes works by Charles Ives, Aaron Copland, Gunther Schuller, and Dan Welcher. Although his Rachmaninoff performance is not without some exciting moments, and features commendable playing by Dallas' then-provincial ensemble, Johanos generally offers a solid, middle-road reading.

For a more vibrant and idiomatic—indeed, definitive—reading, I recommend Kyril Kondrashin and the Moscow Philharmonic Orchestra, recorded by the Soviet Melodiya label in 1963. But, don't expect any audiophile miracles. Soviet recordings were notoriously inferior to those produced in the West (RCA Victor/BMG Gold Seal CD 32045-2, coupled with an equally outstanding performance of Rachmaninoff's *The Bells*).

So, why all the fuss? Because, the Vox/Turnabout *Symphonic Dances* remains one of the most spectacular orchestral recordings ever made! The engineer/producer team of Marc Aubort and Joanna Nikrenz came to dominate the classical recordings from Vox/Turnabout, but a handful of Dallas Symphony recordings under Johanos were produced by Thomas Mowrey and recorded by New York-based engineer David B. Hancock (1927–2001). I had a personal connection with David. His son was a student at SUNY Potsdam in the early 1980s, and he took an audio technology class I taught. I met David through his son and learned of his dislike of condenser microphones, at a time when seemingly everyone was using them (usually Neumann, AKG, or Schoeps). David preferred ribbon microphones, especially those made by Charles Fisher in Cambridge, MA.

FISHER'S RIBBON MICS

Fisher described his ribbon microphone in an article published in *Audio* magazine in 1965. He had modified a pair of RCA 44BX ribbon microphones, replacing the two required transformers with a single, higher-quality transformer and a solid-state pre-amplifier. Concurrently, Hancock had been using a pair of RCA 44A ribbons that he had modified himself. The microphones used for the *Symphonic Dances* recording was a combination of ideas from both men. The story is much longer, of course, and the article is interesting reading. Fisher applied for a patent on August 2, 1965 and in 1969, U.S. Patent 3,435,143 was assigned to his design.

THE RECORDING ENGINEER

David Hancock was a trained musician—a pianist with a special love of chamber music—who graduated from The Juilliard School and whose fine playing can be heard on a number of recordings. (His wife Eleanor was also a pianist and a respected teacher.)

An excellent example of David Hancock as a pianist can be found on Monitor LP MC 2017, which contains violin sonatas by Franck and Debussy, with violinist David Nadien. Hancock also served as the recording engineer and he cut the lacquer master. A CD recording has been reissued by Cembal d'amour, CD 151, and it is available from Amazon.

David Hancock's musical background profoundly influence his judgment when it came to recorded sound, and led him in some unusual directions when it came to his equipment choices. In addition to those superb ribbon microphones, Hancock used an Ampex 350-2 tape transport that he custom-modified for 30-ips operation. The stock Ampex recorder had a serious limitation in that it would only accommodate 10.5" reels which, when running at 30-ips, were limited to a little over 15 min. of music on a 2,500′ reel. Hancock removed the Ampex hardware from the factory-supplied stainless steel top plate and remounted everything on a large, reinforced plywood top plate, with the take-up and supply reels positioned so the recorder would accommodate 14" reels, enabling recording times of more than 30 min. per reel (the same as a 15-ips recorder with 10 0.05" reels). Hancock's recorder modifications ensured exceptionally smooth motion and speed stability. Victor Campos once told me that Hancock's custom-modified recorder was known to many who knew him as "The Plywood Ampex."

Instead of the stock Ampex 350-2 electronics, Hancock used a pair of modified Ampex PR-10-1 mono electronics units. He preferred these because they had a simpler audio signal path with fewer transformers. (The PR-10 was a portable recorder with excellent electronics, but a problematic transport. Ampex later used the electronics in its 354 recorder.) A common, external-bias oscillator drove the modified PR-10-1 electronics and provided a constant-velocity recording curve developed by Hancock. I am grateful to Lonn Henrichsen for providing details on the modified Ampex recorder. Lonn met with David Hancock back in the 1960s, saw the recorder for himself, and took notes during their conversation.

Hancock recorded Rachmaninoff's *Symphonic Dances* on February 20, 1967 in McFarlin Auditorium on the Southern Methodist University's campus, which was the Dallas Symphony Orchestra's regular home. According to the original notes, Hancock used four Fisher ribbon microphones for this recording, but the exact arrangement was not specified in the jacket notes. As producer, Mowrey described their placement in a note posted on the Gearslutz.com website in

2008. (Thanks again to Lonn for bringing this posting to my attention.)

"There were two pairs, one for the whole orchestra, suspended about 15' above stage level, approximately 10' back into the (dead dry) auditorium from the conductor, Donald Johanos. The other pair was suspended about 15' above the woodwinds. The mics were exactly six feet apart in both pairings. The orchestra pair were angled outwards at about 45°, and the woodwind pair were in parallel orientation. For a second series of recordings with the same group (Ives and Copland), Dave Hancock and I added a third pair of these mics, which were placed out in the auditorium, quite widely spaced, in a (futile) attempt to add a bit of bloom to the sound."[2] The spaced arrangement is unusual for microphones with a figure-eight directional pattern—normally they're used in a coincident arrangement named after Alan Blumlein, the engineer who developed it.

THE RECORDINGS

In the 1960s and 1970s, the quality of Vox/Turnabout's disc cutting and pressings was usually what you'd expect for a budget label. But for his own recordings, Hancock cut his lacquer masters himself, from the original 30-ips tapes. (He didn't believe in "production copies.") It's his original cutting that became such a highly-prized disc among audiophiles. Hancock's LP masterings are easy to spot. He always etched his initials, DBH, in the run-out space, enclosed in a rectangular box. On both sides of the Rachmaninoff, to the right of the box, is the designation "ORIG." I own two copies of this LP's first cutting. One is pressed on heavy vinyl and the other is thinner. The thinner record is actually the better sounding of the two. The heavy pressing has a fair amount of steady-state groove noise and it is a bit less detailed than the thinner copy. Hancock's LP looks like an audiophile record. Visually, it's obvious the record has very wide dynamic range and high peak-cutting levels.

If you're lucky enough to own a copy of the original cutting and you have a really good LP playback system, it's easy to understand why this record is still considered one of the most impressive orchestral recordings ever made. Most striking, at least to those who hear live orchestral music on a regular basis, are the instruments' palpably realistic timbres. Both the winds and the strings have a body and warmth that is unbelievably lifelike. Hancock did not ride gain on the original recording or in the cutting of the master lacquer, resulting in spectacular dynamics. The heavily scored passages on this recording are about as close as you'll get to the impact of a live orchestra. The soundstaging is among the best ever captured on an orchestral recording, and the level of clarity and detail are remarkable.

The "dead dry" hall noted by Mowrey is clearly evident on this recording, resulting in a close perspective on the orchestra. It's more like hearing an orchestra in a large recording studio than in a good concert hall. To my ears, the perspective and level of hall ambience is not unlike Sheffield's last direct-to-disc recording with Erich Leinsdorf and the Los Angeles Philharmonic performing Stravinsky's *Firebird Suite* and Debussy's *Prélude à l'après-midi d'un faune* (Lab 24). That recording was made in 1985, in the same MGM soundstage studio used for their Wagner and Prokofiev discs eight years earlier. But this time, Sheffield dispensed with its condenser figure-eight stereo microphone in favor of ribbons. The brightness of the AKG condenser capsules gave a greater sense of ambience than the ribbons; the ribbons laid bare the dryness of the MGM studio. Hancock's recording sounds like it was made in a larger space than the Sheffield, but there are striking similarities in the sonic characteristics of the two recordings, including how the ribbon microphones captured the timbres of the instruments.

Aside from the dry recording venue, if the Rachmaninoff recording has any additional weakness, it's in the reproduction of the brass. Ribbon microphones can overload more easily than condensers on loud brass instruments. On this recording, the loud trumpets, particularly in the last movement, have a slightly buzzy quality. The original cutting is also a bit softer in the treble than subsequent audiophile reissues.

The dynamics of Hancock's original cutting may have provoked complaints from customers with cheap LP playback equipment. I've seen later Vox/Turnabout pressings that were re-cut by someone else and, although I never actually played one, visually it's obvious that the cutting levels were lower and the dynamics narrower. Vox/Turnabout reverted to the "lowest common denominator" attitude that was permeating the American recording industry by the 1970s. If you find a used copy, look for "DHB" in the runout space. If it's not there, don't buy it.

DBX II-ENCODED LP

In the 1970s, a few American labels had brief flirtations with dbx II-encoded LPs. dbx II was an encode/decode, companding-type noise reduction system designed for slower-speed reel-to-reel tape recording. (dbx I was the professional version.) In the 1970s, someone decided surface noise on records would disappear if LPs were dbx-encoded. Vox/Turnabout issued a dbx-encoded version of *Symphonic Dances*, which can easily be identified since the cover artwork is completely different from the original, and it sports a large, round dbx sticker (TV 34145S). The non-dbx version during this period also had the new cover, but no sticker. Needless to say, David Hancock

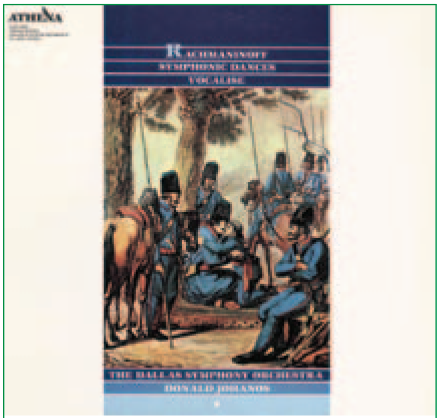

Photo 2: Engineer Doug Sax remastered the Athena audiophile LP edition of *Symphonic Dances* from the original 30-ips tapes, and the excellent pressings were made in Japan. The cover illustration was that used on Vox/ Turnabout's dbx-encoded LP from the 1970s.

had nothing to do with the cutting of the dbx version of this recording. This edition need not detain us any longer.

THE AUDIOPHILE REISSUES

The reputation of Hancock's original recording was only enhanced by the interest in audiophile recordings that began in the mid-1970s and continued throughout the 1980s. The problem was that his original cutting was long out of print and used copies were difficult to find. So, in 1988 Doug Sax, of Sheffield Lab fame, cut a super-disc LP from Hancock's original 30-ips tapes, marketed on a label called Athena (Analogue Productions, ALSW-10001) shown in **Photo 2**. The sound on this edition is spectacular, superior in every regard to Hancock's original cutting, with better dynamics, soundstaging, clarity and treble extension. The record was pressed in Japan, with surfaces like velvet. Used copies of this LP fetch premium prices on eBay on the rare occasions when they're offered for sale. For about 10 years, the Athena LP was the preferred edition of *Symphonic Dances*.

In 1991, Sax digitally remastered the original tapes for a CD issued by Analogue Productions, a division of Acoustic Sounds, which is the audiophile record dealer based in Salina, KS (Analogue Productions, APCD 006). The remastering was done with tube electronics, and the CD sounds very good. But, compared to the Athena LP, I find the CD a bit soft in the treble and lacking in overall impact. Although the CD has plenty of tube warmth, the transients are a bit mushy compared to the crisp, detailed Athena LP. Interestingly, both the Athena LP and the Analogue Productions CD used the cover illustration from Vox/ Turnabout's dbx-encoded LP and not the original.

In the 1990s, higher-resolution digital mastering became the norm and, with the introduction of the DVD in 1995, 24-bit recordings at sampling rates of up to 96 kHz became available. Classic Records (now owned by Acoustic Sounds) soon took advantage of the new format, offering 96-kHz/24-bit remasterings of analog recordings on DVD. Classic Records called the new format "digital audio disc" (DAD). Remember, the DVD-audio format wasn't introduced until 2000. Classic Records DADs used a regular DVD's audio portion, and the discs would play in any DVD player.

In 1998, Classic Records did the first high-resolution digital release of *Symphonic Dances*, offering a 96-Hz/24-bit remastering on an Analogue Productions DAD 1004. It was around that time I used a Parts Connection DAC-3.0, which was my first 96-kHz/24-bit digital processor. For my first DVD player, I purchased a Pioneer DV-525, one of only a handful of DVD players that could deliver a 96-kHz/24-bit datastream to the S/PDIF output without down-sampling to 48 kHz. With this set up, the Classic Records remastering sounded like a breakthrough in digital audio technology. For the first time, I felt as though I was hearing exactly what was on Hancock's original tapes, free of the euphonic colorations (impressive and pleasing though they may be) of LP cutting, tube re-mastering, and so forth, and free of the Red Book CD's limitations. The Classic Records DAD has held up extremely well and it sounds more impressive than ever on my Oppo BDP-95 Universal Blu-ray Digital Player (reviewed in *audioXpress*, January 2012). If this had been the last remastering of *Symphonic Dances*, I'd have been quite content.

ANALOGUE PRODUCTIONS EDITIONS

The Classic Records DAD sold out fairly quickly, so *Symphonic Dances* remained out of print for about 10 years. Now, thanks to Analogue Productions, it's back in print and sounding better than ever on SACD and on a special two-disc, 45-rpm edition pressed in Germany on 180-gram virgin vinyl by Pallas (Analogue Productions, CAPC 34145SA and APC 34145S). Kevin Gray remastered both from Hancock's original 30-ips tapes (see **Photo 3**). The vinyl edition was cut while Gray was working at AcousTech Mastering, and the discs sport reproductions of the original Vox/Turnabout labels (see **Photo 4**). The SACD was remastered at his new firm, CoHEARent Audio (www.cohearent.com). AcousTech Mastering ended operations in 2010. The SACD is a dual-layer, hybrid disc, compatible with all conventional CD players, and was authored by Gus Skinas on the Sonoma system at the Super Audio Center in Boulder, CO (www.superaudiocenter.com).

Photo 3: David Hancock's original 30-ips tapes—two 14" reels—were used for Analogue Productions's latest remasterings of *Symphonic Dances*. "Constant velocity" refers to the equalization curve used on his custom-modified "Plywood Ampex" tape recorder. Ampex 456 tape didn't exist in 1967, so Hancock must have spun the first tape onto a new reel and stored both in new boxes, at some point. (Photo included with the SACD edition, reproduced courtesy of Analogue Productions)

The 45-rpm vinyl edition improves upon the old Athena LP in several respects, most notably in dynamics and impact. The 45-rpm cutting enables an entire 12" side for each movement, the longest being just a little more than 13 min. (Side 4 contains the original filler, Rachmaninoff's orchestral transcription of his Vocalise.) The new 45-rpm discs also have a slightly larger soundstage and a bit more treble detail than the Athena LPs. I did find that the Analogue Productions LPs were not quite as dead silent at the Athena LPs. The first disc (Sides 1 and 2) in my original copy suffered from a few more ticks than I felt was reasonable for an audiophile pressing. Acoustic Sounds sent a replacement, which was much better.

If Classic Records 96/24 DVD edition provided a window on the original analog tapes, the SACD remastering has wiped that window clean. The difference in the treble region is readily audible on my system. Strings on the SACD sound silkier. By comparison, the 96/24 DVD has a bit of graininess. The shimmering quality in the cymbal crashes is more lifelike, with the DVD taking on just a bit of edge. There's also slightly more detail in the treble. By itself, the DVD still sounds spectacular, but the SACD is just that much better. The conversion to 44.1-kHz/16-bit PCM for the CD layer is extremely well done, but the differences between the SACD and the CD layers are definitely audible. (I compared the two by simply switching layers on my Oppo BDP-95.) The CD's treble region has some graininess not present on the SACD, especially noticeable on the violins. The SACD has a larger soundstage, better clarity and articulation in the treble, and an airiness not quite there on the CD. But, the CD layer is still excellent and far better than the old Analogue Productions CD. Even if you don't have an SACD player, the disc is worth owning, and an example of how good a Red Book CD can be, if executed with the best current digital technology.

Comparing the 45-rpm and SACD editions, it's obvious that Gray's goal was to transfer, as faithfully as possible, the original tapes' sound to the medium in question, letting each technology speak for itself. The vinyl records have a bit more warmth, and some additional heft in the low end. But, they're not excessively colored. As to which is a more accurate representation of Hancock's original tape, I'd put my money on the SACD. But, the fact that they're so similar in sound shows just how far digital audio has come over the past 25 years. There's a lot of truth to the belief that the better digital audio gets, the more it sounds like analog. On the Super Audio Center's home page, it notes the Sonoma DSD recorder "sounds like an analog recorder without the flaws."[3] Indeed.

Photo 4: Analogue Productions reproduced the original Vox/Turnabout label for its deluxe 45-rpm vinyl edition of the *Symphonic Dances.*

VINYL OR DIGITAL?

I still love playing vinyl and, despite my enthusiasm for the SACD format, I wouldn't be without either edition of this classic recording. Both can be purchased from Acoustic Sounds (www.acousticsounds.com), Elusive Disc (www.elusivedisc.com), and MusicDirect (www.musicdirect.com). Like previous vinyl and digital editions of this recording, these are likely to sell out. As Crazy Eddie used to say, "Get it now!" *aX*

REFERENCES

[1] D. Kelley, "*Symphonic Dances*, Op. 45," biographical and program notes, 1967.

[2] Gearslutz.com, "Remote Possibilites in Acoustic Music & Location Recording," www.gearslutz.com/board/remotepossibilities-acoustic-music-location-recording/291343-decca-tree-w-m50s.html.

[3] Super Audio Center, LLC, www.superaudiocenter.com.

RESOURCES

Analogue Productions, www.analogueproductions.com.

C. P. Fisher, "A 'New' Ribbon Microphone," *Audio*, August 1965.

———, "Ribbon Microphone," U.S. Patent 3,435,143, 1969, www.google.com/patents/US3435143.pdf.

Gearslutz.com, "Remote Possibilites in Acoustic Music & Location Recording," www.gearslutz.com/board/remotepossibilities-acoustic-music-location-recording/291343-decca-tree-w-m50s.html.

By Vance Dickason (United States)

A New Mid-Bass Transducer

A detailed look at Morel's TiCW 634Nd

The TiCW 634Nd is an interesting new release from Morel, the Israel-based, high-end OEM driver, and branded home and car audio manufacturer. The mid-bass transducer comes from Morel's new Titanium Series that I wrote about in *Voice Coil* (September 2012), a sister magazine to *audioXpress*. This new series of 6" mid-bass woofers is the next generation of Morel's hybrid neodymium/ferrite motors with titanium voice coil formers. There are six models in the Titanium series, and this month I tested the new TiCW 634Nd.

The TiCW 634Nd has a rich design, most notably with Morel's proprietary cast aluminum frame configured with the motor mounted from the front and bolted to the frame's back (see **Photo 1**). This six-spoke frame provides minimum surface area for reflections into the rear of the cone and a totally open area below the spider mounting shelf for enhanced cooling. Other features include an injected damped polymer cone and dust cap,

Photo 1: The Morel TiCW 634Nd top view (a) and bottom view (b)

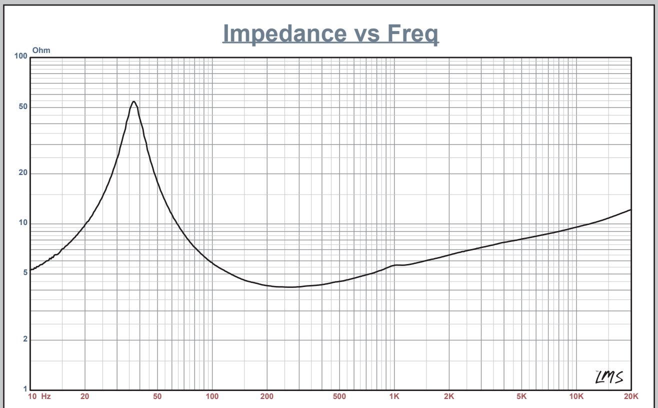

Figure 1: Morel TiCW 634Nd woofer free-air impedance plot

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
F_S	38.6 Hz	37 Hz	35.6 Hz	35.3 Hz	40 Hz
R_{EVC}	3.56	3.53	3.56	3.53	3.7
Sd	0.0117	0.0117	0.0117	0.0117	0.0119
Q_{MS}	4.62	4.69	4.12	4.64	4.98
Q_{ES}	0.34	0.33	0.36	0.33	0.37
Q_{TS}	0.32	0.31	0.33	0.31	0.35
V_{AS}	18.6 ltr	20.3 ltr	22 ltr	22.5 ltr	25.5 ltr
SPL 2.83 V	86.9 dB	86.8 dB	86.3 dB	86.5 dB	88 dB
X_{MAX}	5.5 mm	5.5 mm	5.5 mm	5.5 mm	5.5 mm

Table 1: Morel TiCW 634Nd woofer

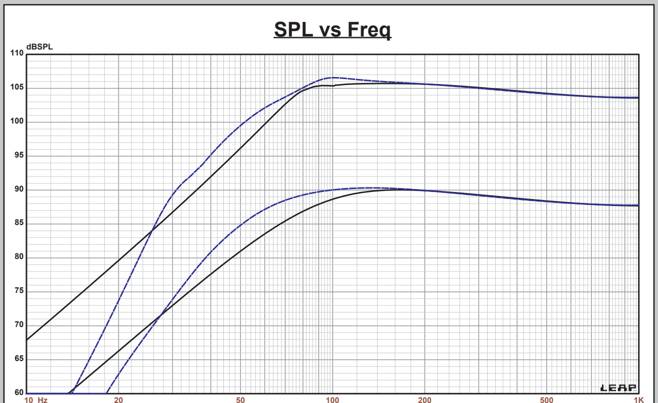

Figure 2: Morel TiCW 634Nd computer box simulations (black solid = sealed 1 at 2.83 V; blue dash = vented 2 at 2.83 V; black solid = sealed 1 at 14 V; blue dash = vented at 16 V).

NBR surround, 3"-diameter vented titanium voice coil former wound with two layers of Morel's Hexatech aluminum wire, a hybrid neodymium and ferrite motor with a copper shorting ring (sleeve), a flat cloth spider, and gold plated terminals. According to Morel, "the use of a low-induction coefficient titanium bobbin immensely improves the electroacoustic parameters of the driver. Furthermore, the Titanium-rigid characteristic affects the sound tonality, yielding a crisp-sounding driver."

I initiated the Morel TiCW 634Nd's testing using the LinearX LMS analyzer and VIBox to create both voltage and admittance (current) curves with the driver clamped

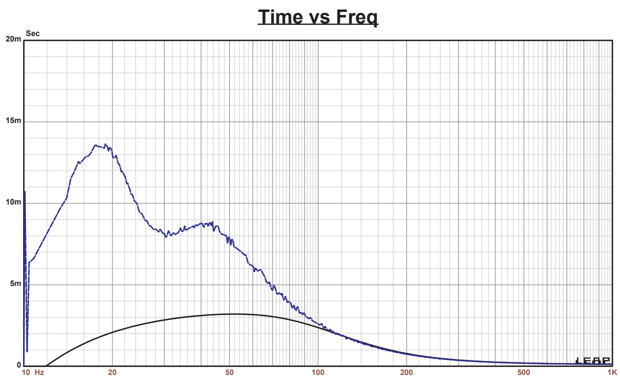

Figure 3: Group delay curves for the 2.83-V curves in Figure 2

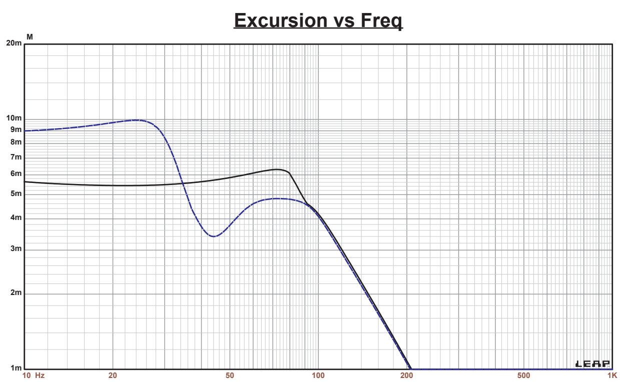

Figure 4: Cone excursion curves for the 18.25-V curves in Figure 2

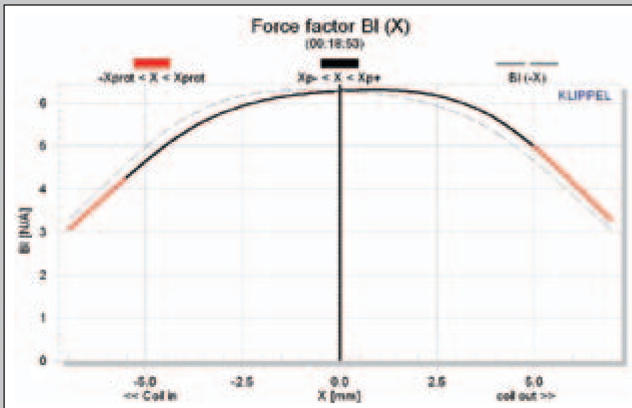

Figure 5: Klippel analyzer Bl (X) curve for the Morel TiCW 634Nd

to a rigid test fixture in free air at 0.3 V, 1 V, 3 V, 6 V, and 10 V. I discarded the 10-V voltage and current curves because they were too nonlinear to get a solid curve fit in LEAP 5. The eight 550-point stepped sine wave sweeps for each TiCW sample were post-processed and the voltage curves were divided by the current curves (admittance) to create impedance curves. Phase was added using LMS calculation method and, along with the accompanying voltage curves, uploaded to the LEAP 5 Enclosure Shop software. Besides the LEAP 5 LTD model results, I additionally created a LEAP 4 TSL model set of parameters using just the 1-V free-air curves. The final data, which includes the multiple-voltage impedance curves for the LTD model (see **Figure 1** for the 1-V free-air impedance curve) and the 1-V impedance curve for the TSL model, were selected and the parameters were created to perform the computer box simulations. **Table 1** compares the LEAP 5 LTD and TSL data and factory parameters for both Morel 6" samples.

LEAP parameter calculation results for the TiCW 634Nd were reasonably close to the factory data. Given this, I used the LEAP LTD parameters for Sample 1 to set up two computer enclosure simulations, one sealed and one vented. For the Butterworth closed-box simulation, I used a 330-ci enclosure with 50% fiberglass fill material. For the vented box, I used a 504-ci QB3-type vented alignment with 15% fiberglass fill material and tuned to 42.3 Hz.

Figure 2 shows the results for the Morel Titanium Series woofer in the sealed and vented boxes at 2.83 V and at a voltage level sufficiently high enough to increase cone excursion to X_{MAX} + 15% (6.3 mm for the 634Nd). This yielded a F3 = 81 Hz with a box/driver Qtc of 0.68 (perfect for a home theater LCR) for the 330-ci sealed enclosure and −3 dB = 60 Hz for the 504-ci vented simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was reached resulted in 105.7 dB at 18.25 V for the sealed enclosure simulation and 106.5 dB with the same 18.25-V input level for the larger vented box. (See **Figure 3** and **Figure 4** for the 2.83-V group delay curves and the 18.25-V excursion curves.) Note that the voltage was limited on the vented simulation as it was exceeding the X_{MAX} + 15% number at 35 Hz. With an appropriate high-pass filter, the vented simulation would handle more voltage and produce a commensurately higher SPL.

Klippel analysis for the Morel 6" woofer produced the Bl(X), K_{MS}(X) and Bl and K_{MS} symmetry range plots shown in **Figures 5** to **8**. The Bl(X) curve for the TiCW 634Nd is nicely broad and mostly symmetrical curve (see **Figure 5**). Looking at the Bl symmetry plot, this curve shows a small coil forward (coil out) offset at the rest position of 0.88 mm that goes to a trivial 0.3-mm offset at the 5.5-mm physical X_{MAX} of the transducer (see **Figure 6**).

Figure 7 and **Figure 8** show the K_{MS} (X) and K_{MS} symmetry range curves for the Morel 6". The K_{MS} (X) curve is moderately symmetrical in both directions with a small amount of coil-in offset at the rest position. Looking at the K_{MS} symmetry in **Figure 8**, the coil-in offset is 1.29 mm at the rest position, decreasing to 0.7 mm at the 5.5-mm physical X_{MAX} of the TiCW 634Nd. Displacement-limiting numbers calculated by the Klippel analyzer for the 634Nd were XBl at 82% Bl = 4.3 mm and for XC at 75% CMS minimum was 3 mm, which means that for the Morel 6" woofer, compliance was the most limiting factor for a 10% proscribed distortion level. The last Klippel analysis curve shown gives the inductance curves L(X) for the TiCW 634Nd (see **Figure 9**). Like the

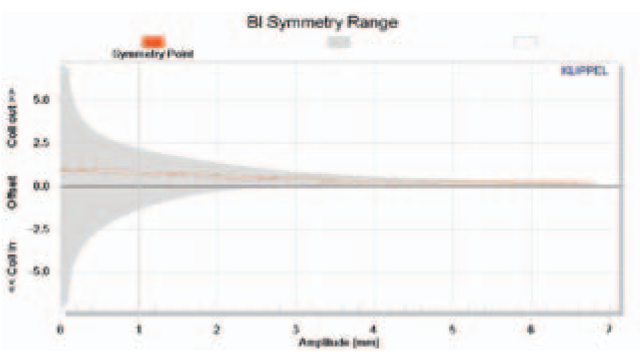

Figure 6: Klippel Analyzer Bl symmetry range curve for the Morel TiCW 634Nd

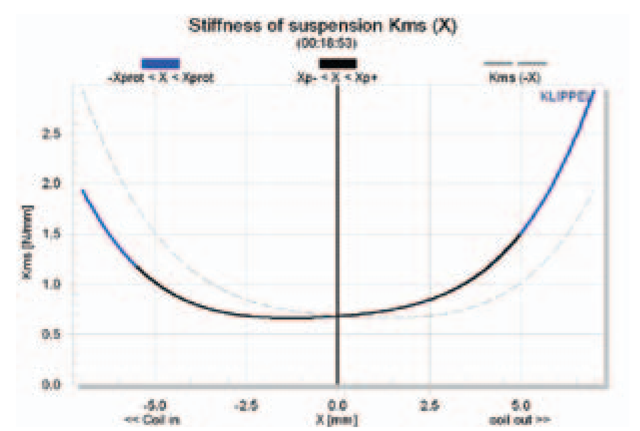

Figure 7: Klippel analyzer mechanical stiffness of suspension K_{MS} (X) curve for the Morel TiCW 634Nd

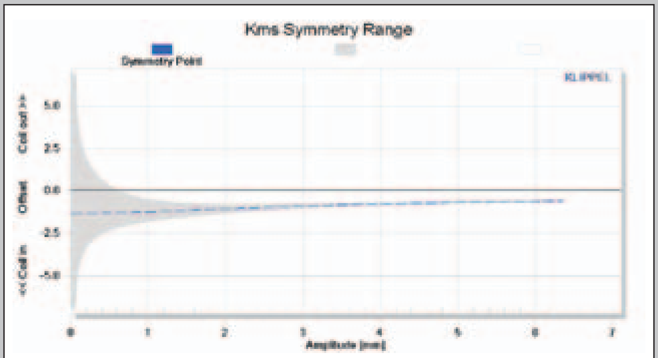

Figure 8: Klippel Analyzer K_{MS} symmetry range curve for the Morel TiCW 634Nd

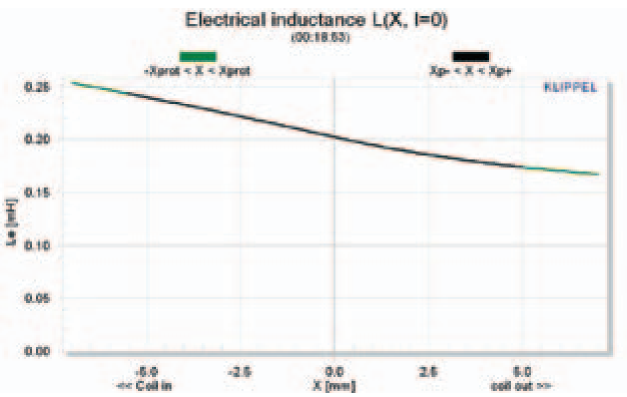

Figure 9: Klippel Analyzer L(X) curve for the Morel TiCW 634Nd

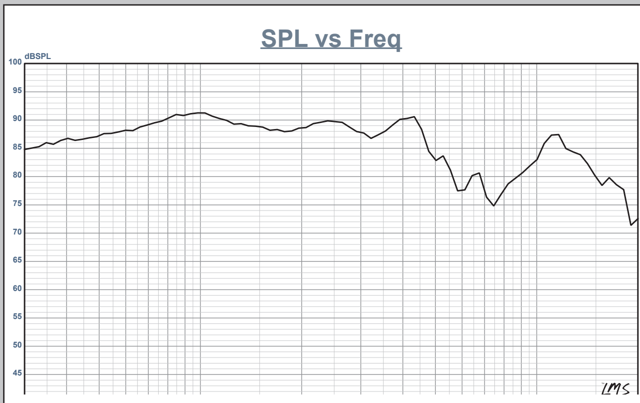

Figure 10: Morel TiCW 634Nd's on-axis frequency response

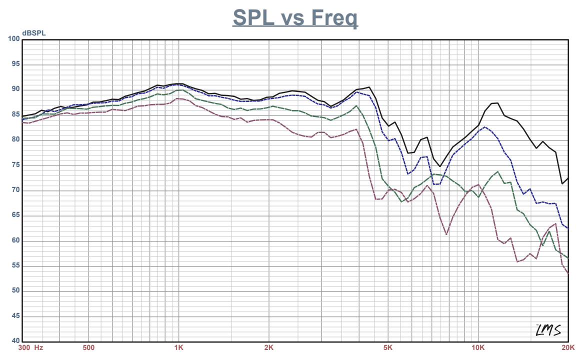

Figure 11: Morel TiCW 634Nd on- and off-axis frequency response (0°, 15°, 30°, and 45°)

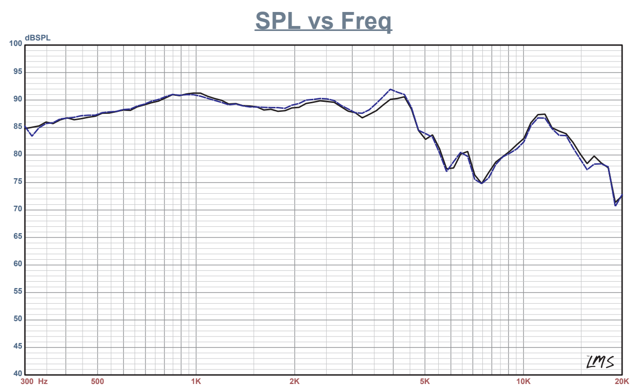

Figure 12: Morel TiCW 634Nd two-sample SPL comparison

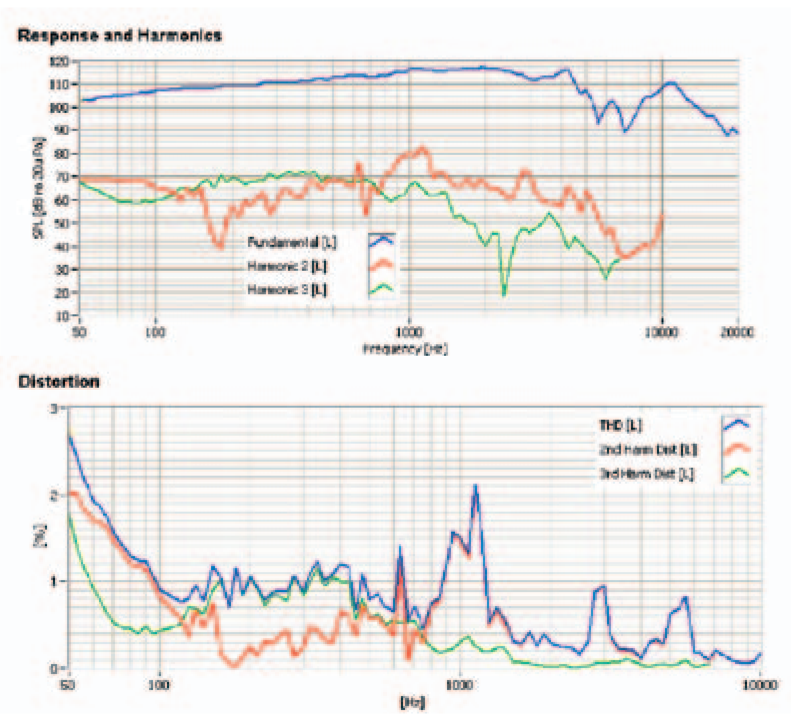

Figure 13: Morel TiCW 634Nd SoundCheck distortion plot

Wavecor FR084, the presence of a copper shorting ring limits the inductive swing from X_{MAX} in to X_{MAX} out of only 0.05 mH, which is excellent. Limiting self induction in woofers is key to obtaining low distortion and clarity.

Following the Klippel testing, I mounted the Morel 6″ woofer in an enclosure with an 8″ × 17″ baffle that was filled with damping material (foam) and then measured the DUT on- and off-axis from 300 Hz to 20 kHz frequency response at 2.83 V/ 1 m using a 100-point gated sine wave sweep. **Figure 10** shows the TiCW's on-axis response displaying a smooth rising response to about 5.2 kHz, making this driver probably easy to blend with almost any 1″ dome. **Figure 11** has the on- and off-axis frequency response at 0°, 15°, 30°, and 45°. A −3 dB at 30°, with respect to

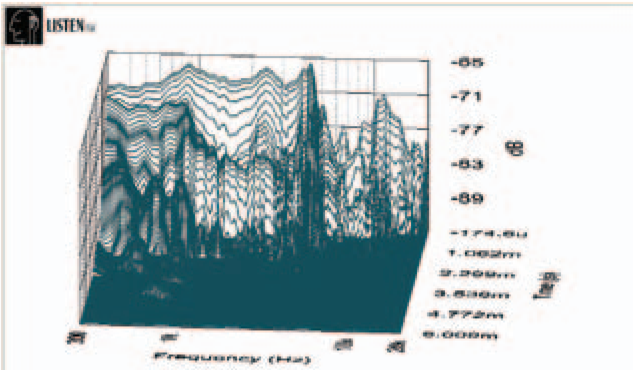

Figure 14: Morel TiCW 634Nd SoundCheck CSD waterfall plot

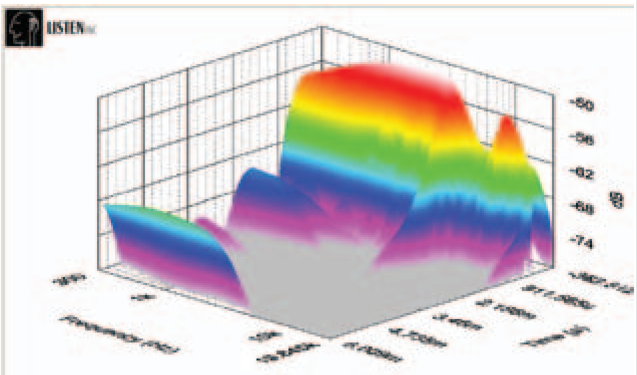

Figure 15: Morel TiCW 634Nd SoundCheck Wigner-Ville plot

the on-axis curve, occurs at 3.2 kHz, so a cross point in that vicinity should be fine. The last SPL measurement gives the two-sample SPL comparisons for the 6″ Morel driver, showing a close less than 0.5-dB match up to 3 kHz, with about 1-dB variations above that frequency (see **Figure 12**).

For the last group of tests, I employed the Listen SoundCheck analyzer to measure distortion and generate time/frequency plots. I used SoundCheck's software noise generator and SPL meter utilities to set up the distortion measurement. I mounted the woofer rigidly in free air and set the SPL to 94 dB at 1 m (1.95 V), and measured the distortion with the Listen microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 13**.

I changed the SoundCheck sequence file to set up the analyzer to get a 2.83-V/1-m impulse response for this driver and imported the data into the SoundMap time/frequency software. The resulting CSD waterfall plot is shown in **Figure 14** and the Wigner-Ville logarithmic surface map (for its better low-frequency performance) plot is shown in **Figure 15**.

Reviewing the data, this is a well-designed driver, but that's expected from a company with Morel's reputation for excellence. Visit morelhifi.com for more information on these and the other Morel Titanium Series drivers (including upcoming tweeters with Titanium voice coil formers). *aX*

 By Richard Honeycutt (United States)

Power Supplies for Hollow-State Equipment

Power supplies explained: sections, transformers, rectifiers, and more

The power supply is one of the most important subsystems of any piece of electronic equipment. Actually, "power supply" is a misnomer, since the power is supplied by a generator somewhere, through the power grid and, ultimately, a power receptacle. But electronic equipment requires DC electricity of a specific voltage, and hollow-state circuits typically use electricity of a specific AC voltage. The subsystem commonly called the "power supply" takes in electricity, usually 120 VAC, and changes it to the form needed by the circuit. I will follow tradition and call this power-conditioning subsystem the "power supply." Although older hollow-state units used batteries for power, I will confine my discussion to AC-powered circuits (e.g., amplifiers and ham equipment).

Power supplies are specified by output voltage and current, noise and ripple in the output, and voltage regulation—the variation percentage in output voltage that results from a given change in load current ("load regulation") or a given change in input voltage ("line regulation"). A typical hollow-state amplifier requires a 6.3 or 12.6-V filament supply and a "plate supply" (often called B+ after the "B" battery used in the earliest radios) of about 100 to 600 VDC. The filament current ranges from less than 1 A for a phono pre-amp to 5 A or more for a power amplifier. The B+ supply's output current can be as low as a few milliamperes to more than 1 A for very high-power circuits.

Noise and ripple in the B+ voltage appear in the amplifier's output, so the lower these are, the better. For hum and noise in the amplifier output to be 70 dB below the maximum audio output, the hum and noise in the B+ voltage must be less than 0.0316%. (Low hum also requires attention to the filament supply and to layout of parts, wiring, and grounding.) Voltage regulation especially affects power-amplifier output power stability.

POWER SUPPLY SECTIONS

Figure 1 shows the sections of a typical power supply. The voltage changer converts the "mains" voltage (120 VAC or 230 VAC, typically) to the proper voltages for the filament and B+ supplies. A single transformer with several secondary windings is typically used. The rectifier converts AC voltage from the transformer secondary winding(s) to pulsating DC. Rectifiers can be either hollow state or solid state. Hollow-state units are mainly used for musical instrument amplifiers in which a "soft" B+ supply (meaning one with poor load regulation) is desired. A soft supply enables the amplifier to produce high-power peaks, but only somewhat lower-power sustained notes or chords. This characteristic partially determines what musicians call the amplifier's "feel." Solid-state rectifiers are longer-lived and less expensive than hollow-state ones.

The filter purifies the DC voltage by eliminating AC ripple and noise from the power line. There are several topologies of filters, each with certain strengths and weaknesses. The regulator is seldom included in supplies for modern hollow-state amplifier circuits. But the better, pre-1960 amps (e.g., Altec-Lansing commercial amps and Hammond organ amps) did employ voltage regulators.

POWER TRANSFORMERS

A transformer consists of two conductors located in close proximity. Low-frequency transformers for audio and power frequencies utilize coiled conductors, wrapped around the same core. The coil into which power is fed is called the "primary," and the coil from which power is drawn to feed another circuit is called the "secondary." The number of turns in the primary and the secondary coils determines the relationship of the transformer's primary and secondary voltages and currents in this way:

$$\frac{V_S}{V_P} = \frac{N_S}{N_P} = \frac{I_P}{I_S}$$

V_S and V_P are secondary and primary voltages, respectively; I_S and I_P are secondary and primary currents, respectively; and N_S and N_P are the numbers of turns of wire in the secondary and primary, respectively.

If you play around with these equations, you find that the primary-to-secondary impedance ratio is proportional to the square of the N_P/N_S turns ratio. These relationships hold for a transformer with zero winding resistance, no magnetic leakage, and a magnetically perfect core. Most power transformers have negligible magnetic leakage and essentially perfect cores. To account for the non-zero winding resistance, transformer manufacturers usually specify the secondary voltages

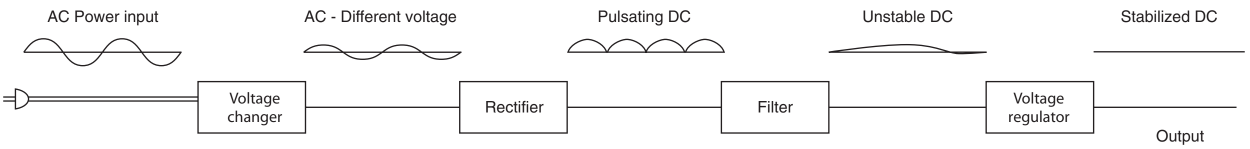

Figure 1: There are multiple sections to a typical power supply, including: the voltage changer, the rectifier, the filter, and the voltage regulator.

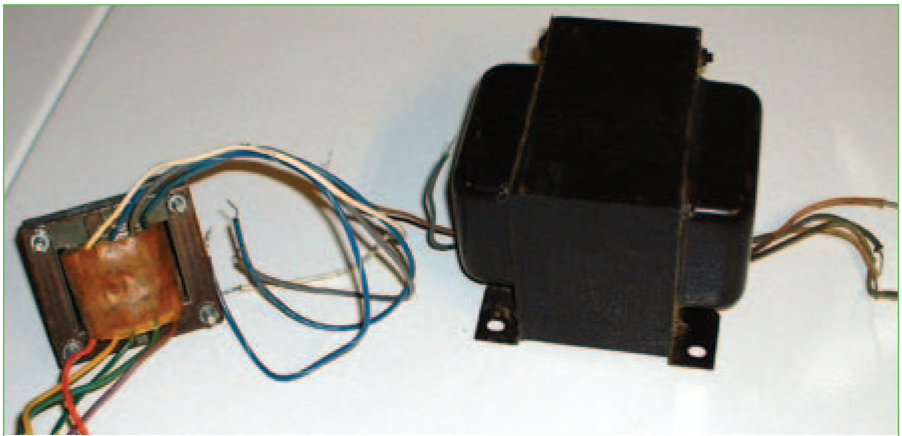

Photo 1: Power transformers vary in size. Some large ones have steel case enclosures.

based on the maximum output current drawn. Thus, a 12.6-V, 0.3-A filament transformer will have an output voltage somewhat above 12.6 V when no current is drawn from the secondary, and the secondary voltage will drop progressively as more load current is drawn. In fact, manufacturers often take line-voltage variations into account, so if the "120-V" line voltage is actually 105 V, the transformer will still output the rated 12.6 V. We seldom have to account for transformer imperfections when designing a power supply.

A transformer primary acts as a large inductor connected in parallel with a resistor. The inductor represents the winding's inductive effect, and the resistance primarily represents the reflected resistance of the secondary. (So, if the secondary is providing 12.6 V at 0.3 A, it has an effective resistance of 42 Ω (i.e., 12.6 V/0.3 A). This is reflected in the primary circuit as 42 Ω multiplied by $(V_P/V_S)^2$ = (120 V/12.6 V)^2, or 400 Ω. The inductance is large enough that its reactance is large compared to 400 Ω. Thus, when the transformer is delivering a full-load current, negligible current is drawn by the inductance. (When no load current is drawn from the secondary, the small "magnetizing current" drawn by the primary from the AC line corresponds to the current in the primary inductance.)

Inductive reactance is provided by $X_L = 2\pi fL$, where f is the AC's frequency and L is the inductance. In the U.S., the commercial power frequency is 60 Hz. In many other countries, 50-Hz power is used. A given primary winding's inductive reactance will be higher in the U.S. So, power transformers are specified for the proper power frequency: 50 or 60 Hz. There is no problem with using a 50-Hz transformer on a 60-Hz circuit. But 50-Hz transformers are somewhat larger and heavier, so the only reason to do so is to build a circuit that can be used with either power frequency. Because a 60-Hz transformer has a lower inductance than a 50-Hz one, using a 60-Hz transformer on a 50-Hz line will result in a larger magnetizing current, so the transformer will operate less ideally.

Photo 1 shows two power transformers. The larger one has a steel case enclosing the windings to prevent the transformer's electric and magnetic fields from introducing noise into the tubes and signal wiring. **Figure 2** shows the schematic symbol of a multi-winding power transformer designed for hollow-state circuits. The 350-V winding is for B+, the 35-V winding is for control grid bias of the output tubes, and the 6.3-V winding is the filament supply.

RECTIFIERS

A rectifier is a device that passes current in only one direction. A power supply's rectifier is made from one or more diodes. **Photo 2** shows a hollow-state and a solid-state diode, and **Figure 3** shows the schematic symbols for them.

Hollow-state diodes consist of at least an electrically heated filament and an anode or plate. Some hollow-state diodes also have an indirectly heated cathode. There are also hollow-state devices that consist of two diodes in a single glass envelope.

Hollow-state diodes have a higher internal resistance than solid-state diodes. The 5U4GB diode shown in **Photo 2** was a popular high-current rectifier diode in the 1950s and 1960s. It contains two plates and a directly heated filament. **Figure 4** shows this diode's plate characteristics when the diode is used to feed a capacitive filter with 200 V_{RMS} on the plates. Notice as more load current is drawn, the output voltage drops. If the max-min change in output voltage is divided by the corresponding max-min change in output current, then we find that the internal resistance under these conditions is about 11 Ω. A solid-state diode's internal resistance, used under the same conditions, would be a fraction of an ohm. Thus, the output voltage would change very little as output current was increased.

There are a variety of ways diodes can be connected to form rectifiers. The simplest is a half-wave rectifier. **Figure 5**

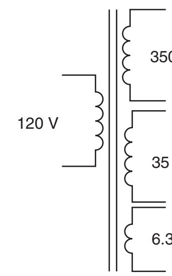

Figure 2: Here is the schematic symbol for a multi-winding power transformer designed for hollow-state circuits.

Photo 2: The solid-state diode (left) is dwarfed by the 5U4GB hollow-state diode (right).

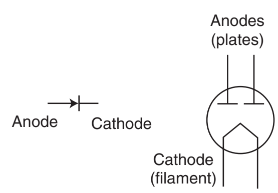

Figure 3: These are the schematic symbols for a solid-state diode and a hollow-state diode.

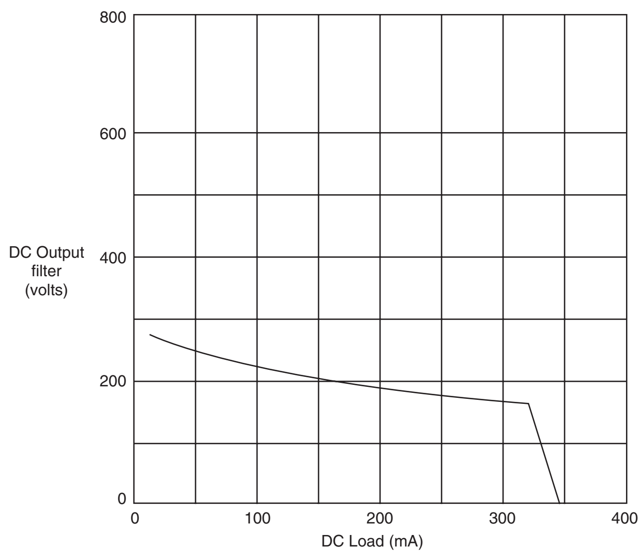

Figure 4: The diode's plate characteristics are used to feed a capacitive filter, with 200 V_{RMS} on the plates.

shows both a hollow-state and a solid-state half-wave rectifier. For both of these, the diode is positive-biased (conducting) on only the AC's positive half-cycle. Only half the AC wave is used to create DC; hence, the name "half-wave rectifier."

The pulsating DC output is really AC and DC mixed. The DC output voltage of a half-wave rectifier is 0.45 times the RMS voltage of the transformer secondary. There is actually more AC than DC. In fact, if the AC "ripple" is calculated as a percentage of the DC output, the ripple percentage is 121% and the percent ripple =

$$\frac{V_{RIPPLE_{RMS}}}{V_{OUT_{DC}}}$$

The ripple frequency is the AC line frequency (60 Hz in the U.S.). The combination of low ripple frequency and high ripple percentage requires a more aggressive filter for a half-wave power supply than for other types.

Figure 6 shows hollow-state and solid-state full-wave rectifiers. These use a particular characteristic of transformers to permit the conversion of both AC half-cycles into DC. This characteristic is the AC wave at one end of a transformer secondary is 180° out of phase with the wave at the other end (i.e., the AC at the bottom end of the secondary winding is inverted compared to the wave at the top end). This means when the wave at the top is negative (the top diode is not conducting), the wave at the bottom is positive (the bottom diode is conducting). The two diodes take turns supplying the pulsating DC output. This type of rectifier requires a transformer with a center-tapped secondary winding. The center tap is grounded.

The DC output voltage of a full-wave rectifier using a center-tapped secondary winding is 0.9 times the RMS voltage of half of the winding, or 0.45 times the full secondary voltage. Transformers for hollow-state, full-wave rectifiers sometimes have the secondary voltages specified as "volts plate-to-plate." This is twice the voltage from each end terminal to ground, since the secondary is grounded in the center. A full-wave rectifier's ripple frequency is twice the line frequency (120 Hz in the U.S.), and the ripple percentage is 48%.

Figure 7 shows a different kind of full-wave rectifier: a full-wave bridge. For simplicity, only the solid-state version is shown. **Figure 8** shows how current is conducted in a full-wave bridge rectifier to convert AC to DC. A full-wave bridge rectifier does not require a center-tapped transformer secondary winding. Its output voltage is 0.9 times the transformer's secondary voltage. Like the full-wave center-tapped rectifier, the full-wave bridge rectifier has a ripple frequency of twice the line frequency and 48% ripple.

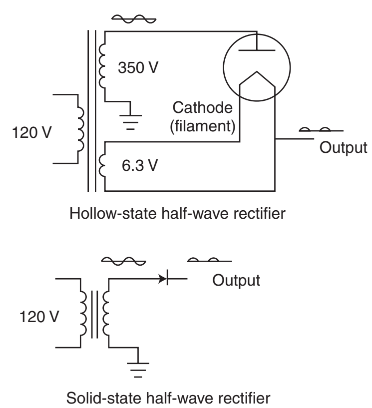

Figure 5: Half-wave rectifiers are the simplest form of rectifier.

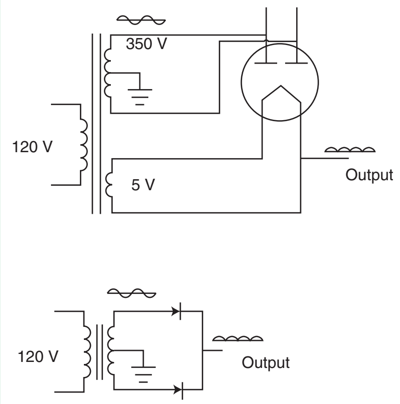

Figure 6: Full-wave rectifiers permit the conversion of both AC half-cycles to DC.

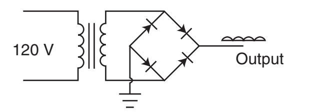

Figure 7: A solid-state full-wave bridge rectifier is shown here.

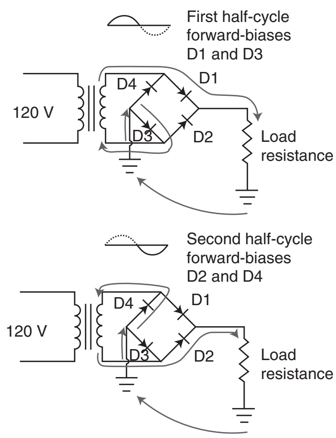

Figure 8: The arrows in this figure show the current flow in a full-wave bridge rectifier.

In next month's column, I'll look at voltage-multiplier rectifiers, then I'll move on to filter circuits. *aX*

*Author's note: The amp-test page referenced in my article, "Differences in Amp Sound: What's the Truth?" (*audioXpress*, November 2012) is now available on my website, www.edcsound.com/amptest.htm. Because of confusion about release dates, some readers thought the amp test webpage would be posted by the time they recieved their November issues of* audioXpress*. In fact, it was posted on October 27, 2012. Apologies for the misunderstanding.*