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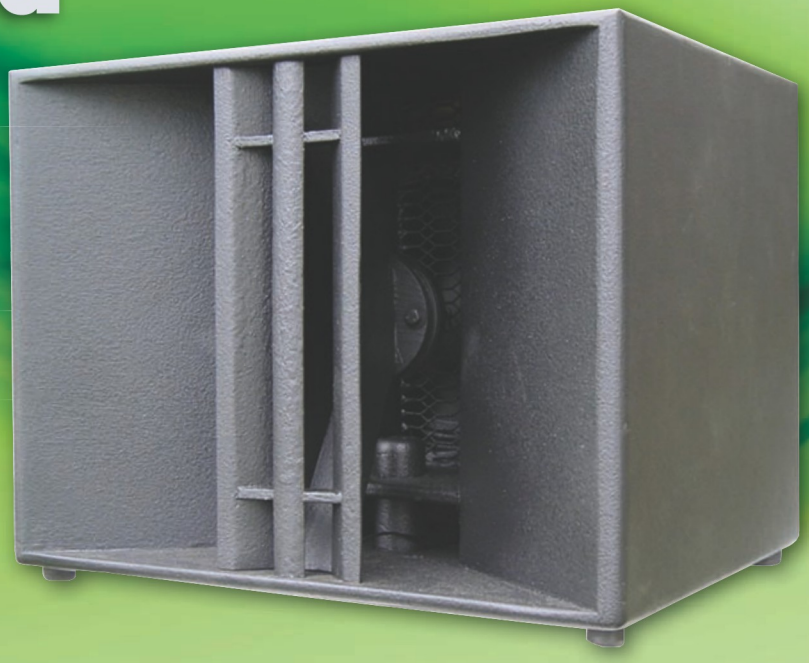

Expert Tips on
CINEMA DESIGN

Error-Correction
POWER AMP, Pt. 2

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Like almost every institution, *audioXpress* is affected by the worldwide downturn in economic viability. Our advertising linage has been down for 2008 by 50%. Thankfully, we are seeing a slight rise in circulation income due, we suspect, to some of you responding to my message in the July issue about our condition. Those two- and three-year subscriptions are a welcome help. What we really need, however, is a couple of thousand new readers.

With that goal in mind, you will notice *audioXpress* contains a special second edition of **Reliable Reviews**. We welcome reviewers from the staff of *$ensible Sound* magazine, along with Don Nowak as advertising representative, specializing in consumer products. I believe strongly there is a place in audio journalism for more reviews of reasonably priced, quality audio equipment. We also know from years of talking with you that DIY types buy all manner of equipment for their systems, in addition to what they build. I hope you will find these reviews of interest and invite your comments. This new section must earn its way, however, so between five and six pages of advertising are needed to support ten pages of reviews.

We are instituting a campaign to garner as many of your e-mail addresses as you are willing to share. This provides us with a way to communicate with you more inexpensively, and to add communication channels for much more information about your activities as audiophiles. This could help us provide a monthly news e-mail medium for much that we cannot find room for in the magazine, or on our website.

We are busy preparing one of our business-to-business periodicals, *Multi Media Manufacturer*, for bi-monthly digital delivery. This allows us to deliver the information inexpensively and quickly, especially to overseas subscribers. Costs for conventional printing and especially delivery are rising each year. We are trying to be prudent in experimenting and planning for ways to continue to deliver the articles and ideas that make audio the compelling avocation it is.

We are being aided as well by our Dutch friends from *Elektor* who are launching a North American edition of *Elektor*. We provide customer service, support in developing marketing, circulation facilities, and advertising aid among many other services. We are both learning from each other to our mutual benefit.

The end of this year marks the completion of 40 years of the Audio Amateur enterprise. We began in 1970 with mostly a family staff and a $500 savings withdrawal. 1970 was not an auspicious year for the decision. Since that time the landscape has become festooned with audio periodical tombstones: *Electronics World, Popular Electronics, Stereo Review, Audio, Wireless World*, to name only a few. All through these four decades our magazines have been a cooperative effort. I hope you will think we still are and will offer your cooperation as generously as you have up to now. --E.T.D.

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paX, Part 2

Continuing with the paX amp design, covering the voltage amplifier and input buffer stages.

Last month I discussed the principle of error correction and developed an error–correction power output stage. In this installment I will use H.ec in a voltage amplification stage and present the complete amplifier.

BACKGROUND

Let me back up to the conceptual circuit of error correction, as defined by Malcolm Hawksford, shown in **Fig. 1A**. You know that if the summers S1 and S2 are unity gain summers, you get Vout = Vin, an ideal gain-of-one amplifier stage. But for the voltage amplification stage (Vas) that drives the output stage, you need much more than unity gain.

Common amplifier gains are around 26 to 30dB, a gain of some 20 to 30 times. I like the lower value because it is better to have less gain in the power amp than to have a lot of unnecessary gain, only to need to turn down the volume control to get rid of it. It turns out you can use H.ec in voltage amplifiers quite easily if you insert an attenuator in the Vout sense leg, similar to what you would do in a regular negative feedback circuit.

I did that in **Fig. 1B**. The sums are slightly different: Vc = (Vout/ß)-Ve, but still Ve = Vin-Vc. Substitution and rearranging shows that now Vout/Vin = 1/ß. So, if you make ß, as is customary, from a simple resistive 1:20 voltage divider, you now have an amp with gain, linearized by error correction.

One other very important note: When I found that the open-loop gain was no longer part of the amp transfer equation, I said that the gain of the amp block no longer had any bearing on the final result. That is true in theory, but not in practice.

In the case of the power output stage, I said the gain is "about 1." A realistic value would be 0.98 at mid frequencies and light load, down to 0.95 at higher frequencies and higher loads. That means that the error-correction circuit needs to add 0.02 to 0.05 times the signal level to the input to straighten out the amp. Intuitively, I believe it is advantageous to have only small signals in the error-correction circuitry that helps it to work linear and with low distortion.

But in the case of the Vas, if the forward gain block has only a gain of 1, the error-correction circuit will need to add 19 times the signal level, and it would be much more difficult to design simple circuits that could handle those levels with high linearity. So, what you need is to make the open-loop gain of the Vas amplifier block such that the signal levels in the error-correction circuit are minimized. This will be the case when the open-loop gain is as close as possible to the closed-loop gain. Then, there will be quite small differences between the input of the actual amplifier and the scaled-down output (by ß), and that small difference then is the error signal.

This, of course, is totally at odds with what you would do for a negative feedback amp. There, you would try to get the highest open-loop gain you can get away with, stability wise, to have a lot of excess gain for the feedback loop to work with. Not so with H.ec, and a corollary to this is that your Vas can now be a very simple gain-of-20 amp.

VOLTAGE AMPLIFIER

The particular topology I chose is in **Fig. 2**. It's very simple. U1 is a unity-gain buffer. Vin applied appears at the buffer output across R7.

The output current through R7 comes from the supplies, of course, so you find a signal current Vin/R7 in the current mirrors formed by Q1 and Q3. The DC quiescent current for the buffer (I'll tell you later which buffer type I used) happens to be about 7mA. With an R7 value of 220Ω and a current of 14mA peak-to-peak available, this buffer can handle a little over 3V peak (about 2V RMS) before running out of class-A, which, of course, is good for linearity. With an amplifier gain of 20 that is enough to deliver 100W in 8Ω.

That same signal current in Q1, Q3 is mirrored through Q2, Q4 and flows through R16 to generate the output voltage. As a first approximation, be-

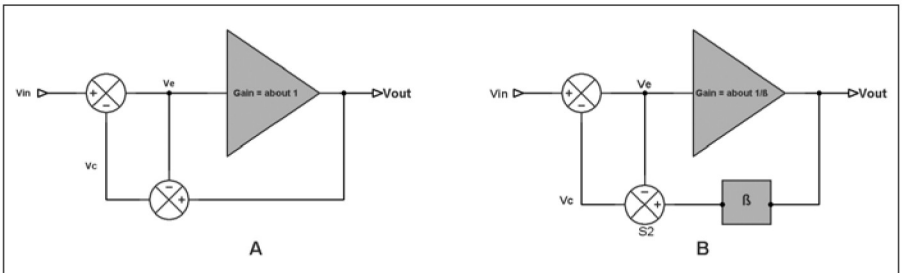

FIGURE 1A: Basic H.ec topology for gain-of-1 stage
FIGURE 1B: Arbitrary gain stage.

cause the same signal current flows through R7 and R16, and the signal across R7 is the same as Vin, the gain (open loop) of this circuit is simply R16/R7, which is about 20 times. So, this is the Vas stage.

But in Part 1 I said that the amplifier output stage needs to be driven by a low (and constant) source impedance (ideally zero Zsource) for the output stage H.ec to work; so add a low output impedance emitter-follower stage to the basic Vas stage (Q10, Q9). This emitter-follower stage is also biased at about 7mA. If the voltage across D3 is equal to Vbe of Q10, the voltage drop across R6 and R8 will be the same, so if you chose R6 = R8, the same current that is flowing through Q2, Q4 will also flow through Q10, Q9. In practice, the diode voltage and Vbe don't exactly match, so make R6 slightly smaller than R8. D3 and D4 provide a measure of temperature stabilization to the emitter-follower output stage bias current.

Now that you have the main gain block for the Vas, enclose it in an er-

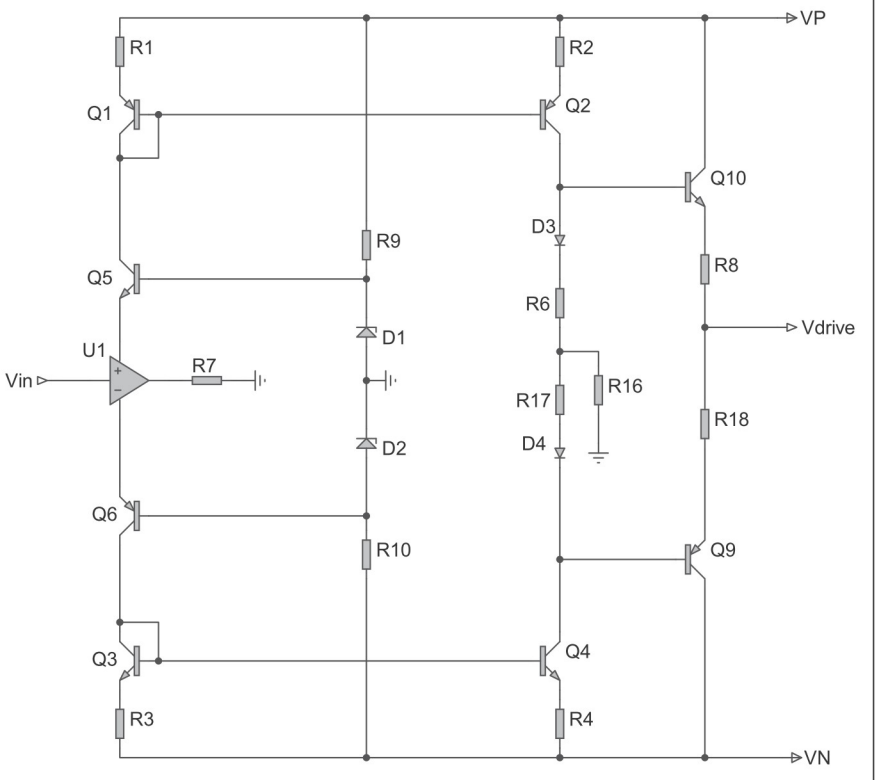

FIGURE 2: Vas basic circuit.

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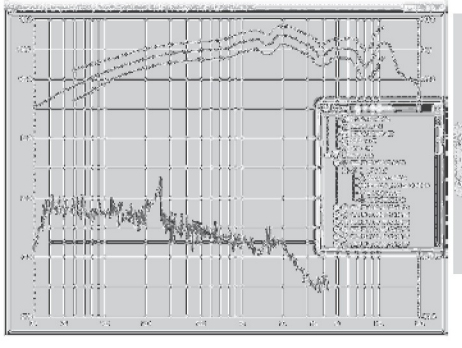

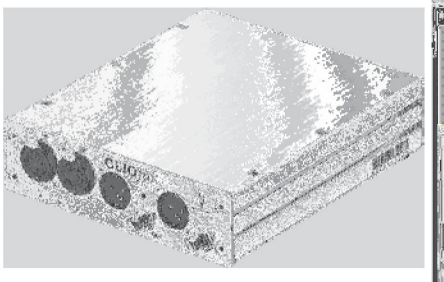

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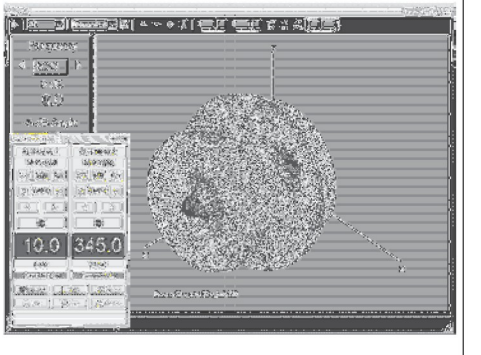

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ror-correction loop as you did with the output stage (**Fig. 3**). Note here, that, first, although both the EC sense and the EC generation resistors are given as "R," they are not equal. The sense resistance value at the right is the sum of resistor R plus the output resistance of the ß-network (R11//R12). It is the total resistance of R plus R11//R12 that determines the error current into the CCII. That same current generates the EC voltage when flowing through the left side R, and these voltages should be equal, so similarly, the left side "R" is actually the series resistance of the nominal R and the output resistance of whatever Vin source will drive the Vas. I'll come back to this issue later.

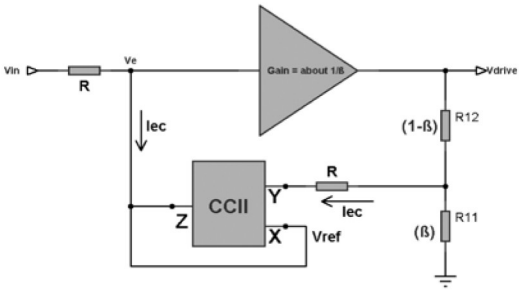


FIGURE 3: The CCII as the H.ec element for the Vas stage.

Second, note that the gain of this amp is not strictly 1/ß. Terminal Y is a virtual earth point. To calculate the exact attenuation from Vout to the node R11, R12 and R, you need to account for the fact that for the attenuation, seen from the Vas output Vdrive, R appears in parallel with R11. So, the attenuation, and thus the gain, is slightly higher than 1/ß.

You need to consider both of these issues when you dimension the error-correction sense and generation resistors. Just as with the output stage, I use an AD844 current conveyor for the EC circuit, while the buffer is used in the amplifier input stage to drive R7. The complete Vas circuit is shown in **Fig. 4**.

I have yet to mention the diodes D9, D12, D13, and D14. What happens if you overdrive the amp? The positive feedback loop is regenerative and will continue to increase the input signal until stopped by some physical limit such as the supply voltage or current or voltage limitations in the circuit. In the Vas stage, this is limited by the four diodes across the error-correction generation resistor R29 (**Fig. 4**).

When the amp is overdriven, the error correction current out of pin 5 of the AD844 CCII will increase considerably. This current will start to generate a large correction voltage across R29, to the point where the threshold voltage of the diodes is reached. At that point, the effective impedance of R29 collapses to just the dynamic diode impedance that is only a few tens of ohms. Further increases in the error-correction current will not generate more error-correction voltage and the positive feedback loop is broken. This makes the clipping clean and recovery fast. But you run the risk that the diodes increase the distortion because they may already conduct small current before the threshold, which would upset the error-correction accuracy.

It turns out that because of the very small error-correction voltages across R29, this can be avoided. In **Fig. 6** you see that if you use a single pair of soft clipping diodes, the distortion starts to

FIGURE 4: Full circuit of H.ec Vas stage.

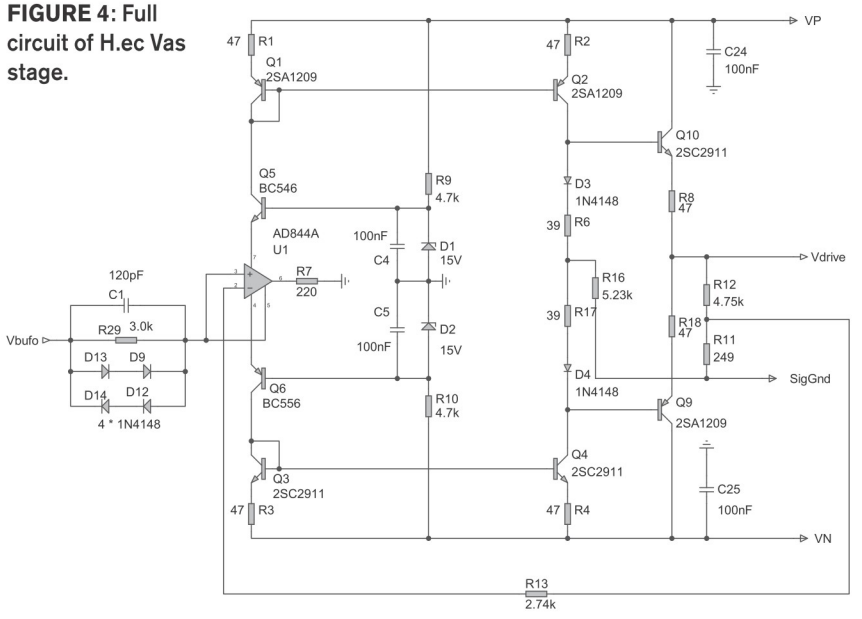
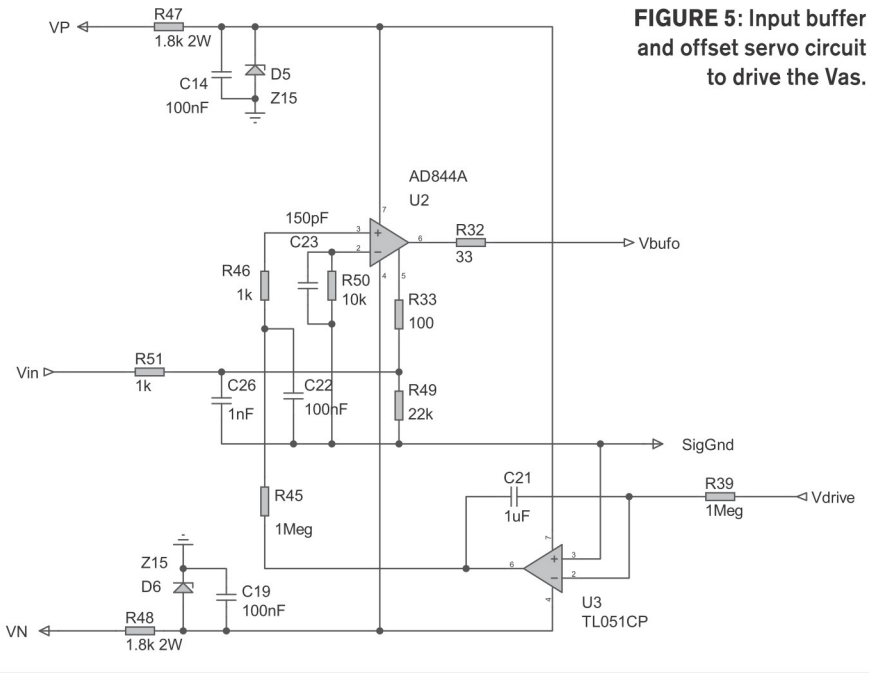


FIGURE 5: Input buffer and offset servo circuit to drive the Vas.



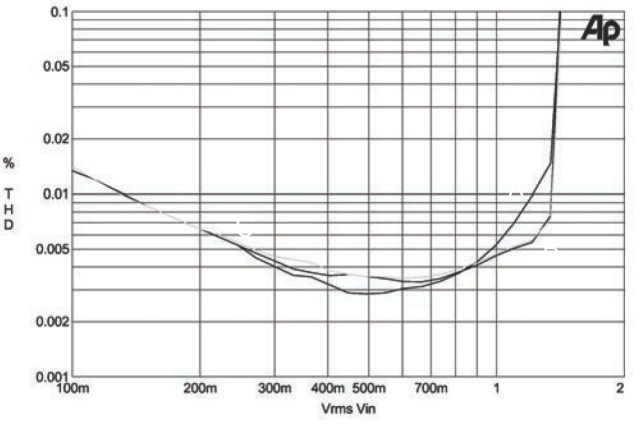


FIGURE 6: Amp distortion as a function of output level, with single diode (A), dual diode (B), and without soft-clipper diodes (C).

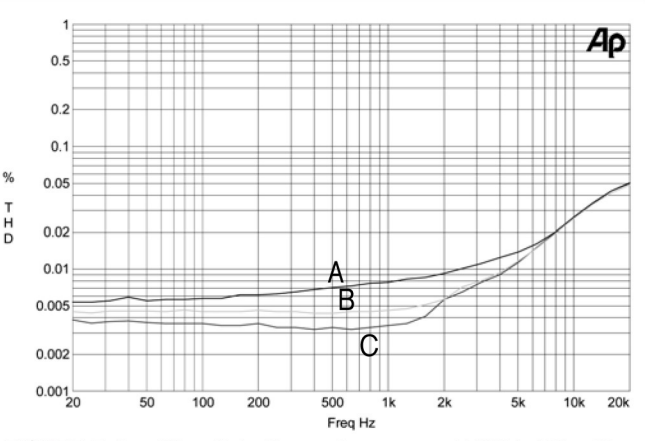


FIGURE 7: Amplifier distortion vs. frequency at 50W in 8Ω, with single diode soft clipper (A), dual diode soft clipper (B), no soft clipper (C).

rise before maximum output. With the dual pairs, no difference is apparent.

Figure 7 shows this effect from a different perspective. The soft clipping diodes for the negative signal part have been temporarily removed so only the positive signal is affected. You can see that without the diodes, clipping recovery is delayed (negative part), while with the diodes (positive part), there is only a hint of delay.

BUFFER STAGE AND DC OFFSET

All that is needed now for a complete amplifier is the input buffer and the DC offset servo (**Fig. 5**). Just as with the output stage, you need to drive the Vas input resistor, which develops the EC voltage (R29), with a low impedance source. You guessed it: I will again call upon the open-loop buffer in an AD844 to drive the Vas from Vin. The signal enters the buffer in U2 through R51 and R33 at pin 5 and exits at pin 6.

The enemy of your speakers (and the amp as well) is DC offset at the output terminal. Most amps have some sort of means to avoid that. In this amplifier, the output stage itself doesn't need any additional measures: the error correction ideally duplicates the driving voltage (from the Vas) to the output terminal, with zero offset, although there will still be

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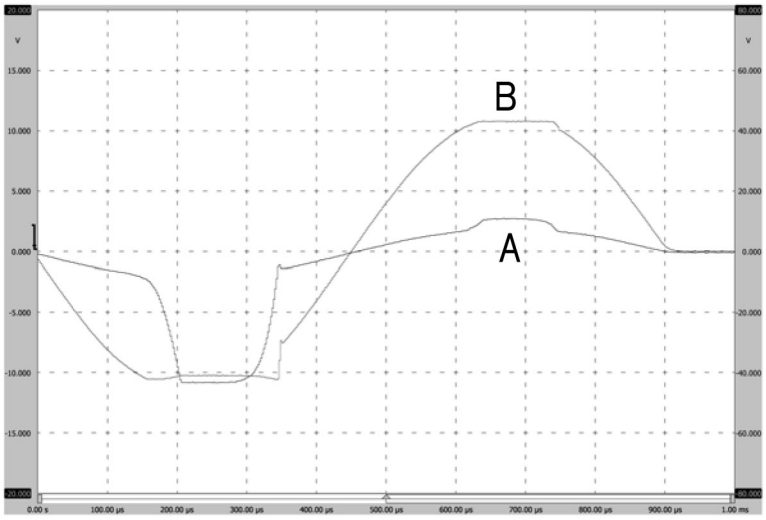

FIGURE 8: Error-correction current (A) and amplifier output clipping (B) with soft clipper (positive signal) and without soft clipper (negative signal).

TABLE 1: AMPLIFIER PARTS LIST

Category	Reference	Value	
Resistors 0.25W MF	R1-4, R8, R18	47	
	R6, R17	39	
	R7	220	
	R9, R10, R12	4.75k	
	R11	249	
	R13	2.74k	
	R16	5.23k	
	R19, R20, R30, R31	2.15k	
	R21-24, R33, R60	100	
	R25, R34	4.99k	
	R26, R27, R40-43, R46, R51	1k	
	R28	10	
	R29	3.0k	
	R32	33	
	R35, R36	619	
	R37, R38	680	
	R39, R45	1 Meg	
	R44	7.5k	
	R50	10k	
Resistors 5W ceramic	R52-R55	0.22E	
Resistors 2W MF	R47, R48	1.8k	
	R64	1.5E	
	R65	6.8E	
Capacitors	C1, C3, C23	150pF	polypropylene or mica 0.2″ pitch
	C2, C4, C5, C9, C14-16, C19, C22, C24, C25	100nF ceramic	0.1″ pitch
	C6-C8, C13, C21	1µF/63C foil	0.3″ pitch
	C10	220nF/50V ceramic	0.2″ pitch
	C11, C12, C17, C18	470µ/63V	0.2″ pitch, 0.5″ diameter
	C26	1nF foil	0.2″ pitch
ICs	U1, U2, U4	AD844AN	
	U3	TL051CP	
Transistors	Q1, Q2, Q9	2SA1209	
	Q3, Q4, Q10	2SC2911	
	Q5, Q11, Q17, Q18	BC546B	
	Q6, Q14, Q19, Q20	BC556B	
	Q13, Q16	STD03N	(+ isolation pad and mounting hardware)
	Q12, Q15	STD03P	(+ isolation pad and mounting hardware)
Connectors	J9	Header 2*1 pin	signal input
	J10	Header 2*3 pin	to protection board
Diodes	D3, D4, D9, D12, D13, D14	1N4148	
	D1, D2, D5, D6, D10, D11	Z15 0.5W	
	D7, D8	1N4003	
Miscellaneous	L1	5µH	see construction guide
	RV1, RV2	200E trimmer	
	PCB		
	IC socket (4)	DIL08	

some offset of a few millivolts from the error-correcting AD844. The only requirement is that the Vas has negligible DC offset.

The Vas, however, has higher gain and will amplify its own AD844 offset with that gain. I decided to implement a DC servo to keep the Vas offset under control. The servo uses a standard low-offset op amp, a TL051CP. I will use the uncommitted current conveyor in that input AD844 (U2, **Fig. 5**) to couple the servo signal to the amplifier.

The way this works is as follows: Remember that whatever current flows in or out of the low impedance input pin 2, also flows (in opposite direction) out of or into pin 5. You also know that the voltage at pin 2 and 3 will track accurately. When you couple the servo signal to the reference input pin 3, it will cause a current in R50 to keep pin 2 at the same level. That same current will flow in R33, and in this way the offset correction will be added to Vin. R49 ensures that that current, which essentially is DC, can flow in R33 even if the Vin comes from a coupling capacitor.

Finally, this input buffer and servo amp have their own ±15V supply from two zeners, D5 and D6. With this circuit, compound offset of the whole amp is just a few millivolts.

POWER SUPPLY

The power supply for this amp should deliver about 2 * 42V DC. Although the output devices are rated for higher voltages, it is not advisable to increase the supply in trying to achieve more output power. The allowed dissipation at higher Vce is much less than the rated DC dissipation due to secondary breakdown limitations of the SOA (Safe Operation Area). With loads that dip substantially below 4Ω, even temporarily, the SOA may be exceeded and an output Darlington destroyed. The supply for the error-correction circuit in the output stage is bootstrapped from the output; that part of the circuit can actually drive the output devices beyond the supply voltage.

Consequently, it is the Vas that determines how close the output can

get to the supply rails. Because the Vas will clip first, there is no clipping overdrive and overdrive delay in the output stage. The output can swing within a few volts of the supply, which is better than in most amps (unless you use separate higher voltage Vas supplies).

The power supply is uncritical. The error correction not only corrects internal amp nonlinearities, but also any power supply ripple or noise that makes it into the circuits. (The supplies for the error-correction AD844s are stabilized separately with zener diodes.) Therefore, a classic rectifier-and-reservoir capacitor supply delivering 2 * 42V DC under load is sufficient for about 100W in 8Ω, 200W in 4Ω.

The power supply circuit details are given in **Figs. 9** and **10**. For a stereo version, a 300 or 400VA transformer should be sufficient; after all, you will not drive both channels at full power for a long period (except maybe on the test bench). You can use the common transformer with a center-tapped secondary, but you can use other configurations.

One nice option for a stereo amp is to use two completely separated supplies, each with a 250 or 300VA transformer. Completely separated supplies for each channel make it easier to maintain strict star grounding.

It is important to get the grounding right. Ground currents back to the supply from the speaker, as well as from the reservoir caps, can be quite large and have all kinds of ripple and noise. Such ground currents generate voltages across the ground return wiring. If the ground for the signal is connected to this ground wire at another point, you have effectively produced a small ripple signal that appears in series with the input signal. This may seem far-fetched, but with very low distortion and very clean amplifiers, even a few mV of these noise signals can be enough to ruin the linearity and distortion figures of an otherwise very good amplifier.

Usually you try to prevent this by using a star ground: All ground returns come together at a single point, and no signal return runs through the same ground wire with another return loop. The power connections to the PCB also get to a central star point. On the PCB itself, I've tried to maintain this. There are two or three separate traces that bring the returns for the regulator zeners and the decoupling capacitors to the same point.

The signal returns from the input network, the gain setting resistors R7, R16, and the attenuation network R11, R12 are brought to a separate point. All return wires from the supply, the transformer, and the speaker are returned to this point, the star ground.

The signal grounds are returned to this point via a low-value resistor R28, which is then by definition the "clean" ground point (pad J1 on the PCB is the clean ground and should be used

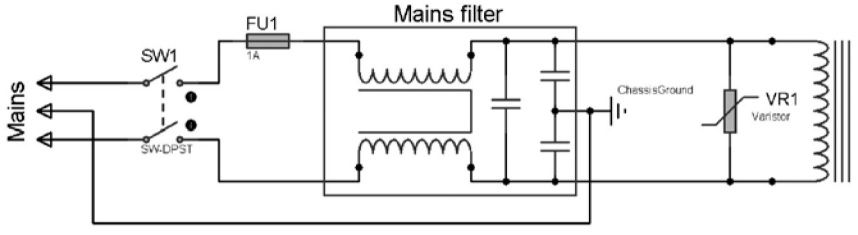

FIGURE 9: DC power supply. Diodes are paralleled, dual, fast, soft recovery 3-pin TO220 types.

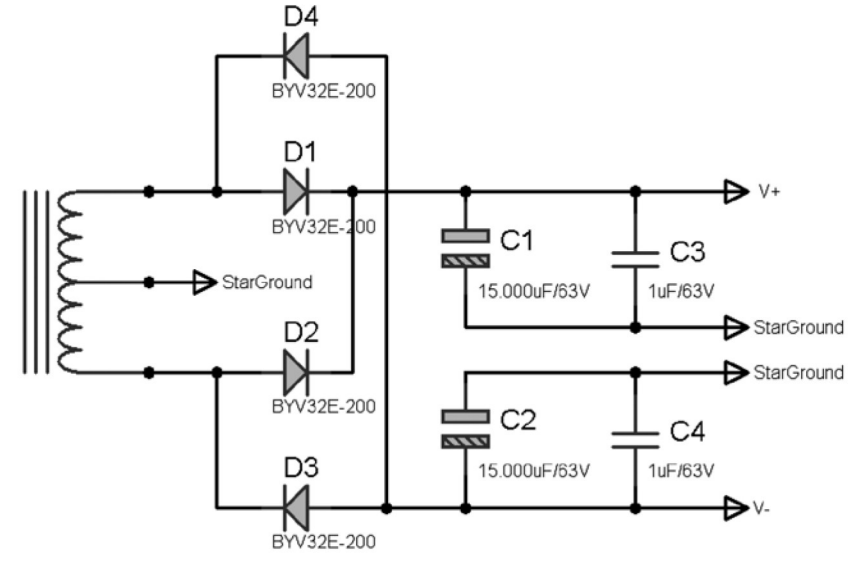

FIGURE 10: Mains wiring for the power supply, mains filter, overvoltage clamp, and fuse.

as "ground" for measurements as well). Any error voltages generated by ripple and pulsating currents in the ground wires cannot end up in the signal, except through radiation. In turn, you can minimize this by making all high-current wires short and as far away from sensitive signal areas as possible.

Finally, the rectifier diodes should be types that recover quickly from voltage reversal when switching off, and do so "softly"—that is, without very sharp current steps. Diodes that take a long time to reverse themselves and in the process generate sharp current steps may cause high-frequency noise that is difficult to filter out. Fast, soft recovery types don't need capacitors or snubbers; indeed, such capacitors would only increase the transmission of mains-borne noise to the amplifier.

I recommend you use Philips BYV32E-200 diodes, which are TO220 dual types with a common cathode; the two diodes should be paralleled. They have a relatively small threshold voltage, and are quite inexpensive. They don't need to be heatsinked, so you can put them on a PCB in free air. For the reservoir capacitors, you should use a minimum of 15,000µF per supply polarity. 63V rating is adequate.

You should use a mains filter (**Fig. 10**) to keep out high-frequency noise and switching pulses riding on the mains. Good mains filters are not cheap but worthwhile; use a type with at least 12A rating for US (120V) mains. Ground the filter ground lug as well as the mains safety ground wire to the chassis only. Use a slow-blow fuse of 6.3A for a stereo version, larger if you use larger reservoir capacitors. A 150V AC varistor clamps any high-level pulses that make it out of the filter.

CONCLUSION

So, there you have it, a complete, high-quality yet simple error-correction power amp. A complete parts list is in *Table 1*. Although harmonic distortion measurements do not always correlate with sound quality, they do give an indication of linearity and behavior of an amplifier. Therefore, a few performance curves are given in **Figs. 7** and **8**.

If you want to build this amp, there is a detailed Construction Guide for the amp on the *audioXpress* website. Working step-by-step, checking your work after each major part will help to avoid errors and, if errors are made, to quickly isolate and fix them. Also check on my website for any last-minute information, additions, or corrections.

This is not a very difficult project, but will give you an excellent amplifier, which reproduces your source music faithfully—adding nothing, taking nothing away. This amplifier is stable and will happily drive a wide range of different speakers. In short, an amplifier that will let you enjoy your music for many years to come! *aX*

[This article originally appeared in the April and May '08 issues of *Elektor*—Eds.]

Further reading: A MOSFET Power Amplifier with Error Correction, Robert R. Cordell, Journal of the Audio Eng. Soc., Vol.32, No.1/2, 1984 January/February; also available at www.cordellaudio.com.

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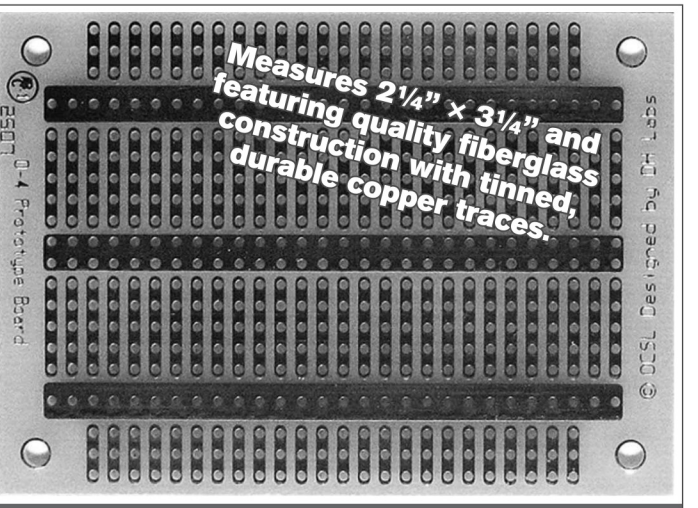

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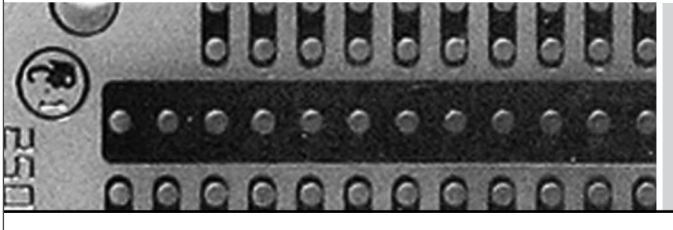

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The Jack 10, Part 1

You can put this speaker to work in a variety of situations.

By Bill Fitzmaurice

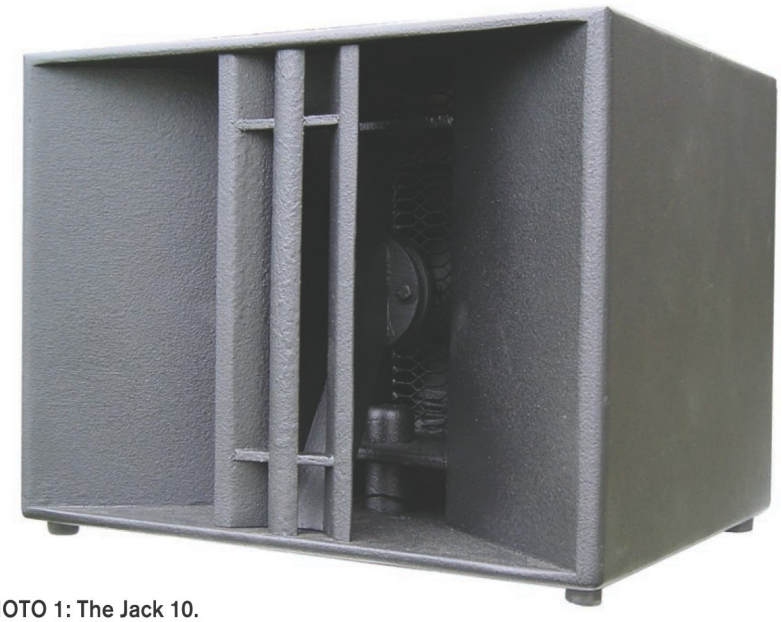

PHOTO 1: The Jack 10.

Jack 10 is a "Jack of All Trades" pro-sound speaker: you name it, it can do it. You can build it as an old-school woofer-only electric bass cab, or as a modern full-range bass or keyboard cab. Use it as a PA top along with subs, or as a full-range PA without subs. You can use it one per side for small room PA needs, or stack it vertically in a line array for thousand-seat venues. If maximum versatility is what you need from a speaker of minimum size and weight, not to mention cost, Jack 10 is one to consider.

I designed this cabinet to operate with 10″ pro-sound woofers (**Photo 1**). Optimal specs are fs 55Hz, Qts .35, Vas 60 ltr, Xmax no less than 3mm. The recommended premium driver is the Eminence Deltalite II 2510. Less expensive options are the Eminence Basslite 2010 and Beta 10. The PA compression horn driver version uses the Eminence ASD-1001 HF driver. If you substitute a different driver, the maximum allowable diameter is 4″, maximum depth 2½″. These drivers are available at www.speakerhardware.com and all the other hardware you'll need to complete the build.

The cabinet is constructed from ½″ plywood, braced to remain vibration-free at very high power levels. You may use softwood (spruce, pine, or fir) plywood, or semi-hardwoods such as birch or poplar. A Jack 10 requires a little more than half of a 4 × 8′ sheet. Buy a grade with five or more plies, but don't waste money on high-priced premium grades or marine plywood.

SECURING THE BOX

Dado or biscuit joints are an option, but this design is so strong that butt joints are quite adequate. You may fasten the joints with screws; if so, you need a pilot/countersink bit and driver bit for your drill. Use drywall screws, 1.25″ long. You also may fasten the cabinet with 1 5/8″ ribbed shank paneling nails, using a nail punch to set their heads below the surface of the wood. Another option is a pneumatic brad nailer, using 1.25″ brads.

Fasteners don't hold the cabinet together; they merely hold the parts in place while the adhesive in the joints sets. Use a polyurethane base construction adhesive, such as PL Premium, applied with a caulking gun. Polyurethane expands as it cures, filling gaps to ensure airtight joints. However, urethane is also messy to work with.

After applying a bead to a joint, release the pressure on the caulking gun trigger and plug the end of the tube before it oozes all over your workshop —and you. When the adhesive cures it will expand out of the joint. Do not wipe or scrape uncured adhesive; you'll just spread it around and make matters worse. Wait until the adhesive has cured, then use a sharp chisel to scrape away the excess.

You'll probably use at least twice as much adhesive as you need on your first few joints until you figure out how little is required. Do not use traditional woodworking glue, because any air leaks in the joints will severely compromise

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performance, while expanding urethane adhesive works so well filling gaps up to 1/8″ wide that you don't need absolute accuracy in your parts cutting. The design is very forgiving of errors, so if things are a bit "off," don't be overly concerned. The only rule that can't be violated is that every joint must be absolutely airtight.

BUILDING THE BOX

The first construction step is to cut out the cabinet top and bottom (**Fig. 1**). Draw the joints on them with the other parts as shown. The lines indicate where the front edges of the baffle and horn panels join the top and bottom. After drawing these lines, place a scrap piece of plywood against each line and trace the outline of the full joint. Using a small drill bit—1/16″ or so—drill pilot holes through where the joints with the horn plates and baffle will be. Don't come closer than an inch from where the end of the jointed part will go. Space the holes 5 to 6″ apart for assembly with screws, 3 to 4″ with nails, closer as required with smaller parts.

Cut the baffle (**Fig. 2**). The driver hole is centered. First draw a 2½ × 8 3/8″ box, then at both ends use a driver frame to trace a curved section extending another 3/16″ or so. To cut the straight edges clamp or screw a guideboard or saw sled to the baffle at the correct distance from the cut line. Raise the saw out of the shoe, place the shoe tight against the guide, start the saw, and make a plunge cut. Practice this technique on some scrap wood before trying it for real. Cut each line not quite to the end, finishing the cuts with a jigsaw. Support the baffle on boards, so that you don't saw into the bench top.

One grille option is expanded metal,

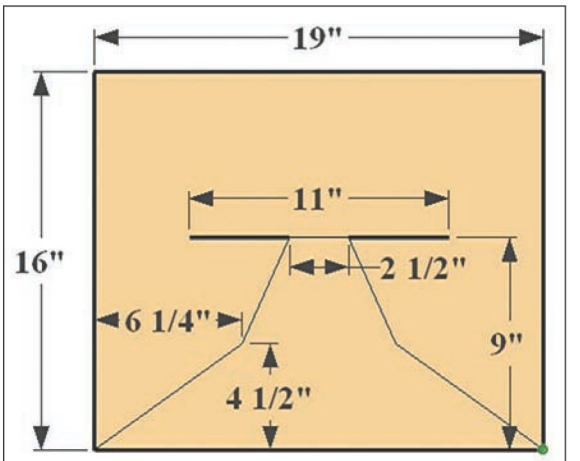

FIGURE 1: Top and bottom layout.

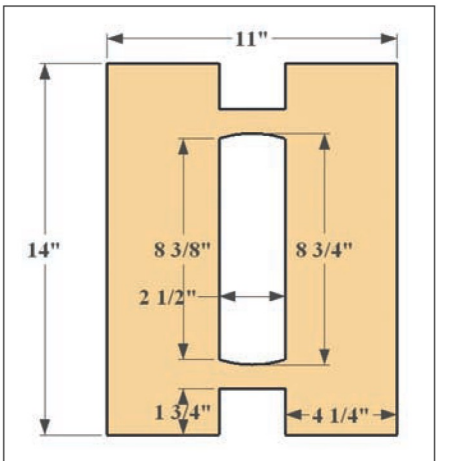

FIGURE 2: Baffle layout.

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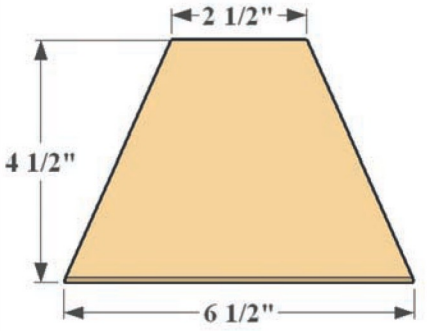


FIGURE 3: Vent ducts.

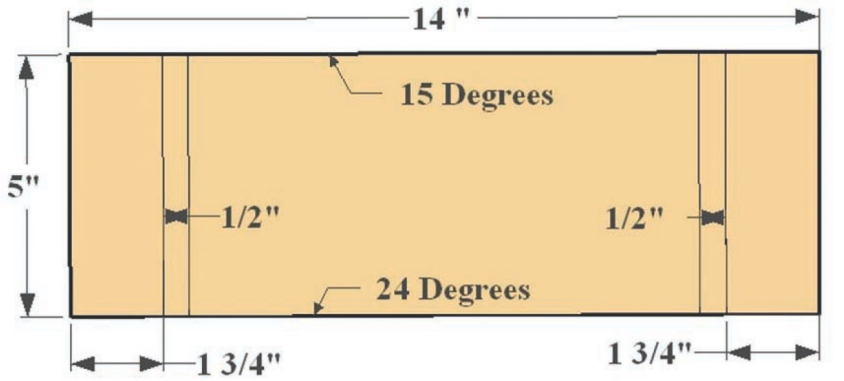


FIGURE 4: Throat panels.

which you can find at many hardware and building supply stores. To put a grille in the horn throat, cut a piece 3 5/8″ wide, slightly shorter than the height of the baffle. Use a hammer to pound the edges flat where the horn plates and vent ducts joint to the baffle. Tack the grille in place with a few staples. Glue with urethane and the expanding adhesive will go right through the grille for an airtight rattle-free fit.

Cut two vent ducts (**Fig. 3**). Ease the leading edges with a sander, just enough to take off the sharp edges. Sand the faces for finishing prior to assembly, because you won't be able to easily reach them afterwards. Cut two throat panels (**Fig. 4**). The edges are both angled, the angles facing opposite directions. The diagram shows the wider face, which is that on the inside of the horn. The standard width is 5″, but if you use a throat mounted expanded metal grille, cut it 1/16″ narrower to compensate for the thickness of the grille. Sand these also before assembly.

Draw joint lines on the panels with the vent ducts, per the diagram. Clamp a short 2 × 2 guideboard to one joint line. *Assembly diagrams show guideboards but not clamps*. Put adhesive on the joint, then clamp a duct plate to the guideboard, making sure the narrow end of the duct plate is flush with the 24° edge of the panel (**Fig. 5**). Fasten the duct plate and remove the guideboard. Add the second duct plate. Using the same technique, add the second throat panel to the assembly (**Fig. 6**).

Cut two side braces (**Fig. 7**). Write on them how they install per the diagram. Clamp a guideboard to the bottom, one edge against the baffle joint line. Apply adhesive to the joint, clamp the baffle in place, and fasten the joint (**Fig. 8**). Use a piece of waxed paper or polyethylene between the guideboard and parts so they won't stick. Leave the guideboard in place for now.

Apply adhesive to the edges of the throat assembly where it will joint the bottom and baffle, including the rear edges of the duct plates. Put the throat assembly in place. Screw a piece of scrap plywood to the top edge of the baffle, pushing the throat assembly tight to the bottom (**Fig. 9**).

From the rear drill a pilot hole through the baffle into each duct plate; drive screws to pull the assembly tight to the baffle. Drive two or three screws through the baffle into the horn panels to pull that joint tight. It doesn't matter if the screws go through the panels; their only purpose is to hold the joint until the adhesive sets, after which you

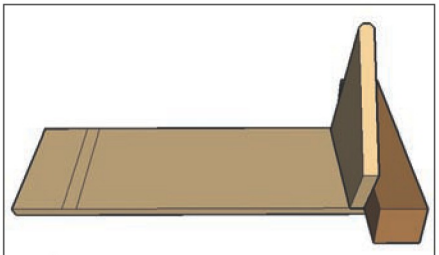


FIGURE 5: Attaching vent duct to throat panel.

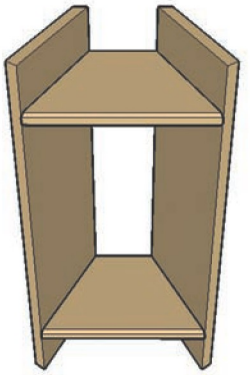


FIGURE 6: Throat assembly.

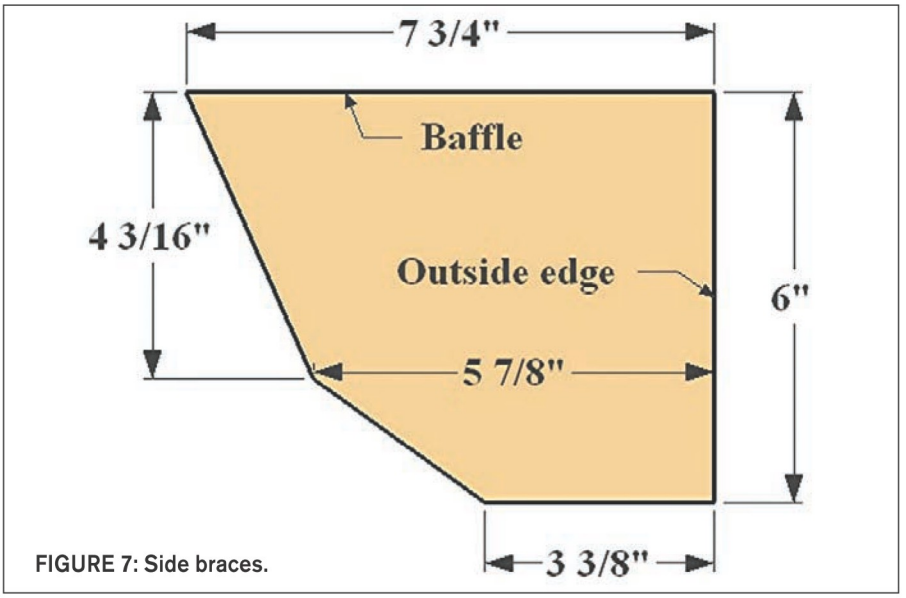


FIGURE 7: Side braces.

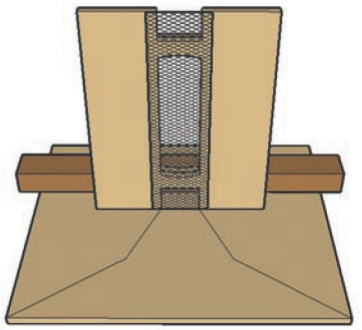


FIGURE 8: Attaching baffle to bottom.

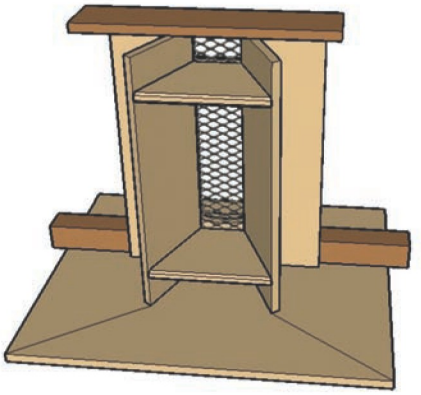


FIGURE 9: Attaching throat assembly.

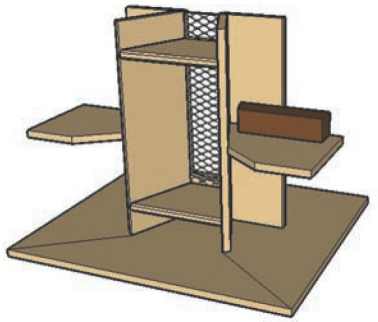


FIGURE 10: Attaching side braces.

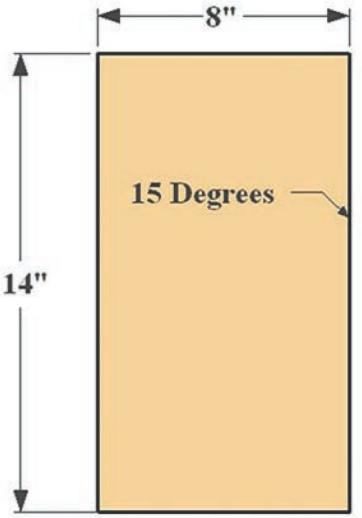


FIGURE 11: Mouth panels.

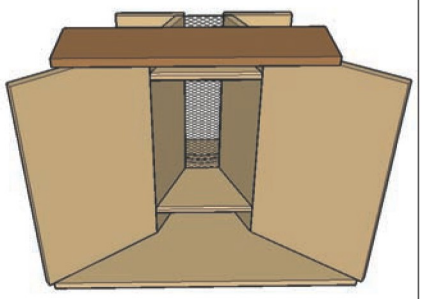


FIGURE 12: Attaching mouth panels.

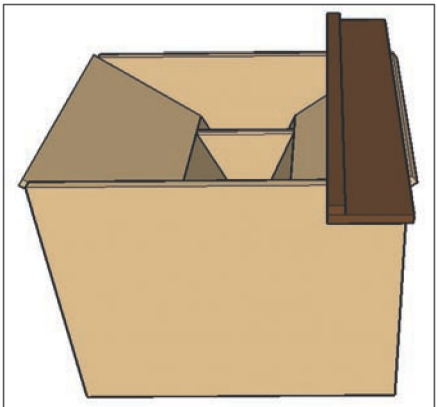


FIGURE 13: Saw sled placement for mouth panel trimming.

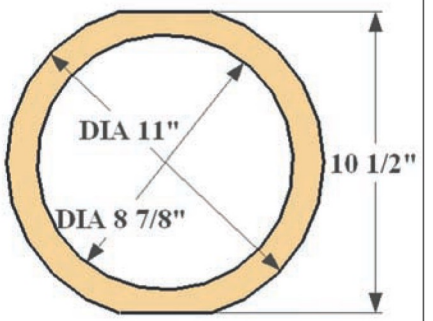


FIGURE 14: Driver spacer.

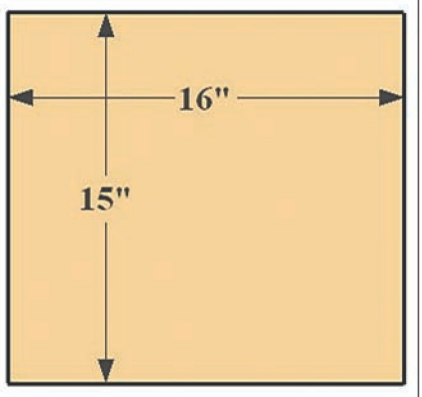


FIGURE 15: Sides.

can remove them and fill the holes with adhesive. Fasten the joints to the bottom then remove the guideboard and scrap jig. Attach the side braces to the assembly, spaced halfway up the baffle. Use a short guideboard and clamps to hold each brace in place for fastening (**Fig. 10**).

Cut two mouth panels (**Fig. 11**). One edge is cut at a 15° angle; the other edge is square. The 15° edges mate to the throat panels. A scrap plywood jig temporarily screwed to the panel edges holds them securely in position while you fasten them to the bottom and braces (**Fig. 12**). Make sure the panels are square and true to the baffle and bottom.

Fastening is not required on the joints between the throat and mouth panels, but be sure the joint is filled with adhesive. If screwing a mouth panel tight to the braces bows the panel inward, leave a gap and fill it with adhesive. The ends of the panels will extend a half-inch or so beyond the front corners of the bottom. Leave the jig in place until you're ready to install the top.

This is a good place to stop for the day and let the adhesive set, so be sure to use waxed paper or polyethylene between the jig and parts so they won't stick together.

ASSEMBLING THE BOX

Add the top to the assembly, using a guideboard to hold everything square and true until after fastening. Put the cab face up, screw a circular saw sled to the cab, positioned with the cutting edge flush to the sides (**Fig. 13**). Trim the excess off the mouth panels.

Cut a driver spacer per **Fig. 14**. Note the trimming at the top and bottom so the spacer doesn't extend over the vents. Install the spacer, centered on the baffle hole. Put the driver in place and use a pencil, nail, or punch driven through the frame holes to mark where to drill holes for 1½″ attachment screws. Place the driver so the screws will not go through into the horn. Put the driver aside and drill the pilots.

Cut and install the cabinet sides (**Fig. 15**). Before attaching them, run a straightedge from the top to the bottom across the braces. If a brace is

proud, sand it down flush. If a brace is shy, apply adhesive to fill the gap between it and the installed side; when installing the side check it with a straightedge to be sure you don't pull it too tight to the brace, causing it to bow. When fastening use a pair of 2′ long pipe clamps or woodworking clamps—one to pull the sides tight to the assembly for fastening, one to align the assembly square to the sides.

Frame the rear of the box with two layers of plywood strips, 1″ wide, producing 1 × 1″ mounting flanges to attach the back. For a painted finish, set the flanges a half-inch from the rear edge. For a carpet finish, set the flanges back 5/8″. Cut the back (**Fig. 16**). A couple of 1½″ holes backed with a piece of ¼″ plywood makes for an easy and inexpensive jack mounting plate. Cut two back braces, half-moon-shaped (**Fig. 17**) to maximize stiffness in the middle of the panel, where stress is the highest. Space the braces 8″ apart on the inside of the back (**Fig. 18**).

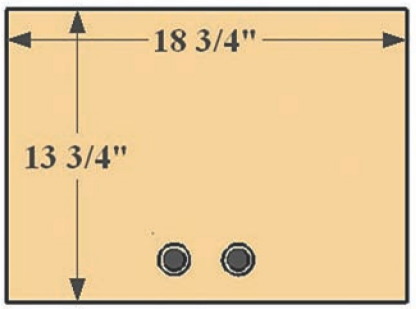


FIGURE 16: Back.

A throat chamber filler improves midrange response. It's made from 1″ thick urethane building insulation foam. Start with a 7½″ diameter disc. Draw a 4″ diameter circle, centered onto one face. Using a sander, taper the edge, producing a "discus" shape. Draw parallel lines 2½″ apart, 1¼″ to either side of center, on the non-sanded face. Remove that section with a saw, either power or hand-held, leaving two half-moon-shaped pieces.

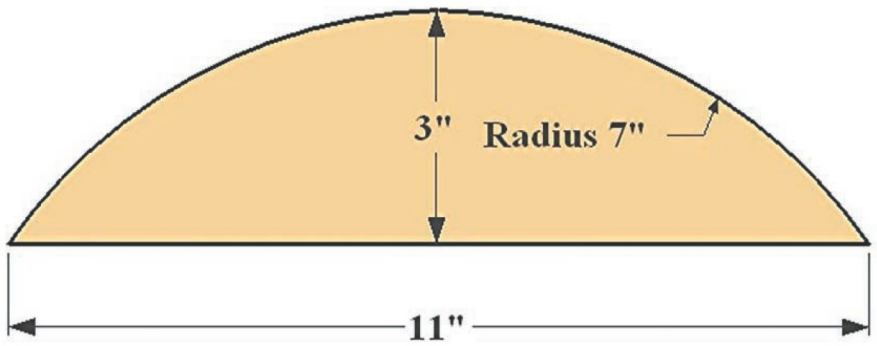


FIGURE 17: Back braces.

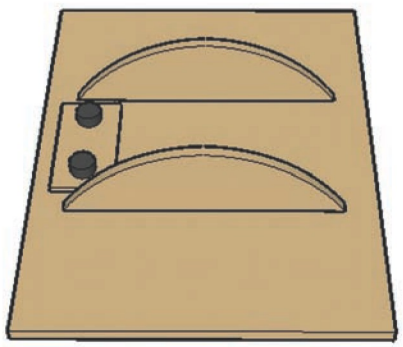


FIGURE 18: Back braces and jack plate installed.

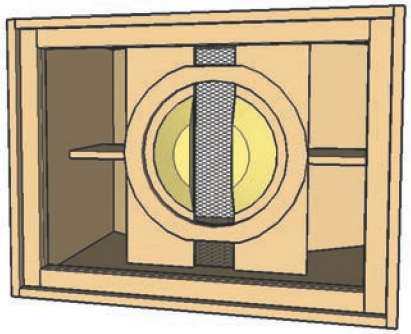


FIGURE 19: Back flanges and throat chamber filler installed.

Sand the smaller face, giving a finished thickness of ¾″. Trial-fit the fillers and driver. Be sure there's at least 3/8″ clearance between the filler and the cone with the driver screwed down tight. Install the throat fillers on the baffle to either side of the throat (**Fig. 19**). Use adhesive and a few screws with large washers, so they won't just go through the foam. Remove the screws when the adhesive has cured.

FINISHING THE BOX

Put the back in place and drill pilot holes for the attachment screws, spaced 4 to 6″ apart. Drill holes in the jack plate for your jacks. Depending on your amp, for instrument use you'll probably need to use phone jacks, while PA may require Speakon jacks. Use both, wired parallel, and you can hook up to any source.

You may use side- or top-mounted handles, flush or surface mount. The center of gravity is approximately on the same plane as the baffle. Cut holes for flush-mount handles, avoiding braces. Drill pilot holes for attachment bolts. The cab is light enough to use a strap handle on the top, screwed in place with 1½″ screws driven through the top into the baffle.

Fill all the screw heads, holes, saw cuts, joint gaps, and defects in the plywood. I prefer polyester auto-body filler for this job, because it sets fast, holds tight, and doesn't shrink. After the filler has set true, round off all the cabinet edges with a router and quarter-round bit if you have one, a sander if you don't. Then sand the entire exterior with 40 or 60 grit sanding discs.

Apply your finish. For best results use Dura-Tex, the same stuff the pros use (available from www.speakerhardware.com). This water-base urethane acrylic goes on with a standard paint roller, drying quickly with a lightly textured finish that's very durable and easily recoated if you ding it. It costs not much more than paint, coverage is quite good, and you can finish a cabinet of this size in about 15 minutes exclusive of drying time. Use a small trim roller to reach tight spots.

The cabinet interior must be insu-

lated (damped) to prevent midrange frequencies from reflecting off the interior walls either back to the woofer or out of the ports. The best material for this job is open-cell convoluted foam, 1 to 2″ thick, widely available in department stores as mattress toppers. Staple or glue it into the box to keep it from shifting. Do not cover the ports. Line the back with foam, avoiding where it will joint the flanges. Do not block the driver magnet with foam. A carpet finish will act as a gasket for the back; otherwise, use adhesive-backed foam weather-strip for an airtight seal.

Install and wire the woofer, going around the bolt circle gradually tightening until the gasket is fully compressed. If you're building the cab for electric bass with no high-frequency horn, connect the wires from the driver chamber to the jacks; install the back.

Many bass players prefer the sound of a woofer only; **Fig. 20** shows the Jack 10 SPL response when loaded with the Deltalite II 2510, and that of the same driver in a vented box of the same dimensions. The difference in the lower midrange, the "punch" bandwidth, is considerable. The trade-off (there's always a trade-off) comes in the upper mids from 1kHz to 2kHz, but because power demand is halved when you go up one octave it's a very worthwhile trade. For those using the cab for PA or full-range bass or keyboards, I'll be back in Part II with the high-frequency horn part of the project. *aX*

PARTS SIZES

These are the nominal sizes of the major parts. If pre-cutting makes them a bit oversized, trim them to final size when installed.

Top, bottom	16″ × 19″
Baffle	11″ × 14″
Throat panels	5″ × 14″
Mouth panels	8″ × 14″
Sides	15″ × 16″
Back	13¾″ × 18¾″

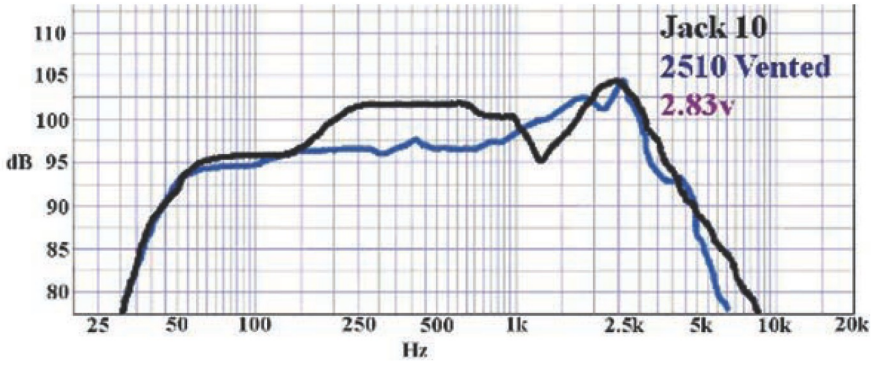


FIGURE 20: SPL Jack 10 versus vented box.

21st-Century Cinema Design

Viewing and listening requirements for a THX-quality theater design.

By Richard A. Honeycutt, Ph.D.

When the first public motion pictures were shown in the early 1900s, the primary concern was that the image be large enough and bright enough for the paying audience to see. Sound in the auditorium was only important in that people needed to be able to hear the organist who improvised theme music to accompany the video. When Warner Bros. introduced talkies in 1926, suddenly the acoustics of the theater became important. Not only did the movie sound need to predominate over projector noise and externally introduced noises, but also the amplifiers and speakers of the day could only provide very small amounts of acoustic power.

By the 1970s, color movies were the rule rather than the exception, Cinemascope® had been introduced, and stereo sound was used in most major theaters. But it was the introduction of Dolby and THX® that spurred the advent of serious home theaters. Because *audioXpress* readers might consider designing their own home theater systems, a discussion of the design considerations for a commercial THX-approved cinema may be of interest, even though these standards may be overkill in a home theater.

THX is commonly associated with high-quality surround sound, but the THX certification process involves optimizing every aspect of the cinema experience.

LAYOUT

The seating plan in relation to the screen depends upon both visual and acoustical considerations. Anyone who watches movies in a commercial cinema knows that sitting too close to the screen produces business for the chiropractor. However, THX also specifies that the viewing angle defined by the left and right screen edges and the furthest viewing position should not be less than 36°,

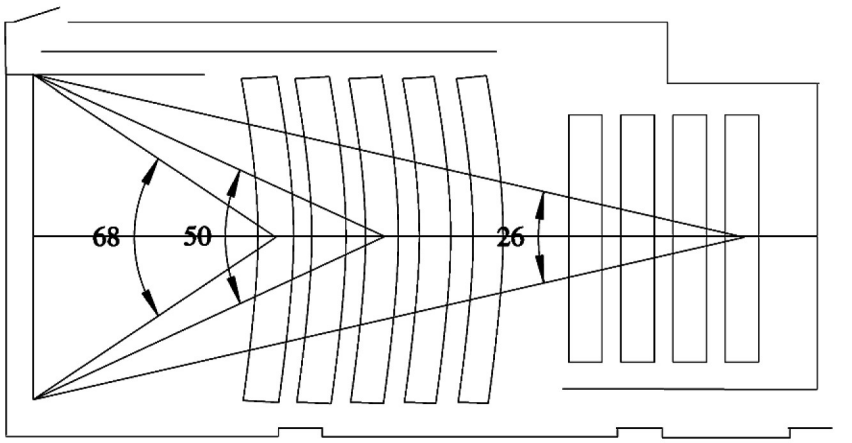
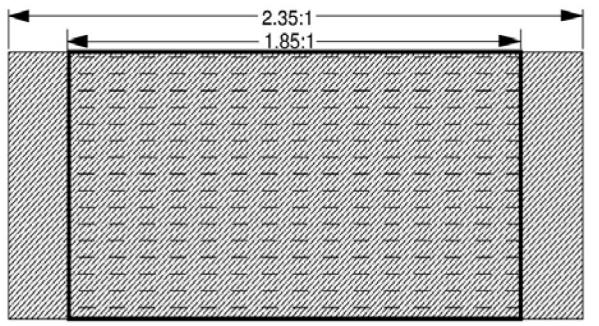
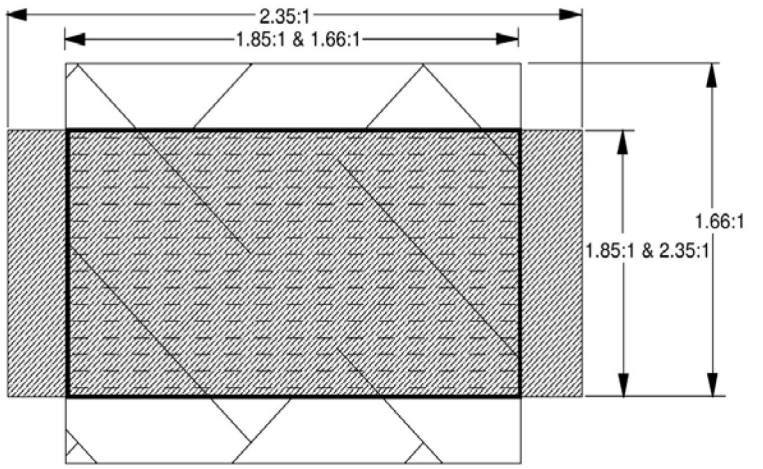


FIGURE 1: Viewing angles.



A. Adjustable masking for 1.85:1 and 2.35:1



B. Adjustable Masking for 1.66:1, 1.85:1, and 2.35:1

FIGURE 2: Masking for different aspect ratios.

and in no case can be below 26° (**Fig. 1**). These specifications ensure an optimal visual and auditory experience for the patrons.

TV viewers have become familiar with two aspect ratios: the traditional 4:3 ratio, and the HD ratio of 16:9 (or 1.78:1). Movies, however, are generally produced in the US with aspect ratios of 1.85:1 (standard) or 2.35:1 (Cinemascope®). European movies may have a ratio of 1.66:1. Most theaters are designed with adjustable screen masking to accommodate various aspect ratios (**Fig. 2**).

For optimum stereo effect, you should place the left and right speakers at the edges of the widest aspect ratio projection area. This means that the masking cloth must be acoustically transparent, because it will partially cover the left and right speakers when a movie of narrower aspect ratio is presented.

You can use black muslin or grille cloth as an insert to the masking cloth, or acoustically transparent masking material such as Harkness 2000M. Even with such cloth, you must take care to avoid covering the front of the loudspeaker's high-frequency section with framing for the masking cloth, or with multiple layers of the cloth. Some commercial theaters have installed fixed screens with 2:1 non-adjustable aspect ratios in order to avoid the cost and bother of adjustable masking. These chop the edges off a Cinemascope movie, and the top and bottom off a standard movie, and therefore good consultants discourage this practice.

In order to keep the sound even front-to-back and side-to-side, the left, center, and right speakers must be placed above listeners' heads: between one-half and two-thirds the height of the screen. If the front rows are too close to the screen, the front listeners will be severely off-axis, and will experience poor frequency response in the reproduced sound.

Virtually all professional cinema speakers have a coverage pattern of 90° horizontal by 40° vertical, so the angle produced by the center speaker and the right and left front

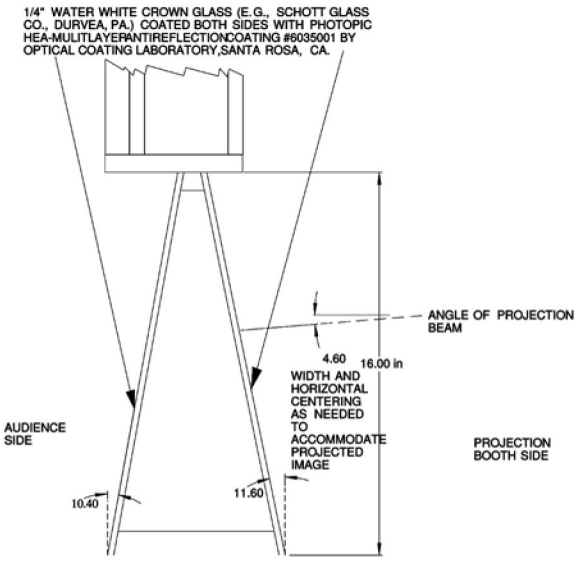


FIGURE 3: Projection port design.

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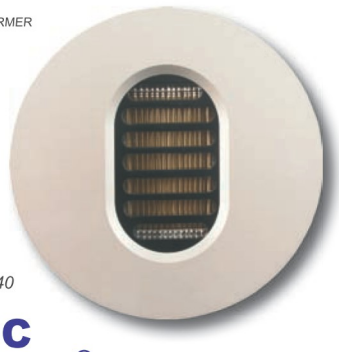

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corner listeners must be less than 90°. Because a 90° × 40° horn typically has 6dB less response at the +45° horizontal and +20° vertical angles, especially at higher frequencies, it is a good idea not to expect the main speakers to cover the extremes of their advertised patterns.

VIDEO PLAN

The THX specification must take into account film projectors, which have no keystone correction (compensation for projection angle not perpendicular and centered to the screen), so there is a maximum image distortion of 5% specified, with a preferred level under 3%. This image distortion would result from the projector being misaligned with the axis perpendicular to the center of the screen. Digital projectors can correct for this distortion.

In order to isolate projector noise from the theater seating area, the projector is mounted in a separate projection room. The screen is illuminated through a double-glass port. The reason for double glass is to insulate against noise.

If an image is projected through glass at any angle other than completely perpendicular, there will be reflected images on the surfaces of the panes of glass, and these will also be projected to the screen, producing a blurred effect. Therefore, the two panes of the projection port should be angled with respect to the axis of the projected beam. Specifically, the glass on the projector side should be angled outward from the projector (farther from the projector at the top) by 7°, and the glass on the theater side should be angled 15° backward (closer to the projector at the top) with respect to the axis of the projected beam. **Figure 3** illustrates a properly designed projection port.

ACOUSTICS

The first acoustical consideration for a high-quality cinema is noise. This can come from outside the building, from other rooms in the building, or from within the cinema itself (Heating, Ventilation, and Air Conditioning noise). Nowadays, most commercial cinemas are built in multiplexes, so that the noise from adjacent rooms is that of another cinema. Except for the rare case of a cinema built under the flight path of an airport serving large aircraft, noise sources outside the building are generally less intense than the sound of an adjacent cinema. Therefore, noise analysis can be greatly simplified in designing a multiplex cinema if all adjacencies are assumed to have the noise level and spectrum of a cinema.

For stand-alone cinemas or home theaters, more care must be given to the noise analysis, because the levels of acoustical insulation specified for multiplex cinemas may well be overkill. However, the greater effort spent in analysis and design can easily be rewarded by reduced construction costs if, as is likely, less stringent measures need be taken to keep noise levels acceptable.

The noise level in a room is specified by the so-called "room criterion" (RC). THX specifies a maximum noise level of RC30, with intermittent increases to no more than RC35, and a preferred long-term level of RC25. **Figure 4** shows the various levels plotted against sound pressure level (SPL). While not as quiet as a classical concert hall or a recording studio, RC25 is still a low noise level. This freedom from noise is needed for the listeners to hear subtle directional cues and other details in the soundtracks of modern movies.

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The noise level in a room depends upon the intensity of the noise source at the outside of the room partition (wall, ceiling, or floor), the acoustical insulation of the partition (specified as "sound transmission class" or STC), and the acoustical characteristics of the room. In a multiplex cinema the noise source and the room characteristics are known, so it only remains to specify the STC of the partition walls. THX specifies a minimum STC 75, which is a very stringent specification, requiring massive walls with resiliently mounted surface layers.

Doors, in general, cannot meet this specification: the best acoustical door has an STC of 53, and this requires a fully gasketed, 4″ thick door. However, because the total area represented by doors is rather small, they do not add a great deal to noise infiltration. Entrances into the cinema are usually through an acoustically treated hallway, to prevent lobby noise from disturbing listeners when someone enters while a presentation is in progress. Noise isolation is one area where the services of a good acoustical consultant are essential in the design of a quality cinema.

HVAC noise is the purview of the mechanical engineer who designs the HVAC system. There are well-known ways of limiting the intrusion of noise from chillers, furnaces, and air handlers through ducts and vents. These methods are routinely applied in the design of new facilities, although they can be quite challenging to retrofit into existing cinemas.

Any perceptible echo in a cinema must be avoided. Considering the high sound level produced by the speakers, you can see that avoiding echoes requires acoustically absorptive wall and ceiling surfaces. Failure to provide enough absorption on these surfaces results in sound reflections that provide false directional cues, spoiling the realism of the surround sound.

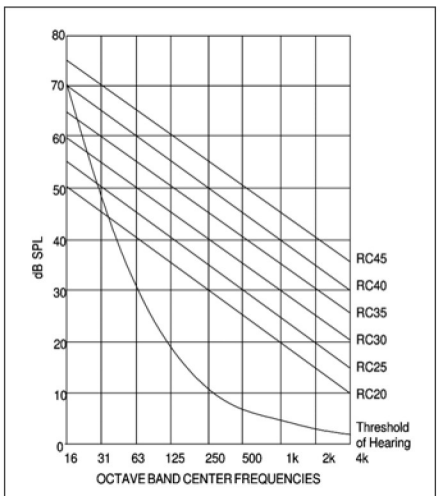

FIGURE 4: Room criterion (RC).

As an example, if the main speakers produce a level of 95dB, and the rear wall of the cinema has a sound absorption coefficient of 99% at 1kHz, the echo produced by sound bouncing off that wall will have a level of 55dB—still quite audible under the proper conditions. The THX guidelines call for 1″ of compressed fiberglass on the sidewalls, over a 1″ to 2″ air space. For the rear wall, the guideline is 2″ of compressed fiberglass over a 2″ to 4″ air space. If acoustical modeling indicates that low-frequency buildup may be a problem, thicker fiberglass and/or thicker air spaces may be required.

The usual ceiling is a 1″ compressed fiberglass suspended ceiling, and if noise from rooms above the theater may be problematic, the suspension should be resilient. A 6″ thick fiberglass noise batt is installed above the ceiling tiles. Cinema seats must have absorptive bottoms, so that unoccupied seats do not add reflecting surfaces when the seats fold up.

Aside from echoes, which are discrete reflections, sound in rooms is also subject to random reflections called reverberation, which causes an apparent prolongation of the sound events. The time required for a sound at the level of normal conversation to decrease to the threshold of audibility is called the "reverberation time" (RT60) of the room. In a THX cinema, THX specifies the target RT60.

Depending upon the volume of the room (cubic feet), the RT60 can be as low as 0.1–0.2 seconds for a 1000ft^3 room to as much as 1–2 seconds for a 1 million ft^3 room. For frequencies below 250Hz, the RT60 is allowed to increase as frequency decreases, to a maximum of twice the mid-frequency value at 31.5Hz. At frequencies above 2kHz, the RT60 can decrease with increasing frequency, to a minimum of one-fourth the mid-frequency value at 16kHz (**Fig. 5**).

B-CHAIN

The electronics involved in cinema sound reproduction are often grouped into two "chains": the A-chain retrieves the sound information from the film or other source

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S/N ratio @ rated W: 90db
Input sensitivity: 1V
Input impedance: 22Kohms
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2.7KHz AP-016B
3.5KHz AP-016C
Weight: 2.6Kgs (5.7lbs)
Dimensions W x H x D:
W: 137mm (5.4")
H: 218mm (8.6")
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Cut-Out W x H:
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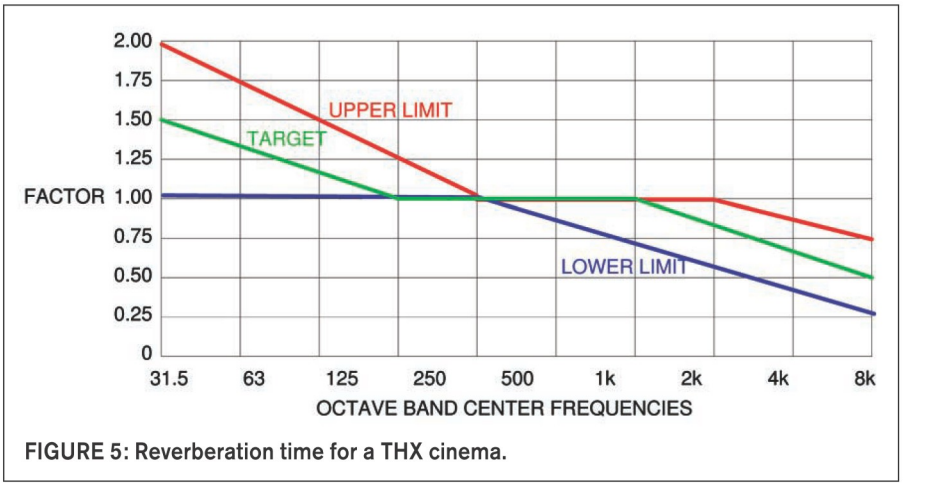

FIGURE 5: Reverberation time for a THX cinema.

and splits it into (typically) main left, right, and center; left surround, right surround; and subwoofer channels. The B-chain includes equalization, active speaker crossover units, power amplifiers, and the speakers themselves. A-chain devices are provided by specialized suppliers, while B-chain elements are offered by a few professional sound equipment manufacturers.

The main (L,C,R) speakers are almost always combination designs using vented low-frequency sections and constant-directivity high-frequency horns. Rather than selecting speakers whose coverage pattern places direct sound as precisely as possible on the seating area, as is done for live sound reinforcement, almost all professional cinema speakers have 90° × 40° patterns. And rather than being aimed at the seating area, the main speakers are aimed parallel to the top of the seating area; *i.e.*, over the heads of the audience. In this way, the decrease in response of the horn for off-axis positions compensates for the proximity of front listeners, giving more even front-to-back levels.

Figure 6 shows a simulation of the levels at 1kHz for the same main speakers, aimed at the seating area (top), and aimed parallel to the top of the seating area (below). Note the more uniform sound levels in the latter case. (The level at the back of the cinema is about the same in either case, but the level in front is noticeably hotter when the speakers are aimed at the seating area. The horizontal aim of the L and R speakers is usually about two-thirds of the way back in the room, centered side-to-side, so as to provide the best stereo effect for all listeners.

In order to prevent early reflections that would be detrimental to directional cues, and to provide loading for the main speakers at low frequencies, the L,C,R speakers are normally mounted in a baffle wall. THX has patented a special design for the baffle wall that optimizes its performance across the full frequency range. The baffle wall must be placed very close to the back of the screen, and must be sturdily built to avoid rattles. The L,C,R speakers should be mounted in such a fashion that vibrations will not be transmitted from the speaker cabinets to the baffle wall, or else extraneous wall vibrations will degrade frequency and transient response.

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FIGURE 6A: Speakers aimed at audience.

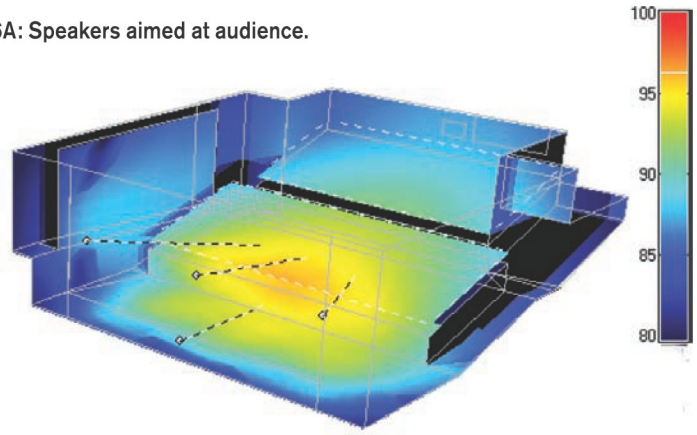
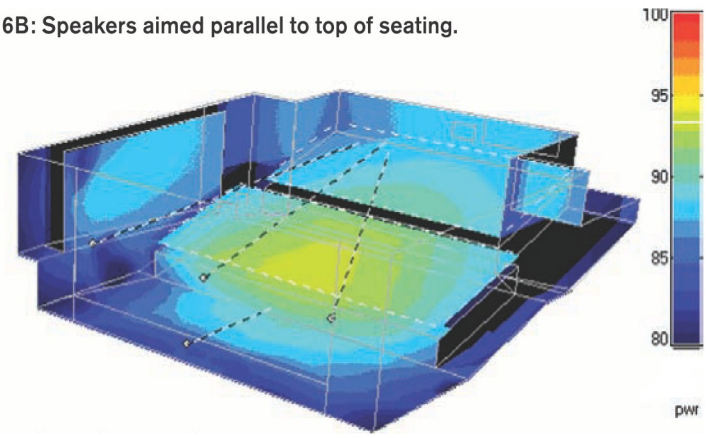


FIGURE 6B: Speakers aimed parallel to top of seating.



The subwoofer (or subwoofers) should be placed in the horizontal center of the screen, under the center speaker. Sturdy, vibration-isolating mounting is even more important for the subwoofer than for the main speakers. Using multiple subwoofers placed at different locations will produce a subwoofer array that will cause hot spots and dead spots in the bass response of the cinema. Only if a qualified consultant specifies spaced subwoofers for particular reasons, after computer-modeling the design, should spaced subwoofers be used.

Surround speakers comprise a left-channel and a right-channel group. The speakers are spaced closely enough that listeners cannot localize the sound to a single speaker: they are to act as a continuous source of surround audio. The surround speakers are located above the heads of the listeners, starting about halfway back on the side walls. If the surround speaker arrays extend farther forward than this, their output tends to mesh with the L and R main speakers, confusing the ear.

Typically, the surround speakers are mounted 10–13′ above the floor, and they are aimed at the farthest listener, although

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Qts	0.40
Vas	18.9 ltrs
X-max	3.5mm
Sens.	87dB
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Qts	0.31
Vas	24.3 ltrs
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the aiming angle can be varied as needed by the consultant in order to provide the most even coverage. Usually the surround speakers are wired in a series-parallel arrangement in order to present a suitable load to the surround power amplifiers.

In a THX design, part of the THX engineering service is to present the owner with a list of THX-approved speakers, amplifiers, and processors from which to choose the cinema equipment. For non-THX designs, the consultant should design for a maximum SPL of 103dBC (for Dolby® SRD) for each screen channel (L, C, R). This SPL capability (18dB above the reference level) should be available at all frequencies from the 31.5Hz band through the 16kHz band. The reference level is measured at a reference seat two-thirds of the way back in the room, but levels should be consistent within 3dB at all seats. Each surround chain (left and right) should be capable of producing 100dBC at the reference seat.

You must choose speaker wiring to avoid line-heating problems or excessive signal loss, and to provide adequate speaker damping. Fortunately, if the line loss is kept under 0.5dB, the other two requirements will be met automatically. Assume that a subwoofer requires a peak power of 800W, and has an impedance of 8Ω. The sub is located 50′ from the power amplifier. The total resistance of the wiring and speaker is then given by:

$R_T = |Z_S| / [\log^{-1}(-0.5\text{dB}/20)] = 8.47\Omega$

where R_T is total resistance, and $|Z|_S$ is the speaker impedance.

This means that each side of the line must be under 0.235Ω, so the wire must have a resistance of 0.235Ω per 50′, or 4.7Ω per 1000′. AWG 17 wire meets this specification. The speaker will draw about 11.2A maximum.

This is below the allowable ampacity (14A) for three AWG 18 wires in a raceway; ampacity for AWG17 is not specified, but it would be higher. The damping factor is given by:

$DF = |Z|_S / (R_{source} + R_S)$

where R_S is the DC resistance of the speaker (assume 6Ω).

Neglecting the very low output impedance of a solid-state amplifier, damping factor is about 1.24. Even with zero resistance in the lines, the damping factor could never be below 1.33, because of the DC resistance of the voice coil[1]. Thus designing for a maximum loss of 0.5dB will lead to a very good choice of speaker wire gauge.

TESTING AND ADJUSTMENT

In most modern cinemas, equalization is provided by digital signal processors, with the target response defined by the "X-curve" specified in ISO Document 2969 (**Fig. 7**). You can make adjustments using a single measurement microphone placed off-center in the seating area. However, better results are obtained by using multiple microphones located as shown in **Fig. 8**.

Mount the test microphone(s) about 9″ above the tops of the seat backs. You should use calibrated microphones, and the four microphones can be used one at a time, and then the readings can be manually averaged; or you can feed the mikes to a microphone multiplexer that automatically switches among the mikes. The time averaging function of the real-time analyzer then performs averaging among the microphones.

In addition to equalizing the main and surround speakers to the X-curve, the consultant performing the final adjustment must set the sensitivity of the power amplifiers for L, C, R, subwoofer, and surround. When the system is fed pink noise at the reference level (such as from Dolby Cat. #85 pink noise generator), each of the main (L, C, R) speakers should produce a level of 85dBC, and the left and right surround channels as a whole should each produce 82dBC. The subwoofer level is set 10dB hotter, due to the recording standards for subwoofer soundtracks.

SUMMING UP

Even though the design is significant and the resulting project may be expensive, cinemas designed to or near THX standards can provide an unforgettable viewing/listening experience. *aX*

REFERENCE

1. For an excellent discussion of damping factor, see the indexed references in *Audio Cyclopedia*, Second Edition, by Howard Tremaine. Howard W. Sams, 1969.

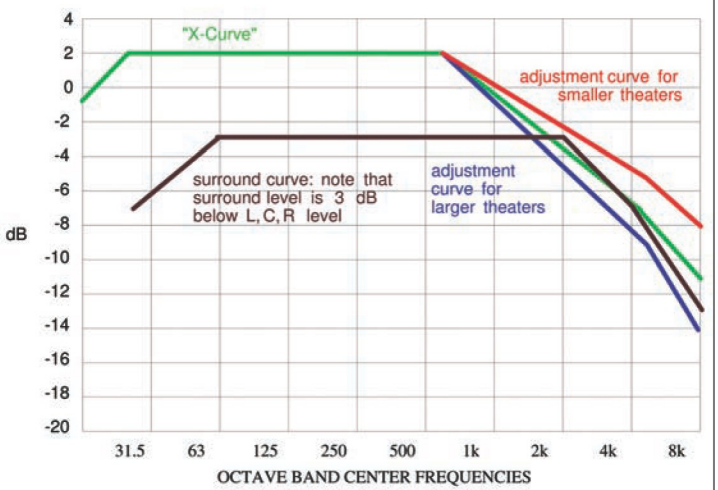

FIGURE 7: X-curve and surround curve.

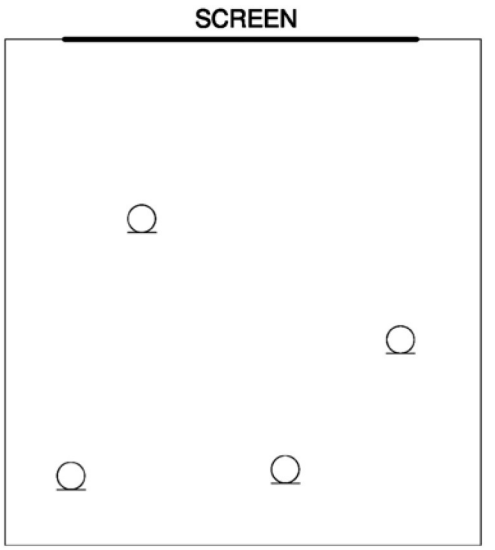

FIGURE 8: Test mike positions.

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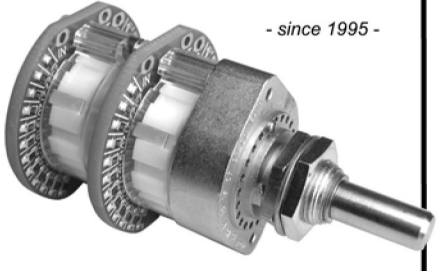

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Jantzen Audio Denmark	19
Jensen Transformers	18
KAB Electro-Acoustics	45
Liberty Instruments Inc.	42
Linear Intergrated Systems	45
Madisound Loudspeakers	31
Midwest Speaker Corporation	30
MISCO/Minneapolis Speaker Co.	CV3
Mouser Electronics	39
Mundorf EB GmbH	13,27
Parts Connexion	9
Parts Express Int'l., Inc.	CV4
PCBCart.com	46
Rocky Mountain AudioFest	7
Saelig Co.	37
Sanus Systems	21
SB Acoustics	23
Solen, Inc.	29
Tang Band Industries Co.,Ltd.	25
TheTubeStore Inc.	8
Vidsonix Design Works	15

CLASSFIEDS

All Electronics	33
Audio Classics Ltd.	33
Billington Export Ltd.	33
Borbely Audio	33
Design, Build, Listen Ltd.	33
ENG Vista, Inc.	33

Pioneer X-Z9 Compact Stereo System, Pt. 1

By David A. Rich

Elite product line
www.pioneerelectronics.com/PUSA/Products/HomeEntertainment/AudioComponents/Hi-FiAudio/X-Z9
$1799 complete system

Speaker
Coaxial midrange and tweeter: 13cm midrange and 3cm dome tweeter
Woofer: 13cm with vented bass loading
Tweeter: metallic
Midrange and woofer: aramid woven cloth with aluminum die-cast frame
Crossovers: 500Hz and 4kHz
Dimensions: 8 ⅓" (W) × 14 ⅝" (H) × 12 1⁄16" (D)
Nominal impedance: 4Ω
Weight: 21 lb 2 oz

Combined receiver and Super Audio CD with inputs for the Internet connection, USB, iPod, and Satellite radio (XC-Z9)

Amplifier
Switched mode without feedback. Digital input and time quantized 1 bit output.
Continuous output power (both channels driven) 40W average power per channel into 20Hz to 20kHz THD 1.0%

Features
Sound retriever for enhanced playback of compressed source material
Digital noise reduction claimed to be useful for cassette or videotape
Full Dot White OEL Display Organic Electroluminescence
Headphone output from switching amplifier (mini jack)
Sources that produce digital data remain digital until switching amplifier's output stage

Music Sources
Internal SACD/CD-R/RW player top panel controls
Internal AM/FM tuner (fully analog IF and with no HD radio support)
Phono input
Analog line in/out on back of unit
Analog line input (mini jack) on front of unit
iPod player connection (analog connection requiring ADC to convert signal back to digital. Support for selected operational function through players TX and RX digital control lines. See instruction manual for details.
XM® Satellite Radio Ready (direct digital output)
SIRIUS® Satellite Radio Ready (module is analog out requiring ADC to convert signal back to digital)
Internet radio using vTuner streaming database
USB interface to playback audio files stored on a USB memory device
DLNA® certified: IP network compatibility enables a link with a networked PC to share your music Microsoft PlaysForSure™ compatible (version 2.01)

Playable file formats: MP3, LPCM, WAV, WMA, AAC, and FLAC
From the user manual: "Some formats are not available for playback although they may be listed as playable file formats. Also the compatibility of file formats varies on the type of server." For complete information on extensions, stream types sampling rate, quantization, bitrate and VBR and CBR support, see the instruction manual. The manual also covers potential playback behavior problems over the network and troubleshooting network problems.

One shipping container has two speakers and an "all in one" SACD/receiver combo unit. The receiver has an extra PC board that adds network capabilities and Internet radio, iPod connectivity, the ability to read audio data from a USB stick, and satellite radio connectivity (Sirius and XM). As with all equipment of this type, it is best to get a money-back guarantee with no restocking fee from the vendor to ensure the unit works with your hardware and software before committing to it.

XC-Z9 Single Box Electronics

The XC-Z9 has a high-tech look with touch controls for the SACD player on the top of the unit. A nicely finished metal top covers the rest of the unit. High-quality plastic replaces what formerly were wood side panels on Pioneer's flagship Elite units. High-efficiency amplifier and power supplies obviate the need for vent holes around the unit, with the sole exception of one at the bottom of the right panel to vent the power supply's fan. For those who enjoy tinkering, I caution you not to take the unit apart without the service manual, lest you find yourself unscrewing PC boards and breaking plastic clamps on the side panels.

The X-Z9 might be the ultimate executive stereo system, a sentiment not lost on the Japanese website, which dutifully portrays this application. The Japanese site holds far more information on the unit than other countries' websites. There's no English to be found, though, on the Japanese site. I got by with the instruction manual, the service manual, and the Babel Fish Translation program of the Japanese website. Unfortunately, Babel Fish only translates text characters at a rudimentary level. Much of the Japanese website has characters that cannot be read as text, which is the only thing Babel Fish can translate; I was able to translate only about half the pages. Many times the exercise left more questions than answers.

OPERATION

Skipping the networking functions for the moment, the key feature is the all-digital path from the SACD/CD player direct to the one-bit digital switching amplifier, which functions as the digital-to-analog converter in the signal chain. Op amps, bypass capacitors, switches, and potentiometers in the signal path are all gone. Hurray! Not only is the signal path cleaner (assuming the DSP code is done correctly), but it is also more reliable than most two-channel equipment.

> "The X-Z9 might be the ultimate executive stereo system"

Traditionally, mechanical switches and potentiometers are at the bottom of the reliability league tables for audio electronics, although these won't take the unit down like a problem in the power supply. Mechanical controls and switches have long been banished from AV receivers, where CMOS switches have replaced them. Lower down on the evolutionary chain, most two-channel units, however, still have them.

Any control knob on stereo equipment with physical stopping points is routing your audio signal through metal touching carbon. In contrast, the rotational sensors common to AV receivers and used on the XC-Z9 only convey information to a microprocessor.

Tone and balance controls on two-channel equipment have stops and are potentiometers. Beyond reliability, you pay in channel separation and tracking errors between channels. Ugh! AV receivers use digital cables (S/PDIF and HDMI) to connect or source to the receiver, putting stereo equipment in the Dark Ages given the analog connection from the SACD player to the receiver: The DAC in the SACD player box connects via cable to the analog input of the stereo receiver. Analog electronics must drive the cable in the SACD player, mute the output on, turn on and turn off, and protect the electronics from damage were someone to mistakenly connect a line output from the receiver to the SACD player's output. At the input of the receiver, it is necessary to buffer and protect. While cables are transparent, dirty plugs and jacks can degrade signals.

To determine how the unit sounds with respect to the digital path (excluding the analog switching electronics that drive the speaker), I sourced signal out of the headphone jack (which has its analog signal derived from the same one-bit switched signal that drives the speaker's power amp) and connected it into my $6000 NHT Xd DSP active loudspeakers. I heard no difference compared to my reference electronics.

The analog electronics in the power amp for converting the one-bit switched signal into analog are a different kettle of fish. The XC-Z9 electronics are designed to drive the S-Z9 speakers. That gives the designers knowledge of the speaker load, and the power needed to drive the speakers to their limits. The 4Ω S-Z9 speakers allow for maximum power transfer from an amplifier with the low power supply voltage rail of 21V used in the XC-Z9. The output of the XC-Z9 electronics is balanced, so the peak-to-peak voltage across the speaker terminal is 42V, ignoring losses in the switching transistors. Assuming a 2V loss in the switching transistors, the XC-Z9 would deliver 45W average power.

Pioneer specs the unit at 40W at 1% THD. The power transistors and drivers

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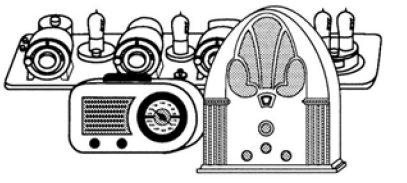

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can handle a larger supply voltage and produce more power, but thermal issues in the power supply and the power amp—when placed in a small enclosed area with only a tiny fan—limit the maximum power supply voltage. The power supply would grow in size were the supply rails to increase, as the power supply would be required to sink more current as the voltage at the speaker terminals increases. The incumbent fits snugly in the box. Clearly this is an element where the aesthetics of the XC-Z9 took priority to maintain the box's compactness and hide the ventilation.

The limits on the maximum power the speaker can handle also come into play. The S-Z9 speaker has been tuned to 45Hz despite its 5″ woofer. I could easily drive the speaker to 20% distortion at 50Hz without clipping the amp in the XC-Z9. I heard lots of port noise as well. Hoffman's (the H in KLH) iron law states that if you want low bass in a small box, you pay with an inefficient design. The little 5″ driver can handle only so much. Above 150Hz the amp places the limit on the SPL.

Because the power amplifier is a switching amp, all sorts of design issues limit distortion performance. For the XC-Z9 THD at full power, Pioneer specs it at 1% from 20Hz – 20kHz. Because the XC-Z9 is customized for the S-Z9 speaker, I did not audition the XC-Z9 with other speakers. With an estimated 20W into 8Ω, you would not need double-blind conditions to hear trouble approaching. Even if power levels were unconstrained, the mismatch of the internal passive analog output filter (required to prevent the digital switching noise from exceeding FCC limits when connected to the speaker cable, which would act like an antenna) into an 8Ω load would result in a rising frequency response at high frequencies. This is a problem with all switching power amps that lack feedback around the output stage.

A class D amplifier from D2Audio (not used in the XC-Z9) actually does a precalibration, measuring the frequency response at the output terminal and correcting the errors in its DSP engine. Hence, I have unlocked the justification for squeezing the XC-Z9 and the S-Z9 into the same box: impedance matching.

DESIGN

Switching to ergonomics, there is no need to fret about universal remote controls with all the electronics in one box. Do not lose the remote, which has over 50 buttons (all but one the same color and most the same size and shape). All you can control from the front panel is the SACD player and switching the inputs.

As a stereo unit, there is no television to display complex graphic user interfaces. To display the information needed for such complex functions as Internet or satellite radio, Pioneer has co-opted the display technology from its automotive division and supplied a Full Dot White OEL (Organic Electroluminescence) design. The design is so well structured you can get more information out of it than from the high-tech GUI of an expensive AVR on your TV screen. For instance, with XM satellite radio, you get the channel number, channel name, what the piece is, and who's playing. Classical music information is limited, but I think that may be a function of XM.

A button click on the remote shows the signal strength of both the satellite and terrestrial feeds. Now you can orient the satellite radio antenna correctly! I did not have a Sirius radio front end, so I do not know what is displayed when using that service.

Strangely, the title and track information for a CD or SACD are not displayed. I thought I had the unit in some special display mode, but I found no way to get out of it. Maddeningly, the instruction manual has not a single picture of the front panel display. This sometimes leaves one to self-discovery by hunt and peck on the remote control and watching the display, even though most key press sequences are discussed in the text. Given the complexity of this unit and its $1000+ price, a 45-page instruction book with minimal graphics is unacceptable.

The networking functions in the Yamaha 2700 networked AV receiver I tested a couple of years ago consumed lots of ICs, board space, and money. Pioneer takes advantage of the next-generation Bridge Co DM850E Media Networking Processor and its associated software. On board are a media processor, an ARM microprocessor, Ethernet Media Access Controller, memory controller, USB2 interface, digital audio interface, and something called a Security Engine. The missing elements are some FLASH ROM, an SRAM, and an Ethernet chip that does the processing to interface the Ethernet cable to the Media Access Controller on the Bridge company chip. The hardware would be useless without the software and firmware supplied by Bridge. Some of the Bridge software can interface DM850E with the iPod.

One worry here is technical support from Pioneer. Suppose the roadblock is the Bridge code when interfacing your peripherals to the XC-Z9. You can envision much hand-wringing as the tech-support person at Pioneer blames your peripherals when the problem is deep inside the networking card. That said, I had no need to call tech support.

I had no problem getting the Ethernet to attach to my simple network running the Internet radio. I must be the only person in the country without an iPod (compressed audio over cheap headphones is not my thing), so I have no idea how well the digital interface works. Because many different iPods exist, definitely test your iPod functionality at the store with the unit. Also bring along a USB drive to make sure the XC-Z9 reads your data. (As an aside, the iPod audio is transferred as an analog signal to the XC-Z9. The bridge chip only interfaces to the digital control signals of the iPod. Some recent products bring iPod audio in digitally, but it is not clear whether Apple supports the feature.)

You cannot test how well your computer's hard drive networks to the XC-Z9 to download music files in a store, which is why you need to have plan B in your back pocket to return the unit. I did not test the download functionality because I do not have any music files on my computer. Can you imagine how much room on your computer it would take to store the complete works of Mozart as an uncompressed file? With so many classical uncompressed CDs free of digital rights management at Berkshire Record Outlet priced equivalently close to a download, I am hard-pressed to find a reason to start downloading. I'm interested to learn how many readers (not your children) are downloading files to a computer.

The Internet radio uses the Vtuner server. The Vtuner website describes what is streamed. Navigation with the high-resolution display of the S-Z9 is a breeze within the limitations of the six-button navigation system on the remote control (up, down, left, right, enter, return) and Vtuner's configuration of the stations.

TERRESTRIAL AND SATELLITE RADIO RECEIVERS

The analog tuner is a simple design. Good tuners in automobile front-ends and table radios with HD radio digital signal processing receivers will pick up many more stations cleanly. That said, the XC-Z9 was more sensitive than tuners I have tested in AVRs with price tags above $2000 (in general, the same tuner module that goes in the $200 AVR goes in the $4000 one). I connected a big indoor antenna with directional control (AudioPrism 8100) and found more stations coming in cleanly than I would have expected. You can force mono but cannot defeat the stereo blend function. I suspect the blend may be aggressive, which accounts for the low background noise on what I think are signals with marginal levels going into the tuner.

Despite the high-tech display, there is no signal strength indicator for FM. A little antenna does appear with three curved bars around it, reminiscent of the signal strength indicator on your cell phone. At first I thought it was a signal strength indicator, but the number of bars never changes, and the user's manual is silent on this issue.

If you use the piece of wire supplied with the unit that they call an FM antenna (they do not even supply a dipole antenna), then all bets are off regarding your reception. Accordingly, you may find yourself using the Internet radio for all but local signals. Note that all the stations you want may not be available from the Vtuner service, or the station may send only a low bit rate signal. In that case, buy an external tuner. HD tuners with remarkable performance start at $99, and some, such as the Sony XDR-F1HD (see review, *aX* 8/09, p.34), have very small form factors.

The XC-Z9 has something called Sound Retriever to enhance the sound of a source recorded or streamed at a low data rate. It is implemented in a DSP chip in the digital signal path of the XC-Z9. The instruction manual devotes fewer than 50 words to this function and when to use it. The Japanese website is more informative if you can read Japanese. A graph on the site which needs no translation shows that the system generates higher energy spectra at the higher end of the audible band. I found it effective for very low bit rate signals (less than 64kbits/sec). It also worked for higher bit rate signals that had been badly encoded—a common problem with Internet radio stations. The stations that do a good job of encoding and streaming a signal do not need the processing.

XM satellite radio signals could benefit since the XM PC board in the SV-Z9 supplies a digital signal directly to the digital signal bus connected to the DSP processor. The first DSP processer stores the code for the Sound Retriever. XM appears to use a significant amount of companding on some stations, so the Sound Retriever did improve the sound quality on many channels.

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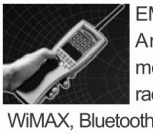

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My impression of the enhancer when dealing with low bit rate signals is that the high-end is extended, but the bass is also pushed for reasons I do not understand. Engaging the enhancer may increase the absolute level of the signal. A number of signal-processing algorithms can enhance compressed digital data. I do not know which method Pioneer adopted.

Strangely, you can enable the enhancer with analog inputs such as a tuner and even CDs. With analog signals it must be a totally different animal, and not one I found I would want to invite into the house, as it never improved any analog signal. In analog mode the frequency response was within 1dB of the response I got by setting the bass and treble controls to maximum below 10kHz. Above 10kHz the Sound Retriever returned to 0dB. I thought the Sound Retriever response would change with signal level or signal spectrum, but changes in input gain made no difference in the shape.

Going from making measurements by sweeping a sine wave to using an MLS noise as the input signal did not change the frequency response of the Sound Retriever. I assume the enhancer in analog mode is totally different from the digital enhancer. Remember that the iPod and Sirius radio signal come in analog. A DSP enhancer cannot enhance anything without seeing the compounded bits. The DSP needs to be part of the process of converting the compacted data back to PCM, implying the signal is truly "enhanced" through improved decoding algorithms.

The unit's "Sound Modes" are called "vivid" (producing a lively sharp sound is what the instruction manual says it does), "exciting" (dynamic, stimulating sound), and "relaxing" (laid back sound). This appears to be similar to the options in some car stereos. I can understand the function with road noise and cheap speakers placed sub-optimally in a car, but the X-Z9 design touches the state-of-the-art. The only sound mode you want is Off!

The loudness contour did not change with volume position; it is the same as the Exciting Sound mode to 1kHz. They can be activated together bringing the bottom end up 14dB! At least the unit is smart enough not to allow the bass control to push this up any further.

INSIDE THE BOX

Inside, the unit is stuffed with electronics. The dominant item is the switching power supply encased in a metal shield to reduce RFI. The shield is 8.5″ × 5.5″ × 2″. The power supply dominates the weight of the unit and requires a fan to cool it. The main PC board, containing most of the analog and digital electronics, is on top of the power supply and is almost as big as the top of the power supply. This board is double-sided with throughholes, as are the other PC boards in the box.

On the side of the power supply is the 2.5″ × 4″ network board. The satellite radio board is about as large as the back right side of the unit. The power amp is on a 4″ × 7″ board toward the left rear. More than half the board holds the analog LC filters that remove the high-frequency noise from a switching power amplifier.

Because the inductors must pass the entire signal to the speaker, nonlinearity effects such as saturation of the inductor under large current flows could result in distortion. The inductor looked like a standard component, but the Japanese website appeared to highlight the design. Unfortunately, I could not figure out what they were talking about using the translation software. Because the outputs signal is balanced, two inductors are required for each channel. Pioneer winds the two inductors on one ferrite, forming a transformer that increases the effective inductance and may reduce even order distortion from the inductor.

The heatsink for the power amp is microscopic at 2″ × 2″ × 4″. The dominant heat producer is the switching power supply, not the power amp. As I said before, the power supply has a fan.

The SACD player lies ahead of the power amp board and to the right of the power supply. The electronics for the SACD player lie under the mechanicals for the player. Dwarfed in size is an all-analog 1980s technology tuner. It is all of 2″ × 2″ × 1″. I assume a DSP-based HD tuner would add about $50-$75 to the price of the unit, so Pioneer probably figured this could not be justified with all the network, iPod, USB, and satellite radio capabilities.

The switching power supply has a similar design to your desktop PC. The design details of a switching power supply are beyond the scope of this review; the basics are on the web and details of making one work properly are voluminous enough to consume complete textbooks. See the sidebar for a quick executive summary.

ICs

Housing all the electronics in one box allows for a unique signal flow. The primary flow is purely digital audio up to the switching transistors of the power amp. The SACD player's digital output (converted to PCM from the DSD stream), along with the digital output of the digital media board, and the XM radio module are selected and sent to a Freescale DSP processor (DSP56374), which may just do no more than the "Sound Retriever" function (I do not have any details about the code in the Freescale DSP), although it is likely a part of the digital volume control. The Digital Noise Reduction function must also be done here. The DSP56374 is a fixed-point 24-bit processor with the ability to perform 150 million multiple accumulates per second.

The signal then enters another DSP processor associated with the switch power amp. It converts the PCM audio signal into the one-bit switching signal that will eventually drive the speakers. The integrated circuit is the TI TAS5504A. In addition to the computa-

tions to develop the one-bit switching stream, the DSP section provides seven orders of bi-quadratic filters, with two operative for the tone controls in the XC-Z9. The sound mode and loudness function may use some of the other bi-quadratic filters sections; I assume most are not used and could be saved for room correction (discussed further). The TI chip also provides functionality for the digital volume control.

The TI TAS5504A drives a pair of TAS5142 chips (left and right). The TAS5142 steps up the low-level digital signal voltage from the TAS5504A to a level that can drive the power transistors' gate. The power transistor's switch lies between the power supplies (21V or ground) and the speaker terminals. The power transistors are in a balanced bridge configuration to reduce distortion and are included as part of the TAS5142.

As discussed, the output of the power transistors drives the second order LC filter connected to the speaker terminals. One TAS5142 chip contains four channels of switchers, which would allow for a balanced stereo pair in one chip. The XC-Z9 uses one TAS5142 for each channel. They parallel two channels of the TAS5142 to reduce voltage drop across the switching transistors when high current is flowing to the speaker.

Class D amplifiers are distinguished by the fact that the output signal switches, in most cases, between two levels. Class D amplifiers can be classified into many sub classes. The most significant difference is the nature of the output signal. Because the output can have only two levels it is the amount of time that the amplifier stays at one level that determines the analog output voltage after filtering.

In the oldest form of Class D amplifiers the time that the amplifier can hold the output at one level can take on any value. After filtering, this type of class D amplifier can produce an output voltage of any level. It should thus produce a time varying waveform without distortion ignoring distortion from real semiconductor

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switches. One limitation of real switches is the finite rise and fall time. These set the minimum time a level can be held.

In another embodiment of a class D amplifier the switching time may only be allowed on a clock edge of a digital clock signal derived from a crystal. In this case the time that the amplifier can hold a single value is quantized. The smallest time a value can be held at is one clock cycle and each larger time interval is a multiple of the clock cycles. The time that one level can be held is thus quantized. This leads to distortion of the recovered analog signal even with an ideal back end.

If a class D amplifier is designed to accept a PCM digital input directly and has a quantized in time output then it can be designed using just digital circuits. The TI TAS5504A is an example of this type of class D amplifier. The TAS5504A also includes an internal digital signal processor that anticipates the distortion at the output from the time quantization and tries to correct it by pre-distorting the signal before it is sent to the all digital Class D amplifier.

The most significant issue with this design is its lack of feedback from the analog output to the processor. This means the amplifier has no power supply rejection and cannot reduce distortion caused by the non-ideal switching of the power transistors. This explains the unit's high full power THD in comparison to class AB amps. Without feedback, the amp's frequency response is sensitive to the speaker's load. The fact that the X-Z9 includes speakers resolves this problem.

The TAS5504A family was released in first-quarter 2006 and has innovative nonlinear signal processing to work around some fundamental problems when crafting an all-digital switching power amp without feedback. The innovations enabled a reduction in the near baseband high-frequency noise associated with the square-wave switching signal while simultaneously dealing with the issue of limiting the number of times the output signal switches state. Energy is used every time a switch occurs to move the filter connected to the speaker terminal from 0V to 21V. In addition, energy is lost when the two switches are on momentarily at the same time causing current to flow from the 21V supply into ground.

While the TI chips perform well, the newer D2Audio chips have less distortion than the TI feedback-free designs. This is accomplished with digital feedback from the speaker terminal. The D2Audio design also has a digital power supply feed-forward correction system that allows the chip to reject some noise on the power supply rails. I do not know the relative costs of the D2Au-

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SWITCHING POWER SUPPLY DESIGN

In a switching power supply, the 60Hz incoming AC is rectified to DC and filtered lightly. A switching transistor is connected between one side of the rectifier's output and one side of the primary of a transformer. The other side primary is connected to the other DC voltage at the full-wave rectifier output. The switch is opened or closed at a high frequency.

The circuit that controls the switching transistor gets feedback from the power supply's output voltage developed on the secondary side of the transformer. The time the switch is activated depends on the current draw at the supply's output. The switching action results in a time-varying voltage with a typical average frequency of 50-200kHz. Large transformers are required to efficiently pass power from the 60Hz power line to the secondary in a linear power supply. In a switching supply the transformers can be very small and more efficient with the primary frequency at 50kHz.

In a switcher, some electronics live on the AC line prior to the transformers. Optical couplers will feed back information about the voltages on the secondary supply of the transformer to the circuitry on the AC line side. For safety, the secondary side is totally isolated from the power cord as in linear power supplies. The XC-Z9 has two separate switching supplies and two transformers. One is dedicated for the power amplifier's output transistors. On the secondary side the 12V analog rail is reduced to 5V for the digital components by another switching regulator.

Efficiency and size reduction are not cheap. The power supply has three analog LSI circuits to oversee the operation. Five feedback loops require five optocouplers. To support all this are 100 discrete transistors, diodes, and passive components. All the switching produces noise on the power supply's output and the feedback can interact with the load at the output of the switching supply. This is why most designers of switching power amps (the unit that amplifies the music signal that drives the speaker) consider the design of the switching power supply (the unit that takes the AC line voltage and produces the DC voltage for the power amp) to be an integral part of the design of a switching power amp (the unit that amplifies the audio signal and drives the speaker). To sum up: it's not easy being green. ■

dio chip and the TI TAS5504A. Using a dedicated speaker for the XC-Z9 as part of the X-Z9 complex mitigates some of the problems associated with a feedback-free digital switching amp.

Pioneer has chosen the Bang & Olufsen ICEpower ASH high-efficiency amplifiers for its high-end audio-video receivers. ICEpower provides modules instead of chips. The modules combine the switching power amp and the switching power supply, preventing noise and distortion that results when a mix and match approach is used. The ICEpower design uses analog feedback loops, an analog input, and the requisite digital-to-analog converter. The measured performance presented on the ICEpower website is significantly better with respect to output noise and distortion than the digital-based amps. The ICEpower is a more expensive solution that may have increased the price of the XC-Z9 significantly.

The headphone outputs also start as digital one-bit signals from the TI TAS5504A. They are filtered with an RC low-pass network and then converted to analog. An NJM2068 buffers the filter and drives the headphone. This is a surprising choice because it is rated to drive 2kΩ loads. I was expecting the traditional NJM4556 that can drive 150Ω loads. The NJM2068 is not a cheaper chip and is often found in many audio products during a variety of internal functions.

The NJM2068 has twice the gain bandwidth and slew rate compared to the 4556. These differences should not be an issue when driving a headphone with an analog signal, but it must have been with the filtered one-bit switched signal. I cannot understand why the output of the 2068 is not buffered by an additional 4556. The fact that the headphone jack is a mini plug in this unit indicates that only wimpy headphones can be connected here. For a unit with such high-end protections, wimping out on the headphone section is a major disappointment.

Analog signals must be fit on top of the digital integrated amplifier architecture. A CMOS switch lets you choose from the analog (two-line) inputs, iPod, Sirius satellite radio, AM/FM tuner, and the output of the phono preamp. These analog signals are converted to digital by an AKM5358 chip, which has a worst case, A-weighted, signal-to-noise ratio of equivalent 15.5 bits and a THD equivalent to 13.5 bits. While this is good enough for the internal analog signals that flow in the XC-Z9 (tuner, iPod, and so on), quality may be sacrificed depending on the performance of the analog device connected to the line inputs.

Given that the SACD player is internal to the XC-Z9, it is unlikely the analog components connected to the XC-Z9 will require higher performance ADC to ensure that significant distortion and noise is added to the incoming signal. The ADC is a good match for the internal phono stage.

The analog line output for a tape recorder requires its own DAC to convert the direct digital input signals back to analog. A TI PCM1742 performs this function. It has a worst case, A-weighted, signal-to-noise ratio equivalent to 16.5 bits and a worst-case THD equivalent to 14 bits.

Note that this unit strangely has no S/PDIF inputs or outputs. Given the digital nature of the signal path, these would have been a natural addition. Those would be used for a CD or mini disk recorder with S/PDIF connection. With the S/PDIF connection you could achieve better performance by removing the ADCs and DACs from the signal chain if the XC-Z9 had the appropriate connections. Some HD radios and external phono preamps also have S/PDIF outputs.

The phono stage is a single op amp design typical of a stereo receiver priced below $500. The distinctive feature of the chosen BA4560 op amp is its large low-frequency gain, which is required to accurately trace the RIAA curve at low frequencies. The phono stage is adequate for use with a turntable in the $200-$300 range, which is likely the case for a secondary system in which the X-Z9 is employed. Use of a higher end turntable and cartridge with an external phono preamp might not be ideal given the performance limitations of the AKM5358 analog-to-digital converter.

Overall, the XC-Z9 is a lot of technology for the money. Usability is improved with the wealth of functionality shoehorned into one box. The associated display and remote control conveniently enable high-tech features. Fitting all this performance into a compact space required much creativity on the part of the design team's electrical engineers.

S-Z9

This is a highly original and compact design of a true three-way speaker from Pioneer's central engineering staff. It is finished in a fine piano black enclosure. They supply a cleaning cloth, which you had better not lose because it costs $40.95 to replace, before shipping and handling. Yikes. Pioneer also supplies the speaker cable (a replacement costs $37.95). The speakers are marked left and right, but for the life of me I could not find or measure a difference. Pioneer could not shed any light on this.

The midrange and tweeter are in a

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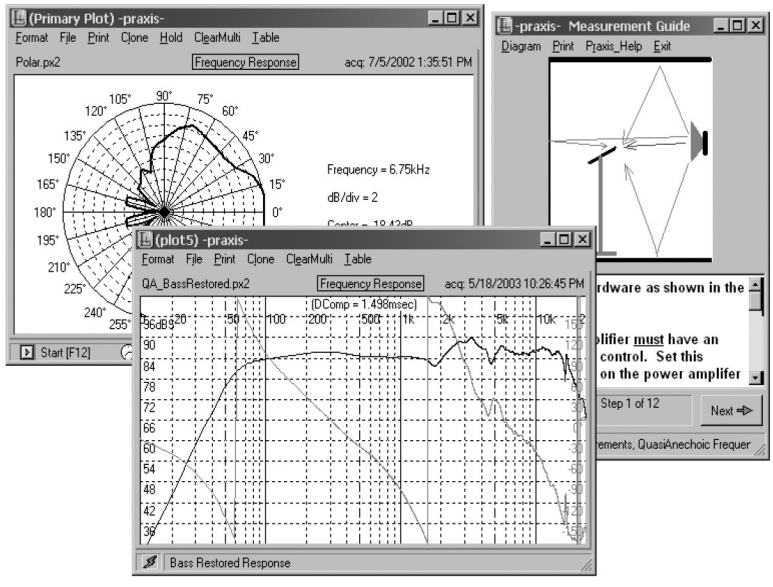

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coaxial configuration. The back of the speaker indicates that it uses technology from KEF and GPAcoustics, which owns KEF. At a minimum, this could be rights to KEF's patented compound loudspeaker drive unit alignment method (US patent **5,548,657**) of the tweeter and midrange. A KEF representative indicated they did not have any direct design input.

The midrange has a stiff concave surround to terminate energy at the end of the cone to the cast frame of the driver. The woofer, which is the same size as the midrange, has a traditional loose convex surround. The cone moves easily when the woofer is pushed, but not so when the midrange is depressed.

The three-way design is important, given the 45Hz tuning of the box and the small size of the woofer. At 50Hz, the cone moves visibly at an SPL of 85dB in a small room. This would cause intermodulation distortion were the driver to run out to 3kHz, but it is crossed over at 800Hz. From appearance, the S-Z9 is a less expensive remake of the unique Pioneer S-2EX from the Pioneer TAD group, though Pioneer indicated to me that TAD was not involved. The driver configurations are the same, though S-2EX is slightly larger with its 7″ woofer.

The small woofer can bottom out when playing frequencies below 150Hz in a large room at a SPL of 80dB. I band-limited an amp to 150Hz that could perform 200W into 4Ω and applied the last five minutes of Mahler's 2nd symphony. The results were as clean as a full-size three-way tower speaker costing $1000–2000 per pair. Clearly, distortion is not an issue when the XC-Z9 drives the unit in a smaller room. A two-way mini monitor or tower speaker could not compete with respect to dynamics or clarity in the midrange between 500Hz and 3kHz. A true midrange is vital when high-end sound is the objective.

In a smaller room, the unit delivers honest bass response down to 45Hz. There was no sign of the unit's having been tuned to push up in the bass like the $10,000 mini monitors at CES, which are accompanied by marketers who talk up the pace and rhythm of the bass. The S-Z9 speakers are amiable as long as the SPLs are kept to reasonable levels. Use of the bass control, loudness, and "exciting" Sound Mode obliterate the speaker's clean low-end and potentially damage the signal when the volume control is pushed. ***rr***

This review of the Pioneer system concludes next month with measurements and listening tests.—Eds.

Echo Busters Decorative Acoustical Treatments

By Tom Lyle

PHOTO 1: Free-standing Echo Buster panels.

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Price: 4′ × 18″ Echo Buster $220, 4′ × 12″ Double Buster $225, 4′ × 9″ radius quarter round Bass Buster $325, 4′ × 12″ × 12″ Phase 4 $395, 15″ × 15″ × 15″ Corner Buster $190 pack of 4; other sizes, non-standard colors, and packages available.

I lucked out when we moved to our new home—I got my very own private, dedicated listening room. In the room are my stereo, shelves with records and CDs, a desk with a computer, network stuff, and a printer. And my listening chair. Paradise.

But there was trouble in paradise: the room was not custom built, and it sounded horrible—the cause of the problem seemed to be its sloping ceilings. The dimensions of the room are more than adequate, with an 8′ plus high ceiling at its tallest that flattens out for more than half the width of the room. But, when I cranked up the volume on the stereo, the bass was constricted, and it had a boosted midrange that included an annoyingly exaggerated cupped-hands effect. I thought the industrial-grade carpet and the window shades (not to mention all the records lining the walls) would be enough acoustic treatment. But no.

> "When I cranked up the volume on the stereo, the bass was constricted, and it had a boosted midrange that included an annoyingly exaggerated cupped-hands effect."

The folks at Echo Busters (Mike, and his assistant Joan, actually) were glad to help me out. I faxed them a hand-drawn diagram with the dimensions of the room, in addition to a top and a side view. They transferred this to a computer blueprint, and then we discussed my problems at length and came up with a few solutions.

CONTROLLING THE SOUND

Echo Busters offers three basic methods of controlling the sound of the listening room. The first is—and seemed rather obvious to me—sound wave absorption. The two products they offer for this are Echo Buster absorption panels (**Photo 1**) and Corner Buster triangular ceiling traps.

The rather lightweight Echo Buster panel is filled with flame-retardant foam and housed in a rectangular wooden frame that is meant to be hung on walls or converted to a free-standing panel. These panels are designed to absorb the sounds that reverberate around a room. These echoes, says Echo Busters, can distort and smear the soundstage. The company claims these panels reduce the reflected sounds to produce a quieter, more revealing listening environment. Music can then be played louder with no discomfort.

The other absorption product they offer is the Corner Buster triangular ceiling trap, which is a fabric-wrapped triangular foam panel with an internal plastic diaphragm. It has beveled edges that are designed to mount into the ceiling corners. Echo Busters says that these are important because sound eventually finds its way into all corners of a room. These resonances can be terminated to treat this acoustical problem. Corner Busters mount with either Velcro or nails.

Echo Busters' second method of controlling the sound of the listening room is bass control. The two types of products Echo Busters offers for this are Phase-4 tuning towers and Bass Busters' low-frequency control towers.

The Phase-4 is a bass trap that has two sides of absorption and two sides of diffusion which allow—says Echo Busters—a multitude of ways to fine-tune a room. You can rotate it to achieve different acoustic benefits. Its goal is to provide control of "one-note" bass while also helping to control midrange and high frequencies. The two reflective

surfaces are made with solid laminate boards and the absorptive surfaces are perforated laminate boards with acoustical foam inside.

The Bass Buster control tower is designed for corner placement with the goal of "turn[ing] acoustically bad room corners into superb high frequency diffusers." They are designed to capture standing waves formed by sound bouncing off walls, floors, and ceilings. They claim that by removing these "boomy" low frequencies, the full bass extension that speakers are capable of producing is achieved. Bass Busters are made with a MDF (medium density fiberboard) cabinet partially filled with foam. The convex plywood front has a perforated surface that is meant to allow bass to pass through while diffusing high frequencies.

The third method of controlling sound is Echo Busters' Double Buster, a diffusion panel. Echo Busters says that "Double Busters create the soundfield diffusion that is the hallmark of great acoustical spaces." The Double Buster scatters sound waves to produce the so-called "disappearing walls" effect. They also absorb low frequencies, while at the same time reflecting the higher frequencies producing a "hemispherical radiation pattern." They are meant to open up a soundstage and add dimension to the sound.

Echo Busters also makes products for noise control—that is, soundproofing. I guess these could be useful in settings where the noise from a listening room is imposing upon others. These products could be helpful to apartment dwellers, and those who share a common wall with family members who do not want to hear the current music selection of an audiophile. But Echo Busters says these products are most helpful in restaurants, schools, offices, libraries, recording studios, and the like.

AESTHETICS

Echo Busters calls these panels and the like "decorative," but what is considered good-looking audio equipment (and all those accessories) to me and what is considered good-looking to the rest of my family are most often two completely different things. If you choose the "Acoustic Art" fabric option available on Echo Buster panels, blending into the décor of a room might be somewhat easier because of the various designs Echo Busters has to offer, but they still might be quite noticeable. But in reality I suppose these panels will end up being placed in dedicated home theater or music listening rooms, not in your average living room.

They are kind of large, and even if the color or fabric design is chosen to "blend in" with the surroundings, they are likely to be noticed. But maybe I'm being a bit harsh. If located in their appropriate positions, perhaps they'll fit in or perhaps even—God forbid—enhance their surroundings.

With so many homes' living rooms doubling as home theaters, there is a chance that they'll just integrate with the rest of the stuff. Although, if you follow Echo Busters' placement recommendations, they might be noticeable not just because they are large, but because there will be so many of them. That said, I had no problem incorporating them into my listening room because, to be honest, no one really cared what they looked like—except me, and when they were all up I thought they looked pretty cool in an audiophile aficionado sort of way.

PLACEMENT

Because of my space and acoustic needs, I chose just the Echo Buster and Double Buster panels. I didn't need any bass control panels or columns because, as I said earlier, too much or bloated bass was hardly the problem. The waveforms bouncing all over the room needed to be controlled, so I followed Echo Busters' rule-of-thumb of deadening the front end of the listening room and livening the back end.

I started with a few panels, adding more later. The improvements in the sound of my room were incrementally positive as I added panels. I don't want to give the impression that I simply hung the panels on my walls in accordance with Echo Busters' plan and all was fine. I needed to experiment quite a bit, especially in regard to sidewall placement. I also was mindful not to over-damp the room, which might have been as detrimental to the sound as not having any acoustic treatment at all.

I started with two 4′ × 18″ Echo Busters directly behind the front speakers (no, there aren't any center or rear speakers). I heard an improvement just by leaning them up against the back wall. After a while, I hung them permanently with the supplied simple hardware—these panels are rather lightweight and don't need much in the way of equipment to mount them on a straight wall. The midrange calmed down a bit, and, surprisingly, some of the bass started to become more articulate—and thankfully more audible.

Next up were the side panels, which were slightly smaller 4′ × 12″ panels located not just at the first reflection point along the side walls, but closer and behind the speaker on the side walls. I again heard improvements in the sound, this time not only in the mid and bass, but also the upper frequencies. Plus the soundstage came into focus.

Then the time came to mount panels on the angled walls. This was a little tricky, and took some time to get right. I ended up using "L" brackets screwed into the outer edge of the panels. There were a total of four per panel, two on each side. Finding the correct anchors to use on the thin layer of plaster over slat walls was a bit tricky, but I received some advice from one of the wandering employees at the mega-hardware store, and they were up in no time.

The panels on the angled walls had a dramatic effect on the sound. I placed two of the 4′ × 18″ Echo Buster absorptive panels on the front end of the room, and two 4′ × 12″ Double Buster reflective panels behind the listening position on the back end. Victory. The result of having all the panels—front, side, rear—up in the listening room made it sound as though I was in a completely different room.

The midrange anomaly that was present before the panels were up was nowhere to be found. In its place was a sound that was recording dependent, and if that recording was up to it, had a soundstage that was virtually 3-D. Between, behind, and beyond the speakers was a soundstage that I could virtually see into. With my eyes closed, with the right recording, the listening room's walls disappeared.

Now there was bass: real, solid, tuneful, deep, gut-shaking, chest-pounding, pitch stable bass that varied little in volume throughout its frequency range. It's as though before the panels were installed my woofers were malfunctioning. Not any more.

There's no need to include here audiophile-reviewer nonsense or needless musical examples. Basically, what these panels did was turn a below-average-sounding listening room into at least an average-sounding one (which I'm thankful for), and maybe even an above average one (which it often ascends to). I can only imagine what these panels would do for a room that has "no problems"—that is, a room that a listener feels needs no acoustic treatment. I can envision that this room could soar to great heights. My room did not soar to greatness, but I sure can live with it now.

Next, I think I might try out the Bass Busters or a couple of Phase-4 units. Who knows where they will take my room? If you have the space and unautocratic decorating guidelines (that is, a sympathetic spouse), I highly recommend these products. ***rr***

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