CROSSOVER APPROACHES

I want to thank G.R. Koonce for his recent article "Passive Crossover Linear Phase Speakers" (June and July '08). This topic is of interest to me, and it is noteworthy that your conclusions and suggestions are similar to mine.

In my article titled "Tweeter Setback" (Sept. '08), I reduced the "delay dispersion" by putting the front of the tweeter behind the front of the woofer. Your article and mine have a common goal: to reduce the effects of phase distortion. Your article is more pristine in that it approaches zero phase distortion. My article acknowledges phase distortion, but asks what can be done to reduce it to inaudibility.

I was very interested in your comments on the audibility of different crossovers. You display unusual (for hi-fi) honesty. Your praise of the third-order crossover agrees with my assessment. In my opinion, fast crossovers help in reducing lobing.

There is one crossover design you did not mention that can result in a linear phase loudspeaker. In the supplementary crossover (CO), one CO (say, the low-pass) is subtracted from the input, with the "supplement" being the other CO (the high-pass). However, I've worked at making a good supplementary CO for years now, with little success, the major problem being that the supplement is a low-order CO, often with ears. This results in a lot of destructive interference, and real-world drivers don't do destructive interference well.

I agree with your recommendation to use a digital electronic CO, because it can give arbitrary delay and a wide CO choice. I also agree that good drivers are very important. I have a pair of Lowther knockoffs that sound delightful in spite of their measured limitations.

I appreciate your good work, and look forward to your next article.

Dick Crawford
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G.R. Koonce responds:

I would like to thank Mr. Crawford for his kind comments on my article and look forward to reading his article. Tweeter setback generally reduces time dispersion, but can have practical problems. In my systems the tweeter location increased the distance between the tweeter and the grille cloth. After I wrote the article, I learned that increasing this distance raises the trouble the grille cloth can cause in terms of adding echoes to the tweeter response. I spent considerable time rectifying this problem, finding thin felt rings on the tweeter faceplate virtually mandatory.

I think Mr. Crawford and I agree on the time dispersion issue. You want to keep it limited, but there is no need to drive it to zero. Thus, minimum-phase crossover designs are probably of more merit than true linear-phase designs. Mr. Crawford brings up the supplementary crossover. I believe this approach did not fit the intent of my article because I know of no passive implementation using a single amplifier per channel. It does not sound as though Mr. Crawford has had much luck with this approach.

Since my article, George Augspurger has brought to my attention a minimum-phase crossover approach he had developed. With his permission, I will document it here for readers to consider. It is amazingly simple, being a low Q second-order design with wide overlap. You simply design a second-order low-pass (LP) with a Q of 0.5 at twice the intended crossover frequency and pair it with a second-order high-pass (HP) with the same Q of 0.5 at half the intended cross-

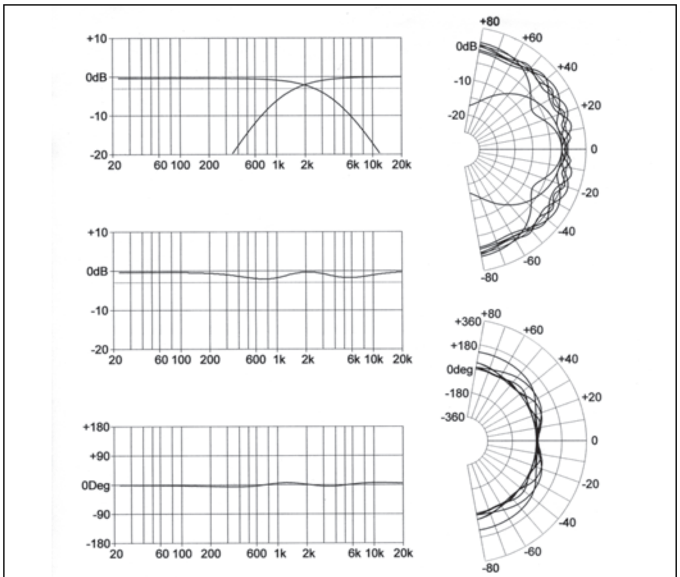

FIGURE 1: Performance of two-way minimum-phase crossover with ideal WTW array.

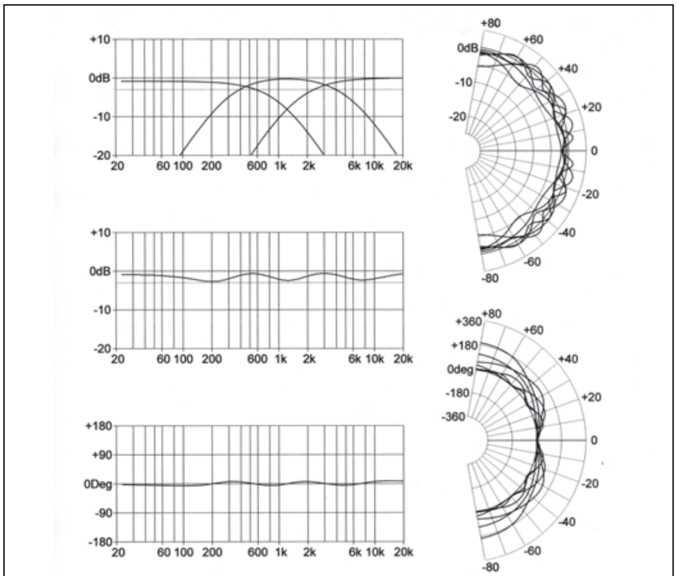

FIGURE 2: Performance of three-way minimum-phase crossover with ideal WMTMW array.

over frequency.

Figure 1 shows the shape of the LP and HP responses and the system performance with a symmetrical WTW array assuming ideal drivers for a design CO frequency of 2kHz. The magnitude anomalies are about 2dB and the phase holds to about ±7°. The approach can be extended to a three-way by designing the bandpass to the same Q and overlap as for the other sections.

TABLE 1: GEORGE AUGSPURGER'S MINIMUM-PHASE SECOND-ORDER CROSSOVER APPROACH.

System	Crossover Frequencies	LP Q	Design f_{co}	BP Q	Design f_{coL} f_{coH}	HP Q	Design fco
Two-Way	2k	0.5	4k			0.5	1k
Three-Way	500, 3k	0.5	1k	0.5	250 6k	0.5	1.5k

Figure 2 shows the performance for a three-way five-driver WMTMW symmetrical ideal array. The performance is not as good if the two crossover frequencies become close together. This is a simple approach with low parts count that ultimately gives 12dB/octave slopes, but still wants drivers with good response overlap. I recommend using symmetrical arrays to limit lobing problems. **Table 1** shows the design parameters for the crossovers in the figures to clarify the approach.

I hope all those working on time dispersion effects and limiting designs will publish so we may all make progress in this area. Right now, I admit I'm still confused about what constitutes the tolerable limit and how it varies with frequency and other variables.

The linear phase speaker article by Mr. Koonce was of great interest to me because I am building an almost identical system. I noted that the 18dB per octave crossover is -6dB at the crossover point, while the 6dB crossover is -3dB. Is this correct? Also, there is a 6dB per octave "Solen split" which is down 6dB at the crossover point. It is supposed to reduce the "problems" of a first-order filter. Any help or comments would be greatly appreciated!

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G.R. Koonce responds:

Mr. Mogavero has asked about the inconsistency in the crossover point between the first-order (6dB/octave) and third-order (18dB/octave) crossovers (COs) shown in my article. I suspect many readers may not be familiar with why the various CO orders have different crossover points, so let me give a brief review.

I will discuss the first-order Butterworth (B1), the second-order all-pass (AP2), the third-order Butterworth (B3), and the fourth-order Linkwitz-Riley (LR4). All of these COs are capable of summing to unity across the frequency band, the definition of an all-pass CO. In all-pass configurations, they do not all have the same system phase characteristic.

Each CO is covered in the accompanying multi-graph figures. The plots are the electrical outputs of the CO when loaded with fixed resistors. You may alternately think of the plots as representing the on-axis acoustic output of an "ideal" coaxial driver.

The top graph is the low-pass (LP) magnitude output (solid line) and the high-pass (HP) magnitude output (dashed curve). The second graph shows the phase shift for the LP and HP outputs. The third graph shows the system output for both drivers wired the same polarity (solid line) and for the tweeter (HP) polarity inverted (dashed curve). The fourth graph is the system phase shift for the same conditions. All plots are for a 1kHz CO frequency.

The B1 CO (6dB/octave) is shown in

Fig. 1. As Mr. Mogavero notes, this type CO has both responses down 3dB at the CO frequency. Note that the phase difference between the LP and HP is 90° throughout.

As **Fig. 2** shows, when two equal signals are vector-added with 90° spacing, the result is 1.414 times each signal, or up 3dB. This is why this CO has the LP and HP down 3dB at the CO point. Note that with either tweeter polarity the system sums to unity. However, only with normal tweeter polarity does the system show zero degrees phase shift and is thus phase linear.

The AP2 CO (12dB/octave) is shown in **Fig. 3**. The LP and HP phases are 180° apart, so when summed with normal tweeter polarity they cancel, producing the infamous second-order notch. With inverted tweeter polarity the system sums to unity because now the LP and HP are in phase. Two equal in-phase signals sum to twice each signal, or +6dB. So all-pass second-order COs cross over at 6dB down. Note that the system phase shift shows a phase reversal occurs at the frequency of the response notch with normal tweeter polarity.

The B3 CO (18dB/octave) is shown in **Fig. 4**. The LP and HP are 270° apart which, because phase repeats at 360°, is the same as 90° apart. Thus again the two sum to 1.414 times each signal and must thus cross over at –3dB. Both tweeter polarities sum to unity. Note, however, the inverted tweeter system has much less phase shift across the frequency band than the system using normal tweeter polarity.

Finally, the LR4 CO (24dB/octave) is shown in **Fig. 5**. The LP and HP are 360° apart, which is the same as being in phase. Thus the normal tweeter polarity sums to unity while the inverted tweeter polarity has the notch. The system phase shift once again shows a phase inversion occurs at the frequency of this notch.

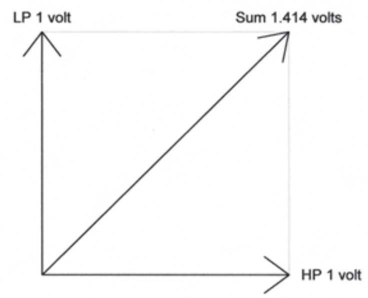

FIGURE 2: Vector summation of two signals at 90° apart.

So with "ideal" all-pass COs the pattern is clear: Odd-order COs LP and HP outputs should be down 3dB at the CO frequency, while for even-order COs they should be down 6dB. As Mr. Mogavero has spotted, the third-order CO I tried (Fig. 29 in my article) was down more than 6dB at the CO frequencies. Review of the second-order CO (Fig. 27 in my article) shows the lower CO point is down about 8dB. This is because when the CO drives real drivers things do not behave as predicted in the ideal.

The COs used in my article were developed by modeling the acoustic response of the system without regard for using ideal AP2 or B3 CO shapes. The drivers' impedance variations and their response anomalies make the acoustic LP and HP responses much different from the expected electrical response of the CO components. It is the acoustic CO response that you care about, and using "ideal" electrical CO networks will likely not produce what is needed in the way of acoustic CO responses. How the acoustic CO responses sum is further modified by the drivers' physical positions and their relative acoustic origin locations. Thus

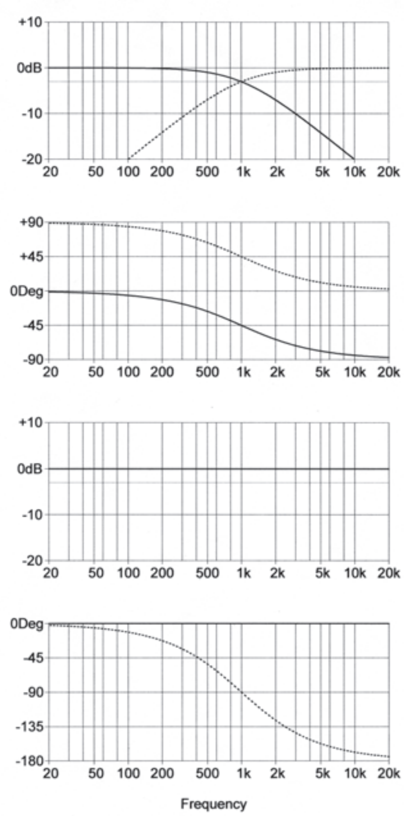

FIGURE 1: Performance of first-order Butterworth crossover.

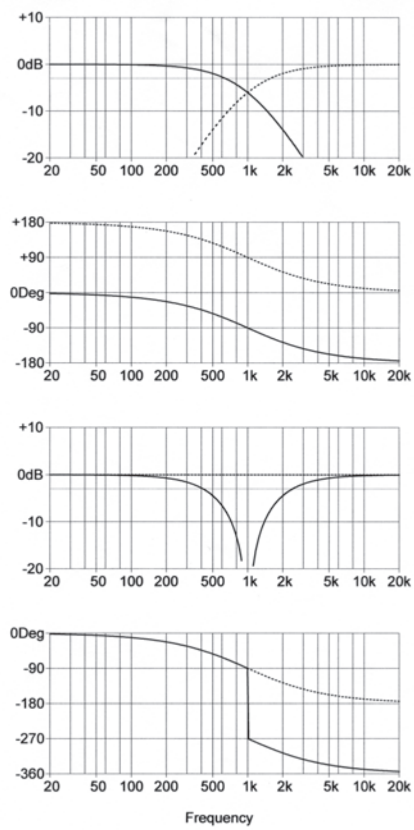

FIGURE 3: Performance of second-order all-pass crossover.

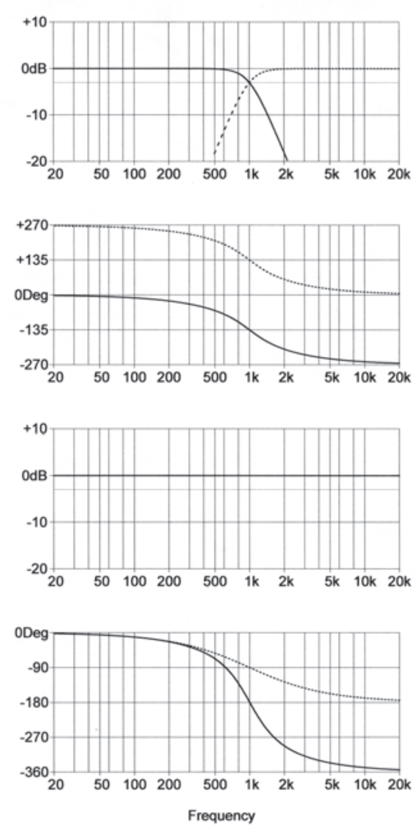

FIGURE 4: Performance of third-order Butterworth crossover.

to produce the desired system response the crossover points are not what ideal CO theory predicts.

I know nothing about the "Solen split" CO and can therefore not comment on it. I hope this answers Mr. Mogavero's questions and clarifies the different CO points. I wish Mr. Mogavero good luck with his speaker project.

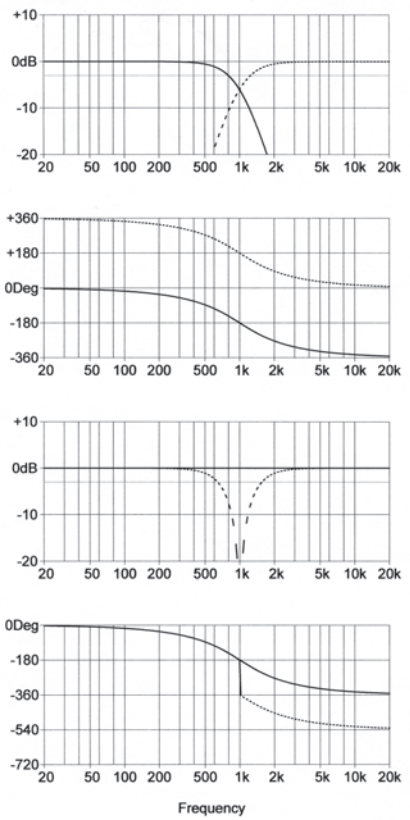

FIGURE 5: Performance of fourth-order Linkwitz-Riley crossover.

GUITAR AMP

I have some questions for Mr. Joseph N. Still about his tube integrated stereo amp (Sept. '08). I am a musician, and would like to build this amplifier to use with my tube preamp. Here are my questions:

1. Can the bias system, which is cathode bias, be replaced by the system you used in your 60W ultralinear amplifier from *audioXpress* 6/04? This system has an adjustable/balance bias system.
2. Can I bridge both output transformers with a switch to get a mono output (parallel them) when not using them in stereo?

I have all the parts to build your 60W unit, but I see you have a new 50W one that uses fewer parts. I know this is a hi-fi amplifier, but would love to use it in my guitar rig; it would give me a clean warm tube output power amplifier.

Dusk Ball
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Joseph Norwood Still responds:

Thank you for your inquiry regarding my article. I hope you'll enjoy it and have little trouble building it!

1. Yes, you can use the bias system from the amplifier in the 6/04 article.

2. You can bridge the 8Ω outputs by using a switch, if the guitar speaker is 4Ω. If you use an 8Ω speaker, parallel the output transformer with 16Ω outputs.

I found a matched pair of 6550s in a fixed bias configuration gave a balanced output, so you might choose to try this configuration before going with the more complex bias system.

The amplifier in the 9/08 article contains an error. The feedback of 12dB is incorrect; the correct feedback is 9dB. I'm very sorry for the mistake!

I assume you intend to build the amplifier on a single chassis, which is available from Mouser Electronics (800-346-6873). The size of the aluminum chassis is 17 × 10 × 3″.

TUBE AMP

This is in reference to Alexander Arion's "Greek Triad" article in the October '08 issue. First, I can't see the advantage of the cascode stage. Maybe I just don't understand it well enough. Second, the coupling capacitor to the grid of the output stage is so big that the response would be down 3dB at about 0.6Hz. I pity the output transformer!

Third, the input is directly coupled without any coupling capacitor, so transients from switching inputs are applied directly to the grid. Looks like a poor design to me. Lack of negative feedback probably is because there is not enough gain in a single 6SL7 stage to permit it. Of course, maybe the author thinks negative feedback is bad.

Ron Anderson
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Alex Arion responds:

Thank you for your inquiries about the Greek Triad. The first stage is not a "cascode" but a SRPP, which is good enough for the job. I selected the 6SL7(6H9C) instead of the classic 6SN7 because of its M factor, bigger than 6SN7(6H8C). The coupling capacitor (1MF) was selected especially to transfer a lot of low frequencies—bypassed by a 47NF. The output transformer is doing well, without problems.

I am not against negative feedback, but if the electronic work is very carefully done, with no noise and good acoustic performance, why do it? Anyway, the amp has been working well from about one year in one of my friends' home.

VOLUME CONTROLLER

I'm not quite sure what to make of "An Automated Level Control" in your October issue (p. 18). The author states that "It does not 'compress' the peaks, but simply 'turns down the volume' as you would do with your volume control." I fail to see the distinction between this circuit and any active compression circuit with a predetermined threshold. This circuit is a compressor, plain and simple.

The author weds an LED and a CdS (not CDS) cell with epoxy and shrink sleeve. Silonex (www.silonex.com) makes a series of "Audiohm" optocoupler devices that perform this same function and have much improved audio characteristics over CdS. You can see a similar compressor circuit in their Technical References.

With the part values as shown, a steep attenuation of high frequencies throughout the circuit starts at the input where even at full volume control setting R18 and C9 reduce the HF -3dB point to just 3.8kHz. It will be reduced even further as the volume control is lowered. By the time the audio signal makes it to the output, the overall HF -3dB point is only 2.7kHz.

Op amps U1 and U2 have limited gain-bandwidth and should have small capacitors across their feedback resistors R14 and R2 to avoid oscillation. Any variations in R4 will change the LF -3dB point as the RC product of R3 + R4 and C2 change.

The whole project seems more geared toward ham radio enthusiasts, where the extremely low (by hi-fi standards) high-frequency turn-down could make Morse code tones and voice more intelligible in the presence of a high noise background. I'm not familiar with ham radio conventions; perhaps that is why the schematic is shown with input on the right and output on the left. Overall, the project is fine for ham operators, I guess. However, given the limited performance of the AD820 op amps and that LM386, I just don't think that one of the applications would be "music systems."

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J.R. Laughlin responds:

If you study the circuit a bit better to understand how it works, you will see that it does not *compress* or *limit* the peaks but simply divides down the entire audio signal such as a volume control will do. D1 and C6 store the approximate peak value, and the LDR divides down the volume so that the determined peak value is not exceeded. R9 was not used. R10 can be used to adjust the time constant of the circuit. Some information about compression and limiting can be read at sound.westhost.com/compression.htm.

The Audiohm couplers use the CdS cell like mine does. I set the high-end rolloff for my particular application; suit yourself for yours. The basic -3dB bandwidth of the amps is around 1.9MHz, which for any gain value used here would provide sufficient bandwidth for the audio range.

Read the "ADJUSTMENTS" section for how to adjust the low- and high-frequency rolloff. Note that use of the LM386 is entirely optional; you can connect any amp of your desire in place of it, externally. I can find no need for caps across R14 and/or R2.

HELP WANTED

I need help finding a source for ½″ bituminized felt. If you remember, Leak and others used it in their speakers to reduce wall vibrations. I have used layers of shingles, which work OK, but would prefer the right materials.

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aX

CONTRIBUTORS

Stuart Yaniger ("The Impasse Preamplifier," p. 5) resides in Benicia, Calif.

Cyril Bateman ("Simulating Inductors and Networks," p. 16) has done extensive work on capacitors, and resides in Norfolk, England.

Dennis Colin ("Noise Measurements of the LSK389B Dual JFET," p. 24) graduated with a BSEE from the University of Lowell (MA) and is currently an Analog Circuit Design Consultant for microwave radios. Previously a band keyboardist and a recording engineer, he has been published in the *Journal* of the Audio Engineering Society. Colin has demonstrated the audibility of phase distortion at Boston Audio Society, and has designed the "Omni-Focus" speaker (bipolar coincidental with phase-linear first-order crossover), ARP 2600 analog music synthesizer, 1kW biamp and PWM supply at A/D/S, and Class D amps.

Paul J. Stamler ("Tri-Way Low Voltage Supply, Pt. 2," p. 28) is a recording engineer/producer, musician, and technical writer; he also hosts a radio program, "No Time to Tarry Here," featuring traditional folk music and related stuff. He has delighted in 78s since he was a boy, when they were still being made.

Gary Galo ("A De-Emphasis Test CD," p. 32) is Audio Engineer at The Crane School of Music, SUNY Potsdam, where he also teaches courses in music literature. A contributor to AAC since 1982, he has authored over 230 articles and reviews on audio technology, music, and recordings. He has been the Sound Recording Reviews Editor of the *ARSC Journal* (Association for Recorded Sound Collections) since 1995, was co-chair of the ARSC Technical Committee from 1996 to 2004, and has given numerous presentations at ARSC conferences (www.arsc-audio.org). Mr. Galo is also a frequent book reviewer for *Notes: Quarterly Journal of the Music Library Association,* has written for the *Newsletter of the Wilhelm Furtwängler Society of America,* and is the author of the "Loudspeaker" entry in *The Encyclopedia of Recorded Sound in the United States,* 1st edition.

Arto Terho ("Showcase: 20W Push-Pull Amp," p. 38), resides in Finland.

Ed Simon (Book Reviwe: *Sound FX*, p. 39) received his B.S.E.E. at Carnegie-Mellon University. He has installed over 500 sound systems at venues including Jacob's Field, Cleveland, Ohio; MCI Center, Washington D.C.; Museum of Modern Art Restaurants, New York; The Coliseum, Nashville, Tenn.; The Forum, Los Angeles; Fisher Cats Stadium, Manchester, N.H.

John Sunier ("Super Fidelity," p. 46) is a CD reviewer for *Australian HiFi and Home Theatre Technology.* His website is www.audaud.com.

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but briefly, presented.

Case then moves on to two special chapters, the snare drum and the piano. The discussion of control and imaging here should be of interest, or perhaps controversy, to most readers. The specifics should guide many of the readers in the studio.

The final chapter is on mixing down the multi track recording in basic terms and with forms of automation. Today this is the conclusion of the studio recording process. In the days of vinyl there would have been a chapter on actually getting the sound to fit into the groove. It is assumed the digital files from the studio can be transferred unmodified to the distribution media with the current technology.

This book, although clearly aimed at current pop music recording techniques, offers many excellent explanations of basic to advanced audio theory and technique. Much is applicable to live sound, and the basics should help many audiophiles better understand why they hear the things they do.

Case does not cover many of the other recording issues in this book. There is very little on microphone technique. The actual selection of the gear and its interconnection is reasonably avoided, because that would limit the life of the book. So in one sense the title is correct; it is not intended to be a complete recording guide. It does contain more than enough information that this book, with the use of the references, could be the core of an excellent modern audio education.

The one shortfall that bothered me is the lack of attribution for some of the data presented. As an example, figure 3.1 is a very valuable chart of audio thresholds related to levels and frequency, but it bears no reference as to where this information came from.

This book hits its target reader dead on, but goes way beyond that. At $39.95, this book is a very low-cost way to get a seat in the first-class compartment on the audio clue train. aX