By Doug Ford (Australia)

Passive Surround-Sound Decoder

Improve sound system quality by eliminating out-of-phase information

This article describes a design for a (relatively) simple passive-matrix surround-sound decoder and the interesting things I discovered during the process. The decoder features an analog time delay, implemented with several cascaded all-pass networks (see **Photo 1**).

My home sound system consists of five Event 20/20BAS powered studio monitors with a powered subwoofer. I was using a Yamaha AV amplifier for audio signal control (i.e., volume, source selection, and surround decoding). I didn't need the Yamaha's power amplifier stages, because these were built in the speakers.

Over time, I became increasingly aware that the Yamaha's surround decoder was rather inadequate. It seemed to mainly produce center-front output. The left, right, and rear output always seemed weak and poorly steered. In addition, the Yamaha's audio path (i.e., A-D, DSP, and D-A) exhibited a significant latency (delay) of more than 50 ms. These defects irritated me so much I decided to design and evaluate a basic "passive-matrix" surround decoder.

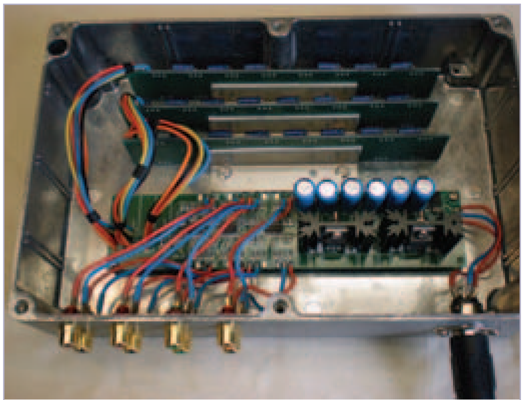

Photo 1: The decoder features an analog time delay, implemented with several cascaded all-pass networks.

SURROUND DECODERS

A passive-matrix decoder is conceptually simple: the incoming left and right audio signals are passed unprocessed to the front-left and front-right speakers, the left and right signals are summed to produce the center speaker signal, and the signal to the rear speakers is the difference between the left and right inputs. "True" surround decoders detect various features in the incoming stereo signal, in addition to the "sum and difference" signals derived in a passive matrix. The signal's amplitude-envelope characteristics are extracted to steer the apparent sound source to the correct location. When this is correctly done, movie sound can be very good. Unfortunately, these decoders are rarely optimized for music, and decoder performance with music is generally terrible.

A passive-matrix decoder has no "time-variant" signal paths. The gain and frequency response of all signal paths is constant and unchanging, unlike a "real" decoder. As a result, music reproduction should be pretty good. However, movie sound decoding and reproduction won't be particularly inspiring unless some additions are made to the basic passive matrix. First, the frequency response of "difference" program signal content to the rear speaker channels is limited to 7 kHz. This response limitation ensures that small differences in interchannel phase, which can occur over some transmission channels at high frequencies, don't enable center (mono) signals to be inappropriately decoded as rear signals. For this reason, the response of the passive-matrix decoders' rear channels should also be limited.

Second, the rear channel signals should be delayed by approximately 10 to 20 ms. Although some authorities claim that this delay is intended to "decorrelate the rear audio," I believe it simply alleviates the Haas effect: Human hearing's tendency to detect sound direction from the first sound's arrival direction rather than the loudest sound. Since the rear speakers (in real lounge rooms!) are always closer than the front speakers, our hearing would "focus" on sounds from the rear speakers, even if their output was weaker than the front speakers' sound. Thus, the block diagram of my passive-matrix decoder was defined once I decided to include low-pass filtering and time delay for the rear channels (see **Figure 1**).

TIME DELAYS

But, how do we implement a time delay? Most designers would bite the bullet and use digital delays: analog-digital conversion, delay of the digitized signal, then digital-analog conversion. I didn't want to take this approach. The design cycle would be too long, there were too many possibilities for error in a one-off, and the S/N ratio would be less than 90 dB if I used 16-bit architecture.

Analog delay lines? My thesis used SAD1024 charge-coupled sampling analog delay lines in the early 1980s, but it resulted in terrible S/N ratio (35 dB on a good day), terrible distortion (3% on a good day), and I doubt they're available any more.

Spring delays? Acoustic hose delays? A mic and speaker in the bathroom? Continuous tape loop delay? All these options have been used at one time or another,

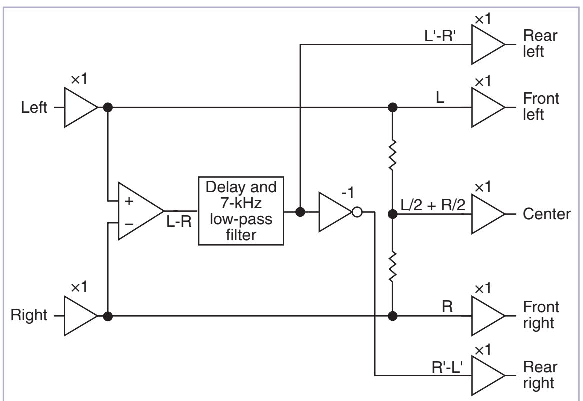

Figure 1: The decoder's block diagram includes a low-pass filter and time delay for the rear channels.

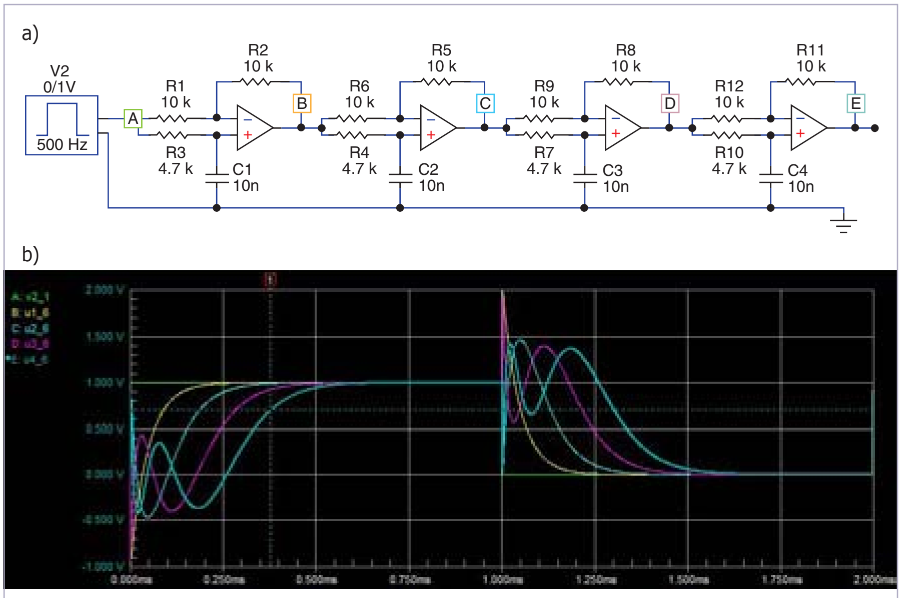

Figure 2: The circuit shows four all-pass networks, implemented with op-amps (a). The simulation waveform shows the response waveform, at each successive stage, to a 500-Hz squarewave (b).

but they're all completely inappropriate in this application.

How about a bunch of cascaded all-pass networks? Hmm. That just might work, particularly since I only need relatively short delay times!

An all-pass network has the interesting property that although its frequency response is flat, its phase response varies from 0° at low frequencies to 180° at high frequencies (although you can flip this to 180° ~ 0° just by swapping a resistor and capacitor). If you cascade several all-pass stages, it starts to behave like a time delay (see **Figure 2**).

This circuit shows four all-pass networks, implemented with op-amps. The simulation waveform shows the response waveform, at each successive stage, to a 500-Hz square wave. Each stage's delay is approximately T = 2RC, where R and C are the components at the non-inverting op-amp inputs. The resistors at the inverting inputs can be pretty much any value, as long as the input resistor and feedback resistor are equal in value. In my example, R = 4,700 Ω and C = 10 nF. The time delay of four stages is: 4 × 2 × 4,700 × 10E-9 = 0.376 ms. The cursor in the waveform is positioned at this time.

As you cascade more and more stages, the number of "oscillations" in the waveform increases proportionally. Even though the frequency response of the cascaded stages is flat, the useable response is limited by the non-ideal time-domain response. In practice, the response should be limited to f = 1/(πRC) to minimize the amplitude of the step-response "oscillations."

Since the surround decoder's delay line response is already limited to 7 kHz, the RC time-constant can be calculated as RC = 1/(F × π), giving RC = 45.5 µs. Since each stage's delay is 2RC, we can get a delay of around 90 µs (0.09 ms) per stage. So, for 10-ms delay, we need (10 ms/0.09 ms) = 111 stages of all-pass networks!

I decided to design the all-pass networks onto plug-in cards, with eight quad op-amps (32 networks) per card (see **Figure 3**). But, I implemented them with 10 kΩ input-and-feedback resistors rather than the 2.7-kΩ resistors shown. Each card would have a 2.9-ms delay (i.e., 32 × 0.09 ms). The simulated waveform response of a card, including the effect of a 7-kHz low-pass filter, is shown in **Figure 4**.

MAIN DECODER CIRCUIT

The main passive-matrix decoder's circuit diagram is shown in **Figure 5**. There are a couple of this circuit's features worth discussing.

- The left and right input buffers are configured as balanced inputs, but the non-inverting impedances are very low (10 Ω). This enables the connection of grounded signal sources to the decoder via standard phono

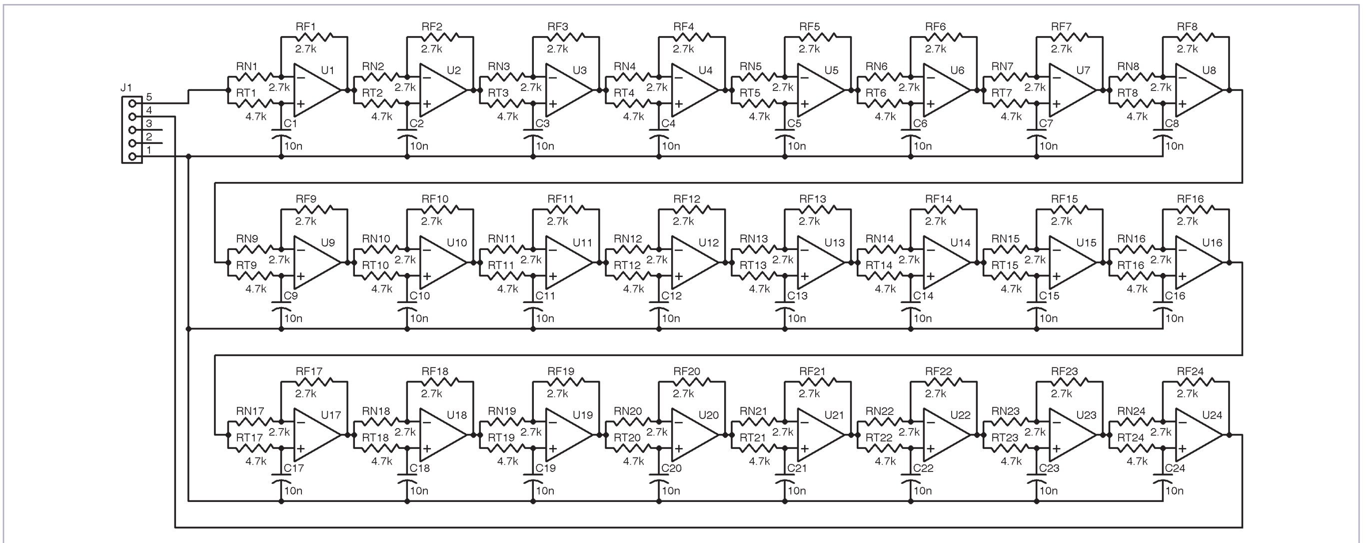

Figure 3: The all-pass networks are designed on plug-in cards, with eight quad op-amps (32 networks) per card.

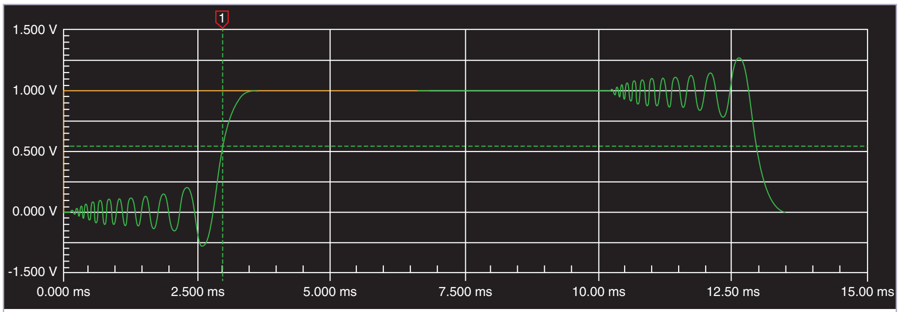

Figure 4: The simulated waveform response of a card, including the effect of a 7-kHz low-pass filter.

cables without creating ground loops. In practice, I didn't use this feature. I actually connected the decoders' phono grounds directly to the circuit common, since the decoder was powered from an ungrounded plug pack.

- The input buffers' response is rolled off at high frequencies (–3 dB at 160 kHz) to minimize the potential for RF-input interference.

- The six output buffers have low-resistance "ground-compensated" returns for the phono connectors. Again, in practice, I returned the phono grounds to the main circuit common.

- Only one "delay" module socket is shown. The PCB was designed for four modules.

- The power supply circuit is not shown here, but it consists of a diode bridge, three 470-µF reservoir caps, and 7812/7912 regulators. Since the quad op-amps use 8 mA each, a fully loaded decoder's total consumption will be 280 mA per rail. For this reason, the regulators were given substantial heatsinking.

- In addition to the five decoded audio outputs, a sixth output was included for a subwoofer. Since most active subwoofers are fitted with internal adjustable crossovers, the response at this output has not been rolled off.

- Op-amps U2B and U2D are second-order, low-pass filters (–3 dB at 7.5 kHz). One is positioned before the delay modules to reduce excitation of high-frequency oscillatory transitions in the delay lines. The other is at the delay line output to reduce high-frequency noise from these lines.

- The front, center, and subwoofer outputs are direct-coupled, with response down to DC. The rear outputs are AC coupled (–3 dB at 8 Hz), since a substantial build-up of DC offset may be expected from large numbers of cascaded all-pass op-amp networks.

I selected ON-Semiconductor MC33079 quad op-amps for all the decoder's stages. They're intended for audio applications. They have low noise (4.5 nV/rt-Hz), low offset (0.15 mV), high GBW (16 MHz), and low THD (0.002%). There are better-specified op-amps available, but they are either expensive or simply not stocked.

I built the decoder with three delay modules, giving a total delay of 8.7 ms. This amount of delay is sufficient to overcome the difference in acoustic flight time between the front and rear speakers in my lounge room. I didn't have enough space in the diecast box to mount the delay modules onto the main decoder card, so I connected them via short wiring harnesses (see **Photo 2**).

TEST AND COMMISSIONING

I used a Rigol oscilloscope and an Audio Precision 2712 signal analyzer to test it, and then I applied power. There was no smoke and the front outputs' performance was looking good. Then I discovered the delay lines were oscillating

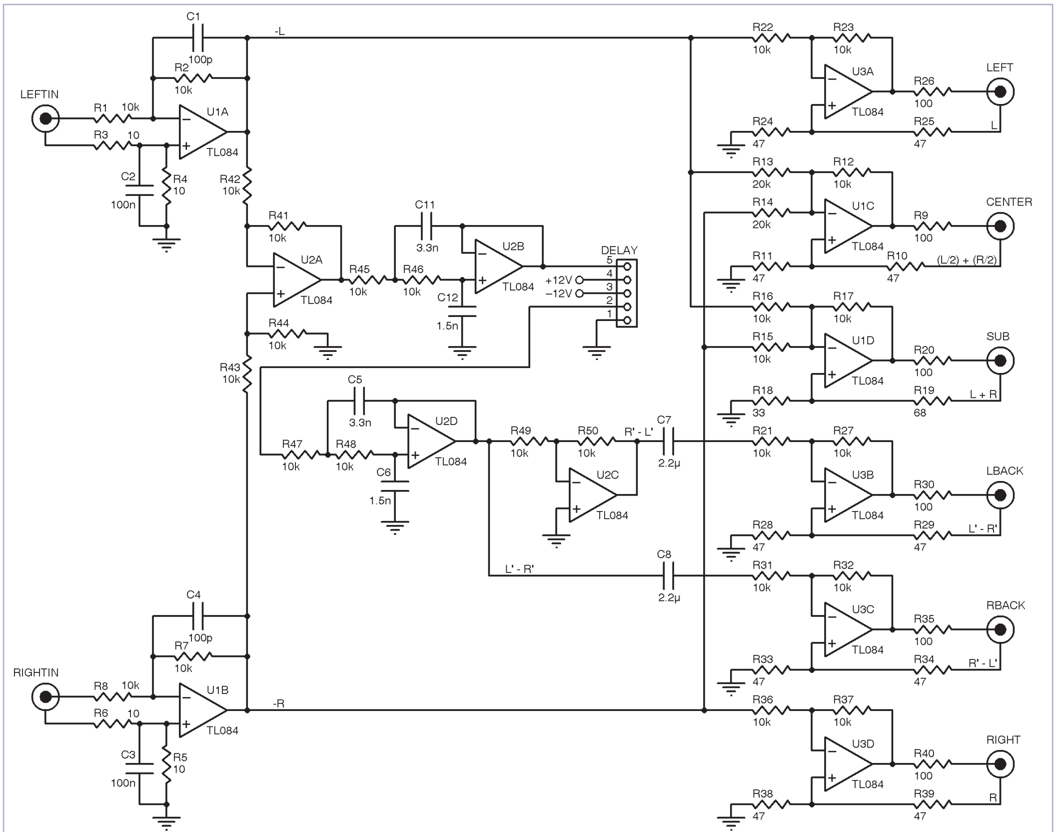

Figure 5: The main passive-matrix decoder's circuit diagram is shown here.

at full voice, at around 2 MHz. I'd paid attention to the layout of these cards. They have a solid ground-plane, and the supply to each op-amp is decoupled. Unfortunately, I'd skimped on the decoupling caps, and used 33 nF on each pin. The MC33079s prefer to have a lot of decoupling! I temporized by putting a 1-μF electrolytic on each device (see **Photo 2**). This helped, but it didn't solve the problem.

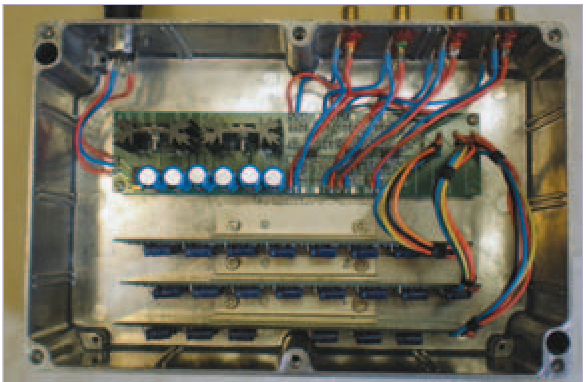
Photo 2: Short wiring harnesses are used to connect the delay modules.

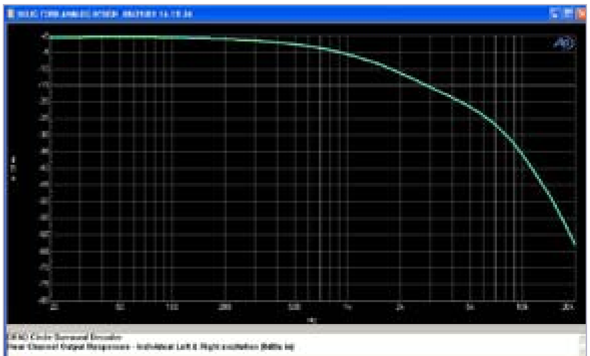
Photo 3: The frequency response reveals a problem with 6 dB down at 1 kHz and 27 dB down at 7 kHz.

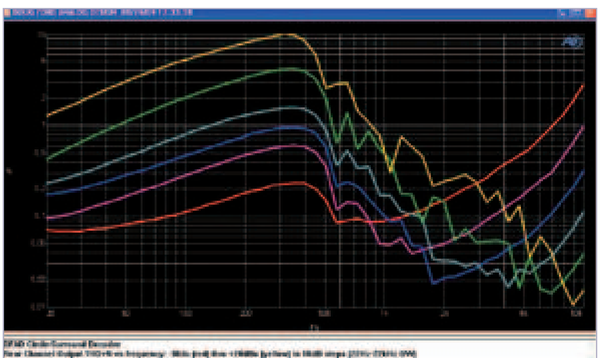
Photo 4: The distortion is very high and amplitude dependent.


Photo 5: The all-pass cards show the 33-nF bypass capacitors and 10-μF electrolytics replaced with 4.7-μF 1206 capacitors.

I discovered that each all-pass stage had a frequency response peak of around 2.5 dB at around 2 MHz. I suspect this peak might actually be caused by stray capacitance from each op-amp's inverting pin to the ground plane—maybe a fraction of a picofarad or so. It was just enough to cause response peaking, anyway. But when you have 32 stages, each with a 2.5-dB peak at 2 MHz, the total gain around the card is 80 dB at 2 MHz. It's no wonder the things were screaming.

I bypassed each 10-kΩ feedback resistor with a 100-pF capacitor—the smallest value I had on the shelf—providing a high-frequency rolloff of −3 dB at 160 kHz at each stage. That made them stable! I noted the DC offset was remarkably low, at less than 5 mV. I was expecting the offsets of 96 op-amps to accumulate and produce some hundreds of millivolts of total offset. I suspect three factors were working in my favor: the individual op-amp offsets were quite low (0.15 mV, typically); the offsets were randomly distributed—some positive, some negative; and most of the bias current offset was negated by making the resistance at each non-inverting input (4.7 kΩ) nearly equal to the resistance at the inverting inputs (two 10 kΩ in parallel).

The rear outputs were stable, but they still had a few small problems. First, the frequency response was pretty awful: 6 dB down at 1 kHz and 27 dB down at 7 kHz (see **Photo 3**). Second, the distortion was high and amplitude dependent (see **Photo 4**). The cause of the poor high-frequency response was obvious: Although the response of each all-pass stage was good to 160 kHz, they were individually 0.16 dB down at 7 kHz. There are 96 stages, so there was 15-dB excess rolloff at 7 kHz!

The excessive THD was caused by the 10-nF capacitors in the all-pass networks. I had used 0805 components with X7R dielectric. This dielectric isn't particularly linear with voltage (or with temperature), but it's not as bad as Y5V dielectrics! I learned a valuable lesson. Even though X7R might be useable in domestic audio applications, where maybe two or three might appear in a signal path, don't use them in premium audio applications or where a hundred or so are in the signal path!

X7R caps may be reasonably used in coupling applications where the applied AC voltage is low (e.g., where the rolloff frequency is much lower than in-band signal levels). They should not be used in midband frequency-selective applications (e.g., tone controls, equalizers or tuned circuits). They are particularly ill-suited for use in all-pass networks, since they have the full AC signal applied to them, via a relatively high source resistance, in a topology that is sensitive to component nonlinearity, and where these nonlinearities will sum cumulatively.

Before I made any modifications, I had a quick look at the distortion waveform on the oscilloscope. Although distortion was clearly visible, it wasn't particularly nasty. The output waveform had some of the characteristics of soft valve-like rounding and some of the characteristics of slew limiting. Encouraged, I listened to some program material. The distortion wasn't particularly objectionable, and its audio impact was outweighed by the poor frequency response. It sounded a bit like an accountant at a backstage party. Not unpleasant, just rather dull.

I fixed the poor frequency response by replacing all the 100-pF feedback capacitors (96 of them) with 22-pF caps. These lifted the individual rolloff points to 720 kHz and reduced the rolloff at 7 kHz from 15 dB to 3 dB. I also tweaked the responses of the second-order low-pass filters by reducing C6 and C12 from 1.5 nF to 1 nF. This peaked the filters by 1 dB each, largely compensating for the remaining rolloff in the all-pass delays. I nuked all the 10nF X7R capacitors and replaced them with parallel pairs of 4.7-nF capacitors with NPO dielectric.

Finally, I replaced the 33-nF bypass capacitors and 10-µF electrolytics with 4.7-µF 1206 capacitors. The all-pass cards now look like the image in **Photo 5**.

FRONT OUTPUTS

The frequency response of the front-left, front-right, center, and subwoofer outputs aren't worth showing. They're simply flat to 20 kHz, and 0.5 dB down at 50 kHz. The noise at these outputs is ridiculously low. There is no trace of hum. All the noise is simply white spectrum: NOISE = –107 dBu A-weighted, or –99 dBu (10 Hz to 80 kHz), or –104 dBu (10 Hz to 22 kHz). The THD+N is extraordinarily low, and it is dominated by simple white noise at input amplitudes below 5 dBu (see **Photo 6**).

Left-to-right crosstalk is very low (–104 dB, all frequencies), but right-to-left crosstalk is marred by a bit of clumsy layout, enabling capacitive coupling of rear-delayed output to the left channel (see **Photo 7**). I won't lose sleep over it, but I'll certainly retrack the main PCB if I'm ever required to make more decoders.

MEASURED PERFORMANCE FOR REAR OUTPUTS

The time delay developed by the all-pass networks is 8.8 ms (see **Photo 8**). This amount of time delay should be ample for my lounge room, but I can always add another card (2.9 ms) if required.

I was expecting relatively poor signal-to-noise ratio from the rear decoder outputs, due to the large number of op-amps (nearly 100) in the signal path. I was flabbergasted to find that the noise was –91 dBu (A-weighted) or –90 dBu (10 Hz to 80 kHz)! Since the rear channels'

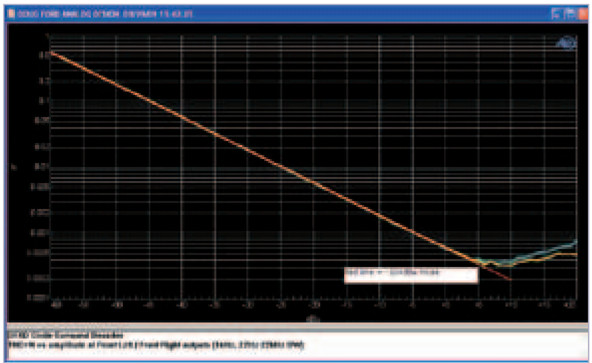

Photo 6: The THD+N is very low and dominated by simple white noise at input amplitudes below 5 dBu.

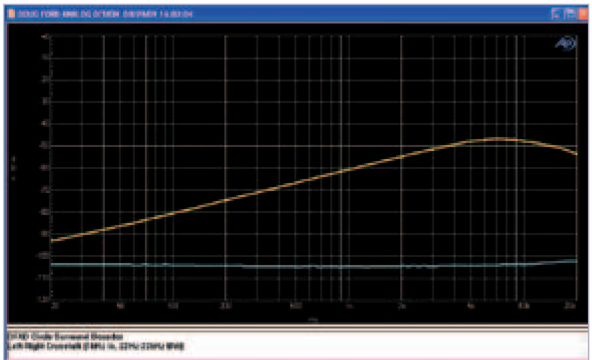

Photo 7: The left-to-right crosstalk is low, but the right-to-left crosstalk could be improved by re-tracking the main PCB.

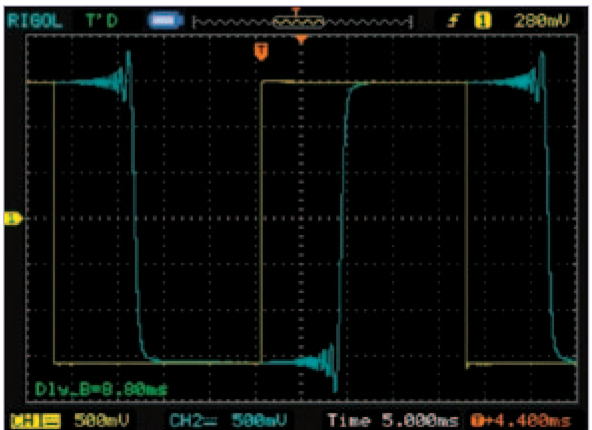

Photo 8: The time delay developed by the all-pass networks is 8.8 ms.

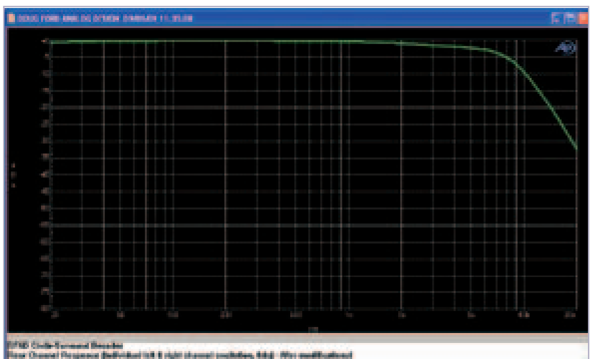

Photo 9: The rear channel's frequency response shows improvement in comparison to Photo 3.

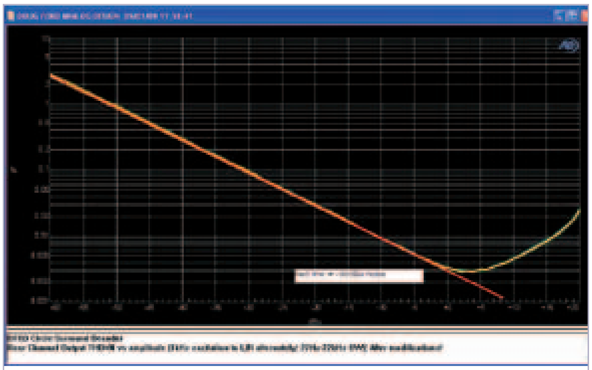

Photo 10: The 1-kHz THD+N versus amplitude is shown.

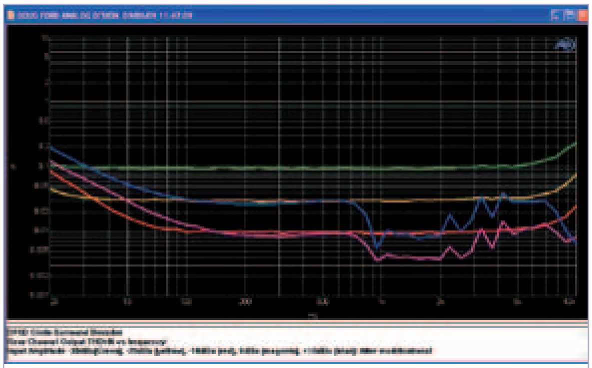

Photo 11: The THD+N versus frequency shows a dramatic reduction compared to Photo 4.

clipping amplitude is just over 20 dBu, the rear channels' dynamic range is 110 dB. This result is at least 20 dB better than a typical 16-bit digital system! Similarly, the dynamic range of the decoders' front outputs is better than 120 dB. Nice!

The rear channel's frequency response, shown in **Photo 9**, is vastly improved in comparison to that of what is shown in **Photo 3**. The response is 4 dB down at 7 kHz, rolling off at 24-dB/octave above this.

The 1-kHz THD+N versus amplitude, is quite remarkable (see **Photo 10**). The THD+N is dominated by noise for input amplitudes up to 0 dBu and is still below 0.01% at 15 dBu. Remember, the signal has been passed through nearly 100 op-amps! The THD+N versus frequency, shown in **Photo 11**, shows a dramatic reduction compared to **Photo 4**. At lower amplitudes (−30 dBu and −20 dBu) the THD+N is dominated by noise. A rise in low-frequency distortion can be seen at 0 dBu and higher. This distortion is probably generated by nonlinearity in the 4.7-nF NPO capacitors. This THD is unlikely to be objectionable, due to the previously noted characteristics of such distortion. If I needed to reduce this THD, I'd simply select NPO capacitors with a higher voltage rating.

LISTENING TESTS

At the time of writing, the listening tests haven't been exhaustive. The decoder has only been in play for a few days, but I can certainly provide my initial impressions.

Compared to the sound of the Yamaha's signal path, this decoder immediately sounded cleaner, fresher, and less gritty. It vastly improved clarity at all listening levels! The elimination of the latency delay was subtle, but noticeable (e.g., better lip-sync).

When playing movies and free-to-air TV, the amount of decoded rear-channel material was surprising, particularly when compared to the Yamaha's decoder. I'm delighted when I can hear actors speaking from different positions across the front of the soundstage and simultaneously hear ambient noise from behind. It's probably a familiar experience to anyone with a good surround decoder, but it's new to me!

I found I had to reduce the rear speaker gains by more than 6 dB, since these speakers are so much closer to the listener than the front speakers. This corrected a significant imbalance in the perceived levels.

When playing music, the soundstage lacks precision in comparison with simple stereo listening, although it is much wider and provides a highly immersive experience. Music through this decoder makes the Yamaha sound dreadful in comparison.

The overall improvement has made me more critical of the audio quality found with some TV channels and programs. For example, many ABC news announcements seem to have excessive distortion on speech sibilants, which appears as an annoying R-L difference signal in the rear speakers.

I was also surprised at the amount of location traffic noise present in the audio for "Packed to the Rafters." Its audio has a high level of low-frequency background noise!

The audio content of TVS transmissions is incredibly variable. The audio changes from mono to hard-left to hard-right, and it occasionally occurs in the span of a few minutes.

WORTH THE EFFORT

You can make a good audio delay with a bunch of op-amps connected as all-pass networks. You don't need to use digital delays, and the MC33079's audio performance is excellent. Let me amend that. The MC33079's audio performance is excellent when you correctly decouple them and remove the worst evidence of 2-MHz peaking. Be aware of capacitor selection criteria when designing SMT audio equipment. Know the differences between X7R, Y5V, NPO/COG, and other dielectrics. Follow these tips and you can build simple passive-matrix surround decoders that can sound pretty good on both music and movie/program material. *aX*

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