A VARIO-MATRIX SURVIVAL GUIDE -- R. Scott Varner

Forgive me for starting off with such a cynical attitude, but it must be said that QS is as dead as any '70s quad system could possibly be. Only ambisonics or SQ will see future development for encoded surround sound. So why produce an article on maintaining QS equipment. I've got a good answer. Even though the QS 'system' will never be resurrected, the Vario-Matrix technology left behind can produce excellent results playing today's music. Vario-Matrix (V-M) decoding offers very smooth separation enhancement, low distortion and wide dynamic range, several interesting circuit typologies, and the ability to extract the maximum amount of either surround or ambience information from a stereo recording. Sounds pretty good, even by today's standards, doesn't it? So, if you own a QSD-1, QSD-2, QRX receiver, or such, count yourself lucky. But, if it ever malfunctions, you're in trouble; do you really think the local service tech, or even Sansui, is going to know how to properly repair ten year old quad equipment? This article, then, is devoted to providing the information you need to keep your V-M device in good shape, and possibly make it better.

Before you service or modify a circuit, you should have an understanding of how it operates. Relatively little has been written on QS encoding and decoding theory, compared to SQ, so some background information is in order. QS took the original Schieber formulas and, by adding $\pm 90^{\circ}$ phase shifting to the rear channels, created the option for placing sounds (at least theoretically) around a 360° circle using 4-2-4 pair-wise-mixing. Attendant stereo/mono compatibility problems were the same as Scheiber and the public demanded higher separation between corner signals. Sansui responded by offering rather bizarre phase modulation and frequency contouring to their QS-1 and other early units. Soon, however, they introduced their Vario-Matrix decoding, which kept frequency response flat and increased corner separation from 3dB to an average 25dB.

The standard Sansui encode/decode block diagrams are shown in Figure 1. During encoding the front channel signals are given a certain amount of phase shift, called the reference, or 0°, shift, while Lb and Rb are shifted by $\pm 90^{\circ}$. Both Lf/Rf and Lb/Rb are blended towards each other by .414 or 7.7dB. The decoder circuit is not quite a mirror of the encoder. Between Lt and Rt the signals are either in-phase or of opposite phases, so simple addition and subtraction circuits can be used and minimal phase shifting at the output is needed only to correct the phases of the dominant signals. Contrast this to even simple SQ decoding, which has accurate decoding only over the bandwidth of the phase shifting. I've always wondered, by the way, why Sansui made these decoding illustrations with variable pots in the decode circuit, since no QS decoder ever included this feature. Nevertheless, it provides an excellent approach to demonstrate the simplest form of a variable-matrix decoder.

Consider the very common situation of a center-front vocalist with a certain amount of rear reverb present, down, say, by 10-15dB. Given a sensing circuit (not shown) that detects the phase between Lt and Rt (in other words, a phase discriminator), it would respond by

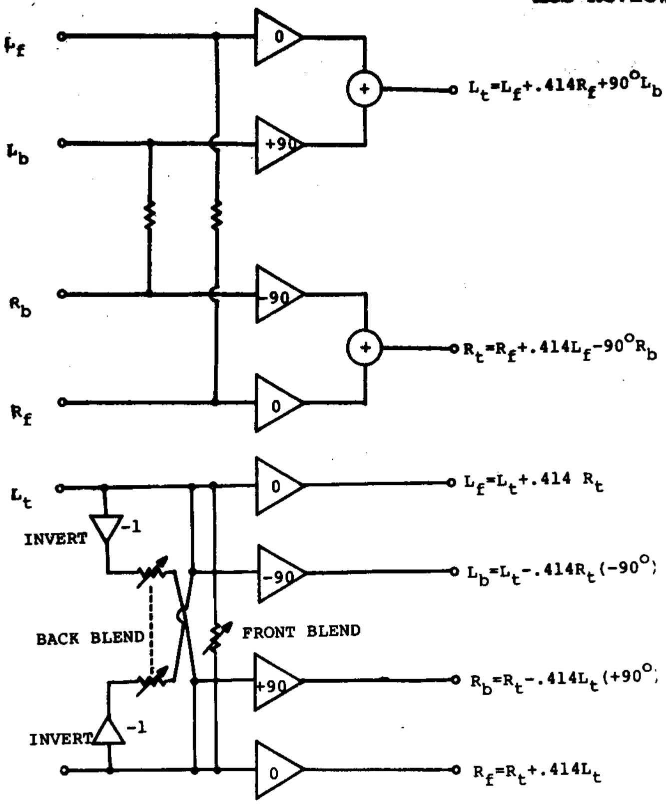

Figure 1. Basic QS encoding (top) and decoding (bottom) circuits.

decreasing the blend on the front channels and **increasing** the blend on the rear. In comparison to fixed decoding, this eliminates the vocalist in the rear, but doesn't attenuate any of the very important rear ambience. (In the case of SQ's 'variblend', which had a similar purpose, blending was not part of the primary decoding process and was applied to the rear channels only.) Thus, without gain-riding, front/back separation is greatly increased and the subdominant sounds are preserved. For a center left/right signal, front and rear blend would be equal and result in 7.7dB L/R separation just as in the basic matrix.

This simple approach takes us part of the way there, but what about a corner signal, say Lf? The phase discriminator would decrease the front blend all the way, increase the rear blend a bit over the fixed decoder and leaves us with 7.7dB corner separation, which is the maximum you'll get on an undecoded QS disc between any

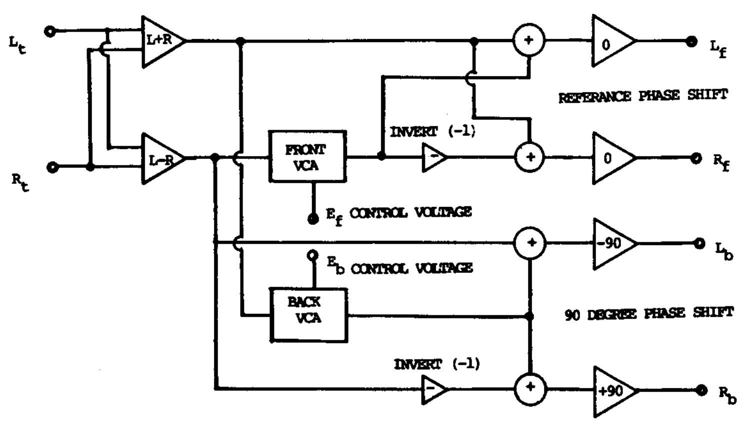

Figure 2. Type B Vario-Matrix block diagram.

corner positions.

Getting closer to real-life V-M decoding, this limited separation can be improved by working with L+R and L-R signals and blending them, sometimes by more than unity gain, to get the desired result. Never mind QS; this is what's done with FM stereo technology that we're so familiar with. The 'main' signal of an FM station is mono, or L+R. A subcarrier represents the L-R or difference information. If you add L+R to L-R, the R signal cancels and you have the original left stereo channel. Invert the difference signal so that you have R-L, add it to L+R, and the right stereo channel is back.

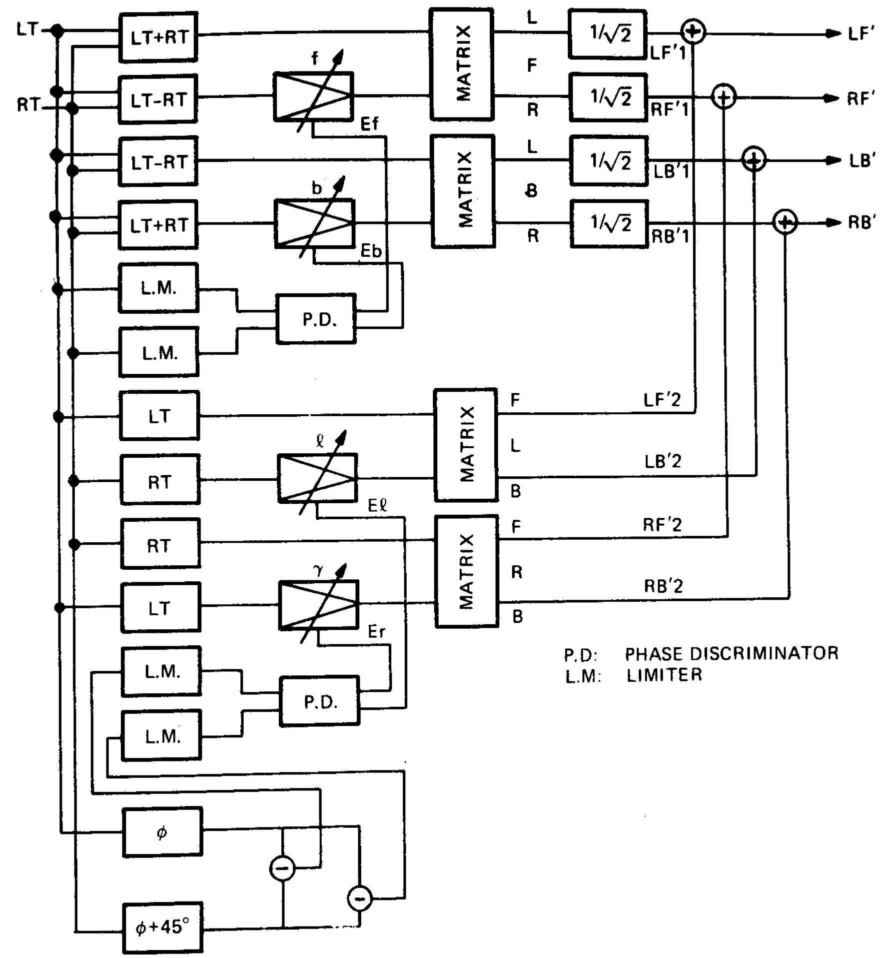

Figure 3. Type A Vario-Matrix block diagram.

The QS implementation of this is shown in Figure 2. Sum signals and difference signals are sent to the front and rear channels respectively. A phase discriminator, not shown, controls the level output of the front Voltage Control Amplifier (VCA) and back VCA. Very important is that these VCAs have a maximum gain of not unity but of 2.42 or 7.7dB gain. So, for a Lf signal, the phase discrimin ator senses an in-phase relationship between Lt/Rt, decreases the output of the back VCA and increases the front VCA to maximum. Not only does this reduce the rear channel level but the extra 7.7dB L/R gain completely eliminates the original 7.7dB QS crossblending. TA-DAH! Complete corner separation. Rb would have been silent anyway, since it is diagonally opposite Lf, a basic virtue of even simple QS decoding.

The aforementioned design was put to use with discrete components in the QRX-3000 series and the QC-01. The QSD-1 and D-2, Photolume decoders and QRX-4000 series receivers carried it a step further and added L/R phase discrimination to F/B. A working block diagram is shown in Figure 3. The two basic V-M circuits were called Type A (F/B and L/R direction sensing) and Type B (F/B sensing only) and are shown in Figure 4. As you can see the differences are quite small, and both can be adapted to synthesize surround, produce ambience effects, or decode SQ with F/B and enhancement, which, using the two different type of decoder circuits provides a choice of six circuit configurations. It's interesting to note that a high quality 'B' circuit for QS, Hall and Surround requires only three ICs, no phase shifters and only a smattering of discrete components. Research has shown me that, in comparing equivalent QS and SQ circuits, the QS will always be simpler. This holds true even if you wanted to apply gain-riding logic to QS (quite feasible), compare surround synthesis circuits, direction sensing, or phase-shift requirements. Even Tate enhancement applied to QS would be less complex than the SQ version.

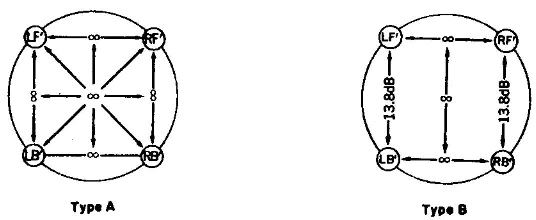

Figure 4. Type A and Type B Vario-Matrix decoding comparison.

While we're on the subject, let's compare Tate to V-M. Even though V-M maintains excellent constant power relationships between the one dominant and subdominant sounds, it accomplishes this at the expense of diffusing the location of the lower level sounds. The crosstalk-cancellation technique used by Tate, on the other hand, leaves subdominant sounds in their properly decoded locations. Comparing stereo synthesis or decoding ability between QSD-2 and a Tate 101, the Tate is **very** amplitude correct but the D-2 has an added space and depth to it. The D-2 will very rarely offend you with noticeable 'logic' action; the Tate will about 10% of the time. This is the trade-off between high separation (Tate at 35-50dB compared to the D-2's average 25dB) and smooth action. The 'Best of Both Worlds Department' says that Tate enhancement applied to Schieber's RM/QS would be magnificent for stereo decoding.

A Vario-Matrix Circuit Journey

Enough with theory; now down to the fun stuff. I can't reproduce full circuit diagrams here, but Sansui can supply copies free of charge.[1] The schematic shown in Figure 5 is representative of a

single-band Type A V-M decoder and will be used as an example. Parts numbers on this diagram correspond to those in the QSD-2, and, because the D-2 is the most common unit, I will concentrate on working with that design. A quick note about the Photolume decoder: produced as a kit by Bob Coleman about 1976, it is almost an electrical twin of the D-2. Missing were the output phase shift circuits, making it essentially a variable matrix RM decoder in the QS mode. Other Sansui clones were manufactured by Compcor (Great Britain) and Vidox (Sweden).

The best way to understand the operation of the D-2 is to start by looking at the chips used.[2] There are three basic types; one is used twice, for a total of four in the circuit. The 1327 is the Phase Discriminator IC; its pinout and circuit are shown in Figure 6a. It consists of one buffer amp, two limiter amps and the actual phase discriminator portion. The limiters start working at 5mV input and function over a 60dB range to ensure the P.D. section isn't affected by amplitude variations in the signal. The P.D. circuit delivers two D.C. voltages complementary to each other and proportionate to the phase difference between Lt/Rt. These are Ef and Eb control signals for the front and back matrix coefficients. In a Type A system, two 1327s are used; the second one produces control voltages representing the absolute ratio between Lt and Rt. This is what the buffer amps are for; they drive the phase shifters to produce Lt and Rt+0^{o} and Lt-Rt-45^{o}. When fed to the P.D. this combination creates the El and Er control signals.

The D.C. signals Ef, Eb, El and Er are filtered to eliminate ripple and fed to IC3103, shown in Figure 6b. These are FETs used as variable resistors in conjunction with the 1328 matrix chip. As a control voltage (such as Ef) increases, the corresponding resistance of the FET decreases. Connected to summing points in the 1328 chip (Figure 6c), this causes gain changes and thus a variable matrix action. The FET method is superior to the common bi-polar transconductance approach found in full-logic SQ decoders. There are two other aspects of V-M you should be aware of. First, even though gain control is employed it is not the same as the signal attenuation found in other schemes. Second, even though V-M is used for corner signal decoding, the direction sensing is based on front/back and left/right differences. These combine for smooth, accurate, high-separation decoding and the FETs used as variable resistors allow for low distortion.

A signal traveling through the D-2 first goes to the PNP transistors Q1, Q3. These input transistors along with switched resistors TR601 and TR602 provide phase blending for the Surround and Hall functions. Two NPN transistors Q2, Q4 provide phase reference for TR601. SQ decoding via V-M requires that the rear blending coefficients (L+R) be phase shifted by 90^{o} compared to other signals present in the 1328 IC. TR602 finishes up modifying the phase-shifted signal. Note that these six transistors have nothing to do with QS decoding and serve only to increase the flexibility of the V-M decoder.

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[1] Sansui Electric Company, 333 West Alondra, Gardena, CA 90247, 213/532-7670.
[2] All three chips are available still from Sansui. Phase Discriminator IC 1327, part no. 03600900, is $23.10; Matrix IC 1328, part no. 03602100, is $19.80; and FET Array IC 3103, part no. 03601000, is $16.00.

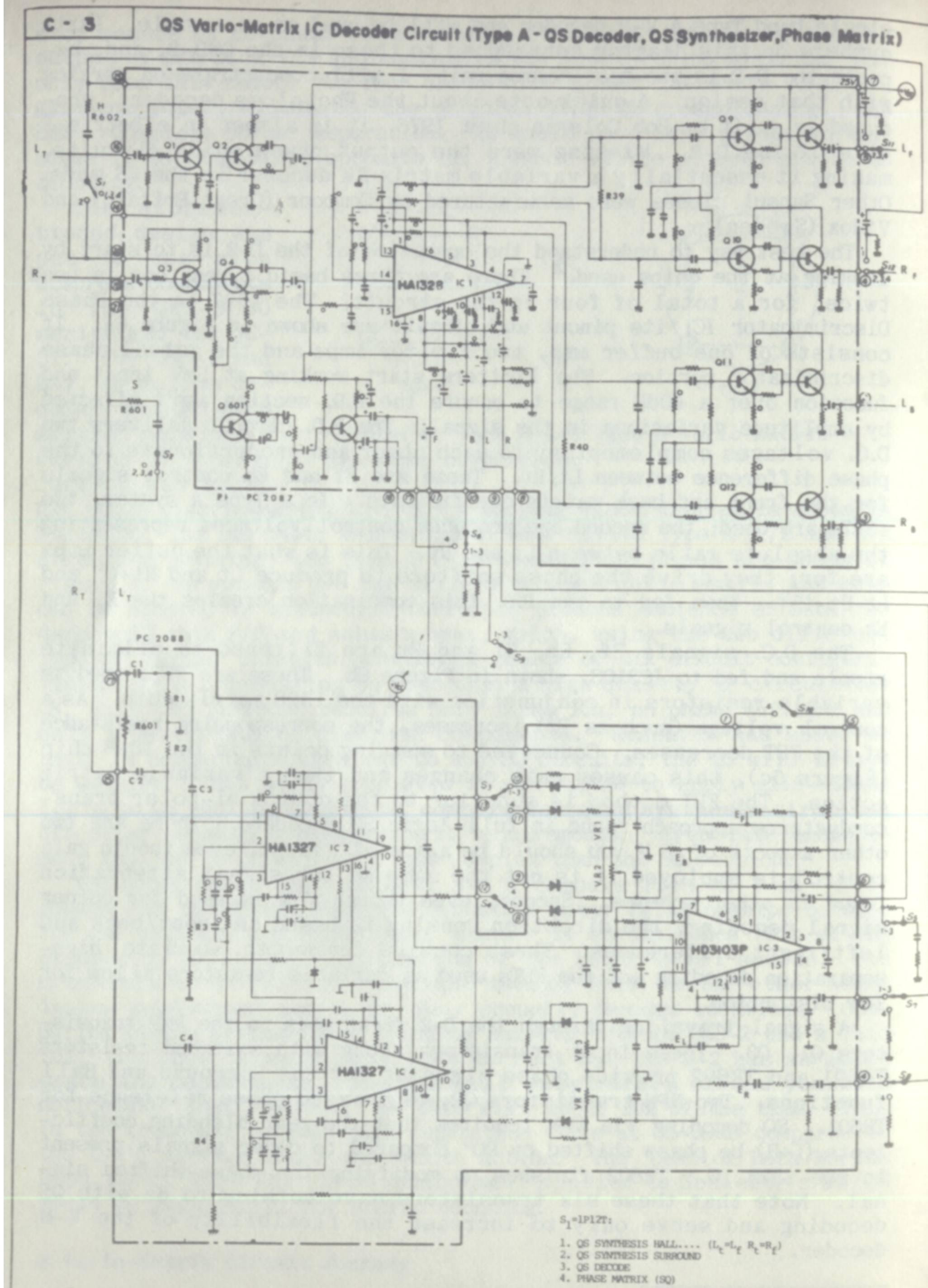


Figure 5.

* Design and specifications subject to change without notice for improvements.

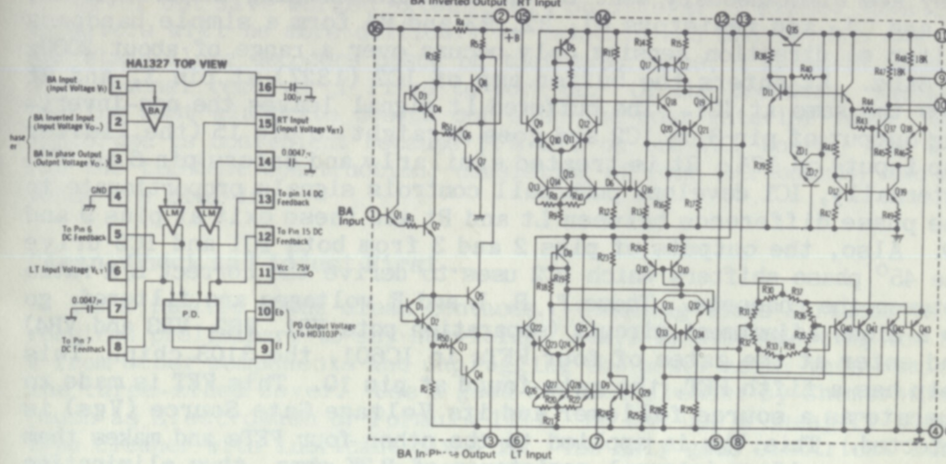


Figure 6a. Phase Discriminator IC 1327 pinout and circuit diagrams.

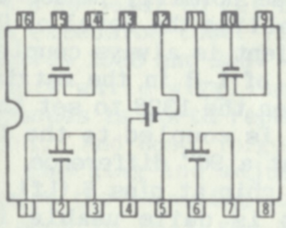
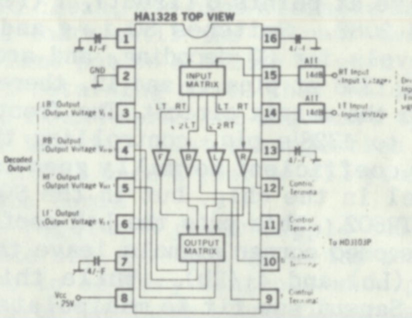
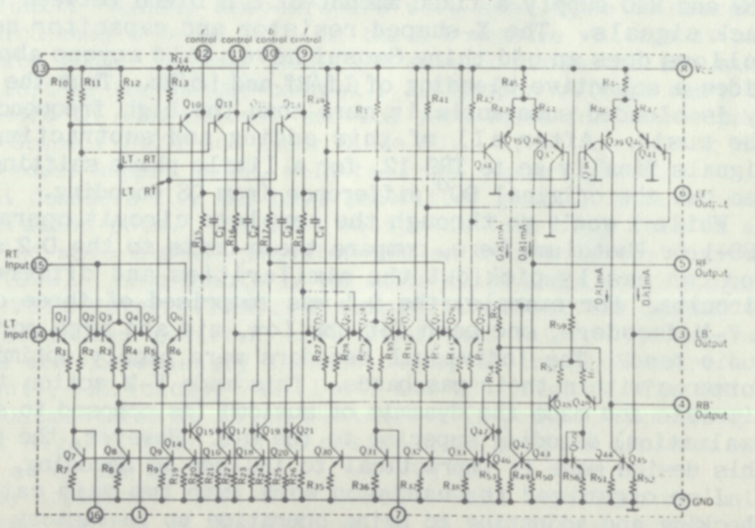


Figure 6b. FET IC 3103.





e 6c. Matrix IC 1328 pinout and circuit diagrams.

While Lt and Rt are entering the matrix IC1328 at pins 14 and 15, they are simutaneously sent to the direction-sensing chips. C1, C2, C3 and C4, and resistors R1, R2, R3 and R4 form a simple bandpass filter so direction sensing only occurs over a range of about 200Hz to 8kHz. Lt enters the buffer amp of IC2 (1327) at pin 1, and Rt does the same at IC1. The buffered Lt signal leaves the non-invert-ing output of pin 3 on IC2 and goes straight to pin 15 (the limiter amp input) on IC1. Rt is treated similarly and enters pin 6 on IC1. Internally, IC1 develops push/pull controls signals proportionate to the phase difference between Lt and Rt and these exit at pins 9 and 10. Also, the outputs of pins 2 and 3 from both IC1 and IC2 drive the 45^{o} phase shifters which IC2 uses to derive the correct L/R ratio between the channels. These F, B, L and R voltages are filtered, go to a diode adjustment circuit (separation pots VR1, VR2, VR3 and VR4) and enter at the gates of four FETs in IC601, the 3103 chip. This chip has a fifth FET, the gate found at pin 10. This FET is made to operate as a source follower and its Voltage Gate Source (Vgs) is detected. This Vgs is applied to the other four FETs and makes them acquire a uniform internal resistance of 2.3K ohms, thus eliminating several independent adjustments. The variable resistances of the FETs leave at points 6 (front), 7 (rear), 2 (left) and 4 (right) on PC Board 2088. Switches SO i, g and h put the L/R coefficient at fixed levels for SQ decoding, and are otherwise normally passed to matrix IC 1328 at pins 11 and 12, thereby controlling the level of Rt and Lt in the matrix circuit. The front coefficient is always coupled straight to 1328's pin, controlling the level of L-R in the matrix. The rear coefficient normally goes to pin 10 on the 1328 to set the L+R level in the chip, but in the SQ mode it is coupled to the IC through TR602. This puts the L+R coefficient at a 90^{o} difference.

The decoded corner signals leave the matrix chip at pins 6 (Lf), 5 (Rf), 3 (Lb) and 4 (Rb). While this output is quite usable in itself, Sansui saw fit to manipulate the signal a little further. R39 and R40 supply a fixed amount of L/R blend between the front and back signals. The X-shaped resistor and capacitor network which follows does an odd thing Sansui never told anyone about: it provides a selective blending of Lf/Rf and Lb/Rb. Thus the low frequency is blended substantially more than the high frequency portion of the music. After all of this adding and subtracting the corner signals finally go to TR9-12, for a little phase shifting to straighten out the original 90^{o} difference from QS encoding.

While I won't go through the complete circuit operations of the QSD-1 or Photolume here, compare these units to the D-2 schematic and you can easily pick out the similarities and differences in the circuits. For example, the D-1 was comprised of three complete Type A V-M decoders, one each for the low, mid and high portions of the audio band. The individual decoders were highly optimised for performance within their passbands. This made V-M action totally unnoticeable and made the dynamic or musical (as opposed to single signal evaluation) decoding superior to the D-2. However, the complexity of this design made it impractical to include SQ decoding.

I've organized the hands-on work into two main categories: 1) checking and adjusting to bring operation up to factory specs, and 2) modifications to improve performance. For ease in performing this work I would rate units in the following order: QSD-1, Photolume,

QSD-2 and QRX receivers. Honestly, I've never touched a Sansui receiver but have worked extensively on others. I expect Sansui's receivers will be more difficult to maintain and modify than any of the stand-alone decoders based on that experience; I've never seen a four-channel receiver I liked to work on. Of the decoders, the D-1 has the most elaborate design but also the most interior room. The Photolume is convenient because it's all on one P.C. board. The D-2 has the tightest construction, consisting of four P.C. boards, two of which are plug-in.

Clean, Check and Adjust First ...

I'm a fanatic about clean contacts. Cleaning these surfaces is a required preliminary to any servicing. Start by disconnecting the D-2 from other components and unplugging the power cord; then remove the three-sided cover. Use a good contact cleaner by Chemtronics (such as Electrowash or Formula 111). Do not use a cheap TV tuner type cleaner with lubricant mixed in. The only good electrical lube I've found is Cramolin; the Red & Blue is an excellent two-step process. Tweek, which lowers contact resistance, is a good product to finish up cleaning with. To clean, use pipe cleaners saturated with solvent, working back and forth until the solvent evaporates. It's not east to get to the volume pot on a D-2 so be sure you have an extension tube. There is a retaining plate that holds in P.C. boards 2088 and 2087 -- remove it and spray the edge card type contacts while wiggling the P.C. boards. Just a suggestion -- this process is worth repeating at least once a year on all your equipment. And don't forget the patch cords.

Checking and adjusting the regulated power supply comes next. Leave the retaining support plate off, but do plug in the power. (Do I need to tell you to be sure to be very cautious when power is applied?) Put electrical tape over the large black filter cap at the back of the unit. Sansui decoders are very sensitive to correct voltages, which should be at 25V. On the D-2 measure voltage at pin 5 on P.C. board 2088. Adjust VR601 (it's well hidden on the main P.C.B to the left of the volume control). If proper voltage can't be obtained, resort to conventional service techniques for series pass regulators. Suspect TR602 and Zener Diode 601. After completing the cleaning and power supply check, reinstall the support plate.

All Sansui decoders I've met have benefitted greatly from adjusting the separation pots. The best equipment for doing this is an adjustable sine wave signal generator and an auto-polarity Digital Multi-Meter (DMM). Also usable is FM white noise and a conventional Vacuum Tube Volt Meter (VTVM). A simple IC circuit is needed to simulate a center-back position; more about that later. The separation trimpots are conveniently located on top of the 2088 board and are VR1 (front), VR2 (back), VR3 (left) and VR4 (right). The trick to these adjustments is to think backwards; if you are inputting a center-front signal, then VR2 is designed to produce minimum and equal output at Lb and Rb.

So start by applying a left-only 1-kHz, 1-volt signal to the unit with the volume control at maximum and the QS position selected. If you have a DMM, put the red lead to Rf output and the black to Rb. To adjust is very simple, just turn VR4 until the readout is zero. This signifies a strict imbalance condition between Rf and Rb. Using a VTVM measure alternately between Rf and Rb until outputs are low and equal. Change to a right-only input and measure from Lf and Lb and adjust VR3. This balances center right. To adjust center back apply equal level signals to Lt and Rt, either from a true mono source or jumpering at the inputs. Rotate VR2 to acquire matching output levels at Lb and Rb. To adjust center front, a center-back position must be simulated. This requires equal-level Lt and Rt inputs with 180° phase difference between them. A simple IC inverter as shown in Figure 7 is the best way to go. Make sure the outputs are equal level and apply to Lt and Rt. Measure Lf and Rf and turn VR1 to achieve minimum crosstalk and equal balance as in the previous steps. If you don't build an inverting circuit and are doing this by seat of the pants, then match the physical position of VR1 to VR2 and hope for the best.

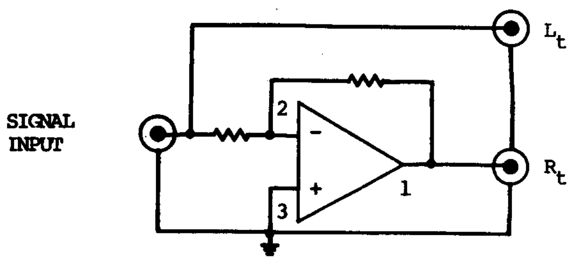


Figure 7. Center-back signal simulator. Resistors = 10K, IC = TL071 or equivalent. Pin 7 = +15V, Pin 4 = -15V.

To tune-up the QSD-1 separation pots requires adjustment in three separate frequency bands. Use signal frequencies of 100 Hz, 1 kHz and 10 kHz. Also, instead of measuring at the output jacks, remove P.C. board 2466 and test at points 11 (Rf), 12 (Lf), 7 (Lb) and 8 (Rb). But, if you test at these points, be sure to capacitor-couple to your meter, and do it the easy way -- hook up a cap to each point to avoid repeated switching. About 1µF caps will do fine.

The separation pots are right on top of the V-M PC Boards and are left to right: right, left, back and front. Set the signal source at 100 Hz and adjust the pots on the low frequency board, 2463; go to 1 kHz and put the mid-range board (2463) through its paces as well; and finally switch to 10 kHz and do the high range board (2465). After adjusting the separation pots, measure the final outputs at the back panel and adjust the individual level trimpots on the mother board for equal balanced levels. Match Rf to Lf while injecting a Cf input, and match Lb to Lf and Rb to Rf with corresponding Cl and Cr signals. Do this only in the QS mode.

The Photolume unit did away with these pots entirely, suspecting no one would be able to trim them correctly. You can either leave them as is or replace the fixed resistors with trimpots and adjust like the D-2.

Having accomplished all the preceding steps is reason to celebrate. Excluding any gross circuit malfunction, you basically have a brand new, up-to-snuff Vario-Matrix decoder. Before doing any more work audition your unit and listen for the difference. If something still seems not quite right check the IC pin voltage as shown in Table 1. Replace any suspicious ICs (use sockets for replacing); then check the discrete devices. You probably won't have any problems, however, as the Hitachi chips were very high quality and the design in general is a safe and good one.

Pin Number	*IC1*	*IC2*	*IC3*	*IC4*
1	*5.5*	*7.2*	*7.1*	*22.0*
2	*0*	*17.8*	*17.8*	*14.2*
3	*7.3*	*6.0*	*6.0*	*22.0*
4	*7.3*	*0*	*0*	*22.0*
5	*7.3*	*4.9*	*5.1*	*22.0*
6	*7.3*	*4.6*	*4.5*	*14.2*
7	*5.0*	*4.6*	*4.5*	*22.0*
8	*25.0*	*5.1*	*5.1*	*23.3*
9	*12.0*	*16.1*	*16.1*	*22.0*
10	*11.7*	*16.2*	*16.2*	*16.2*
11	*11.7*	*25.0*	*25.0*	*4.0*
12	*11.7*	*5.0*	*5.2*	*13.7*
13	*12.0*	*5.2*	*5.1*	*22.0*
14	*7.6*	*4.8*	*4.5*	*22.0*
15	*7.3*	*4.7*	*4.5*	*13.8*
16	*4.8*	*13.8*	*13.9*	*22.0*

Table 1. IC pin voltages with no signal applied.

... Then Modify

Depending on what you are willing or want to do, modifications and upgrading can be very simple, or complex, especially if intended to provide greater versatility or fidelity. What follows are suggestions only — the most useful ones from my actual experience. But, before you try any, I'd suggest a simple precautionary measure -- to provide spike and high voltage transient protection for the unit's power supply. GE makes several varistor units to accomplish this, among them the GE-750, which is wired in parallel across the main voltage input line. The RCA SK-400 protector, on the other hand, works like an adaptor at the plug/socket. You don't think it's needed? Try a scope across the 120v house line and see how 'dirty' it is. Turn your vaccuum cleaner, or whatever, on and off for a real scare. This varistor protection is valuable also for the Tate II, especially very early units, which the manufacturer suggested leaving powered continuously. Computer filters offer no more protection at a higher cost, and other protective circuit measures (Zener protected inputs/outputs or a fused +v out) I consider overkill.

How can I mention sonic improvement without considering coupling capacitor replacement? The D-2 has about 24 truly low quality electrolytic/tantalum caps in its signal path, the D-1 more than triple that number. You'll have to decide for yourself if these capacitors degrade performance. My own opinion is an emphatic **yes**! For those wishing to embark on wholesale capacitor upgrading here are some tips.

In order of desirability for high performance audio capacitor materials are as follows: Teflon, Polystyrene, Polypropylene, Polycarbonate, Mylar, Electrolytic and Tantalum. Mica and ceramic aren't used for coupling purposes and so don't apply. Availability, regretably, is almost the reverse of desirability, and, as you try to find the larger capacitor values, it usually pushes you into the less desirable types. A general rule on values is to select equal or greater capacitance and working voltage than specified in the schematic. My favorite general purpose replacement cap is sold by ETCO.[3]

[3] ETCO Electronics, North Country Shopping Center, Rt. 9 North, Plattsburgh, NY 12901, 518/561-8700. additional suppliers for these and other parts include Mouser Electronics, 11433 Woodside Ave., Santee, CA 92071, 619/449-2222, and Old Colony Sound Lab, Box 243, Peterborough, NH 03458.

It's a CDE metallized polycarbonate, 4 µF at 200V. This cap cost me $4.50 about six years ago, brand new, but thanks to 'cosmetic defects' ETCO sells them for only 49 cents each! And not one I've purchased from them has been defective. If you are a fanatic, a smaller polypropylene or polystyrene can be wired in parallel around it. The size of the 4.0 µF cap is about .5" X 1.75". The question of just where to put them all is important. The QSD-1 is the easiest since there's plenty of room on back of the smaller P.C. Boards or on the mother board. The Photolume can be raised on spacers and the caps placed underneath. The D-2 is tough and the best I've come up with is to mount a small companion enclosure to the side. A plain perf board can hold the caps and wiring will connect it. I'd suggest two-conductor with shielding, grounded at only one side. Don't forget the caps that couple the 1328 chip to the FET 3103 chip. Caps to be replaced are as follows: on the 2087 chip, C1, C2, C7, C8, C12, C35, C36, C37, C38, C604, C605 and C611; on 2612, C1, C2, C3, C4, C5, C6, C11, C12, C13, C14, C601 and C602; and on 2088, C39 and C612.

The musical benefit is mainly in the mid-range and higher frequencies, and any waveform with rapid attack/decay. You'll hear it on harpsichord notes, and notice violins separate into individual instruments. I've never heard 'better bass', improved dynamic range, or less noise from using high quality caps; the difference has always been a matter of transients, detail, definition. Once again, compare the similar circuit points of a Photolume or D-1 to the D-2 schematic to see where the equivalent points are.

The fact that fixed resistors are switched in and out to facilitate Surround and Hall offers another option for improvements: replace these with continuously variable pots. Resistor R601 (5.6K) accomplishes out of phase blending between Lt and Rt for Surround and R602 (3.9K) produces in-phase blending for Hall. These are inconveniently located by the mode switch and are hard to get to but well worth the trouble to replace. You can arrange pots (dual, single), with or without series resistors, in several combinations but one way I've found works best. Use one dual, 10K linear pot wired as shown in Figure 8, with a 900-Ohm resistor in series on each side. This

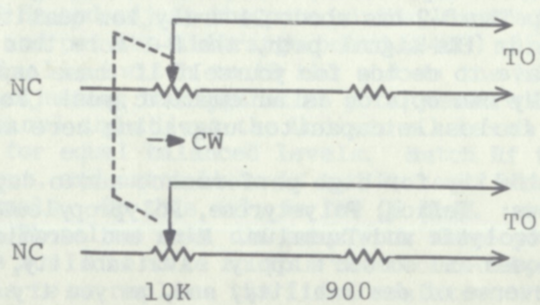


Figure 8. Dual, 10K linear pot for use as described in the text.

evens out the unequal amount of blend between Hall and Surround and gives you a wide range of smooth control. When set at 12 o'clock it's close to the original factory amount of Surround blend; and 2 o'clock puts it back to the original Hall blend. Rotate clockwise for increased blend and note two special items: this contol has no effect in the QS mode, and when in Hall Lt and Rt are passed to Lf and Rf totally undecoded.

The remaining mods are the simplest of all. They involve merely clipping or removing components to change the factory operating parameters. First to go is R39 and R40 on PC board 2087. This will result in wider L/R separation at the expense of a very small loss in F/B separation. If you're not interested in SQ decoding, remove C5 and C6 on the same board and you'll eliminate the low-grade phase shift prior to IC1328. On PC board 2088 clip R601 (220K). This blends Lt and Rt before the signal goes to the phase discriminator ICs, reduces L/R separation and increases the front channel activity. Removing it gives the 1327 chips more sonic information to grab on to. Last but not least, if you primarily play stereo sources through your D-2, the output phase shift is unnecessary and actually degrades the center left or right imaging. Back to PC board 2087, remove caps C31, C32, C33 and C34 to eliminate this phase shift.

Postscript

Is there no end to the versatility of Vario-Matrix (or, for that matter, to this article)? While customizing a friend's Photolume unit, I discovered a few more options worth passing on. My friend is devoted to synthesizing from stereo, and he has usually approached this by decoding (from stereo, SQ, etc.) and encoding to QS. However, there is a more direct and desirable route to take. You can simply tap off the synthesis circuit in the QS decoder and pass it on as Lt and Rt. On a QSD-2 this can be done at P.C. board 2087. Point 14 is the left output, and point 21 the right. Because DC bias voltages are present, you must capacitor-couple to the output jacks, which could easily be the tape-out jacks on the back panel. When done in this fashion it is more appropriate to consider this a 'phase balance' control. Just as the common L/R balance is really an amplitude control, using these outputs as described controls the in-phase to opposite-phase balance. When in the Hall mode you control the in-phase relationships; in the Surround mode you determine reverse phase blending. The obvious use is for creative control in recording. I was also delighted to find it helped substantially with my Tate II in the Surround mode.

While a continuously variable pot is most desirable for controlling the phase blend options, there is another alternative. Replace the fixed resistors R601 and R602 with internally-mounted 15-turn trimpots. Once phase blend is critically adjusted, changing the mode switch also selects the blend you prefer; you won't have to readjust it each time. Sansui had a different amount of blend in all their units and none was really correct. To fine-tune the phase for Surround, replug the decoder into your system after completing the modifications and apply a left-only signal to the input. A test tone works best, but FM white noise or music is fine, too.

Now adjust all output balance controls so you are monitoring only the right-front speaker. With the unit in the Surround mode, adjust the pot replacing R601 for absolute minimum output from right front. Most of the crosstalk will be in the very low frequencies. Notice that if you turn the trimpot too far you'll cross a break point where you hear a large and rapid increase in level. Just before this point is the desired reference point for maximum Surround separation; a front-panel control would allow deviation from this point to suit the program needs. To adjust for Hall, switch to this mode and listen to Rb only. Tune the corresponding pot for minimum output. When going through these steps, let the decoder season by playing music through it for about twenty minutes, a practice you should follow prior to tweaking the internal separation pots, too.

Earlier I suggested removing R39 and R40 from the output stage to increase L/R separation. A more useful design will change their values and make them switch selectable. R39 is optimum at 100K and R40 at 10K. A DPDT wired in series with these resistors will bring them in or out of the circuit. I enjoyed this the most when listening to classical music in the Hall mode with minimum blend. The net effect was to impart a very stable L/R soundstage to music that doesn't really need aggressive enhancement. In the Surround mode I invariably preferred the resistors switched out. ****

References

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EDITORIAL (continued from page 1)

but let's not quibble about that. After all, it didn't stop us last time. Besides, it looks as if they put some genuine thought into the matter when they chose their word.

So, my vote goes to 'ambiphile', or more specifically 'audio-ambiphile'. It is alliterative, concise, and, even if it does come close to someone's trademark, means exactly what we want it to — 'someone who likes sound on all sides' -- with no restrictions on how, or with how many channels, we do it. ****

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ENCODE/DECODE (continued from page 4)

(858 E. Congress Park Drive, Centerville, OH 45459, 800/762-4315) has two Audio-Technica cartridges for sale: the AT-15SS (MCM part no. 39-080) for $87.85 and the AT-14SA (part no. 39-070) for $46.50. They also carry the replacement stylus for the Panasonic EPC-450C-II and accept VISA and MasterCard.

About Scott Henderson's letter, I too prefer music all around rather than ambience. With discrete reels and discs gone, and SQ the only matrix system seemingly in use, perhaps new material can be mixed this way. But I would like to see four-channel discrete Compact Discs, and I'm sure many other quadraphiles would, too. ****