009

ORBAN/PARASOUND STEREO MATRIX MODEL 300B

OPERATION AND MAINTENANCE MANUAL

orban

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12/3/71

THE ORBAN/PARASOUND STEREO MATRIX

Operation and Maintenance Manual

Introduction:

The Orban/Parasound Stereo Matrix is an instrument of absolutely unique design. It is a new concept, quite as fundamental as a program equalizer or compressor. Just as a program equalizer permits you to manipulate the frequency response of a signal, and a compressor permits manipulation of dynamics, the Stereo Matrix permits manipulation of the stereo field through control of the frequency response and the phase response of the two channels with respect to each other.

Since the concept of the Matrix is almost totally unfamiliar, it is imperative that the contents of this instruction manual be read and thoroughly understood before the instrument is put into use. Failure to do this can only result in less than optimum results, or even in disaster.

Concepts:

The Matrix has three operating modes.

1) **Matrix:** In this mode, the Matrix creates stereo space from a number of mono-miked sources. The source of the mono-miked tracks is ordinarily the master tracks of the 4, 8, or 16 track master. The Matrix has five inputs: left, right, reverb, synthesized center, and phase-shift center. Inputs to any of these are spread spatially about a nominal point in the stereo space. The spreading is frequency-dependent (that is, different instruments applied to the same input will appear in different places) in all inputs except for phase-shift center. The effect of the phase-shift center is to diffuse the apparent width of the track about the center.

Left spreads from left to center, right from center to right, and reverb widely about the center, with full separation. The synthesized center spreads about the center in a frequency-dependent manner, except that all components below about 200 Hz appear equally in both channels. Separation is less than in the reverb input; this means that the synthesized center material appears more tightly bunched up in the center.

The average ratio of difference to sum generated by the left and right channels is frequency-dependent, and varies between 0 and 1.00, averaging about 0.65. This ratio is 1.00 in the reverb channel and 0.65 in the synthesized center input, both frequency independent. This assures that the stereo signal will be positively correlated, and that the lateral modulation will exceed the vertical. In addition, the matrix has been designed so that the mono mix from the stereo output will have the same musical balance as the output reproduced stereophonically, eliminating the classical 3 db center-channel buildup.

2) **L/R Quadrature:** This mode is used to create compatible stereo from stereo mixed with conventional pan-pot techniques. The conventional stereo is introduced to the L and R inputs; the compatible stereo appears at the L and R outputs. Internally, the Matrix shifts the phase of one channel 90° with respect to the other, giving the well-known "quadrature split".

3) **Center Phase Split:** Here, the L and R inputs are introduced directly to the L and R outputs. A phase shift, adjustable by the front panel **center phase** control from 0° to 90°, may be introduced between the components of the phase-shift center input which appear in the left and right outputs. With 0°, the classic 3 db center-channel buildup in the mono mix of the stereo output will occur; with 90°, it will be entirely absent. A 90° shift will, however, substantially spread the apparent width of the center component. In the Matrix mode, it is only necessary to use a 65° shift, so this spreading effect is substantially reduced. The Center Phase Split mode has been supplied only as a convenience to those who are conservative and who wish to fall back on a technique which has already appeared in the liter-

ature. The stereo space generated by the Matrix mode is, in our opinion, far more effective than the stereo space generated by the center phase split technique.

Philosophically, the Matrix is an attempt to generate the kind of stereo effect which used to be generated by the classical technique of miking with two or three mikes in the same acoustic space. This technique is known to give an excellent sense of acoustic space and environment, but it provides no effective means of controlling instruments, and mono mixes are usually subject to acoustic phase distortion.

The multi-track technique, conversely, provides excellent control of the individual tracks, but the "pan-pot" technique of placing tracks provides little more than a series of little mono recordings, spaced one-by-one (or, more crudely, bunched left, center, and right) across the stereo stage. The necessary phase and frequency differences to provide an illusion of real space are absent.

The Matrix replaces pan-pots with a far superior space-generating system, which gives greater apparent loudness, greater psycho-acoustical instrument separation, and complete and totally predictable control over the mono sum signal, resulting in no phase distortion problems under any operating conditions, providing only that each of the inputs is presented with completely independent program material.

Installation:

The inputs to the matrix are 10K balanced bridging, with a nominal input level of +4 dbm. The outputs are designed to be loaded into 600 ohms, and terminated under all operating conditions. They are balanced, and clip at +26 dbm into 600 ohms. While the amplifiers are protected against output short circuits, these are to avoided if possible.

Because the Matrix is a very complex active filter, a certain amount of noise is generated by the circuitry. If the unit is used correctly, overload/noise ratios in the order of 90 db are obtainable. Output noise is approximately -65 dbm; overload point is +26 dbm. A 14 db overload margain would call for peaking the output at +8 VU or +12 dbm. Operating at +4 dbm results in a signal/noise ratio of about 69 db and a peak factor of 22 db.

Bypassing the output transformer gives an overload point of +20 dbm and noise of approximately -71 dbm. This is a far better operating point for +4 dbm nominal levels. However, we were inspired to go to +26 by the current insanity in the industry which insists that amplifiers be capable of delivering undistorted power some 10 db beyond total tape saturation, where no VU meter is guaranteed to remain undamaged. Users must make their own decisions on these matters.

If the unit is used without transformers, it is unnecessary to terminate the outputs. In addition, midband distortion figures will be maintained down to the lowest frequencies.

The unit has slightly less than unity gain from each of its inputs to the output.

It is essential that gain balance be maintained between the output channels. If this is not done, mono compatibility and stereo frequency balance will both be adversely affected. If the phase shift center is used, the channels should have identical phase response; otherwise, phase cancellations will occur in mono mixdown. The other four inputs have considerably greater immunity to channel phase errors. For this reason, it is advisable to use the synthesized center for cymbals and other instruments particularly sensitive to degradation by phase distortion if it is desired to place these intruments in the center.

Operation in Matrix Mode:

The multi-track master should have its individual tracks operated on for equalization, compression, etc., before application to the matrix. The tracks are applied directly to four inputs: left, right, and the two centers. The producer should choose roughly where he wants the tracks to appear, and these tracks should be introduced to the suitable inputs by mixing through the board down to four tracks. Under no circumstances should a master track be split between any of the four tracks.

It is recommended that the soloist or featured instrument be introduced into the phase center, since this is the only input that it is easy to hold on dead center. The synthesized center will tend to spread its material to the left or right of center.

The dimension controls are essentially frequency-band pan-pots, and they are used to adjust the left-right balance within each of the spread fields from each of the inputs (except for phase-shift center). The operator should familiarize himself with their effect by manipulating them while listening to one input at a time. In operational practice, they are used to get a smooth, even, and well-balanced stereo field. Their adjustment should be different for input material with different spectral contents.

The reverb input should be fed by the echo return bus. It has the effect of spreading and diffusing the echo return throughout the entire stereo field, thus simulating reverberation in a hall with good acoustics. Note too that this input is not entirely mono-compatible. The original mono input to this reverb input will appear in the mono mix, but it will be down 1.5 db in level with respect to its perceived level when listened to stereophonically. This is necessary to obtain maximum separation and width of spread. It was felt that the mono mix could, in most cases, well stand the increase in clarity that a loss of 1.5 db in reverb level could bring. If the producer feels that it is absolutely necessary to maintain mono/stereo level equality, then the reverb signal could be introduced into either of the center inputs.

The reverb input alone is equivalent to a standard Orban/Parasound Stereo Synthesizer (without separation control), and may therefore be used to electronically process old mono material into effective pseudo-stereo.

The center phase control on the front panel adjusts the phase angle between the components of the phase center input which appear in the left and right outputs. This control is variable from 0° to 90°. As it is rotated from 0° to 90°, the loudness of the phase center component appearing in the mono mix of the output channels will stay constant, but the loudness of the phase center component listened to stereophonically will increase smoothly up to +3 db at 90°. Simultaneously, the apparent width of the phase center source will increase from a point source (at 0°), to the full width between the speakers (at 90°).

In order to obtain mono/stereo compatibility, the center phase control (which operates in both the Matrix and Center Phase Split modes) must be set at 65° in the Matrix mode and 90° in the Center Phase Split Mode. In the L/R Quadrature mode, it is removed from the circuit.

The purpose of having the control is to permit adjustment of the mono/stereo loudness ratio of the phase center component. For example, in the Matrix mode, a setting of 0° will cause a 1.5 db center channel buildup in the mono mode; a setting of 90° will cause a 1.5 db center channel dropoff. In the Center Phase Split mode, reducing the shift below 90° will cause greater and greater mono center-channel buildup, until a 3 db buildup is finally reached at 0°.

The mix should be monitored through the Matrix, and all operations on the signal should be made while listening to the Matrixed output. Mono compatibility will be automatically maintained provided the instructions above have been followed. The only front panel control (other than the mode switch) which affects mono compatibility is the center phase control.

Operation in the Center Phase Split Mode:

A classical center phase split mix is made by mixing down to a left, right, and center, and introducing these into the left, right, and phase shift center inputs. The center phase control is set at 90°. Left components appear as a point source from the left speaker, right components appear as a point source from the right speaker, and center components appear diffused about the center. In terms of stereo effect, there is no particular improvement over pan-pot techniques, and the only advantage is full mono/stereo musical balance compatibility. However, this technique generates more vertical component than the Matrix technique, so its mechanical compatibility is reduced with respect to the Matrix technique.

It should be noted that any signal introduced into the reverb input in this mode will be fully mono/stereo compatible. However, use of the reverb input (although highly recommended) will take the technique beyond the standard center phase operation.

Operation in the L/R Quadrature Mode:

This mode is to be avoided if at all possible, because it passes left and right components through phase shift networks which will affect transients. However, it is the only way available to process a pan-potted two-track mix into a form which makes it stereo/mono musically compatible. It will also have the effect of spreading the apparent width of the center component, just as the Center Phase Split mode does, and for identical reasons.

To use the Matrix in this mode, apply the left and right pan-potted master tracks to the left and right inputs of the Matrix. The processed signal will appear at the outputs. No adjustments are possible or necessary.

By adding the left and right outputs in this mode, it is possible to derive a mono signal from a standard pan-potted stereo master in which the center channel does not build up 3 db.

Maintenance:

The circuitry of the Matrix is to be found on four plug-in printed circuit cards (AP-1, AP-2, AP-3, and SM-4), as well as on one power supply card which is mounted on the side of the Matrix chassis with screws and standoffs. All of its wires clip on, and if the power supply is ever removed, notes should be made of which wires are connected to which slots in the board so that the assembly can be correctly reassembled.

A full set of drawings has been supplied with this manual, showing the mainframe wiring (in block form), as well as showing detailed schematics of each of the circuit cards.

The system diagram illustrates the overall topology of the system. Each of the five inputs is introduced into its own bridging transformer (T1-T5). S1 is the mode switch, and serves to switch the left and right inputs to the left and right processors on AP-2 (matrix mode), to the two sections of the quadrature generator, AP-1 (L/R Quadrature mode), or directly to the output summing amplifiers (Center Phase Split). S1A and S1B are input selectors to AP-1, and choose either Phase Center to both phase-shifters (Matrix or CPS), or Left to the upper phase-shifter and Right to the lower phase-shifter (L/R Quad.). The phase-shifters on AP-1 are arranged so that the difference between the phase shift of the two sections is approximately 90° from 20 to 20,000 Hz.

S1C and S1D serve to route the Left and Right input signals either to the left and right phase-shifters on AP-2, or directly into the summing bus on SM-4 through pins 9 and 15. S1E serves to disable the center phase control in the L/R Quad. position. The center phase control is a gain control for the top phase-shifter in AP-1. In the Matrix and CPS modes, this top phase-shifter serves as a difference signal (note that its output is a phase-splitter), while the bottom phase-shifter supplies a sum signal. By varying the ratio of sum-to-difference, phase shifts from 0^o to 90^o may be obtained by matrixing in the combining busses of SM-4. S1F and S1G switch the output of AP-1 so that it is the aforementioned sum-and-difference feed in the Matrix and CPS modes, but so that each phase-shifter goes straight through to the left and right outputs respectively in the L/R Quad. mode. In the Matrix and CPS modes, the Phase Center input feeds both phase-shifters on the AP-1 card.

The Synthesized Center and Reverb inputs feed their respective phase-shifters on AP-3 without switching.

The outputs from all the phase-shifters eventually appear at the inputs to SM-4. This card consists of four phase-splitters (sometimes called "phase inverters"), and two operational amplifier combining circuits. These opamps are sufficiently powerful to drive +26 dBm into 150 ohms, and therefore also serve as output program amplifiers.

The frequency bands which the stereo synthesis technique generates in the left and right channels are caused by phase additions and cancellations in the combining amplifiers. The outputs of each of the phase-shifters in AP-2 and AP-3 are applied to their own phase-splitters, which has two outputs differing in phase by 180^o. These phase-shifted outputs are then used as difference signals to feed the matrix. The sum signals come directly from the inputs (through a single emitter-follower) except for the Synthesized Center input. Here, the sum signal is also processed through a phase shifter located on the AP-3 board.

The time constants of the phase-shifters are adjusted by means of the dimension controls. After matrixing, the effect of changing these time constants is to move certain frequencies from one output channel to the other.

The output opamps drive a 150 ohm to 600 ohm stepup transformer, which is connected to the output connectors.

AP1, AP-2, and AP-3 consist of a number of phase-shifters, all of which are realized substantially identically in terms of circuit layout. The only differences between the phase shifters are the choice of time constants, and the fact that the upper part of AP-3 has a two-stage added sum-signal phase shifter.

The phase-shifters consist of a number of bridge-driven passive RC single pole/ single zero phase shifters. It can be shown that networks of this type, given infinitely high load impedance and drive by ideal transistors, will have a frequency response that is flat in amplitude but with a phase response varying from 0^o to 180^o as a function of frequency. The radian frequency at which the phase shift is 90^o is equal to 1/RC, where R and C are the series resistor and capacitor respectively. The value of the transistor load resistors (3.9K) does not affect this time constant. In our circuit, the ideal transistor is closely approximated by a conjugate NPN/PNP pair. The pair exhibits very high current gain, very high input impedance, and very close to unity gain in this particular phase-inverter circuit. Distortion and noise are virtually non-existent. By alternating NPN and PNP types, it is possible to get DC level shifting which makes direct-coupling practical. It should be noted that the conjugate transistors behave like a transistor of the input type (NPN or PNP) of the pair, and that the second transistor of the pair is controlled wholly by feedback across th emitter resistor. The emitter of the second transistor of the pair is effectively the collector of the pair.

Faults are extremely unlikely in AP-1n AP-2, and AP-3. However, any faults which occur will probably be transistor failures, and these can be traced like any other failures. The transistor types used (2N4123 or 2N4125) were chosen because of their low-noise properties, and these transistors should be replaced with exact replacements. There are so many active stages in the Matrix that even small noise contributions from each transistor could rapidly add up if not kept under control. If any of the 3.9K resistors are replaced, care should be taken to avoid noisy replacements.

This brings us to a discussion of the SM-4 card. The four phase-splitters on this card are identical in form to the phase-splitters on the AP cards. The output amplifiers deserve some discussion, however.

The output amplifier is basically a power operational amplifier operated in the inverting summing mode. Special steps are taken to isolate the DC level on the summing bus from the bias point of the opamp.

The opamp consists of a uA749c dual IC operational amplifier with an added discrete complementary symmetry output stage. We will discuss the left circuit; the right is identical.

The output stage of the uA749c is a PNP transistor with no collector load. The collector load is external in this circuit, and consists of the two diode-connected transistors Q409 and Q411, plus the load resistor R441. Q409 and Q411 serve as bias diodes for the output transistors Q410 and Q412, and each is thermally connected to its output transistor to prevent thermal runaway. Bias in the output stage is determined by the characteristics of all four transistors in association with R439 and R440, which provide stabilizing DC feedback. In addition, they work with CR401 and CR402 to limit short-circuit current through the ouput stage to 100ma, thus protecting the stage from damage.

The Matrix is powered from a single-polarity power supply. The supply voltage for the output amplifiers is +32 volts. The desired quiescent voltage at the output is +16 volts for a symmetrical clipping point. This voltage is generated by the voltage divider R436/R437, which provides a +16 volt reference for the non-inverting inputs of both output amps. The output is held at the proper voltage by means of the DC feedback loop consisting of R435, C411, and R434. AC feedback to the summing junction is provided by R433/C403. The combination of C401 and the DC feedback loop with C411 forms an active 12 dB/oct highpass filter which rolls off below the audible range. Values of the components have been chosen so that the filter is stable and does not exhibit low-frequency ringing. The high-frequency compensation of the amplifier is done with C402 and R438, to prevent supersonic oscillations.

The AC feedback into the summing junction forms a perfectly conventional active combining network that needs no further discussion.

The only circuit left to discuss is the power supply. The AC from the power transformer is rectified by the bridge CR501-504 and applied to an electronic regulator circuit. The first part of this regulator consists of a +32 volt supply with a uA723 IC voltage regulator and external series pass transistor, Q501. This supply supplies the output amplifiers directly and all the other electronics through two post-regulator stages, Q502 and Q503. The +32 supply is current-limited to about 220 ma, which is sufficient to protect it against any condition of short-circuit of the output stage. HOWEVER, THIS REGULATOR AND THE OTHERS WILL NOT WITHSTAND DIRECT DC SHORTS TO GROUND! BE CAREFUL

WHILE TROUBLESHOOTING; a short will destroy Q501, Q502, Q503, or a combination of these! The power supply was designed for minimum high-frequency output impedance; this made necessary use of somewhat delicate devices. These devices will never fail under ordinary operating conditions, but can be destroyed by internal shorts.

R502 and R503 form a voltage divider which compares the output voltage with the reference voltage at pin 3 of IC 501. C503 frequency-compensates the internal opamp in IC501.

Q502 and Q503 form two independent postregulator circuits which reduce the residual ripple from the +32 volt supply to a few microvolts. These supplies supply all power for the many transistor phase-splitters in the Matrix circuitry, since these are rather sensitive to power supply disturbances. The outputs of these regulators should have less than 20 microvolts ripple and noise. They are essentially capacitance amplifiers. The +32 volt supply should have no more than a few millivolts ripple and noise. If the +32 volt supply appears to have failed, chances are that the failure is Q501, not IC501. Similarly, failure of either +30A or +30B is almost certainly due to failure of Q503 or Q502. Any inexplicable increase in hum should be diagnosed by checking power supply ripple and noise.

Because of the power transformer used, it is not possible to restrap the unit for other than 110-120 volt AC operation. Operation at other power line voltages must be accomplished through the use of an external transformer.

Factory Service:

The factory will be happy to perform checkout and maintenance of the Matrix throughout the life of the unit. After the initial warranty period, there will be a reasonable charge for labor and materials.

Warranty:

The Orban Associates Division of the Kurt Orban Co., Inc., guarantees the Orban/Parasound Stereo Matrix for one year from date of outright purchase, or 90 days from time final payment is made, against defects in materials and workmanship, and will repair any such defect without charge for parts or labor, excepting only shipping charges. This warranty is void if the Matrix has been subject to physical or electrical abuse, or operated contrary to the instructions in this manual.

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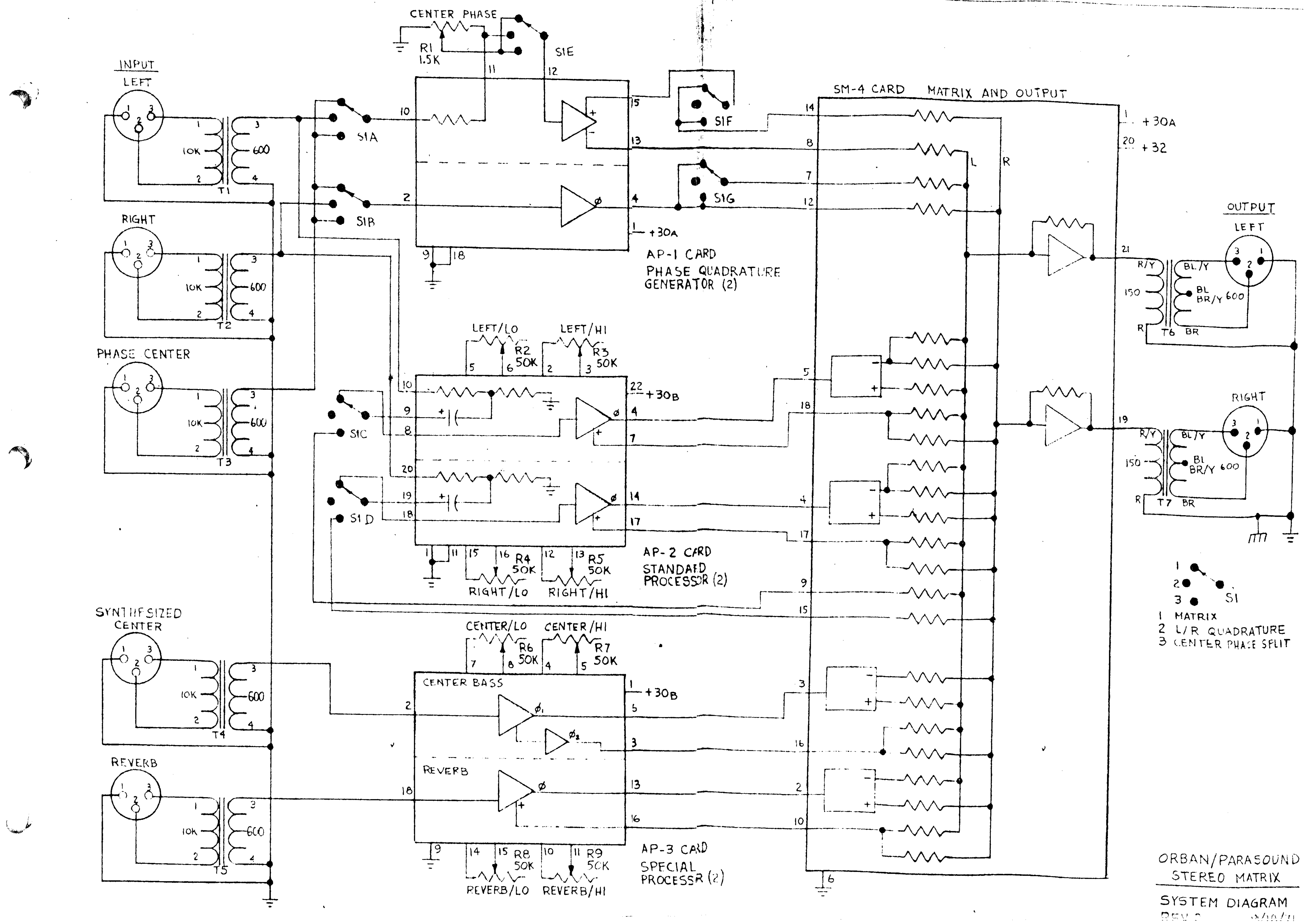
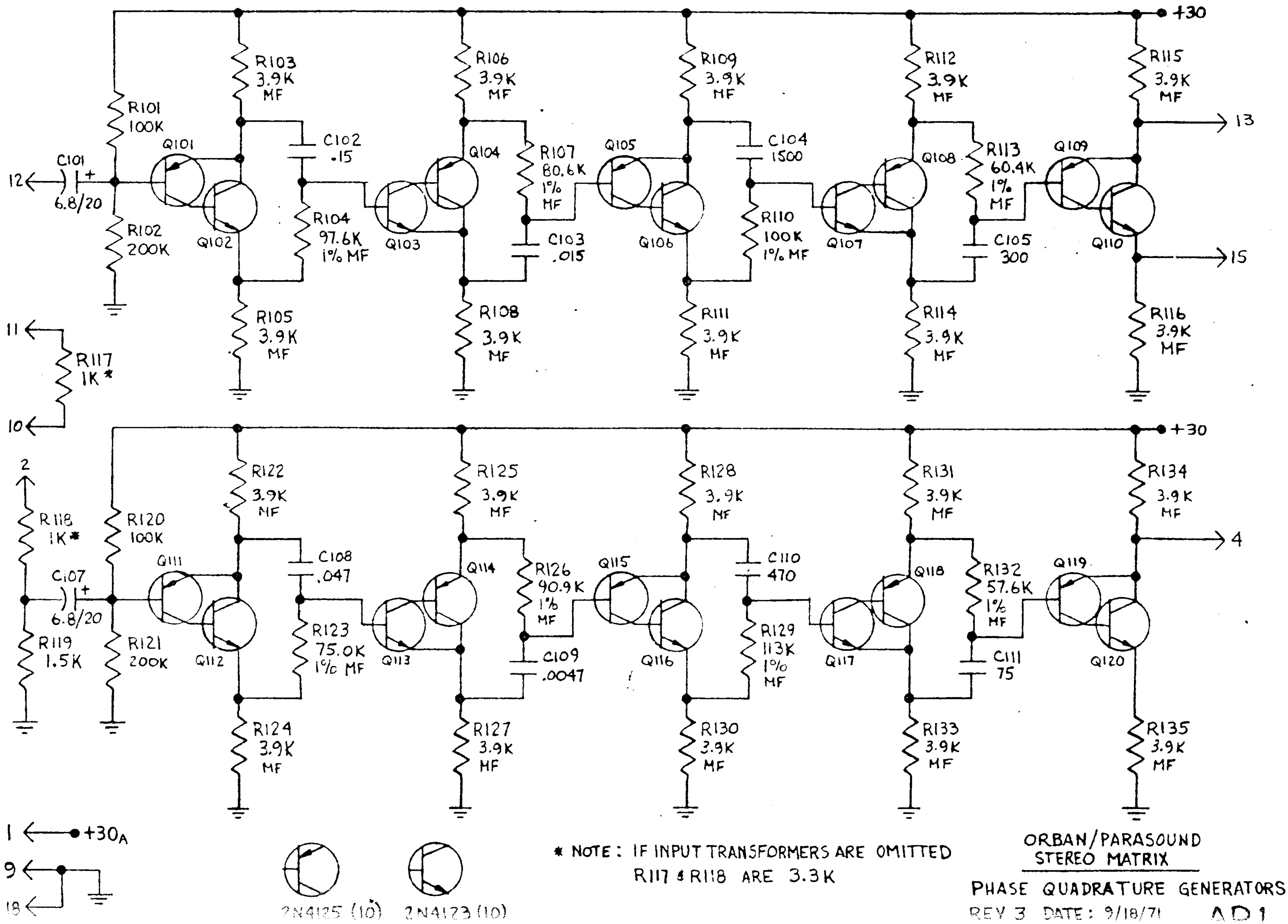

CENTER PHASE
R1
1.5K
S1E
INPUT
LEFT
10K
600
T1
S1A
S1B
S1F
S1G
SM-4 CARD
MATRIX AND OUTPUT
+30A
+32
L
R
RIGHT
T2
+30A
AP-1 CARD
PHASE QUADRATURE
GENERATOR (2)
OUTPUT
LEFT
R/Y
BL/Y
150
BL
BR/Y 600
T6
BR
PHASE CENTER
T3
LEFT/LO
R2
50K
LEFT/HI
R3
50K
+30B
S1C
S1D
RIGHT
R/Y
BL/Y
BL
BR/Y 600
T7
BR
AP-2 CARD
STANDARD
PROCESSOR (2)
R4
50K
R5
50K
RIGHT/LO
RIGHT/HI
S1
1 MATRIX
2 L/R QUADRATURE
3 CENTER PHASE SPLIT
SYNTHESIZED
CENTER
T4
CENTER/LO
R6
50K
CENTER/HI
R7
50K
CENTER BASS
+30B
REVERB
T5
REVERB
R8
50K
R9
50K
REVERB/LO
REVERB/HI
AP-3 CARD
SPECIAL
PROCESSR (2)
ORBAN/PARASOUND
STEREO MATRIX
SYSTEM DIAGRAM



* NOTE: IF INPUT TRANSFORMERS ARE OMITTED R117 & R118 ARE 3.3K

ORBAN/PARASOUND STEREO MATRIX

PHASE QUADRATURE GENERATORS

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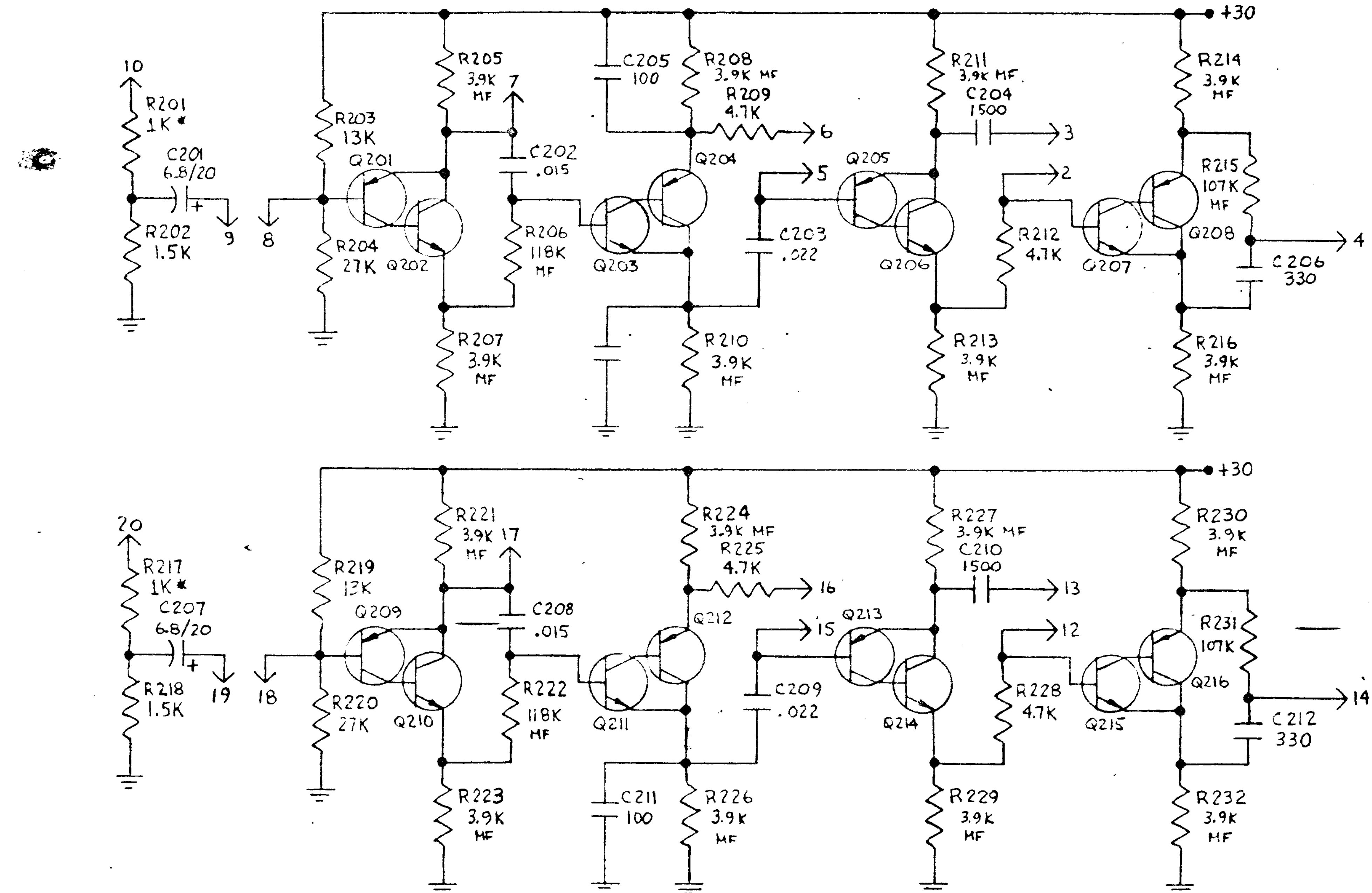
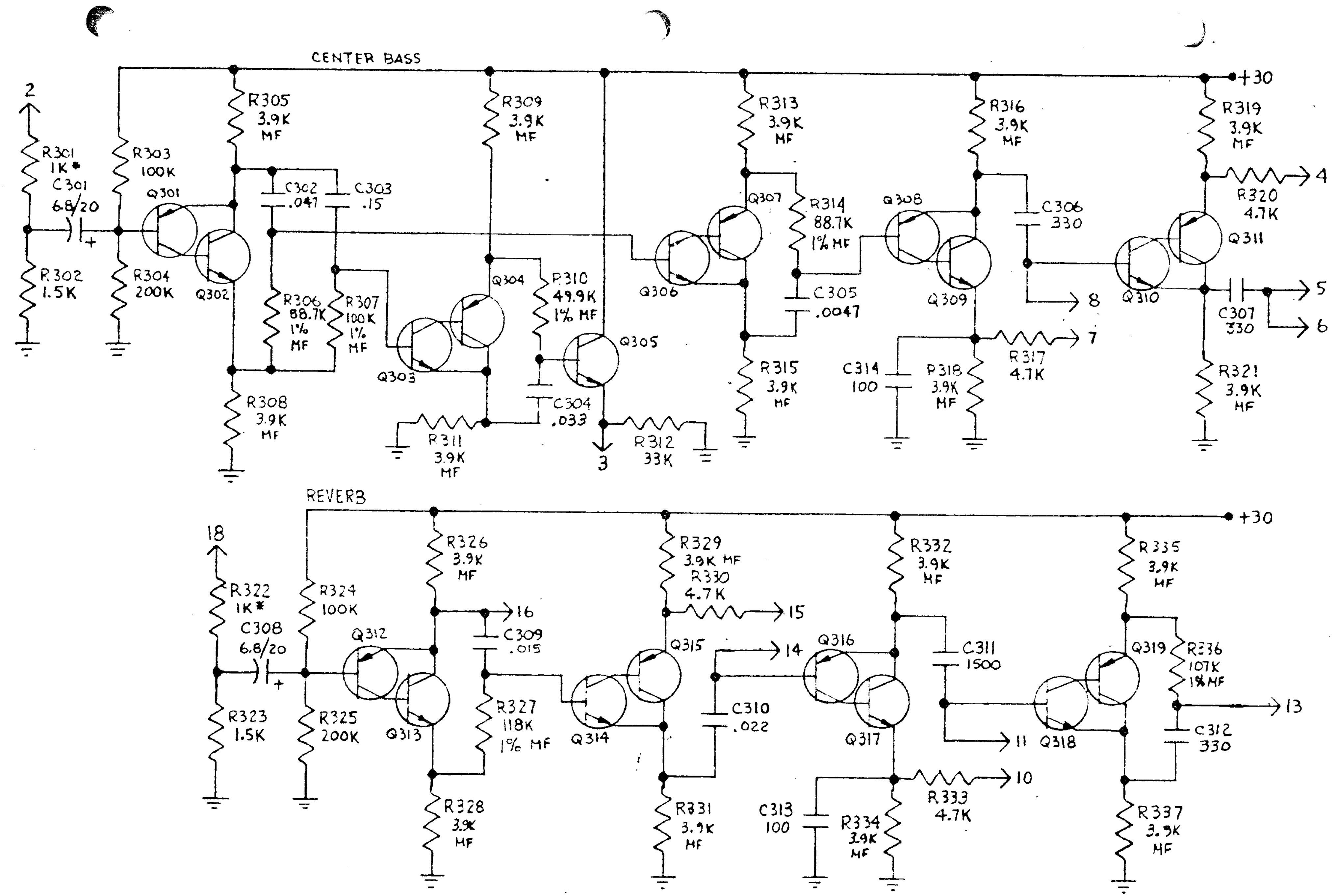
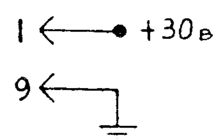


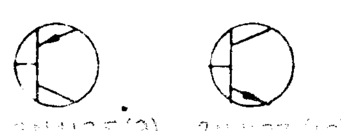
* NOTE: IF INPUT TRANSFORMERS ARE OMITTED R201 & R217 ARE 3.9K

ORBAN / PARASOUND
STEREO MATRIX

STANDARD PROCESSORS



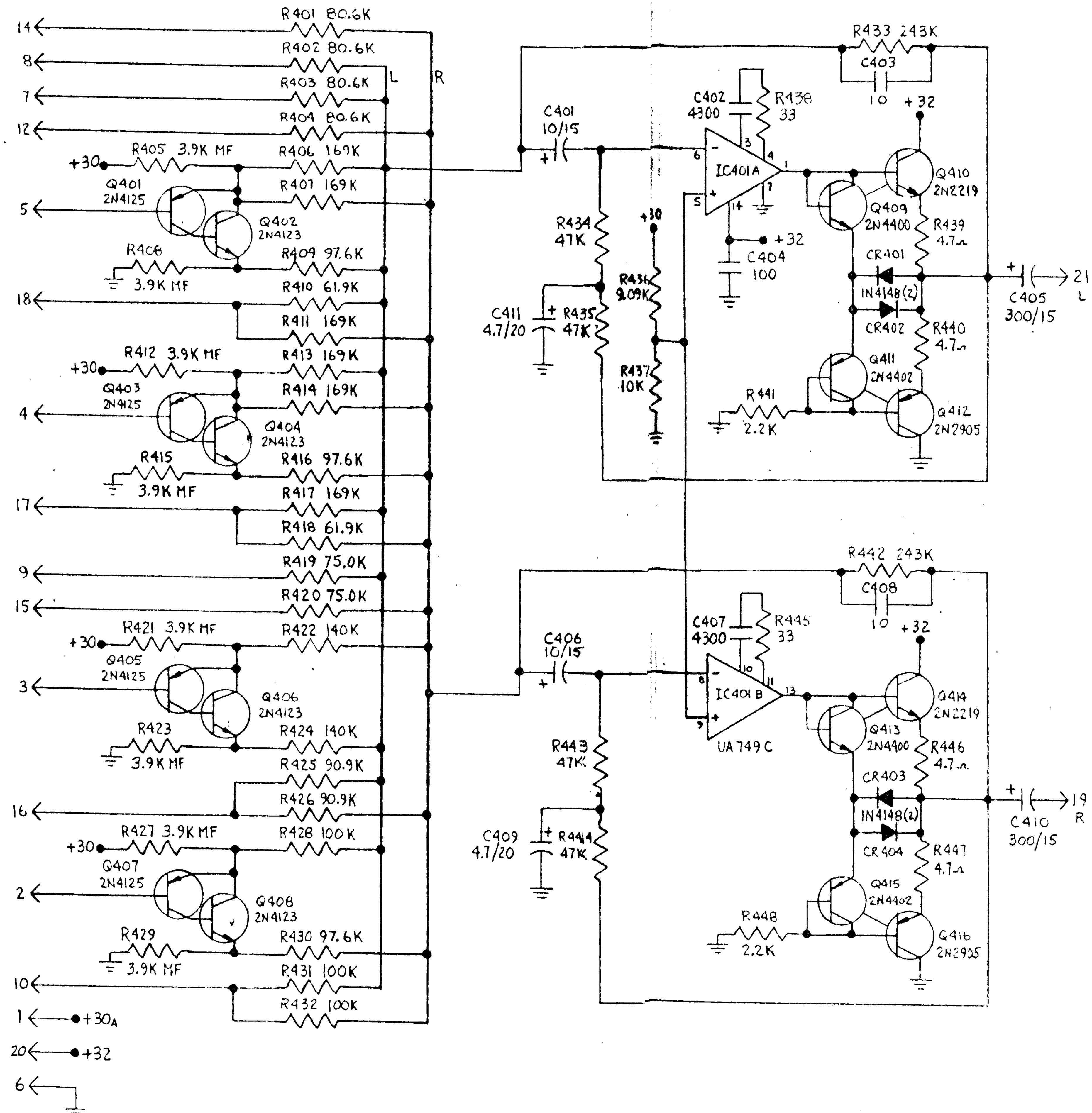




* NOTE: IF INPUT TRANSFORMERS ARE OMITTED R301 & R322 ARE 3.9K

ORBAN / PARASOUND
STEREO MATRIX

SPECIAL PROCESSORS



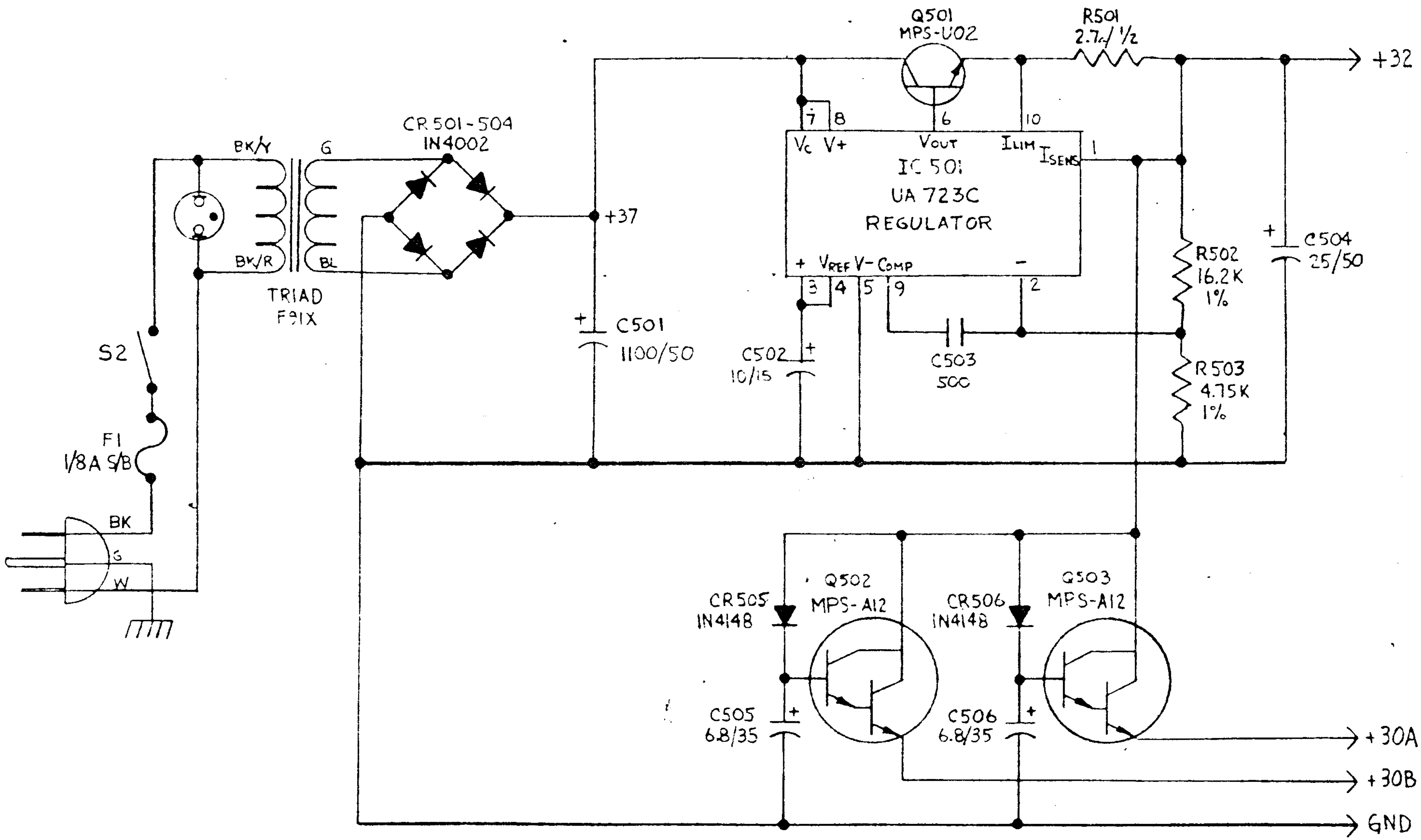
THESE TRANSISTOR PAIRS
ARE THERMALLY COUPLED:

Q409 - Q410
Q411 - Q412
Q413 - Q414
Q415 - Q416

ORBAN/PARASOUND
STEREO MATRIX

MATRIX & OUTPUT
REV 3 DATE 9/18/71

SM4



ORBAN / PARASOUND

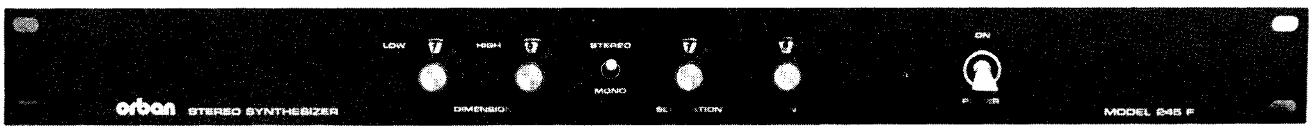
STEREO MATRIX

POWER SUPPLY

len feldman

Lab Report

Orban 245F Stereo Synthesizer



General Description: The Orban Stereo Synthesizer is a signal processor that turns any monophonic signal source into a pseudo-stereo program. The resulting pseudo-stereo is created in such a way that it does not degrade the quality of the mono recording.

Synthetic stereo in the Orban 245F is created by a complementary comb filter derived by a patented phase cancellation technique. The spectrum of the incoming audio signal is divided into five bands, each approximately two octaves wide. These bands are alternately placed in the left and right channel outputs to produce a convincing stereo illusion of channel separation, depth, directionality and even channel balance.

One of the chief virtues of a device such as this is that it provides complete mono compatibility. This occurs because the electrical sum of the two output channels is proportional to the mono input to the synthesizer. A listener/user can recover the original mono signal simply by summing the two "stereo" channels. This means that FM mono listeners, for example, will hear the original mono source when listening in mono to an FM stereo station that employs the synthesizer for creating a stereo effect with mono program sources. It also means that, in the case of a stereo disc that was mastered using the synthesizer, lateral modulation on that disc will also represent the original mono signal, since lateral modulation is the sum of the left and right stereo channels.

Control Layout: All of the controls on the slim front panel of the Orban Stereo Synthesizer are clustered near its center. Two DIMENSION controls at the left vary the relative positioning of the comb-filtered frequency bands. A STEREO/MONO switch just to the right of the dimension controls allows for a quick comparison between the original mono sound and the stereo-synthesized outputs. A SEPARATION control further to the right on the panel allows for reduction of the apparent stereo separation by partially re-mixing the channels together. According to Orban's instruction manual, this control may prove useful in permitting more natural headphone listening or in reducing the vertical excursion of a disc cutter during the making of a master disc. Reduction of apparent separation will also tend to increase mono loudness in FM stereo broadcasting.

A GAIN control located to the right of the separation control is located between an active-balanced input stage and the signal processing circuitry of the syn-

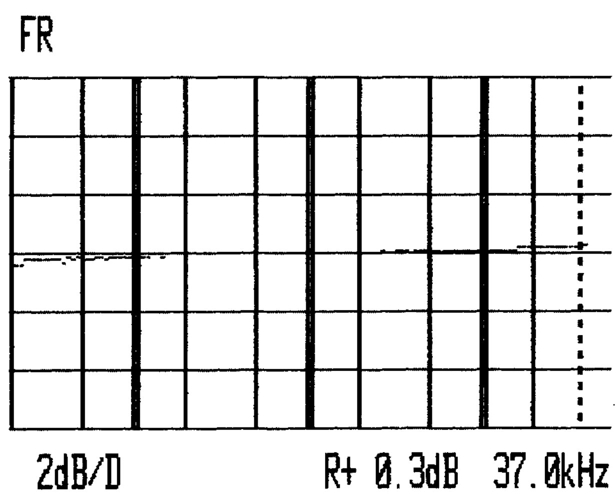
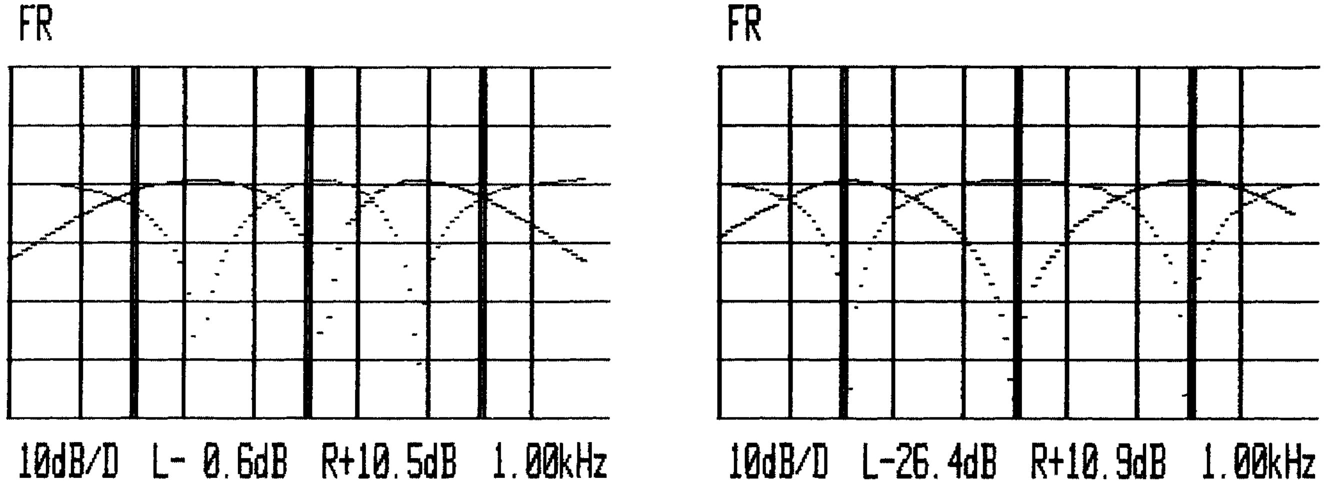


Figure 1. Frequency response of the 245F with switch set to mono. Vertical scale is 2 db/division. Sweep is from 20 Hz to 40 kHz.



A) B)

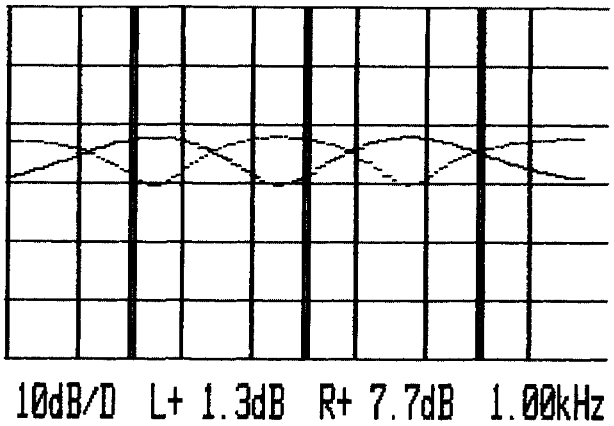
Figure 2. Plot of left and right channel response, with LOW and HIGH dimension controls set to "10" and "0" (A) and to "0 and 10" (B). SEPARATION control is at maximum.

thesizer. Therefore, while this gain control cannot prevent overload of the input stage due to excessive drive levels, it will alter overall gain of the system to match levels of other program sources or to accommodate succeeding equipment into which its output is to be fed. In its full-open position, the gain control will provide 9 dB of signal gain from input to output. A power switch is located at the right of the front panel. With the synthesizer in the signal path, its power must be turned on even if the stereo/mono switch is in the mono position, since the signal goes through several active stages even when stereo synthesizing is bypassed.

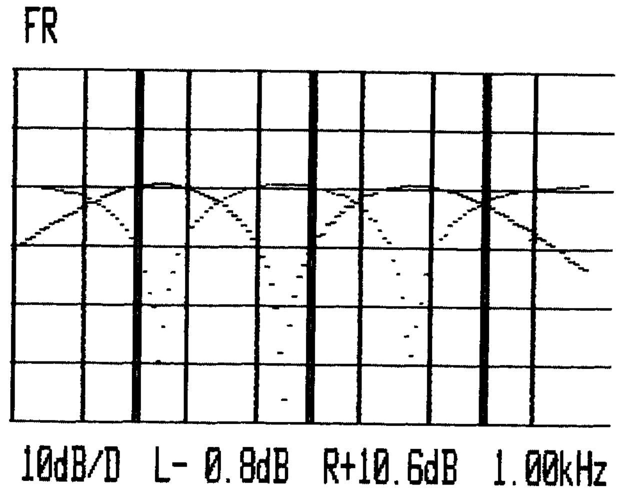
Test Results: There are relatively few meaningful measurements that can be made on a device such as the Orban 245F. These are tabulated in our VITAL STATISTICS chart at the end of the report, and all of them essentially equaled or bettered the manfacturer's published specifications. Frequency response, with the selector switch set to mono, was essentially flat throughout and beyond the audio spectrum, as indicated by the response plot taken with our Sound Technology 1500A test system and reproduced in *Figure 1.* Harmonic distortion, with the switch set to mono, hovered around the negligible 0.01 percent mark at all audio frequencies—well below the 0.1 percent limit set by Orban in its published specifications.

Our frequency plotting system enabled us to depict the action of the comb-filter, channel splitting circuitry of the synthesizer graphically, since that instrument allows the superimposition of two response plots (in this case the left and right channels) on a single graph. To obtain the response plots shown in *Figure 2A* we set the LOW and HIGH dimension controls at 0 and 10 respectively, with the SEPARATION control set to maximum (10). For the plots of *Figure 2B,* we reversed the settings of the low and high controls, while maintaining the setting of the separation control at its maximum. Comparing the results of *Figures 2A* and *2B,* the shift in frequencies at which maximum separation between channels occurs is clearly evident.

In *Figure 3A* we plotted response for both channels again, this time having adjusted all three controls to their mid-points. Here, the effect on degree of separation is apparent, while further shifts of frequencies of maximum separation are less apparent but nevertheless evident. Finally, in *Figure 3B,* we left the two dimensions controls set to their mid-positions, but increased the separation control once more to its



A)



B)

Figure 3. Response plots, as in Figure 2. However, dimension controls are both set to mid-point ("5"). Separation is set to 5 (A), and to 10 (B).

maximum setting. Note that only the amount of maximum separation has increased, while center frequencies (those frequencies at which the maximum difference in amplitude between channels occurs) remain as they were in *Figure 3A*.

Comments: Being something of a purist myself, I questioned how effective a stereo synthesizer could be, no matter how cleverly it was designed. Having listened to this device with a variety of program material, I must confess that my respect for Orban Associates has grown tremendously. Of course, much is dependent upon the nature of the program source and, as suggested by Orban, each type of program material requires a somewhat different setting of the 245F's controls for optimum stereo synthesis. We found, as did Orban, that if a single "best compromise" setting is desired (in other words, if you don't want to keep readjusting the dimension controls with every change of musical material), that best setting is around 3.0 for the LOW dimension control and about 7.0 to 8.0 for the HIGH dimension control.

As an experiment, since our lab is fully equipped with FM stereo generating equipment, we fed the outputs of the stereo synthesizer to our FM stereo signal generator, picked up the resulting program material on a high-quality FM tuner, and conducted a series of A-B comparison tests with the FM tuner set to the mono mode. The comparison tests were made by switching the Orban 245F's Mono/Stereo switch back and forth between the two settings. Once we adjusted all gains correctly, there was no audible difference, confirming Orban's claim regarding mono compatibility of the sum signal derived from the two synthesized stereo channels.

Of course, a more important test is the one we conducted with the tuner set to its FM stereo mode and the synthesizer set to simulate or synthesize a stereo effect. I won't claim that the effect delivered was as good as some of the best stereo program material I've heard on records or over the air, because it wasn't. But I will say that the effect (call it stereo, synthesized stereo, or whatever you please) was a good deal better than some of the poorer stereo mixes I've heard in recent years that were "real" stereo to begin with. And that, believe me, is meant as a compliment to the Orban designers. If you have need of a stereo synthesizer for whatever purpose, this is definitely one to listen to and consider.

ORBAN 245F STEREO SYNTHESIZER: Vital Statistics

SPECIFICATIONS	MANUFACTURER'S CLAIM	LAB MEASUREMENT
Frequency Response (Stereo sum signal)	20 Hz to 20 kHz, ±1 dB	20 Hz to beyond 40 kHz
Total Harmonic Distortion	Less than 0.1%	0.009% at 1 kHz; 0.01% at 20 kHz
Signal-to-Noise Ratio	Better than 80 dB	78 dB (82 dB A-weighted)
Available Gain	9 dB mono; 14 dB peak, stereo	Confirmed
Absolute Overload Level	+26 dBm	+26 dBm
Maximum Output (into 500 ohm load)	+19 dBm	+20 dBm
Power Requirements	115-230 V 50/60 Hz, 2 VA	Confirmed
Rack Panel Requirements	1¾" x standard 19"	Confirmed
Shipping Weight	7 lbs.	Confirmed
Suggested Price:		

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