

# PHONO PREAMP

BALANCED LINE LEVEL

OUTPUTS +4dBm

63dB GAIN AT 1 kHz

LOW DISTORTION: .003% THD

LOW NOISE -75dB

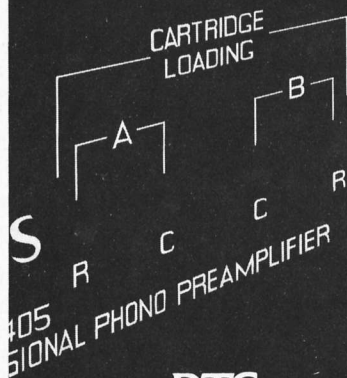
ACTIVE PASSIVE HYBRID DESIGN

RTS Systems introduces the first product in our new Series 400 professional audio line: the Model 405 Professional Phono Preamplifier. This high performance pre-amp reveals a new degree of excellence in recorded sound.

Following two years of design and field tests, our engineers have developed innovative new circuitry which we call *Active-Passive Hybrid*. APH is a combination of active low- and passive high-frequency equalization resulting in lower noise and greater headroom.

The 405 brings together technical excellence and superb sound in the most demanding applications. From disc-mastering rooms to performance-conscious broadcast stations, the sound quality is remarkable. Outstanding features include adjustable gain for each channel, adjustable cartridge termination (R&C), switchable 20 Hz rumble filter, and IHF unbalanced outputs (hi-fi level).

Let us tell you more about our distinguished high-definition phono pre-amplifier. Call or write RTS Systems for details.

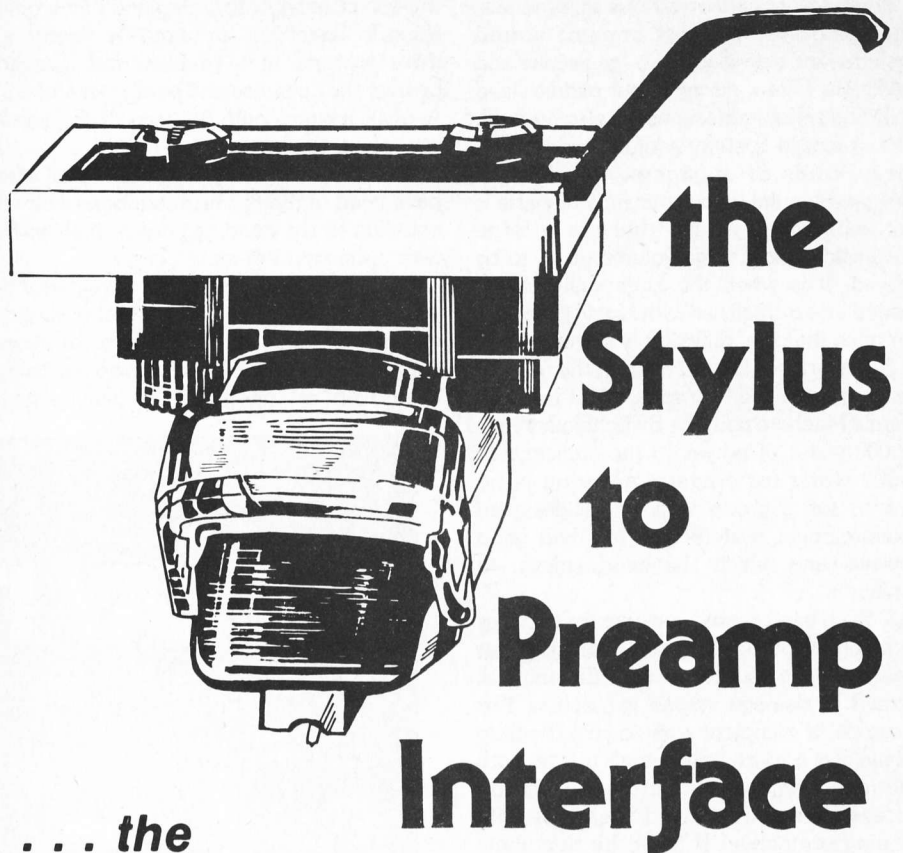


**RTS  
Systems**

1100 WEST CHESTNUT STREET

BURBANK, CALIFORNIA 91506 213/843-7022

TX: 66-2404 ANS. BK.: IMAGETRANS-LSA



# the Stylus to Preamp Interface

... the  
weakest link in the  
audio reproduction chain

by William Isenberg

In every record project there is a crucial moment when everyone concerned gathers round to find out what has happened to the sound when the final mix was transferred to disk. No matter what was done at the mastering facility, inevitably a reference cut or test pressing must be compared to the master tape. Ideally this means the playing of that disk in the same room used for mixdown, so as to be absolutely sure of the sound.

Such comparisons are truly unfair, but

there is no alternative. Even under ideal conditions, a record cannot sound as good as the original tape since no analog system can improve fidelity or reduce distortion. All we can hope for is the least possible number of devices in the signal path. If a certain piece of equipment is not being used, it should be bypassed with patchcords, rather than running program through it.

When it comes to that all-important playback, it is equally important that top quality gear be used. This means a high quality belt or direct driven turntable properly fitted with a grade "A" tonearm and the best cartridge available. The playback equipment must be above suspicion, or there is little point in listening. If any changes

Mr. Isenberg is currently associated with RTS Systems Division of Compact Video, Burbank, California. He is responsible for circuit design of the professional audio product line. Previous employers include Pioneer North America, Pasadena, Pioneer of America, Long Beach, Scientific Audio Electronics, Los Angeles, James B. Lansing Sound, Northridge, Record Plant, Los Angeles, Daniel Flickinger & Associates, Hudson, Ohio and Mastersound Recording, Atlanta, Georgia.

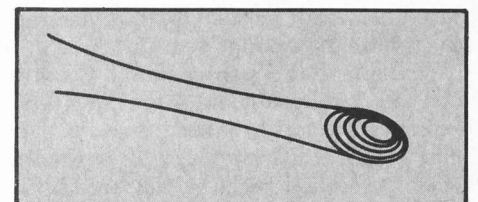


Figure 1: Flux Loop  
for Phono Cartridges

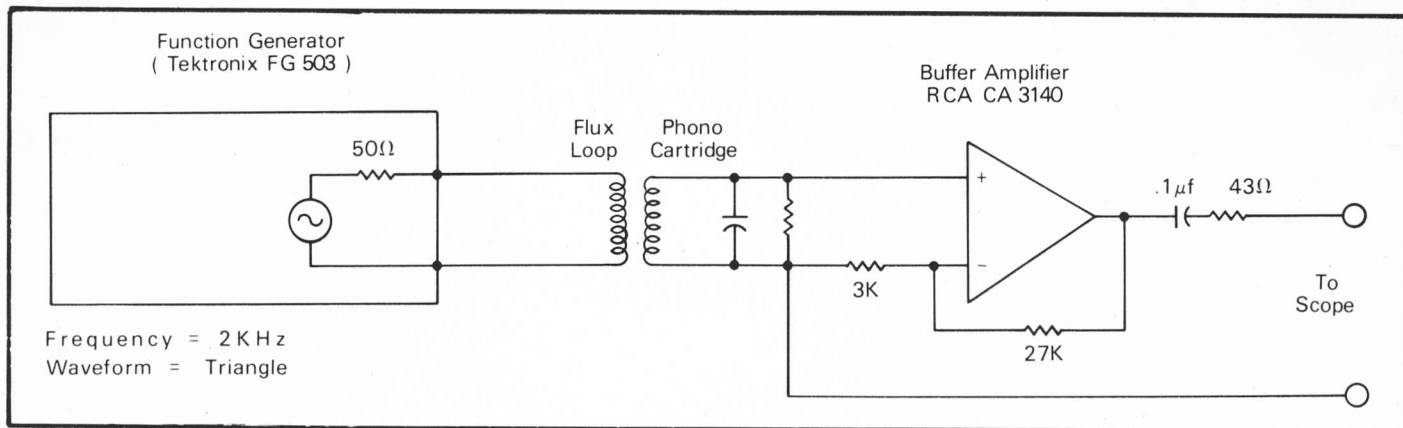


Figure 2: Phono Cartridge Flux Loop Test Setup

are made, particularly to cartridge or electronics, it would be a very good idea for all concerned to listen first to a representative sample of familiar discs, including some direct-to-disk material. Since we are talking about a professional situation, one might think that professional equipment would be the best choice to insure reliable service. However, this may not be the case.

The prime users of professional turntables are radio stations, syndication services and discos. In these applications rugged, reliable construction and ease of maintenance come first, but a recording studio user will also be very picky about the sound quality as well. This can be a problem because it is quite possible that the slip-cue felt-covered turntable which works so well over at the local radio station has so much rumble that it becomes unusable in a recording studio. The electronics can also be a problem, because the type made for radio usage often has to perform in a strong radio field. Our friends in radio know how hard it is to eliminate RF pickup. Thus we find that a preamp designed for radio probably has lots of RF chokes and bypass capacitors to kill RFI. Sad to say, these things tend to kill the high frequency response as well. Again, the fact that we are picky about sound quality is causing us to wonder about alternatives.

As far as sound quality is concerned, we can probably get what we want by using hi-fi gear. This will result in good sound, but not for long, as most hi-fi gear isn't rugged enough for studio use and has many features that are great for the hi-fi market but too complex for professional application.

Certain hi-fi products will be okay if not abused. The trade refers to this type as "esoteric." In this category, extra gadgets and features are left out and the prime thrust is better sound. Sad to say, there are many opinions of what good sound is in the hi-fi business. The only way to find out is borrow some promising units and take them to the studio and listen carefully.

If test equipment is available the units

should be checked out *before* listening. It is possible to waste a good deal of time trying to figure out what sounds good or bad since the objective measurement figures may run contrary to the subjective evaluation. However, some folks insist on listening first because "if it sounds bad you'll hear it." If there weren't so many factors involved I could agree, but experience has convinced me that complex problems are solved more easily if divided into bite-sized chunks.

#### A Playback Cart

Since a disk playback is not done every day in most studios, the most practical package is probably a roll-around cart. If the cart can be locked up when not in use it will be more likely to have a cartridge with a good stylus when needed.

Once the gross mechanics have been dealt with, what about those all-important details that affect sound quality? The two most critical areas are the cartridge chosen and the preamp. In addition, we face a serious challenge in proper matching of these components.

Until recently proper termination of phono cartridges has been one of the more neglected areas in disk reproduction. A flat response is difficult to maintain because the mechanical and electrical systems tend to resonate at different frequencies in the upper end of the audio spectrum. For many years cartridge loading was done on a compromise basis which included a standard load resistance of 47K and the assumption that anywhere from 200 to 500 picofarads of capacitive loading would be presented to the cartridge. Typical cartridge inductance evolved to be anywhere from 500 to 900 millihenrys. A glance at a reactance chart shows resonance to fall in the octave from 10 kHz to 20 kHz. This electrical resonance is somewhat damped by the 47K load resistance, but the overall result is not outstanding as far as transient performance is concerned.

Having done some work with microphone transformers I became very aware that

termination makes vast differences in the sound quality of microphone preamps. Square wave testing is a powerful tool for evaluating the transient performance of transformers which are step-up in the case of mike preamps. The usual ratio is 1 to 8 or so with the primary source impedance usually 150 ohms which the transformer reflects into the amplifier as 10K ohms. This is done to obtain the best noise performance using a typical bi-polar transistor amplifier. Unless the secondary of the transformer is properly terminated, the capacity present in circuit will resonate with the inductance of the secondary winding and cause a peak in the response at some very high frequency. On the surface of it there would be no harm in this, but contrary to some opinions, many musical instruments have substantial transients and overtones well above 20 kHz. When bells and triangles are struck a whole spectrum of ultrasonic information appears to excite any spurious resonances in a mike transformer. This is plainly audible as a shattering sound on impact instead of the distinctive click heard in person. Broadband sources such as ride cymbals generally sound harsh and nasty through a preamp with transient distortion.

#### Measurement Procedure

There is no substantive difference between the secondary of a microphone transformer and a phono cartridge. It is not hard to measure the response of a mike preamp as long as the generator has the proper source impedance, but how to go about measuring a phono cartridge? Vibrate the stylus? With what, and with what kind of precision? Even if it were possible, the stylus mechanical resonance would influence the results. In order to treat the electrical and mechanical areas separately there must be an electrical method of exciting the cartridge without the stylus being used.

There seemed to be no answer until Lyman Miller and I were part of a group listening to Peter Butt describe the calibration of magnetic tape duplicators

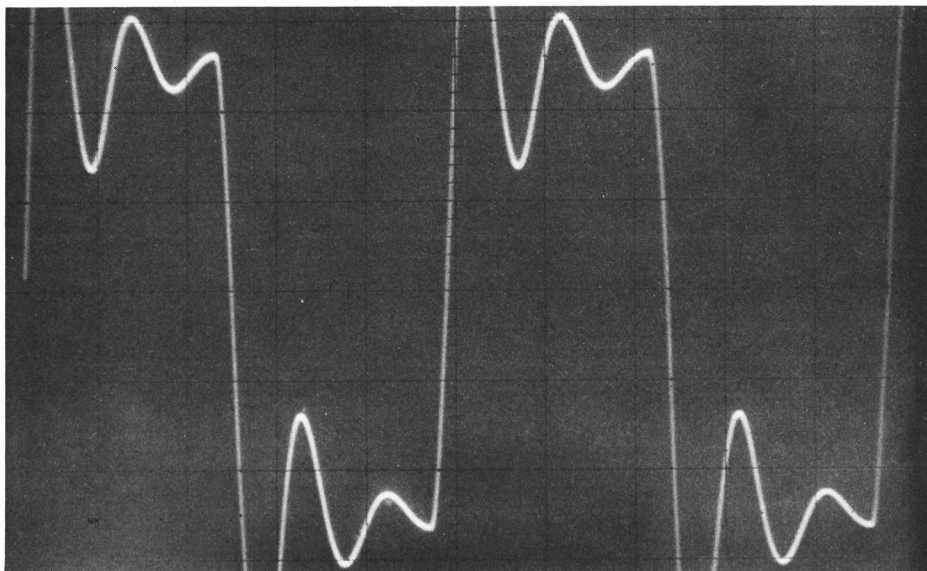


Figure 3:  $F = 2\text{kHz}$ ,  $100\ \mu\text{sec}/\text{div}$ .  
Capacitive Load Only, Overshoot 42%.

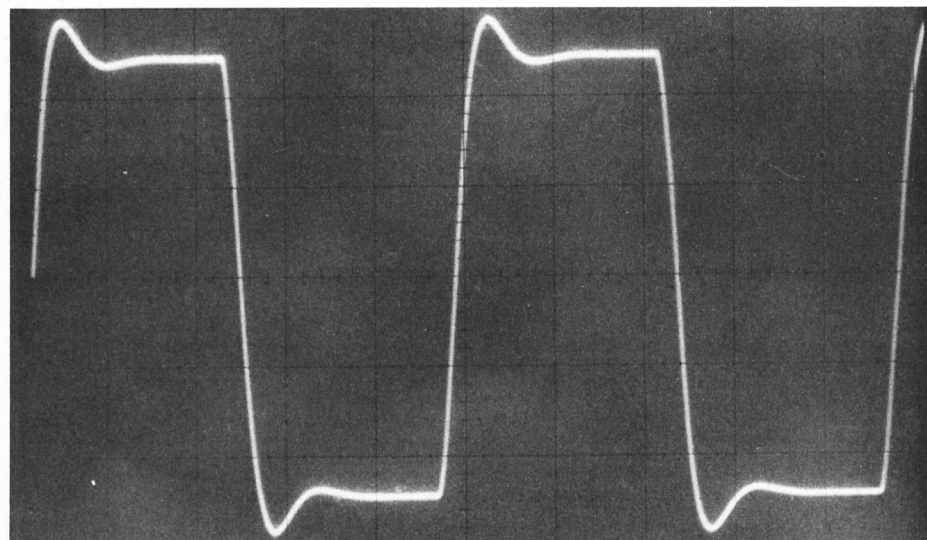


Figure 4: Resistive and Capacitive Loading.  
Risetime  $26\ \mu\text{sec}$ , Overshoot 9%.

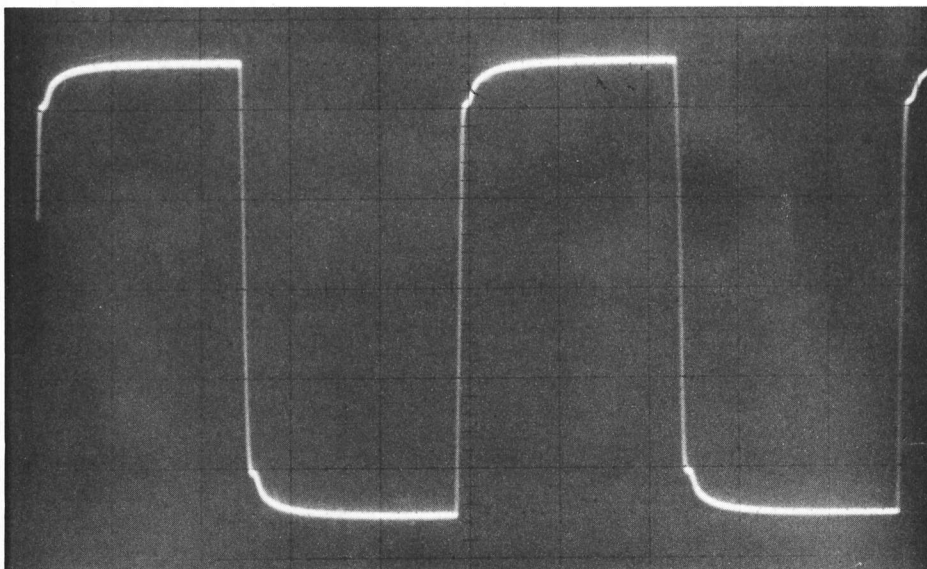


Figure 5:  $F = 2\ \text{kHz}$ ,  $100\ \mu\text{sec}/\text{div}$ .  
No Load, Rise Time  $8\ \mu\text{sec}$ .

using a *flux loop*. This gadget makes it possible to check performance of a reproduce channel without worrying about tape-to-head interface problems such as poor tape wrap or azimuth. Eureka! All we had to do was remove the stylus assembly from a phono cartridge and insert a ten-turn flux loop (Figure 1). Lyman pursued the concept and read a paper on flux loop calibration of phonograph reproducing systems at the AES convention in Los Angeles in May, 1976. My own effort along these lines did not progress as rapidly, but the insight obtained proved to be quite an eye-opener.

The biggest single problem with the phono cartridge is that it has to be installed in a tone arm away from the preamp. Tone arm lead capacity is difficult to reduce below 100 pf. To simulate this a capacitor was connected in parallel with the cartridge while using a flux loop to excite the coils (diagram in Figure 2). This caused substantial overshoot and ringing; the square wave was badly distorted, as shown in Figure 3. By connecting a 47K resistor in parallel, the damping of the square wave got much better, but the rise time became unacceptable (Figure 4). Now what? By transformer standards this performance was *terrible*. Was this the only way to listen to records? Not exactly. Moving coil cartridges which operate at a lower impedance and thus are relatively immune to capacitive loading are available but they have problems of their own, such as non-replaceable styli, high cost, and more hum and noise. What really intrigued me was that with both resistive and capacitive loads removed from the cartridge driven by the flux loop, square wave performance improved incredibly (Figure 5). If the cartridge could somehow be operated with *no loading whatever*, a dramatic sonic improvement could be expected. The only thing to do was either put the entire phono preamp in the tonearm headshell, or use some kind of buffer to lower the impedance such that the cartridge would not be loaded by tonearm wiring. A method was devised using two transistors to buffer the cartridge as required. With electrical problems bypassed, it became possible to investigate the mechanical situation. Various cartridges were installed into the buffered headshell and the DIN 45541 test record (sold by Gotham Audio) was played. Incredible! Some cartridges performed very well, but others had very pronounced peaks in the high frequency end. One highly rated cartridge had a stylus assembly which resonated so strongly that it reached a 10 dB peak at 22 kHz! Flat as a pancake all the way from 10 Hz (tonearm resonance) until 10 kHz and then a curve that shot up like a rocket from Vandenburg! No wonder the spec sheets recommended a 500 pf load. Needless to say, my personal system did not use this cartridge.

## New Cartridge Designs

Fortunately, new cartridges appear with regularity and the CD-4 quadrasonic system has served as a stimulus to improve the status quo. The latest cartridges available have lower inductance which reduces electrical problems and better designs for the stylus/cantilever assembly which addresses mechanical resonance difficulties. These benefits are complemented by a growing consumer awareness of the situation and the appearance of preamps with adjustable cartridge loading. Some preamps have switches on the front panel to make it easy.

Once the cartridge output is presented to the preamp, a very important process begins. The amplifying devices used and the connection of circuits in conjunction with them can affect listening greatly. It is very important that no spurious responses be generated by the preamp. The two types likely to cause the most trouble are harmonic and intermodulation distortion.

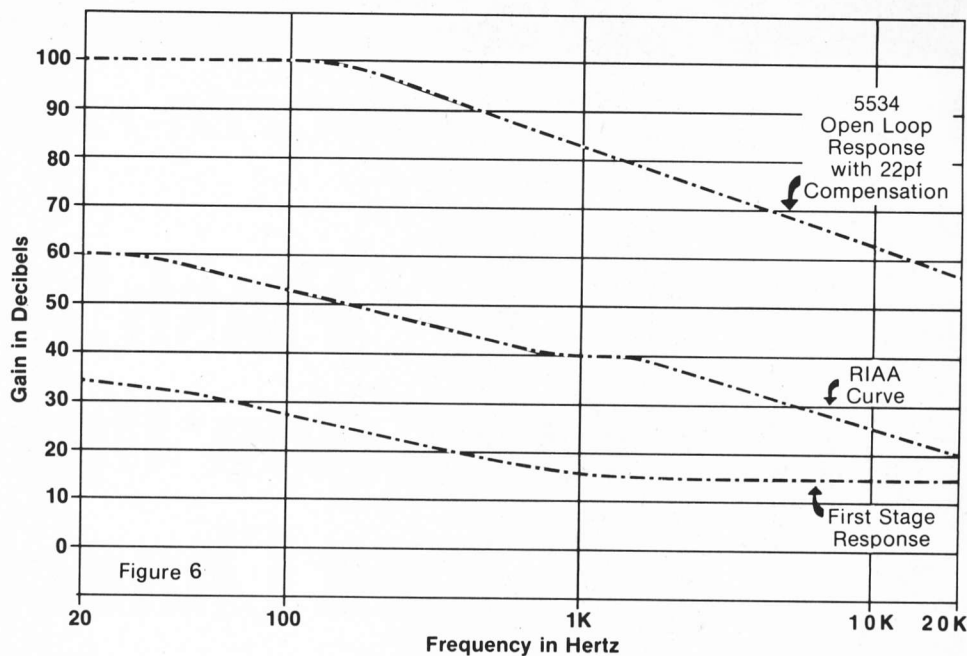
By far the most noticeable is third harmonic. Third harmonics do not normally occur in music (other than fuzz-tone guitars and synthesizers) and as a result just a little bit is audible. If a preamp has a lot of third harmonic distortion, voices will buzz, brass will sound bigger than life, and everything will tend to smear in a concave manner. In fact, this sonic effect is often called "transistor smear" or "transistor sound." It is not musical and also causes listening fatigue.

Second harmonic distortion can also cause trouble, but in a different way. Since music naturally contains second harmonics (octaves), it would seem that some of this distortion would be a good thing. Having too much of a good thing can be a problem, however, when it slowly dawns on you that vocals sound very closed miked and intimate, woodwinds swell and bloat, brass sounds muffled and strings get sleek and fat. The perspective becomes convex as things seem to bulge towards you. (Fat) Vacuum tube equipment is prone to this type of distortion. Some people cherish old (or new) tube equipment because of the less annoying characteristic sound and relative lack of fatigue.

Intermodulation distortion is difficult to measure using the SMPTE method because it always turns out to be very low. A more stringent test has been devised which uses two tones much higher in frequency. This process goes by the general name of twin-tone and involves two tones separated by 1 kHz. If the amplifier were perfectly linear, no 1 kHz beat note would be produced. Real amplifiers aren't so good. The sonic result is muddiness and loss of articulation.

### Phono Pre-Amp Circuitry

At this point it might be a good thing to consider the amplifier internal circuitry, sometimes called topology. Remember that



an amplifier does not amplify anything. What is called an amplifier and looks simple is in fact a complex servo system which takes power obtained from the power supply and uses that power to create a replica of the incoming signals.

Most amplifiers used for audio signal processing are known as operational amplifiers or "op-amps." Originally developed for analog computers, this flexible device has found application in many audio amplifiers and signal processors. When monolithic op-amps were first designed, performance was limited. The first integrated circuit op-amp to be widely accepted is the Fairchild  $\mu$ A709 which came out in 1965. Although many other devices have appeared since and come with as many as four units per package, they all have certain things in common as far as the audio designer is concerned.

- Low cost.
- Ease of application (fewer parts)
- Small size
- Low power consumption

So much for the good side. But you don't get something for nothing. Here are some of their deficiencies:

- Distortion rarely specified
- Noise can be a problem
- Large product variation
- Occasional stability problems
- Inconsistent sonic characteristics.

This puts the designer in a tough spot. To keep costs down and make a salable product, it is necessary to use ICs, but a truly high performance design requires a discrete transistor approach. The most desirable advantage of discrete amplifiers is the total control which is available to the designer. This doesn't come cheap, however. It takes about 8 transistors to

make a good discrete op-amp and lots of other parts as well.

### RIAA

Just as important is the overall circuit configuration used to implement the RIAA playback equalization. Almost all phono preamps on the market use a feedback circuit which attempts to accomplish the entire task with a single network/amplifier. This may be good enough for a compact stereo or a small receiver, but component quality stereo and professional usage requires less compromise.

The RIAA curve requires a 40 dB change over the frequency range of 20 to 20 kHz. Since a 40 dB change represents a voltage ratio of 100 to 1, the network used imposes severe demands on the amplifier used. This is shown graphically in Figure 6, where a standard RIAA curve (40 dB @ 1 kHz) is the middle trace. The open loop (no feedback) response of a signetics NE5534 op-amp with 22 pf compensation is shown at the top for comparison. Note that there is 43 dB of loop gain (difference between open loop and closed loop) at 1 kHz which degrades to 37 dB at higher frequencies. Feedback theory indicates that error caused by having only 40 dB of loop gain will be approximately -84 dB or about .006%. This is not as good as it is possible to do with the 5534 or a good discrete amplifier. If the loop gain is 60 dB instead of 40 dB, theoretical error becomes -126 dB or .00005%. This is much lower than the -70/80 dB signal-to-noise ratio obtained by most good phono preamps. It is obviously preferable to have error (distortion) be 40 dB below the noise level than approximately equal to it.

Given the fact that this amplifier chip has only 83 dB of gain at 1 kHz, it appears that the maximum closed loop gain shouldn't be

**Direct boxes & "Mic-splitters"**  
DESIGN AUTHORITY

Passive Direct Box SM-1A	Active Direct Box SM-2
Deluxe Active Direct Box SM-3	Single "Mic-splitters" MS-1A
Quad "Mic-splitter" MS-4	8 x 2 "Mic-splitter" MS-8
12 x 2 "Mic-splitter" MS-9	16 x 2 "Mic-splitter" MS-10

**Direct Boxes:** Both active and passive SM-1A for guitars SM-2 and SM-3 for keyboards and electronic instruments.

**"Mic-splitters":** Low impedance in and out. Will handle +6 dBm. Will pass phantom voltage. Isolated grounds.

**Thousands in use around the World!**  
We also manufacture audio transformers, snakes, audio modules

SEND FOR YOUR FREE COPY OF OUR NEW CATALOG  
1111 Las Vegas Blvd., North  
Las Vegas, NV 89101  
(702) 384-0993, (800) 634-3457  
TWX (910) 397-6996

Quality Engineered Sound Products

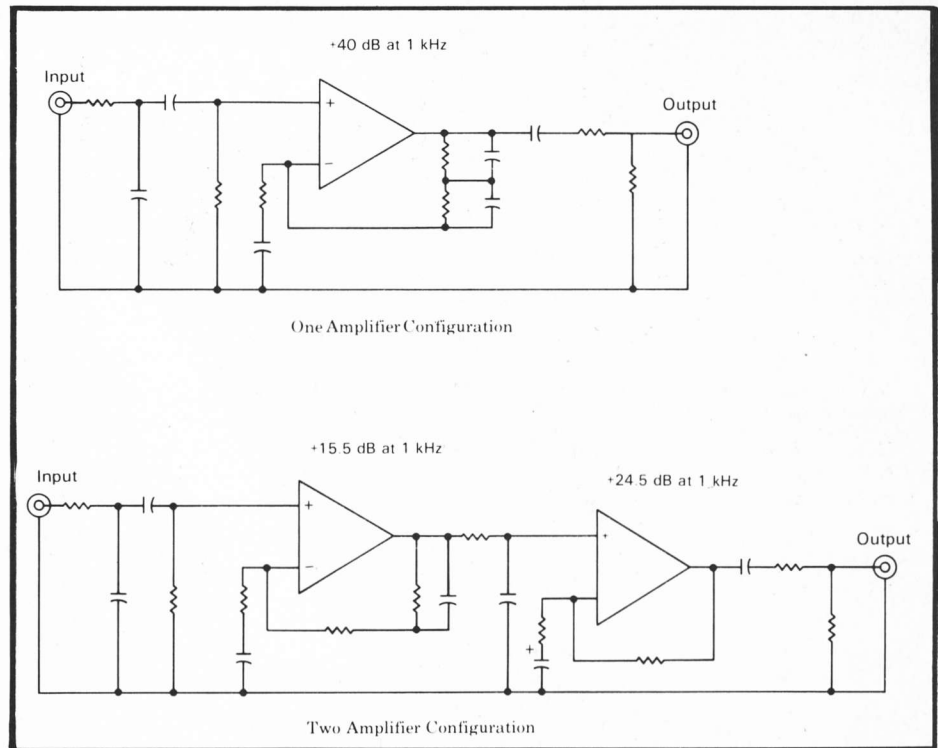


Figure 7: Comparison of Amplifier Configurations  
(Note that overall gain is the same for both.)

any greater than 23 dB @ 1 kHz in order to maintain 60 dB of loop gain. Since 40 dB overall is required, the best way to get it is to cascade two stages. The lower curve on the graph, Figure 6, shows the 50 - 500 Hz portion of the RIAA curve implemented with 67.5 dB of loop gain at 1 kHz, 66 dB at 20 Hz, degrading to 42 dB at 20 kHz. This configuration has 15.5 dB of gain at 1 kHz, requiring an additional 24.5 dB to reach a total of 40 dB. A second amplifier with no EQ network is quite acceptable as there will be 58.5 dB of loop gain at 1 kHz and 32.5 dB @ 20 kHz. In order to accomplish the HF equalization (-3 dB @ 2,122 Hz) a passive RC network is used between the amplifiers. This network is independent of the 50 - 500 network used in the first stage and because it is passive there is no departure from the ideal 20 dB/decade ( $\approx 6$  dB/octave) slope as unity gain is approached. The output of the second amplifier is available to drive cables without changing equalization due to capacitive loading. Schematic diagrams of the one- and two-amplifier configurations are shown in Figure 7.

To evaluate sonic differences the author has built many "little grey box" preamps in the past five years (Figure 8). The first few used IC op-amps of various types, and, when ICs available at the time proved to be less than perfect, a discrete op-amp was designed using 8 transistors. This made the phono preamp a bulky box containing 32 transistors and many other parts (Figure 9), however the effort was justified as sound

quality developed an "effortless" character which I have not heard from any IC op-amp. As better sounding IC's became available, the IC version was re-designed to take advantage of the newer devices (Figure 10). One advantage is that newer ICs can be plugged into old sockets, which produces amazing changes in sound quality. All of these grey boxes use the two stage configuration described earlier.

#### Calibration

To get the best performance, the turntable-tonearm-cartridge-preamp system should be calibrated. This means that all members of the group play in tune, so the sound doesn't suffer.

Calibration should not prove to be a serious problem if it is undertaken carefully. *Warning: Not all test records are created equal. Some are more equal than others!* (My personal favorite is DIN standard 45541 available through Gotham Audio). With careful adjustment of cartridge termination and perhaps the tone controls (if any) it should be possible to get response within  $\pm 0.5$  dB of flat at least to 15 kHz. Do not use a VU meter on the console or a tape machine to measure frequency response. Some of them are unacceptable above 10 kHz. If you have a Hewlett-Packard or Sound Technology distortion analyzer the meter section is ideal. You will probably notice that the meter responds to rumble and record warp well enough to make measurements difficult. The preamp rumble filter may help as you adjust the top end termination. At the

**NEW!** Everything you expect ...

## ... in a PARAMETRIC EQUALIZER

- +20 to  $-\infty$  db equalization in 3 overlapping ranges
- Low and high level inputs and outputs
- Widest bandwidth variation in the industry
- Mono (Model PQ-3), and stereo (Model PQ-6) versions
- Quality construction, dependable, easily serviceable
- The prices are right!

**FURMAN SOUND, INC.**  
616 Canal St. • (415) 456-6766  
San Rafael, California 94901

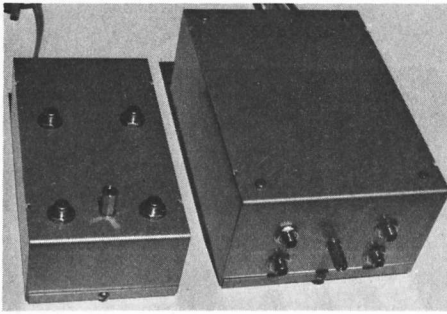


FIGURE 8: GREY BOX PREAMPS

bottom end it is likely there will be a slight rise in response caused by tonearm/stylus resonance. Very few tonearms have any damping mechanism to adjust, but some cartridges have an integral brush which is an effective substitute. Now that frequency response is as flat as it will get, don't forget to set the two channels equal in level. This should be done while playing the mono section of the test record.

If an oscilloscope is available you are ready to undertake an important step toward better reproduction. Compare the two channels for phase difference as you would to check azimuth on a two-track tape machine. Adjust the phase by loosening the cartridge screws slightly and swiveling the cartridge as viewed from the top. Then tighten the screws again. This assumes that

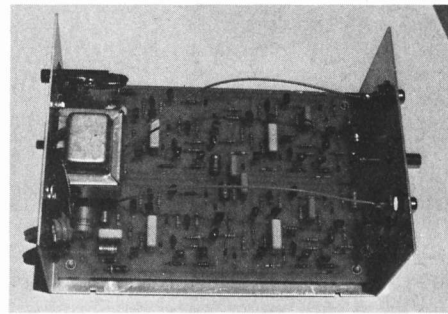


FIGURE 9: DISCRETE PREAMP INTERIOR

overhang, tracking force and all the routine things have been done as well.

If you are really brave and have a distortion analyzer you can adjust anti-skating to optimum instead of taking their word for it. Find a tone on the record long enough to permit the analyzer to lock on. Don't expect any marvels of low distortion, either. Anything under 3 percent is an incredible miracle. Before the tone quits, adjust anti-skating and stylus pressure for the least odious result.

Now you have a calibrated disk playback system. It may not be quite as good as the stuff at your friendly disk mastering facility, but at least when you play a reference on your monitor system, you'll be giving it a fair chance.

Hears to fidelity.

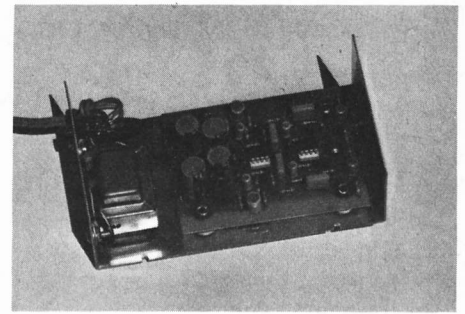


FIGURE 10: I.C. PREAMP INTERIOR

**References:**

**Dynamic Range Requirements of Phonograph Preamplifiers** — Tom Holman, *Audio Magazine*, July, 1977.

**Record Warps and System Playback Performance** — Larry Happ and Frank Karlow, *Journal of the AES*, October, 1976.

**Understanding Phono Cartridges** — S. K. Pramanik, *Audio Magazine*, March, 1979.

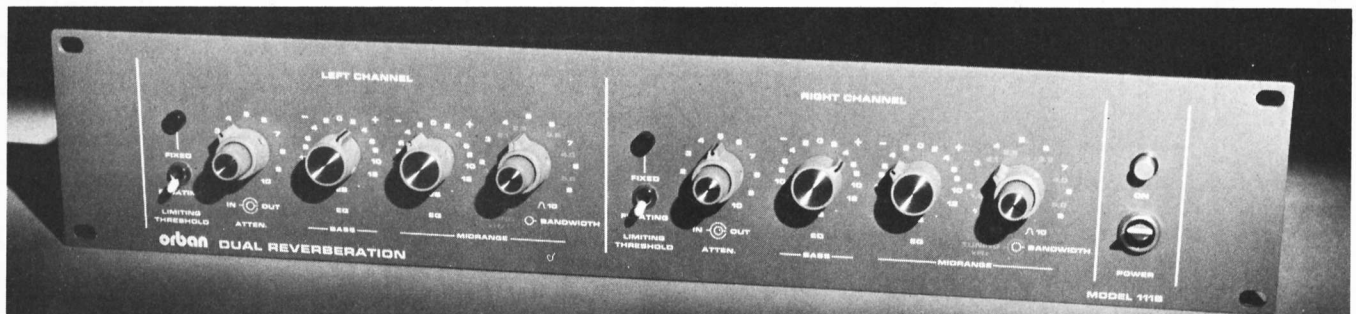
**Disk Cutting in Practice** — Tony Bridge; **Disk Cutting in Theory** — Hugh Finimore, *Studio Sound Magazine*, July, 1975.

**New Factors in Phonograph Preamplifier Design** — Tom Holman, *Journal of the AES*, May, 1976.

**An Overview of SID and TIM** — Jung, Stephens, Todd, *Audio Magazine*, June, 1979 (contains extensive bibliography).

**Noise Specs Confusing?** — Jim Sherwin, *National Semiconductor Corporation*, AN-104, May, 1974.

**Flux Loop Calibration of Phonograph Reproducing Systems** — paper read by Lyman Miller, Palo Alto, California. May, 1976 AES 54th Convention.



## Two More Springs For No More Money

Now Orban advances its price/performance leadership in compact, professional reverb systems. Our 111B Dual Reverb now comes with *six springs per channel at no increase in cost.*

You get:

- Lower flutter
- Higher echo density
- Smoother, more natural sound

Plus, these Orban standard features:

- Advanced signal processing
- Floating threshold to minimize twang
- Midrange parametric equalizer
- Bass shelving equalizer

For the 111B Reverb and other fine professional audio products, see your local dealer, or contact Orban for the location of the dealer nearest you.

Hear it at AES Booth 28

**orban**

Orban Associates Inc. 645 Bryant Street San Francisco, CA 94107 (415) 957-1067