NEW FACTORS IN PHONOGRAPH PREAMPLIFIER DESIGN*

TOMLINSON HOLMAN

Advent Corporation, Cambridge, MA. 02139

An investigation into why various phonograph preamplifiers sound different has produced some new measurement techniques which correlate well with subjectively observed differences. A preamplifier design which performs well on these new tests as well as on conventional ones has evolved. Closely matched A–B listening tests have been performed, and, in general, the unit which measures better does sound better.

INTRODUCTION: The normally quoted specifications of a phonograph preamplifier, such as frequency response, noise, and distortion, are of limited usefulness in ultimately determining how a preamplifier will sound. This is due to a variety of factors: for example, 1) no conventional test measures performance under transient conditions; 2) test results (with the exception of some noise measurements) are not weighted for human perception or annoyance value; and 3) the response to signals out of the audio pass band may affect in-band performance through intermodulation in the preamplifier or, more likely, later in the system. The measurement environment consisting of oscillators, meters, etc., represents a gross simplification of the real-world conditions in which the preamplifier operates. It is therefore not surprising that conventional measurements have low correlation with the critical listening experience. Conventional specifications may give engineers limited information about how a cricuit behaves, but specifications cannot be said to be at a state where preamplifiers, sorted by specifications, will be ranked subjectively in the same order.

Preamplifier listening test comparisons, like those of other components, must be handled with some care. The cartridge must be properly loaded, both resistively and capacitively, for each preamplifier under test. Preamplifier gains must be very accurately matched (to within a fraction of a decibel). Grounding and shielding must be done with great care since the output of the preamplifier is asked to come physically close to the input (for switching purposes), and some units may oscillate under these conditions. Listening tests should be "double-blind" so that the participants are not influenced by their predispositions. The monitor system should have appropriate frequency range and flatness to prevent weighting the results unnaturally. What is typically found, under careful test conditions, is that variations are subjectively heard which the observers call frequency response differences, most often having the quality of "brighter" versus "duller."

HIGH-FREQUENCY INTERACTIONS

Various factors influence the high-frequency steady-state response of a cartridge phono preamplifier system. The cartridge–cable system presents an irregular source impedance to the preamplifier which loads the cartridge–cable system with a complex load impedance. Cartridge designers frequently state the proper load (resistive and capacitive) to ensure response to the specifications, yet few turntable, tone arm, cable, or preamplifier manufacturers specify impedance completely.

Measurements made through an equivalent electrical circuit with typical moving-magnet cartridges in place reveal a high-frequency rolloff associated with the electrical circuit. Corresponding "rollups" in the mechanical circuit of the cartridge (the damped stylus mass–groove wall resonance) yield a "flat" response. This high-frequency rolloff is predicted by the equivalent circuit of the system by inspection; however, of fifteen currently available phonograph preamplifiers tested (stand alone phono preamplifiers, system preamplifiers, and phono preamplifiers in integrated amplifiers and receivers from all price categories), all but one exhibited anomalous high-frequency behavior when fed a test signal from an actual phonograph cartridge. This misbehavior is known as "cartridge inductance interaction."

A good way to measure this interaction is to compare the frequency response of the cartridge–preamplifier system with and without the addition of a high-impedance buffer stage having low known input capacitance (Figs. 1 and 2). Of course the cartridge terminating resistance of typically 47 kΩ is used at the input to the buffer since this represents the hypothetical load that the preamplifier should present to the cartridge system (Figure 3).

* Presented October 31, 1975, at the 52nd Convention of the Audio Engineering Society, New York.

Use of this technique reveals differences which are confirmed aurally. An example is from a preamplifier in a current well-respected receiver. The RIAA equalization accuracy measured conventionally with a voltage source is quite good—about ±0.5 dB from 20 to 20 000 Hz. However, in the equivalent system with a 200-pF capacitor representing cable capacitance, the interaction effect is large, peaking with respect to the reference 1½ dB at 7 kHz, crossing over at 10 kHz, and having a loss of 6½ dB at 20 kHz. In Fig. 4 the top curve is the voltage source error, the middle pair of curves show the interaction (the difference between the curves) with the 200-pF capacitor, and the bottom pair of curves show the interaction with no cable capacitance. Although the choice of cable capacitance has an influence on the degree of the interaction, it shows little influence on the kind of interaction.

Three separate effects of the cartridge inductance interaction have shown up in measurements. One is a gradual high-frequency rolloff; another is a gradual high-frequency rollup; the third is of the form shown in the figure. These interactions arise from two sources: one is simple input capacitance, which may be an intentional cartridge termination capacitor; the other is more complicated. This interaction involves the amplifier's open-loop gain, bandwidth, and input impedance; the RIAA feedback loop; and the cartridge system. For example, an examination of the input impedance versus frequency for the preamplifier documented in Fig. 4 shows a 40% drop in input impedance at 20 kHz with respect to 1 kHz. In a preamplifier in which the interaction is less severe than in the first case, a variable capacitor was connected to the input of the buffer amplifier to try to match the preamplifier's response. A good match could not be made, indicating that the dropping high-frequency response in this case was not entirely due to input capacitance but probably to the interaction of the open-loop input impedance with the feedback loop.

In this case there was not enough feedback at high frequencies to maintain a high-input impedance. The open-loop input impedance of a bipolar transistor is only moderately high (40–70kΩ); negative feedback is used to keep the input impedance high. The falling input impedance with increasing frequency is effectively in parallel with the cartridge termination resistor and therefore interacts with the cartridge system. Two additional factors contribute to errors accumulating at high frequencies, the cartridge impedance goes up with increasing frequency making the high-frequency load of the preamplifier input impedance more important, and stability considerations frequently dictate a fairly low open-loop dominant pole compensation [1] which reduces the available high-frequency feedback. Miller effect capacitance of the input device can also play a role, especially with high first-stage gain. In one or two cases an opposite effect has been found, but the cause is similar—the input impedance goes up with increasing frequency due to an effective negative input capacitance formed by the amplifier and feedback loop which causes a high-frequency "rollup."

Hallgren has shown [2] a model for the cartridge impedance which includes a normally overlooked term, the sum of frequency-dependent resistive losses which he calls

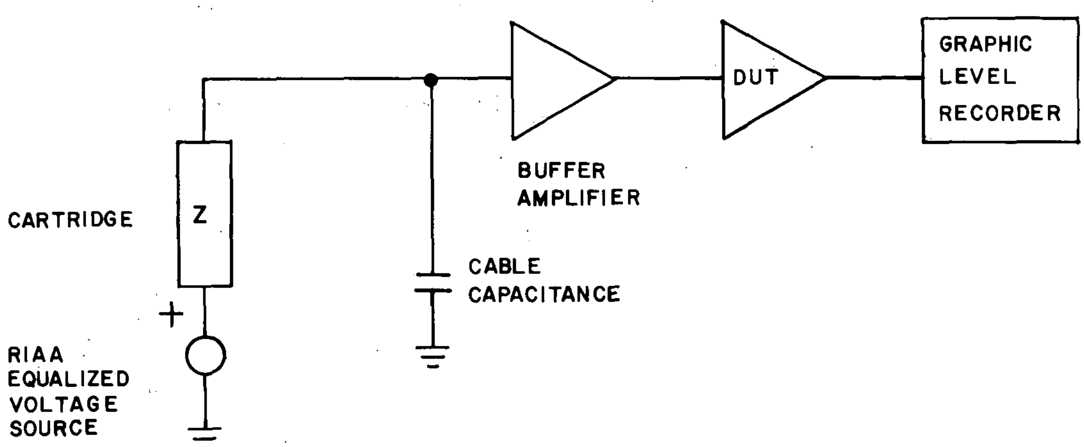


Fig. 1. Cartridge–preamplifier system with buffer stage.

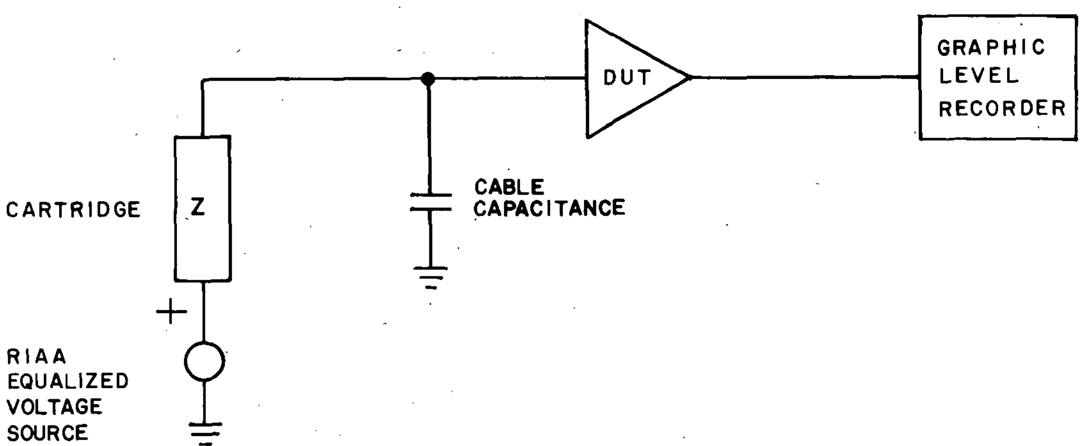


Fig. 2. Cartridge–preamplifier system without buffer stage.

"RLT." A frequency-dependent resistive loss arises from the eddy current and hysteresis losses associated with a coil wound on a magnetic material [3]. By including this term, unique for each cartridge, in a model with the dc resistance of the cartridge, the inductance of the cartridge, the total parallel capacitance, and the load resistance, the frequency response of the electrical system can be predicted with great accuracy. Such a model is a useful design tool as it completely predicts what we have done experimentally with the buffer amplifier.

INFRASONIC RESPONSE

An often overlooked and important area of preamplifier design is the amplifier's infrasonic response, where the cartridge appears resistive and the above interaction is unimportant. Infrasonic rolloff acts to reduce the effects of driving loudspeaker systems well below their nominal cutoff frequency where the cone is essentially unloaded (especially with vented-box loudspeakers), and to reduce overload and intermodulation in tape machines and power amplifiers caused by infrasonic cartridge output. In more than one case a tape machine was returned to its manufacturer for the most gross kind of distortion (even accompanied by periodic signal cutoff), which proved to be overload caused by a particularly nasty combination of cartridge, tone arm, record warp, and low-frequency response of phono preamplifier and tape machine. In a listening test in which two otherwise identical preamplifiers were compared, the preamplifier which incorporated an infrasonic filter produced less audible intermodulation at high playback levels and a more solid stereo image. Infrasonic rolloff dramatically reduces visible woofer motion and amplifier overload caused by record warps.

Phase effects associated with infrasonic cutoff should not be neglected. The optimum filter should attenuate greatly in the difficult 7-Hz region but should not introduce audible group delay on low-frequency program material. A listening test with worst case choices of test signal and listening conditions has shown that the 20 ms worst-case group delay of a three-pole complex filter is just perceptible. The group delay sets a practical upper limit on the complexity of the infrasonic filter.[1]

Various observers have reported on the usefulness of damping in the tone-arm–cartridge resonant system. Members of the Boston Audio Society have reported that damping reduced audible amplitude modulation effects, improved stereo imaging, etc. Tone-arm damping may work in two senses: it may simply be reducing the infrasonic response of a system, and/or it may be making ever more highly compliant cartridges track record warps better in suboptimal arms [4]. The first mechanism (reducing the infrasonic response) adds to the case for infrasonic rolloff.

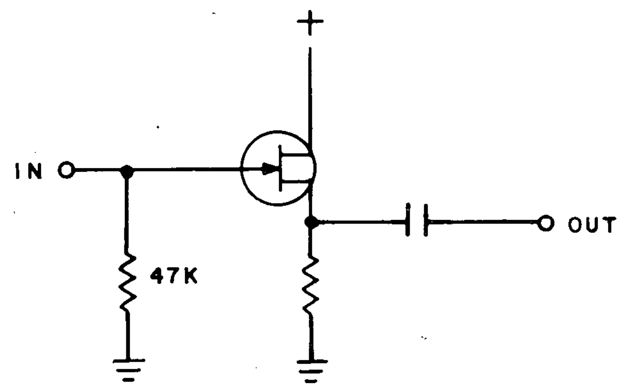


Fig. 3. Buffer amplifier.

NOISE

Noise performance of the cartridge–preamplifier system is also a case of interaction. The noise performance of the preamplifier should be designed for a typical cartridge

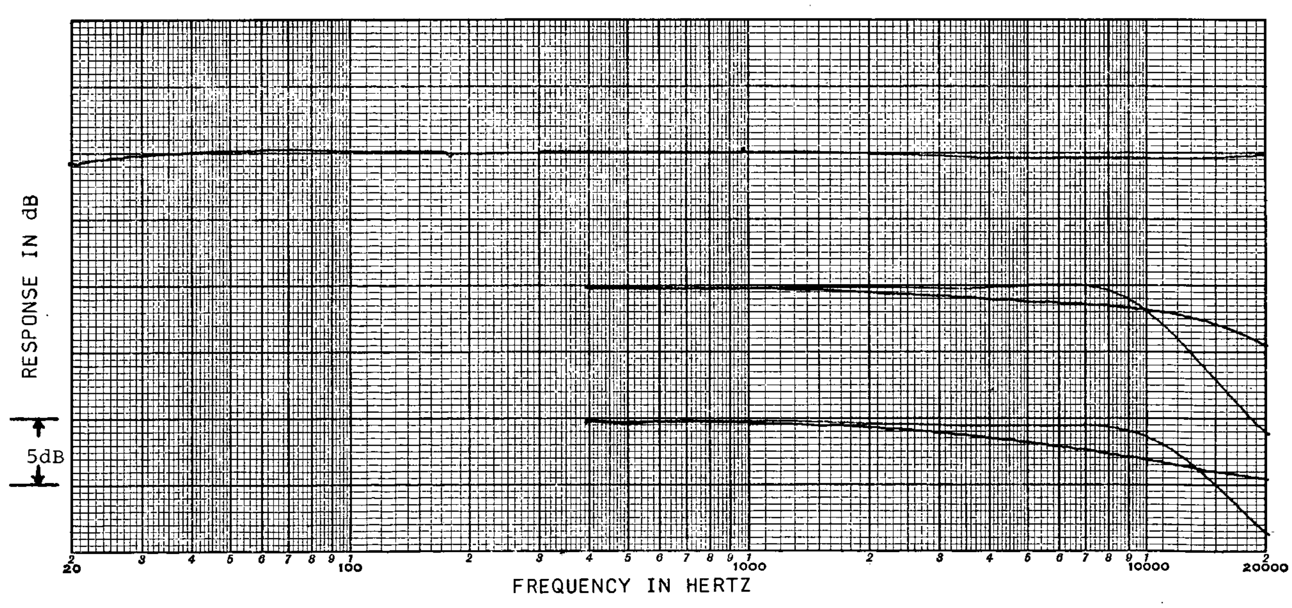


Fig. 4. Effect of cartridge interaction. Top curve—voltage source frequency response; center curves—interaction (difference between two curves made with and without buffer amplifier) with 200 pF cable capacitance; bottom curves—interaction with very low impedance.

[1] Group delay audibility test: 10 ms positive pulse at one second repetition rate dc coupled to headphones rated flat to 10 hertz.

source impedance. Also, design (as well as measurement) should be done on a weighted basis so that the performance is optimized for the low-level chracteristics of human hearing. Combining these requirements with some well-known techniques for making low-noise amplifiers ensures a design where the noise is dominated greatly by the cartridge as shown by Hallgren. For optimum noise performance within the range of expected source impedances, the bipolar transistor is the most readily usable technology. Until recently there were no field effect transistors available which could compete with low-noise types of bipolar transistors for the range of source impedances presented by cartridge systems. Today there seem to be a few expensive types available which have competitive noise performance. Time and increasing device manufacturers' interest in consumer markets should reduce the cost differences so that FETs may play an increasing role. Another important consideration is the method of application of feedback. For the case of phonograph cartridges driving preamplifiers, a series feedback topology is superior to a shunt feedback (virtual ground input) form [5]. When we take these points into consideration, we find that we have a design which is limited by the noise inherently associated with the real part of the cartridge impedance. However, such a design will probably not measure as well as some on the conventional short-circuited input unweighted measurement since it is optimized for an appropriate source impedance and weighting.

The special case of the moving-coil cartridge with its very low source impedance (2–40 Ω) and low output level requires special design techniques for low noise. Among the available techniques are 1) to use a transformer to step up the impedance to a range where a low-noise transistor has an optimum source impedance, 2) to use a number of low-noise transistors in parallel to optimize the effective source impedance, or 3) to use a very large geometry device (a power transistor) selected for low noise and operated at an optimum current level.[2]

Having chosen the most likely class of input device and feedback topology, two factors are then under the control of the designer; the choice of a specific device and the operating point (chiefly collector or drain current). Choice of device and operating point are made easier by making the simplifying assumption that the typical cartridge has a known source impedance in the most sensitive frequency range of human hearing. Then the possibilities are more easily investigated by choosing the best candidates and operating points and, once a device is chosen, optimization is quickly convergent on the lowest noise solution.

Two noise reduction techniques have appeared in recent designs. One is to use quite low impedances in the RIAA feedback loop for minimum noise generation from the feedback resistances with the consequence of needing a complementary common collector output stage to drive the low impedances. The second involves the use of a synthesized input impedance through the use of an extra feedback loop which bootstraps the cartridge termination resistor to reduce its noise contribution. One commercial embodiment of the bootstrap method produced a signal-to-noise ratio of 85 dB re 10 mV, 1 kHz input, ANSI "A" weighted with a cartridge input.

Because the thermal noise associated with the source impedance is the dominant factor in phonograph system noise, any meaningful future improvement will have to come from the phonograph cartridge designers and disc manufacturers. Further, commercial disc recordings rarely approach the cartridge–preamplifier system noise level in noise performance.

OVERLOAD

Sine-wave input overload has been the subject of a numbers race which has limited meaning. In fact, the observed impression of overload distortion does not correlate with the sine-wave overload number except in the crudest way, since if the sine-wave overload number is high, it is likely that other areas of preamplifier design have received more careful attention. Unlike other audio designs (most notably microphone preamplifiers) where the designer must prepare for enormous dynamic range and sensitivity change from transducer to transducer, the phonograph preamplifier designer's task is less demanding—moving magnet and reluctance cartridges have a small range of sensitivities, and are subject to a definite tracking limit, with the best tracking cartridges having lower sensitivities. Taking the worst case combination of a high output, good tracking cartridge, and recorded level yields a peak input voltage of 135 mV ref 1 kHz [6]. This converts to a 1-kHz rms value of 95 mV. It should be emphasized that this is a genuinely worst case combination which is not expected to be approached typically in practice. A large study has been made of the velocity versus frequency on commercial records [7] which may serve as a guide to the preamplifier designer.

The 1-kHz sine-wave overload number is inadequate to describe the overload characteristic over the whole frequency spectrum. At low frequencies many preamplifiers are not capable of as high an output level as at midband due to the large low-frequency gain requirement of the RIAA curve, and therefore lack of distortion-reducing feedback. At infrasonic frequencies the charge on large emitter bypass capacitors which are used conventionally may change over the cycle causing high distortion. At high frequencies inadequate slew rate and lack of output drive capability limit the distortion performance. A low impedance, high-frequency load is imposed on the preamplifier by the feedback loop. The RIAA network looks like a low impedance at high frequencies in series with the gain-setting resistor which sets the high-frequency load. Thus the preamplifier needs to be able to supply the load imposed by the feedback loop at the highest frequency of interest as well as the external load. An optimum design would have an output overload point invariant with frequency.

SLEW RATE

Slew rate is a principal high-frequency limitation caused by the necessity of changing the charge on capacitors located in the signal path of an amplifier. Fortunately for

[2] Suggested by Rene Jaeger of dbx.

phonograph preamplifier designers, the maximum slew rate is fixed by a physical process (the acceleration of the stylus) which is further processed by an electrical low-pass filter discussed before under cartridge impedance. However, the RIAA recording preemphasis does place the phonograph preamplifier in an unusual position: the signal that the preamplifier sees at the input is high in transient content.

The slew-rate requirement for the preamplifier can be set from measurements of slew rates from cartridges or from data derived from cutterhead specifications. Using specifications from a modern, high-velocity cutterhead for tone-burst peak velocity combined with half-speed cutting, and played by the highest sensitivity cartridge which tracks well (although the cartridge would definitely mistrack under these conditions) yields an accleration of 13.2×10^6 cm/s^2 and a slew rate of 0.026 V/μs. This compares with measurements from commercial recordings of 5×10^6 cm/s^2 and 6×10^6 cm/s^2.[3] Using a preamplifier with 40 dB of gain at 1 kHz, the required output level is thus 2.3 V rms at 20 kHz with low distortion.

Inadequate slew rate could lead to transient intermodulation distortion. However, the very fast transients which produce transient intermodulation distortion are limited in rise time and level by the finite acceleration of the stylus and by the electrical low-pass filter consisting of the cartridge source impedance and the cable and load system, thereby lessening the chances for transient intermodulation distortion.

DIFFERENCE TONE INTERMODULATION

The phonograph preamplifier, along with other similarly equalized preamplifiers, is in a tough setting for one form of distortion—high frequency intermodulation. J. McKnight tells the story that in the early days of tape recording in this country, a user complained of hearing low-frequency disturbances accompanying his bird-song recordings. None of the conventional distortion tests showed anything wrong with the amplifier until two high-frequency tones with a close spacing were introduced and a strong difference tone was produced. This was traced by the engineers to a lack of available charging current for the feedback-loop capacitors. RIAA equalized preamplifiers have nearly a 40-dB difference in gain from one end of the spectrum to the other, so that a difference tone distortion (second-order intermodulation, f_1-f_2) of 0.1% for a flat amplifier could lead to almost 10% for the equalized case. Also, since the recording is preemphasized, the input to the preamplifier contains a disproportionate share of high-frequency energy which accentuates the difficulty. Second-order intermodulation (f_1-f_2) has been measured on a number of units with signal generator sources. The level of the source generators was based on peak recorded levels of commercial records and normal cartridge sensitivities. The tones were 13.0 kHz and 13.1 kHz mixed 1:1 at a composite level of 40 mV rms. The resultant difference tone varied in percentage from 1% for a simple two-transistor design to unmeasurable (less than 0.02%) for the topology to be discussed.

[3] David Griesinger made the latter measurement; we confirmed his with the former.

SQUARE-WAVE TEST

One would suppose that after cartridge impedance interaction, noise, and conventional as well as the other forms of distortion measurements had been made and found to be substantially similar, then phonograph preamplifier designs would sound alike. They do not. My impression, substantiated by others, is that there are still "frequency response differences" on playing program material, which in at least one case goes counter to the measured differences. There is one element that has been overlooked—the response to transients. None of the conventional tests or the proposed new ones test phonograph preamplifiers with asymmetrical transient conditions similar to program material. While attempting to measure the transient response of preamplifiers, some rather surprising results were found. Many preamplifiers did poorly on a test which was not thought to be particularly severe—the reproduction of a 1-kHz square-wave spectrum. Square waves were chosen over other kinds of nonsinusoidal test signals as they were found to correlate perfectly with asymmetrical signals, and the resultant spectrum is easy to analyze. As a test source, a fast square-wave generator with good symmetry was used. This square-wave signal was passed through a signal-pole 30-kHz RC low-pass filter to an accurate RIAA preemphasis network which incorporated rolloff in addition to the single 30-kHz real pole beyond 50 kHz. The signal, now at an rms level equivalent to the 3.54 cm/sec "0" VU sine-wave test signal, but containing the sharp transients associated with the RIAA preemphasis, was applied to the input of the device under test. The output from the preamplifier should be a reconstructed square wave with only odd harmonics present. In fact, the preamplifiers had very different output spectra, ranging from identical to the input spectrum to a unit in which the second harmonic is down only 13 dB representing 22% second harmonic (see Table I).

Several sources are possible for this kind of distortion. Any capacitors in the amplifier which are subject to asymmetrical charge and discharge cycles are a cause. (Whether the case of slewing at adequately fast but nonetheless asymmetrical rates is a problem has not been studied.) The RIAA compensation capacitors may have values and be placed where they limit the potential slew rate and are asymmetrically charged. RF bypass capacitors used with good intention to eliminate interference may limit slewing and have asymmetrical charging.

The first work done on asymmetrical performance measured eight preamplifiers ranging from two-stage tube designs to multiple-transistor designs. Results from these early tests correlated well with listener perceptions of the preamplifiers under study. Later work showed some anomalies when, in particular, single-ended input FET and passive equalizer designs ranked worse on measurement than on listening. In the case of the FET input preamplifiers, the asymmetry took on a different form as measured on an oscilloscope in the time domain than had been seen with more conventional designs. Generally the preamplifiers had slewed well in one direction and badly in the other to produce the asymmetry. Both FET designs tested had overshoot on the square wave in one direction, which yielded the same asymmetrical condition with attendant high even-order distortion products. Apparently this test may be overly conservative. The required rise time and symmetry to pass this test are well beyond what can be expected from phono cartridges; still, a number of designers have commented upon the efficacy of the test for finding asymmetries.

Perhaps a test that would satisfy the condition that the generated signal be as much like asymmetrical musical waveforms available from cartridges would be an asymmetrical pulse test analyzed by means of a Fourier transform. Pulse level and spectral content should be selected to be realistic, and both polarities of pulse should be tried since many of the preamplifiers tested showed considerable differences in reproducing a ramp of one polarity versus the opposite polarity.

A NEW DESIGN

A new preamplifier design has evolved along with the refinement of these measurement techniques and the criteria described above. The most basic decision is the choice of topology. The topology is strongly influenced by the type of active devices employed, and the choice of devices is in turn heavily influenced by the required noise performance. Design for low noise is done on the basis of the elements discussed above. The most important segment of the design from a noise standpoint is the configuration and operating point of the first stage. Also, the first stage should have enough gain to overcome noise contributions from later stages.

In the design shown in Fig. 5 the differential bipolar input configuration has been chosen for a number of reasons. One is the inherent nonsaturating quality of a differential amplifier supplied with an emitter current source. Another is the good isolation between feedback loop and input which contributes to noninteraction with the cartridge source impedance. The differential configuration allows greater freedom of choice of impedances in the feedback loop so that the high-frequency load (and consequently, the slewing performance) may be optimized.

Open-loop compensation of phonograph preamplifiers is fairly tricky due to the extensive feedback in the RIAA

Table I. Square-wave distortion.

Device:	Frequency (kHz) 2	4	6	8	10	12	14	16	18
Source spectrum	−67	−67	−67	−67	−67	−67	−67	−67	−67
Two transistor phono preamplifiers in a $400 receiver	−13	−18	−20	−23	−25	−26	−27	−28	−29
An expensive preamplifier	−16	−20	−23	−25	−27	−29	−30	−31	−32
A transistor preamplifier with an unusual cascode configuration	−30	−33	−35	−37	−39	−40	−41	−42	−43
A preamplifier using an integrated circuit	−47	−51	−54	−56	−58	−59	−61	−62	−63
An inexpensive tube preamplifier, 2 stages	−50	−54	−57	−59	−61	−62	−63	−64	−65
An expensive tube preamplifier, 3 stages	−54	−58	−61	−63	−64	−65	−66	−67	−67
The topology of Fig. 5	−67	−67	−67	−67	−67	−67	−67	−67	−67

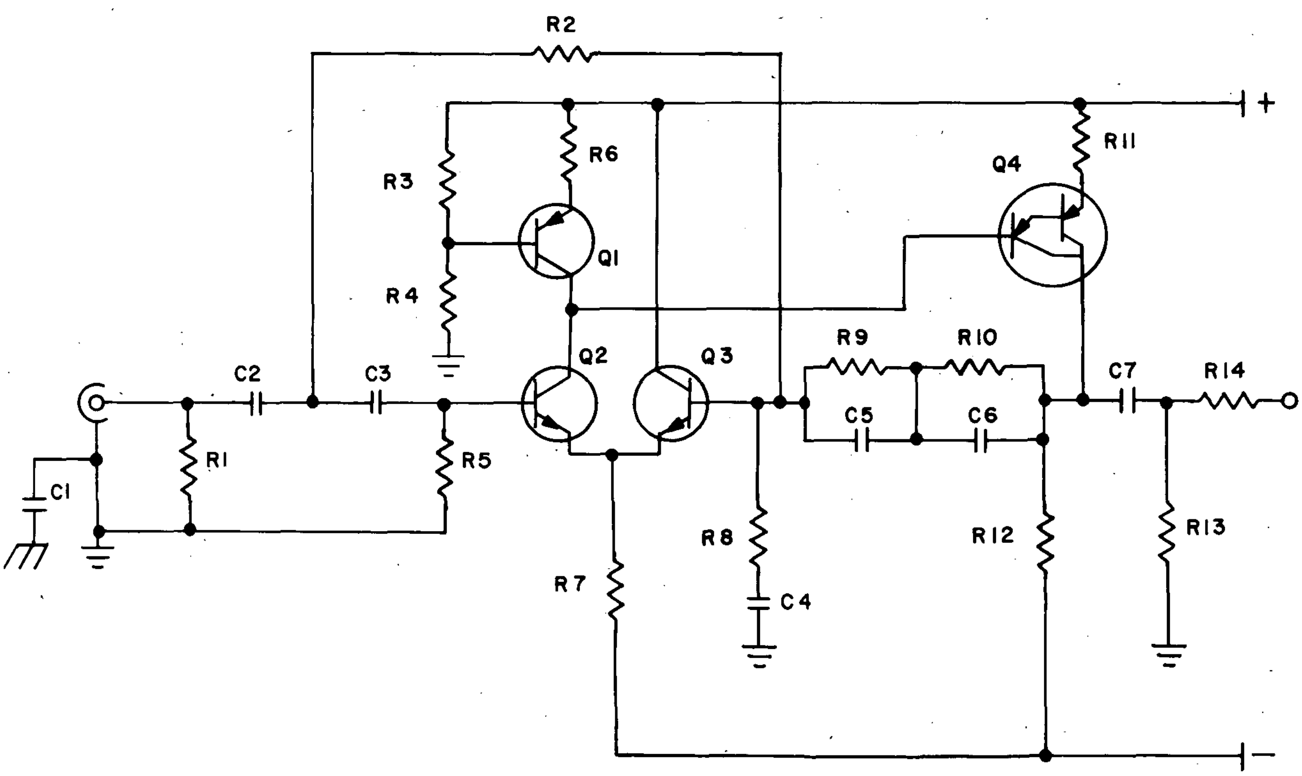


Fig. 5. New design topology.

compensation loop around the amplifier. Many designers choose fairly low-frequency dominant-pole compensation, which generally is safe in the steady state, but which does not allow the amplifier to slew well. In this preamplifier design, the dominant pole is set by the total capacitance at the input stage collector which is on the order of only a few picofarads. Since there is a fairly large amount of current available to charge this capacitance, the slew rate is very good. Also, all other capacitors in this design are scaled such that there is adequate current available to charge them quickly.

The use of current source loading on the first stage produces very high stage gain, which yields large open-loop gain. Large open-loop gain leads to large (closed-loop, RIAA feedback) loop gain, which keeps the input impedance high and the distortion low. In fact, the preamplifier has no measurable cartridge inductance interaction, and distortion at 7 V output at 1 kHz is less than 0.04%, consisting of second and third harmonics and therefore completely inaudible. The distortion decreases monotonically at lower levels.

The usual high-fidelity design practice of low-frequency filtering by the use of synchronously tuned RC stages (those with a number of poles on the negative real axis of the s plane) yields a "soft" response corner. A better solution is to move the poles off the axis and space them so as to produce a desired response. If we rule out inductors, this implies that feedback must be used to generate the appropriate response function. In this preamplifier the input RC network is bootstrapped from the feedback input of the amplifier to obtain a complex pole pair which, when combined with a real axis pole produced by C4 and R8, yields an 18-dB per octave high pass with good shape.

The measured performance of a number of prototype units is given in Table II. Results of the cartridge inductance interaction test (unmeasurable) and square-wave even-order distortion test are particularly attractive. No conventional specification has been sacrificed to obtain high performance on these new tests, and the unit is only slightly more expensive to make than the simplest circuits.

Several public A–B demonstrations have occurred that test the efficacy of the square-wave transient test. At a meeting of the Boston Audio Society with 130 audiophiles in attendance, the prototype preamplifier was compared with one which showed closely matched frequency response with a cartridge. Once levels were adjusted accurately, there was general agreement that transients were better reproduced by the prototype preamplifier with the differences manifest as apparent frequency response changes between the preamplifiers. Another public demonstration was conducted on the WBUR radio program "Shop Talk" during which two preamplifiers were com-

Table 2. Measured prototype performance.

Frequency response	1. Voltage source, audio passband	±0.5 dB, 25 Hz–20 kHz
	2. Infrasonic response	−1 dB, 20 Hz with −19.5 ms group delay re 1 kHz −3 dB, 15½ Hz −12 dB, 10 Hz −21 dB, 7 Hz −35 dB, 4 Hz
	3. Cartridge impedance interaction (high-inductance cartridge, 20 Hz–20 kHz)	Unmeasurable, less than 0.2 dB
Noise	1. Referenced to 10 mV rms, 1 kHz input, ANSI "A" weighted; cartridge source	−82 dB
	2. Short-circuited input, ANSI "A" weighted	−86 dB
Total harmonic distortion		
	1. 1 kHz, 7 V rms output	0.04%
	2. 20 kHz, 5.6 V rms output	0.04%
	3. 20 Hz, 7 V rms output	0.06%
	Note: Harmonic distortion decreases monotonically at lower output levels. Distortion products are a mixture of second and third harmonics.	
Intermodulation distortion		
	1. IM using SMPTE method of 60 Hz and 7 kHz tones mixed 4:1 and preemphasized by RIAA record function	
	Measured at 6 V rms composite output level	0.05%
	Measured at 1 V rms composite output level	0.008%
	2. Difference tone intermodulation using 13.0- and 13.1-kHz tones at a composite level of 40 mV rms	Unmeasureable, less than 0.016%
Preemphasized square-wave input even-order distortion products, 1 kHz		
	1. Voltage at input of device 19 mV rms, 600 mV peak; worst case even harmonic level	−75 dB, 2nd harmonic
	2. Average of all even harmonics in pass band	−78 dB
Sine-wave input overload, 1 kHz		100 mV rms
Slew rate	Measured slew rate at output with preemphasized square-wave input	Over 7 V/μs, instrumentation limited

pared. Neither was the prototype design; each matched in frequency response with a cartridge source. One performed well and one badly on the square-wave test. The difference was plainly audible even without instantaneous A–B comparisons.

APPENDIX

Response of a preamplifier to RF interference is becoming an increasing problem. A bill now before Congress would legislate the performance of high-fidelity devices with respect to their susceptibility to RFI. Any design for RF filtering at the input of high-fidelity amplifiers will need to be subjected to close scrutiny for not causing any audible difference while still rejecting the RF. Also, the legislation may be so restrictive that it prevents making preamplifiers which do not interact with cartridge source impedances. The technique for predicting and measuring cartridge inductance interaction discussed above should be helpful in the optimum design of any required filters.

ACKNOWLEDGMENT

Contributions toward the research and development of this preamplifier have been made by Frank Kampmann, Joe Seale, Gary Chanson, Andy Petite, Bruce Gregory, Don Lloyd, and Henry Kloss. Jay McKnight and Bernhard Jakobs provided valuable information. Mark Davis and Frank Kampmann reviewed the manuscript and made useful suggestions. D. Wright typed the manuscript through its revisions.

REFERENCES

[1] D. Campbell, W. Hoeft, and W. Votipka, ''Applications of the uA739 and uA749 Dual Preamplifier Integrated Circuits in Home Entertainment Equipment,'' Fairchild Application Note APP-171 (Jan. 1969). Presents different compensation schemes for phonograph preamplifiers with tradeoffs.

[2] B. I. Hallgren, ''On the Noise Performance of a Magnetic Phonograph Pickup,'' J. Audio Eng. Soc., vol. 23, pp. 546-552, (Sept. 1975).

[3] R. M. Bozorth, *Ferromagnetism* (Van Nostrand, Princeton, N.J., 1951), pp. 769-788.

[4] L. Happ, and F. Karlov, ''Record Warps and System Playback Performance,'' Preprint 926 (D-5), presented at the 46th Convention of the Audio Engineering Society, New York, 1973.

[5] H. P. Walker, ''Low Noise Audio Amplifiers,'' *Wireless World*, pp. 233-237 (May 1972).

[6] C. Huntley, ''Preamp Overload,'' *Audio Scene Canada*, pp. 54-56 (Nov. 1975).

[7] J. Kogen, B. Jakobs and F. Karlov, ''Trackability - 1973,'' Audio, Vol. 58, No. 8, p. 16 (Aug. 1973).

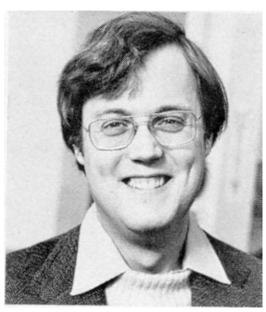
[8] R. Williamson, ''Magnetic Pickup Loading,'' *Wireless World,* (Letter to the Editor), (June 1973).

[9] G. J. King, ''Magnetic Pickup Loading,'' *Wireless World,* p. 592 (Dec. 1973).

[10] C. D. Motchenbacher, and F. C. Fitchen, *Low Noise Electronic Design* (Wiley, New York, 1973).

[11] J. G. Graeme, G. E. Tobey, and L. P. Huelsman, *Operational Amplifiers–Design and Applications* (McGraw-Hill, New York, 1971).

THE AUTHOR



Tomlinson Holman was born in Illinois in 1946. An early interest in sound recording led to work in the production aspects of theater, television, opera, and motion pictures while studying engineering and communications at the University of Illinois. Subsequently he made film sound recordings and mixes, taught a film sound course at the University, and started a small business which built custom audio equipment.

In 1973 Mr. Holman joined Advent Corporation as Audio Design Engineer and was appointed Chief Electrical Engineer in 1975.

He is a Member of the Audio Engineering Society.