THE B.A.S. SPEAKER

THE BOSTON AUDIO SOCIETY
P.O. BOX 7
BOSTON, MASSACHUSETTS 02215

VOLUME 3, NUMBER 4
JANUARY 1975

In addition to this month's two publications described below, the Speaker is crammed with articles and letters from members. The emphasis this month is on phonograph cartridges and arms. Al Foster describes his joy in his newly installed Decca arm, and links its performance with Leigh Phoenix's article on tonearm damping. Experimenter Bob Graham took Phoenix's article right into the shop, and he describes his similar pleasure at the performance of an STP-damped SME arm. And a template for adjusting stylus overhang has been provided by Al Foster.

The December BAS meeting featured a discussion of the Micro-Acoustics QDC-1E cartridge, and because the audience inundated the speaker, Arnold Schwartz, with questions about styli, cartridges in general, and the pressing process, this meeting became more of a tutorial than a sales pitch for the QDC-1E. The resulting meeting summary was condensed from three separate reviewers' versions, with each reviewer elaborating on various points. Thus this meeting report is a veritable review of the entire phono reproduction method. Even if you were at the meeting, read this summary and find out what information lay between the spoken lines.

To complete the package, Dan Shanefield has provided the BAS with his review of the December meeting of the New York Audio Society, which featured a discussion of the Audio-Technica line of phono cartridges.

If phono systems aren't your bag, then a note (more like a concerto) by Rene Jaeger on noise-reduction units or Harry Zwicker's hint on FM multipath detection and minimization may satisfy you.

Note that substantial contributions to this month's Speaker have come from outside of the Boston area (pieces by Phoenix and Shanefield, along with more than one letter), which we think is superb. Please keep on making this your society.

THIS MONTH'S PUBLICATIONS

Leigh Phoenix's article on tonearm damping is the result of an article we received about six months ago. The original was an extremely detailed, mathematical treatment of the tonearm/cartridge/stylus mechanical system. The editors felt that in its original form all but the PhD mechanical engineer would be scared off by the content, yet the revolutionary conclusions Leigh reached were, if correct, obviously of vital interest to the members of the BAS. Therefore we asked Leigh to simplify the mathematics and to return to us an article more accessible to the layman. This he did, and the result is the extremely enjoyable and highly informative publication presented this month. We can absolutely guarantee that you will see the results of Leigh's theoretical investigations appearing in commercial equipment in the near future. (But if you read the suggestions by Bob Graham, you may save some of the expense and beat the commercial competition.)

In a nutshell, Leigh's article establishes the necessity for tonearm damping to tame low-frequency resonances. Damping, by the way, is a force that is proportional to the velocity of a moving object (in this case the arm), as opposed to friction, which is a force that is constant for a moving object, regardless of velocity or pressure. Most manufacturers have, in the past, attempted to reduce tonearm pivot friction to a minimum, but few have ever tried to sell their arms on the basis of proper damping (e.g., note the number of hi-fi salons that offer damping modifications for SME arms). Bob's article helps to explain why damping is so sensible, and we ask you to absorb and spread this information to your friends in the audio community. —Harry Zwicker

This month's hardware offering is a pink/white noise generator. Chief Engineer at dbx, Inc., BAS member Rene Jaeger promised the simple generator at the November meeting and he has delivered quickly on his promise.

The usefulness of noise as a means of quickly spotting differences in frequency response is well known, and the Soundcraftsman pink noise disc is widely used in this application, as is FM hiss. Unfortunately, the disc is equalized more or less to the Fletcher-Munson relative loudness curve, and FM hiss may be degraded by the frequency response of the tuner and by faintly received signals. What's needed is a noise generator that's simple to build and that has repeatable, close-tolerance performance, and this is what Rene has designed.

It's a worthwhile project for experimentally inclined audiophiles, but it is one that requires a little care; perhaps you should have built at least one simple kit before tackling it. Try to find a friend or service agency with an oscilloscope who will check the unit for oscillation before you plug it into your system. If you are neat and careful, you will have no trouble, but high gain op-amp circuits, even though assembled by skilled engineers and technicians, sometimes go into oscillation for no anticipated reason.

Rene has also designed a lazy man's noise generator in the form of a "pinking" filter for FM interstation hiss. This is certainly the simplest and least expensive way to get a noise generator, but again, the frequency response of your tuner (especially with the common fault of rolled-off bass) will affect performance. We hope to have a note on this next month.—Jim Brinton

UPCOMING PLANS AND ARTICLES

We have decided to embark on two series of articles which we feel the commercial publications have more or less ignored: test equipment and book reviews. For the first topic, we propose to discuss the usefulness and the cost of obtaining an oscilloscope, using audio oscillators (including the BAS op-amp oscillator), test records and test tapes (possibly including the sale of a non-lab-standard yet useful BAS test tape), LED power displays, and anything else that comes to mind. You are sincerely invited to contribute requests for a discussion of some particular type of equipment you would like to hear about, or to submit articles to The Speaker for publication. As for the book review series, we do not intend to review the newest books on the market, but rather to bring to the attention of the members those classics that should be considered musts for the shelves of any serious audiophile. We would guess that anyone reading this newsletter likes to read or to be informed about audio. So with the help of a few resident pros, we shall begin to collect interesting titles and book reviews for dissemination in The Speaker. Again we ask you to send us such reviews or your comments about the idea. Subjects to be covered will include not only audio and hi-fi as such, but also the sense of hearing and psychoacoustics, electronics construction, and music.

Next month's issue will include a sequel to the A-B testing article by Dan Shanefield. We also hope to begin a series on interpreting and using manufacturers' magnetic tape specification sheets. We will include copies of these sheets when they can be included with The Speaker at no increased postal cost. A brief schematic and description of an adapter for changing the de-emphasis of an FM tuner from 75 to 25 microseconds is also in the offing, so don't buy the Switchcraft adapter ($8) just yet. Also in the works is a method for extending the bass response of the Advent 201 to 20 Hz.

RECOMMENDED LISTENING: THE BSO

Regardless of our prejudices, almost everyone who has heard either of the two BSO programs conducted by Klaus Tennstedt this December has experienced one of the most satisfying concert performances in memory. This 49-year-old product of East Germany has been "visiting" in the West for about four years, and has apparently conducted no other domestic orchestra prior to his appearance with the BSO. Without falling into the "capsule review" pit, let us simply say to all of you who receive transcribed broadcasts of the BSO, be absolutely sure to tune in when his Brahms program and his Bruckner's Eighth are being presented. And if your tape machines happen to be in gear at the time, especially for the dynamic Bruckner with its un-Boston-like brass, all the better. Be advised that the latter work is played, at least in the actual performance, without any intermission, and that it takes about 75 minutes.

If you do receive the BSO service in your area, how is the quality? And how long is the distribution delay? Please send us natives your comments, and we will relay them to the BSO Transcription Trust.

BAS TAPE PURCHASE

The BAS will continue to offer Maxell UD-35 reel-to-reel tape and Advent CrO_2 cassettes indefinitely, at $3.95 per reel or $39.45 per case and $1.95 per cassette or $23.50 per case, tax included. We will require that purchases for more than half a case of either type of tape be paid for in advance. Delivery will be made at the meeting following payment, but we will try to sell any remaining odd lots to anyone with the cash in hand after the firm orders have been filled. Note that the first $500 of tape was sold out within 15 minutes at the December meeting, and this with no prior firm orders.

BAS OSCILLATOR

Anyone interested in purchasing an oscillator kit is strongly encouraged to send a $10 deposit to Larry Kaufman at P.O. Box 7. The final installment of parts for the kits will not be purchased until sufficient deposits are on hand to obtain quantity discounts.

RECORDING HIGHS AND LOWS

By the volume of contributions received I should have taken the hint as promised, but I have been coerced into extending this column for a couple of more issues. Please take a moment and let me know if you have any real feelings toward some of the best or worst recordings in your collection; I can't believe everyone out there plays only test tones and tuners into their systems. Send lists, large or small, to either Joel Cohen, P.O. Box 135, Brookline, Mass. 02146, or the BAS at P.O. Box 7.

- Bruckner/Symphony Number 1/Jochum/BSO/DGG 139131/Probably one of the most "together" symphonies of Bruckner in listenable length (about 40 minutes). The beautiful adagio movement provides an excellent test for inner groove tracking; the strings at the conclusion of the second movement should sound clean and detailed with no "screeching" if your playback system is properly adjusted.—Alan Southwick

EQUIPMENT FOR SALE

Acoustic Research tuner, $110. Laurie Cote, 783-0920 evenings.

New 5-inch oscilloscope kits, same as Heathkits built for RCA electronics course, 70-mV/inch vertical and 1 V/inch horizontal sensitivity, 3 Hz to 5 MHz vertical bandwidth. Several units available at $60 each. Larry Kaufman, 449-2000, ext. 516.

LETTERS

The following collection of letters is printed relatively uncut and unedited. The first letter, from musician Dave Ranada, bears directly on the preceding recordings column. Have any of you ever purchased a record strictly on the basis of the recommendation of some anonymous BAS member, or possibly from the list that The Absolute Sound publishes? Often members have purchased discs after hearing only a few minutes at a BAS meeting, and even more often when they have heard a disc at the home of a friend. But somehow there is little interest in a column where favorites are only listed. A written review, however long, cannot replace a listening session, but with the quality of records (even Philips) on the wane, aren't recommendations on pressing quality and sound worth attention? The cost of a record is so small in comparison with the expense of the average BAS member's system, so what if we end up with a few recommended records that we don't like? Your comments please.

The other letters are somewhat less controversial, and require no further comment.

Musical Performance Versus Recordings

I must state some serious reservations about both the format and content of the first series of "Recording Highs and Lows" in the November 1974 issue of The BAS Speaker (Vol. 3, No. 2). They are neither informative nor necessarily valid as presented, and they contain little of use to the record purchaser. As an audiophile, the characteristics of recordings I'm most interested in include: pressing quality, sound quality (mike techniques, hall sound, frequency response, audible compression, distortion, etc.) and the instruments used (builder of the harpsichord, organ, etc.). Musically important parameters include attention to phrasing and articulation and their effects, choice of tempi and attempts at historical authenticity in regards to instrument choice, or use of rubato and ornamentation. I maintain that such qualities cannot adequately be described or summarized in reviews of such short length as the column's format dictates. I urge that this Ed Canby style of non-informative review be abandoned in favor of a more lengthy (sentences will be allowed) and hopefully more informative layout.

The "exciting sound and dynamic range, good pressing" heard from Columbia MS6366 may not be what the present day purchaser will encounter. Columbia's pressing quality has varied tremendously, and more modern pressings of older recordings may be quite poor (e.g., Stravinsky: Odeipus, M31129). If the recording is old and from an American company, some notice should be given as to the date of purchase.

The Verdi Requiem on London OSA1275 is described as "melodic." Since the music itself is quite melodic it is hard to tell just what is meant by "very melodic." Are the lyrical aspects of the music stressed at the expense of rhythmic definition?

Riley's "A Rainbow in Curved Air," described as "ultra-modern jazz on synthesizer," is neither ultra-modern, jazz, or on a synthesizer. The piece, an 18-minute 40-second elaboration (rhythmic-melodic) of a single chord (A major), uses several electronic instruments (electronic organ, harpsichord, rocksichord) but not a synthesizer. Electronic music techniques are used extensively, particularly tape loops to create ostinati of varying phase relationships. The effect is a strange combination of medieval organum and Javanese gamelan music, though very modern and Western in sound.

There is no reason for anyone to be disappointed or embarrassed at the purchase of the Brandenburg Concerti on VICS6023, as it is one of the better sets. The "lemon-sucking sour" brass is explained by the cover, which plainly states "performed in their original instrumentation" including "clarin trumpet" and "natural" horns. These instruments have no valves and depend completely on the players' ability to stimulate a resonance at particular points on the instruments' overtone series. For complex reasons, the resonances will not always be "in tune" with the other instruments in concerti One and Two. In addition to this, some notes on the natural horn are not playable unless the player handstops his instrument by placing his hand into the flaring bell of the instrument. Hand stopping results in a change of tone color in certain notes of the horns' range. Thus, the sourness of the brass is due to the instruments used, and is in fact what Bach would have heard and expected from his players.

The complaint is only relevant to two out of the six concerti, and no mention is made of the fine playing of the other soloists or of the "sourness" of some of the woodwinds. I happen to prefer the slight irregularities produced since they help the ear discern the lines of polyphony. From the audiophilic viewpoint the recording is a bit too distantly miked, with a touch too much hall reverb both in quantity and quality (reverb time). Musically the performances are generally a bit too slow (compared to Marriner and Richter) but are played with an intelligent knowledge of structure, phrasing, and ornamentation. The playing is quite competent. Edward Tarr's performance of the Tromba line in the second concerto is extremely exciting, considering that the trumpet he uses has no valves. This RCA recording is the best of the budget versions, and it is able to stand with the rest of the better Brandenburg recordings: Britten, Dart, Harnoncourt, Marriner, Menuhin, and Ristenpart (note: not Newman, Klemperer, Casals, Karajan, Kehr, or Richter).

I hope these notes have been helpful, particularly to future reviewers. The point to realize is that sonic qualities and musical parameters cannot be concisely summarized in reviews of the length and format presented in the BAS Speaker.—Dave Ranada

Remarks on Amplifier Testing

Thanks for publishing my article in the November Speaker, and thanks also for the acknowledgements amidst the Dyna-Marantz-Phase discussions.

By next year I'll have the whole Absolute Sound-approved chain of components, so I can A-B each one of them against the corresponding Ortofon, McIntosh, and Bose equipment that they seem to hate. Actually, I agree with Jim Brinton's point of view, presented in the September Speaker, saying that the whole chain is probably not necessary for hearing differences if they are there. But I think that having the whole chain is useful for convincing people.

For example, I can just hear The Absolute Sound guys asking, about your 400-500-700 amplifier study, "Did the Boston Bunch use a recording with sharp peaks like the Sheffield III or Ambiphon? That's the real test of a high power amplifier. And was the preamp overload-proof? Did the loudspeakers have super-low mass, to be ringing-proof?"

I imagine that the answer to the first two rhetorical questions is "Yes." Regarding the third one about the loudspeakers (AR-LST), you folks did mention your doubts about it. But would The Absolute Sound staff ever believe anything done on Acoustic Research speakers? Maybe you are not obsessed with the questions raised by The Absolute Sound, like I am, but being obsessed is sometimes a useful trait in hunting down an elusive truth. I remain obsessed for the while.

Sorry about the above critical-sounding note, especially when I've hardly done anything yet myself. I actually was thrilled (literally) to read your 400-500-700 report, since this is exactly what is needed—an independent group with ample facilities, willing to change their minds if necessary. I'm just trying, in this letter, to contribute at a distance, by being a devil's advocate. — Dan Shanefield (New Jersey)

The Dahlquist DQ-10

I own four Dahlquist DQ-10 speakers (with the newer woofers). Rest of system: Sony 2251 direct drive; SME (non-detach); Shure V-15/II; Crown IC-150; Sony TA-3200F; Dynaquad; Sony 755 deck. Listening room is 23 by 18 feet and live sounding. After living with this system for several months I am continually impressed with how good the Dahlquists sound. On A-B comparisons with FMI 80's, double Advents, FMI 100's, Dyna A-35's, and JBL 100's, the Dahlquists wipe out the competition in most categories, especially midrange clarity and imaging. While I have not been able to compare the Dahlquists with more worthy and comparably priced competitors (e.g., AR-LST's, Quad's, Magneplanars, etc.), I am inclined to suspect that the new woofer speakers are better than (your review of) the old woofer speakers by a margin large enough to be worth retesting.

In addition, you tested these speakers on the floor. The manufacturer suggests experimenting for best effect, and in certain cases elevating them. BAS members are certainly the sort of people who will (1) experiment with positions, and (2) follow your advice (if it is sound) for best position ing. So an assessment made at the best position will provide more useful information than an assessment made with the DQ-10's only on the floor.

The Speaker is great, and so is the BAS. Are you interested in my drumming up more (paying) members here in Winnipeg?—Les Leventhal (Winnipeg, Canada)

COMMENTS ON "COMMENT ON SIGNAL PROCESSING"

It is fair to take exception to the statement in the November BAS Speaker that "noise-reduction gear makes high-amplitude signals a bit noisier." We must say what we mean by "a bit noisier."

If the signal we are concerned with has an amplitude of 1 volt and the noise associated with it is 60 dB below that level (as with a good record), and if we pass that signal through a processor with a noise level of -60 dB (referenced to 1 volt), then the processor will add 3 dB of noise to the signal. This could be considered a noticeable degradation.

If we decide that a 1-dB degradation is all we can tolerate, then the processor must have a 7 to 8 dB better noise figure than the source, in this case a -67-dB noise level relative to 1 volt. If the processor is a before-and-after device (such as a tape noise-reduction system), it will add its noise to the signal twice and so must be an additional 3 dB quieter—a -70-dB noise level in our example. Suppose this is a popular system promising 10-dB effective noise reduction and is used with a tape recorder having a 50-dB signal-to-noise ratio. At zero VU input (a 1-volt signal) the output noise will be -50 dB, all coming from the tape recorder. The signal will mask the noise to some extent, but it is still only 50 dB away from the signal, instead of the 60 dB mentioned above. At lower input signal levels, for example zero input, our -60-dB processor source noise will get boosted by the encoder to -50 dB. This noise is then added to the tape noise (also at -50 dB) during recording, and on playback the net noise reaches the decoder at -47 dB, to be attenuated on decoding by 10 dB to -57 dB. So we lose only 3 dB of our signal and gain 7 dB on the tape recorder's noise.

All this is to say that the statement we wish to take exception to (i.e., that noise-reduction gear adds noise to high-level signals) is true—for some systems. But let us now examine a before and after (Dolby plus dbx –Ed.) system which claims 30 dB of noise reduction; we will use the same signal and tape recorder as before. The signal enters the encoder at 1 volt-96 dB above the residual dbx input noise—and leaves at 1 volt or zero VU, our choice for unity gain.

If the signal fundamental is below 1 kHz, its high-frequency (noise) spectrum is pre-emphasized by 6 to 12 dB; the noise part now leaves the encoder at -48 to -54 dB. The signal and its noise are recorded and passed to the decoder where a complementary de-emphasis of the high frequencies gives us an instantaneous signal-to-noise ratio of -58 to -55 dB. We have thus gained 5 to 8 dB noise figure over the tape recorder itself at zero VU.

In the absence of signal, our source noise at -60 dB will be pre-emphasized by 12 dB (assuming it is primarily hiss) to -48 dB, then compressed by 2:1 (in dB) to -24 dB, pass through the tape recorder unmolested (tape noise is at -50 dB – Ed.), be expanded to -48 dB and de-emphasized to -60 dB in the decoder, and emerge 36 dB above the decoder's (-96 dB) output noise. The "signal" noise now will be 26 dB above the tape noise (50 minus 24 dB – Ed.), while it would have been 10 dB below the tape noise (50 minus 60 dB) without the processor—a 36-dB signal-to-noise (noise-to-noise ?) improvement.

Of course, what we need to know is how little noise, or better, what signal-to-noise ratio, we really need for music listening. Some listeners hear acutely enough to detect noise some 90 dB below a pure 800-Hz tone in an anechoic environment. I would venture a guess that at listening levels of 90-dB SPL in a quiet, well-damped living room, most of us would not be bothered if noise were 50 to 70 dB down, depending on program material and spectrum masking. In material with predominant low-frequency energy, such as pipe organ, tympani, or bass, the higher signal-to-noise ratio (70 dB) would be required because of the poor masking of hiss by these low tones. Also, even with material where high-frequency energy is predominant, masking is poorer with purer toned instruments such as orchestral bells, harp, and piano, and again 70 dB is required.

So if we reach for the 70-dB figure to cover all cases and add 20 dB on top to cover peaks, we find ourselves with a 90-dB instantaneous dynamic range criterion. Not unreasonable when we consider that records can achieve this sort of range at mid-frequencies. But careful attention must be paid to noise at every step in the recording-reproduction process to ensure the retention of a 90-dB dynamic range.

A good microphone will have a self-noise of less than 20-dB SPL equivalent—that is about the loudness of a pin dropping—and a 1% distortion point of 130-dB SPL—that is about as loud as you can bear—giving us more than 100 dB usable output signal range. A good mike preamp could handle this without adding either noise or distortion.

At the other end of the chain, 110-dB dynamic range preamps and power amps can be built; some already come very close. As for loudspeakers, horn units can reproduce 130-dB SPL with fairly low quantities of some kinds of distortion, one electrostatic design (the Dayton-Wright) can reach 120-db SPL over most of the audio frequency range, and some direct-radiator dynamic speakers can crank out 110-dB SPL maximum (no comment on distortion or bandwidth).

So it seems we might be able to preserve and reproduce 90 to 100 dB dynamic range in the living room, except that I forgot to mention what happens between the mike preamp and the listener's preamp. You see, this is where we could use some form of signal processing. Ever since Mercury's "Living Presence" was overshadowed by the spectre of stereophonics, the concept of a "sound happening" being heard from one point of view has gradually lost ground, until now the concept of a sound event localized in space unfortunately has been replaced by "sound" hung on the wall(s) somewhere between your speakers (two or four of these; and you decide where you want the speakers—anywhere in the room will do). This sound degrading circumstance has come about through the notion, remarkably widespread, that all the instruments taking part in the "sound happening" should be heard equally well. Therefore, many, many microphones are distributed lavishly about the "happening," each one discretely amplified and equalized, with its output stored in bits and pieces on Scotch tape and later fed to a mixing console where every one of those formerly discrete signals, each different in terms of phase or time, is allowed to mix with every other signal until none is discrete enough to act coherently. The resultant indiscretion is then packed onto two or four channels and again stored on tape. Eventually the signals on this tape will be searched once more for any signs of discreteness such as "excessive" peak-to-average ratio, "too much" out-of-phase low-frequency energy, or "too wide" a dynamic range. These "excesses" will be removed prior to finally cutting the record.

It must be remembered that each one of these operations adds a veil of noise and distortion to a process where excess noise is routine. Therefore introducing a noise-reduction system after the microphone could be helpful, but only if carried through directly to the user at his preamp. Allowing the listener to perform the mixing, equalizing, and time-domain correction is the only way to maintain our 90-dB goal. You see, we cannot mix or equalize those encoded signals without losing the information we need to decode them. (I was wrong. J. B.)

Not that we would want to mix things up or unequalize our flat microphones. But could we live with only two mikes adjusted properly, with no chance to change the sonic perspective in midsong, or to dance from instrument to instrument tossing highlights to and fro—a carnation for the first violins, a rose for the bassoon? And all those lovely knobs, switches, and meters at the studios; they'll get so rusty. Wouldn't that be a shame ?—Rene Jaeger

A SIMPLE MULTIPATH INDICATOR FOR ALMOST EVERYONE

Almost all deviations from perfect FM signal pickup at the antenna, for example, multipath and alternate-station interference, cause rapid, audio-frequency signal-strength variations in (or amplitude modulation of) the radio-frequency input to the tuner's IF amplifier strip. The unwanted amplitude variations are, to some degree, reduced by the automatic gain control (AGC) signal system before reading the FM demodulator. But because this AGC control path is intended principally to keep the IF signal constant as the tuner is varied from station to station, the AGC signal is normally filtered to ignore audio-frequency amplitude fluctuations occurring above about 10 Hz. Thus rapid multipath amplitude variations are not used by the AGC line, and are only crudely indicated by fluctuations on the signal-strength meter (which displays some form of this AGC signal). All interfering AM variations are simply fed through to the demodulator, unknown to the user, where the detection circuits can sometimes reject at least a portion of the spurious AM signals with a minimum of audible distortion. Detection of the presence of multipath is therefore difficult, making its correction (by fine tuning or antenna rotation) impossible.

In tuners, however, a signal similar to the AGC signal, but not so severely filtered as the AGC line (and thus able to contain the more rapid amplitude fluctuations in signal strength) is fed to a phono jack labeled "oscilloscope—multipath vertical output" on the rear panel of the unit. When this signal is displayed on an oscilloscope, fluctuations in the display height are seen and the orientation of the receiving antenna can be changed to minimize them until the AM distortion of the RF signal, including multipath, is at a minimum. Although an oscilloscope is valuable to any audiophile, most of us do not have them, and those who do normally don't leave such a bulky instrument permanently connected to the tuner strictly for multipath detection.

A very simple alternative to the use of a scope, and one which nearly everyone can use, is simply to listen to the audible detected AM signal available at the "vertical" output jack. One can easily connect this jack to one channel of a spare high-level preamplifier input and monitor this audible signal while tuning for center of channel and rotating the antenna until one obtains the minimum audible level regardless of distortion. At this point the signal is optimized, and one can return to the tuner position for listening. Note that the audio output at the "vertical" jack may be of a very low level, but may be superimposed on a quite high dc voltage. Therefore, do all switching at zero preamp volume, and crank up the gain only to monitor the test signal. And don't forget to turn the volume back down before switching back to the tuner position, or the transient pops will blow your speakers. If your preamplifier/amplifier is dc coupled, add a capacitor to the multipath output before connecting.

How simple and inexpensive can one get? Of course if your tuner does not have a multipath jack, you will have to do some digging in the unit, but most manufacturers are now providing multipath outputs free of charge, hoping that you will buy one of their several-hundred-dollar scope displays to use them. Buy a 99¢ phono cable instead, and you can do almost as well. — Harry Zwicker

Comment. Since Harry came up with this idea, at least three BAS members have tried it, and with excellent results. Laurie Cote now uses the approach with his Pioneer TX-9100, as does Al Foster with his. I have compared the "audio-tuning" approach with oscilloscope tuning on my elderly Scott tuner and find that both multipath and center-of-channel tuning are almost identical regardless of whether the scope or the listening approach is used. Note, though, that when Harry says that one tunes for a null regardless of distortion, he means just that; the signal near the null point may be severely distorted indeed. —Jim Brinton

COMMENTS ON `TONEARM DAMPING"

After reading Leigh Phoenix's article on damping, I was sufficiently impressed with his findings to re-evaluate my own tonearm's performance. I remembered my earlier days in hi-fi, when the Weathers turntable "mounted on a seismic platform" (as the ads used to say) together with its (then) revolutionary 1-gram cartridge mounted in the Weathers damped tonearm generated "oohs" and "ahs" whenever a salesman would toss the arm onto the spinning disc. Nothing bad ever happened, of course, because of the viscous damped arm; it just settled very gently into whatever groove happened to be beneath it at touchdown. Very few of us thought of this as anything more than an impressive demonstration; I certainly didn't realize the importance of damping then, and it has been only recently that I've even encountered damping again.

A couple of years ago, I owned the absolutely beautiful/ugly (take your pick) Transcriptors Hydraulic Reference turntable. It was impressive, with the records sitting on those 24k gold pads; too bad the Transcriptors' idea of anti-shock-suspension was to mount the whole thing on hockey pucks. But the Fluid Arm—that was different! It was a single jeweled unipivot, surrounded by a thick, gooey damping material. The arm was very low mass, and, together with the damping, which I now realize is very important, probably made it one of the best tonearms around.

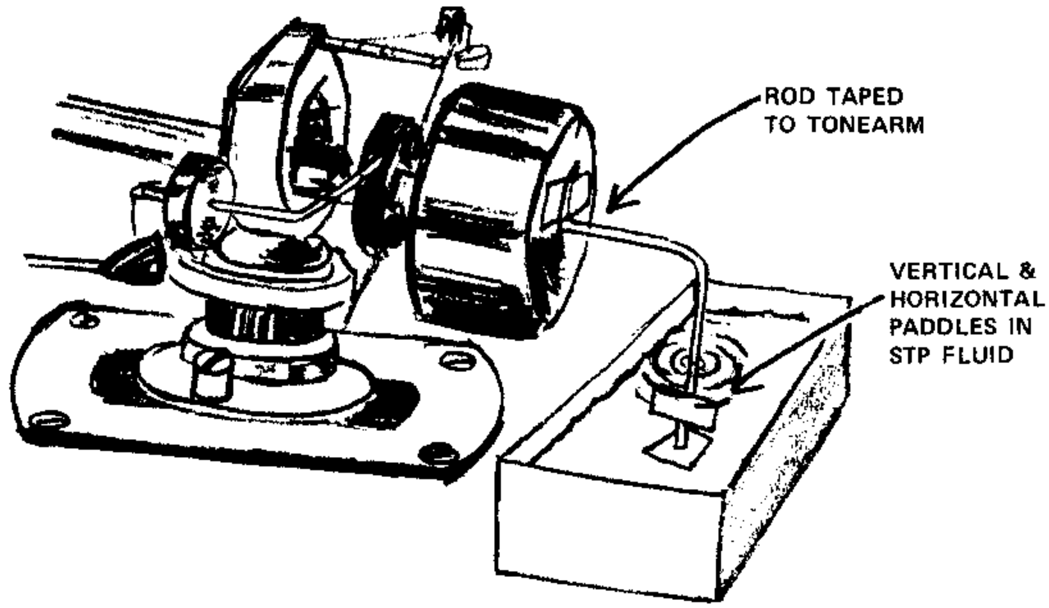
I sold the Transcriptors, hockey pucks and all, and resurrected my AR turntable, eventually improving it by adding the SME 3009 Series II arm. Now we come to the real reason I'm a believer in damping. When I added some of the newer cartridges to the old SME a couple of years ago, I began to notice some trouble. An ADC 10E Mk IV, for example, wobbled all over the record—I just couldn't keep the stylus from moving around, independent of the tonearm. Likewise, the XLM was less than spectacular, and warped records were now a real annoyance. Happily, SME came out with their new series arms, and I purchased one. NOW I would have perfection—I thought!

Almost, but not quite. True, the arm did perform a lot better, but there were still some warped records, particularly organ discs, where the arm bounced several times after each sharp warp, and I could hear the organ tones being modulated by this. Still, it was better than before, and the XLM was now my favorite, so I couldn't bring myself to give up the combination.

Well, all the problems are now in the past. After reading the article on damping, I tried an experiment. I mounted a piece of thin brass rod on the back of my tonearm, and fastened two "paddles" on the end of this rod. These paddles are submerged in STP oil additive, which is held in a small metal trough. The paddles are mounted perpendicular to each other, so there is force applied whenever the arm assembly is in either a vertical or horizontal motion (see sketch).

The principle, of course, is viscous damping. You can demonstrate this principle by placing your hand, fingers together like a paddle, in a bucket of water and moving the flat of your hand slowly through the water. You will feel almost no resistance. If, however, you try to move your hand quickly, you will feel considerable resistance; the quicker you try to move, the more resistance you will feel.

This is the principle on which a damped tonearm works. It will follow the relatively gradual motions of a warped record, but the damping action will "put the brakes" on any bounce following a particular warp. This is superior to merely taking an undamped arm and tightening up the



bearings so that you increase the Coulomb friction, which is constant with velocity; that would only cause the arm to stick and do other unpleasant things. By starting out with a low-mass, low-friction arm and then adding controlled resistance with fluid, you will have a really first-rate damped tone arm.

As for my own arm and cartridge, the XLM has never sounded so good. The previously annoying records are now playable with absolutely no "bounce," no warbling tones. The stylus stays virtually motionless (to the eye) relative to the tonearm, so I'm not generating low-frequency tones to modulate all my music.

Although my system is somewhat clumsy-looking, the improvement in performance is so profound, particularly with a high-compliance cartridge, that I would not consider being without it. I'm sure that adding fluid damping to the old SME arm would have made the XLM very usable. How many cartridges like the XLM are considered "too fussy" for most tonearms, when the addition of damping might have made them compatible?

The only maintenance that might be required would be an occasional oil change—perhaps every 1000 miles of stylus travel or so, to get rid of accumulated things that might end up in the bucket of oil (like dust or a housefly). This device could be added to any tonearm, even those that could not otherwise have viscous damping. You could use a heavy paper clip for the rod, and glue some small paddles on with 5-minute epoxy. The container for the STP could be just about anything. It should be wide enough to allow the full horizontal motion of your tonearm, and deep enough to completely cover both paddles with fluid. The amount of damping force exerted can be varied by changing the size of the paddles. I've found that the vertical motion paddle works well if it is between 1/4 and 1/2 inch in diameter. The horizontal paddle size is somewhat less critical.

Try it—you'll like it! ! —Bob Graham

MY EXPERIENCE WITH THE DECCA INTERNATIONAL TONEARM

Low-mass, very high-compliance cartridges designed to track at forces of less than 1 gram have revealed the limitations of the vast majority of conventional tonearms and have given rise to serious problems of compatibility. These problems, as Leigh Phoenix indicates in his article, often result in uncontrolled and misplaced resonances owing to the lack of tonearm damping.

My experience with the Decca International tonearm, which has damping at its pivots, has convinced me of the necessity of damping and has given me a fair indication of the audible and visible effects of its absence. To observe these effects, simply play a record with a mild but sharp warp on your present turntable and observe if the stylus, instead of remaining stationary with respect to the cartridge body, tends to bounce up and down in the cartridge body (viewed from the side of the cartridge) as it crosses the warps.

Some minor deflection at the beginning of the warp is to be expected in any realistic tonearm, but the arm, rather than the stylus assembly, should track most of the warp. In a proper tonearm/ cartridge combination, the stylus beam will never go out of position and the stylus tip will never move with respect to the cartridge body. Any tip movement away from the rest position indicates too much mass and/or too little damping.

The net effect of a poor tonearm/cartridge combination, whenever a large undamped resonance falls within the audio band, is that oscillations may be excited by the recorded signals, producing a rising bass characteristic with distortion and mistracking, and whenever the resonance is below 10 Hz, disc defects such as warps and ripples will throw the arm into gross oscillation, thus producing severe variations in tracking pressure and dramatically reducing trackability. This results in distortion and a lack of detail.

To test the effects of inadequate damping, I tried several cartridge and arm combinations: the Philips 212, Dual 1218, and Decca International arms, and the Audio-Technica 20SL, the Micro-Acoustics QDC-1E, and the Shure V-15/111 cartridges. The Decca tonearm was used as my "damped" standard and was mounted on an AR turntable. The Audio-Technica and the Micro-Acoustics cartridges were used because both track best about 1.5 grams. The Shure was to represent the high-compliance cartridges that track around 1 gram. The Philips, Dual, and Decca arms all have very low mass and negligible friction.

I mounted the cartridges in the arms and observed their 1-kHz traces on an oscilloscope and inspected visually for needle bounce as described above.

The Shure did not fare well in either the Philips or Dual arms. An unstable 1-kHz trace was observed on the scope and the stylus often deflected toward the body of the cartridge with the record warp. Sonically, on musical passages, the Shure suffered from a lack of focus and directionality. Instead of remaining stationary, the instruments would wander around the stage. However, in the Decca arm, the Shure cartridge became a new beast! Instruments became stationary and the detail, particularly in the low bass, was uncanny. And for the first time, I could hear a pure, uninterrupted 1-kHz tone from a record!

I then tried the Audio-Technica and the Micro-Acoustics cartridges in the Philips and Dual arms. The MA seemed to be the most insensitive to the arms and produced as stable a trace as did the Shure mounted in the Decca arm. The AT was slightly less stable than the MA, and its performance could have benefited from the damped Decca arm.

Would my results have been different if I had used the Shure SME Improved arm instead of the Dual and Philips? Or would the Shure type III cartridge have been sonically identical in the SME and the Decca arms? I doubt it. Hi-Fi News & Record Review (August 1974) reports that the low-frequency resonance of the SME Improved (fixed shell) with the Shure V15/III cartridge occurs at 13 Hz, a very dangerous place to have oscillations. This location is bad because record eccentricities (e.g., offcenter holes) cause excitation frequencies of 0.5 to 0.75 Hz at 33⅓ rpm, record warp excites in the range from 0.55 to 5 Hz (depending on its severity), and the turntable/ platter combination can generate "squat" (vibration of the playing system with respect to the base, caused, for example, by jarring) from about 0.1 to 1 inch, generating natural frequencies in the range from 3.5 to 10 Hz. The ideal tonearm/cartridge combination would have its natural resonant frequency well above 10 Hz. A practical limit would place the lowest frequency of vibration closer to 20 Hz (see Gary T. Nakai, "Dynamic Damping of Stylus Compliance/Tone-Arm Resonance," The Journal of the Audio Engineering Society, Vol. 21, No. 7, Sept. 1973; pp. 555-562.)

So far as low friction is concerned, would the use of the SME arm have drastically altered my results? The SME and Decca arms probably have the lowest friction of the group; however, as Phoenix indicates, unnecessarily low pivot friction can increase the amplitude of the resonance, making the problem worse.

Would their differences in mass have made a difference? No figures are available in my literature on the effective mass of the arms I tested, but visual inspection tells me that the SME, probably the lightest of the group, is not four times lighter. All the arms are roughly the same or at least there is not enough difference in effective mass to drastically alter my results. Because, as Phoenix points out, a fourfold reduction in tonearm/cartridge effective mass only doubles the natural resonant frequency of the combination. Mass is not the issue; controlling or minimizing unwanted resonances is. Damping, when properly applied, does not interfere with the normal horizontal or vertical movement of the tonearm/cartridge, but it does act to minimize resonances or rapid oscillations unrelated to the recorded information. There are currently only two damped tonearms on the market: the Decca and the Keith Monks Audio Ltd. (KMAL). Both sell for about $135, and both are available at Suffolk Audio in Boston.—Al Foster

DECEMBER NEW YORK AUDIO SOCIETY MEETING

At the December meeting of the New York Audio Society, Mr. Jon R. Kelly, General Manager of Audio-Technica (AT) USA, described the AT line, claiming three main advantages for their cartridges:

1. Twin magnets having lower total mass than the usual single magnet,
2. Emphasis on quality control and reliability, and
3. Mechanical damping of stylus resonances.

Their top-of-the-line diamond styli have square cross sections at the point where they are mounted into the stylus tubes, providing more accurate alignment than the usual round cross sections. A steel wire is ordinarily attached to the back end of a stylus cantilever tube to fix the tube's axial position. During manufacture, the stylus tube is moved axially until a rubber hinge is compressed to just the right tension; more compression would give more damping but less compliance. Audio-Technica holds the wire in its optimal position with a setscrew instead of the usual drop of solder, claiming that this allows more precise adjustment.

Mr. Kelly said that the AT design for the hinge, which involves rubber disks instead of the usual rubber doughnuts, has improved the mechanical damping qualities. (He would not elaborate on this, saying that there was not enough time to explain it in detail.) The AT handout literature contains the following statement about their cartridges; "elimination of high frequency mechanical resonances within the audio range decreases groove wear." This emphasis on the mechanical damping is in contrast to the ADC XLM literature which mentions "Controlled Electrodynamic Damping (C.E.D.)" and to the Shure article in Audio (August 1974, p. 22), which mentions electrical filtering of resonance output. Theoretically, mechanical damping might have advantages over electrical damping or filtering, especially at low frequencies.

Recent reviews of the AT cartridges showing square-wave response have appeared in Audio (August 1973, p. 32 and March 1974, p. 36) and in High Fidelity (December 1974, p. 51). The stylus resonances did appear to be minimal, although it should be noted that these tests do not clearly distinguish between true damping and simple electrical filtering of the output. The March 1974 Audio tests are unfortunately flawed by electrical filtering in the test circuitry, as shown by the excessive curvature in the waves obtained from all the cartridges.

In addition to its recommended use with CD-4 records, Mr. Kelly said that the Shibata diamond tip gives better sound with stereo records than does the elliptical diamond. AT advertisements have also been stressing this recently. He claims that the Japanese developers of the AT products have extensively tested both kinds of stylus for stereo use, using live-versus-recorded comparisons, a technique which he said is much more popular in Japan than here. He said that AT will soon publish extensive data showing decreased record wear when the Shibata, as opposed to the elliptical, is used with stereo discs.

Vertical tracking forces of 1 to 2 grams were recommended with the Shibata, and lower forces would give poorer results, even with stereo discs. He claimed that most cartridge manufacturers have found by trial and error that ultra-high stylus compliance gives poor results with CD-4 discs. Therefore AT does not utilize unusually high compliance. It was claimed that ultra-low weight tonearms are not required with the AT line, as they seem to be with either ultra-high compliance or poorly damped cartridges.—Dan Shanefield (New Jersey)

IN THE LITERATURE

Thanks to Dana Craig for his contributions—H. Z.

Journal of the Acoustical Society of America, Nov. 1974

- Simulation of the Effects of Recruitment on Loudness Relationships in Speech, by Edgar Villchur, Foundation for Hearing Aid Research, Woodstock, N. Y. Listed for its author, rather than for high-fidelity content, but still an interesting article, complete with demonstration "soundsheet."

Audio Scene/Canada, Aug. 1974

- Erroneously listed as Stereo Scene/Canada, in the December issue.

Boston, Jan. 1975

- High Fidelity: The Sound and the Fury, by Len Feldman. Two parts (stereo and quad), strictly for beginners; save your money.

db, Sept. 1974

- Amplifying the Orchestra. Specific microphones suggested for each type of musical instrument.

db, Nov. 1974

- Burwen's home studio system (unbelievable I).

Electronic Design, Apr. 12, 1974

- Schematic of a practical class D switching power amplifier, with some theory; comment and reply in Aug. 2, 1974 issue.

Electronics, Aug. 8, 1974

- Article on cable test of Dorren four-channel discrete broadcasting system.

Electronics, Aug. 22, 1974

- Article on variable speech control (VSC) system which permits slowing down or speeding up speech playback with minimal effects on intelligibility. Not suitable for high-fidelity as yet. Includes "soundsheet" which could be demonstrated at the next BAS meeting if interest is expressed (to Dana Craig, BAS).

Electronics, Oct. 17, 1974

- Brief technology update on audio.

Electronics, Dec. 12, 1974

- Seimens Promises Cordless Headset. Infrared lamp floods room with modulated signal which is picked up by receiver in the headset.
- Videobeam Projection TV Looks for Big Market. News update on previous sales and on model 1000A with detent UHF tuning.

Popular Electronics, Jan. 1975

- Stereo Scene: Mikes and Miking. Skip it.
- How to Read FM Tuner Specs. Skip this too.

Radio Electronics, Dec. 1974

- Better TV Sound. Networks to use 15-kHz bandwidth in the future.
- New Concepts in FM Tuner Design.
- Design of Audio Feedback Circuits. Tutorial on, for example, RIAA feedback loops.

Radio Electronics, Jan. 1975

- Miscellaneous: Ben Bauer, SQ patents (p. 12): 5000-watt electronic organ in Carnegie Hall (p. 14): advertisement for Crown VFX-2 electronic crossover, $300 (p. 80).

Wireless World, Nov. 1974

- Active Crossover Networks. Design uses single woofer and electronically crossed-over mid- and high-frequency units to reduce size and cost: "better sound."
- Audio '74 at Harrogate. British hi-fi show review.
- Nonlinearity of Air in Loudspeaker Cabinets. Examination of distortion caused by different types of loading at low frequencies.
- Quadraphonic Broadcasting - Current Proposals and the Way Ahead.

DECEMBER BAS MEETING

Business and Open Discussion

A full house of about 80 members met on December 15 at BU. Jim Brinton began the discussion with a request for new contributions to The Speaker, including full articles, short comments, postcard contributions to the "Recording Highs and Lows" and the "In the Literature" columns, and volunteers for editors. Articles on music topics are especially requested, as are suggestions for topics which other members of the society can address. Perfect style and spelling are not required, as the editors will take care of these details. See if you can't come up with some good material after a short session pounding the typewriter on one of these cold winter evenings. Contributions should be sent to P. O. Box 7.

Peter Mitchell demonstrated samples of a live concert he recorded on an Advent cassette unit, using a pair of nude model 814 Thermo Electron electret capsules (their larger diameter models). These microphones are omnidirectional but have a small rise about 8 kHz on axis, which is compensated by an off-axis rolloff above the same point. Peter suggested facing the microphone about 90 degrees away from the sound source (toward the ceiling) to flatten the response; this orientation also provides a good sense of the hall acoustics. He cautioned also against placement too close to the audience, where the ultrasonic component of applause can overload the microphone preamp. These cautions aside, Peter claimed that these units are comparable to AKG C451's, which would cost the recordist about $400 a pair, while a pair of 814's will cost around $50. (The similar Group 128 microphone, which uses the smaller Thermo Electron capsules, sells for $149 each and reportedly sounds "beautiful.") The specification sheet is included in this issue of The Speaker for reference by those interested in a group purchase of the raw capsules; further details will be given in a later issue. The demonstration of portions of Haydn's Symphony Number 104 as performed by the Boston Philharmonia was highly pleasing, particularly in view of the simplicity and low cost of the mikes, and evoked sufficient interest in the microphone units that a long discussion of the stability and channel match of the electrets evolved. Peter reported that two of his three units were very well matched, while the third appeared to have lost about 2 dB of sensitivity in the soldering process. Al Southwick warned that Sony advises users to keep their electrets below 120°F, and several other members repeated rumors that electrets loose their sensitivity at the rate of about 2 dB per year—and even more rapidly with exposure to salt air. Peter replied that this sensitivity loss was exponential with time, and that most of the loss would have occurred by the time a user received his elements. He also lives within 1000 feet of the shore and has not yet observed any ill effects due to the environment.

In other business, it was announced that several further BAS meetings will be held at GTE Sylvania in Waltham. Owing to the lack of public transportation to this location, arrangements will be made for car pools from BU's Sherman Union out and back for those in need of a ride. The February meeting will be held on the ninth, one week earlier than usual, in order to accommodate the invited guest speaker. The BAS newsletter will not be distributed at this meeting because of the short preparation period, and all copies will be mailed to members about one week after the meeting. Peter Mitchell announced that his comments on the tests of the TEAC 2300 tape unit reported in High Fidelity were incorrect (see last month's Speaker) and posted the reply from the editors. Andy Petite of Advent replied that cassette recorders are tested at a 3 dB higher level than reel-to-reel recorders, and that most of the measured "IM" results are not accurate *as* published. A spectral analysis of the output during the IM test of the Advent 201, made with several different tape formulations, indicates mostly modulation noise distortion and very little of the spectrum associated with IM distortion.

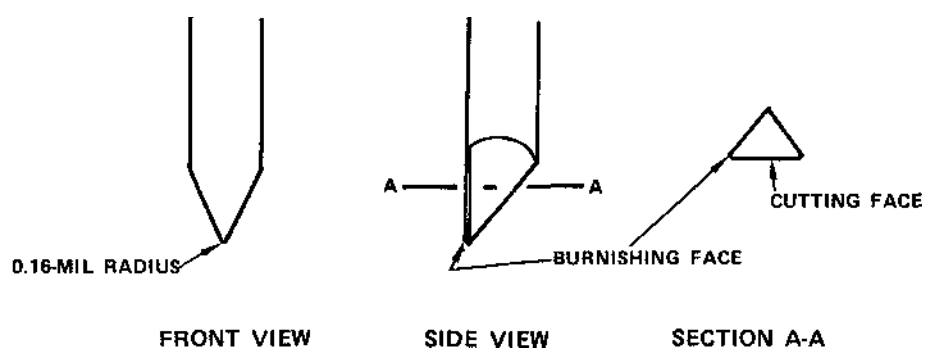
Al Foster distributed Maxell 1800-foot UD tape at $3.83 per reel and Advent C-90 cassettes at $1.90 each; a $500 supply was exhausted within minutes, and a further order is to be placed. Dave Letterman announced that due to 20% delivery and slow service; the domestic record buying service is now discontinued. The imported disc buying service will, however, be continued; another order will be placed when the total purchase exceeds about $60. The Burwen "Perfectly Clear" records are also sold out. Al Foster reported his pleasure with the viscous damped Decca arm. Henry Nicholas, formerly of Audio Lab and Tech Hi-Fi, was reported to have opened a hi-fi service shop in Harvard Square under the name Stereo Lab.

Meeting Feature: Arnold Schwartz—The Micro-Acoustics QDC-1 Cartridge

The featured speaker was Mr. Arnold Schwartz, president of Micro-Acoustics, manufacturers of high dispersion tweeters and speaker systems in addition to the phono cartridge under discussion at this meeting. He was formerly head of recording research at CBS Laboratories and more recently president of Micropoint, a firm that fabricates record cutting styli. His presentation included discussions of the disc-cutting process and the phono cartridge in general and his electret-transducer cartridge in particular; he concluded with a discussion of high quality test records, particularly the CBS STR-100 series.

Cartridges and the QDC-1. To begin the tutorial section of his talk, Mr. Schwartz stated that the performance that the phonograph record and the phono system must reproduce accurately is the mixed down stereo master tape used to create the disc stampers rather than the acoustic performance heard in the hall. It is somebody else's job to capture the sound accurately on the master tape. From his point of view, then, the true test of fidelity is the A-B comparison between the master tape and the phono chain. It is Mr. Schwartz's claim that his cartridge "sounds like the master tape" and, of course, therefore contributes no sound of its own. He also claimed that everything on the master tape is capable of being accurately transferred to the disc medium, although this is not always accomplished.

The tutorial continued with a discussion of the disc-cutting process. The signal from the master tape is used to modulate the velocity of the cutting stylus. Grooves are cut in a "lacquer," which is a 6-mil-thick acetate film on an aluminum baseplate. The cutting stylus, a shaped diamond point on the end of a sapphire rod, is wrapped with a heating coil to aid the cutting, but the mastering process is basically one of chiseling the acetate out of the film. The cutting stylus is pointed in the direction into the disc, but the face doing the cutting is flat. The trailing end of the stylus is wedge shaped, so the entire cutting point is rather like a short ship traveling backward in the water. At the edges of the "stern" the sharp corners are beveled to provide burnishing faces which smooth the melted acetate to a roughness of less than 0.003 mil (0.1 micrometer, 1 mil = 25 micrometers), which is an invisible optical roughness, regardless of the degree of microscope magnification used to examine the disc. (But see High Fidelity and Stereo Review, October 1968, for scanning electron microscope views of a cut disc.) The maximum groove width is 3 mils, or about half the thickness of the acetate film, so that the total dynamic range of the groove is in the order of 1000, or 60 dB of audio signal.



The very bottom of the cut groove has a radius of about 0.16 mil, which corresponds to a recorded frequency limit of about 100 kHz at the outer edge of the disc. (If the tip radius were made smaller, it would become increasingly difficult to separate the final pressing master disc from the vinyl record.) Note that this 0.16-mil radius is just small enough to allow CD-4 discs to be cut. The process of metalizing the lacquer, pressing the stamping mother from this master disc, and plating the mother to provide the metal stampers used to press the final discs was then reviewed (see Stereo Review, October 1974).

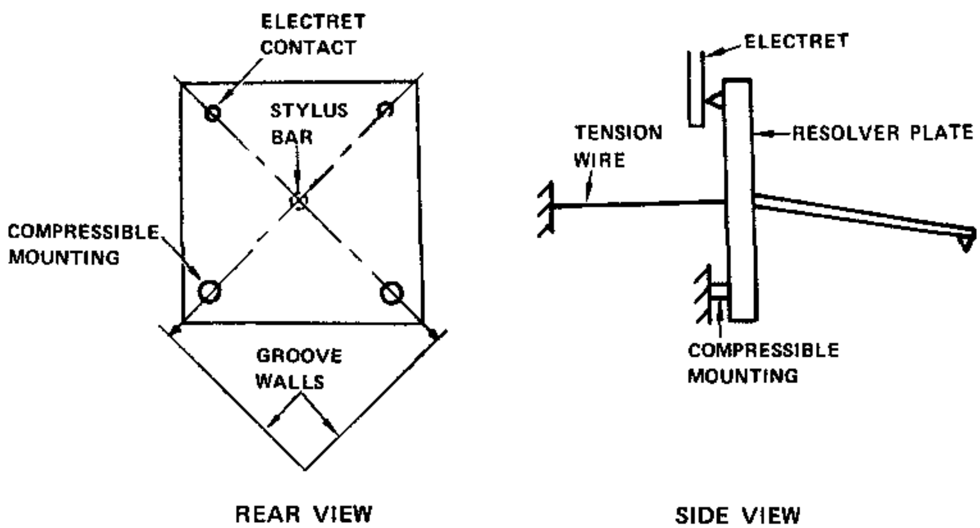
With the minimum cutting dimensions in mind, the talk proceeded to the requirements on the playback stylus to accurately reproduce the grooves cut in the disc. (See Stereo Review, January 1969, for a discussion of styli and the stereo disc.) The evolution from conical to elliptical to Shibata has arisen from the desire to trace as small a length along the groove as possible, while keeping the net contact area as large as possible by contacting the groove as high as possible along its vertical sides. The former requirement reduces tracing distortion at high frequencies, where the radius of the cut groove undulations approaches the minor diameter of the playback stylus (e.g., the 0.2-mil dimension of some ellipticals). Although the smaller the effective radius of the stylus, the lower the tracing distortion, the distortion can never be reduced to zero. In a similar vein, if the playback stylus is made more and more identical to the cutting stylus (e.g., the Shibata design), the distortion will be reduced more and more, but the playback stylus will then not only replay the information within the groove but will also recut it, thus scraping off some of the information. The tradeoffs in a small radius in the playing direction and in the shaping of the stylus to reduce distortion are further complicated by the desire to reduce record wear at the required tracking force, the second requirement above. Here keeping a large contact area is important, which is done with elliptical styli by using a large radius stylus in the direction up the side of the groove. Although the net contact area of the elliptical stylus is smaller than that of a conical of about 0.7-mil radius in both directions, the lower tracking force for the elliptical stylus keeps record wear low (see Stereo Review, October 1968). With the Shibata shape, the contact area is even larger, and the same wear is experienced at higher allowed forces. One difficulty with these non-symmetrical stylus tips is that of keeping the vertical tracing angle correct. If this is not done, the stylus tip will engage an even longer portion of the groove in the playing direction than would a conical tip. This contact with several recorded wavelengths of groove wall at the same time greatly increases distortion and reduces transient accuracy.

Mr. Schwartz related an incident at CBS Records to demonstrate that however critically the original stylus is cut and polished, its wear will introduce flats along the contact area that will increase distortion audibly before wear can be observed visually, even with a microscope. It seems that the women who listened to the finished records were discarding styli (for increased distortion) long before he could detect any signs of tip wear, and in a blind A-B test of new versus discarded styli, the selection accuracy was nearly perfect. His point was that we should not trust inspection to determine wear, but rather we should recheck distortion on a reference "testing" record often enough to detect wear at the earliest possible point.

Moving beyond the stylus to the cantilever assembly, Mr. Schwartz discussed the resonances that result from the interaction of the tip mass with the vinyl plastic (the stylus/groove resonance) and the tip resonance of the cantilever and the stylus suspension (in free space). For the combined resonance, the frequency of the peak can be moved out of the audible range by making the tip mass as low as possible, which is accomplished by using a nude diamond cemented to a hollow titanium alloy tube. Less expensive but more massive designs use a diamond only for the tip, and cement it to a metal or sapphire rod and then to the tube; this can be aluminum or steel in less expensive designs. The shape of the tube (its taper, thickness, and diameter) also affect the tip mass and thus the resonant frequency. In the QDC-1 the resonance was claimed to be 35 kHz, above which the response falls off rapidly.

In response to questions, it was admitted that some of the first production models of the QDC-1 had misplaced internal damping blocks, which lowered the net resonance to the 15-kHz range. This misplacement was blamed on the shipping process rather than on production problems. Newer versions of the QDC-1 have an externally visible, non-critically located damping block, and a serial number greater than 6000.

At the other end of the stylus supporting beam in the QDC-1 is the mechanism that translates the mechanical motion of the tip into an electrical signal. This transducer assembly consists first of a rectangular resolver plate, with the stylus attached to its center. This resolver plate rocks on four pivot pins in response to the mechanical vibration of the stylus. Two of the pivot points are elastomer buttons that fix the resolver plate to the cartridge body, but allow the top portion of the plate to push in and out, and to rock back and forth, in response to the beam motion. The two other pivot points are cone-like pads that bear against the electret elements, the actual transducers, which produce the electrical signal from the mechanical forces. The motion of the resolver plate is simply a rotation about the diagonal axes connecting opposite pairs of suspension pads. Rotation about each axis independently resolves the motion of the stylus beam into the two stereo signals, left and right channels. Because the pivot axes are precisely perpendicular to the groove walls (see the indicated position of the walls in the sketch), undulations in each of the groove walls are transferred directly to only one of the movable cone-like pivots. This provides excellent (theoretical) separation of the left and the right channels, and a minimum of mechanism between the stylus and the electrets. One final bit of mechanical contrivance, a centering wire that keeps the middle of the resolver plate fixed with respect to the cartridge body, completes the assembly. Stylus replacement involves the complete renewal of all of this mechanism, up to the electret elements themselves.



The mechanical motion of the two (independent) pivots is changed into an electrical signal by using the electret principle, which is similar to the principle of an electrostatic speaker or to the electret capsule microphones in comparison with the usual magnetic transducer operation. An electret is an electrical analog to a magnet. Where a magnet has magnetic poles and a built-in magnetic field, the electret has permanent positive and negative charges at its poles and a built-in electric field. (Although Mr. Schwartz declined to identify the electret material in his cartridge, the most popular choice currently is Teflon, which has at least a 20-year charge retention life.) To make a transducer element, the electret is usually used as the dielectric (or insulating) material in a capacitor with electrodes plated on the two opposite faces to form a permanently charged capacitor (one with a permanent dc voltage across it). As the surfaces of this capacitor are squeezed to move its plates closer together, or *as* the permanent mechanical bias pressure is relieved to allow the plates to separate, an ac voltage is developed across the capacitor which, within limits, is proportional to the displacement. (For the physicists, $V = Q/C$ and $C = k/d$, so that $V = (Q/k)d$. In the QDC-1, $d = 5$ mils at rest and Q is constant.) In the QDC-1 there are two separate metalizations on the front surface of the electret element (each 25 x 180 mils) and a single one on the backside, so that the two separate channels of mechanical stylus motion result in two separate varying electrical signals, each proportional to one direction of stylus displacement.

The phonograph record is, however, cut with the cutting stylus <u>velocity</u>, rather than displacement, proportional to the signal amplitude. In the usual magnetic cartridge, the output is proportional to the velocity, and therefore the signal can be fed directly into a preamplifier (where the RIAA curve is applied). In the QDC-1, however, a passive electrical network is required to alter the displacement-proportional output to the required velocity-proportional signal. This is built into the cartridge itself. An added advantage of the network, according to Mr. Schwartz, is that the output impedance of the cartridge and network system is totally resistive (3.9 k ohms), thus reducing phono noise of a perfect preamplifier stage by roughly the square root of (47/3.9) or a factor of 3 to 4. (But most magnetic cartridges have 600-ohm dc resistances—is this noise reduction factor really correct ?—Ed.) In addition, the cartridge is not affected by varying input impedance caused by the preamplifier feedback circuits or by large values of connecting cable capacitance. The output of the system is within the norm for magnetic cartridges (3 mV at 3.5 cm/sec), but the advantages given above obviously result from the higher efficiency of the electret transducer over its magnetic counterpart.

Final comments on the QDC-1 cartridge included the statement that its compliance is lower than average, obviating the need for ultralight tonearms as required by, for example, the XLM. The weight of the cartridge could also be much lower than that of most magnetics, owing to the lighter materials used in its construction, but some material is actually added to the unit to make it fall within the weight norms and to keep the cartridge/arm resonance about 10 Hz on the average. Tracking force with the "elliptical" stylus is about 1.5 grams, although Mr. Schwartz claimed no higher record wear with his stylus than with other ellipticals at 1 gram. A claim for superior rise time of the QDC-1 cartridge caused some argument as to why such a characteristic would not show up in the frequency response, but Mr. Schwartz held to his claim that the rise time of the QDC-1 is superior to that of almost all competing magnetic units. (It was unclear whether the claim was for faster rise time or for shorter delay time from the moment of groove deflection to the resulting output signal.) Another point raised was that in most magnetic cartridges, where the stylus beam is suspended by a ring of elastomer, nonuniformities in this material would result in deteriorated separation or inner channel balance. Also in regard to magnetic cartridges, he stated that the magnetic output is a nonlinear function of displacement, and that a push-pull arrangement is used to cancel the nonlinearity. But the electret is inherently linear in output; so as long as the two electret portions are matched, linearity and channel balance in the QDC-1 should be perfect.

Several members of the society have auditioned the QDC-1, as reported in the November issue of The Speaker, as also have the authors of The Absolute Sound, but to date the results have been mixed at best. A further report on the audible performance of the unit is promised when the results become conclusive, probably in about two months. A demonstration at the meeting was also inconclusive, with the only definite comment being the presence of harshness in the 10-kHz range.

Test Records. As former head of recording research at CBS Labs, Mr. Schwartz was a co-developer of the CBS test record series. He gave a history of test records prior to CBS's and a description of the method used in the production of the first absolute standard recording, the CBS STR-100 disc. Originally, test records such as the Cook series were checked using specially calibrated phono cartridges. These, however, were calibrated against still other test records, so there was really no original, absolute groove amplitude calibrated test record, with absolute recorded levels against which to judge a reference cartridge in the first place. CBS Labs rectified the situation by producing a calibrated test record whose levels were verified by non-playback means, specifically, through the use of optical inspection processes to measure the amplitude of the groove undulations. The CBS series also offered a new concept of test material in the inclusion of bands such as the swept frequency signal, which is more useful than the earlier discrete frequencies in finding resonances and narrow-bandwidth aberrations in the phono cartridge. The rate of the sweep was designed to coincide with the most common chart recorder, a General Radio unit, so that the presentation would be completely unambiguous. (An earlier Clarkson swept-band record was not directly compatible with any such chart recorder.) Another test signal available from early test records consisted of mixed tones useful in measuring the IM distortion of the cartridge. The CBS record was intended to expand the usefulness of this type of test signal.

The CBS test record series is especially noted for the care and accuracy involved in the cutting of the proper amplitudes in the master. These levels were measured directly on the finished disc by one of three optical methods. The first involves the measurement, with a microscope, of the peak-to-valley groove excursions of a sinusoidal signal; this measurement is possible only at the lower recorded audible frequencies. From the amplitude of this motion, the electrical output of an ideal phono cartridge can be calculated. A second measurement technique, required at the higher frequencies where the actual groove modulations were not measurable with optical microscopes, was to illuminate the recorded disc with light and to examine the pattern of reflected light on a screen. The desired pattern of light and dark spots for a given amplitude and frequency could be calculated, and the match between the actual and the theoretical results compared. (This is similar to interferometric methods used to measure optical lens aberrations too small to measure directly with physical probes or with an optical microscope.) The presence of distortion products on the disc can also be directly detected by this method by examining the sharpness of the light pattern. A third method of obtaining accurate amplitude undulations as a function of frequency is to use one of the above two methods for one reference frequency, to calibrate the output of a cartridge at this reference point, and then to vary the speed of rotation of the test disc to vary the apparent recorded frequency as the varying output from the cartridge under test is measured.

In the production of the CBS STR-100 disc, the mastering was done at night when building vibrations and line voltage fluctuations were minimum, and the signal generation process was automated so as to reduce the variation from disc to disc over several recording sessions, which operator intervention could cause. The separate test bands on the resulting CBS STR-100 disc contained the following signals: (1) swept frequency, (2) lateral and vertical tracking tests, using amplitudes corresponding to +6 to +18 dB laterally and +6 to +12 dB vertically above the reference 5 cm/sec (0 VU) peak recorded velocity (vertical amplitude is limited to +12 dB by the 6-mil thickness of the acetate coating on the lacquer, (3) tonearm resonance test, with a lower recorded limit of 10 Hz, (4) IM distortion, and (5) transient response, checked with a square wave (which is cut as a triangle wave owing to the velocity-proportional, or differentiating, nature of the usual magnetic cartridge).

Another test record, the STR-111, added discrete vertical and lateral modulations to provide independent measurement of the vertical and lateral compliance of the cartridge. (The Shure ERA-III disc does not specify the direction of the cut.) Additional bands sweep an extended low frequency, subaudio range for arm resonance tests, and also include frequency response to 50 kHz, and add additional square-wave test signals.

Under questioning, Mr. Schwartz claimed that the ringing found by testing facilities using the record (High Fidelity, Stereo Review) are not in the disc but are evidence of resonances in the stylus, which can increase further as the damping material supporting the stylus beam begins to age. In the QDC-1 this ringing can also be seen as peaks (evidence of a resonance) at 18 kHz, where the beam/resolving-plate elasticity and the stylus-mass resonance is excited, and at 35 kHz, where the stylus tip and the elasticity of the groove wall has a resonant peak. Mr. Schwartz concluded his presentation with the statement that all measurements are still only attempts to predict what the human ear will hear, but that the ear is still the final element in what sound is perceived.

As for the availability of the CBS test records, CBS Records took on the distribution chores some years ago and has not made the discs widely available (and has possibly let the quality slip). It is rumored that CBS Labs will again take on the distribution task at an early date, and perhaps the discs will once more become a reliable and widely available tool for cartridge testing. Letters from those interested in this action just may help in speeding up this process.—Joel Cohen, Keith North, John Schlafer

THERMO ELECTRON
CORPORATION

85 FIRST AVENUE
WALTHAM. MASS 02154
TEL 617 890 8700
CABLE TEECORP

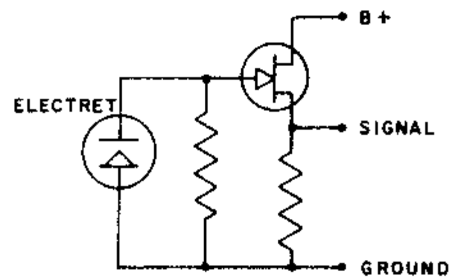
SPECIFICATIONS - ELECTRET CONDENSER MICROPHONE MODEL 814C

Small microphone featuring smooth extended (infrasonic and ultrasonic) frequency response, wide dynamic range highly resistant to shock and vibration. Recommended for sound measurement and professional audio applications. Includes a self-contained high voltage electret and hybrid integrated circuit.

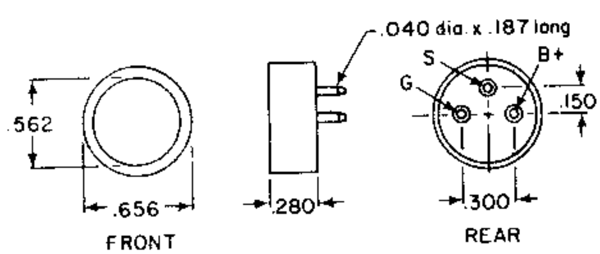
PATENTED: Canada, #832, 651, #882, 386; Great Britain, #1, 230, 402, #1, 230, 403, and #1, 230, 404; USA #3, 612, 778 and other patents pending.

SPECIFICATIONS - 814C

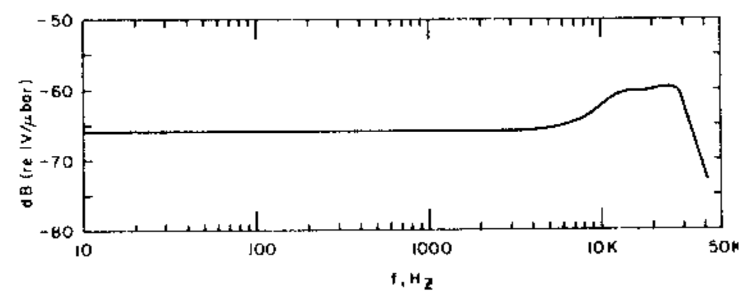
Directional Response	Omnidirectional
Frequency Response	10-35, 000 Hz
Sensitivity	-66dBV (Re: 1V/μbar)
Noise Level (A Weighted)	24dB SPL
Distortion (10% THD)	145dB SPL
Vibration Susceptibility	80dB SPL/g
Hum Susceptibility	10^{-7} V/gauss
Supply Voltage	8. 0-30. 0 volts
Current Drain	1 ma maximum
Output Connections	3 connector pins



MICROPHONE SCHEMATIC



DIMENSIONS IN INCHES



FREQUENCY RESPONSE
(OUTPUT MEASURED OPEN CIRCUIT AT 8.0 VOLTS SUPPLY)

For further information, contact: Dielectric Product Operations
Thermo Electron Corporation
101 First Avenue, Waltham, MA 02154
Telephone: (617) 890-8700, Ext. 251
Cable: TEECORP

TEM-S-4P (8/72)

The Role of Damping in Tonearm/Cartridge Performance

S. L. Phoenix*

INTRODUCTION

Last spring I decided to replace my AR tonearm/Shure V-15 Type II-improved cartridge combination, having it on good authority that an ADC-XLM cartridge mounted in one of a handful of "suitable" tonearms would give far better performance. The task seemed simple enough—ask the "right" people what performance characteristics a suitable tonearm should have, buy one, assuming it was in the affordable price range, and mount it on my AR turntable. The proper tonearm characteristics, it was said, were low vertical and horizontal bearing friction, low effective tonearm mass, precise antiskating compensation, and low tracking error, along with a few other things. It then occurred to me that I would save money if I could modify my AR tonearm to satisfy the specified performance requirements. Several shop-hours later I had substantially reduced the effective tonearm mass, rewired the arm with extremely fine wire, developed a lateral bearing with extremely low friction, extensively modified the vertical bearing system (again lowering the friction), and developed an antiskating device similar to that on the Shure SME arm. With anticipation I finally mounted a new ADC-XLM cartridge in the modified arm, only to be very disappointed in the results. To be sure, the resonant frequency was a respectable 7 Hz, but the tonearm literally wallowed along, even on recordings with negligible warp, so that a 7-Hz signal permeated everything.

Having some background in vibration theory, I decided to do an analysis to see if I could determine a better way of improving the performance. The predictions of the completed analysis were startling and I used the results to markedly improve the performance of my modified AR tonearm; indeed, I have heard no arm that betters its present performance. Warped records that had long been retired to storage because of their unplayability were now tracked with ease. Some of the findings that resulted in this performance are reported in this article.

SIMPLE ANALYSIS

Let us consider the function of the tonearm. When the tonearm/cartridge system is in play, we wish to have the stylus tip follow the record groove and displace relative to the cartridge, generating an electrical signal. Meanwhile, the cartridge shell and tonearm should remain a fixed distance from the local mean record surface, riding the warps and other inaudible low-frequency excitation as a cork might ride the waves on an ocean. Whether or not these objectives are achieved in any system is a function of the dynamics of the tonearm/cartridge mechanical system, specifically, its response as a function of frequency.

*Assistant Professor in the Sibley School of Mechanical and Aerospace Engineering, of the College of Engineering, Cornell University.

Let us consider a conventional tonearm/cartridge system. We will analyze in detail only vertical motion, though most of what is said holds also for lateral tonearm motion. We assume first that the effective tonearm/cartridge mass is of value m (the precise meaning of which is not important to the discussion) and that the compliance is 1/k (the usual manufacturer-specified cartridge compliance). The natural system resonant frequency (f_n) in Hz is then related to these values by

$$f_n = \frac{1}{2\pi}\sqrt{\frac{k}{m}}$$

There are two additional physical quantities which don't get mentioned often in discussing phono systems. We will call them ζ_s, the damping factor of the stylus assembly (in system), and ζ_p, the tonearm pivotal damping factor (in system). It turns out that f_n, ζ_s, and ζ_p tell the lion's share of the tale about how the tonearm/cartridge system performs. In analyzing the dynamics of the system in terms of these parameters, we imagine that a sinusoidal groove of frequency f and amplitude Y is cut on the record and we consider the sinusoidal displacement amplitude Z of the stylus <u>tip relative to the cartridge</u>, or better still, the amplitude <u>ratio</u> $|Z/Y|$. We would like $|Z/Y|$ to be <u>equal to one</u> for <u>f above 20 Hz</u> (or so) and would like $|Z/Y|$ to be zero for <u>f below 20 Hz</u>. If these criteria are met, then the function of the tonearm/cartridge system mentioned earlier, i.e., for the arm to follow warps and for the stylus to follow the audio information, is perfectly satisfied.

Well let's see what really happens—and the best way is to look at two graphs. In Fig. 1 (The expressions for these plots are given in the Appendix—Ed.) we consider the case where $\zeta_p = 0$, that is, where the damping in the tonearm vertical pivot is virtually zero—a situation that is common in many precision tonearms. Now notice the role played by ζ_s and f_n. For $f \geq 3f_n$, in all curves we see that $|Z/Y| \approx 1$, so that if f_n is, say, 10 Hz, then for frequencies above 30 Hz we get the response we want. Also for $f < 0.80 f_n$ or so, we get a rapid drop in $|Z/Y|$ with decreasing excitation frequency f, which is also what we desire. But at $f \approx f_n$, we see that the behavior is more complex. We can get anything from no peak to a gigantic peak in $|Z/Y|$,

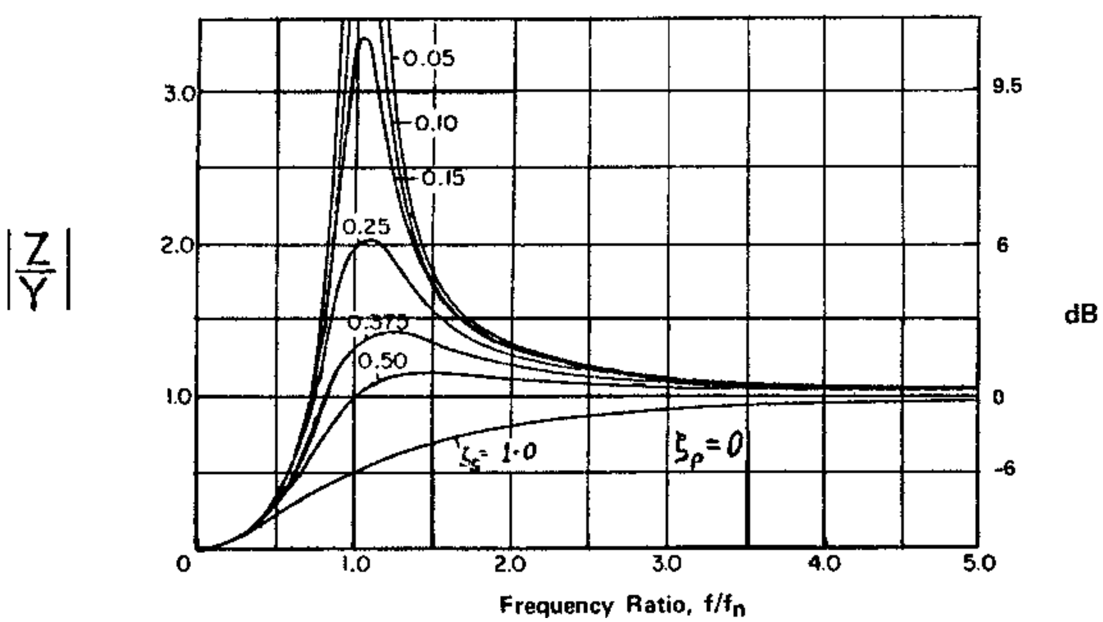


Fig. 1. Plot of frequency response $|Z/Y|$ for various values of stylus damping factor ζ_s

a

depending on the value of the stylus assembly damping factor, ζ_s. One can easily be convinced that the best value for ζ_s is 0.5, because |Z/Y| is "flat" down to $f = f_n$, below which |Z/Y| rapidly rolls off. On the other hand, a value of say $\zeta_s = 0.15$ will give a peak value of |Z/Y| 3.3 at $f = f_n$ and the response will be up approximately 10 dB at $f = f_n$. In this case, if the disc groove or disc warp has frequency components in the neighborhood of f_n, very large displacements will result. Such large displacements will yield audible intermodulation distortion, large variations in tracking force, undesirable amplifier loads, and large bass speaker cone excursions at the natural resonant frequency fn. It is clear that specification of fn alone is not enough to characterize the system, but that the amount of damping is also critical.

But what value of ζ_s is typical for current high-performance tonearm/cartridge systems? Most high performance cartridges employ a rubber-like viscoelastic ring through which the stylus cantilever passes; in addition, some form of metallic spring is sometimes employed. The ring must be viscoelastic in nature, but it must also be stable enough to resist creep in extended use. Therefore some degree of molecular crosslinking (polymerization) is necessary, and this limits the amount of damping possible. We can estimate ζ_s pretty well given the viscoelastic properties of the current stable polymeric materials used in this application. (Surprisingly enough, specific stylus geometry has little effect on the value of ζ_s.) Without going into details, I will simply state that if the natural frequency f_n is about 10 Hz and only a rubber-like ring generates the compliance 1/k, then $\zeta_s \approx 0.12$; if a metallic spring is used, the value could be lower. A value of $\zeta_s \approx 0.2$ would, in my view, be a technological upper bound if any long-term stylus stability is to be expected. As it is, the viscoelastic ring dries out and hardens with age, thus decreasing ζ_s and increasing 1/k and consequently f_n with time. One sees in Fig. 1 that our typical value of $\zeta_s = 0.12$ generates a peak value in |Z/Y| of roughly 4, resulting in a 12-dB peak at $f = f_n$.

What this means is that we cannot rely on the stylus assembly alone to adequately damp the system. We point out also that reducing tonearm mass does not radically improve the situation. A <u>fourfold</u> reduction of tonearm/cartridge effective mass m only <u>doubles</u> f_n, which in turn, by the properties of the rubber-like ring, increases ζ_s by approximately 40%. And at the same time f_n may now be in the audible region, yielding an undesirable 5- to 10-dB "hump" in the frequency response near f_n.

Let us now see what the effects of tonearm pivot damping are, and again we look at a graph. In Fig. 2 we consider the case now where $\zeta_s = 0$ and we look at the effects of varying ζ_p —the tonearm pivot damping factor. (We have already pointed out that for conventional high-compliance cartridges, ζ_s is a rather low 0.12, so that the assumption that $\zeta_s = 0$ is not unreasonable.) We see in Fig. 2 that ζ_p has a marked effect on the response amplitude |Z/Y|. The behavior is similar in character to that of Fig. 1 for $f \geq f_n$, except that here a value of $\zeta_p \approx 1.0$ is necessary to get anything near a "flat response" down to f_n. But the effects of ζ_p on the rolloff behavior of | Z/Y| differ markedly from those of ζ_s in the region where the excitation frequency f is less than the resonant frequency f_n. In fact the frequency response can be both <u>flattened</u> and <u>extended downward</u> by increasing C_p. A desirable value for ζ_p is clearly $\zeta_p \approx 1.0$, where the frequency response is reasonably flat yet a reasonable rolloff below f_n occurs. The behavior for small values of ζ_p (very large peaks at $f = f_n$) is very similar to that for small values of ζ_s shown in Fig. 1.

It is clear that some pivot damping is necessary to reduce the height of the frequency response peak at $f = f_n$. But too high a value of ζ_p causes susceptibility to warp frequencies well below f_n, especially if f_n is low (say 5 Hz) to begin with, as it would be with a high-compliance cartridge in a massive arm. On the other hand, if a light tonearm is used with a relatively low-compliance cartridge, then f_n will be high to begin with, and high values of ζ_p could be used to advantage.

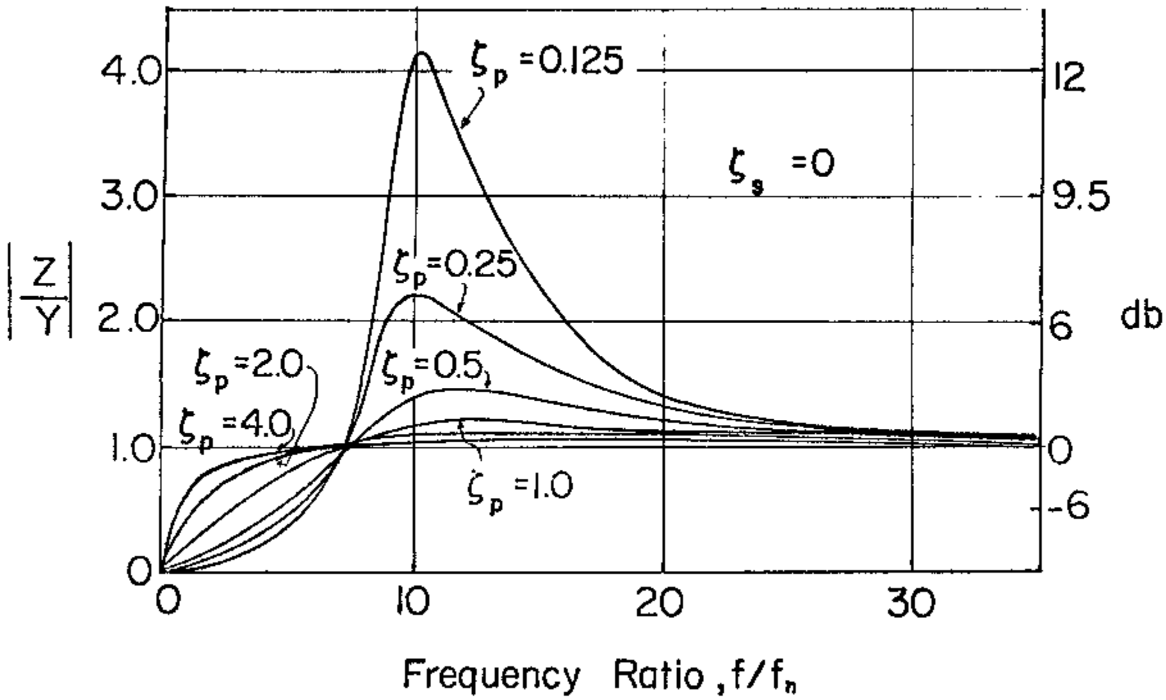


Fig. 2. Plot of frequency response |Z/Y| for various values of tonearm pivot damping factor ζ_p

But what value of the pivotal damping factor ζ_p is typical for conventional tonearm/cartridge systems? We can estimate this by imagining that the cartridge plus arm is moving vertically in a sinusoidal fashion at the system resonant frequency of 10 Hz and with an amplitude of 10^{-2} cm (at the cartridge), a motion which is probably not visible to the naked eye. (This motion would be generated by a periodic warp at, say, 10 Hz.) Then we can show that the effective pivot damping force as measured at the stylus is approximately $F_d = 2\zeta_p kX$, where 1/k is the stylus compliance and X is the cartridge vertical amplitude. If 1/k is 25×10^{-6} cm/dyne, a value typical for high-compliance cartridges, then $F_d = 800\zeta_p$ dynes or $800\zeta_p$ mg force. But pivot friction, which acts much like damping, is frequently quoted in test reports as being 20 mg or so, and the damping force (which is not measured) is probably of the order of 40 mg at $X = 10^{-2}$ cm [follow on through with this assumed value—Leigh shows it to be reasonable—Ed.], so that an effective value for F_d for many tonearms is likely to be around 60 mg for $X = 10^{-2}$ cm. Hence, from the above equation, a value of $\zeta_p \approx 0.075$ seems reasonable (though this value will depend on X because of the nonlinear behavior of friction and pivot damping).

Finally, to find the peak height when both ζ_p and ζ_s act and are small, we can show that we may simply add ζ_p and ζ_s to get an effective value $\zeta_e = \zeta_s + \zeta_p$, and use this value in Fig. 1. For our case, $\zeta_p \approx 0.075$ and $\zeta_s \approx 0.12$, so $\zeta_e = 0.195$ and the peak |Z/Y| is roughly 2.5 or about 8 dB. This is in agreement with the tonearm report figures in High Fidelity magazine. The main point here is that ζ_p is far lower than it should be, and in fact it may be absurd to reduce the pivot friction to miniscule proportions because this indeed will make the effective value of ζ_p lower and the peak higher. Indeed, for the example tonearm vibration amplitude of $X= 10^{-2}$ cm and for the desirable value of $\zeta_p = 1.0$, $F_d \approx 800\zeta_p = 800$ mg, a value which is far greater than the friction value of 20 mg quoted in magazine reports.

a

It is clear that an adjustable pivot damping device, to be set depending upon the cartridge chosen, would be highly beneficial on most tonearms. Indeed it should act both vertically and horizontally. Pivot damping also would help control other external sources of vibration such as that coming from the turntable foundation.

Our analysis and discussion suggest the following conclusions:

1. Specification of the damping factors ζ_s and ζ_p is far more relevant to the tonearm/cartridge system performance than specification of the system natural frequency fn.
2. It is not possible to specify an optimal value for f_n without knowing the magnitudes of ζ_s and ζ_p as well.
3. The stylus assembly damping factor ζ_s for virtually all high-performance tonearm/cartridge systems is likely to be far below that necessary for optimal damping, because of material property limitations.
4. Pivotal damping is highly beneficial yet highly inadequate (in fact totally lacking) on most tonearms.
5. Reduction of the tonearm vertical and lateral friction to miniscule proportions is a waste of effort and indeed can be detrimental to performance. Instead, using a highly viscous bearing lubricant would have the beneficial advantage of both eliminating this dry Coulomb friction and generating viscous damping along with a more acceptable form of friction (that with lubricated surfaces).
6. There is no benefit in increasing f_n above 20 Hz unless a large value for ζ_p is used. In fact it is highly detrimental to do so in virtually all instances.

ADDITIONAL THOUGHTS

We have been speaking up to now of conventional tonearms, all of which are proven to be inadequate, and one might properly ask how this discussion reflects on the behavior of some very recent (revolutionary?) designs.

Let us consider first the new Transcriptors Vestigal arm. The designers of this arm are rather fortunate that the arm does not perform as they claim it does, i.e., that it does not have a natural frequency f_n in the region of 200 Hz or so. For one thing, that would require about a 500-fold reduction in effective tonearm mass, and this plainly is just not possible; moving the pivot point close to the cartridge does not yield anything like this decrease in effective tonearm mass. Also, by Fig. 1, a natural frequency of $f_n \approx 200$ Hz or so would be disastrous. The arm may, in fact, have a natural frequency (when fitted with an ADC-XLM) of more like 15 Hz, but its pivot damping situation is open to investigation. To be fair, it may perform quite well—but not for the reasons the manufacturer claims.

There is also interest in the new Dual 701 turntable. It appears to this writer that the Dual arm uses what amounts to a tuned damper, that is, the arm is in two sections separated by a viscoelastic element which is supposedly optimally chosen. Unfortunately, the system performance depends on the cartridge choice (where are the adjustments ?). And secondly, one high peak at f_n gets replaced by two broader but lower peaks which tend to straddle the original peak. One may still wind up with some low-frequency resonant susceptibility. So, as in the case of Fig. 2 where ζ_p can be increased too much, we cure one evil and create another. Nevertheless, the arm may perform very well—but at what cost !

For those technically inclined, I have found the automobile engine lubricant STP to be a very successful lubricant in both the vertical and horizontal bearings of my modified AR tonearm. For one thing, the film strength is high enough to deal with the dry (Coulomb) friction that may cause the arm to stick. My vertical bearings are set pretty tight because this seems to give the best performance in conjunction with the STP lubrication. (The vertical "butterfingers" damping device in the AR arm has, of course, been disconnected.)

APPENDIX

As mentioned in the newsletter introduction to Leigh's article, this publication is an extreme condensation of a mathematical treatment we received several months ago. For those of you who would like a bit more of the theory and mechanical analysis, we present this short appendix taken from the original paper.

In this paper, the effective mass at the stylus tip is defined as m and the mechanical stiffness of the stylus assembly is k, or the stylus compliance is 1/k, as in the BAS publication. The damping parameters are C_1 for the stylus and C_2 for the arm, as represented by "dashpots" in the schematic for the mechanical system. Leigh worked out the system response function, $H(\omega)$, as a function of frequency, $\omega = 2\pi f$, for two forms of excitation. Both of these were sinusoidal with time, but one was for the stylus forced with time (as for a warped disc or for the groove modulations of a normal disc) and one was for the arm pivot forced with time (as would be the case for acoustic feedback). For the former excitation, the amplitude of the response $|H(\omega)|$ is given by

$$H^2(\omega) = \frac{(\omega/\omega_n)^4 + (2\zeta_2\omega/\omega_n)^2}{[1 - (\omega/\omega_n)^2]^2 + [2(\zeta_1 + \zeta_2)\omega/\omega_n]^2}$$

where $\omega_n = \sqrt{k/m}$, $C_C - 2\sqrt{km}$, $\zeta_1 = C_1/C_C$, and $\zeta_2 = C_2/C_C$.

This expression, with $C_2 = 0$, is plotted in Fig. 1, while the function for $C_1 = 0$ is plotted in Fig. 2. You can investigate the range of the approximation that ζ_1 and ζ_2 can be added to give the curve of Fig. 1 at your leisure.

For the second form of (system) excitation, replace the numerator in the equation given above with the factor $(F_0/k)^2$, where F_0 is the force of the excitation. Note that the effect of damping at low frequencies is similar to the damping with ζ_1 and ζ_2 discussed for stylus excitation in the publication. No plot of this function is given in this appendix, although one appears in the original paper. It can easily be plotted from the denominator of the equation given above.

As a final point of detail, we give the following expression for the effective damping force caused by pivot damping:

$$|F_d| = 2\zeta_2\sqrt{km}\,\omega X$$

In Leigh's shortened publication, where he finds the value of ζ_2 for most common arms, he sets the excitation frequency ω equal to the resonant frequency, $\omega_n = k/m$, to give his simplified expression $F = 2\zeta_2 kX$ used to verify that ζ_2 is small, with the typical value of 0.075.

White or Pink: Adding a Little Noise to Your Life

How To Build a Noise Generator

Rene Jaeger

Most hi-fi systems are best suited to music reproduction; nevertheless, many audiophiles spend hours listening to noise through them. I suspect the rationale here is akin to the search for the elusive quark; but noise is a useful tool for measuring the performance of audio components.

Noise, as a wideband signal, can be used for comparing preamplifiers, power amps, and loudspeakers. And because of its random frequency distribution, noise may be used to test loud-speakers in non-anechoic situations like livingrooms. Hiss-like random noise won't excite the standing waves which would color interpretation of measurements made using sine wave signals.

But this article isn't about test methods, it's about noise and the construction of a high-quality generator of either pink or white noise.

Noise's Source

The most common source of electrical noise is random movement of electrons. In the world of electrical currents, the electron is a veritable planet to its nucleus, as determined and dependable in its orbit as the earth's movement about its sun. But given the universe of atoms and electrons contained in a single resistor, we find (in our accelerated time scale) a fury of random electron movement such as to produce energy at every frequency we can measure. This random energy is called thermal noise and it follows a frequency distribution law which says that if we know the value of a resistor and the bandwidth over which we wish to determine noise density or power, we can use the formula

$$E_n^2 = 1.65 \times 10^{-20} \times \text{bandwidth in Hertz} \times \text{resistance in ohms} \quad (1)$$

or

$$E_n = \sqrt{(1.65 \times 10^{-20} \times \text{bandwidth} \times \text{resistance.})} \quad (2)$$

We can see that given a bandwidth broad enough, say that of light (10^{15} Hz), we will have electrons fairly leaping off the ends of the resistor.

Rearrangement and inspection of this formula shows that noise density is the same for a given bandwidth in every part of the frequency spectrum:

$$E_n^2/\text{bandwidth} = 1.65 \times 10^{-20} \times \text{resistance.} \quad (3)$$

Thus, the 20-Hz-wide band from 20 to 40 Hz will normally have the same noise power as the band from 20,020 to 20,040 Hz. Restated, the octave from 10 to 20 Hz has 27 dB more noise power than the octave from 20 to 40 Hz. When played through a music system, this sort of naturally occurring noise sounds bright, tipped toward the high-frequency end of the audible spectrum. This is what's known as white noise.

The FM tuner, sometimes used to listen to music, also is a good noise generator, and in some circles is used for little else. Over the audio spectrum, it generates noise that is essentially white; however, the noise density above 2 kHz is rolled off by the tuner's deemphasis network, giving a spectrum weighted toward the mid-frequency range. We might call this noise above 2 kHz red noise because it has more energy per unit of bandwidth at lower frequencies than at higher—not too useful for testing equipment for its relative flatness, say.

What we are seeking is pink noise—noise with a spectral density that is constant for any percentage bandwidth regardless of what part of the spectrum is examined. Using this definition, an octave of pink noise from 10 to 20 kHz has the same power as an octave from 20 to 40 Hz.

Building a Noise Generator

I don't know of any naturally occurring pink noise sources, but we can make white noise pink by passing it through a filter whose response falls off at 3 dB per octave. A simple RC network gives a rolloff of 6 dB per octave (Fig. 1). By "spoiling" this network , we obtain a shelving response with an approximate slope of 3 dB per octave between f_c and about $3f_c$. Several networks may be cascaded to give a close approximation to the 3-dB slope, with the tolerance decided by the spacing (and tolerance) of the resistors and capacitors.

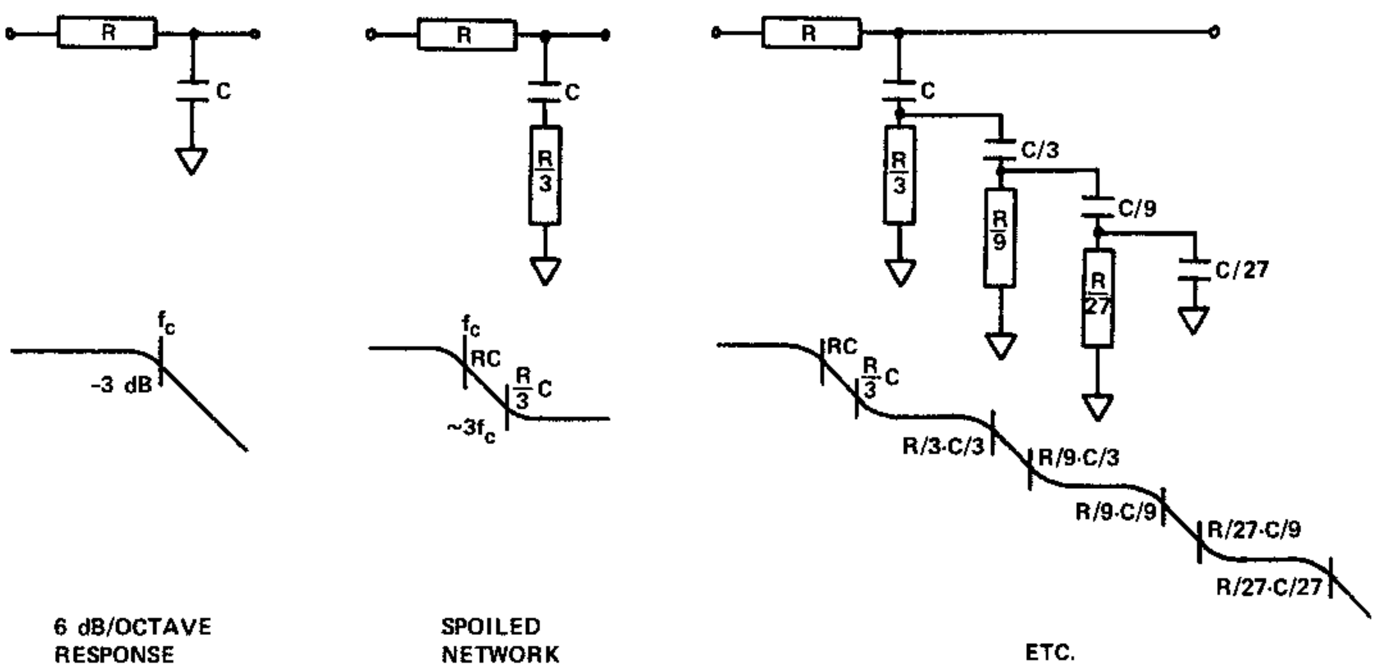


Figure 1

Fig. 2 is a schematic of the noise generator and "pinking" filter. A few notes are in order:

- The noise generator and amplifier must be well shielded to avoid pickup of unwanted noise. In the generator, we are dealing with an input impedance of 1 megohm and a gain of about 60 dB.

- Care must be taken to isolate the FET gate from the amplifier output and to minimize gate-to-ground capacitance. Lead length on the specified 1/4-watt Allen-Bradley resistor and between the FET and resistor must be as short as possible. For isolation, I recommend a two-compartment box with a tight cover; this might be built of brass shim stock soldered together at the seams, with the resistor/FET pair in one compartment and the 301 amplifier in the other.

- The stage after the FET must be a 301-type IC; a 741-type does not have enough high-frequency gain for this application. Note that the noise voltage generated by the resistor over a 20-kHz band (found using eq. 2) is about 17.5 microvolts. This noise is amplified by a factor of 10 by the FET, and 100-fold by the 301. A 741 just won't do the job. The 741 shown in the schematic is in a unity gain stage.

- Do not try to put more gain in the box with the generator. If higher levels are needed, add an external 10x amplifier stage (see Peter W. Mitchell, "IC Op Amps The Audiophile's Friend," BAS Speaker, September 1974).

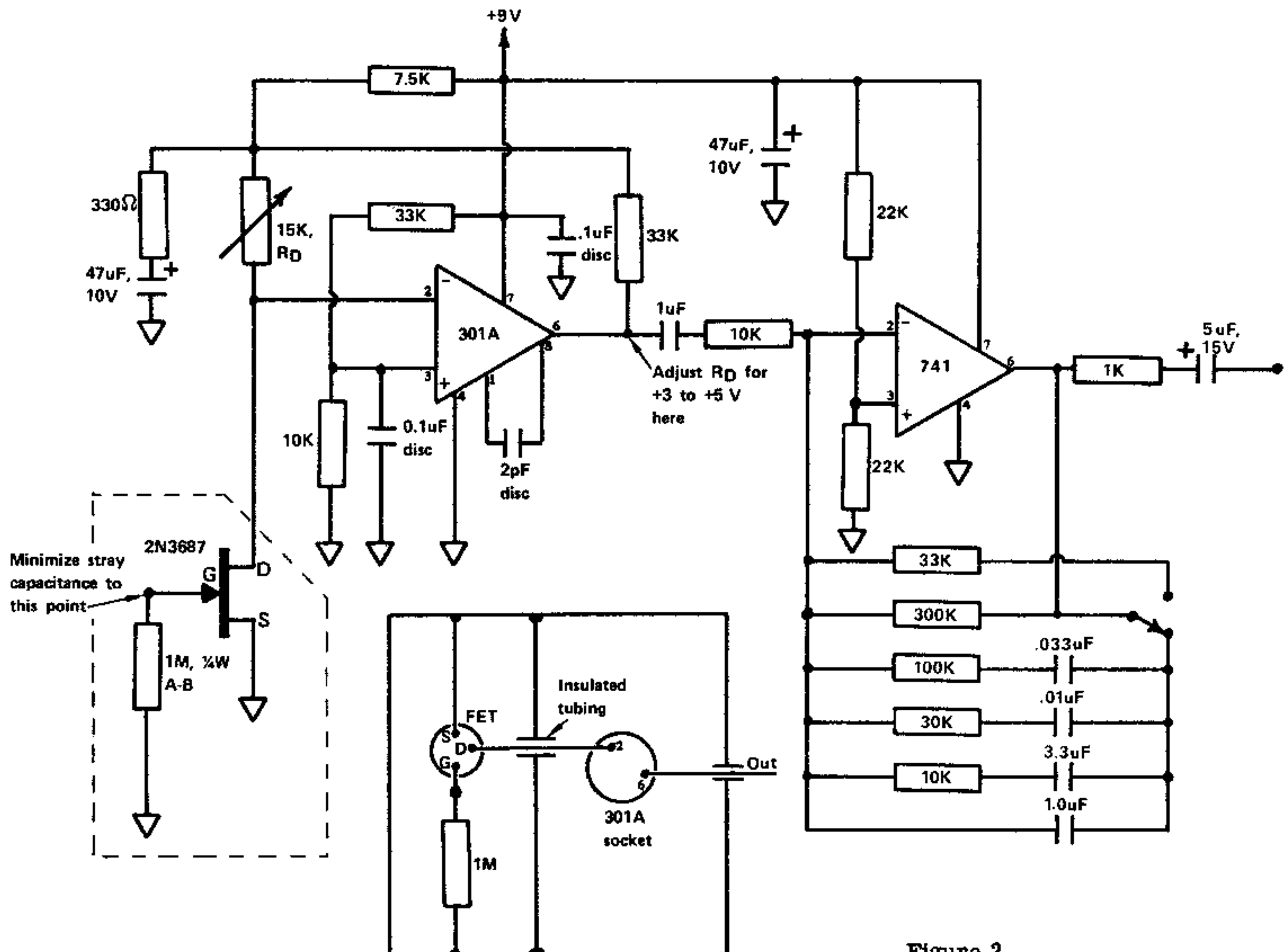
- Do not try to obtain outputs greater than 300 millivolts rms using a single 9-volt supply—you must leave headroom for naturally occurring peaks which will rise more than 20 dB above the rms level of the noise.

- The pinking filter is straightforward. It may be adjacent to the noise amp, since its gain is unity at high frequencies where feedback could be a problem.

- Use of 1 percent tolerance resistors and 2.5 percent tolerance capacitors will give a slope conformance of about ±0.5 dB. Five percent parts will give a conformance of about ±1.0 dB. The second amp stage can be either a 301 or 741—as can be the outboard 20-dB gain stage if you elect to build one.

- Power drain is about 5 milliamperes at 9 volts.

- Finally, be sure to look at the output of the generator on an oscilloscope. If by chance there is unwanted oscillation present, the time to find out about it is before you connect the unit to your hi-fi system.



+9V
7.5K
330Ω
47uF, 10V
15K, RD
33K
.1uF disc
33K
47uF, 10V
22K
301A
1uF
10K
741
0.1uF disc
10K
2pF disc
Adjust RD for +3 to +5 V here
22K
1K
5uF, 15V
2N3687
Minimize stray capacitance to this point
G
D
S
1M, ¼W A-B
33K
300K
100K
.033uF
30K
.01uF
10K
3.3uF
1.0uF
Insulated tubing
FET
S
D
G
1M
2
6
301A socket
Out

Figure 2

STYLUS OVERHANG TEMPLATE

The odd looking shape on this page is not the latest tonearm design from Gem Hi-Fi, but is rather a genuine stylus overhang template. This template is a standard design, and the procedure for its use is the same for any length of arm and any platter. Care was taken in the reproduction to ensure that the exact dimensions of the KMAL (Keith Monks Audio Limited) template were maintained.

Glue the template to light cardboard and then cut along the dotted lines. Be sure to precisely round out the hole for the platter's spindle indicated by the +. Place the hole around the spindle and lay the template flat on the turntable. The stylus tip should now be placed on the center of the + marked "stylus." The cartridge body or tonearm shell should now be parallel with the lines on the template when viewed from above. If not, adjust your tonearm overhang until this condition is obtained.

A stylus with improper overhang will have a tangent error greater than 1 degree. This tangent error will cause more rapid wear of the side walls of the record and will produce tracing distortion, particularly in the high frequencies. The detrimental effects of improper stylus overhang increase toward the center of the record, and this often accounts for the increased distortion one hears in the inner grooves.—Alvin Foster