THE B.A.S. SPEAKER

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In This Issue

This is a usual, run-of-the-mill, jam-packed issue of The BAS Speaker. The feature articles include another contribution from Dan Shanefield, this time on the audibility of phase shift in high-fidelity systems, especially loudspeakers—a hot topic of late, as a number of firms now are touting phase-coherent speaker systems. Shanefield helps us decide whether they are worth the shouting.

Speaker designer Roy Cizek explores a topic heretofore somewhat devalued—the size of speaker wire needed for best combined amplifier-speaker performance. This topic has generally been overlooked since some observers decided merely that larger gauge was better, but Roy not only has done the math to show why heavier wire is better, but notes deleterious effects of small gauge wire on other aspects than (the normally referred to) damping factor. You will be surprised at the number of parameters a simple thing like choice of speaker-wire gauge can affect. You may only think you own a 400-watt amplifier.. .

This month marks the first in a hoped for series of "Sonnets (bad pun) from the Japanese" so to speak, beginning with descriptions of some forthcoming Japanese products and a discourse on one of the better Japanese high-fidelity magazines.

There are designs for two "turn-on/off transient eliminators" and information (not good) on the Atlanta low-TIM power amplifier project. Tonearm damping surfaces again as a BAS member takes a bite out of (adoptive fellow member) "Jaws" Rabinow, and the Audio International fiasco seems close to being resolved.

And there's more. Read on.. .

Membership dues are $12 per year (October 1 to September 30) or portion thereof. Dues include a one-year subscription to the BAS Speaker. (Note that almost the full amount of dues is allocated to production of the Speaker. The local activities of the BAS are strictly self-supporting.) For further information and application form, write to: The Boston Audio Society, P.O. Box 7, Kenmore Square Station, Boston, Mass. 02215.

Stolen

While visiting Boston in June BAS member Desmond Fretz' suitcase was stolen with his new Audio Pulse Model One inside. Beware of buying a used unit cheap; it may be his stolen one. The serial number of the missing delay line is 01054.

For Sale

- KLH-9's, 6 panels (3 pairs). In mint condition, recently rebuilt with individually selected and matched tweeters. $1500/pair or best offer. Prefer to sell all three pairs to one buyer. Leave message for Dick Goldwater (617) 492-1364.
- Dynaco 400, excellent condition, $285. (617) 749-2219.
- Mint (brand new) latest Mark Levinson preamplifier (JC-2) with ML white ash case. (919) 449-4132.
- Revox A77, Mark IV, 1/2-track, high-speed deck, 30 hours use, $800. Gately SM6A mixer, 6 in, 2 out, like new, $350. Pair AKG C451 mikes with N46E power supply, used five times, $500. Desmond Fretz (215) 822-1226, after 6 p.m.

Wanted

- Dynaco PAS-3X preamplifier or other decent-sounding, modestly priced replacement for my PAT-4. David Temple, 300 South Street, Medfield, Mass. 02052. (617) 359-2915.
- Advent 100A Dolby B, any condition, price negotiable. Also dbx 122. Call Ken, (617) 646-3427.

Help Wanted (Non-Human)

I am trying to get a copy of George Frow's book A Guide to Edison Cylinder Phonographs, published by Anthony of St. Austelle, Britain, in 1970 and due for reprint soon. I've been unable to locate it even at the Edison Historic Site in New Jersey. If anyone knows where I can get a copy, please send me the source and price (if known). — David Weinberg (Maryland)

Speaker Mailing

The following members have decided to rotate the responsibilities of labeling, stamping, and bagging the newsletter for a particular month. If you are interested in assisting, please contact the appropriate distribution manager.

Month	Name	Community	Home Phone
July	John Ouzts	Brookline	738-1035
September	Jim Topali	Watertown	924-4944
October	Carl Covell	Somerville	776-3894

August Speaker

Due to vacations and the lack of people to fill in, the August Speaker will be combined with the September issue. This dual issue will be approximately double size.

More on Audio International

As we noted last month, there are two Audio Internationals in Germany. A highly placed source at Audio tells us that because of German federal legal pressure, the questionable Audio International (Berlin) has been forced to write all its customers offering them either a full refund or eventual delivery (in one or two months). This latter option is seen as a bit shaky since A.I. has yet to stock the promised Revoxes. This seems to indicate that A.I. was indeed an attempted fraud, but hopefully the authorities have the situation under control.

Boston member Ken Deen, who originally reported on this, has received one of the letters and has requested the refund. We'll let you know if he gets it. — Joyce Brinton (Massachusetts)

Low-TIM Amplifier Kit Delayed

Damon Hill, of Atlanta's Audio Forum, writes to let us know that Electronics One is no longer selling the circuit board for the low-TIM amplifier construction project that appeared in the February issue of Audio. This means his attempts to put together a kit of parts will be delayed unless he can find someone to make the boards.

Correction on Sony Cable Capacitance

Unless Sony has changed since I bought my PS-1800A turntable and tonearm, there is some incorrect data in The BAS Speaker listing indicating that all Sony's have 210 pF lead capacitance. I used a capacitance bridge from both ends of the tonearm cables (with cartridges disconnected, of course) and measured 133 pF left channel and 143 pF right channel—just some more data to cloud the issue. — David Weinberg (Maryland)

PCM Again

Regarding the PCM article in Audio Amateur (3/75) mentioned on page 10 of the March 1976 issue of The BAS Speaker, I believe your correspondent to be seriously in error about the background noise on the PCM-recorded Odyssey disc Y33200. The audible noise is not due to the recorder, but is studio noise (air-conditioning rumble, etc.). Indeed, I believe that I can even hear some traffic noise in the background. How many records have low enough tape noise for this to be possible? — Stanley Lipshitz (Ontario)

Other Audio Societies

We have recently become aware of several more audio societies. While we don't have a lot of information about them, here is what we do have:

• Saint Louis Audio Society. A new organization. For information write: Saint Louis Audio Society, 7435 Cornell, University City, MO 63130.

• Chicago Audio Society. The first meeting of the group took place on June 8th. Thirty responses were received from a newspaper ad in the Chicago Tribune. What was more encouraging was that six individuals volunteered to help and have come up with some good ideas. For further information contact: Phil Marrow, 387 N. Edgewood Ave., Lombard, IL 60148.

• Melbourne (Australia) Audio Club. This group has been active for about two years and publishes its own monthly newsletter. They have a membership of over 200 and are still growing. The activity of the membership is astounding—between the regular monthly meetings at least nine special interest groups meet and many of these actually listen to music (something too few audio club members ever seem to find time for). — J. and J. Brinton (Massachusetts)

The Digital World Approaches

The audio world is becoming increasingly digitized. The first indications were gimmicks: digital frequency readouts on FM tuners. Then the Heath AJ1510 tuner came out using digital frequency synthesis for station selection as well as readout. The LED power readouts on the Bose 1801 and Harmon-Kardon Citation 16 power amps are obtained by an analog-to-digital conversion, as are similar pseudo-VU meters available elsewhere.

Now we have the Infinity Class D switching amplifier, which has a recognizable signal at the input and at the output, but everything in between is digital. The Audio Pulse and Sound Concepts delay lines are digital signal processors, as are the more advanced professional versions. (The Sound Concepts unit processes analog data digitally; the Audio Pulse converts the analog data to digital before processing.)

Why such a profusion of digits? Basically, because there are many tasks that need to interface with computers, and many tasks that need greater accuracy than is available on an analog readout. Because of this, the audio world is beginning to use the spinoffs of products originally developed for other purposes.

For example, several years back, the military went to pulse code modulation (PCM) for secure, high-rate communication. Infinity uses a similar technique in their amplifier. Digital shift registers are a basic part of any computer, and of much other advanced hardware. These same shift registers are in use as the basic operating technique in audio time-delay units.

<u>Onward to Microprocessors</u>. The big new tool in the digital world is the microprocessor. A microprocessor is basically the grown-up, high-capability version of the integrated circuit in your pocket calculator. It is the central processing unit of a computer, albeit a small computer. Of what conceivable use could a computer be in an audio system? The only reason to use such a device is because it can do something that more normal circuitry cannot, and the tasks that a microprocessor could do in an advanced preamplifier are so useful that you will probably be seeing top line "computer preamps" within 6 to 8 years.

For example, at my office I have a computer that has been used as a radio receiver, but not in the sense with which most of you are familiar. The receiver consists of an extremely quiet RF preamplifier and an analog-to-digital converter. After that, all work is done in the computer. This includes filtering, limiting, compressing, and decoding signal in the presence of much noise.

As an example of how sophisticated such filtering can be, there is one extremely sharp notch filter program used solely to remove power-line hum. And as the frequency of the power-line signal drifts—tenths of a hertz—the filter will automatically change its center frequency to compensate. It is easy to do this task in digital circuitry, because you do not have to be concerned about the problems of component drift with aging and temperature.

Another task we have implemented on this computer is a spectrum analyzer. Using a mathematical technique known as Fourier transformation, we can display the frequency response of a given signal from dc to 256 Hz in 1/4-Hz increments. Of course, this range could be expanded further through the full audio range, with more suitable resolution.

If you had a "smart" preamp with a microprocessor in it, it would be easy to program in a room equalizer, with broad-band or narrow-band response. An equalizer is most useful with a spectrum analyzer, and that could be part of the preamp, too.

The possibilities are endless. Pop removal from records, autocorrelators for noise removal, programs to compress or expand a signal to match Dolby A, Dolby B, dbx, anything.

What makes this type of approach attractive is that it is far easier to change a computer's program than it is to change any given piece of hardware. A manufacturer could increase the performance of his preamp or add new features very simply.

If you want to experiment, microcomputers with power supplies, input and output facilities, and extended memories have become available from several manufacturers. The biggest is MITS in Albuquerque. The audio kit manufacturer, Southwest Technical Products in San Antonio, is selling a microcomputer kit. Present prices run from about $300 up, depending on options.

This whole discussion may seem a little (or perhaps very) absurd. However, I would like to relate something that science (and science fiction) writer Arthur C. Clarke noted a few years ago when writing about the hazards of prophecy, and about how rapidly the real world moves. Clarke referred to "the failure of imagination" and "the failure of nerve." The former is self-explanatory; the latter is simply this: I haven't got the nerve to tell you what I really think is going to happen.

-- Mark Saklad (Massachusetts)

More on Audio Epistemology

Epistemology is a concept recently attracting much attention in some circles. Neither the subject nor its application is necessarily intuitively obvious, but for potential illumination of complex questions it merits some concentrated attention.

I am going to examine the matter briefly here—but please be aware that since there are about as many different schools of epistomology as of philosophy itself, I'm not going to try to present other than my own observations and conclusions.

Metaphysics is properly the first study of philosophy, and epistemology the second. Metaphysics deals with the nature of existence itself, while epistemology studies the origins and significance and validation of human knowledge.[1] Metaphysics asks the question, "What is it?" Epistemology asks, "How do you know?" and "So what?" It is that last question we find so interesting in audio.

Historically, the concept underlying today's consumer audio industry was that of higher fidelity sound reproduction, and although contemporary recordings commonly contain program material that never existed as live sound prior to playback, the idea of sound reproduction can provide a worthwhile lead toward a working definition of what it is that puts the "hi" into one's "fi."

Consider the case of an imaginary "perfect" recording of imaginary live music being played on an even more imaginary "ideal" reproduction system. Here an arbitrary listener is experiencing hypothetical sound waves identical to those originally generated at an imaginary performance. (Undoubtedly our listener is also comforted by the imaginary certainty that the system will never require hypothetical repair and cannot use more of his imaginary budget.)

Unfortunately this cannot happen. An identical sound field cannot be reproduced. Period. But this isn't enough to stop us—people still can't fly either. Instead they do something else that has the same effect: they build airplanes.

Likewise with high fidelity: the perfect audio system would be one that comes as close to reproducing those original sound waves as the laws of physics will permit. (By way of contrast an imperfect system would be one that falls short of these ultimate limits. It can be considered imperfect only because of its failure to do perfectly everything a system can do.) Our perfect system would be marked by its refusal to induce any added acoustical degradation upon the original sound field beyond that necessitated by the physics of sound. (Please note that this definition assumes a perfect signal source. But an imperfect source would still give us no degradation beyond its own, and the limits set by the physics of sound.)

This leads to an interesting situation: since it is the degree of excess degradation that determines equipment quality, we more accurately direct our attention to the phenomenon of low fidelity—especially in the area of equipment evaluation. A practical sound reproduction chain is an integration, a whole united by a common purpose: the optimal approximation of perfect sound reproduction in the sense described above. The purpose of quantitive measurement should be to objectively determine how significantly non-optimal a particular approximation is.

This means that when testing equipment everything counts. An amplifier is everything it is and does everything it does. If a designer were careful enough to isolate the tape outputs so that capacitive loading can't degrade the preamp's frequency response: it counts. If he didn't: it makes a difference that will show under the right (wrong?) circumstances.

Any double-blind testing must be done with care to avoid masking qualities one is testing for. The object of such testing must be to evaluate what the equipment will do to music when used in music reproduction systems and operated within its characteristic range. That is, when normally used. If a designer manages to produce equipment with a wider "normal" range of usability than the competition has, then that fact is what we want to know.

And in fact every sort of component must be tested this way in order to properly evaluate it: Very carefully. All audio equipment is primarily acoustical in function (even if it sees only electrical signals) and should be judged on the basis of its acoustical effects. The kicker in doing this is figuring out how to accurately determine what these effects are and why they are what they are.

The "why" part is, of course, one of the most fascinating aspects of the whole matter. For anyone who can feel at home in a five-dimensional coordinate space in which the traditional time and frequency domain measurements become special cases of a more general analytic approach, Richard Heyser's delay plane offers a lot of hope.[2] A systematic mathematical treatment of such questions could conceivably advance the state of the art by a few orders of magnitude. In any event, his system promises to be rather interesting.

References.

1. Ayn Rand, Introduction to Objectivist Epistemology, The Objectivist Inc., New York, 1966.
2. Richard C. Heyser, "The Delay Plane, Objective Analysis of Subjective Properties," J. Audio Eng. Soc., Vol. 21, pp. 690-701 (Nov. 1973); pp. 786-791 (Dec. 1973).

— Richard Bowser (Nebraska)

Mahler Festival in New York

The New York Philharmonic is presenting a Mahler festival during a four-week period in Carnegie Hall from September 26 to October 25. This will mark the first time an American orchestra will have performed all nine symphonies as well as the uncompleted Tenth and various songs during such a short period of time. Two of the three symphonies that employ choruses and vocal soloists will be conducted by James Levine, the Metropolitan Opera's new music director and a gifted Mahler conductor (judging from his performances of the First and Fourth Symphonies, released by RCA last year).

Not only is this series of concerts a "first," it is quite likely a "last" within the lifetime of many of us, considering the enormous expense of presenting this music in concert. To dedicated Mahlerites, the prospect of hearing all this music in a period of one month is sheer paradise. Any BAS members who share my enthusiasm for the music of Mahler will consider the trip to New York well worth it.

I suggest that anyone interested in subscriptions or individual tickets act as soon as possible. There are enough Mahler nuts in New York City alone that every concert will probably be sold out well in advance.

Tickets for the festival will be sold on a subscription basis, with prices ranging from $40 to $80. The subscriptions went on sale June 1 by mail order. Remaining single tickets will go on sale at the Carnegie Hall box office August 29th. Further information can be obtained by calling (212) 799-9595.

— Stephen Sarper (Connecticut)

Also, Fidelity Research has come out with a dynamically balanced tonearm (FR-64) and moving-coil cartridge (FR-1 Mk. W). The arm is somewhat similar to the FR-54 (which appeared in Discwasher ads) except for the pivots.

Satin's new M-117 cartridge is said to sound less colored than their previous models. The M-117 is a high-output moving-coil cartridge with the pivots for its moving parts coinciding at one point and with movement of the cantilever around its axis prevented. Frequency response extends to 50 kHz, with channel separation of 40 dB up to 10 kHz and 35 dB at 50 kHz.

— Norihisa Sayanagi (Connecticut)

A Magazine That Couldn't (?) Happen Here

Stereo Sound, a quarterly magazine published in Japan, costs about $5 an issue (approximately 500 pages). The magazine is known for testing many pieces of equipment of the same category. For example, tests of 75 integrated amplifiers, 41 tuners, and 41 record players have been featured in one or a series of two issues.

A taste of Stereo Sound can be obtained by reading AR-3a ads in American audio magazines published around 1970 (High Fidelity, Sept. 1970, page 40).

The most current issues I have at hand (Autumn 1975 and Winter 1976) include tests on 80 speaker systems. Six critiques gave individual evaluations, although three evaluators had to listen together due to time limitations. Each critique gave a score and a listening impression for each of the 80 systems. One of the critiques states that after listening to speakers all day every day for a week, one evaluator became ill and had to be put to bed.

Measurements were also taken with B&K and other instruments. Frequency response in an anechoic chamber (at 0° and 30° off axis), impedance versus frequency, bass distortion (75, 150, and 300 Hz at various levels), efficiency, and practical impedance were all measured.

Associated equipment included the following: Cartridges: B&O MMC 4000, Elac STS-455E, Ortofon VMS-20E, EMT XSD-15, Empire 4000DIII, Shure V-15 Type III, Decca Mark V, Ortofon SPV-GT/E, Denon DL-103; Record Players: Technics SL-1300, Lux PD-121 with SME 3009S2, Sony PS-8750, Denon DP-3700F; Amplifiers: Mark Levinson LNP-2, Yamaha B-1 and VC-1, McIntosh C-28 and MC-2105, Pioneer Exclusive C3 and M4, Lux C1000 and M6000, Denon PMA-7007, Onkyo Integra A755II, Diatone PA-V850, Lux SQ38 FDII, Denon PMA-235.

The following is a list of the frequency speakers were included in the "top 13" by the evaluators (the numbers give the frequency with which each speaker appeared in the reviewer's individual "top 13" lists):

Technics 7 (SB-7000)	1	Leak 2075	1
Electro-Voice Interface A	3	Infinity Monitor II	1
Spendor BC-H	2	Allison: One	2
Quad ESL	2	Cabasse Sampan Leger	1
Marantz 4GII	1	Dahlquist DQ-10	1
Dual CL-172	1	JBL L-16	1
Celestion Ditton 66	1		

These speakers were included in the sample but were not considered "tops" by the reviewers; however, they were high scorers:

Diatone DS-28B	Diatone DS-281	Sansui LM-033
Tannoy New Rectangular York	Technics SB-6000	Yamaha NS-451
Yamaha 1000M	Celestion VL-6	Pioneer CS-T88
Diatone DS- 50C	Advent 2	

Some other interesting articles that appeared in these two issues of Stereo Sound are the following:

- "Essence of Craftsmanship in Classic Audio Equipment": An article on old equipment such as Marantz tube amplifiers; includes a number of full color photographs and some interesting discussion.

- "Audio High Technique": An article devoted to the search for the "best sound." An example is a listening session trying to determine the best combination of JBL consumer and professional units, using active crossovers and multi-amplifier setups.

- "My Handicraft": An article for the do-it-yourselfer. An example is construction of a special horn-loaded corner enclosure for Tannoy driver units, with full explanation on the history of Tannoy speakers. — Norihisa Sayanagi (Connecticut)

More on Japanese "super records" next issue. — Ed.]

Biting Back at "Jaws"

I disagree with Jacob Rabinow's comments on tonearm damping in the April Speaker. Since zero-mass arms and cartridges are not available, damping is desirable. (Would you remove the shock absorbers from your car?) He grossly overstates the disadvantages of pivot damping while at the same time fails to mention the unavoidable overshoots and ringing that occur in undamped arms. I'm surprised no one at the meeting brought this matter into proper perspective. I believe Rabinow is quite incorrect in his conclusions because they are deduced using an assumption of gross overdamping. For a horrific account (together with oscilloscope photographs) of how a first-rate arm (SME) behaves with a highly compliant cartridge (ADC-XLM)—an account which really brings home forcefully the need for damping—see the first half of the article "Pickup Cartridges for CD-4 (Part 2)" by Roderick Snell in Hi-Fi News and Record Review, September 1975, pp. 59-64.

— Stanley Lipshitz (Ontario)

Problems with Moving-Coil Cartridges

I have found that both the Supex and Ortofon moving-coil cartridges are sensitive to temperature changes. The Supex's output will increase 2 to 3 dB at 15 kHz with a change in temperature in my room from 68° to 80°. The Ortofon shows similar tendencies. The Supex has been found variable in channel matching and frequency response here. The right channel output on one was 4 to 5 dB lower at 15 kHz than that from the left. Also the channel balance (which at 1 kHz is rated at 0.3 dB) was actually 1.5 dB. The Ortofon was balanced in output at 15 kHz but also was 1.5 dB different at 1 kHz. — Nathan Garfinkle (California)

Sonus/Formula 4/AR Combination

Having felt dissatisfied with my Dual 701/Shure V15 III record-playing system for some time, Peter Pritchard's interesting BAS lecture spurred me to action. One week later I purchased a used AR-XA and sent for a Formula 4 tonearm; one month later I had a Sonus Blue Label cartridge.

Mounting the Formula 4 on the AR took 8 hours including removing the AR arm and its portion of the suspension system, readjusting and rebalancing (by adding mass) the remaining suspension, installing the arm, and modifying a Miracord dustcover to replace the AR's. Since the Miracord dustcover is one inch larger in each horizontal dimension than the turntable, it can be used while records are being played. Actual arm adjustment and installation was both simple and enjoyable since all critical adjustments can be accomplished easily. And if I say so myself, the new turntable/arm/dustcover combination is quite handsome.

Penguin Stereo Guide—Part IV

This is a continuation of the list of outstanding records that Dr. Brian Leeming has culled from The Penguin Stereo Record Guide.

- Prokofiev, Symphony No. 5 (B Flat), Op. 100, Berlin Philharmonic, Karajan, DGG SLPM 139040.
- Prokofiev, Piano Sonata No. 7 (B Flat), Op. 83, Maurizio Pollini, DGG 2530 255.
- Puccini, La Boheme (Complete, 2 discs), Victoria De Los Angeles, Beecham, RCA Victor Chorus and Orchestra, HMV SLS 896.
- Puccini, Madame Butterfly (Complete), Mirella Freni, Pauarotti, Karajan, Vienna Philharmonic Orchestra, Vienna State Opera Chorus, Decca SET 584-6.
- Purcell, Dido and Aeneas (Complete), Janet Baker, St. Anthony Singers, Lewis, English Chamber Orchestra, L'Oiseau-Lyre SOL 60047.
- Purcell, King Arthur (opera, complete), Heather Harper, St. Anthony Singers, Lewis, Philo-musica of London, L'Oiseau Lyre SOL 60008/9.
- Rachmaninov, Piano Concerto No. 4 (G Minor), Op. 40, Michelangeli, Gracis, Philharmonia Orchestra, HMV SXLP 30169.
- Rachmaninov, Symphony No. 2 (E Minor), Op. 27, Previn, London Symphony, HMV ASD 2889.
- Ravel, Piano Concerto (G Major), Benedetti, Gracis, Philharmonia Orchestra, HMV SXLP 30169.
- Ravel, Daphnis et Chloe, Chorus Royal Opera House, Covent Garden, Monteux, London Symphony, Decca "Ace of Diamonds" SDD 170.
- Ravel, Introduction and Allegro for Harp, Flute, Clarinet, and String Quartet, Melos Ensemble, L'Oiseau Lyre SOL 60048.
- Ravel, String Quartet (G Major), Op. 10, Italian Quartet, Philips SAL 3643.

Record Recommendations—Ensayo/MHS

I'd like to add to Christopher Gupta's note (May 1976) on recordings by Ensayo available through MHS. There are four other works (all of them very good) available:

- MHS 1623/Ensayo NEL 2002: Tchaikowsky, Serenade for Strings in C, Op. 48; and Dvorak, Serenade for Strings in E, Op. 22.
- MHS 1475-76/Ensayo NEL 2003-4: Satie, Piano works, Laurence Allix.
- MHS 1736/Ensayo NEL 2009: Mozart, Divertimenti for String Orchestra, Nos. 1-3, K. 136-138.
- MHS 1280/Ensayo NEL 2010: Ravel, Introduction and Allegro for Harp, String Quartet, Flute and Clarinet.

-- Randy Phillips (Missouri)

Japanese Products

A new generation of direct-drive turntables is coming out of the Japanese market. Sony and Technics already have introduced their models here, but there are still a number of others. Victor (JVC) has come up with models TT-101 and TT-81, Denon with model PP-7000, and Pioneer with its prototype PL-X.

These turntables all incorporate quartz frequency-locked, direct-drive systems. The specifications included in the advertisements in Japanese magazines are impressive, although how much improvement can be heard is uncertain. Just as in the Sony PS-8750 ads in U.S. magazines, the new generation direct-drive turntable ads tend toward emphasizing the system's immunity to stylus drag. (This seems to have significance in Japan— Stereo Sound's tests include measurements in that area.)

The turntables mentioned are very expensive. Fortunately, Japanese companies have not forgotten the mass market and are coming out with a number of inexpensive belt and direct-drive turntables and record players. Some models that still are not imported to the U.S. are models from Denon and Micro.

Using the Shure in the Formula 4 confirmed Alvin Foster's experience with damped arms: the cartridge no longer sounds vague and shallow; it now delivers a good stereo image with little wander and good detail. Frequency response did suffer from the low capacitance of the Formula 4's cables, however.

By this time, I was more than ready to audition the Sonus and it did not disappoint. The cartridge is very smooth and detailed with deeper and more detailed bass than the Shure. I found myself constantly saying "I never heard that before !" when listening to familiar records. The Sonus also provides one with excellent breadth and depth of image. Music that was adequately reproduced by the Shure was well reproduced by the Sonus and with the additional ability to locate subsections and soloists with precision even during complex passages.

Listening fatigue is virtually absent even when listening at high levels for extended periods of time. Record surfaces must be kept absolutely spotless, however, as the Pathemax stylus seems to deny passage of dust around itself and is sensitive to all groove imperfections.

Though I have not listened to all the high-end cartridges, I would unhesitatingly recommend auditioning the Sonus Blue Label/Formula 4 combination. The arm had no trouble handling the cartridge (no cantilever deflection could be detected) and the cartridge sounds beautiful. For a total investment of $263 + tax ($50 for the AR-XA, $90 for the Formula 4, and $86 for the Sonus), I have been thoroughly rewarded.

For reference, I listen to AR-LST 1's driven by a Dunlap-Clarke 500 with a Davis-Brinton phono preamp running into the high-level section of a Dynaco PAT-5. The Davis-Brinton preamp saved my PAT-5's life. . .it's super. Also, the performance of the Dual's tonearm can be drastically improved (but not to the level of the Formula 4) by the use of Dual's new low-mass headshells.

— Robert Henry (Massachusetts)

Minority Report on Shure Versus Supex

[Ed. Note: California member John Puccio's opinion differs from Foster's and Leonard's (BAS Speaker, Jan. 1976) not only on the relative merits of the two cartridges but on those of transformers and pre-preamps. Also, it's good to point out that many product evaluations perforce are based on single samples—though less so here than in some major magazines—and therefore there's the happy possibility that everyone is right. . . Confusing isn't it? — J.B.B.]

Foster and Leonard state that the Supex appeared to sound better, and end up attributing this to the presence of a high-frequency peak in the Supex. But they admit in paragraph six, page 2, that they had not equalized both cartridges and A-B'd for comparison to confirm this. In conclusion, they state, "...there is sufficient evidence to suggest that the differences we heard between the Shure and Supex combinations were primarily a function of frequency response, not of operating principal." Unfortunately for their argument, nowhere in the article did they bother to point out what this "sufficient evidence" was. I have equalized both cartridges for identical response and still found the Supex to sound vastly superior. Now, whether this superiority is worth three to four times the price of the Shure is up to each person to decide. It's my opinion that a small expenditure for a good cartridge is worth more than some huge expenditures in speakers. And besides, I've heard the "high-end-peak-makes-it-sound-better" theory applied to the Decca as well, and the theory doesn't hold up any better there either. I believe that some pickups simply possess better transient response than others, thus producing a more clearly delineated sound. Such is the case with the Decca and most of the moving-coil cartridges.

Second, Foster and Leonard claim that the transformer used with the Supex produced more of a high-end peak than the pre-preamp. Perhaps this was true of the specific combinations and samples they tested, but my experience has differed. When I bought a Supex, about 1½ years ago, the pre-preamps I heard with it sounded brighter and more edgy than with Supex's own transformer. After reading Foster and Leonard's article, I was moved to measure the high-frequency output of my Supex both with and without the transformer. The frequency response, using two different test records, did not change with the transformer (SDT-77).

And finally, Foster and Leonard claim a greater high-frequency peak in the Supex than in the Shure. My Supex and Shure measured almost identically. Sample-to-sample variations must surely account for this discrepancy between Foster and Leonard's samples and my own. My own, by the way, were both quite flat, showing only a 1 to 2 dB rise above 10 kHz, at slightly different points, in each case. I suggest that Foster and Leonard look for a better sample Supex. The Supexes with such peaks are exceptions.

All of this is not, however, to suggest that the Supex is the best cartridge in the world. It has its faults, and among them is a tendency to sound somewhat "two-dimensional." My own preference in cartridges at this time runs to the Denon, provided one has the money. At a more reasonable price, I favor the Grace 8C or 8L. — John Puccio (California)

Quad 405 Addendum

Regarding the comment in the March issue of The BAS Speaker to the effect that the Quad 405 specifications did not look impressive on paper, beware that you do not fall into what appears to be a very common error in this regard. Quad quotes an input slew rate limit of 0.1 volt/millisecond, which corresponds to an output slew rate of over 5 volts/millisecond, comparable to most top-quality power amplifiers. What is very impressive is its performance into reactive loads, something very rarely specified. For two very thorough reviews of the Quad 405 see Hi-Fi News and Record Review, April 1976, and Studio Sound, March 1976. Another review is scheduled for the June 1976 issue of The Gramophone. — Stanley Lipshitz (Ontario)

A Turn-On/Off Transient Eliminator

The Audio Research SP-2 is a very fine preamp, but it does emit rather strong warmup thumps during the first 22 seconds after turn-on. Several years ago, picking up a suggestion from The Stereophile, I began using an Amperite thermal time-delay relay to delay the turn-on of my Mc-30 power amps for about 10 seconds so that preamp thumps wouldn't get through to the speakers. This worked well, the only problem being severe pitting of the relay contacts. (I did use a capacitor across the relay contacts as per the Stereophile article.)

When I replaced the Mc-30's with a Quad 405 early this year, the thumps became an intolerable problem as the 405 was instantly transmitting arcing transients to the speakers. I investigated the possibility of obtaining another type of relay with contacts designed to switch more power, but such relays cost $20 or $30 or more.

I solved this problem with the approach shown in Fig. 1. The heater of the Amperite relay is still controlled by the preamp switch, but the relay now controls power to the coil of a dc relay whose contacts are normally closed. The normally closed relay is then connected across the preamp output jacks to short them. (This will cause no damage to any normal preamplifier.)

When the Amperite's contacts close after the prescribed delay (in this case 30 seconds), the short is removed from the preamp outputs. The result is a smooth turn-on of the system, with only a slight thump from the 405 itself when the preamp power switch is turned on.

This scheme uses inexpensive parts and requires no modification of either the preamp or the power amp. An inexpensive 12-Vdc relay with two sets of SPST contacts can be used, along with an appropriate power supply.

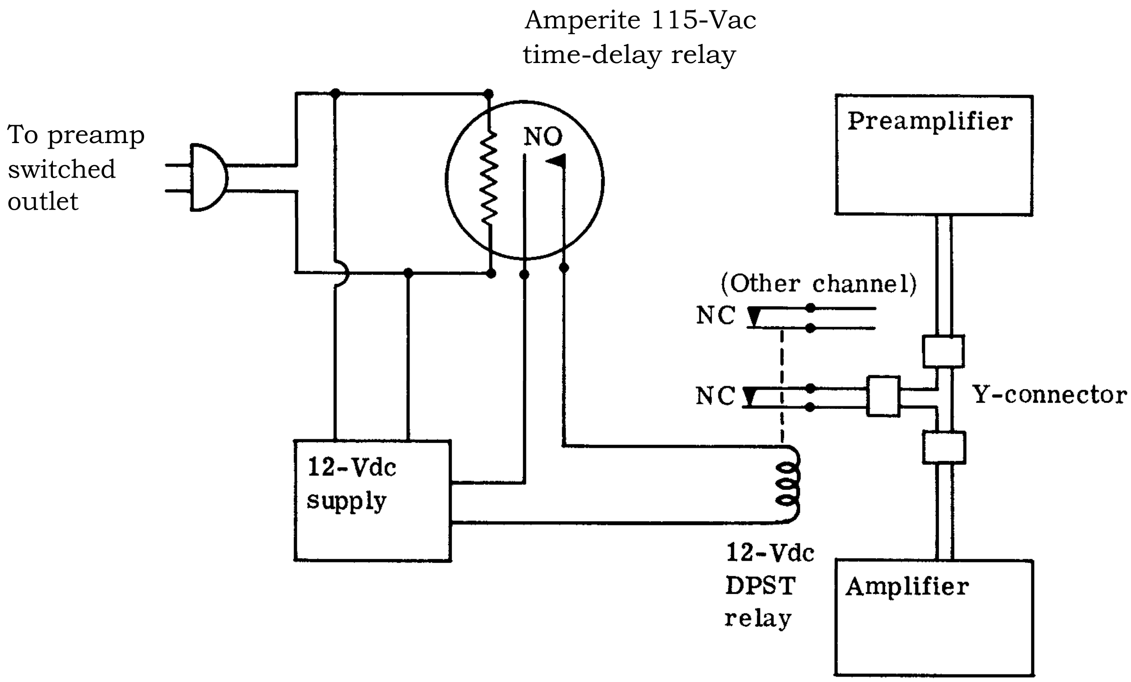


Fig. 1. Turn-on/off transient eliminator schematic

The Amperite relays are available with delays of 2, 5, 10, 15, 20, 30 and more seconds. While mine (model 115 N030) has 115-volt heaters, others are also available with 6.3-volt heaters. The ones I use plug into a standard octal socket and cost $2.70 from Lafayette. Inexpensive 12-volt dc relays are available for a few dollars from Lafayette, Radio Shack, and similar outlets.

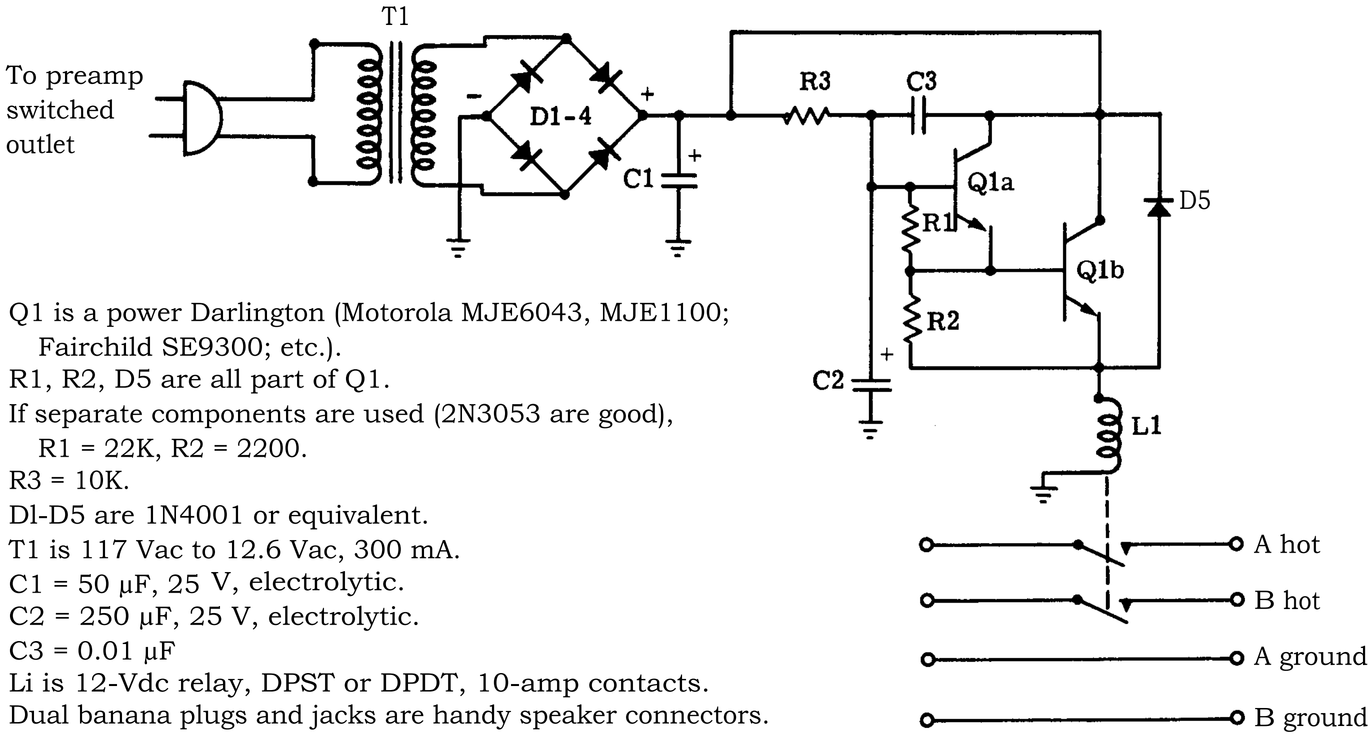
It's not as sophisticated as the muting system used in the Crown IC-150 preamp, but it works !

— Thomas Shedd (Illinois)

[Note that this device as described eliminates only those transients originating in or prior to the preamp; any transients from the amplifier will continue to reach the speakers. Amplifier transients can be eliminated by using the same basic idea but using the second relay to switch the speakers on and off (as in the following article) rather than shorting the preamp output. In this case, use a relay whose contacts are normally open, or you will fry your output transistors. Also, the 12-Vdc power supply can be eliminated by using a 115-Vac relay, since the Amperite unit can switch 115 Vac at 3 amps. — Bob Borden]

Another Transient Eliminator

Are you tired of bangs and pops and speaker strain that occurs when you turn your high-power high-fidelity system on or off? Then try this simple relay cricuit (Fig. 1), to be added between the power amp and the speakers. It is powered and activated from a switched preamp ac outlet. The turn-on delay is over five seconds, while turn off is virtually instantaneous, thus sparing the speakers from all preamp and power amp transient voltages.



Q1 is a power Darlington (Motorola MJE6043, MJE1100;
Fairchild SE9300; etc.).
R1, R2, D5 are all part of Q1.
If separate components are used (2N3053 are good),
R1 = 22K, R2 = 2200.
R3 = 10K.
Dl-D5 are 1N4001 or equivalent.
T1 is 117 Vac to 12.6 Vac, 300 mA.
C1 = 50 µF, 25 V, electrolytic.
C2 = 250 µF, 25 V, electrolytic.
C3 = 0.01 µF
Li is 12-Vdc relay, DPST or DPDT, 10-amp contacts.
Dual banana plugs and jacks are handy speaker connectors.

Fig. 1. Turn-on/off transient eliminator schematic

Most of the parts are available from Radio Shack, Lafayette, You-Do-It, etc. All fit easily in a 4- x 5- x 7-inch MiniBox. The values shown work well, but if more turn-on delay is desired, increase C2 (not R3). If relay does not close, reduce R3 by 30% to 50%.

Upon ac turn-on, the +18 volts dc charges C2 through R3 (in series to ground through R1 and R2 and the coil resistance). When the base of Q1a reaches 8 to 10 volts, the relay activates, closing the speaker leads. When the ac is switched off, C1 (being small) discharges rapidly through Q1a&b and the coil, opening the relay almost instantly.

Some cautions: The relay should have 6- to 10-amp contacts. Do not omit C3 or Q1 will be an RF oscillator and relay operation will be unreliable. And do not use common speaker ground wires in the box. Little or no heat sink is required on Q1 since it is saturated when on.

— Abbott W. Lahti (Massachusetts)

In the Literature

Audio, July 1976

- Audio Etc.: The saga of the acoustic suspension loudspeaker. (p. 14)
- Behind the Scenes: Discussion of the new Elcasete, 3¾ ips, 1/4-inch tape cartridge. (p. 20)

- Three Car Stereos Tested: The Pioneer KP-500 is the winner over the Panasonic CQ-840EU and Motorola TC877AX FM/cassette units. (p. 28)
- Car Stereo Directory. (p. 36)
- Car Speaker Placement: Tom Holman concludes that in-door placement is best, at least for the small car used in these tests. (p. 54)
- Reviews of the ESS electronic crossover (p. 56)

Audio Engineering Society, Journal of the, May 1976

- Synthesis of 4-2-4 Matrix Recording Systems: From CBS comes a mathematical discussion of the SQ matrix, with particular attention to optimal decoding and mono/stereo compatibility.
- Low-Frequency Tracking Behavior of Pickup Arm-Cartridge Systems: James Kates of AR states that low-frequency tracking is governed by stylus compliance, stylus damping, tonearm effective mass, and tonearm pivot damping. An equivalent circuit model is presented. Conclusions are that: "For existing pickups, using a stylus compliance of 20 x 10^{-6} cm/dyne as typical, an effective mass of 20 grams is too heavy. Reducing the mass to 10 grams and adding pivot damping greatly improves the low-frequency tracking-force requirement, but further reductions in effective mass yield only marginal reductions in total tracking force." Reference is made to a July 1963 AES Journal article by Ben Bauer, where damping the pivots rather than using cartridge damping alone is described.
- New Factors in Phonograph Preamplifier Design: By Tomlinson Holman of Advent. We thank Holman for letting the BAS in on this work early. (p. 263)
- Advance in Turntable and Tonearm Design: Sony engineers describe crystal-controlled servo turntable motor and the carbon-filament tonearm as used in the $900 PS-8750 table. (p. 276)
- Richard Heyser was elected a Fellow of the Acoustical Society of America (p. 332) and the 53rd convention (Zurich) is summarized. (p. 286)

dB, June 1976

- The Sync Track: Clarifies dB's and VU units on tape recorder meters. (p. 8)
- Compander Increases Dynamic Range: New Telefunken device operates in four bands and claims 30 dB improvement in dynamic range. (p. 16)
- Controlling Room Ambience: Contains a table of absorption coefficients of various materials as a function of frequency. Emphasis is on the effects of reverberation on intelligibility. (p. 32)
- In summaries of Zurich AES show and Chicago NAB gathering, the BASF Unisette demonstration is discussed; competitors, in addition to the Elcasete, will be the Otari two-track, 3¾ ips cassette system. (p. 37)

EDN, May 20, 1976

- Special Report—Linear IC's: New state-of-the-art op-amps include National's BI-FET LF156 and RCA's Bipolar/MOS CA3130. A tabulation compares these with the 741A and the 108A-superbeta. Several super-741's are also described. (p. 46)

EDN, June 20, 1976

- Low-Cost Oscilloscopes: 18 scopes are compared, tested, and tabulated (unusual for this journal). (p. 80)

Electronic Design, June 21, 1976

- Siliconix advertisement summarizes the line of power V-FET's from this domestic manufacturer. (p. 13)

Electronic Engineering Times, June 7, 1976

- A new type of high-efficiency switching audio amplifier is described. (p. 16)

Electronics, May 27, 1976

- Play's the Thing in Home Video Games: IC's, the FCC, and falling prices mark the future for TV games. (p. 80)
- Shift Register With Feedback Generates White Noise: Circuit produces a power spectrum flat within ±1 dB over the entire audio range. (p. 107)

Electronics, June 10, 1976

- Quadraphonic Sound Poised Again: Tate Ltd., a British company, has signed with National Semiconductor to develop an IC version of Tate's SQ decoder and enhancement system. In a breadboard version, this system claims 30 to 40 dB separation with 0.05% distortion and 70 dB S/N. (p. 68)

High Fidelity, July 1976

- Worthwhile editorial urges disclosure about compression used on discs so those with expanders can undo the damage. (p. 24)
- Reviewed this month are the new Crown preamplifier, a JVC portable stereo cassette similar to the Sony 153, the ADS 200 mini-loudspeaker (softspeaker?), and the Onkyo T-4055 FM tuner,
- In the Musical America section, a nice preview of upcoming young talent (p. MA-13) selects only the Greenwood Consort and BSO trumpeter Rolf Smedvig from the Boston area. But in reviews of the Boston Opera Company's "Montezuma" and the New England Chamber Opera's "The Death of King Philip," young vocalists Valerie Walters and Beverly Morgan find the recognition which soon will be commonplace (p. MA-24).

IEEE Spectrum, June 1976

- Jacob Rabinow elected to the National Academy of Engineering (one of 104) for inventions in computers, power transmission, and post office automation. (p. 96)

Popular Electronics, July 1976

- Stereo Scene: Discusses phono tonearm adjustment and maintenance. (p. 22)
- Reviews of the Pioneer electronic crossover and the MXR audio equalizer. (p. 74)

Radio Electronics, July 1976

- The Audio Pulse time-delay unit is discussed, with some technical detail. (p. 43)
- Reviewed this month are the Tandberg 10XD ($1300) open-reel machine and the Pioneer SX-1200 super-receiver. (p. 49)
- CB in HiFi: How to get the truckers and Rubber Duckies out of your stereo. (p. 62)

Stereo Review, July 1976

- The New London Cassettes: Suggests that even these new sources leave a bit to be desired. (p. 30)
- In the review section, J. Hirsch hits Consumers Union again, and then describes the merits (only) of the ho-hum Akai cassette deck, the Stax SR-5 headphone unit, Design Acoustics D-2, and the long-lost Lenco L-85 "IC" turntable. (p. 36)
- Record Wear: Has some nice photos of stylus tips, which illustrate just how difficult it is to examine one when wear is in doubt. (p. 56)

Wireless World, May 1976

- A rather complete review of loudspeaker quality and design appears in this issue, and for a change it does not stress phase effects. (p. 45)
- But several letters do. (p. 61)
- Foil Antenna Array Hidden by Wallpaper: May answer some questions "The Idea File" has been asked. (p. 57)
- Reviews of two European audio shows are interesting. (p. 76, 89)

— H. Zwicker and Dana Craig

June BAS Meeting

Business Meeting

The June meeting of the Boston Audio Scoeity was held in the rooftop facility of Bolt Baranek and Newman in Cambridge. As of this writing, there still is no "home port" for the BAS; we are looking for a permanent meeting site, and until one is found we will be something of a gypsy organization—rather inconvenient for a group of almost 1000 members.

The business meeting was to the point, with BAS president Jim Brinton leading an appeal for volunteers to help run what has become a very large operation—probably the largest international consumer audio group, and possibly more "effortful" even than the UK's national association of gramophone societies. Last month's Speaker includes a rundown of the skills required to help the BAS survive: Q.B.

Brinton noted that the BAS has lasted to this point mostly because of the relatively uncomplaining efforts of six to eight members, including the elected officers. After from two to four years bearing the weight of the organization—which is, after all, (a) non-profit and non -salary-paying, (b) "volunteer," and (c) growing like blazes—these pillars of the BAS are beginning to groan (audibly) under the load. Unless some greatly increased delegation of responsibility becomes possible by September, the BAS may cease to publish The Speaker, publish it only irregularly, or the BAS may totally disappear.

This projection brought, however, no cascade of volunteers from the assembled membership, leading some officers to wonder whether guilt-based child raising might not in fact be preferable to more permissive techniques.

In pleasant contrast to this was Frank Farlow's exposition of the committee approach to Speaker fulfillment (that's a publishing term for stuffing The Speaker in envelopes, putting on stamps and addresses, and getting them to the post office). Farlow announced that the June "stuffing party" would occur at his home in Brookline.

The usual mixture of equipment and records was for sale during the meeting's intermission. Items on the block included Scott Kent's record "New Angle on Harpsichord," Insight Records "Pigs Eye Jass" (Vol. II from that firm) via Ira Leonard, and Davis-Brinton phonograph preamplifiers at their dealer cost of $167.50 plus tax.

Dr. Brian Leeming delivered discs as part of the BAS overseas record-buying service. He notes that this service is hanging by a thread. Not only are too few people partaking of the opportunity to get excellent European pressings of fine music, but almost unbelievably, many of those who have ordered are failing to take delivery. If this sort of thing were peculiar to the summer months, we might blame it on vacations, but similar behavior has plagued not just Leeming's year-long attempts to organize this worthwhile service, but also the earlier service run by David Letterman. (There was some muttering that the BAS could not get takers on a diaper changing service if one were offered.)

At any rate, Leeming has a large number of catalogs detailing the offerings of many European labels. Recent additions include Norse and Swedish catalogs. He also has access to selective guides from British sources. (For a small example, see the extracts from the Penguin Guide being reprinted monthly in The Speaker.) Contact Leeming at (617) 277-2247. — Jim Brinton

Meeting Feature: Geoff Langdon, AKG

Geoff Langdon is Technical Manager for AKG/USA and as such bridges the gap between the technical and marketing departments of the firm; it is his task to explain the needs, roles, and developments of each side to the other. With such a job description, Langdon makes an excellent lecturer on the proper application of microphones, although for a spokesman for such a fine product line, he seemed at times defensive when presented with some of the tough questions for which the BAS is becoming known.

Mike Types and Characteristics. Microphones can be characterized by their operating principle and/or their directional characteristics. Among "dynamic" microphones, there are moving-coil microphones, which operate like miniature "dynamic" loudspeakers in reverse, and ribbon microphones, in which an ultra-lightweight aluminum strip is suspended between the poles of a permanent magnet, fulfilling the functions of both the diaphragm and voice coil of the moving-coil (often simply called "dynamic"—as if not all microphones were dynamic) types. Condenser microphones are usually constructed with one or two conductive membranes and a fixed backplate, forming a variable capacitor which is kept polarized by one of three mehtods: by an externally applied voltage, by a "permanently" applied static charge, or by a radio-frequency current generated by a crystal oscillator in the microphone circuitry. In the first two types of capacitor microphone, the capsule output is at a very high impedance and must be converted so that it can be sent down a cable without severe losses. (Ribbon mikes have the opposite problem of too low an output impedance and often a special transformer must be incorporated to raise their output to practical levels.) In the third case, the diaphragm motion causes frequency modulation of the applied RF current and the resulting signal can be demodulated with the possible benefit of somewhat lower noise since the capsule capacitance presents a lower impedance at radio frequencies than the garden-variety capsule does at dc.

Moving-coil microphones have the relative advantages of lower cost, lower inherent noise, freedom from overload, overall robustness of construction, and no need for external power supplies. Many ribbon microphones have enjoyed reputations for "warm" and "natural" sound as compared with condenser microphones (although some people say they have no top end), and in the past decade there has been notable improvement in the ruggedness of the available ribbon models (particularly from Beyer). The microphones with the cleanest transient response have tended to be ribbon and condenser designs with their relative freedom from subsidiary resonances in the audio band. Electret condensers, whose membranes are given a long-term (several years) static charge during manufacture, have undergone rapid improvement in the last few years due to the availability of new transducer materials. Langdon forecast that as materials continue to improve, the electret design would eventually replace traditional condenser microphones because of economies of manufacture, ease of use, and freedom from "arcing" in extreme humidity (although a high-humidity environment shortens the life of the electret's static charge). At present, however, electret elements cannot be made to perform as uniformly as the best traditional condenser elements, and they have not been widely adopted for highest-quality recording or broadcast.

In general, condenser microphones have prevailed in studios where their superior transient response and polar patterns have been valued; their higher output levels have also helped to overcome some of the noise generated in other parts of the recording chain. Electret elements do lose their charge with time and will "age" more rapidly if exposed to high temperatures and humidity. They must then be replaced; in the case of the new AKG electret element the cost is comparable to that of the stylus in a medium-priced phonograph cartridge.

Apart from differences in operating principle, microphones also may be classified according to their directional characteristics. Two fundamental types are recognized: pressure transducers and pressure-gradient transducers; a microphone can be one or the other or both. Pressure microphones have only one side of the element exposed to the sound source and they respond equally to any impulse regardless of the direction it comes from. (Note that any object can cast "shadows" at higher frequencies where its dimensions approximate sonic wavelengths; as a result, even a "pure" pressure transducer could not be truly omnidirectional at the highest audio frequencies unless it were smaller than desirable for an adequate signal-to-noise ratio.)

Pressure-gradient tranducers measure the instantaneous pressure difference between two nearly adjacent points which are both exposed to the sound source. In principle these are bi-directional microphones (figure-8 response) since at 90° and 270° incident angles, sound waves will reach both points in phase, with no net change in pressure difference; but at 0° (on-axis) or at 180° the pressure difference will "track" the waveform of the sound. A cardioid characteristic can be obtained through the electrical addition of a bi-directional and an omnidirectional characteristic (either from separate elements or from two parts of one element, often the means used in cardioid ribbon mikes) since the rear lobe of a figure-8 microphone is 180° out of phase with the front lobe, cancelling the corresponding signal from the omnidirectional element. A cardioid characteristic can also be created by having the rear-incident sound pass through an acoustic delay network so that it meets its front-incident component at the membrane simultaneously and is cancelled out. This is the method used in most cardioid microphones made today.

AKG manufactures moving-coil microphones, traditional condenser microphones, and has more recently added a line of electrets designed as "semi-professional or amateur" items. These have a mid-range and high-frequency rise and a low-frequency rolloff to make up for the deficiencies of "typical consumer tape recording equipment, to put `punch' in the sound that the consumer likes to hear . . .," and to improve intelligibility in highly reverberant recording environments. Speaking thus, Mr. Langdon aroused a controversy (or was it just a widespread misunderstanding?) that hung around for the remainder of the session. He noted that customer demand had led AKG to elevate extreme treble respone (about 2.5 dB at 20 kHz, beginning around 14 kHz in the standard cardioid capsule) of the C-451 microphone and proposed that in nearly all cases a microphone "is not intended to be an absolutely accurate, flat, linear reproduction device. It's a 'creative' tool." (Spirited disagreement on this point.—Ed.) Since a microphone's measured specifications can give only a bare outline of its sonic performance, makers and users of microphones can come into conflict where no standard of "naturalness" exists, or, as is more usual, when there are differences of opinion about which deliberate falsification of the sound is appropriate. It is clear that a microphone can be evaluated only in use and not on the sole basis of specifications, as in the case with loudspeakers or phonograph cartridges.

Mr. Langdon further pointed out that if a microphone's 1-meter on-axis frequency response were flat in an anechoic environment, it would tend to roll off at high frequencies in a free-field environment. This rolloff tends to balance the effect of a designed-in rising response. A new design for the omni-directional capsules in the C-451 series, by which the sound is let in only along the rim of the capsule rather than through the front, also is for the sake of more nearly flat free-field response, although the capsule resonance is also manipulated "to put a little bit of `punch' up around 16 to 18 kHz." He expressed the view that a microphone without deliberate coloration would not be popular enough to make it worth producing.

He also pointed out that most consumer hi-fi stores are not commercially able to stock high-quality microphones or to be familiar enough with them to help potential customers, and he recommended dealing with professional equipment dealers for advice and the possibility of trying out some equipment before deciding what to buy.

Miking Technique. A typical method of recording in stereo is to space two microphones 4 to 12 feet apart at some distance in front of the ensemble (A-B method). Various A-B setups are shown in Fig. la. The stereo effect is brought about by intensity differences, arrival-time differences (and thus phase differences if and when audible as such), and timbral shift (the tendency—especially of pressure microphones—to pick up less treble information from distant sources). None of these effects corresponds to what a listener realistically hears in any one position in the hall. (The playback system has to bridge the gap from there to the ears.) There is no form of manipulation or mixing of such microphone signals that would extract from them a compatible monophonic signal; if the two channels are simply summed, the result is a doublet with very unfavorable octave-related frequency response and pickup pattern variations. Langdon demonstrated this by moving a white-noise source around and between two spaced microphones, either cardioids or omnis, recording and then replaying their summed output. The result was the obvious emphasis of parallel, sliding bands of frequencies and, one infers, the corresponding suppression of the frequencies in between. This effect was almost eliminated when cardioid microphones were put onto a stand's stereo bar with their membranes approximately in the same plane with respect to the sound source and with their elements nearly touching; stereo perspective was at least as clear with this "co-incident" (X-Y) method, which also maintained mono compatibility.

There is a second method of stereophonic recording which generates a compatible monophonic signal: Mid-Side (M-S) recording, and it has certain intriguing advantages of its own. A microphone of any desired polar characteristic is placed as though for a single-mike mono recording, and a figure-8 microphone is placed as close as possible to this mono microphone, but aimed so that one of its nulls faces directly toward the center of the ensemble. (See Fig. lb.) A monophonic signal is of course available from the first (or mid) microphone. The second (side) microphone gives all the needed "cues" for directionality since its symmetrical left and right lobes are "in antiphase." By matrixing the microphone outputs (M + S, M - S) in any desired proportions, a compatible stereo pair of signals can be derived as though two directional microphones had been symmetrically placed as in X-Y recording (assuming that the matrixing can be done precisely enough). The apparent width of the stereo image can be adjusted by varying the proportion of "M" to "S" in the mix: As the AKG booklet "M-S Stereo Recording Techniques" points out, "A change in directional characteristics makes it possible to adjust the recording to any given [set of] room characteristics. Broadening the polar characteristics from [matrix-simulated] crossed figure-8's to crossed cardioids narrows the apparent width of the sound image. At the same time it reduces reverberation picked up from the rear. This effect is the opposite of that obtained by varying microphone distance, since, in the latter case, moving further back narrows the source but increases the reverberation. Attenuating the M or S signal before matrixing expands or contracts the width of the image, respectively, with the same effects upon the reverberation as changing the directional characteristic."

By way of further explanation: it is feasible to construct a condenser microphone with two capacitor elements—one on each side of the backplate, for example, arranged so that the electrical outputs of the two elements are summed—and to vary independently the polarizing voltage on each element so as to alter the effective directional response of this dual microphone continuously from omnidirectional through cardioid super- and hyper-cardioid to bidirectional. This can even be remotely-adjustable. For M-S recording it is desirable to mount two such microphones in the same microphone body so that the capsules are very close together and also for convenience. Neumann and AKG both make such microphones (AKG is about to come out with a new version of theirs with not only the several classic patterns per channel but also with the added option of using the four capsules separately for "clover-leaf" quad recording) and Schoeps makes a stereo microphone with three patterns (omni, cardioid, bi-directional), obtained from a single-diaphragm capsule per channel and mechanical, non-remote pattern selection.

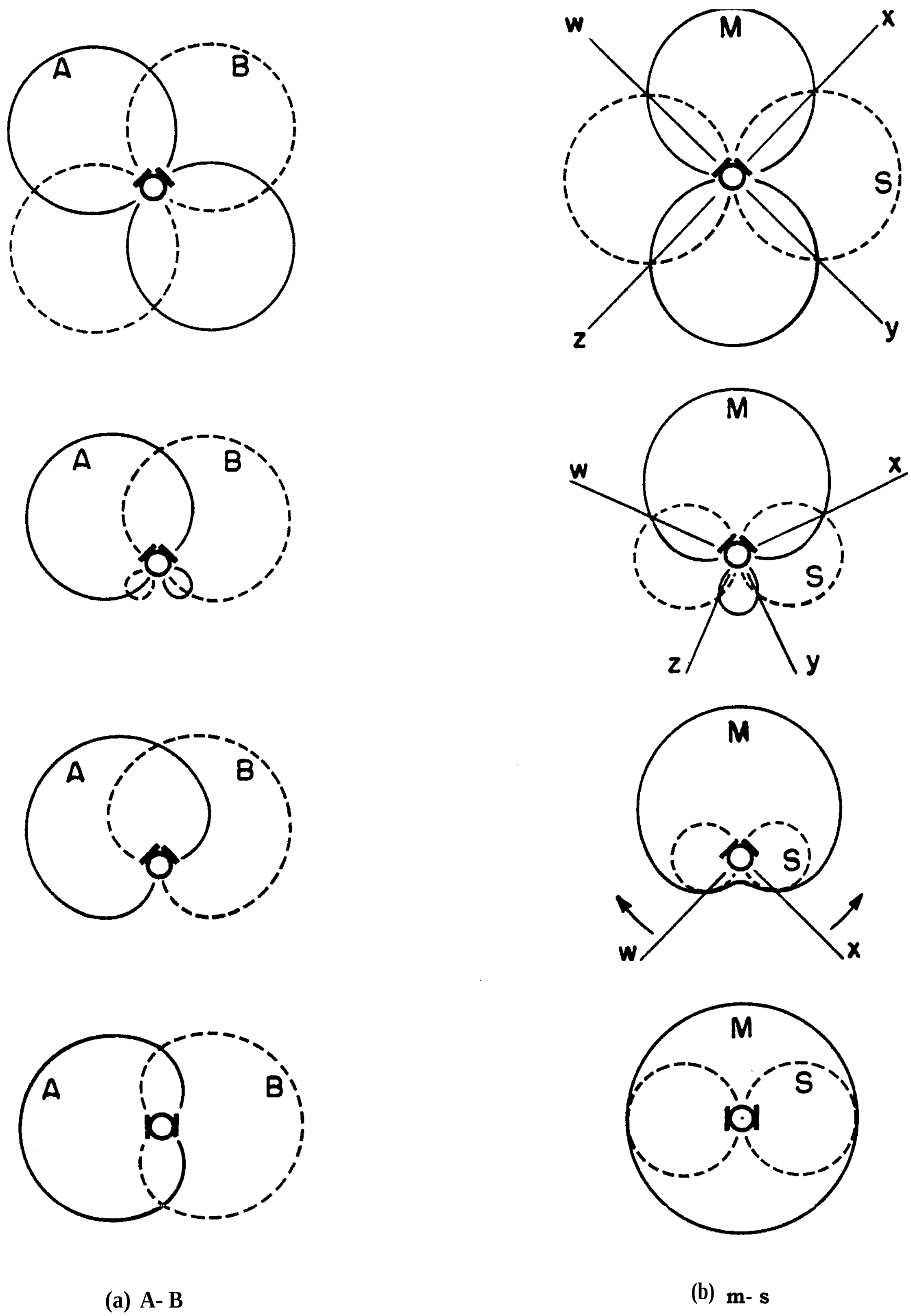


Fig. 1. A-B and M-S microphone configurations

"Matrixing" can even be used to derive an omnidirectional signal from two cardioid elements (place their elements facing, touching, and in the same plane, and sum their inputs in phase), a cardioid from a figure-8 and an omni (as above), or even a figure-8 from either two cardioids (combine their outputs out of phase), or a cardioid and an omni ("subtract" the omni's output from the cardioid's). All the microphones must match in sensitivity. For interview situations, a figure-8 pattern is more useful and more natural sounding than cardioids at some random distance put back to back and added in phase (e.g., Shop Talk), which creates another irregular doublet when mixed into mono.

Mr. Langdon also demonstrated a means of comparing the response of one microphone against another microphone's (of the same model and of known good health) by placing the mikes side by side and summing their outputs out of phase to find the null point (if any) and listening to the resulting frequency response. The sound of the "nulled" signal will indicate the differences between the two outputs and normally should be severely attenuated throughout the low end and mid-range due to cancellation.

Lastly, Langdon demonstrated that by putting two omnidirectional microphones side by side and summing their outputs in antiphase, a differential "noise cancelling" effect is created. Since any distant sound is effectively equidistant from the microphones, it is cancelled, but close speech will be picked up more by one mike than by the other and thus will rise above any background noise.

In response to a question about phase linearity in microphones, it was pointed out that although minimum phase disturbance is available from some types of omnidirectional microphones, there is no recognized sonic advantage to be gained from optimizing this characteristic.

AKG Z. All AKG microphones are nominally at 2000 output, and are meant to be bridged. The impedance is not entirely independent of frequency, especially in directional moving-coil microphones. Too low a load impedance would lower the clipping point of a condenser mike; a very high load impedance would perhaps be undesirable from a noise standpoint but gives the microphone as such no difficulty.

With low-impedance microphones, the principal cause of RF interference is cable defects. Dynamic microphones have signal levels as low as a phonograph cartridge and AM signals can sometimes compete for audibility; condenser microphones, as well as power amplifiers (!), can sometimes become overloaded due to the detection of radio signals fed back into their outputs from connecting cables.

—David Satz (Massachusetts)

A New Method for Estimating Phase Distortion and Experiments on Phase Audibility

Daniel Shanefield

Background. Like many developments in physics, this study arose from an inability to do something. Although my attempts to fool people with live-versus-recorded comparisons have succeeded when the listeners were far from the sound source(s) (more than twenty feet), my most diligent attempts to fool the same listeners have utterly failed when they were up close to the loud-speakers (less than twenty feet away). This was true with both "one-eared" and "two-eared" experiments.

There are several possible explanations, and among them is "phase distortion"--by which I mean that the bass and the treble are delayed by different amounts of time during the record/playback process. According to this hypothesis, at a distance from the speaker, a large percentage of the sound is reflected, and therefore is phase-distorted, for both the live and the recorded cases. You can't tell the difference, and therefore you can be fooled. But up close, where the sound is mostly direct, the live sound would not be greatly phase-distorted, while the recorded sound would be distorted because of imperfections in the record/playback process. Presumably the ear could tell the difference and wouldn't be fooled.

New commercial loudspeakers with improved phase response have been popping up all over the place. Some are "linear phase,"[1] some are "minimum phase,"[2] and some are claimed to be essentially "phase

I say "essentially" because it is not practical to be exactly coherent, since a motion of your head up or down from the center axis of a two-way, non-coaxial loudspeaker can put you out of exact coherence when it comes to frequency pairs such as 800 Hz and 8000 Hz. If exact coherence is really important, then the whole thing is hopeless from the standpoint of commercial loudspeaker design.

At the other end of the scale, extreme phase distortions (corresponding to differential delays of 10 milliseconds or so) have been shown by telephone researchers to be audible and bad. But that extreme sort of distortion is not what we are discussing here, either.

It has been hypothesized from time to time that a fair degree of phase coherence is necessary for realism because live musical sounds have steep initial wavefronts (almost like square waves), and the only way to preserve these fronts during recording is to keep the high frequencies and low frequencies traveling together. But this is probably wrong, because live musical sounds do not have steep initial wavefronts, and, quite the contrary, they take at least a few tenths of a milli-second to build up to full volume. That has been shown for music and handclaps by Duncan et al.,[4] and you can see it yourself if a storage oscilloscope is available. A few tenths of a millisecond is several complete cycles at 8 kHz, so initial wavefronts do not have to be steep—at least from that line of reasoning they don't.

However, it is well known that we don't fully understand these things, and "lines of reasoning" do not always jibe with audio realities. If phase distortion is audible, maybe it does affect realism, even if we can't say why. Anything that's a "distortion" and is audible should probably be eliminated. So is it audible?

Many previous experiments have said, "No" for monophonic sound. But V. Hansen and E. R. Madsen of B&O in Denmark have claimed[5] that small monophonic phase changes can be audible under some circumstances. An interesting debate on the subject has been appearing in almost every issue of Wireless World magazine this year, much of it in the letters column. Evidently a few acousticians believe that Hansen and Madsen are correct, and a few others don't. (Of course, everyone agrees that interaural phase changes are audible, and this contributes to stereo localization.)

I think that Hansen and Madsen overlooked a serious potential problem in their technique, which I happened to uncover while trying to duplicate part of it. They used Koss ESP-9 electrostatic headphones, which I also used, and they assumed that the acoustic output of these headphones has a wave shape that is a very accurate reproduction of the electrical input. However, my impulse tests show that the ESP-9 in an essentially anechoic environment rings a little bit, and it therefore acts in some ways like a slightly reverberant room. Hansen and Madsen admit that a reverberant situation will give a falsely enhanced audibility to phase shifts, due to destructive interference effects. This causes loudness changes at certain frequencies, which the ear can hear very well indeed. So maybe we just don't have perfect enough transducers to do the experiment unequivocally.

Personally, I don't think we really need to know whether a pure signal of some kind is audible, although it is an academically interesting subject. What we do need is an experiment that directly compares a phase-coherent loudspeaker with an incoherent one, keeping everything else identical, and playing music. We need to determine which one is more realistic.

People at the B&W Company in England have recently published a report that seems at first sight to be concerned with just that very experiments They arranged listening tests of music with two nearly identical loudspeakers, one phase-coherent and the other non-coherent. The jury was polled on its preferences, which turned out to be strongly in favor of coherence.

Frankly, I think their results are inconclusive. First of all, their experiment was evidently not done blind, and we really might expect a jury to choose a sophisticated-looking speaker (their new Model DM-6) as opposed to a plain one (which the incoherent one certainly was). Secondly, the "better" sound was chosen without immediate access to the live performance, so "better" might not be "more realistic."

I suppose you can see from my criticism just what experiment I'm going to report, so let's get on with it. But, before we can compare different degrees of phase distortion, we have to be able to measure it.

The New Test Method. Here is a new method for estimating the amount of phase distortion present in any one link of the record/playback chain, or in the whole chain. The advantage of this method is that it is easy to use, as compared to phase meter approaches (which are not as simple as they might appear to be), and compared to the "raised cosine"[7] or "sine-squared"[8] and fast Fourier transform[9] approaches. (By the way, I think the Fourier method is subject to considerable error, unless the frequency response and other critical attributes are measurable to a high degree of accuracy.)

My new method involves running a square wave through an octave-type graphic equalizer. If the slide that controls the 8-kHz frequency band is set at +6 dB, and the other slides are all set at -12 dB, each square wave will become a skinny spike, as viewed on an oscilloscope. Now, if the 60-Hz slide is also raised to +6 dB, the waveform becomes the thing shown in Fig. 1. I call this an "S-wave."

a

If an S-wave is now run through a tape recorder, in most cases it will become "phase-distorted" and look something like the wave shown in Fig. 2, because there now is a difference in the delays applied to the treble and the bass frequencies.

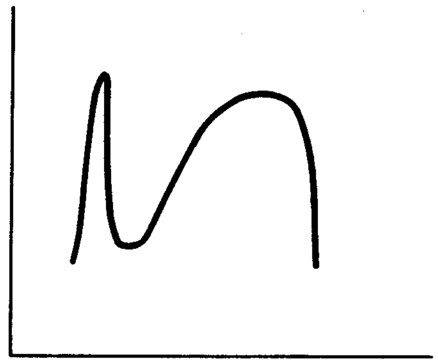


Fig. 1. An S-wave with no phase distortion

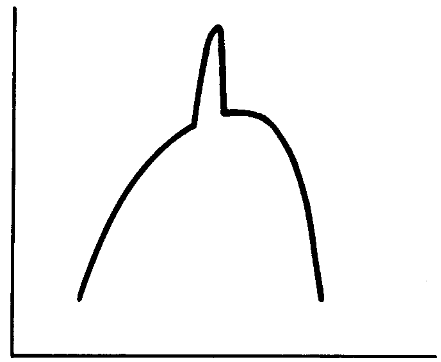


Fig. 2. A phase distorted S-wave

A disadvantage of this testing method is that one cannot easily obtain a continuous reading of phase shift versus frequency during a sweep through the audible spectrum. However, we usually don't need a continuous reading, and looking at only four or five points on the frequency scale will tell us a lot.

For improved accuracy at the treble end, it is best to break the frequency span into smaller steps such as 8 kHz/2 kHz, then 2 kHz/500 Hz, then 500 Hz/120 Hz, etc. This way, small time delays in the higher of the two frequencies being studied will show up better.

Using a sequence of S-wave tests, I have found that, while the extreme bass and treble of the Tandberg 3300X cross-field tape recorder are badly phase-distorted (unequally delayed), the range from 120 Hz to 8 kHz is essentially constant phase. (Note that this is not "linear phase "[1] or "minimum phase"[2]; this is essentially zero phase distortion.[3])

S-waves can also be sent through a complete record/playback system, and my experiments with that can be summarized as follows. The recording chain consisted of an S-wave going through a Bose 901 equalizer, a Dynaco 400 power amplifier, a single Bose 901 loudspeaker facing forward (not reflecting), an air link, a Thermo Electron 814 microphone, and a Tandberg 3300X tape recorder with Maxell UD tape at 7½ ips. Playing the Tandberg back through the equalized Dyna 400, the Bose 901 speaker (facing forward again), and the 814 mike to an oscilloscope showed no phase distortion visible with the S-wave test, from 120 Hz through 8 kHz. Therefore, the whole system was essentially phase-coherent.

However, a variety of other devices such as dynamic microphones, other tape recorders, and electrostatic headphones each showed gross phase distortion (as in Fig. 2) when they were individually substituted into the chain.

An Impulse Test for Ringing. In addition to measuring phase distortion, S-waves can be used to test speakers and microphones for ringing. A very-low-frequency square wave is fed into the graphic equalizer, which causes a gap in the time between each successive S-wave. Overswing across the zero-amplitude line on the oscilloscope display indicates ringing (a form of poor "transient response"). The pulses are too short to allow full ringing build-up, so the method is less than ideal. But it is convenient, and it does quite graphically show up any tendency toward unidirectional overswing. The room reverberations can usually be separated out, since they come much later.

By adjusting the equalizer pass bands to find those worst-case frequencies that maximize the ringing, it was found that electrostatic transducers (ESP-9 phones and B&W Model 70 speakers) and also Magnapan speakers are not Simon-pure after all, and do ring slightly. This was also true with pure treble as well as pure bass. Good-quality cone-type speakers turn out to be just as effectively damped. (I suppose we should have expected this.' The Mylar diaphragms might have low mass, but they also have very little mechanical damping action—not much more than in a bass drum!) For confirmation of this, see the[10] advertisements for the B&W Model DM-6 speaker appearing in some British publications.

Experiments on the Audibility of Phase Distortion. And now for the main event. Using the 814 mike, Tandberg, and Bose chain, I monophonically recorded repetitions of a 698-Hz xylophone note (with its overtones, of course). The loudness was kept at moderate levels so that overload was not a problem. It was played back using separate graphic equalizers as a crossover, splitting the signal into the below 1-kHz part, which went into one 901 speaker, and the above 1-kHz part, which went into another 901 speaker. (The frequencies above 8 kHz were filtered out altogether, since they would have been phase-distorted.) Putting the two speakers approximately side by side (or one above the other) gave no difference in realism over putting them several inches in front and in back of each other.[11]

The graphic equalizer itself has some effect on the phase, so the essentially zero phase distortion (coherent) signal was not obtained with exact side-by-side placement. Also, the treble tends to come from the apex of the speaker cone, while the bass comes from areas farther forward.[12] A displacement of 1¼ inches did produce essentially coherent sounds. Both these and the grossly incoherent (5-inch displacement) sounds were compared to live performances of the xylophone notes. (This was also repeated with a variety of other musical notes.)

There were plenty of "differences" in the sounds. In fact, I've never found two loudspeakers that sound exactly alike under ordinary circumstances,[13] so the sound does depend on which Bose 901 handles the treble, etc. Watch out for this when you read other people's reports on similar experiments !

No relative speaker position was clearly the most realistic when quickly or slowly A-B'ed against the live performance. (This is an example of what I called the LAB test in a previous article.[14])

In an earlier series of experiments, I made a silly mistake, which *The BAS Speaker's* editors kindly corrected. If I haven't made another one this time, the conclusion is that a fairly high level of phase distortion does not affect realism.

This business of speakers each sounding different dredges up another one of those deep philosophical problems. Unless we attack it with very clear thinking, it's liable to become a virtual Loch Ness monster. Suppose all loudspeakers sound different (and S. K. Pramanik of B&O states very definitely in reference 13 that they do). Then how can we ever expect one to sound like the live performer, if it can't even sound exactly like a duplicate loudspeaker? It seems to be impossible to remove all such differences, or at least it seems impractical.

Here is my way of pushing the philosophical monster back down. I'm willing to accept a *difference* between the live and recorded sounds, just as I'll accept a difference between duplicate live instruments, each being equally "realistic." What I am trying to do is prevent a blindfolded listener from *identifying* which sound is recorded. This is what does happen at a distance, where listeners can actually be fooled. (Do you remember those three D, I, and B questions in my 1974 article?[15]) Then, even if a listener practices for a while and does learn to identify the sources, I'm trying to get the honest listener to say that neither one is *best.* It should sound like two different "live" instruments. That is what I mean by "realistic." So much for philosophy; let's get back to experimental details.

Loudspeaker Design Considerations. The main criticism I can see for the whole study is that the Bose 901 facing forward is possibly a too-imperfect device to prove anything, primarily because of the diffraction peaks in its frequency response curve caused by its multiple drivers. (It's not meant to be used facing forward for up-close listening.) Also the side-by-side arrangement of the two speakers causes additional interference peaks, because the crossover that feeds them is not optimized to prevent this. But actually, the whole loudspeaker-room system was carefully equalized using my pseudo-performer method, [14] and it did sound quite good in spite of the diffraction.

It certainly would be useful if other people tried similar experiments with a variety of loudspeakers. Just please be observant of all the snares mentioned above. This is a tar pit surrounded by quicksand.

What is the true explanation of my failure to fool listeners up close, if it's not phase distortion? I don't have a strong opinion at this point.

It's interesting that my best results in listener-fooling have been obtained with a Magnapan MG-II loudspeaker, played through a large, thin curtain which is strongly lit up from the front and dark in the back. This speaker has a fair amount of phase distortion, probably because of its crossover design. Maybe the Maggie's relatively good performance is due to its size, to its bipolar radiation pattern, or to its unusual frequency response curve. Whatever it is, it doesn't have a monopoly on realism, though, because a giant pile of conventional speakers arranged to be bipolar and big will sound just about as good.

I have a feeling that the ear is sensitive to subtleties in the back-reflections off the walls of the listening room, and that is how we can tell the live from the recorded sounds. This might be an interference effect that gets translated into a frequency response effect, and it might be affected by the size and shape of the loudspeaker. Maybe small speakers have high-Q (finely tuned) environmental interferences and resonances, causing strong colorations while large speakers such as Magnapans or electrostatics have diffuse and weaker colorations of this type. Or, maybe it's the shape of the wavefront, with large speakers providing a more nearly planar-shaped wave.

For a xylophone-to-microphone distance of about a yard or more, the wavefront that hits the mike is nearly planar. If the loudspeaker is put where the mike was, maybe the speaker should produce a similarly planar wavefront. (However, I suppose that a closer mike distance might work better with a non-planar speaker, and I guess this ought to be explored further.)

As soon as the weather gets warm enough, I'm going to repeat my best live-versus-recorded comparisons outside, up on a pair of ladders. (If you see any of my neighbors, please tell them, "It's just audiophilia-nothing really dangerous.")

References

1. "Linear phase" means that the phase distortion is not zero but is linear with increasing frequency. An unusually clear example of this is shown in graphs for a loudspeaker reported by S. Ishii et al. of Panasonic, in a paper delivered at 52nd AES Convention, preprint 1059-3L (available from the Audio Engineering Society, New York, N.Y. for $2.00 post-paid). Some Technics and Infinity speakers are claimed to be linear phase.
2. "Minimum phase" does not mean zero phase distortion either, but in this case it refers to the least amount of such distortion that is theoretically possible when using conventional LC crossover designs. Many speaker systems are designed this way. See R. C. Heyser, Audio, Dec. 1974, p. 71.
3. This is really zero phase distortion, or nearly zero. It is being claimed for the Sony 880-2 tape deck and for some of the newer Ohm, Dahlquist, B&O, and B&W loudspeaker designs.

4. M. G. Duncan, et al., J. Audio Engineering Society, p. 610 (Fig. 5C), Oct. 1975.
5. V. Hansen and E. R. Madsen, J. Audio Engineering Society, p. 783, Dec. 1974.
6. J. Bowers and S. Roe, Hi-Fi News and Record Review, p. 56, April 1976.
7. I. Nomoto, et al., J. Audio Engineering Society, p. 9, Jan. 1976.
8. Advertisement for B&W Co., Wireless World, p. 70, Feb. 1976.
9. T. Muraoka, et al., 52nd AES Convention, preprint 1088-6G, available from the AES for $2.00.
10. See Reference 8, or Hi-Fi News and Record Review, p. 24, March 1976.
11. For discussions of somewhat related listening experiments, see articles in The BAS Speaker by Dennis Colin, June 1975, and by Mark Davis, Sept. 1975.
12. J. Moir, Wireless World, p. 74, April 1976.
13. S. K. Pramanik, Wireless World, p. 531, Nov. 1975.
14. D. Shanefield, BAS Speaker, March 1975.
15. D. Shanefield, BAS Speaker, Nov. 1974.

Speaker Wiring: What Size Wire is Sufficient?

Roy Cizek

[Ed. Note: Where speaker wire is concerned, the National Electrical Code is misleading. That skinny, 22-gauge stuff sold as "speaker wire" in some of the sleazier hi-fi emporia is safe, if fire is your only worry. But to get the performance from your system promised by the speakers' and amplifier's specifications, you should use the largest wire convenient. Speaker designer Roy Cizek's exploration of this topic, heretofore thought to be of relatively little consequence, is the most thorough seen to date and uncovers more reasons for thoughtful wire selection than you might have suspected existed. "Narrow-gauge" wire not only can degrade damping factor, but amplifier power output, system frequency response, and more.—Jim Brinton]

How should we determine the minimum size of wire that will suffice for connecting up a loudspeaker to an amplifier? Well, one might first proceed by noting that if an amplifier were putting out 400 rms watts continuously into a 4-ohm load, the current flowing through the wires connecting amplifier and load would be 10 rms amperes. This is so since power into a resistive load is equal to current squared times resistance:

$$P = I^2R, \qquad \text{so equivalently} \qquad I = \sqrt{P/R}.$$

By the same formula, 400 rms watts into an 8-ohm load would mean a current of 7.07 rms amperes was flowing.

Now, by the 1971 National Electrical Code (National Fire Protection Association Handbook of the National Electrical Code, John H. Watt, Ed., p. 292) two-conductor, 18-gauge flexible insulated wire can safely carry 10 rms amperes continuously. Considering that 400 rms watts is a rather high continuous power to be putting into most loudspeaker systems, it would seem by this criterion that 18-gauge wire should be quite satisfactory for hooking up to even very powerful amplifiers, and that we might easily come down a few gauges to no. 20 or no. 22 for amplifiers or receivers that can put out only 100 rms watts per channel or less.

This is apparently the prevailing way of looking at the question of wire size, judging from the manner in which most systems are wired and from some of the extremely small gauges that are commonly sold as "speaker wire."

But It's Wrong. Unfortunately, the additional series resistance introduced between amplifier and speaker by using too small a wire size for a given length run can have a number of undesirable effects on system response and efficiency, quite apart from any question of the speaker leads becoming a fire hazard.

Consider, for example, a loudspeaker connected to a power amplifier with a 20-foot run of two-conductor no. 18 wire. This amounts to a total of 40 feet of no. 18 wire in series with the speaker. From a wire table we find that the resistance of no. 18 wire is approximately 6.385 ohms per 1000 feet, or 0.006385 ohm per foot. The resistance of 40 feet of no. 18 wire is therefore 0.2554 ohm. Since this resistance of the wire is in series with the loudspeaker, the same current must flow through both and the power dissipated in each is proportional to its resistance. This means that if we drive 100 watts of signal into the combination of a 4-ohm load and a 0.2554-ohm series resistance, then

$$\frac{0.2554}{4 + 0.2554} \times 100 \text{ watts} = 6 \text{ watts}$$

will be dissipated in the wire's resistance, and only

$$\frac{4}{4 + 0.2554} \times 100 \text{ watts} = 94 \text{ watts}$$

in the 4-ohm load. Since the sound output of a loudspeaker is ideally proportional to the electrical power in, this drop of 6% in the power delivered to the loudspeaker results in an equivalent 6% or 0.27 dB lowering of the efficiency of the system [$10 \times \log_{10}$ (94 W/100 W) = -0.27 dB].

There is, however, an additional factor to be taken into account, since most amplifiers can deliver more power into low-impedance loads than into high-impedance ones. An ideal constant-voltage amplifier output, with no limit on the current that can be supplied, but only on the maximum voltage available from the power supply, could put out twice as much power into a 4-ohm load as into an 8-ohm load ($P_{max} = E^2_{max}/R_{load}$). If such an amplifier were capable of putting out a maximum 100 watts into a 4-ohm load, then increasing the resistance of the load on the amplifier to 4.2554 ohms, as by inserting 40 feet of no. 18 wire in series, would reduce the maximum power output to 94% or 94 watts. Then, by our previous calculation, <u>of this 94 watts</u> from the amplifier, only 94% or 88.4 watts would actually be driven into the 4-ohm load, since 6% or 5.6 watts would be dissipated in the wire's resistance. Total loss is 11.6% or 0.54 dB.

Now with any real amplifier there <u>are</u> limitations on the current that can be supplied, so that the rated maximum power output into 4 ohms is <u>not</u> actually double that into 8 ohms. Therefore, depending upon the particular amplifier being used, the actual power available into a 4-ohm load connected up with 20 feet of no. 18 wire will be somewhere between 88.4% and 94% of the amplifier's rated output into 4 ohms. The 94% figure considers just the power lost in the wire; the 88.4% figure includes also the maximum possible offset of the change in the total impedance presented to the amplifier.

The same 20 feet of no. 18 wire used to connect up an 8-ohm load would cause the actual power delivered to this load to drop to somewhere between 94% and 97% of the amplifier's rated output into 8 ohms. Tables 1 and 2 at the end of this article show the reduced power available into 4- and 8-ohm loads, when connected up with a 20-foot long run of two-conductor wire of gauges 12 through 22.

Some people recommend installing a fuse in series with a loudspeaker as a protection against overload. This also introduces additional resistance between amplifier and speaker. A type 3AG 1¼-amp Slo-Blo fuse, for example, has a resistance of approximately 0.285 ohm. If this fuse is used with a 20-foot run of no. 18 two-conductor wire, the total added series resistance due to wire and fuse is 0.540 ohm. This combination would cause available amplifier power to drop to between 78% and 88.4% of rated power for a 4-ohm load, and to between 88.4% and 94% of rated power for an 8-ohm load. In general, lower amperage fuses of the same type will have even higher resistance, as will physically larger types with the same amperage rating. Tables 3 and 4 show the reduced available power into 4- and 8-ohm loads when connected with a 3AG 1¼-amp Slo-Blo fuse and a 20-foot run of two-conductor wire, gauges 12 through 22. (It is well to note that these calculations need not apply to fuses in amplifier output stages contained within a feedback loop of the amplifier.)

Another way to look at the effect of speaker-lead resistance is to consider its effect on the amplifier's damping factor, as actually seen by the loudspeaker. By definition, the damping factor of an amplifier into a given load is equal to the load impedance divided by the amplifier output impedance:

$$\text{Damping Factor} = \frac{\text{Load Impedance}}{\text{Amplifier Output Impedance}}$$

so equivalently

$$\text{Amplifier Output Impedance} = \frac{\text{Load Impedance}}{\text{Damping Factor}}$$

This means that three amplifiers having damping factors of 20, 100, and 1000 into a 4-ohm load must therefore have output impedances of 0.2 ohm, 0.04 ohm, and 0.004 ohm, respectively (= 4 ohms/20, 4 ohms/100, and 4 ohms/1000). When a length of wire is used to connect an amplifier to a loudspeaker, the resistance of the wire must be added to the amplifier's output impedance to calculate the damping factor seen by the loudspeaker. For the three amplifiers just mentioned, the actual damping factors into a 4-ohm load connected up with 20 feet of two-conductor no. 18 wire are 8.8, 13.5, and 15.4, respectively [= 4/(0.2 + 0.2554), 4/(0.04 + 0.2554), 4/(0.004 + 0.2554)]. In fact, even starting with an ideal amplifier of zero output impedance and infinite damping factor, 0.2554 ohm in series will reduce the damping factor into 4 ohms to 15.7 [4/(0 + 0.2554)]. With 0.285 ohm more series resistance, as from a 3AG 1¼-amp Slo-Blo fuse, these damping factors will be further reduced to 5.4, 6.9, and 7.3, respectively. An infinite damping factor would be reduced to 7.4. Tables 1, 2, 3, and 4 also show the effects of added wire and fuse resistance on some amplifier damping factors, for both 4- and 8-ohm loads.

Now, damping factor is one measure of an amplifier's ability to control those resonances in a loudspeaker that show up in its impedance curve. An impedance versus frequency curve for a typical two-way loudspeaker system is shown in Fig. 1.

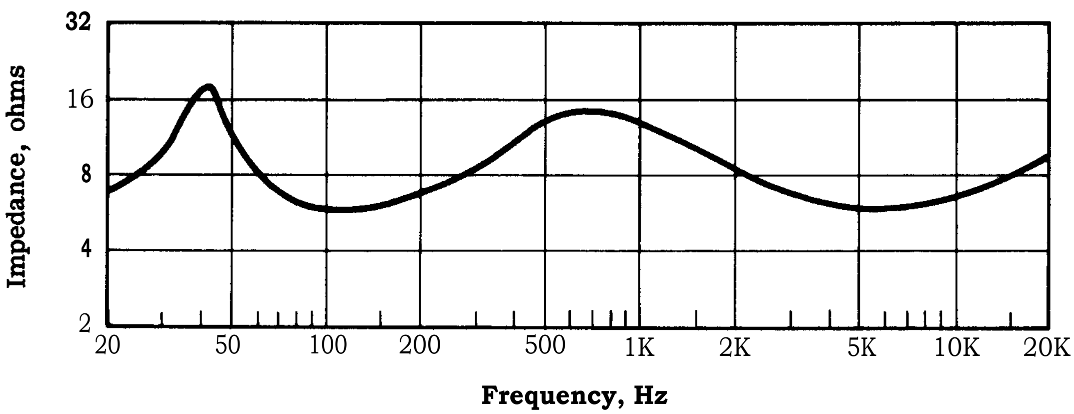


Fig. 1. Magnitude of impedance versus frequency curve for a typical 8-ohm two-way loudspeaker system

Note that although the impedance of this speaker is nominally 8 ohms, its actual impedance varies from a maximum of 18 ohms at 42 Hz to a minimum of 6 ohms at 120 Hz and at 6 kHz. The peak in impedance at 42 Hz is caused by the fundamental resonance of the woofer in the cabinet. The peak at 650 Hz is due to the combined effects of the tweeter's fundamental resonance, the woofer's voice coil inductance, and the various reactive components of the crossover network. If such a loudspeaker is designed to have good frequency and transient response when the effective amplifier damping is high, then decreasing this damping by using too small a wire size or a fuse in the line will tend to produce peaks in the *frequency response* corresponding to those in the impedance curve, as well as poorer transient response and increased ringing.

An effective damping factor of 8 or lower may introduce changes in response of up to 1 dB. At low frequencies and moderate levels, due to the crowding of equal loudness contours (see Robinson and Dadson, "Threshold of Hearing and Equal-Loudness Relations for Pure Tones, and the Loudness Function," *Journal of the Acoustical Society of America*, Vol. 29, No. 12, Dec 1957, pp. 1284-1288), the *apparent level* difference to the ear may be more like 1½ or 2 dB, in addition to the *increased ringing at the resonant frequency*. Speakers using vents or drones (or drone cones) are particularly susceptible to this problem, since their sharp low-frequency cutoff is accompanied by a greater tendency towards ringing to begin with (see, for example, A. N. Theile, "Loudspeakers in Vented Boxes: Part II," *Journal of the Audio Engineering Society*, Vol. 19, No. 6, June 1971; pp. 475 and 477-478).

The fact that too much added resistance in series with a loudspeaker can cause deterioration of frequency and transient response, as well as a drop in the power available to the speaker, is a strong reason for avoiding the use of smaller wire sizes for connecting to the amplifier output. Special attention should be paid to long runs of wire to rear channels or remote speakers.

Some points to note are:

- Generally, 4-ohm or lower impedance speakers will be more sensitive to the effects of wire resistance than will 8-ohm or higher impedance speakers.

- Contrary to common practice, it can be especially important to use heavier wire with smaller amplifiers or receivers, since they have low power output and low damping factors to begin with.

- Standard solderless connectors are available from electronics parts distributors for connecting no. 12 or even no. 10 wire to the screw or binding post terminals on amplifiers or loudspeakers. They are easily crimped-on with a hand-crimping tool and come pre-insulated. Some suitable solderless connectors are AMP 32968, Waldom T2150 or T2050 (insulated ring types), and AMP 324172, Waldom T2723 (insulated flanged spade or forked type).

- Since we are concerned about total wire resistance, and not about current-handling capacity, there is no problem with reducing the diameter of a heavy-gauge conductor to that of 18 gauge to fit into spring-loaded terminals. Simply cut away some of the strands of the heavy wire until it fits the terminal, and tin. Do not thin down a greater length of the wire than necessary, say a maximum of one-half to one inch from the end.

Table 1. Effects of 20 Feet of Two-Conductor Wire Connected to a 4-Ohm Load

Wire Gauge	Total Wire Resistance, ohms	Available Power Into 4-Ohm Load (Watts) Out of a Possible:					Power Loss, dB	Effective Damping Factor for an Amplifier With Rated D.F. Into 4 Ohms of:			
		25	50	100	200	400		20	100	1000	
22	0.646	19-22	37-43	74-86	148-172	297-344	1.30-0.65	4.7	5.8	6.2	6.2
20	0.406	21-23	41-45	82-91	165-182	330-363	0.84-0.42	6.6	9.0	9.8	9.9
18	0.2554	22-23	44-47	88-94	177-188	353-376	0.54-0.27	8.8	13.5	15.4	15.7
16	0.1606	23-24	46-48	92-96	185-192	370-385	0.34-0.17	11.1	19.9	24.3	24.9
14	0.1010	24	48-49	95-98	190-195	381-390	0.22-0.11	13.3	28.4	38.1	39.6
12	0.0635	24-25	48-49	97-98	194-197	388-394	0.14-0.07	15.2	38.6	59.3	63.0

Table 2. Effects of 20 Feet of Two-Conductor Wire Connected to an 8-Ohm Load

Wire Gauge	Total Wire Resistance, ohms	Available Power Into 8-Ohm Load (Watts) Out of a Possible:					Power Loss, dB	Effective Damping Factor for an Amplifier With Rated D. F. Into 8 Ohms of:			
		25	50	100	200	400		20	100	1000	
22	0.646	21-23	43-46	86-93	171-185	342-370	0.67-0.33	7.6	11.0	12.2	12.4
20	0.406	23-24	45-48	91-95	181-190	362-381	0.43-0.21	9.9	16.5	19.3	19.7
18	0.2554	23-24	47-48	94-97	188-194	376-388	0.27-0.14	12.2	23.9	30.4	31.3
16	0.1606	24-25	48-49	96-98	192-196	384-392	0.17-0.09	14.3	33.3	47.4	49.8
14	0.1010	24-25	49-49	98-99	195-198	390-395	0.11-0.05	16.0	44.2	73.4	79.2
12	0.0635	25	49-50	98-99	197-198	394-397	0.07-0.03	17.3	55.7	111.9	126.0

Table 3. Effects of 20 Feet of Two-Conductor Wire and 1¼-Amp Slo-Blo 3AG Fuse Connected to a 4-Ohm Load

Wire Gauge	Total Series Resistance, ohms	Available Power Into 4-Ohm Load (Watts) Out of a Possible:					power Loss, dB	Effective Damping Factor for an Amplifier With Rated D.F. Into 4 Ohms of:			
		25	50	100	200	400		20	100	1000	
22	0.931	16-20	33-41	66-81	132-162	263-324	1.82-0.91	3.5	4.1	4.3	4.3
20	0.691	18-21	36-43	73-85	145-171	291-341	1.38-0.69	4.5	5.6	5.8	5.8
18	0.540	19-22	39-44	78-88	155-176	310-352	0.88-0.44	5.4	6.9	7.4	7.4
16	0.446	20-22	40-45	81-90	162-180	324-360	0.92-0.46	6.2	8.2	8.9	9.0
14	0.386	21-23	42-46	83-91	166-182	333-365	0.80-0.40	6.8	9.4	10.3	10.4
12	0.348	21-23	42-46	85-92	169-184	339-368	0:72-0.36	7.3	10.3	11.4	11.5

Table 4. Effects of 20 Feet of Two-Conductor Wire and 1¼-Amp Slo-Blo 3AG Fuse Connected to an 8-Ohm Load

Wire Gauge	Total Series Resistance, ohms	Available Power Into 8-Ohm Load (Watts) Out of a Possible:					Power Loss, dB	Effective Damping Factor for an Amplifier With Rated D.F. Into 8 Ohms of:			
		25	50	100	200	400		20	100	1000	
22	0.931	20-22	40-45	80-90	160-179	321-358	0.96-0.48	6.0	7.9	8.5	8.6
20	0.691	21-23	42-46	85-92	170-184	339-368	0.72-0.36	7.3	10.4	11.4	11.6
18	0.540	22-23	44-47	88-94	176-187	351-375	0.57-0.28	8.5	12.9	14.6	14.8
16	0.446	22-24	45-47	90-95	179-189	359-379	0.47-0.24	9.5	15.2	17.6	17.9
14	0.386	23-24	46-48	91-95	182-191	364-382	0.41-0.20	10.2	17.2	20.3	20.7
12	0.348	23-24	46-48	92-96	184-192	367-383	0.37-0.18	10.7	18.7	22.5	23.0