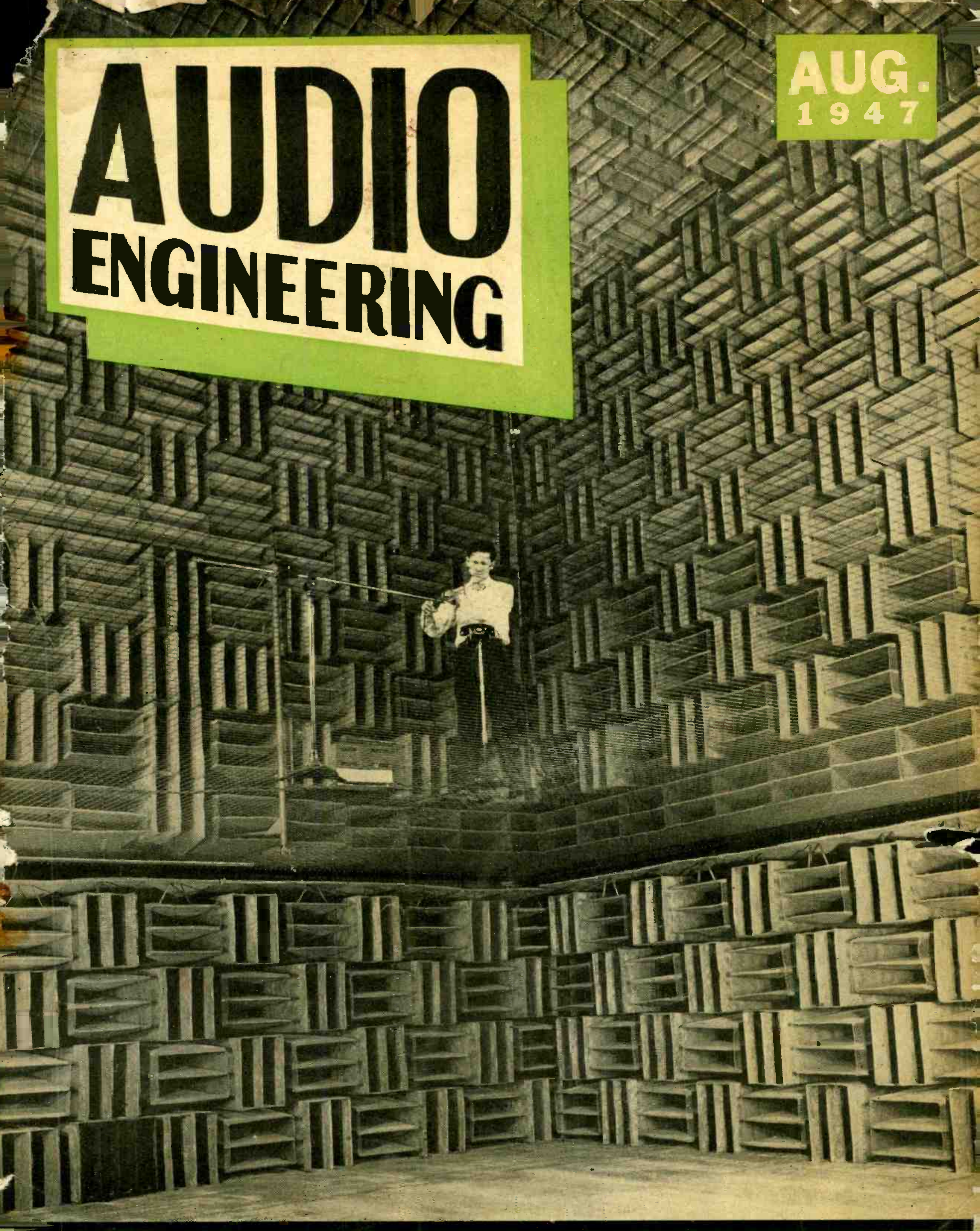
AUDIO ENGINEERING

AUG. 1947



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CONTENTS AUGUST, 1 1947

COVER ILLUSTRATION

Making a ringer test in the soundproof room at Bell Telephone Labs at Murray Hill, N. J.

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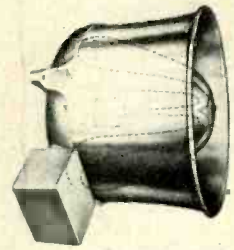
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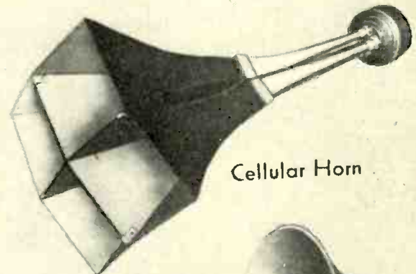
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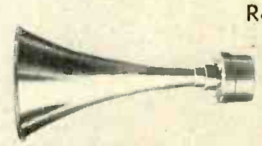
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Letters

The Big Wind

Reverberations from the impact of S. Young White's extraordinary article, *Too Much Audio,* in the May issue of this magazine are still being felt. The high-power turbojet generator, driven by compressed air, seems to have applications in disturbing the amours of pigeons and regimenting the peregrinations of salmon, in addition to those already discussed.

From the salmon-tender:

Dear Mr. White:

We have examined your paper in the May issue of AUDIO ENGINEERING titled "Too Much Audio", with considerable interest. In the Time magazine write up of your activities, it was mentioned that you had succeeded in getting guppies to line up around a sound pattern. We would like to know how and with what equipment you did it, and also if the experiment could be duplicated on a large scale with other fish. One of the major problems in connection with the passage of anadromous fishes at a large dam such as Bonneville Dam on the Columbia River, is getting the downstream migrants, or fingerlings, safely past the structure. It has been established from various experiments that considerable mortality occurs in the spillways and turbines. This mortality is known to be near 15 per cent at that particular dam. The Corps of Engineers plans to construct several additional dams, the most recently approved being McNary Dam at Umatilla, some distance above Bonneville. It does not require a mathematician to point out that a 15 per cent loss at each of six to ten dams equal in hazard value to Bonneville would mean the virtual extermination of a major portion of the salmon in the Columbia. McNary Dam and the other proposed dams will be over 50 per cent higher than Bonneville. Add to this the cumulative effects on the migrant fingerlings of passing several dams and you have a fair picture of the threat to the $10,000,000 Columbia River salmon runs as of the present.

Mechanically screening a river as large as the Columbia is a practical impossibility. At the present time we are experimenting with a system of lights which we hope may guide small fish into safe channels past the dam. The recent development of sonar with its potentialities has given up the hope that we may discover a sound frequency that will either frighten or attract fish into safe channels. There has been a great deal of chit-chat about the effect of various frequencies upon various fish and mammals, but little real scientific work has been done. We know that 20 kc to 30 kc does not frighten schools of fish. We know that small fish and frogs have been killed by subjection to 400 kc. Apparently an extremely low frequency or one extremely high would work as a fish "scare". We do not know that, nor have we been successful in obtaining equipment to find out. None of the commercial sound people in this region have equipment for our needs.

What we need, at least for our initial experiments, is an instrument capable of producing a wide range of frequencies to frighten (or attract) small fish in a hatchery pond. In addition we need equipment that will measure both the frequency of the waves being emitted, and their intensity.

Any information or suggestions that you may be able to offer would be greatly appreciated.

Very truly yours,

J. T. Barnaby,
In Charge North Pacific Fishery Investigations,
U. S. Dept. of the Interior

From the Pigeon-Hater:

Dear Mr. White,

I hope I won't trouble you with my peculiar interest in supersonic devices. I have one in mind that I'd like to experiment with. I can name it, but I can't build it. It's a "Wigeon Phistle".

We have a tiny garden here at our apartment on West 54th Street, and the blankety-blank pigeons keep us jumping from dawn to dusk, or they would destroy our flowers. Even a few minutes of relaxed vigilance, and they've eaten up our petunias. What they don't chew up, they tramp on. You can hear 'em stomping up and down on the flowers, just for meanness.

Isn't there something electronic, in a frequency so high as to be inaudible to humans, that would sound somewhat like the whistle on the "Queen Elizabeth" to pigeons, at close range? Would it be possible to have or operate such a device, by plugging it into house current? We do have DC, which is bad; for I presume the lack of alternating current stops us.

I've wished for an air-gun, quick and ruthless poison, sudden death and destruction for pigeons, but I'd rather have a pigeon whistle, if it's possible to make one. I would appreciate your comment. In New York, many would buy "Wigeon Phistles" for that early-and-late mooing and cooing, flying and flapping, all the day long. To say nothing of saving flowers.

Sincerely yours,

Walter H. Brooks
34 W. 54th St.
New York City.

Dear Mr. Birdlover:

It is true that my little 20,000 watt unit would be distinctly unpleasant to even the flying rocs you mistake for pigeons. In fact, we give a factory guarantee it will stunt their growth for seven generations both forward and backward, a claim no one of our non-existent competitors can possibly make.

The writer is that disgusting breed known as a "famed Government scientist", so his opinion is infallible. "Savant" will be accepted as passable, if you have a perfect Harvard French accent. Brace yourself, brother, here comes the bad news.

You need about five watts at 24,000 cycles—actual energy in the air. You have your choice of two ways that it cannot be done. My gimmick merely requires a fifty-shaft horsepower compressor to run it—one of those cute machines you see parked on the street to supply air to those noiseless tools that break up the pavement.

One way is a magnetostriction oscillator at 3% efficiency, so you need a half kilowatt broadcasting station to run it. You also water cool it, and the nickel rods break every half-hour.

The other way is a Hartmann whistle—also 3% efficient, run from a quarter-horse compressor. After listening to the compressor for ten minutes you will be glad to have the pigeons back.

There is still another way—if you have steam heat, run the whistle on, or off, steam. This either means roasting in summer, which is a minor discomfort compared to your present agony, or luring them back in the winter so you can delure them with the whistle. A type z16-997 atomic power plant will warm your garden so you will have wintertime flowers to save from the wintertime pigeons.

Invite me in for a drink sometime, and when I see the premises I might be able to figure out nine more ways it cannot be done. I shall tell you in pidgin English, of course.

Thanx for a charming letter.

S. Young White
202-09 43rd Ave.,
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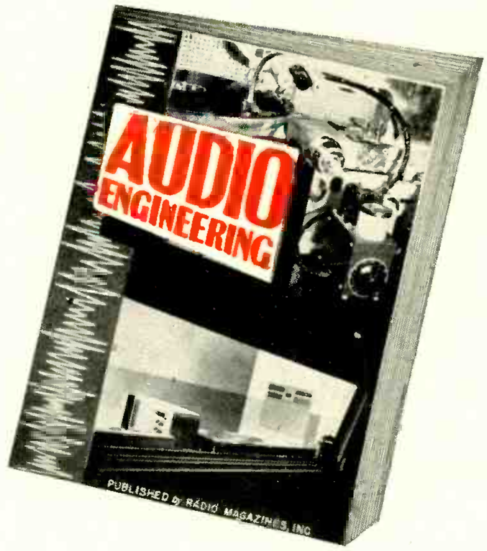

Each issue covers 5 or more of these subjects

1—Broadcasting

2—Transmitter and receiver manufacturing

3—Sound-on-film equipment

4—Recording (disc, wire and tape)

5—Public address

6—Industrial sound equipment and applications

7—Acoustic treatment of studios, rooms, auditoriums, etc.

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Ass't Director of Technical Development
National Broadcasting Company

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EDITOR'S REPORT

LISTENER PREFERENCE TESTS

• For several weeks we have had on hand four excellent articles on listener preference tests, one of the best of which is presented in this issue. In addition, many others have written in, offering to submit articles on this subject. Under the circumstances, please, if you feel like writing on this subject, confine your comments to a letter. We do want contributions, though, particularly on recording and p.a., so if there is anything along these lines which you feel would be interesting to others, let's hear from you.

PITCH

• In discussing high-fidelity reproduction of orchestral music, our standard is usually the original performance of the orchestra. Highest fidelity is achieved when the listener in his home feels that the is actually seated in the concert hall. But our real standard should be the composer's original score, which the conductor of the orchestra reads. If the music is to sound as the composer intended it, then the performance of the orchestra must faithfully follow the composer's score.

No orchestral rendition can sound as the composer chose unless it is played at the pitch for which it was written. Because standards of pitch now used differ from those in effect during the early part of the 19th century, the works of Beethoven and other classical composers are no longer played at the proper pitch. In 1813, the standard pitch for middle "A" was 1847 vibrations per second, corresponding to 423.5 cycles per second in our terminology. Some conductors started to use slightly higher pitches and gradually this practice spread. By 1846, the London Philharmonic had raised their "A" to 425.5 cycles. In this country, the pitch was often even higher, wobbling with the heat, the humidity, and the disposition of the oboeist (who sounds "A" in all orchestras).

Thus the international standard pitch of "A" = 435 cycles, at 59° F. which was established in France in 1859, was usually ignored by orchestras. Even when the orchestra pitch was fixed at 440 cycles, many orchestras in Russia, England, and in this country used 445 cycles and more. Finally, in 1945, the French proposed to raise the international standard pitch to 440 cycles provided those who were using an orchestral pitch of 445 cycles or higher would come down to 440. Up to the present time, nothing seems to have come from this proposal. Many conductors feel that higher pitches make for more "brilliant" renditions.

Under the circumstances, it is apparent that those who listen to classical orchestral compositions of the early 19th century cannot hear them as the composer desired. If the change of pitch makes so much difference as the trend seems to indicate, then it would be well worth while for broadcast and recording studios to require that such classics be played in the pitch used by the composer. If programs were selected so that only music of the same period was presented at the same time, it would be unnecessary to change the pitch until the whole program was changed.

This is a small step toward more faithful reproduction, but it is one which is easily taken. It may seem inconsistent to advocate abandonment of standardization of a pitch frequency, when the broadcasting and recording industries are so greatly in need of fixed standards, but most certainly we should not sacrifice valuable esthetic qualities for the sake of pitch-frequency standardization. And there is no good reason why the standard orchestra pitch should not be retained for all present-day musical compositions. It would be most interesting to see how this works out.

NAB CONVENTION

• The National Association of Broadcasters will hold their annual Convention and Exposition in Atlantic City, N. J., September 15th to 18th, inclusive. There will be many interesting exhibits and technical papers presented. Our organization has taken a booth, and we hope to see you there.

—J. H. P.

Ever wish you were Aladdin?

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Psycho-Acoustical Aspects of Listener Preference Tests

C. J. LeBEL*

This authoritative discussion of the factors involved in listener preference tests merits careful study by all radio/electronic engineers.

THERE COULD be no better way of opening a discussion of this subject than by an extremely relevant quotation from Lewis Carroll:

"'Twas brillig, and the slithy toves
Did gyre and gimble in the wabe;
All mimsy were the borogoves,
And the mome raths outgrabe."

Even the most extended study will show the above to be completely explicit, viz.,

"All mimsy were the borogoves.
And the mome raths outgrabe."

However, all the researches of the past fifty years have failed to yield a usable interpretation of these esoteric remarks. Perhaps Carroll was just an early radio set advertisement writer.

Unfortunately, the basic problem of much audio research is similar in nature. Almost anyone can make a test and get consistent results, but an engineer of long experience may find real difficulty in defining the scope and validity of the result. More scientifically put, consistency is a necessary, but not the sole, requirement for accuracy.

Applying this philosophy to the tests which have recently been described[1,3,4,5,6,17] and the organizations sponsoring them, we cannot doubt that their *observations* are both consistent and correct. We need not, and in fact cannot necessarily accept all the *conclusions* which their authors and critics have drawn.

One quirk of psychology has fixed nearly all these tests in a common groove: The engineer continues a test only until he (whether debunker or idealist) has secured what he considers a valid result. Since most experimenters have secured results which fitted in well with their previous ideas, the work has generally stopped with the result only partly defined. As Bolt has jokingly said, "If the first experiment checks theory, don't repeat the experiment."

Tests

Three methods of test of listener preference exist:

In the *direct acoustic* test[1] the listener hears live music in the adjacent room. The two rooms are joined by an acoustical system of adjustable properties. In the test reported, the listeners had a choice of unrestricted transmission and of a 5-kc. low-pass system.

Using popular music, over 1000 listeners voted

Prefer full range 69%
Prefer 5-kc. range 31%

Using semi-classical music, about 200 listeners voted

Prefer full range 66%
Prefer 5-kc. range 34%

Using speech, no less preference was shown for the full range.

While no test of the hearing acuity of listeners was made, the writer's experience in his own tests has been that 1200 observers chosen at random are likely to be a satisfactorily representative sample. Deviations from the results with a group of 10,000 observers are small, less than the inherent deviation due to uncontrollable variation in test conditions.

Due to the small room necessarily used to approximate home acoustics, the musicians had to play at an exceptionally low level. That the resulting tone and balance were correct was checked and approved by a number of musical directors of long experience. Playing an instrument at lower than normal intensity will reduce the harmonic content, but the resulting sound level, as reported by Olson, ranged from 70 to 80 db — definitely louder than optimum. This would tend to increase the effect of harmonics present. It is evident from the musical directors' verdict that the two effects were small enough to cancel one another out.

The results were well documented, with virtually every technical question well answered. This Olson paper should be a model to all for its completeness of data.

---

*370 Riverside Drive, New York City 25.

In the *indirect* method we may examine the acoustical characteristics of musical instruments in common use, remembering Burris-Meyer's saying: "Man has had a long time to throw away instruments of displeasing tone."

If we examine results of a test by Snow[2], we find that the highest frequency whose absence could be detected by listeners 80% of time, and the limit of musical tone were:

	80% Limit	Tone Limit
snare drum	14 kc.	14 kc.
violin	14 kc.	9 kc.
flute	14 kc.	9 kc.
cymbals	12 kc.	12 kc.
soprano saxophone	12 kc.	11 kc.
oboe	12 kc.	12 kc.
piccolo	9 kc.	10 kc.
trumpet	8 kc.	8 kc.
clarinet	8 kc.	10 kc.

It is evident that many popular instruments have readily audible musical tone output in the frequency range in dispute, yet remain popular.

In the *electronic* method we use an electroacoustic reproducing system of adjustable properties. In some cases a live program was the source[3,5]; in other cases, lacquer originals[3], also transcriptions[5], and phonograph records[6].

Results secured by Chinn and Eisenberg[3,4] may be indicated by the following characteristic figures retabulated from their curves and data for average listeners; the figures given are percentages:

Test Pair	Range, cycles	Music: Classical	Music: Popular	Speech: Male	Speech: Mixed
1	(60- 8000	19	33	52	34
	(120- 5000	38	26	25	34
	(No preference	(43)	(41)	(23)	(32)
2	(20-10000	12	39	21	15
	(60- 8000	67	22	55	64
	(No preference	(21)	(39)	(24)	(21)
3	(20-10000	15	33	24	23
	(120- 5000	58	34	48	45
	(No preference	(27)	(33)	(28)	(32)
Number of observers		105	123	105	123

In the ensuing discussion[4], it was brought out that, with experience, listeners may overcome this initial dislike and discover new enjoyment in wide range. Their data[3] shows:

Test	Range, cycles	Classical Music: Average Listeners	Classical Music: F.M. Listeners
1	(60- 8000	19	31
Pair	(120- 5000	38	28
	(No preference	(43)	(41)
2	(20-10000	12	16
	(60- 8000	67	61
	(No preference	(21)	(23)
3	(20-10000	15	28
	(120- 5000	58	59
	(No preference	(27)	(13)
Number of observers		105	96

The effect of listening to wider band programs is apparent at a glance in test pairs 1 and 3 above, as the percentage preferring greater bandwidth is nearly doubled. It is also covered in the discussion by Bryan Groom[4], an amplification of incidental remarks made in the past by a number of authors in *Wireless World* and *Wireless Engineer* articles on English broadcasting systems.

It is interesting to note that the musicians had a hearty and completely unmistakable dislike for the system at maximum bandwidth *for music only* (especially see Test Pair 3):

Test Pair	Range, Cycles	Classical Music: Average Listener	Classical Music: Musician	Male Speech: Average Listener	Male Speech: Musician
1	(60- 8000	19	20	52	62
	(120- 5000	38	28	25	10
	(No preference	(43)	(52)	(23)	(28)
2	(20-10000	12	7	21	25
	(60- 8000	67	83	55	48
	(No preference	(21)	(10)	(24)	(27)
3	(20-10000	15	5	24	48
	(120- 5000	58	73	48	40
	(No preference	(27)	(22)	(28)	(12)
Number of observers		105	20	105	20

It is evident from these data that the musicians had a much greater desire than had the average listener for full bandwidth on *speech*, but very obviously certain characteristics of the reproduction of music were such as to repel them even more violently than they repelled the average listener.

Webster and McPeak[6] have reported a number of tests using the electronic method in perhaps oversimplified form. Listeners were given a choice of single channel and dual channel transmission of a live program, and single channel transcription. They were also permitted to vote whether they felt the two-channel program was better than, equal to, or poorer in quality than live music, but no acoustically transmitted live music was offered for immediate comparison. Virtually none of the listeners felt the system was inferior to live music, but we are disposed to be unsure of the full validity of this comparison, for it places too much reliance on untrained memory. The following is characteristic of their *interpretation* of their main test results:

	First Choice, %
Group 6 String Group—	
transcription	44
live single channel	22
live double channel	34

This was summarized as:

transcription	44
live music	56

Statistically, the validity of such a method of summarizing is probably doubtful at best, and it has been challenged elsewhere[24]. Since there was no attempt to vary channel bandwidth, we believe that their results show only that people *can* like a broad-band system. It is certainly not a test to determine optimum bandwidth, as some readers have asserted. We wonder whether the palatability of the broad band might be attributed to their use of a 3-channel speaker, covering the bands 40-225 cps, 225-800 cps, and 800-12000 cps. This would decrease Doppler effect intermodulation, as compared to the two-channel speaker which Chinn and Eisenberg used. Would this be great enough to be of significance?

Bauer[6] has briefly reported on a test using phonograph records of unknown characteristics:

Cutoff Frequency	Non-technical Listeners	Technical Listeners
3000	71	13
5000	21	58
above 5000	8	29

He has very carefully restricted his conclusions to phonograph record listening, as befits the only source material he used. In view of the known poor quality of much record processing and pressing stock for the last several years, we are not disposed to challenge his data. Undoubtedly the use of vinyl pressings or an H. H. Scott noise suppressor would improve the signal-to-noise ratio enough to modify the result appreciably.

Discussion

Three possible reasons may be advanced for user preference of restricted range:

1. High frequency response of voice and musical instruments is excessive.
2. The user has become conditioned to restricted frequency range, by virtue of long listening to radio sets with such characteristics.
3. The distortions produced in the reproducing system may be highly distasteful to the public, and a restricted frequency range will eliminate some distortion and reduce the unpalatability of the remainder.

It is fairly evident that a combination of Olson's tests, Chinn's tests and Snow's data demolishes both 1 and 2. We are left, then, with only the matter of distortion in the reproducing system. In pressing this point we are severely

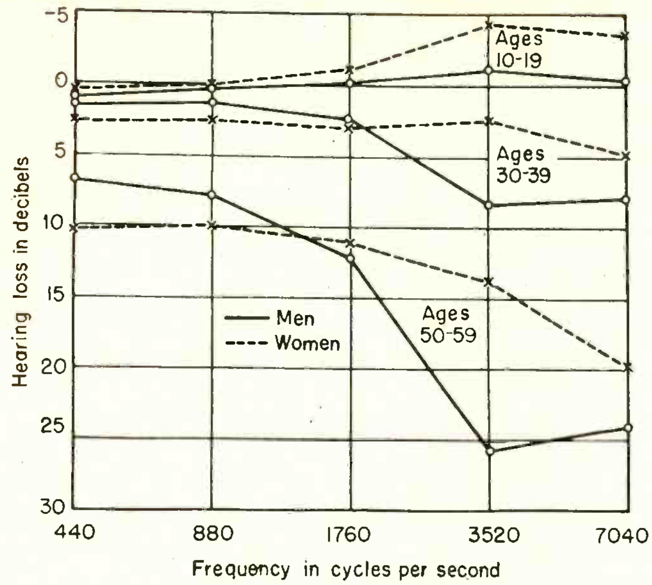


Fig. 1. Mean hearing loss for various age groups, from World's Fair hearing tests (after Steinberg, Montgomery, and Gardner).

(Courtesy Journal of Acoustical Soc. of America)

vey by Beasley the curve is a modal value, and we may expect about half the persons concerned to have losses ranging between zero and the curve, i.e., a loss given as 15 db at 8192 cycles is the most often occurring, but the deviation of the readings is so great that we would find a large number with losses between 0 and 5 db. The World's Fair curves are mean values, and just half of the actual readings would show less loss than the curve. For an idea of these results see *Figure 1*. Note that the mean relative hearing loss at 7040 cycles (with respect to 1760 cycles) is only 6 db for the male and only 1½ db for the female, in the age range 30 to 39 years. This is entirely negligible. Even among the hard of hearing we often find cases with uniform loss over the entire frequency range, i.e., flat response. To say that the public is physiologically incapable of hearing higher frequencies is arrant nonsense.

lacking in experimental data, for only Olson's paper has been satisfactorily documented with information on virtually every aspect of the test. We are given an incomplete frequency response curve by Chinn and Eisenberg, which does not include the loudspeaker response. They give us harmonic distortion data, showing not over 1% over the full band, and not over .5% over the narrow band at 10 db above normal (*i.e.*, at probable peak level), *excluding the loudspeaker*. We are given equally little information on the system characteristics by Webster and McPeak.

Missing Data

It is obvious that no test result will be accepted by a critical world unless the test conditions are rather fully set forth, and that conclusions will be still more suspect unless test conditions are completely given. That this is no light undertaking may be judged when we inspect the areas in one or more of which all writers but Olson have been deficient:

a. Physiological Acoustics
b. Architectural Acoustics
c. Psycho-Acoustics
d. Electroacoustics
e. Psychology
f. Statistics

It must be emphasized in the strongest terms that the lack of much of these data is more a reflection on the low state of the radio art than on the authors criticized. Inspection of the literature will show a period of perhaps twelve years when radio set engineers as a whole paid very little more than lip service to the engineering aspects of making reproduction pleasing to listen tc. Only the organizations in two nonradio corners of the electronic field did any scientific work on the problem (with perhaps three exceptions) and the engineers so engaged were estimated at little over a dozen by one prominent editor. If the Johnny-come-latelies in the quality field make rash statements, and flounder a bit, the graybeards may perhaps be permitted a chuckle or two.

Physiological Acoustics

If a limited number of listeners was used, we are entitled to audiograms of their hearing. If a somewhat larger group, we need proof that it is a true sample of the public hearing[8, 9]. A true sample of the population by age, sex, and occupation is not necessarily a true sample of public hearing. It *might* be questioned whether a random 150 or 200 persons could constitute even a true sample by age, sex, and occupation, for small random samples are not necessarily true samples. The statisticians should get out copies of the Public Health and World's Fair hearing tests and pads of paper, and begin work.

Incidentally, these basic sources of information on American hearing characteristics seem to be badly misunderstood by the average engineer. The curve of hearing for a given age group and sex is a trend, not a universal rule. In the monumental Public Health sur-

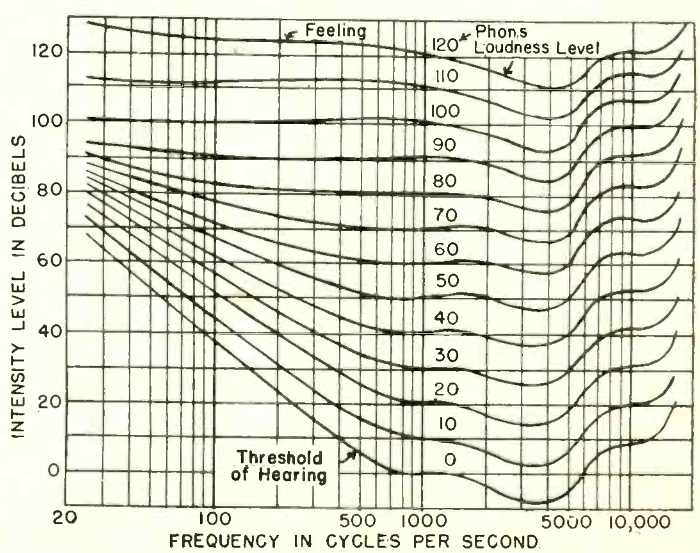
Architectural Acoustics

Studio acoustics and studio microphone placement seem too often designed to emphasize the middle highs, in an attempt to make a $19.95 radio sound as nearly human as possible. If a wide band reproducing system is then used, the result may be much too bright for naturalness or comfort. In Olson's paper we have acoustical data on the room used. In no other study are we given data on the studio acoustics nor on the microphone placement, and no evidence of attempt to match the acoustical conditions to the wide band systems in use.

The acoustics of the listening space also deserve attention. Was an unused studio used, without change? If changed, how close to the acoustically average home was it? The reverberation characteristics of the studio and of the listening space have a profound effect on the balance of tone and on the optimum overall air-to-air curve[13]. A bright, hard

Fig. 2. Contour lines of equal loudness for normal ears (after Fletcher and Munson)
0 db = .000204 bars = 10^{-16} watts per sq. cm.

(Courtesy Journal of Acoustical Society of America)



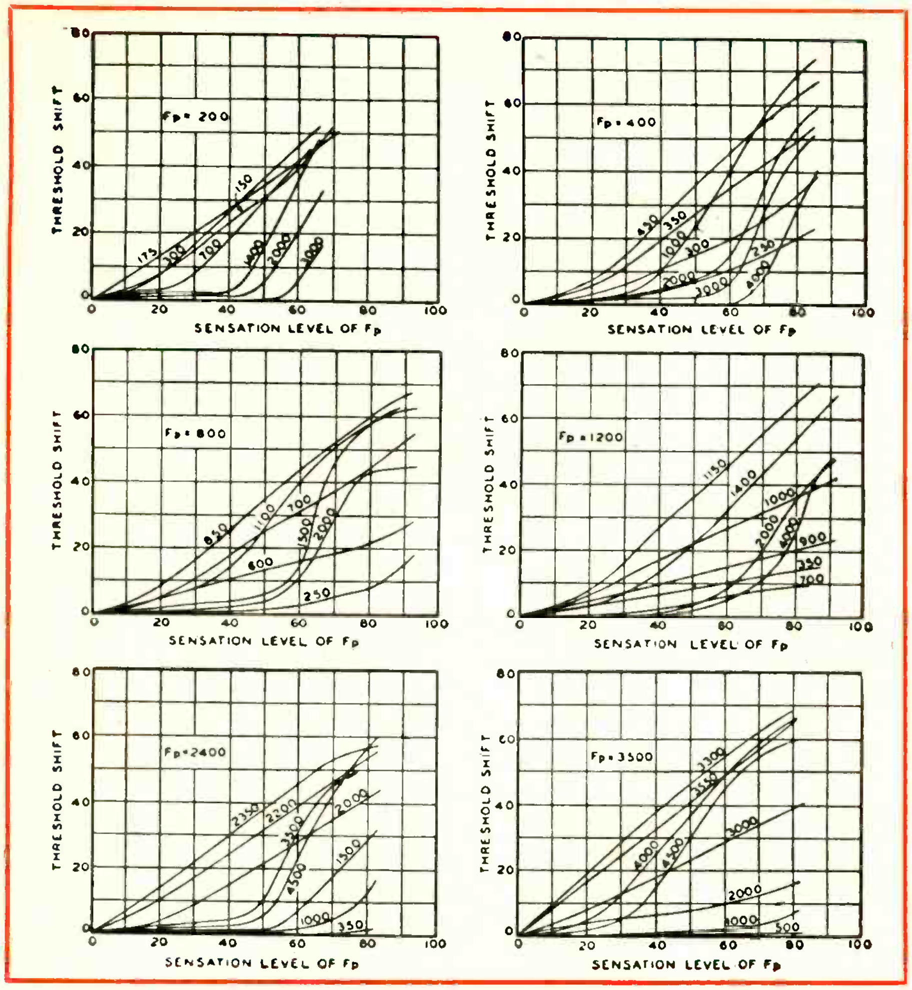


Fig. 3. Curves showing the ability of different frequencies F_p (indicated on each plot) to mask other frequencies (indicated by the numbers attached to the curves). The sensation-level of the masking tone is shown by the abscissa, and the elevation in the threshold of the tone by the ordinate (after Wegel and Lane). (*Courtesy Physical Review*)

listening room will tend to make everything sound shrill, and a dead room will often sound more muffled and boomy than the shape of the reverberation-frequency curve alone would suggest. We must not forget that studios and rooms which experts would pronounce "good" by ordinary broadcasting standards are not necessarily correct for *best listening* to wide band systems.

Psycho-acoustics

This brings up the question of scale distortion due to difference between original and reproduced sound levels. Theoretically, the equalizer correction curve could be computed from the appropriate equal loudness contours (see *Fig. 2*); practically, it is dependent also on the acoustical conditions in each enclosure[13] and should be experimentally determined. Lacking such a correction, at usual music volume-differences the high-frequency response would have to be sloped off to restore a good tonal balance. Lacking a sloping network, the cutoff filters would be used—which is just what was done.

It is desirable to remind the reader briefly that choice of selection and choice of orchestration can have a profound effect on the result. Certain treatments are damaged much less by decreased bandwidth than are others. Was this perhaps the reason why the preference for increased bandwidth was proportionately greater with popular music than with classical music in the Chinn and Eisenberg tests?

There is a point on which many engineers are badly mistaken, and this has even crept into some articles: belief that room noise will mask the higher frequencies due to its intensity. Since room noise may exceed 50 db, this would be a valid point *if* true. Actually, masking is a well-studied specific fact, not a general bogey. Studies[10] have shown that very great low-frequency energy is necessary to mask a high-frequency tone, as is shown in *Figure 3*. For example, a masking tone of 400 cycles will not shift the threshold of hearing of a 4000 cycle tone until the 400 cycle level has reached 60 db. This could be rephrased to say that the masking effect of a 400 cycle tone on a 4000 cycle tone would be 60 db down from the level of the 400 cycle tone. Since the average of room noise[15] in a home is 43 db (with a standard deviation of 5.5 db) we may consider the effect negligible most of the time, since the noise energy drops very rapidly above 800 cycles[16]. This has been discussed in detail by Fletcher[25].

There is another point on which many engineers are in error, of which the following comment on the acoustical filter method of test is representative: "*However,* the results would apply to binaural listening and not to monaural listening, which is common practice for present-day broadcasting . . ." Practically all listening is binaural (physiologically speaking), regardless of whether it is to a live orchestra or to a loudspeaker, so we are sure the authors quoted meant "two channel" and "single channel" transmission, but we still disagree with their remarks. There is nothing in the literature and no reason whatever to believe (in testing a system with given electroacoustic characteristics) that the number of channels in use will affect the choice of *optimum* bandwidth. This was emphasized by Maxfield in the discussion after the Olson paper.

Electroacoustics

In no electroacoustic system test has the overall air-to-air characteristic been published, and we can only guess. In one case the loudspeaker is a "dual unit of well-known manufacture", in another "an expensive loudspeaker of well-known manufacture"; in another test a radio set "designed by good engineers" was employed. In a fourth instance, phonograph records of "well-known manufacturers" were employed. These are exceedingly weak links in chains of already doubtful strength. Without an air-to-air test how can we know what sort of system we are using?

There are many phases of electroacoustics where our instrumentation is currently weak. In some cases, the ear is a much more strong indication of fault than any instrument normally available. Here are cases of such faults having a possible effect on results:

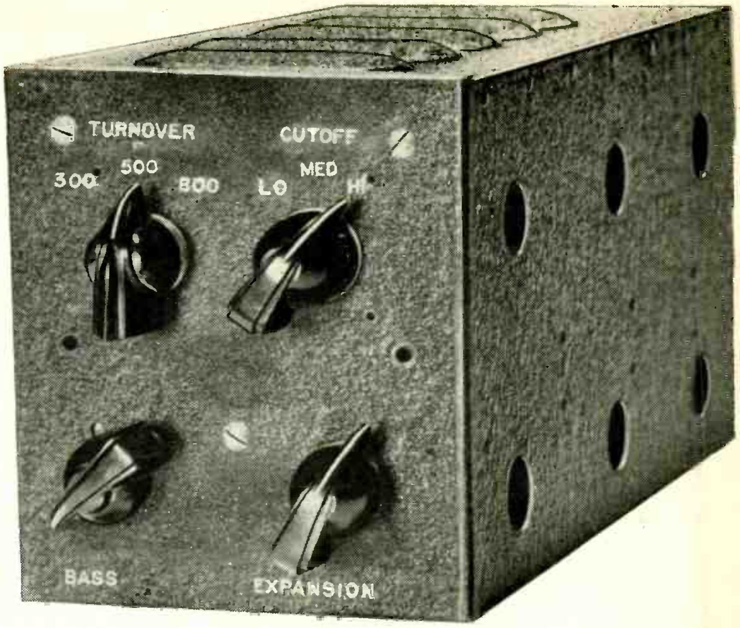
Five models of microphone might normally be used to pick up the original program. Nominally not too different in free field response curve, actually one of them has a rather harsh unpleasant sound, two of the others a velvety smoothness denoting good transient response.

Three types of disc recording heads are most widely installed, with nominally not too dissimilar response curves. One type sounds good, the others definitely harsh, due to poor transient response. The large record manufacturers often have their own special cutting heads, each with its own characteristics.

How about the pickups? One widely used pickup has a definitely characteristic tone which colors everything

[*Continued on page* 44]

Experimental Volume Expander and Scratch Suppressor



C. G. McPROUD*

Utilizing a number of familiar circuits combined in one unit to provide the functions of expansion and needle-scratch suppression.

With the advent of new magnetic phonograph pickups with an extended high-frequency response, many record enthusiasts are beginning to notice that there are things on their records other than music. Although the smoother h-f response characteristics make the scratch noise less offensive than is the case with a pickup which has peaks throughout its reproducing range, some records still have a noise level which is too high for comfortable reproduction.

To date, several noise suppressors have been developed, but the cost of the better units prohibits their wide acceptance by the average listener. The circuit described herein is an adaptation of one employed in some of the models of the Scott Radio (not to be confused with the suppressor developed by H. H. Scott), and while far from perfect, it does offer some relief from records having high scratch levels.

Since some of the requirements for a scratch suppressor are duplicated by a volume expander, the additional components for the latter circuit are comparatively few, and were also built into the amplifier, making a combination device which offers interesting possibilities. Being assembled largely from resistors, it is not expensive to build, and it is sufficiently flexible that the individual user can arrive at any effect desired by experimentation with component values.

This description should not be construed as an argument for expanders. The writer happens to be of the opinion that about 6 db of expansion is desirable, adding a degree of lifelike "expression" which is worth while. Many reputable engineers, however, look askance at expanders, offering myriads of reasons why they are objectionable. Therefore, if the listener prefers this unit without the expander, it may be omitted from the construction without disturbing the operation of the scratch suppressor section.

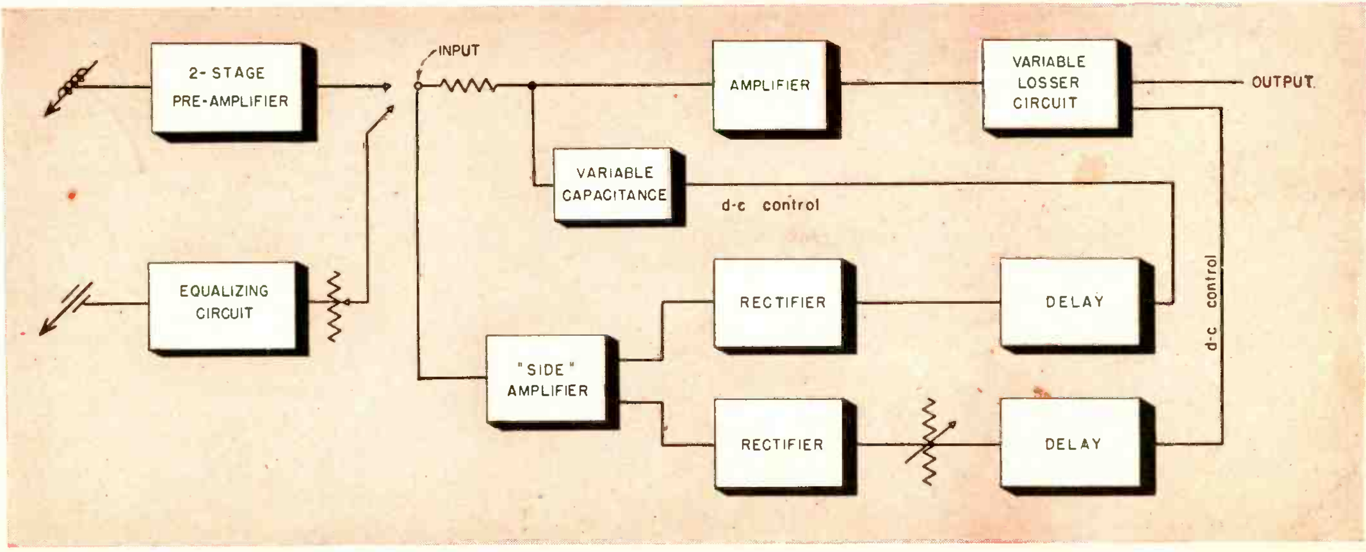
Basic Circuit

The block schematic of *Figure 1* shows the general arrangement of the circuit elements. As shown, a two-stage preamplifier precedes the expander-scratch suppressor unit in order to bring the output of a low-level magnetic pickup up to that required for operation of the controlling circuits. This preamplifier follows the design previously described* and contains the necessary equalization. The output is fed to two circuits — one being the signal channel, and the other the "side" amplifier.

*Managing Editor, Audio Engineering

*Audio Engineering, July, 1947

Fig. 1. Block schematic showing arrangement of components for combination scratch suppressor and expander, fed from either a magnetic pickup with a preamplifier, or from a crystal pickup and a potentiometer volume control.



The signal channel is isolated from the preamplifier circuit in order that the effect of the scratch suppressor shall not be fed back to the side amplifier. Thus, the side amplifier is fed from the input circuit directly, while the signal channel is fed through a resistance network.

The side amplifier output is fed to two rectifiers, and the d-c output of one is fed through a delay network to a reactance tube. A controllable amount of the d-c voltage from the other rectifier is fed through a second delay network to a variable losser tube which provides the expansion.

Referring to the actual schematic of the circuit, *Figure 2*, it is seen that the signal channel consists of a dual triode, one half serving as an amplifier, and the other as the losser tube. The output of the amplifier is coupled to a 0.1-megohm resistor in series with the plate resistance of the losser. The signal output is fed from the junction of the plate and the series resistor. At low signal levels, the bias on the losser tube is at a minimum, and the plate resistance is of the order of 10,000 ohms. This provides a loss of approximately 20 db, which results in a net gain of about 5 db from the input to output of the signal channel. As the bias on the losser tube is increased, resulting in an increase in plate resistance, a greater percentage of the signal voltage is available at the output.

This arrangement is essentially a potentiometer, with R_6 and R_p being the total resistance. R_p being variable, the output signal is dependent upon the ratio of $R_p : (R_6 + R_p)$. The choice of this circuit for the expander is based on two parameters: no variable-mu elements are employed, so that distortion due to variation in plate voltage is eliminated, and a simpler rectifier circuit is used since the required voltage is negative in polarity for increase in output.

Signal Channel

The grid of V_{1a} is fed from the junction of R_1 and R_2, providing isolation from the input signal for application of the reactance-tube circuit. V_{1a} is biased by a cathode resistor, and uses a plate resistor of 0.1 megohms. The plate of V_{1b} is fed through a separate resistor, R_5, in order to avoid a variation in the d-c potential available at the plate of V_{1a}. C_2 provides signal coupling to the "potentiometer," and C_3 feeds the output circuit.

Coupled to the grid of V_{1a} through a 0.01-μf capacitor is the reactance tube, V_2. This tube acts as a variable capacitor controlled by the bias voltage applied to it. Normal bias is furnished by a tap on the potentiometer, R_{10}, which is adjusted for the optimum value. The capacitor C_5, together with the grid-to-plate capacitance of the tube is reflected across the signal circuit, multiplied by the gain of the tube. In effect, therefore, the reactance tube is a high-frequency tone control which is turned so as to suppress the highs when no signal is present. Whenever any high-frequency signal is present, the bias on the reactance tube is increased, and the "tone control" is turned back to normal. The frequency-selective side amplifier provides the variable bias.

The side amplifier consists of a 7E7 as a pentode amplifier tube, with the plate feeding the two diodes through separate circuits. For the reactance-tube control, the coupling capacitor is small, resulting in the development of a d-c control voltage only when high-frequency components are present in the input signal. The entire d-c voltage developed across the diode load resistor R_{13} is fed through the delay circuit consisting of R_{12} and C_7 to the grid leak of the reactance tube.

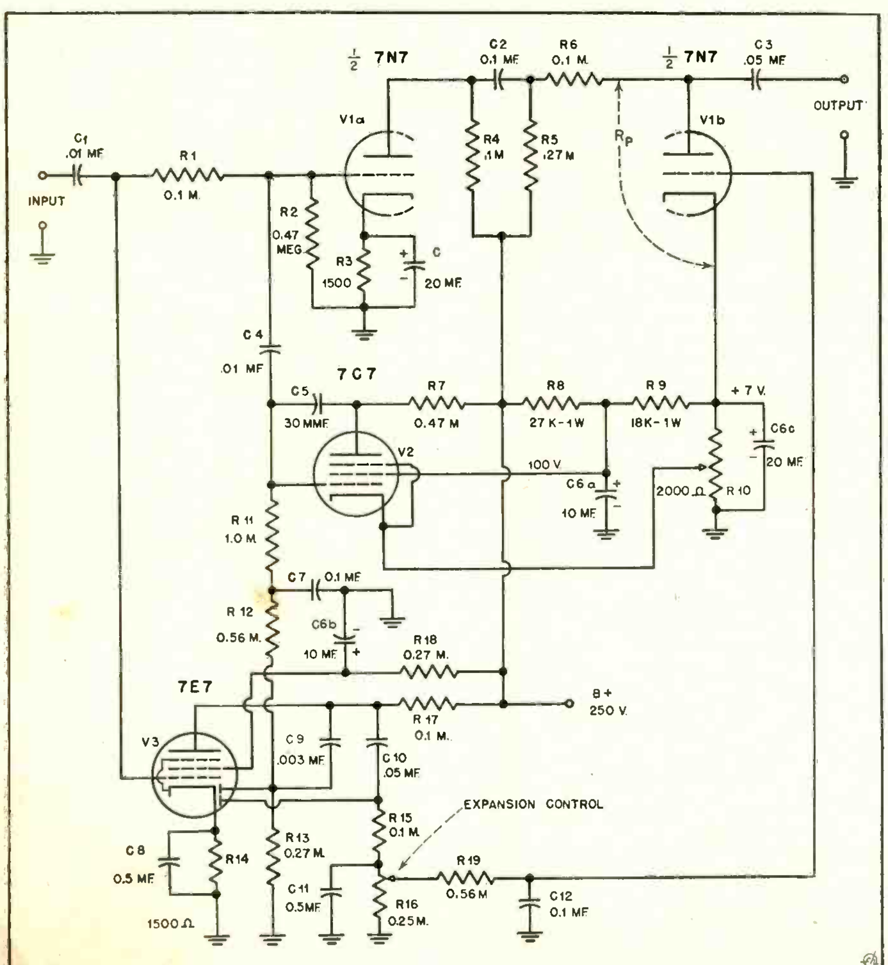
The coupling capacitor feeding the second diode is larger, resulting in a developed d-c voltage which is proportional to the *total* signal. The diode load consists of R_{15} and R_{16}, the latter being a potentiometer used for the expansion control. The delay circuit for this diode consists of C_{11} across the potentiometer for the release time control, and the network consisting of R_{19} and C_{12} controlling the attack time.

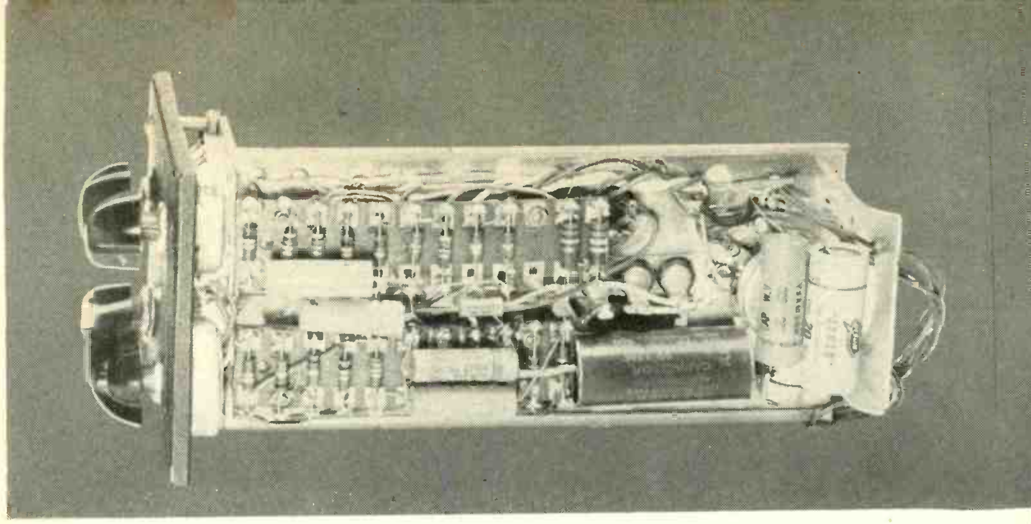
Performance

The principal object of the scratch-suppressor circuit is to reduce the high-frequency response when there is no appreciable amount of high-frequency signal in the input circuit. As constructed with the constants shown in *Figure 2*, the frequency response of the entire circuit is shown in *Figure 3*, plotted with respect to the input signal voltage. With signal intensities over 0.4 volts at the input, the response is essentially flat. However, as the signal intensity is reduced below 0.4 volts, the reactance tube becomes effective, and the response is altered as shown for the curves of 0.3 and 0.2 volts. Below 0.2 volts, the response remains approximately as shown for this value.

These curves were measured with signal intensities as indicated for the entire frequency band. A second series of measurements were made with a 1,000-cps signal of the same intensity applied to the input together with the the high-frequency signal to determine if the presence of low-frequency components was able to supply sufficient

Fig. 2. Schematic of expander-scratch suppressor unit. Two-stage preamplifier used with magnetic pickups is not included in this circuit.





Top and bottom chassis views of completed preamplifier-expander-scratch suppressor.

bias to the reactance tube to prevent its operation. With the component values shown, the 1,000-cps signal required to cut off the reactance tube was 15 db above the applied high-frequency signal.

Figure 4 shows the input vs. output with the expander off, and with the control turned full on. This does not provide a large amount of expansion, but the distortion is relatively low, and the addition of 6 db is entirely sufficient for home use. The maximum value of 7 db is obtained at approximately 10 db above the threshold, and this value continues over the range of the amplifier.

Circuit Adjustments

When constructed in accordance with the schematic of *Figure 2,* the bias applied to the grid of V_{1b} is approximately 7 volts, with the expansion control at zero. C_{12} and R_{19} control the attack time for the expander, set at approximately 60 milliseconds. C_{11} and R_{18} control the release time, the time constant of this section of the network being 125 milliseconds. The total release time is the sum of these two values, or 185 milliseconds. These values are reasonable averages, though a refinement in the design would be an inclusion of a variable control for both of these delays. For optimum results, separate adjustment of both should be available. However, this entails two more controls, and this arrangement attempts to average the delays in order to keep operating controls at a minimum.

R_{10} controls the operating point of the reactance tube. The simplest way to set this control is to feed a 5,000-cps signal to the input, at a level of 0.3 volts. Observe the output level with the arm of R_{10} at the ground end; then adjust the control to obtain a decrease of 10 db in the output.

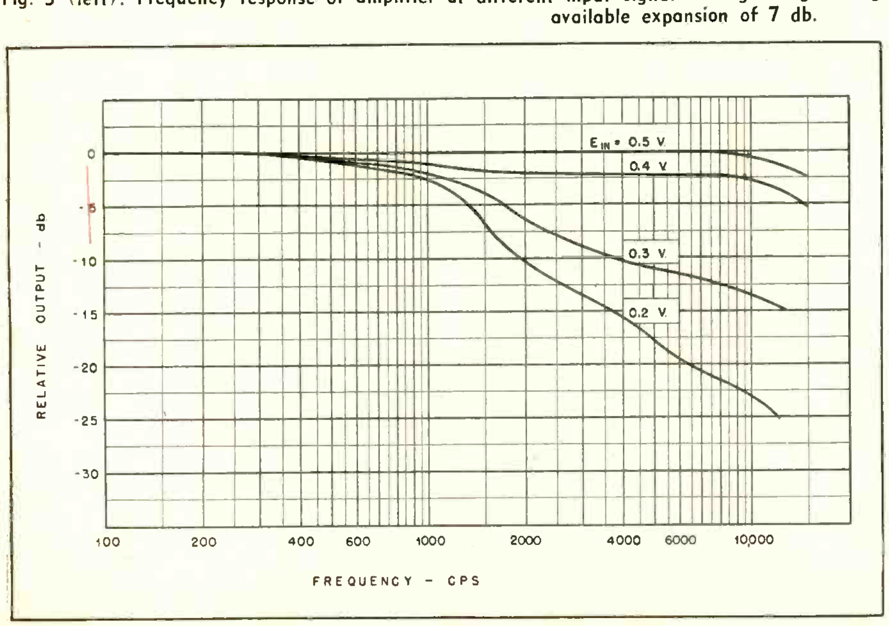
It is necessary that the signal voltage fed to the input be sufficient for the correct operation of the unit. With a two-stage equalized amplifier, the output from the Pickering Cartridge is approximately correct for proper operation of this circuit. The gain of the amplifier must be increased somewhat for the GE pickup, since its output is approximately 15 db below the Pickering. The equalized output from a crystal pickup is too high for the input to this unit, and it is suggested that a volume control be interposed between the pickup output and the input to this amplifier.

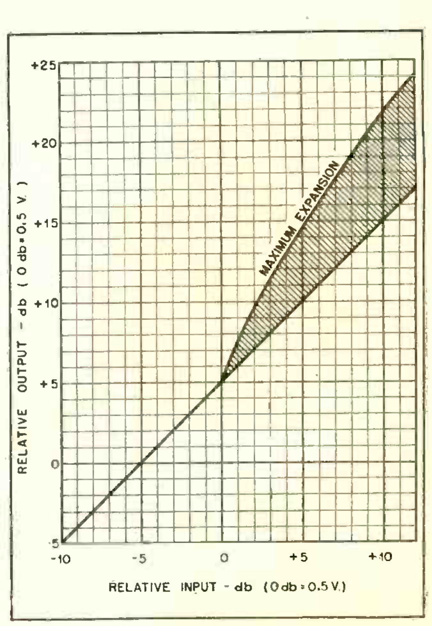
Construction

Obviously, a device of this sort can be constructed on practically any sort of chassis. However, there is a small chassis readily available on the surplus market which was originally intended as a servo amplifier. This chassis is equipped with four lock-in sockets, which governed the selection of lock-in tubes for this particular application. The fourth socket is used for a 7F7 preamplifier stage with the necessary equalization for magnetic pickups. The experimental unit is pictured in three photographs, indicating the tube positions. Note that the shielding enclosure

[*Continued on page* 39]

Fig. 3 (left). Frequency response of amplifier at different input signal voltages. Fig. 4 (right). Input level vs. output level, showing maximum available expansion of 7 db.





Design of Electronic Organs

WINSTON WELLS* PART I

In this series of articles, the author presents a thorough discussion of the design and operation of electronic organs.

"Jesus Maria! I play it so good an' it comes out lika this!"

THUS SPOKE FERDINANDO GERMANI, the celebrated Italian concert organist, upon playing an electronic organ for the first time. This was about a decade ago and, at the time, Germani had just signed a contract to play a series of demonstration recitals for the manufacturer of the instrument.

The maestro was not alone in his dismay, for the performance of the little instrument was a decided letdown to those who were accustomed to the pipe organ at its best. True, it surpassed the finest of pipe organs in dynamic range and rapidity of action. It was compact and readily portable, and its cost was but a small fraction of its nearest non-electronic competitor. But it was lacking in tonal resources, some of the controls were awkward to manipulate quickly and there were other points of design which violated the most sacred traditions of organists and organ builders.

Now it is not to be supposed that, even in this day of wonders, a device so complex as an electronic organ should spring, fully grown and perfect, from the head of its creator, in the fashion of Minerva. Rather, any art must go through a long period of evolution, building a background of trial and error, before approaching the goal set by its pioneers. But an intelligent consideration of the fundamentals upon which the art is based, and an awareness of the objective, can do much to bring the goal closer.

It is the purpose of this article to acquaint the reader with the requirements of good organ design, to discuss the mechanical and electrical structure of modern commercial instruments, and to point out the trend of design in future types.

Design Fundamentals

When we, as engineers, are asked to design a given device, three questions come into our minds:

(1) What is it supposed to do?
(2) How many are to be made at a time?
(3) At what price does it have to sell in order to yield a profit which justifies the undertaking?

Upon the answers to these questions is based the work of the entire project which follows, from the first crude sketches and rule-of-thumb computations to the finished product. The physical structure of the device is then evolved with the objective of satisfying the requirements of performance, production and cost.

Perfection is not ordinarily sought in industrial practice but instead, every effort is made to establish a practical compromise between ideal performance and low production cost. It is the judgment exercised in balancing these two factors which largely determines the ultimate utility of the product.

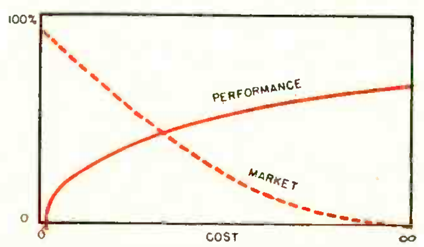


Fig. 1. Cost increases as performance is improved, while market decreases.

It is apparent that the potential sales of a product will increase as the price is lowered, or as the performance is improved. An increase in rate of sales will permit the use of more efficient production methods which, in turn, will lower the cost of manufacture and, consequently, the cost to the consumer. However, it is difficult to make substantial improvements in a product without raising the cost of production.

By means of data obtained through engineering estimates and market surveys, it is possible to plot reasonably accurate graphs for any product, showing the relation between market, performance and cost. The point at which the performance and market curves intersect determines the optimum consumer price and, also, the standard of merit to which the product must be designed. *Fig. 1* shows the manner in which these graphs are applied to a hypothetical product.

Before attempting an analysis and criticism of the current instruments, and before discussing means for improving them, it is necessary that we consider the position the electronic organ occupies in the field of music. While it is inherently capable of some things which are impossible to the pipe organ, its prime function is that of succeeding the pipe organ in the musical world.

The modern pipe organ is the end product of several centuries' development work, and it represents the combined efforts of many men who were outstanding in the fields of music, acoustics and mechanics. The instrument is one of the most functional devices ever to be built by man, its design being shaped by the combination of musical requirements, human anatomy and the mechanical limitations under which the organ builders worked.

The introduction of electronics to the production of musical tones has conquered many of the limitations imposed by the use of pipes and reeds, but it has changed neither the mind nor the anatomy of the human being, nor, in consequence, the organist and his audience. Therefore, we may well examine the pipe organ, and ascertain the reason for its design, before planning an instrument which we arbitrarily call "an electronic organ".

In its modern form, the pipe organ is essentially a sustained tone, multi-keyboard instrument, capable of rendering polyphonic music, and capable of covering the entire frequency range useful in music.

Acoustical Oscillators

The musical tones are produced by organ pipes, which might be described as "acoustical oscillators", the simplest of them resembling the tin and wooden whistles sold in toy shops. The pipes are driven by a regulated air supply, the pressure of which is usually between five and twenty inches, measured on a water manometer. In some cases the pressure may be as high as thirty inches, and a few pipes have been built to operate at one hundred inches.

A separate pipe is used for each note, and for each separate tone color. A set of pipes, all of the same tone color but graded chromatically in pitch, is referred to as a "rank" of pipes, or as a "stop".

*307 E. 44th St., New York City.

Each rank of pipes is mounted on a wind chest, which is supplied with air from the regulator. The receptacle which holds the base of each pipe contains an electro-pneumatic puff valve which, through a series of switches and relays, may be connected to the proper key on one of the organ's manuals or the pedalboard (hand or foot keyboards, respectively).

The common return circuit for the notes of one stop is brought through a switch, located on the console in a position convenient to the organist. The handle of this switch is called a "stop tab", "drawer knob" or "rocker tab", according to the particular form which it takes. Thus, when a stop is "on", the pipes in the rank will speak (play) when their corresponding keys or pedals are depressed. Conversely, when a stop is "off", the pipes of that rank are silent, even though the keys may be depressed.

Very small organs may have only three or four ranks of pipes; the largest may have several hundred. The pipes, along with the more bulky apparatus in the instrument, are housed in an "organ loft", which is located with a view toward obtaining good sound distribution over the area to be covered. The console may be remotely located, provided the organist can hear the sound from the loft above the reverberation and room noise. At very great distances, of course, the time delay effect becomes objectionable . . . the organist may have played several successive notes before he hears the first one!

The Console

The console is the control center of the instrument, and is designed to permit playing and changing of tonal effects with a minimum of bodily movement on the part of the organist. Since the bass passages are played with the feet, the legs, as well as the arms of the organist, are in constant motion, leaving the buttocks as the only means of bodily support. This fact is of considerable importance when deciding upon the location of controls, for they must be so placed that the body will not be thrown off balance while manipulating them.

The pedal board, upon which the bass is played, is so located that its keys are accessible to the organist's feet, as he swings his lower leg from side to side, the keys being arranged in a radial pattern to conform to the arc traced by the foot. There are thirty-two pedals, covering the range of two octaves and a fifth, chromatically, the lower (left hand) end starting with C, and the upper (right hand) end terminating with G. The pedal contacts are usually designed to "make" when the pedal has been depressed about one inch. The pedals which correspond to the "black keys" on a piano, are raised a couple of inches above the others.

The manuals (hand keyboards) are located in front of the organist, and are arranged, one above the other, in "staircase" fashion. The lower one is approximately at elbow height and each successive one is raised by another 2½ inches. The upper ones slope toward the organist, to compensate for the change in attitude of the wrist when playing them.

There are sixty-one keys on each manual, covering a range of five octaves chromatically, starting and ending on C. The exposed portion of a "white key" is 4¼ inches long. They are identical to piano keys, except that the ends facing the organist are undercut. The upper and lower manuals overlap slightly, and this undercut permits the playing of one hand directly below the other, on separate manuals and with minimum interference of motion. The electrical contacts "make" when a key is depressed about one-half inch. The key action is designed to respond to a finger pressure of from 1½ to 4 ounces.

Above the manuals, and to the sides, the stop tabs and other controls are arranged in rows, and are grouped according to their functions. In some consoles, they are mounted upon a semicircular or elliptical panel, so that they may all be reached by one continuous sweep of the organist's arm. (This has proven to be an excellent design, and it might well be used by builders of electronic instruments.)

Modern organ playing necessitates the very rapid changing of registration. ("Registration" is the term applied to the particular tonal effect which is momentarily set up upon the manuals and pedals.) Since a change in registration may easily involve the simultaneous movement of fifty or more stop tabs, it becomes necessary to perform this function mechanically.

The larger pipe organs are equipped with rows of push buttons, called "combination pistons", these being mounted upon narrow panels, under their respective manuals. When a piston is depressed, (usually by means of the knuckles), an electro-pneumatic mechanism is set in motion, which moves the stop tabs according to a pre-arranged registration. These registrations are set up upon a "recorder board" in the rear of the console. There are usually provisions for setting about thirty registrations at one time, although extremes may vary from ten to several hundred.

Upon having depressed and released a piston, the organist may quickly modify the registration thus obtained, by changing a few stops by hand.

Joan Curtis at the Hammond organ.

The organ is not a "touch responsive" instrument. That is to say that, for a given registration, a note will sound exactly the same, regardless of the manner in which the key is struck. Therefore, expression must be obtained by other means. It is standard practice in modern pipe organ design to cover the opening to the organ loft with a set of movable soundproof shutters. These, when closed, attenuate the escaping sound about 30 db in the better instruments.

Expression Control

The "swell shutters", as they are called, are coupled through an electro-pneumatic servo-mechanism to the "swell pedal" (sometimes called "swell shoe" or "expression pedal"). This pedal responds to a forward or backward rocking motion of the foot, the shutters being open when the foot is in the toe forward position. The swell pedal is located so as to be operable by the right foot, since the greater number of the bass notes are played by the left foot.

The pipes of the larger instruments are often divided into several groups, each group being housed in a separate organ loft. In such cases, it is customary to connect the shutters of the separate lofts to individual swell pedals, thus permitting the "fading" of one musical passage with relation to another. By means of a switch, they may also be ganged together electrically.

The utility of the swell mechanism in pipe organs has been somewhat limited, both by the mechanical inertia of the enormous shutters required, and by the difficulty of maintaining a good ratio of sound attenuation between the open and closed positions. From this standpoint, the electronic organ is infinitely superior, since an ordinary bass compensated volume control, connected to a swell pedal, gives all that one could ever wish for in a swell action.

Originally, the swell pedal was limited to the effecting of gentle, gradual changes in loudness of musical passages but, in modern instruments, it is used to obtain marked diminuendos and crescendos, as well as for accenting individual notes in a passage.

The "crescendo pedal" is located to the right of the swell pedal, and is identical in construction. It actuates a progressive multi-contact switch, the contacts of which are connected into the stop actions. As the pedal is "opened", an increasing number of stops are made active until, at the full open position, the organist is using "full organ" (all of the instrument's stops in use). This device is extremely useful in building musical climax, especially on large organs, where there is a fairly even graduation of loudness between successive stops.

Stops

Upon examining a pipe-organ console, it will be noted that the stop tabs are labeled with the name of the instrument or effect which the stop is supposed to represent. This name is followed by a numerical notation, the most common ones being: 16′ 8′, 4′, 2 2/3′ and 2′. These notations refer to the musical "register" pitch of the stop, and are derived in the following manner:

The lowest note on an organ manual is the second octave below middle C (65.41 cps). An open pipe, approximately eight feet long, will produce this note when blown. Hence, any stop whose second octave below middle C coincides with the lowest key on the manual, is termed an eight foot stop.

A stop whose lowest note is three octaves below middle C will need a pipe sixteen feet long, to produce this lowest note. Thus, if this lowest note coincides with the low C on the manual, we are said to have a sixteen foot stop. It will be noted, in this case, that if the the middle C key is depressed, we actually play the octave below middle C. We can best remember this system of notation by keeping the following in mind. When we set up a registration of 8′ stops, the notes which we play on the keys are sounded. When we use 16′ stops, the organ sounds at an octave below the notes which we are playing upon the keys. With 4′ stops, the organ sounds at an octave higher than the notes being played. A 2′ stop will sound two octaves high, a 32′ stop, two octaves low, and a 2 2/3′ stop will sound the twelfth interval (an octave and a fifth) above its nominal position on keyboard.

The same system is used in the notations on pedal stops. However, the lowest note on the pedal board is, nominally, three octaves below middle C, so it is the sixteen, rather than the eight-foot stop, which sounds the note corresponding to the pedal being depressed.

Stops of one register may be played alone, or in combination with those of another. The foundation of organ music depends upon the eight-foot stops on the manuals, with the sixteen-foot stops on the pedals. Those of the lower registers add "body" to a passage, while those of the upper register lend "brilliance."

Couplers

In addition to the stops of various registers, most organs are equipped with "couplers", to which the same form of notation is applied. The control tabs are identical to those used for the stops. When a 16′ coupler is added to a manual, the organ will duplicate, at an octave lower, everything set up in the registration of that manual. Likewise, a 4′ coupler will duplicate the registration at an octave higher. Another control, which is labled "unison off", will silence the original registration, but will allow it to "speak" in the register of any couplers which may be on.

There are also four, eight and sixteen foot inter-manual and manual-to-pedal couplers which, as their names imply, can duplicate the registration of one manual upon another, or upon the pedals. Couplers, when properly used, are of tremendous value in extending the usefulness of a stop, and in making the organ a more flexible instrument.

Another important part of the pipe organ is the "vibrato", sometimes called a "tremolo" or "tremulant". The same terms are also applied to the effect produced by this device.

Vibrato

Vibrato may be defined as a rapid cyclic variation in the pitch of a note, the rate of variation ranging from about 4.5 cps to 12 cps in extreme cases, and the pitch variation ranging from plus to minus 0.5% to 5% of the nominal frequency of the note. We may also define vibrato as the frequency modulation of a musical tone.

As applied to music in general, the frequency and amplitude of vibrato vary among individual performers, and are different for different types of instruments. In the human voice, the rate averages about 6.5 complete beats per second, with a pitch variation of plus to minus 1.3% of the nominal frequency. The violin vibrato averages a little over 7 cps with a pitch variation of about plus to minus 2% of nominal frequency.

While the dominant characteristic of the vibrato is one of frequency modulation, amplitude modulation also occurs in most cases, as well as a whole series of complex phenomena arising from "side band" effects. Since there is always some reverberation in a room, we hear all of the successive pitch variations of the tone and its partials (harmonics included) simultaneously. These beat with each other and their resultants, in turn, beat with them, and so on, ad infinitum.

Vibrato is of profound significance in all sustained tone music, since its introduction into a tone has a direct effect upon the emotions of the listener, regardless of the character of the music being played. The subject should be thoroughly studied by anyone concerned with sustained tone instruments, whether as a performer, composer or designer.

The vibrato is seldom employed in percussive tones, since it softens the attack and gives an effect which is generally displeasing to the ear. An exception is the case of the vibra-harp. This instrument resembles the xylophone. both in structure and in manner of playing. It has a series of tuned cylindrical resonators hung below the resonant bars to bring out the fundamental tone. The necks of these resonators are fitted with motor driven rotating dampers which impart a kind of "amplitude vibrato" to the tone of the instrument, the effect being quite pleasing.

The statements made regarding the vibrato in vocal and instrumental music are generally applicable to the organ. Large pipe organs have several vibratos, these being effective on individual stops or groups of stops. It is highly desirable to be able to apply vibrato selectively to various stops, without affecting the rest of the instrument. The smaller organs usually have but one vibrato, this being effective upon the entire instrument.

Vibrato is effected in the pipe organ by means of a device which pressure-modulates the air in the wind chest at the desired rate, causing the tones of the pipes to rise and fall in pitch. The "on and off" control is brought out to a conventional stop tab at the console.

Varieties of Tone

The reader may have wondered, by this time, why the pipe organ has so many stops, especially since it has been pointed out that a single one may be built to cover the entire compass of pitch necessary to the instrument.

The primary reason is that any one tone color, no matter how beautiful it may be, will become monotonous when heard for more than a few minutes at the most. This goes back to a fundamental law of biology and psychology, which states that an organism loses its power to respond to a given stimulus when that stimulus is applied over too long a period. A one-hour performance by a chorus of angels would become as irritating as a convention of Swiss bell ringers, providing they did nothing to vary the act.

The second reason is that various types of music, and musical passages, require widely different tonal treatment. One would scarcely expect to hear a piece of sacred music played with the "wah wah" trumpet effect which is featured in jazz bands, nor would he relish hearing his favorite popular tune rendered with the somber dignity of an anthem. Music, being an abstract expression of human experience, must be capable of registering that experience in a form appropriate to our understanding.

The third reason is one of tonal balance. It is often desirable to have a given stop available in several degrees of loudness. Since there is no practical method for making the volume of sound from a pipe instantaneously variable, the only solution lies in providing several similar stops, graded as to intensity.

The fourth reason has to do with the "ensemble effect" or "grande celeste", as it is sometimes called. The pipes of an organ, like the instruments of an orchestra, are never perfectly in tune, although the deviations in tuning may be too slight to be directly apparent to the ear. However, when many notes, each differing slightly in pitch, are played in unison, a particularly pleasing effect is produced.

This pleasing quality of the grande celeste seems, in part, due to the complex beats which arise between the various notes and their partials. There is also the fact that the ear dislikes a tone whose pitch is too sharply defined, as much as the eye dislikes a line whose edges are too sharply drawn.

While the grande celeste is undesirable in certain types of musical passages, it is absolutely necessary to the building of musical climaxes, and it can only be obtained through the use of many ranks of pipes, played in unison.

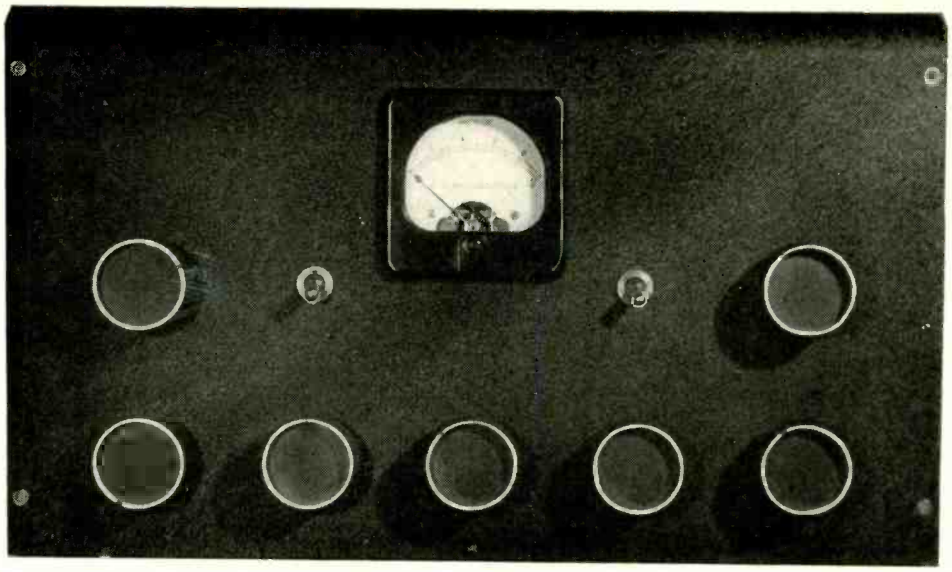
Organs designed in strictest tradition have their stops divided into carefully chosen groups, each group being assigned solely to a given manual (or the pedals). The stops assigned to one manual are not available on another, except through the use of couplers. When a single rank of pipes is so connected that it is available as an individual stop on two or more manuals (or manual and pedal), it is said to be "duplexed". When a single rank of pipes is provided with individual octave couplers, so that it may be used independently as a 16′, 8′ and 4′ stop on one manual, it is said to be "unified". It is, of course, possible that a stop be duplexed and unified at the same time.

Unification and duplexing can extend the resources of an organ enormously, especially a small organ. There are two serious disadvantages, however, to their unrestrained use. The first of these comes in attempting to play the same note on two different manuals simultaneously. Obviously, if the same pipe is being used for both, the note will sound

[*Continued on page* 43]

George Wright and Miss Gaumer of the Hammond Organ Co. with Solovox attachment for organ.

—*NBC Photo*



Multi-Purpose Audio Amplifier

MAURICE P. JOHNSON*

This high-quality audio amplifier combines high gain, unusually high fidelity, and an expander-compressor circuit. It can be easily built from readily obtainable components.

INNUMERABLE AUDIO AMPLIFIERS have been built to serve as "general purpose" systems, and have performed with varying degrees of excellence.

One common complaint against this type of unit, however, is that the very simplicity of the typical circuit limits its scope of application. On the other hand, some systems have been designed for special purposes, often with the result that the complexity of circuits results in a bulky, heavy unit, and again of restricted versatility.

The amplifier to be described incorporates more tubes and circuit features than are usually found in the general run of amplifiers in its power output class. It has been designed to serve equally satisfactorily as a recording and playback amplifier for a small studio, as a high-quality audio system for a-m and f-m tuners, and as a speech amplifier for a "ham" transmitter. The features included allow the unit to perform all the usual audio work of the home, hamshack, or small studio. At the same time, it is compact enough to be portable without being unwieldly.

The circuit and construction shown are the result of careful study and trials of already available amplifiers. Several trial constructions and tests of possible circuit features and combinations were made, and the resulting final design incorporates the features which were considered necessary and equally useful in all the intended applications.

The unit has been tested for each of the above uses, and has been found entirely satisfactory in all cases. When used with a good speaker system, the quality leaves nothing to be desired. With an f-m, or a good a-m tuner, the fidelity of reception is extraordinary.

Design Considerations

In the selection of the output tubes, the requirements for each intended use were carefully considered.

For amateur applications, there are class B modulator tubes available to provide almost any modulation power needed, and those tubes generally used by hams can be driven by ten watts or less of audio. If the speech amplifier is designed to drive such tubes, it need not be altered as the transmitter grows in size and power, which results in a considerable saving in both time and expense as time goes along.

The best class B drivers would be constant-voltage tubes, and the nearest to constant voltage of the types available is the 6B4G series. With 350 volts on the plates, a pair of 6B4Gs in push-pull with self bias will give ten watts of audio power output with less than 5% total harmonic distortion. This will be sufficient for most modulators used by amateurs.

Recording heads and speakers present a variable impedance load, and for such use the 6B4Gs again give best results. Commonly used cutting heads require approximately one watt for average disc modulation, but several serious recordists have pointed out that a reserve power of ten times the average is necessary if instantaneous peaks are to be handled without distortion. This requirement is met readily by the ten-watt output.

High power is not generally needed for normal playback uses, but the dynamic range of this amplifier is increased by an expander circuit, described below. The extra power available allows high-level peaks without excess distortion.

Other requirements leading to the selection of a push-pull triode output were the desire for low hum levels, low distortion, and good frequency response without using degenerative circuits.

Since ten watts of audio meets all the demands to be made of this amplifier, operation of the 6B4Gs in push-pull with self-bias is entirely practical. This simplifies power-supply circuits, and also allows the use of a phase-inverter input instead of an input transformer. Resistance-capacitance coupling throughout the amplifier aids considerably in keeping the hum level low by avoiding inductive hum pickup, as well as inexpensively allowing good frequency-response characteristics.

A dual triode is desirable for phase-inverter use. A 6N7 will provide the high signal voltage necessary to produce full power output with the 6B4Gs. A self-balancing inverter circuit is used which automatically adjusts itself to slight differences in tube characteristics, and this particular circuit has proved the most satisfactory of several tested.

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Tone Control

In some applications, a flat frequency response is desired, but for maximum versatility some control over the response is necessary. In recording, a rising treble response is highly desirable. In a speech amplifier, however, suppression of the highs and lows allows maximum power at the speech frequencies, and an opposite response curve is needed. For these reasons, a popular type of dual tone control was selected for this unit, which makes use of a degenerative cathode circuit built around a 6C5 tube. Dual controls permit independent control of each end of the audio range. Either boost or attenuation of highs or lows is available with the two control potentiometers, so that practically any response curve can be obtained. The circuit is so arranged that unity stage gain is obtained at mid-frequencies, regardless of the settings of the controls. More complete details on this tone control circuit can be obtained from the manufacturer of the special choke required for bass control.

It is accepted practice to limit the dynamic range of the material being recorded to prevent overcutting the disc at high volume levels, while keeping low volume levels at a suitable value above amplifier and disc surface noise. This calls for volume compression to be included in the amplifier. During playback, the volume range of the recording must be artificially increased to restore the reproduction to its original range and brilliance. This action is obtained from an expander.

In phone work on the ham bands, a high average modulation is needed for best operation. However, if the audio level is high, with the conventional speech amplifier, there is always the danger of overmodulation on voice peaks resulting in splatter. The compressor circuit is ideally suited for this application, since it will allow a high average modulation, while suppressing the peaks to prevent overmodulation. The compressor performs this helpful service by suppression rather than clipping of the audio peaks, and therefore gives less distortion.

—(Editor's Note: *The standard Thordarson tone control circuit employed here uses constants which cause the high-frequency boost curve to depart from flat at about 600 cps. Some users prefer to have the departure occur somewhat higher up in the frequency band, so that the response up to 2,000 cps is not affected by the boost control. To obtain this effect, C_{18} should be reduced in capacitance, with a suggested value being 0.005 µf.*)

Because of these advantages, it was decided that an expander-compressor circuit should be included in the design, and it is well worth the three tubes needed for its incorporation.

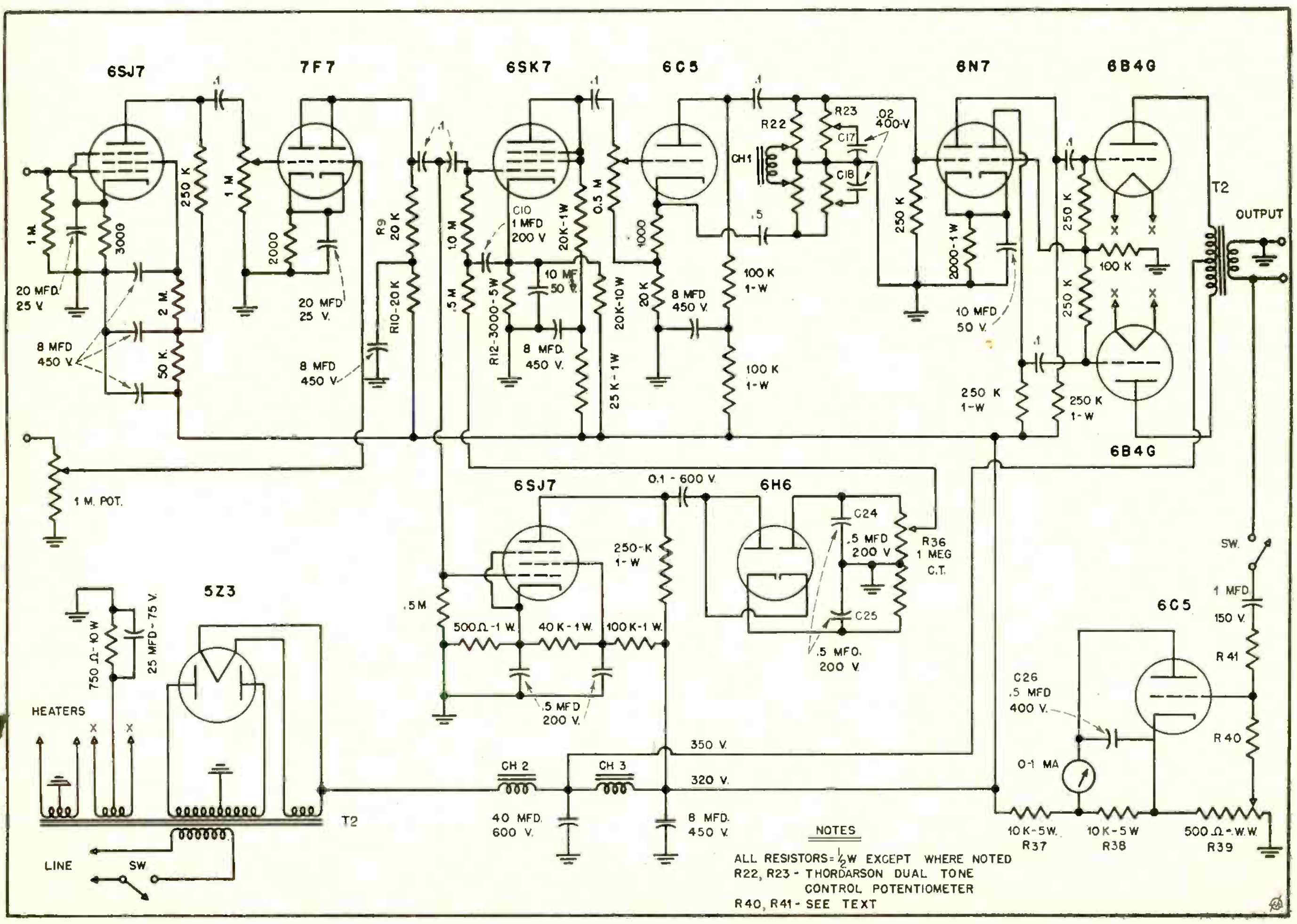
Expressor Circuit

The functions of expansion and compression are both available in the "expressor" circuit used, based upon a circuit originally appearing in *Electronics**. A 6SK7 is selected as the control tube because of its variable mu remote cut-off feature. The plate, screen, and suppressor are tied together to permit operation as a triode. The normal bias on the tube is obtained from the voltage divider formed by *R-13* and *R-16* in series.

Part of the input signal for the 6SK7 is fed to the 6SJ7, which works as a conventional pentode voltage amplifier. The amplified signal output of this tube is fed to the plate of one section and the cathode of the other section of a 6H6 duo-diode rectifier. The remaining plate and cathode are connected to the ends of a center tapped potentiometer, the center tap being grounded. The output of the 6H6 is a d-c voltage which

*Volume Expansion with a Triode," by C. G. McProud, *Electronics*, August 1940.

Schematic diagram of the multi-purpose audio amplifier.



is applied to the signal grid of the 6SK7 tube, along with the regular signal input. This d-c voltage acts as the bias which causes expansion or compression, depending upon the voltage polarity. The RC values of this bias circuit are such that the bias follows the general audio level, rather than the audio itself. Since the time constants of the circuit are rather critical, for proper operation it is recommended that the values given for the parts be adhered to closely. Some control over the time delay may be had by varying the size of *C-10,* but the value listed was found to give a good average delay for ordinary use.

Potentiometer *R-36* allows control of the value of bias voltage. Maximum expansion is had at one end of the control, and maximum compression at the other extreme. Setting the control to the mid position grounds out the control-bias voltage, and thus removes the expressor action altogether. The 6SK7 then operates as a regular low-mu triode. Naturally, intermediate settings of the control potentiometer will give intermediate amounts of expression.

——(Editor's Note: *There are two time constants involved in any expressor circuit, and both should be considered. "Attack" time refers to the delay in operation of the circuit to increase or decrease the volume upon application of a signal, while the "release" time is the delay in return to normal gain. In this circuit, the attack time is the product of* R_{12} *and* C_{10}*, and the release time equals the product of* C_{25} *and half of* R_{36} plus *the attack time. Listening tests have indicated a recommended value of the order of 75 milliseconds for attack time and this delay would be obtained if* R_{12} *were 0.5 meg and* C_{10} *were 0.15 μf. The release time should be longer, usually between 200 and 300 milliseconds. With the suggested values for* R_{12} *and* C_{10}*, this would necessitate the use of 0.35-μf capacitors for* C_{24} *and* C_{25} *with the center-tapped 1-megohm potentiometer* R_{36}*.)*

The input circuit will depend upon the manner in which the amplifier is to be used. The amplifier shown in the photographs was intended for use with a remotely located multi-input preamplifier, which would be used whenever microphone pickups were necessary. Therefore, it was decided that two medium level input channels would be sufficient. One channel could be fed by the remote preamp, and the other channel serves for the turntable or tuner locations, through the use of a simple switching panel. The two amplifier inputs are of the high impedance type.

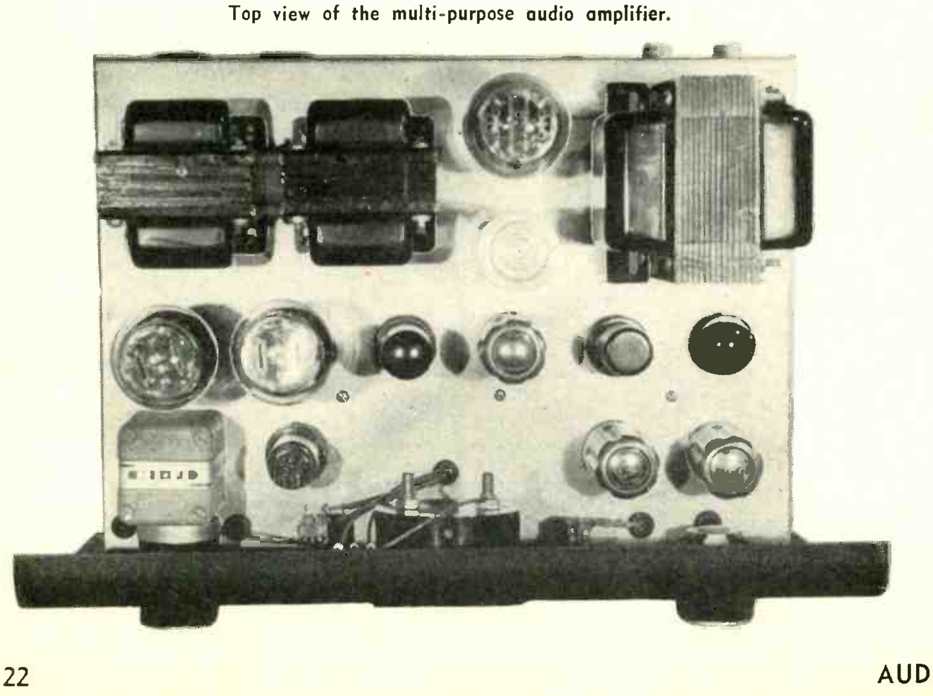
A dual triode is used as an electronic mixer, by feeding the grids individually while the plates are tied in parallel. Several types of tubes and variations of the mixing circuit were tried before the one shown was selected. A lock-in tube was chosen since it had the lowest noise level of types tested for this application.

In many cases, it is recommended that isolation resistors be added in the plate coupling arrangement, when dual triodes are used for this form of electronic mixer. However, the parallel connection of the plates to a single load resistor simplified the problem of shielding the leads and reduced the hum pickup at this sensitive point to a marked degree. A careful check of the mixing action proves that there is no interaction of the controls, and no noticeable distortion occurs in the stage. This is by far the most satisfactory of the many mixer elements commonly encountered.

Each mixer grid is fitted with a volume control potentiometer for controlling the input signal amplitude. A master gain control is added between the expressor and tone controls.

Although not used in the original construction, a 6SJ7 preamplifier stage is shown in the schematic. This was added for those persons who may wish to duplicate the amplifier and desire at least one microphone input without the necessity of using a separate preamplifier. It is recommended that not more than one microphone channel be added to the main amplifier chassis, because of the difficulty of adequately shielding and decoupling the high-gain stage when located with the power supply. If more than one mike channel is needed, a preamplifier is the best solution. It is then suggested that the layout shown in the photos be followed for the amplifier, and two medium level inputs be made by feeding the 7F7 grids directly, with the associated gain controls, in a manner similar to the one medium-level input to the 7F7 shown in the schematic.

Top view of the multi-purpose audio amplifier.



Monitoring Recordings

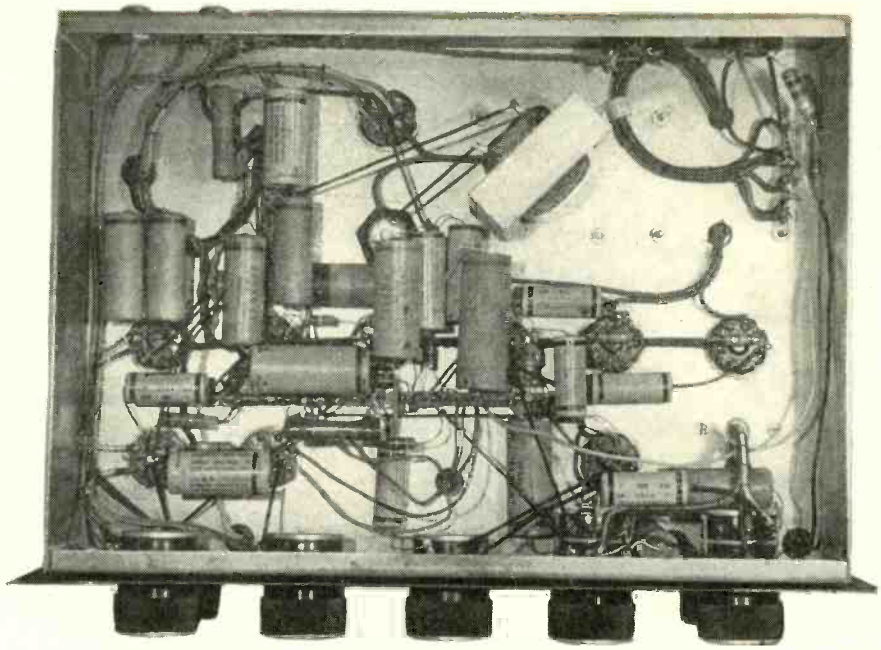
Another feature which has proved to be very useful is the output meter shown connected across the output transformer secondary. Special db or vu meters are available for use in keeping an accurate check on the output levels.

What was desired in this case, however, was not so much an extremely accurate indication of the exact output, but rather a simple indicator of reasonably good frequency characteristics for monitoring the recording level. A simple vacuum-tube voltmeter was constructed with an 0-1 ma meter as the indicator. The circuit is conventional. *C-26* is added to enable the movement to respond to peaks which might otherwise not show up on the meter. A 7F7 section was used for the triode meter tube in the pictured unit, but a 6C5 would be more suitable because of its metal shield. *R-39* is a wire wound potentiometer for adjustment of the bias. This allows the meter to be adjusted to function as a regular output indicator, by setting the bias so that the meter reads zero or nearly zero plate current with no signal. Any signal voltage will then cause the meter to read up scale, due to the rectifying action of the tube. It is also possible to increase the bias beyond the cutoff value, so that the meter will respond only to peaks. One note of caution: With no bias, the tube will draw more than the one milliampere maximum rated for the meter movement, therefore it is important to adjust the bias from a full-negative value toward a less-negative voltage. Cutoff will be readily determined, since the meter will begin to read a steady value of plate current if the bias is reduced below the cutoff value, with no applied signal.

There are many ways of coupling this output meter to the amplifier. Care must be taken to avoid shorting out the input to the meter. The coupling method shown has been satisfactory. The values of *R-40* and *R-41* must be selected by trial. They should be chosen so that the meter will read about half scale with normally used audio levels. It may be desirable to make a variable voltage divider if the amplifier is used repeatedly at varying levels. In the unit shown, *R-40* was made one megohm and *R-41* was made 5 megs to give good meter indications at normal speaker levels of approximately 5 watts.

The power supply is conventional. The components are oversize to give trouble-free operation and avoid excessive heating. Filter capacitors are large, and choke input is used. Separate windings are used for the output-tube filaments so that self bias can be used.

Considering the circuit as a whole, it will be noticed that every attention has been paid to securing freedom from hum without sacrificing good fidelity. Extensive decoupling is used in the plate voltage leads. By-pass and filter capacitors are large. Triodes are used wherever possible, with low values of plate loads. Coupling capacitors are large enough to give good response at the low frequencies. All resistors, and particularly those in the high-gain input stage should be of highest quality to reduce circuit noises as much as possible. The layout has been made with the idea of keeping the amplifier compact and without crowding of components. Care in construction and wiring has been amply rewarded in noise-free and trouble-free operation.



Bottom view of the multi-purpose audio amplifier.

Construction

For those interested in duplicating this amplifier unit, the following information may be helpful.

The unit shown in the illustrations was constructed on a 10 x 14 x 3 inch chassis, and mounted in a desk style cabinet with a 9 x 15 inch front panel.

The controls, from left to right along the bottom of the panel are: bass control, treble control, master gain control, first input gain control, and second input gain control. Above the bass control on the left of the panel, is the output-meter bias adjustment. Beside this is the toggle switch in series with the meter lead. In the center of the panel is mounted the milliammeter. This is followed by the line switch, while the expressor control is mounted to the extreme right.

Looking at the top of the chassis, the power supply is arranged along the rear edge, the input choke and filter condenser mounted above the chassis. The output transformer is at the left rear. The row of tubes in the center, from left to right, consists of: push-pull 6B4Gs, 6N7, 6SK7, 6SJ7, and 6H6. Directly behind the panel, from left to right, are the tone control choke, followed by the 6C5 tone-control tube, and at the right of the meter, the output-meter tube and the 7F7 input mixer. If the 6SJ7 input stage is to be used, it should be located at the right, where the 7F7 is now shown, the 7F7 moved to the left where the meter tube is now located, and the meter tube mounted directly behind the meter movement. These locations have been selected in the interests of low hum and minimum interaction between stages. The components directly behind the panel should be set back sufficiently to avoid interfering with the controls and meter which are mounted on the panel.

The under-chassis view shows that the wiring is made as direct and point-to-point as possible. A row of tie-down lugs is run between the two lines of tube sockets, to mount resistor and condenser leads. The power supply wiring is cabled. The high-gain input leads are all made with shielded wire, which is bonded to the chassis every few inches. All ground points are tied together with bus-bar, even though the chassis is itself at ground potential. Care in wiring throughout will give an amplifier with inaudible hum even at high output levels.

Operation Notes

Some comments on the proper adjustments for best operation of this unit are in order.

With all controls full open, the gain in the amplifier becomes excessive. In fact, when an ordinary commercial shellac pressing is played with a high-quality pickup of the high-impedance type which gives approximately one volt output, the 6B4Gs will be overdriven if the controls are fully opened. The reason for incorporating this extra gain, however, will be apparent when the proper adjustment of the circuit is explained.

For correct action of the expressor, the signal applied to the 6SK7 grid is rather critical. With insufficient signal, the control bias developed will be low, and it will be impossible to obtain complete expansion or compression. On the other hand, if too great a signal voltage is applied to the grid, the expression bias will cause over-compression and over-expansion. Correct voltage will give smooth expansion with no tendency to distort, and will hold the output signal at a constant level when compressing, with no tendency to squelch or cut-off the 6SK7.

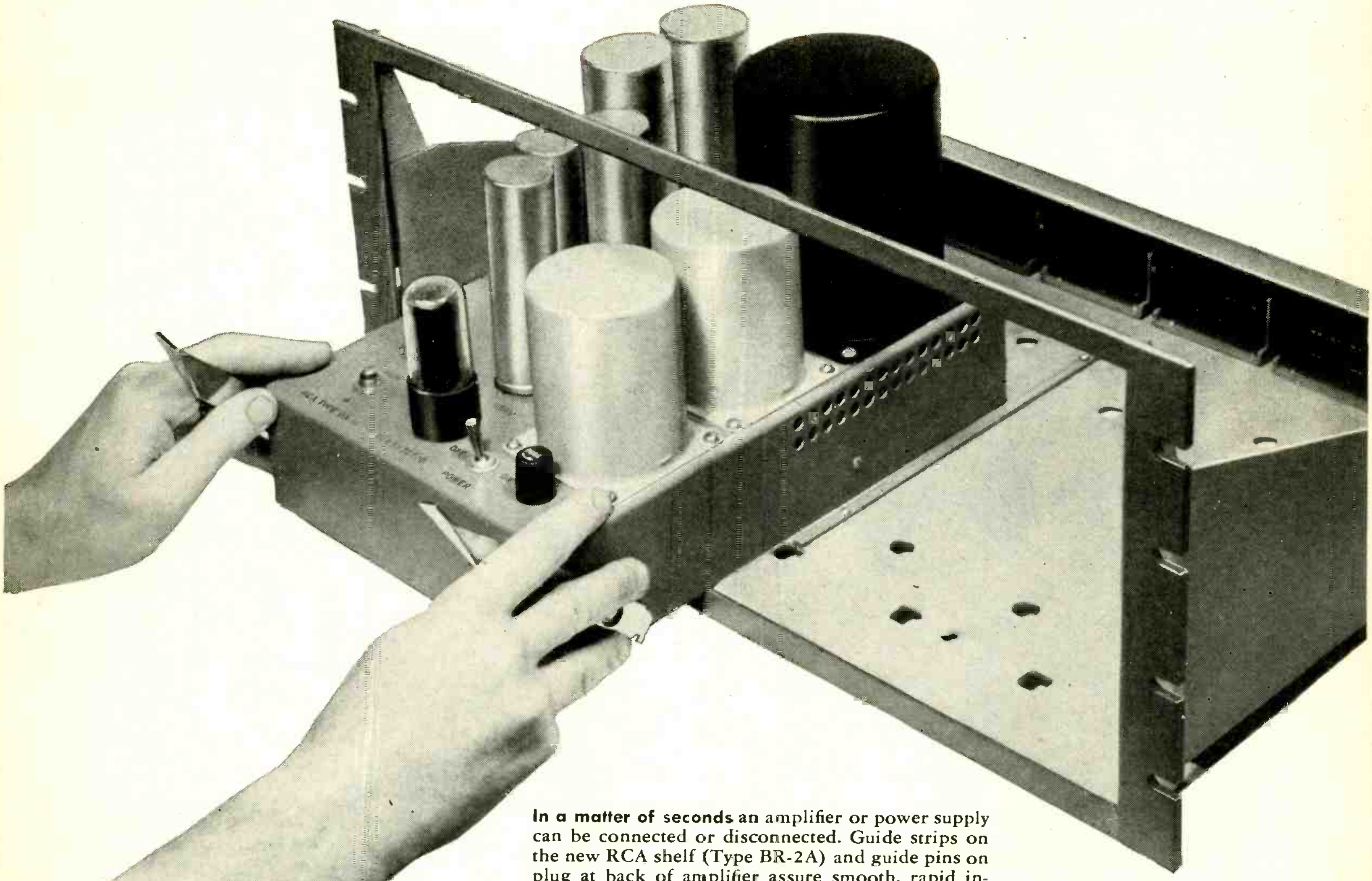
The input gain controls are used to set the signal to the proper level for proper expressor action, and usually this will call for about 0.25 to 0.5 volt signal on the input-mixer grid. Once the input gain controls are set to this level, there will be no need for readjusting them until a different pickup device is used. The master gain control, which is located after the expressor stage, is used as the conventional gain control, to set the output volume level to that desired for any particular application. Under these conditions of operation, a check of the waveform at each stage was made and no distortion was apparent on the c-r tube, even with the master gain control fully on.

Control over expansion or compression is had by varying the bias by means of potentiometer *R-36*. Do not use the input gain controls as a means of varying the expressor action, once the controls have been set to the proper positions.

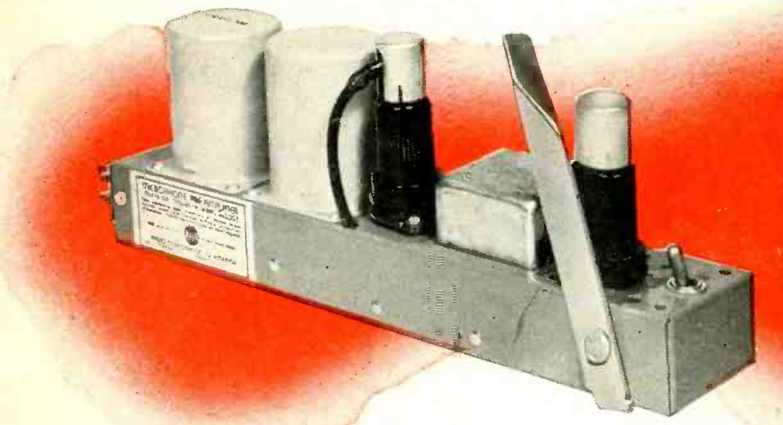
Adjustment of the meter bias control was covered above. The remaining controls are conventional in action, and no

[*Continued on page* 39]

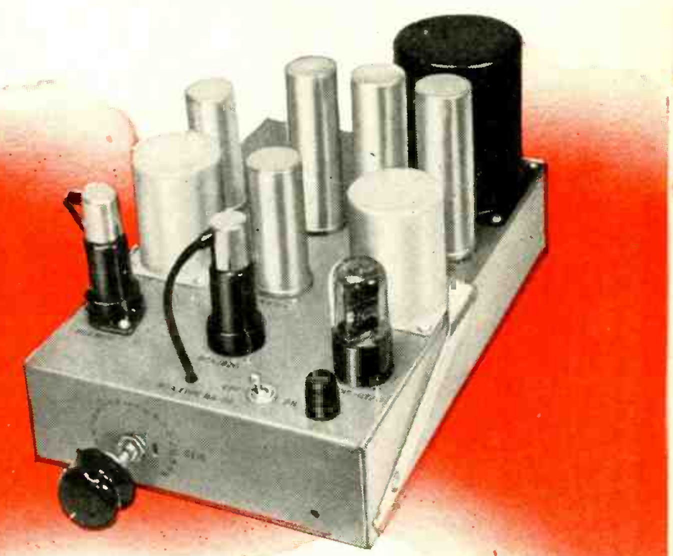
"PLUG-IN" amplifiers for



In a matter of seconds an amplifier or power supply can be connected or disconnected. Guide strips on the new RCA shelf (Type BR-2A) and guide pins on plug at back of amplifier assure smooth, rapid installation. Levers at front hook into slot in shelf and pull amplifier into place. Sockets on bracket at rear of shelf permit self alignment of receptacle with amplifier plug. (Shown is the BX-1A Power Supply Unit.)

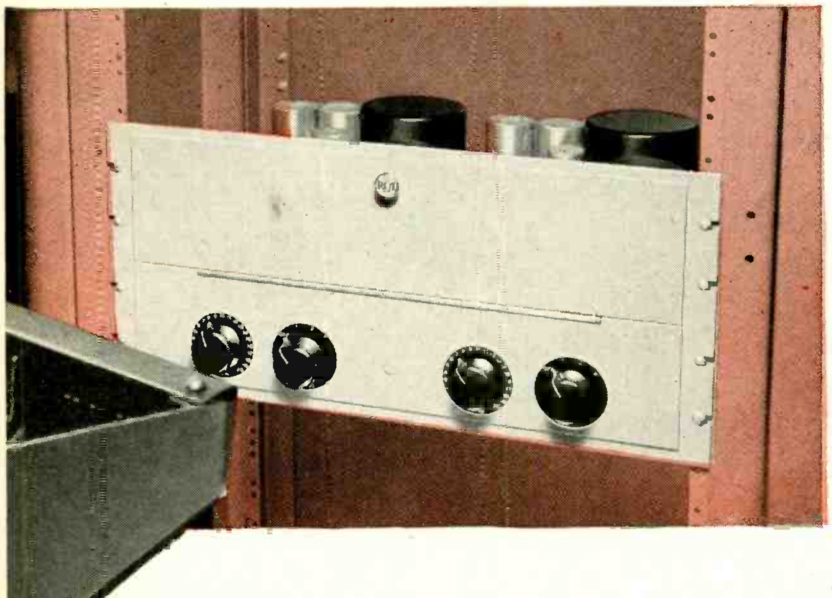


Two-stage Preamplifier (Type BA-1A)—ideal for use as a microphone preamplifier, turntable preamplifier, booster amplifier, or low-level isolation amplifier. *High gain:* 40 db. *High output:* +10 db. *Low noise level:* −80 db. *Low distortion:* 0.5% rms, 50 to 7500 cycles. *Isolation factor:* approx. 90 db; over 100 db with special Volume Control Kit. *Frequency response:* ±1 db, 30 to 15,000 cycles. *Small size:* six units will fit on a 36-B or new BR-2A shelf!



Booster Amplifier (Type BA-2A)—A two-stage unit having applications similar to those for the BA-1A; also valuable where a high-gain amplifier between announce microphone and limiting amplifier is required. *High gain:* 50 db. *Low noise level:* −68 db. *Low distortion:* 0.75% rms, 40 to 15,000 cycles. *Frequency response:* ±1.5 db, 30 to 15,000 cycles. *Compact:* two units can be mounted on one 36-B or BR-2A shelf. Features plug-in capacitors and built-in power supply.

new convenience and speed



The RCA Type BR-2A Shelf fits any standard rack; takes 8¾ inches of panel space. If desired, however, RCA Type 36-B panels and shelves now in use can be easily adapted for plug-in amplifier service.

These versatile, multi-purpose units by RCA will simplify many of your studio setups

As easy to install or remove as an electronic tube! Pull a lever near the front of the amplifier and the plug on the rear of the unit is smoothly withdrawn from its socket—automatically disconnected from the supply voltage. No longer is it necessary to crawl around to the back of hard-to-get-at racks and unsolder or unscrew countless connections. System changes can be made quickly; minutes can be slashed from inspection, servicing, and testing time.

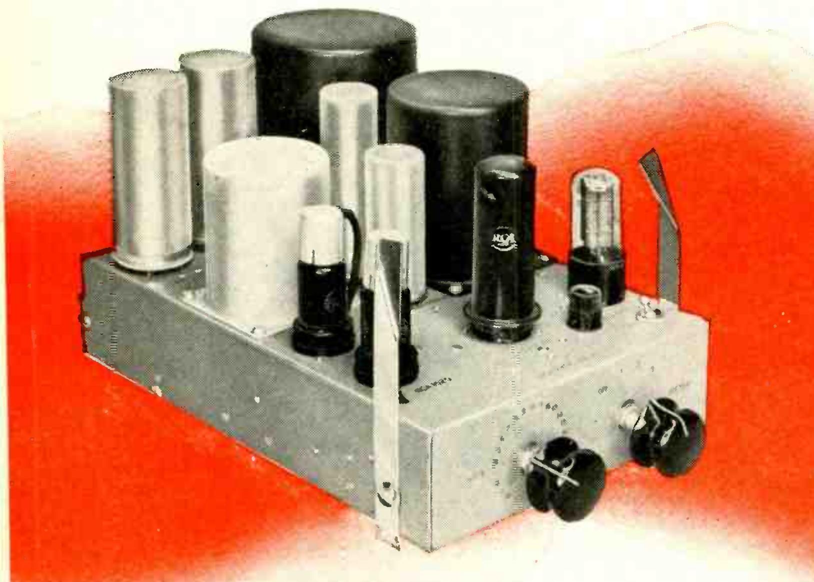
This new RCA line now includes the four amplifiers and one power-supply unit shown. Others will be added in the near future. New, carefully selected characteristics make these units ideal for a large number of studio jobs.

All units use the same standard plug. To assure maximum convenience, a new shelf (Type BR-2A) has been designed. With a few easy changes, however, the conventional RCA Type 36-B panel and shelf can be used, if desired. The necessary accessories are available for this purpose.

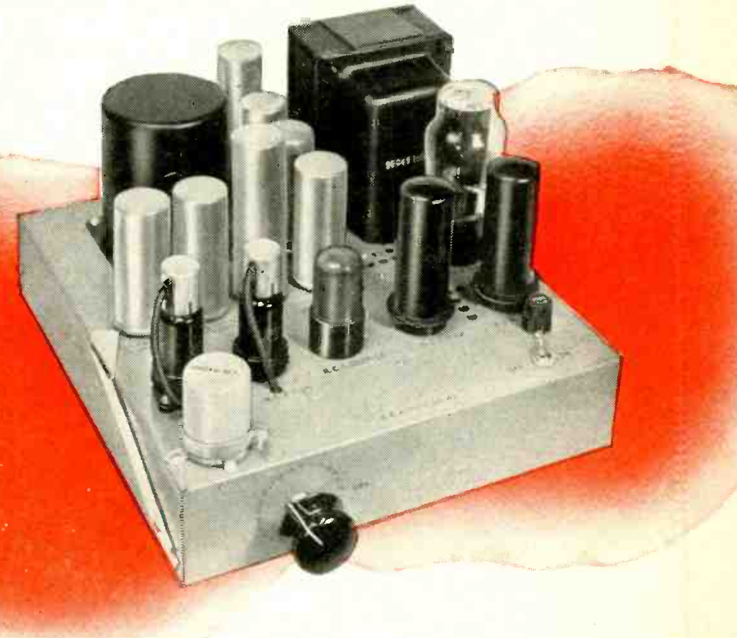
Here, we believe, is a real opportunity to modernize your amplifier system—a quick, convenient way to get better performance at low cost. New descriptive leaflets are yours for the asking. Write: Dept. 115-H, Audio Equipment Section, Radio Corporation of America, Camden, New Jersey.

BROADCAST EQUIPMENT
RADIO CORPORATION of AMERICA
ENGINEERING PRODUCTS DEPARTMENT, CAMDEN, N.J.

In Canada: RCA VICTOR Company Limited, Montreal

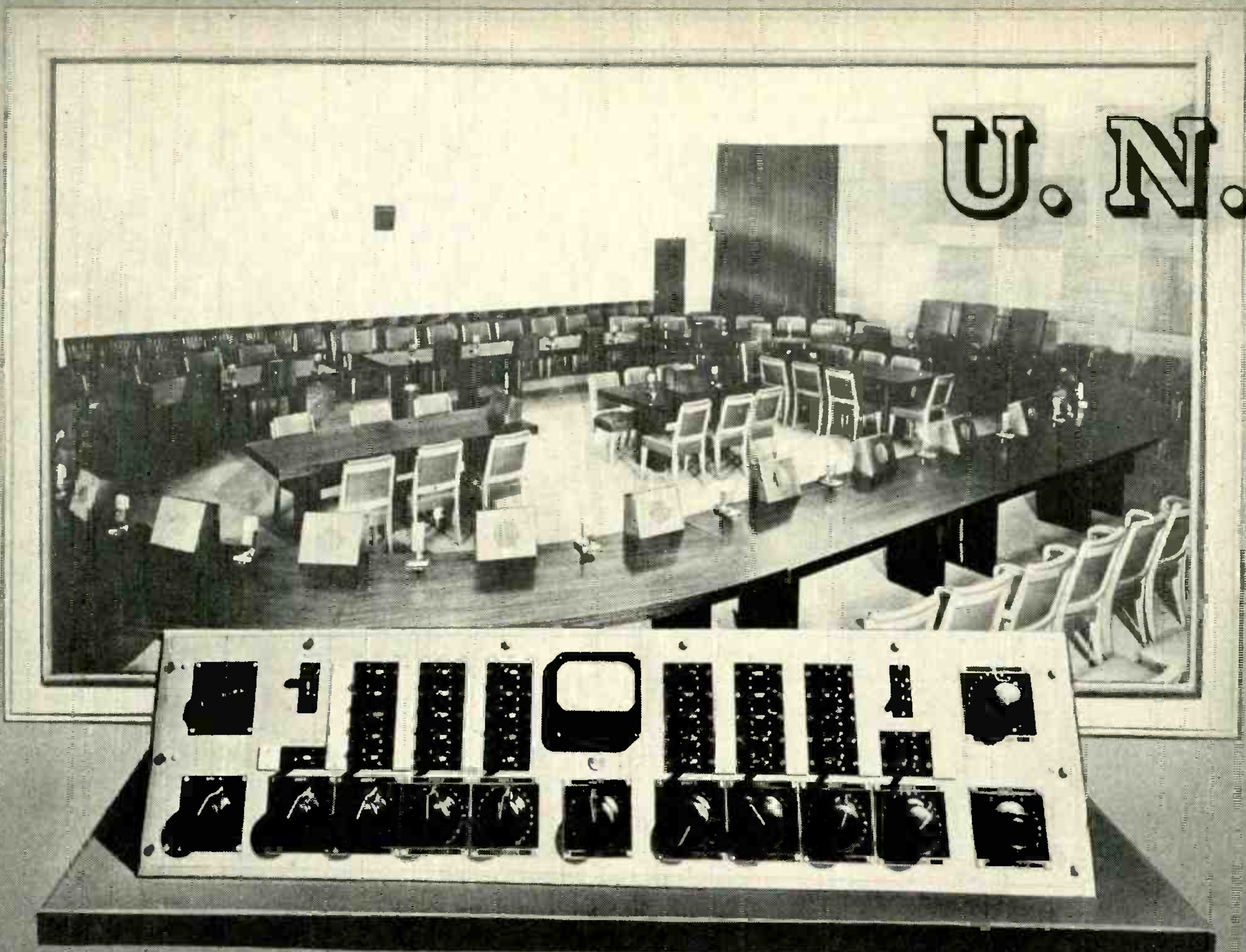


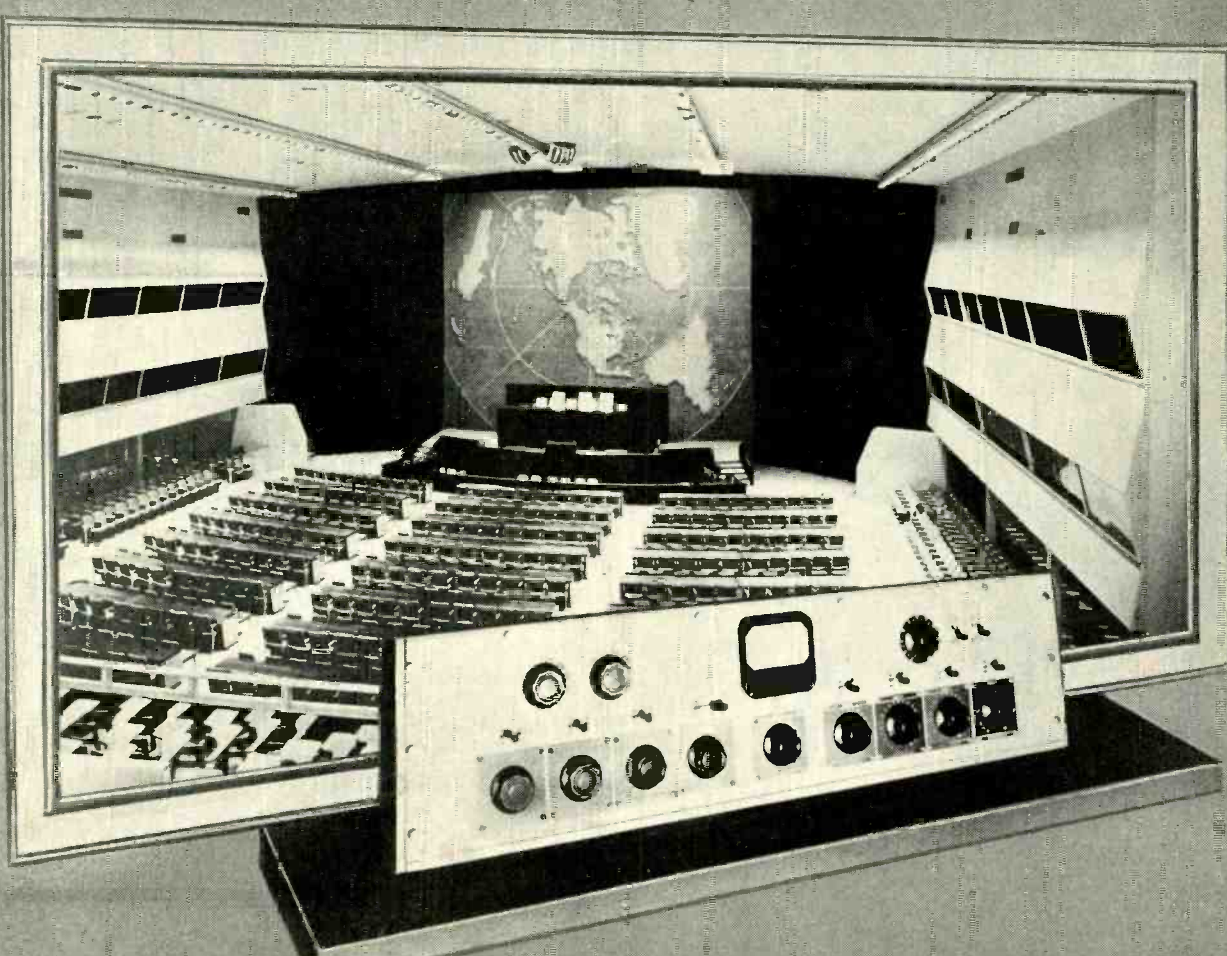
Program Amplifier (Type BA-3A)—one of the most versatile on the market. Ideal as a program, line, or booster amplifier; a high-level isolation amplifier; a cueing or monitoring amplifier, or a driver amplifier (for high-power recording amplifiers, etc.). *High gain:* 65 db for matching input, 27 db for bridging input. *Low noise level* (with maximum gain): –52 db. *Low distortion:* less than 0.5 to 1% rms, depending on output level. *Frequency response:* ±1 db, 30 to 15,000 cycles.



Monitoring Amplifier (Type BA-4A)—Designed for operation at microphone levels. High output of 12 watts is sufficient to drive several speakers or, in some applications, a recording head. Other uses include application as line amplifier for portable and mobile transmitters. *High gain:* 105 db. *Low noise level:* –20 db (with maximum gain); –40 db (with minimum gain). *Low distortion:* less than 3% at 12 watts. *Frequency response:* ±2 db, 30 to 15,000 cycles.

U. N.





Broadcast & Public Address Systems

Describing an elaborate and successful sound installation.

ARTHUR W. SCHNEIDER*

In 1939 at the World's Fair there was dedicated the Court of Peace, and not far from it lay the New York City Building, one of the Fair's permanent structures.

Eight years have passed, and amongst the many changes that have taken place since Hitler's attempt to turn the dedication of the Court of Peace into a mock ceremony has been the conversion of this New York City Building into a temporary home of man's latest concept of the Court of Peace, namely, the United Nations General Assembly.

In 1939 the engineers of Commercial Radio-Sound designed the broadcast facilities and the public address systems for the World's Fair, and these same engineers found themselves very much at home during the installation and design of the systems now in use in the United Nations General Assembly.

There are two systems in use at the General Assembly's headquarters, one for the large General Assembly Auditorium and the other for the Conference Room. Each of these systems is totally independent of the other except for tie-lines between them permitting feeds in either direction. The purpose of both systems is as follows:

a) To make audible to the members of the United Nations, their guests, visitors, press and all others present all words spoken by each speaker; to permit the maximum freedom of movement on the part of the speaker and a minimum amount of concentration on the part of the auditor.

b) To faithfully transmit all sounds picked up to the following services.

1. Broadcast stations and networks
2. Recording for official records
3. Motion picture film recorders
4. Television

c) To accomplish the functions outlined above with the minimum number of controls, ease of operation, and dependability of circuits.

Gen. Mgr. and Chief Engineer, Commercial Radio-Sound Corp., N. Y. C.

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Fig. 1. (top). United Nations conference room.

Fig. 3. (bottom). United Nations general assembly auditorium.

Conference Room

Figure 1 shows a photograph of the actual Conference Room setup which is unique in the application of pickup and reinforcement of the spoken word.

It is necessary to pick up from any one of the delegates sitting around the outside of the large table or from any of the interpreters located within the oval, and permit good hearing with good intelligibility without feedback by all persons in the room. At the same time the intimate character and purpose of the room has to be preserved.

The solution suggested by Commercial Radio-Sound engineers was adapted by the United Nations' engineers for all their Conference Rooms, including those at Lake Success.

As can be seen through the control room window in *Figure 1,* the Conference Room contains a large oval table on which are located thirty-one microphones and loudspeakers for the delegates. Within the table space are two more microphones and speakers for use by the interpreters. The block diagram for the Conference Room is shown in *Figure 2.* To facilitate handling this number of microphones, thirty of those about the table were wired through lever keys into six groups of five each. Each group of five is controlled by one attenuator. The two interpreter microphones within the oval, each through its lever key, are grouped into one attenuator and a single microphone for the chairman of the conference has an attenuator of its own. This grouping of the thirty-three microphones results in only eight controls, the desired microphone being switched in by throwing its lever key. Such an arrangement permits rapid and accurate switching.

These microphone keys are used for a relay interlock operation with the loudspeakers adjacent to the associated microphone, automatically turning off the loudspeaker on each side when the microphone is turned on. Since all the loudspeakers on the table are operated at a low level, the turning off of the two on either side of the line microphone is sufficient to prevent feedback and the remaining loudspeakers permit all the delegates to hear the speaker perfectly.

By throwing these same microphone keys to the opposite side from their "on" position, the associated microphone is connected to a separate cue amplifier and speaker. This arrangement aids the operator in anticipating the next speaker, and permits accurate switching.

The output of the eight-position mixer feeds through two identical circuits of booster and line amplifiers and master gain controls. These provide a continuously protected amplifier chain that meet through a "Regular-Emergency" key. In case of failure of the regular section, throwing this one key to "Emergency" restores operation. Spare amplifiers of all types employed in the system are available on jacks to permit patching if necessary.

A high-pass speech filter is provided with a switch to cut in or out of the circuit at will. This filter cuts off sharply at 200 cps and prevents the "boomy" speech characteristic of some persons. The output of the console is controlled at +8 vu.

Other controls at the console are the monitor, cue, PA gain control, and a utility attenuator. The utility attenuator input and output are available at jacks on the rack and two utility transformers with inputs and outputs are also available. These utility components are useful for setting up special circuits that invariably become necessary from time to time. Another useful control mounted on the rack is the monitor input selector switch. This switch allows spot checking throughout the system in order that any failure may be quickly located.

From the output of the console, the circuit feeds the PA and telephone line dividing amplifiers. The output of these amplifiers terminates in a low value of resistance, in this case 15 ohms. Each feed to a PA power amplifier or to a telephone line, as the case may be, bridges the proper 15-ohm resistor with 600-ohm low impedance matching resistor. Since the source bridged is low impedance, the isolation between outgoing 600-ohm lines is of the order of 30 db. This provides an economical means of supplying a number of isolated circuits from one amplifier.

In this case, twenty-four 600-ohm telephone lines with the isolation mentioned are fed with plus 18 dbm each, all from one 10-watt amplifier. All amplifier equipment is located in three racks directly behind the operator.

Planning a Studio Installation

J. D. COLVIN*

PART II—This is the second of a series covering broadcast studio installations. The methods outlined are also suited to large public-address projects.

As a start toward laying out the console, a list of all the items that are to be mounted either on the panel or in the cabinet should be prepared. The source of this list is the block diagram. Checking off on the block diagram the items already planned to be mounted in the equipment rack leaves those that belong in the console (microphones, loudspeakers and over-the-door on-air lights excepted). All controls, switches, indicating lights and meters should be mounted on the panel while such things as transformers, resistors, non-variable or non-indicating items are to be mounted either behind the panel or somewhere inside the cabinet.

Making the actual panel layout is another job that starts with pencil and erasures on cross-section paper and ends up with "T" square and triangle. The object is to arrange the parts to be mounted on the panel in such a way as to provide the greatest convenience to the operator. An effort should be made to group together those controls that are functionally related. For example, a switch with a pilot light to indicate its operation should be grouped together. A switch in a mike circuit should be located over the attenuator of that circuit, or if for other reasons the switch is located on the announcer's control panel in the studio, a light should be located over the attenuator to indicate when the distant switch is operated and that the attenuator is hot. Until someone conceives of a better arrangement, the panel sketch begins by laying out the variable attenuators for microphones, turntables and nemo inputs across the bottom. The VI meter goes at the top center of the panel. What is left in the way of switches and lights are located functionally in the remaining space.

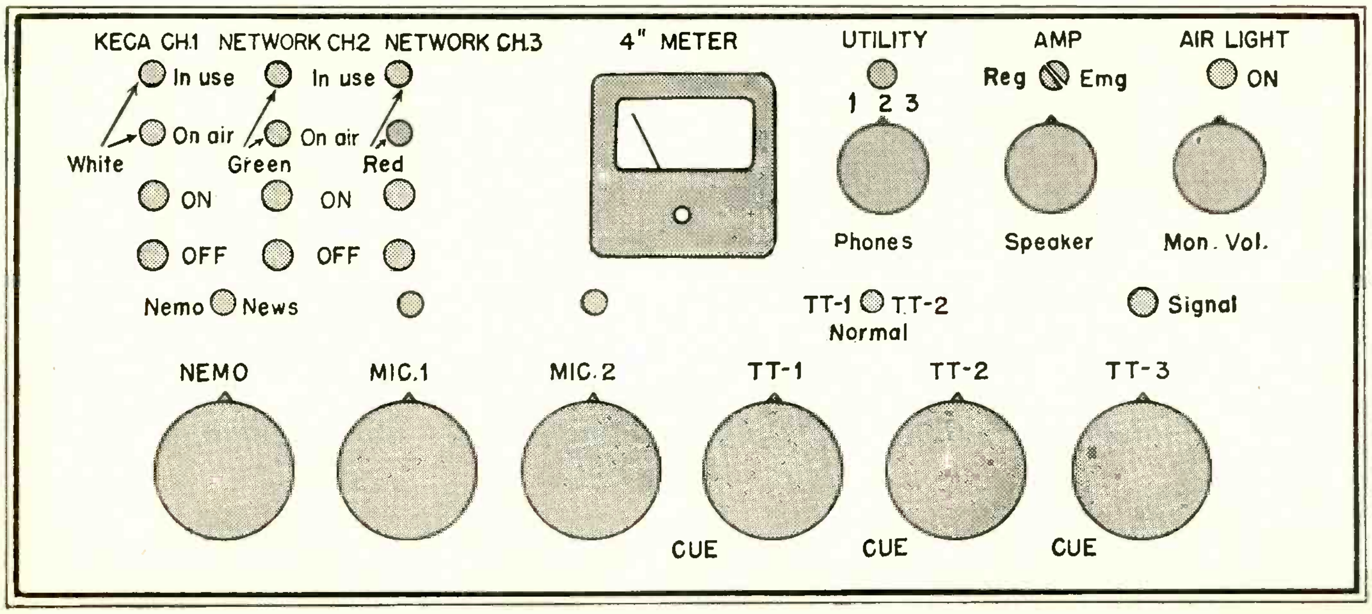
*Audio Facilities Engineer, American Broadcasting Co.

Panel Layout

Figure 1 shows the panel layout that was finally arrived at for the studio being used as an example. The upper left-hand section contains the group of lights and push keys that control the master switching relays. This arrangement is uniform with other control panels of the same station. Indicating lights that show when either *Mic 1* or *Mic 2* are on are located immediately over their respective *mic* faders. The switch described as providing a rapid transfer between *turntable 1* and *2* is located over and between the faders for these turntables. Headphones and monitor speaker selector switches and the monitor speaker volume control are within easy reach but far enough removed from the turntable attenuators so as not to be confusing. Least used are the utility and the regular-emergency amplifier keys and naturally take the spot of lesser operating convenience at the top of the panel. The key for turning on the "on-the-air" sign is also at the top of the panel but more readily found by habit since it is on the extreme right side of the panel.

Console panels should be as low as is possible to make them without undue crowding of parts or excessive sloping. Eight inches is a good height to strive for but very difficult to obtain with a flat panel. Off-set panels as shown in *Fig. 2* are necessary to obtain the minimum of height but are costly to manufacture and require a more expensive cabinet. When operated on a table 28 inches in height, a console panel can be eleven inches high and not make it necessary for the operator to stretch to look into the studio. Most comfortable operation of the faders is had when mounted on a panel that slopes 10 to 15 degrees off the vertical with their center line between 2¾ and 3¼ inches above the table top. This slope of panel

Fig. 1. Panel layout for the studio installation discussed in this article.



also puts the VI meter at an angle of best visibility to the operator. Thus, after having laid out the panel on cross-section paper until a satisfactory arrangement is obtained, the next step is to draw the panel layout accurately and to dimension all the mounting holes. Attention must be given to clearance between each part to make sure there is sufficient room for wiring.

Finish

Finish specified for a panel should be one not having a high gloss to avoid troublesome light reflection. A very high resistance to wear is necessary because of the invariable habit operators have of rubbing the tips of their fingers on the panel while riding gain. An electrolytic finish known as alumilite is probably one of the best from the wear standpoint. It can be had in black, natural aluminum and some pastel shades. This finish can only be applied to an aluminum panel and for best results must be reflector sheet aluminum. Run-of-the-mill aluminum sheets have a tendency to finish streaked. Alumilite finish can be engraved through without any danger of chipping as occurs with most lacquer finishes.

Factors that determine cabinet design are such things as: the panel size and the depth to which equipment mounted on the panel extends toward the rear; other pieces of equipment that are to be mounted on the base of the console as terminal blocks, transformers, etc.; the point at which interconnecting cables between the console and other equipment enter the console; sufficient space to clear apparatus mounted near the top of the panel when the panel is swung outward on its hinge. Cabinets must also be designed for appearance as well as function. A well-appearing studio installation always creates a good impression on a client.

For normal servicing of the equipment on the panel, particularly the faders, the panel should be arranged on a hinge out from the top of the cabinet. For installation of the console it is desirable to have access to the terminal blocks through the rear of the cabinet, or better still have the entire cabinet top removable. The base of the cabinet with terminal blocks and panel attached remain fastened to the table.

Preliminary preparation of the conduit layout was listed as the next step in planning a studio layout on the basis of probability. In the author's experience, by the time the equipment layouts have been completed and before running sheets and interconnecting diagrams can be started, actual construction work on the studio plant is under way. And, since the contractor will start to pour concrete for the control room floor the day after tomorrow, the electrician has to have a conduit layout by yesterday! Since it has happened so frequently in the past it is probable that it will happen again in the future and the only solution is to whip out a preliminary conduit layout.

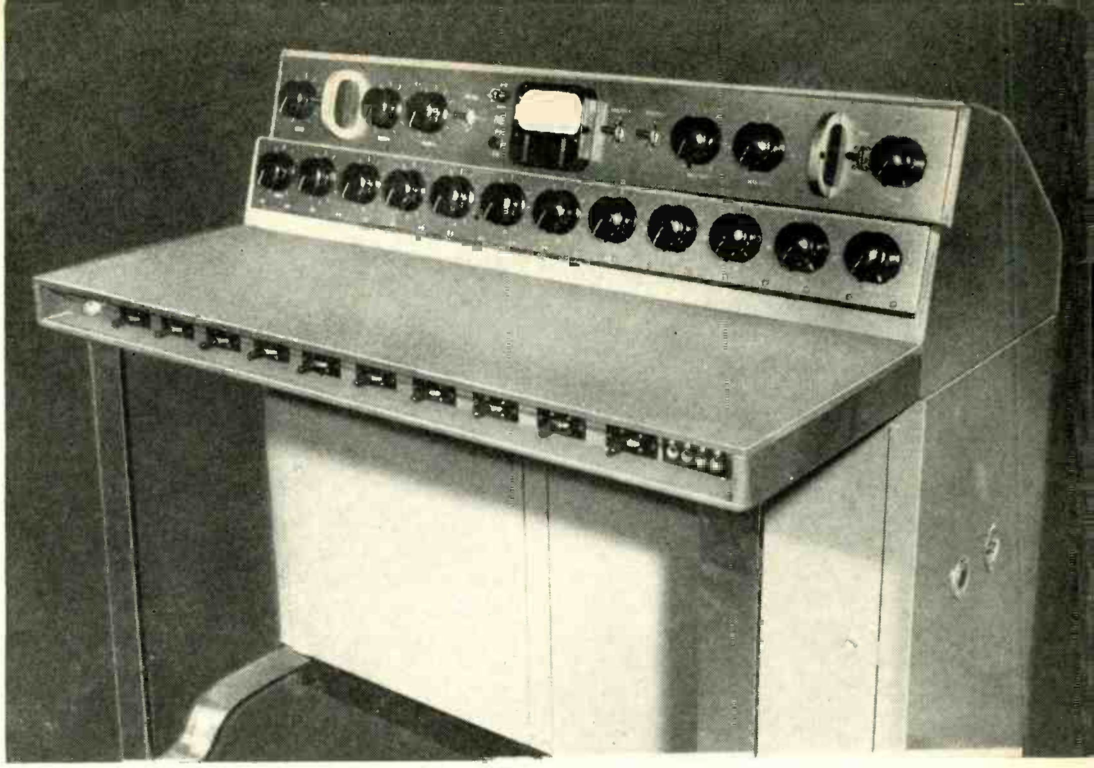


Fig 2. Typical console with off-set panel designed for most convenient operation

Conduit Layout

To start the layout, a plan view drawing of the studio and control room is necessary. On this are drawn, in the proper locations and dimensions, plan views of the equipment rack and console. With the block diagram as a source of requirements, appropriate indications are marked on the plan view drawing where each microphone and turntable receptacle box is to be located. Turntables require a-c power for their operation so it is necessary to indicate a-c outlets convenient to their location. Outlets for loudspeaker and over-the-door on-air light signs must be accounted for and indicated. When all such equipment terminations are accounted for that are shown on the block diagram one is quite safe to assume that nothing has been overlooked.

Conduits, metal duct or troughs in the floor must now be decided upon as a means of carrying the inter-connecting wiring between all terminal points that are indicated on the plan view. The use of conduits and troughs in the floor should be considered only on new installations when they can be located as required. Metal ducts running along base boards are useful in those cases where additions to existing equipment are being made and it is either impossible or impractical to make conduit runs in the floor. Troughs are most useful where there are a great many wires running between two points as is usually the case between a rack and console. Its use makes easy the running of the many pairs and the segregation of these into cables of safe level difference. Troughs also provide a convenient terminating point for individual conduit runs which might otherwise have to be much longer and make extra bends if run direct to their final termination point. The rack end of a trough can run directly underneath the rack thereby providing an easy means of running the cables up to the terminal blocks. At the console end the trough usually ends underneath a wall type cabinet mounted at floor level. The box provides a means of anchoring flexible conduit running up to the console and to bring cable from the trough to the console terminal blocks. The main use of conduits is for runs of but one or a few pairs of wires to isolated points such as mike receptacle, speaker outlets, the main program line to master control etc. Conduits are best suited where runs must be made through walls or from one floor to another.

Circuit Levels

A very definite rule must be observed as to the separation of cables into circuit level groups when run in conduits and troughs. Wire pairs having a level difference of more than 30 db should not to be run in the same circuit or cabled together in a trough. Failure to observe this precaution is likely to cause cross-talk between circuits of greater differences. A level group classification found useful in broadcast installations are *Low, Medium, Zero, Control* and *Power*. *Low* is for circuits from —60 to —30 VU, medium, from —30 to 0 VU, Zero from 0 to +30 VU and High covers any value above +30 VU. *Control* is composed of battery, relay and light circuits. *Power*, as its name im-

[*Continued on page* 41]

Self-Monitoring Recorder Head

S. YOUNG WHITE

Describing a simple method of avoiding overload in recording.

IN HIGH-QUALITY LATERAL RECORDING, the monitor is a very important person. He must bridge the gap between the 90 db variation in input levels and the 30 db or so the record can handle. As direct recording becomes more popular, we must often perform this function automatically. In dictating machines, the user learns to monitor himself after a while. In coin-operated recording devices open to the general public, this problem is very severe; a great many are making records for the first time, and are so scared that they just about whisper into the microphone. The others are probably just as scared, but they shout into the mike.

Telephone Recording

Telephone recording is another art where we have no control over the relative signal strengths of the two voices. Now that we can tap into the line we can anti-side tone some of the local energy. But someone may call from across the street with a voice of great recording energy level, or he may be at a distance of 30 miles or so, just inside of repeater range, and may have a low energy voice.

Audio a. v. c. is a partial answer except that there must be some time constant in the system. Whatever the time constant chosen, it is bound to be wrong for many conditions, and the resultant effects are difficult to explain to the customer.

There is a very interesting effect noted when first recording from the telephone. The "quality" of telephone speech is so bad from a fidelity point of view that one may assume any poor recording system will handle it. This definitely is not so—it requires a pretty good recorder to take it without making the speech quality so much worse that it is unusable.

The high-frequency end requires no special care in design, except to avoid sharp resonant peaks in the system below about 5 kc; otherwise transients in the line will shock-excite the system.

As we go down in frequency the excursions of the recording stylus become greater, of course, and the final low-frequency point, say 200 cycles or so, is represented in *Fig. 1*.

The horizontal scale represents signal level into the system, and the vertical is the resultant excursion of the recording stylus. Dotted line 1 is the noise level of the recording system itself. Shaded area 2-3 is the line noise, which is not under our control, and varies between very wide limits indeed. Line 4 represents the maximum safe excursion of the stylus, as beyond this limit we may run into cross-overs.

Curve 5 is a wide excursion system. If we could afford to record at ten grooves to the inch we would have no problem, but economics dictate that we must run from 100 to perhaps 500 grooves to the inch.

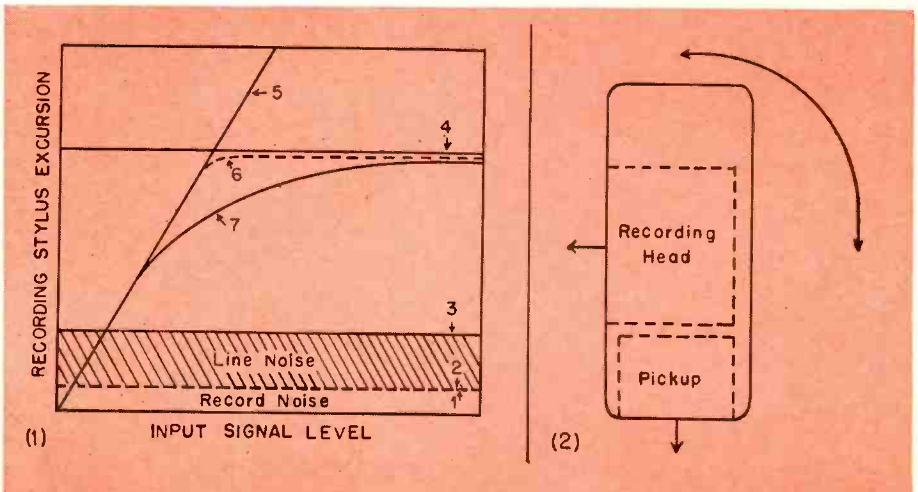


Fig. 1. Curves showing limiting action. Fig. 2. Sketch of turret-type head.

Curve 6 is *curve 5* modified by placing fixed limiting stops at the dangerous limits of excursion. Do not try this, as it gives the most horrible distortion known to man, a sharply chopped flat-top wave.

Curve 7 is the final compromise. It is sensitive and linear for the first half of the maximum excursion. This records the weak signal above the noise level with maximum fidelity. At very close to the half-excursion point it changes to a smooth curve that approaches the limit in an asymptotic manner—always getting closer but never quite reaching the limit.

The distortion introduced by this type curve is surprisingly easy to take. On high-level telephonic speech it is about the equal of a high-fidelity recording of the same level, properly hand-monitored. This of course is a matter of opinion, and the writer is naturally prejudiced in its favor. So-called "impartial witnesses" agree; but perhaps they may be simply too polite to disagree.

Much experimental effort was made before we decided to build such a head. We were never able to modify the amplifier itself and obtain electrically such a smooth distortion characteristic as resulted by redesigning the head to do the trick mechanically.

It is very difficult to apply the system to a crystal head, because if we attempt to restrict the excursion of a Rochelle crystal very much we set up stresses that may ruin the crystal. Best results were obtained by using a rather wide gap magnetic head, with resilient damping material in such thin section that the mechanical resistance built up rapidly with deflection, giving a composite curve of the type desired. The whole head had to be designed from the ground up, in fact.

Turret Head

Such a restrained head is a pretty heavy load on the record when used for reproducing, so another (usually a crystal) head was used for the pickup. Instead of having two arms, however, a turret-type head was designed, as shown in *Fig. 2*. The recording and pickup

[*Continued on page* 44]

In this department the author, who is a very well-known record critic, will review monthly record releases of outstanding technical, as well as musical, quality.

EDWARD TATNALL CANBY

THE ADVENT OF HIGHER FIDELITY and greatly increased tonal range in recording for commercial records has, strangely enough, brought up problems of mike pickup that are leading to a return toward simpler mike techniques, away from the highly complex and hitherto remarkably effective multi-mike techniques developed in the last twenty years. The problems, which are squarely in the area where engineer and musicians work most profitably together, are clear cut enough.

With the limited tonal range and high noise ratio of earlier recording and of AM broadcasting there could be no question of exact reproduction of music. Rather, the problem was how best to simulate the real thing. The attempt to force greater proportions of overtone coloration into both vocal and instrumental reproduction led to the now prevalent and necessary type of pickup where a solo musical instrument or voice is placed only a few feet from a solo mike — whereas no concert audience was ever intended to hear the live musical sound from closer than, say, thirty or forty feet. The actual live sound of most musical instruments and of the singing voice is startlingly ugly at close range. But the close-to-mike pickup does tend to force enough of the higher overtones through the narrow-range electronic system to give a reasonably life-like sound, and in fact by far the most satisfactory sound that can be had with the standard 4000-7000 cycle upper limit. Indeed, developments in multi-mike technique, especially in popular music, have been sensational, considering how little of the musical tone color does get through the ordinary system. Popular music could not exist without the elaborate selective microphoning that balances, for instance, the powerful trumpet against the weak guitar. Large scale symphonic music has not always fared so well, for pre-electronic composers undeniably intended their music—including the many concertos for solo instruments — to be heard at concert hall distance. The engineer faces a fine dilemma here, for true distant-mike pickup has been most difficult, thanks to the lack of overtone coloration. Brilliant compromise results have been attained with the new accentuation technique, which combines features of close-to and distant pickup, and it is in fact remarkable what results can now be attained musically on a record that is played back with a 4000-cycle cut-off!

Those of us who first began working in FM before the war, however, discovered for ourselves in a dramatic manner what extraordinary things can happen when the usual techniques are applied to a true high-fidelity system, minus background noise. A speaking voice at one-foot range seems to hiss in your face; an oboe or similar instrument at two or three feet or even a dozen, is strident and mechanical; the player's breath and the mechanics of finger work are horribly apparent. A flute player spits saliva between every note, perfectly audibly. I shall never forget the performance of a conventional dramatic show in which the heroine had a good cry — into her closely held microphone. The FM effect of this perfectly standard AM technique was as Niagara, or the air brakes on a dozen trains! A high fidelity system is super-realistic in an embarrassing way; it gives you exactly, *exactly* the monaural sound that would be heard at the microphone's position. Close-to pickup, then, is impossibly high fidelity, ghastly "distortion" (actually a lack of distortion) of the musical sound. Engineers, now working with high fidelity, are finding themselves, to their surprise, backing away, farther and farther. The whole beautiful edifice of close-to mike technique is coming up for drastic overhauling, and the field is wide open.

A perfect illustration of the extent of this change was afforded me in a recent recording session, when Keynote recorded an oboe concerto by the Englishman, Vaughan - Williams. There were only fifteen string players, plus the soloist. The studio was large and live (Reeves). A single over-all mike was placed fifteen feet in the air, more than forty feet away from the small group of players, huddled in the middle of the studio. The oboe soloist, far from having his own solo mike, simply stood next to the conductor, not more than three feet in front of the orchestra. He was picked up directly by the same single mike, *more than forty feet away.* The recording, made on new Fairchild cutters and reportedly good to 12,000 cycles in the pressing, is musically of the utmost naturalness as played back on wide range equipment. A circumstance as remarkable as the mike technique itself was the striking similarity in the sound of the music as heard in the studio and in the control room.

[*Continued on page* 43]

MUSICAL ACOUSTICS

BENJAMIN F. TILLSON*

PART III

This is the third of a series of articles on music theory, written especially for sound engineers.

The literature on sound and acoustics would lead one to believe that his hearing would experience the same nodal changes in sound intensity or amplitude as graphically represents the patterns of wave interferences. Each pattern represents a different phase, frequency, and amplitude relationship of its component waves.

In "Scientific Papers" I, p. 409 Lord Rayleigh describes two unison organ pipes mounted side by side, with ends close together, on a common windchest; and that if blown for a short time they sound, but when blown longer but less than a second the sound dies away to a small fraction of that due to either alone. This may not properly exhibit the interference of sound waves in the open because the pressure pulsations emitted from one pipe may be sucked into the adjacent pipe when out of phase. However, he observed that when the two pipes were out of tune they sound a common note which may be higher than that due to either alone.

When a vibrating tuning-fork is rotated before an ear four positions will be found that make it inaudible, but the sound will again be heard if something is interposed between the ear and one prong of the fork. A half-pint creambottle will resonate a fork tuned to "A" equals 440; and the silent position occurs when the plane of the prongs makes an angle of forty-five degrees to the axis of the bottle.

When both waves are of the same frequency, the amplitude "beats" between them have a common frequency, so no change of pitch takes place and the resultant sound is either louder or less loud. At the most such amplitude is doubled and the intensity quadrupled, or increased only 6 decibels, which is inconsiderable. It is well to remember that the energy and intensity of a tone vary directly with the square of its frequency and also with the square of its amplitude. However, the null position is observable in the tuning-fork experiment.

But if one makes a repeated striking of the same piano key, with the damper off, it seems impossible to obtain a null point or noticeable variation of sound intensity. Perhaps that is chance, or else the presence of overtones masks the interference results obtained with pure tones.

When one strikes the arpeggio of a chord (the notes of a chord in sequence), with dampers off, the resultant tone sounds the same as when the keys are struck simultaneously. This might lead one to believe that the ear analyzes complex sounds into their component frequencies, and that the brain synthesizes them again into a pattern dependent only on pitch and amplitude, but ignoring phase differences.

However, when graphs are plotted of mixed frequencies the respective phases of the components give entirely different patterns. This is illustrated by *Fig. 6*, where a fundamental and five of its overtones (with amplitudes varying inversely as their frequencies) are plotted for respective conditions of all starting in phase with maximum amplitudes, or all starting with zero amplitudes at the same instant. Different patterns are also shown for the chord of the Tonic and the Fifth, C and G. Since no difference of sound results, such analysis by the ear seems indicated.

How that analysis is accomplished is a matter for conjecture. Probably the canals and chambers of the external and internal ear, the bony amplifying mechanism (the maleus, incus, and stapes), the rods of Corti on the basilar membrane, and the membraneous diaphragms all have specific resonant frequencies, varying fidelities of response to impressed vibrations, and certain inertia and imperfect forces of restoration; and the nerve ends may also vary in sensitivity throughout the areas of the basilar membrane in the cochlea.

At least it has been shown that the ear does not give a linear response to the sound waves it receives. Just as the non-linear portion of the characteristic curve of a radio tube is used for the detection and transformation of the frequency of incoming signals, so the ear uses a combination of incoming sound frequencies to manufacture "subjective" tones whose frequencies are equal to either the sum or the difference of the incoming frequencies. The unequal response of the ear to different frequencies is shown in *Fig. 7*.

Each curve in *Fig. 7* shows the intensity of sound waves in decibels at various frequencies to impress the ear as having the same loudness level indicated for each curve at the pitch of 1,000 cycles per second. It will be noted

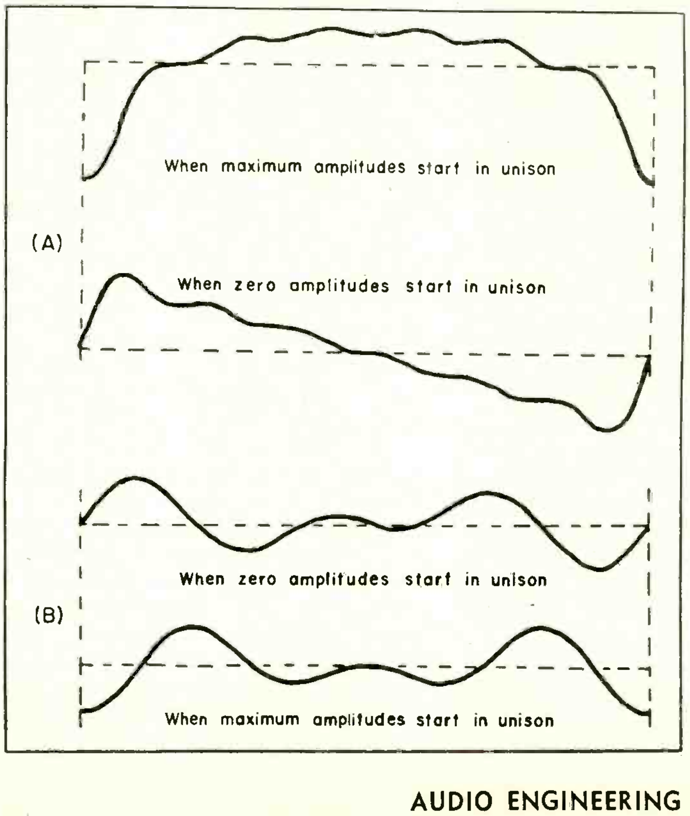


Fig. 6. Graphs of mixed frequencies when phase relations are changed.

*Consulting Engineer, Montclair, N. J.

that below 700 cycles or above 7,000 cycles the intensity of the tones must be greater, and that it is particularly so with the bass notes to give a loudness of 60 db. The decibels of loudness levels are called Phons.

The required increase of intensity of the frequencies above 7,000 cycles does not vary greatly with the various degrees of loudness; but such increase does vary greatly in the frequencies below 700 cycles per second, and a higher fidelity is favored by great loudness. Therefore, any controlled reduction of amplified volume causes a greater loss in bass response and warrants a "boosted bass" in phonograph records and radio amplifiers, but it also makes automatic volume control likely to produce an unbalance between bass and treble tones.

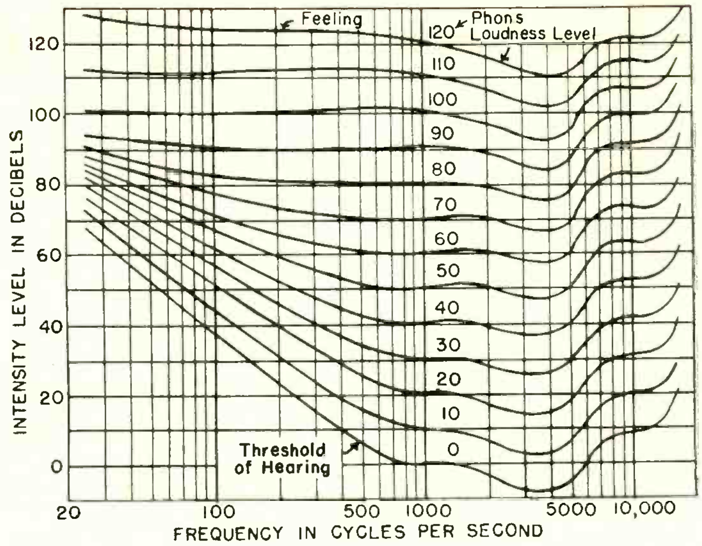


Fig. 7. Intensity of sound waves at various frequencies (Fletcher).

Within certain limits all of our sense organs conform to the Weber-Fechner Law in psycho-physics: that equal increments of sensation are associated with equal increments of the logarithm of the number which represents the comparative ratios of the stimuli (therefore, the relationship between increase of stimulus and resulting increase of sensation).

If one listens to music from a point 100 feet away and approaches to 10 feet, since the intensity of sound varies inversely as the square of the distance, it will be 100 times or 20 db greater. Also, all tones below 700 cycles frequency will sound disproportionately louder and a 100-cycle tone will have a loudness level a further 20 db above a higher-pitched tone. Thus the bass tones are lost as we get farther away or as the volume of a loudspeaker is reduced.

Overloading

When the ear is overloaded it acts as does an overloaded electron-tube amplifier by departing from a straight-line magnification of impressed frequencies and acting as a modulator which originates harmonic frequencies. This takes place when the intensity of the impressed tone exceeds 40 decibels above its threshold of hearing value; and higher harmonics result from increased overloading.

Another result of overloading is the impression in the brain of a shift in pitch, which is particularly noticeable for tones of 100 to 200 cycles per second. At 60 db, the tone seems two per cent lower in frequency, at 76 db three per cent lower, and at 93 db eleven per cent lower (as determined by S. S. Stevens at Harvard). Dr. Harvey Fletcher noted that a *222*-cycle tone at 100 db sounded the same pitch as a 200-cycle tone at 40 db; and that a 421-cycle tone at 100 db was the octave above the *222*-cycle tone and sounded the same in pitch as a 400-cycle tone at 40 db. Since the intensities produced by orchestras may be as high as 100 db it is evident that the crescendos produce dissonances.

Furthermore, if a pure tone receives a cyclical variation of its intensity its frequency is increased and decreased by the frequency of the intensity change; and the fundamental has only one-half intensity and each side-band one-quarter. A different modulation occurs where there is a cyclical variation of the frequency of a pure tone. This produces the musician's vibrato, which is best when varying six to seven times per second.

Visual Perception

An equivalent illusion occurs with visual perception where the speed of a rotating body appears to increase as the illumination is decreased, and vice versa. This indicates that there is a timing function associated with the mechanism of transmission of nerve impulses to the brain. About twenty years ago the writer proposed an electrostatic bio-chemical hypothesis to explain the transmission of nerve impulses as due to the charging and discharging of electrical condensers in the nerve system.

The time interval for charging would depend upon the intensity of the stimulus, and the frequency of charge-discharge cycle would also vary with such intensity. Therefore, an image would move a shorter distance (for a given velocity) upon the retina between successive periodic stimulations of the optic nerve endings when receiving a high illumination intensity than it would with dim illumination. Similarly with the stimuli of sound, the more rapid nerve transmission of high intensities gives a brain sense of a greater interval between each cycle, and therefore a lower pitch of tone.

In the construction of a nerve there is a central "axis cylinder" of about nine per cent of the nerve fibre, encased by a sheath of fatty myelin. The mineralized plasma of the central core makes of it an electrical conductor, while the fatty sheath is an insulator. Such is the manner of an electrical condenser.

Twenty years ago it was dicovered that no graduated impulses are carried by a nerve fibre, and that sensations were observed in steps or quanta and that a certain quantum represents the threshold of perception. If the transmission of nerve impulses were by direct electric current any stimulus would produce a proportionate effect by gradual change. This is not so.

We find that there is a threshold of hearing and that the loudness heard is directly related to the number of nerve fibres excited and the rate at which these excitations occur, since each fibre always carries its maximum impulse. When all nerve fibres have been excited at their maximum frequency no further loudness is possible as sound.

After a nervous impulse has passed through a nerve there is a refractory phase during which time the ionized nerve plasma and tissue is reconstructed and the nerve is unable to respond or conduct. Then there follows a relative-refractory phase during which the excitability, the conductivity, and the speed of propagation gradually return to normal, and upon doing so an inertia effect is exhibited by passing the normal to a supernormal state when the nerve is more sensitive, more highly conductive, and permits a greater speed of propagation, and so conducts quanta less than the threshold values. Then the supernormal state returns to normalcy.

The time interval of the refractory state is one millisecond, and for the relative-refractory state is three milliseconds. Therefore, the maximum number of nervous impulses that a single nerve fibre can send to the brain is 1,000 per second; and those periodic excitations greater than 300 per second will not be transmitted as normal impulses since

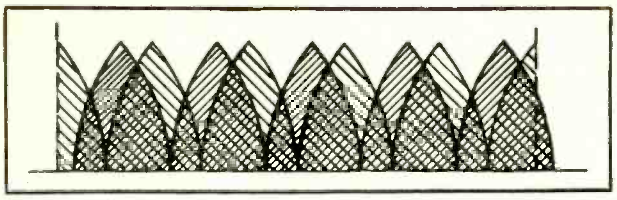


Fig. 8. Analogy of reflected light effects to acoustic reflections. See text.

each succeeding excitation would then lie in the relative-refractory phase. But Adrian determined a longer relative-refractory period for the nerve endings so as to respond to no more than 150 impulses per second. Regardless of the quantitative values the possibility is present for a subjective inter-modulation distortion which must be given consideration.

Consequently a pure tone of 3,000 cycles per second probably sends impulses to the brain at as few as 20 per second, and the rate of discharge cannot exceed this. Nerve stimulation in the inner ear occurs four times in each frequency cycle: at wave crests and troughs, and at change across the static position. The louder the exciting impulse the greater the area of basilar membrane stimulated and the greater the number of nerve endings and fibres excited. Intermodulation of the basilar membrane produces additional frequencies.

When two or more tones are impressed upon the ear they produce beat tones whose frequencies are the sums or differences of both the fundamental tones and their harmonics. These are called Subjective Tones, or Combination Tones.

A continual application of intense sound vibrations will ultimately destroy the cilia (hair cells) and possibly the auditory nerves, as could be expected from continuous ionization or electrolysis of the plasma without normal biogenetical reconstruction or supply.

A complex tone, consisting of a fundamental and its harmonics, also varies in its pitch as its intensity (loudness) is increased; but it varies less than a simple pure tone. But it retains the pitch of the fundamental even though the latter, and even some of the lower harmonics, are filtered out. That is because the difference between each pair of successive harmonics is the same as the frequency of the fundamental. Therefore loudspeakers too small to resonate to the low frequencies give the illusion of bass tones.

A complex tone of 200 cycles with five harmonics was found to increase in pitch by only 4 cycles (two per cent) when its loudness level was lowered from 100 db to 40 db; whereas a pure tone of 200 cycles increased about 20 cycles (ten per cent) in pitch for the same change in loudness levels.

The greatest change in pitch occurs at frequencies variously given by different observers in the range of 100 to 200 cycles frequency of pure tones. Taking as the base 40 db above the threshold of hearing H. Fletcher found a drop in pitch of 8 per cent for a 200-cycle tone at 100 db, and W. B. Snow found a drop of 21 per cent at 120 db, but the maximum per cent of drop in loudness occurred for 100 cycle tones and was 10 per cent for 100 db, 6 per cent for 80 db, and 2 per cent for 60 db. If the frequency of pure tones at 120 db increased from 200 to 400 cycles (pitch rating at 40 db) the actual tones heard increased from 158 cycles (21 per cent less than 200) to 368 cycles (only 8 per cent less than 400 cycles), or a change of 1.21 octaves at 120 db instead of the one octave change which would be heard at 40 db.

This illustrates the danger of dissonance from reproducing at high sound levels.

We, therefore, learn that an auditor's idea of the pitch of a tone depends upon three physical characteristics of sound waves: their frequency, overtone structure, and intensity.

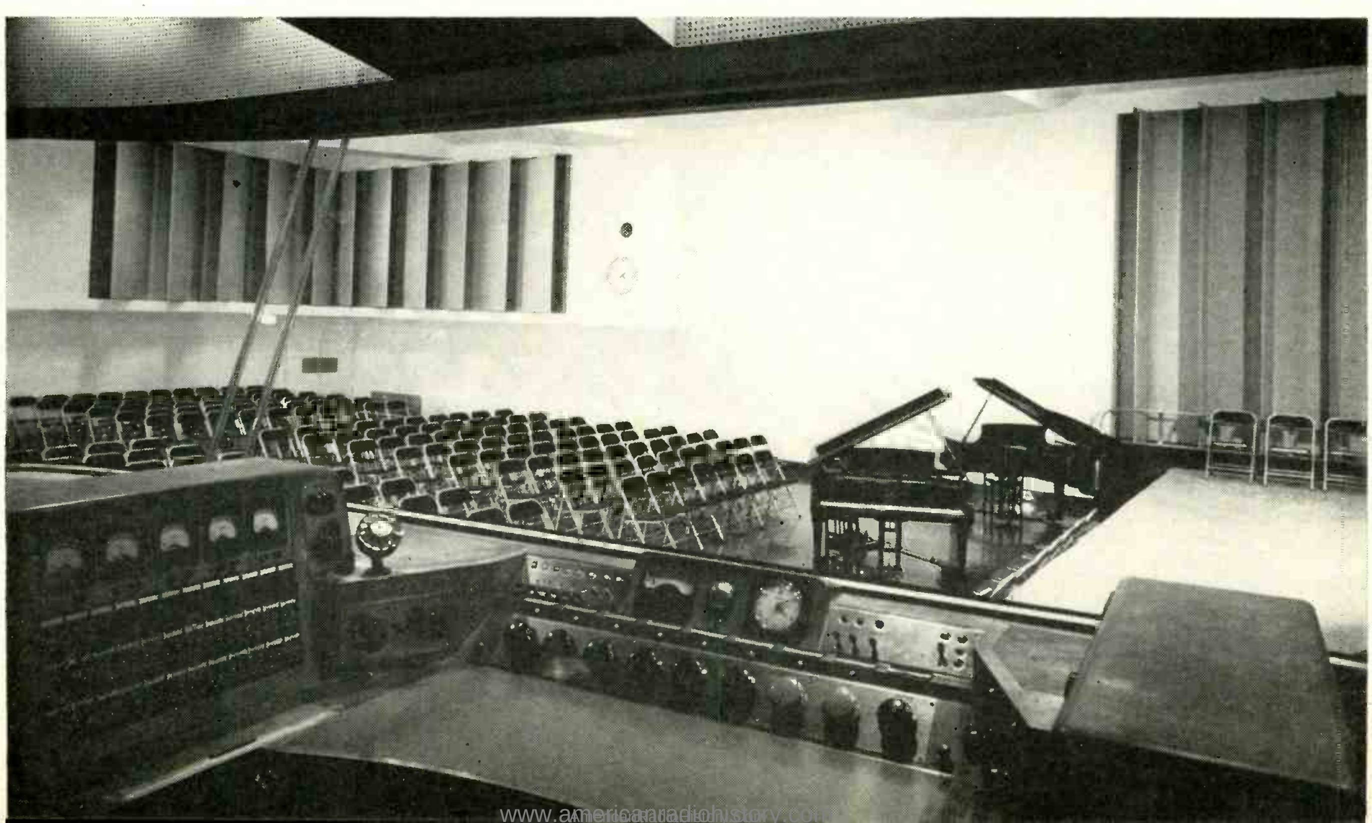
Timbre

The "timbre" or color of a complex tone has already been mentioned as dependent upon the specific partials and their relative intensities. But since the auditory pitch depends upon variable intensities at different frequencies an increase in the loudness of a complex tone produces different percentages of pitch changes of the various partials, thereby further changing the timbre.

Furthermore, Messrs. Chapin, Trimmer, and Firestone reported in 1934 and 1937 to the Acoustical Society of America that a change in phase relation of two harmonic tones of low frequency and high sensation level produces a perceptible change in both loudness and tone quality (timbre). This is contrary to the conclusions already expressed by this writer: that tone structure and not wave form is responsible for the character of the sound heard.

H. Fletcher has determined that about

CBS-New York studio 21 viewed through the control room window. A CBS 2A studio control console is in the foreground. Rotatable wooden panels enable the reverberation characteristics of the studio to be altered.



one-twentieth second is the minimum time for which a frequency must be sustained for the ear to recognize a tone of definite pitch. At a shorter time it hears only a transient sound. Thereby we are relieved of much of the transient distortions in acoustical and mechanical systems if they are masked by higher intensities of musical sounds. Many consonants used in speech are transients of very brief duration, especially the "stop consonants", and their decay is very rapid. They add to difficulty in understanding certain words in telephone conversations. Longer transitory periods result from musical instruments or apparatus because of the greater inertia of the vibrating elements.

The sound of music out-of-doors should be different from that heard in concert halls or chambers, even though the latter are free from reverberation or echo, because there is always some sound reflected from bounding surfaces. The combination of original and reflected tones will affect the amplitude and phase relationship of tone patterns. *Figure 8* gives a simple example of equivalent effects illustrated by light and shadows. A single light bulb located in front of a mirror produces two sources of illumination which cast two shadows of the tops of uniformly spaced, repeated sections of a steam radiator. The combination of two shadows gave the darkest and most prominent shadows with a cyclical variation of amplitudes and lower maximum amplitudes.

Difference Tones

The Difference Tones were first discovered by A. Sorge in 1740 and were independently discovered by G. Tartini in 1754, and were named after the latter. H. von Helmholtz developed in 1863 a theory for their formation in the middle ear from the non-linear response of the ear-drum and its bony linkage, because of the damping of the forces of restitution; and he predicted the difference tones and later confirmed that by experiments.

Combinational tones may be formed between the overtones of two complex tones as well as between their fundamentals, and in some cases even between the combinational tones themselves. When formed between the fundamentals they are called of the "first order". The summation tones are much more difficult to perceive than the difference tones.

The richness of harmonies, their color, and their emotional appeal largely result from the transient dissonances of the subjective tones which they evoke. This has been the unconscious gift of the genius of composers rather than an analytical knowledge of combinational tones on their part. *Figure 9* depicts by music notation, as the black quarter-notes, the combination tones of the first order from simple musical intervals shown by half-notes.

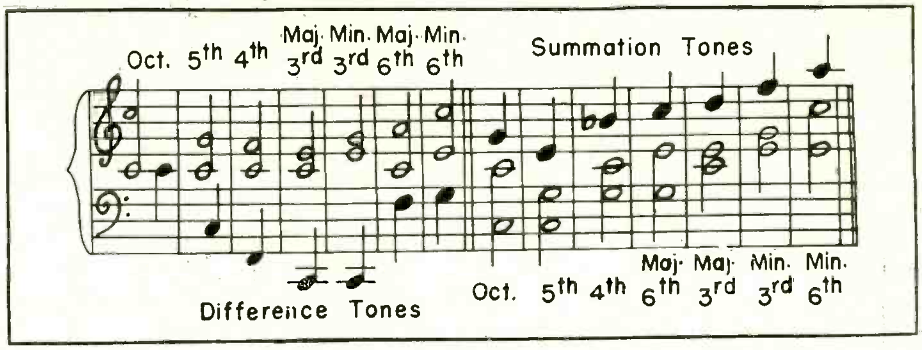


Fig. 9. Combination tones of the first order from simple musical intervals.

Figure 10 shows in music notation the first six measures of the music "My Country 'Tis of Thee", and beneath them the additional subjective first order difference tones with which the ear of the listener enriches it. This makes it evident that the hearing of music and its appreciation is a far more complicated matter than listening to simple tunes. And it points out the need for the reproduction of music to be at the highest fidelity.

If we represent the number of score notes in a harmony by *"n"* then the number of additional resultant difference tones will equal *n* times $(n-1)$ divided by two. Therefore, four-part harmony may add six subjective difference tones, and eight-part harmony might add twenty-eight, all of the first order. Larger symphonic harmonies would greatly increase these numbers, and when played loudly the difference tones between the higher partials would greatly multiply such subjective tones.

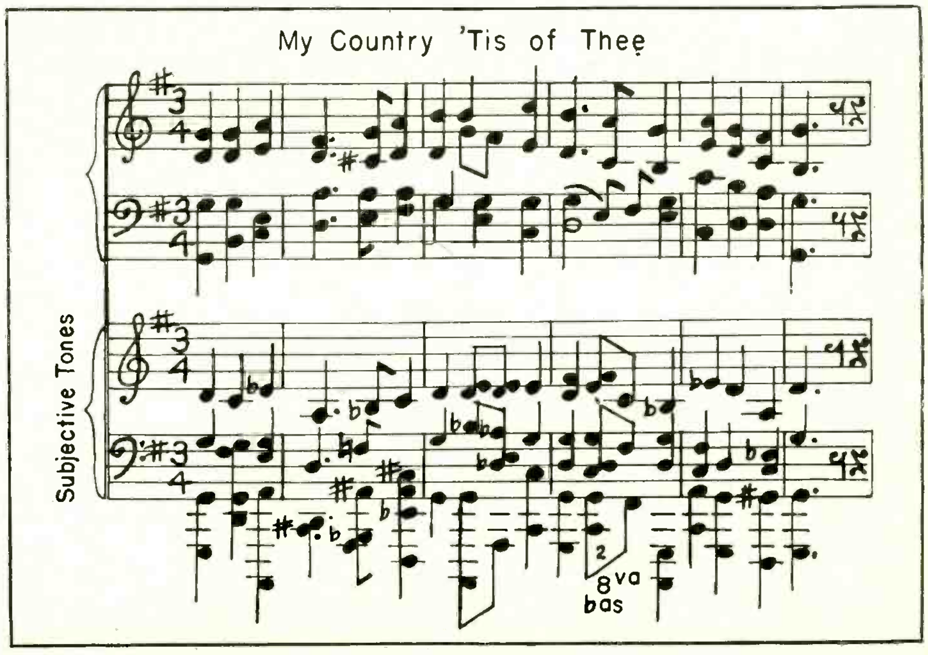
Much has been printed to decry the need for sound reproduction systems which give a faithful undistorted delivery of the high frequencies of musical tones. And listener reaction has been quoted in support of that viewpoint without evidence that the demonstrations did not suffer from intermodulation distortions, confusion of reflected tones, or proper balance of sound intensities to free the ears from the distortions of overloading.

The author's personal experience about thirteen years ago was quite to the contrary. During an auditorium demonstration of audio perspective by a system with an upper limit of 15,000 cycles per second, the quality of the music suffered greatly. It lost tone color and liveness. The snare drum lost its tingle.

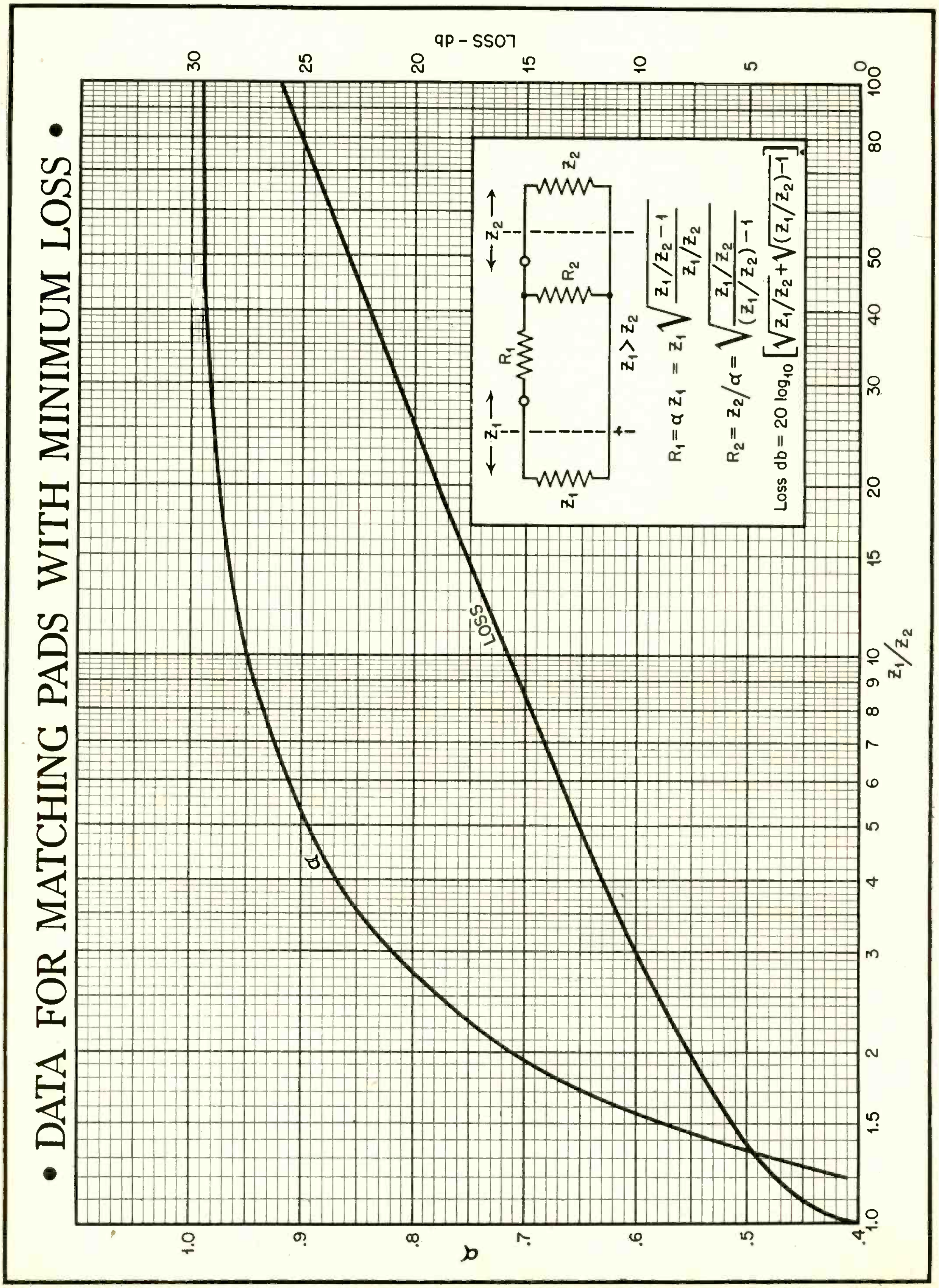
Because of the limitations of the grade of telephone lines between studios and transmitters, and because of the low musical quality of phonograph records, and of phono and radio amplifiers and loudspeakers, it is doubtful whether the public has the opportunity to hear musical fidelity any better today than the 8,000 cycle limit. So there is no chance for ear training to appreciate

[*Continued on page* 44]

Fig. 10. Music notation of a familiar composition and the subjective tones added by the listener's ear.



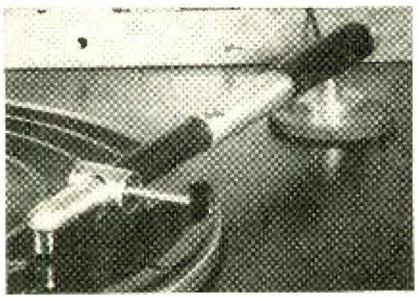
AUDIO DESIGN NOTES



NEW PRODUCTS

PICKUP ARM

Vibro-Master Company, 144 West 54th Street, New York 19, N. Y. A newly designed pickup arm, Type "K", for AM and FM broadcasting and other professional use. This arm combines the best in scientific design for use with the GE Variable Reluctance and Pickering Model 120M pickup cartridges, is equipped with



a new type spring loaded handle reducing stylus and record damage, arm resonance is less than plus or minus 1 db over the audio range of 40 to 15,000 cycles, and provides a proper stylus pressure of 22 grams for the GE and 15 grams for the Pickering.

This arm will not "walk" even on 30 cycle modulated grooves and will neither introduce nor amplify lateral distortion. The arm permits the playing of 16 inch transcriptions and requires a 5/8 inch one-

purpose. The output of this frequency-meter circuit consists of a steady voltage component which is proportional to the average or unmodulated frequency of the input signal and varying components proportional to the frequency deviations.

For further data, please write Furst Electronics, 800 W. North Ave., Chicago 22, Ill.

STUDIO RECORDER

A new studio recorder of console type for instantaneous or wax recordings is now in quantity delivery by the Fairchild Camera & Instrument Corporation. The table accommodates all sizes of acetates as well as 18 inch flowed wax masters. The positive 33⅓d rpm Fairchild drive guarantees absolute timing with its synchronous motor, making this recorder especially useful in motion picture work.

Fairchild's No. 541 magnetic cutterhead microscope, in combination with the precision-built lead screw mechanism, assure uniform cutting at any pitch from 80 to 160 lines—either in-out or out-in. The cutting pitch is continuously variable from 80 to 160, by means of a large, con-

MONOSET

Addition of a volume control unit for the Telex Monoset has been announced by Kenneth Dahlberg, sales manager of the Electro-Acoustic division of Telex, Inc., Minneapolis.

In the development of a volume control feature, Telex engineers have pioneered again in the electro-acoustic field. The Monoset itself is a distinctly different type of headset with under-chin type construction which resembles a doctor's stethoscope. The light weight makes it possible to wear the Monoset for long periods of time without noticeable ear pressure or head fatigue.

LOGARITHMIC VOLTMETER

Instrument Electronics, 42-17 Douglaston Pkwy., Douglaston, N. Y., is introducing the new Model 47 Voltmeter. It is an extremely sensitive amplifier-type instrument, featuring the voltage range of

Recorder Head

[*from page* 32]

heads were mounted in a composite head rotatable through 90 degrees. The user could rotate the heads to perform the desired function. This rotation automatically switched the amplifier from recording to playback, and disconnected the lead screw mechanism. The weight of the whole assembly was correct for recording, but in the playback position the pickup head was pivoted inside the housing so only its own weight rested on the record, the weight of the whole arm being carried on a stop which was automatically engaged in the pickup position.

By placing the head to one side of the record, the power was turned off, and by using quick-heating tubes the whole apparatus was completely controlled by using the head only, there being no need for any manually operated switches or gain controls. This feature had considerable appeal to the unskilled user.

Audio Amplifier

[*from page* 23]

trouble should be encountered in operation, if the above instructions are followed.

Coupling

The uses to which the amplifier is

but the values listed are conservative, and will be correct for most calculations. Bear in mind, also, that the gain is affected to by the tone and expression controls, and the frequency response is subject to almost any variation by means of the dual tone controls.

Power output— +38 vu or 6 watts with low distortion
+40 vu or 10 watts with 5% distortion

Gain — 75 db with expander-compressor off
86 db with expander fully on.
(based on 100,000-ohm grid impedance)
(values for unit not including 6SJ7 preamp)

Hum level— at least 68 db below rated output, with bass control in full boost position. With bass control in normal setting, no hum is audible even with the ear next to the speaker cone.

Response — ±1 db from 60 to 10,000 cycles
±2.5 db from 30 to 15,000 cycles
(The frequency response will be determined primarily by the output transformer, since all other stages are resistance coupled. These values are based on the medium-priced transformer used in the amplifier illustrated.)

Tone control— Bass control varies response from +9 db to —25 db at 60 cycles. Treble control varies response from +12 to —30 db at 7000 cycles. By means of the two fully variable controls, practically any response curve desired can be obtained.

Volume Expander

[*from page* 15]

is used to separate the signal channel from the side amplifier and the reactance tube. A filament transformer can be installed on the chassis if the amplifier with which it is to be used is unable to supply the additional 1.2 amps. If this particular chassis employed, the front panel should be spaced about ⅝″ away from the chassis, and the back panel must be moved forward by the same amount if the enclosing box is to be used.

Similar results can be obtained with octal tubes, using a 6SL7 for the preamplifier, a 6SN7 for the signal amplifier and expander tube, a 6B8 for the side amplifier (no single-ended tube is available in the octal line), and a 6SJ7 for the reactance tube.

[*Continued on page* 41]

[*Continued from page* 39]

Conclusion

The effect of this unit is rather interesting. At the conclusion of a passage of music containing high frequencies, the scratch fades out. There is no apparent change in quality, since the presence of high-frequency signal is sufficient to restore the frequency response to normal, but when there are no highs, there is no needle scratch. This unit is far from being a complete answer to the problem of scratch elimination, but it does make a contribution to reproduction from records that are especially bad. If the listener enjoys expanded reproduction, the circuit should enhance his pleasure immeasurably, because both scratch suppression and expansion are provided in a relatively simple circuit.

Planning a Studio

[*from page* 31]

plies, covers 115 volt a.c. Observance of these level groups determines in many cases how many conduits must be run from between any two points or how many separate cable groups must be made in a trough. An example would be a mike receptacle mounted in the base board and a monitor speaker outlet on the wall immediately above. It would be necessary to run two separate conduits from the rack to the location due to the level difference between the two circuits. Cables grouped into the above classification should be separated by at least three inches when run in troughs. Compromises can sometimes be made in short runs, such as running control circuits with high-level circuits.

Having all equipment indicated on the plan layout of the studio and control room, the actual location and position of troughs and conduits are drawn. Assuming a trough is to be used between the rack and the console this should be drawn first. Conduits are then drawn from all terminal points to the position of their connection. For example, according to the block diagram, the microphones connect first to a jack which is located in the rack. Thus the conduit or conduits running from mike outlets must go first to the rack. If the shortest length of conduit can be had by running direct to the opening under the rack, it should be shown that way. If the shortest path is to run the conduit to some point along the trough, this would be the logical point for it to terminate. The mike cable can then finish its run to the rack in the trough. A neater job of fanning out the wires to the rack terminal blocks can be done if all the pairs come out of the trough, rather than a number of conduits converging from various directions.

Checking off the circuits on the block diagram as each conduit is drawn will assure that none will be overlooked. If any doubt exists as to whether or not a conduit is needed, always put it in. In cases where there are three or more conduits between two points it is always good insurance to run a spare.

With all the conduits required shown on the layout it become necessary to determine what the trough dimensions should be and the size of each conduit. Since the trough will carry all circuit levels except power, there will be five cables in all. Separating these cables by three inches will require a total trough width of approximately 18 inches. The depth of the trough need not be more than three inches although it can be more if a greater depth fits in better with the floor construction. The following table gives a conservative number of pairs that may be pulled through various sizes of conduit.

Type of Wire	Number of Conductors per Conduit				
	½	¾	1	1½	2
Rubber Covered Mic Cable	1	2	3	6	9
Twisted Pair Shielded	2	4	8	15	30

Each conduit on the layout should now be numbered and a table made up that consists of the conduit number, its size, the number of pairs of wire and the kind of wire. To determine the size of the conduits a careful estimate of the number of pairs to be run in each will have to be made. A fairly accurate count of audio pairs can be made by using the block diagram as a guide. It is helpful to mark each circuit indicated on the block with the conduit number through which it is to run. Control circuits are more difficult to estimate at this time unless a complete wiring schematic of the relay interlock and lights has been made. If there is any doubt always assume on the greater side. From the number of pairs the size of conduit can be selected from the table given previously. The exact number of pairs to be run in each conduit or trough will have to be determined after the cross connection sheets are made. When these sheets are compiled the number of pairs to be pulled in each conduit can be added to the table. As to the kind of wire, all microphone leads should carry a rubber insulation over the shield. Audio and control circuits are wired with twisted pair shielded. 18 or 20 ga. conductors are sufficiently large. Power circuits should conform to the local electrical code.

[*To be concluded*]

TECHNICANA

RECORDING POWER AMPLIFIER

● A rather unique design for a high-quality recording power amplifier is described by Kurt Singer, of RCA, Hollywood, in the *J. Soc. Mot. Pict. Eng.* for June 1947. Capable of an output power of 10 watts with a maximum distortion of 0.5 per cent from 50 to 8,500 cps, this amplifier employs a unique feedback system, as shown in *Figure 1*. The amplifier comprises three stages, all push-pull, with 6J7 pentodes in the first stage, 6J7's as triodes in the second stage, and 6V6's in the output stage. The feedback loops extend from the 6V6 plates to the opposite side of the amplifier at the cathodes of the input stage.

Another unusual feature is the provision of a pair of resistances permitting the use of the amplifier when it is necessary to feed loads which must be operated from sources having impedances of 500 or 600 ohms. Since the amplifier uses 35 db of feedback, the output impedance is well below the nominal value of the secondary of the output transformer. When used to feed a loudspeaker, this amplifier is used without the resistors, thus providing a low impedance source.

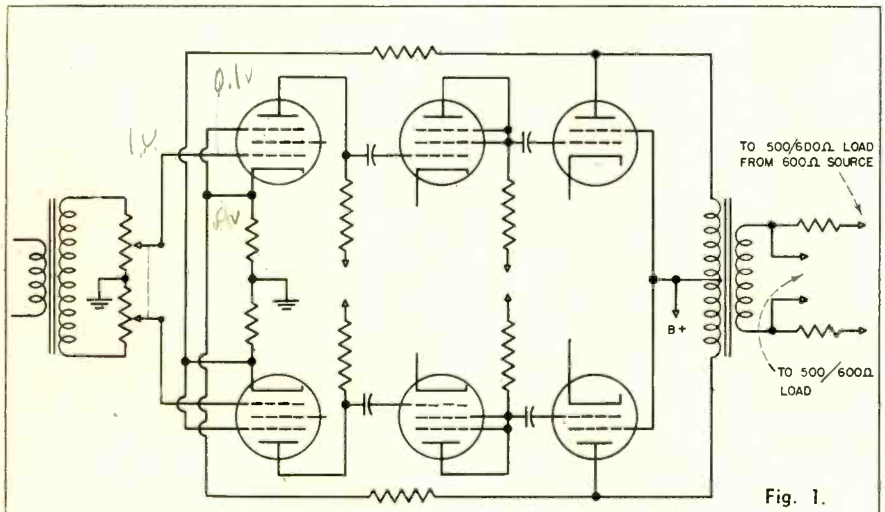


Fig. 1.

The method of arranging the feedback is claimed to reduce cross modulation, and to make the entire system independent of tube matching. It will be noted that this is an unusual arrangement for amplifiers using push-pull stages exclusively. At 2 db below full power output, the intermodulation is less than 0.5 per cent.

AMPLIFIER DESIGN

● The design of a high-fidelity amplifier must take into account a number of items which are not always considered, according to an article by F. Langford-Smith in the March-April issue of *Radiotronics* (Australia). One of the problems encountered when pentodes or beam tetrodes are used is that of compensating for the varying impedance of the speaker load. Two possible methods are available, without resorting to the use of feedback, which perform the required stabilization, but which are inefficient. Of these, the better is to employ an RC network between the plates, but this has the disadvantage of creating a low-Q resonant circuit for the tubes to work into. The other method is to use a shunt resistor which consumes a high percentage of the available output power, leaving the speaker load to receive only a small percentage. Obviously this is inefficient.

The use of a ported reflex cabinet improves the operation at the low-frequency end of the spectrum at speaker resonance, since it eliminates the single peak and replaces it with a double peak of considerably less amplitude. Still further improvement can be obtained by using two ported cabinets with speakers of different resonance points, thus improving the overall impedance

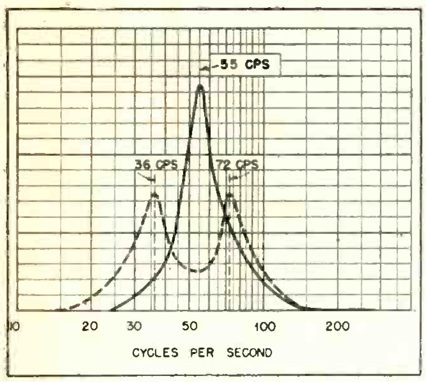


Figure 2

of the load. The two cabinets, when used, should be placed close together, and if impedance measurements are made, both should be connected. Operation of two cabinets widely separated does not give much improvement. Typical curves of impedance for a single speaker in a baffle are shown in *Figure 2* for open-back cabinets (solid line), and for reflexed cabinets (dotted line).

Record Revue

[*from page* 33]

(Altec theatre speaker system.) With the normal type of multi-mike pickup, especially with a soloist, the difference between control room sound and studio sound is usually very marked.

Needless to say there is still much music which benefits from and indeed demands multi-mike or close-to technique. But as FM and high-fidelity recording gradually take over there is no question that change there must be. The very fact that a solo oboist can now be picked up at a forty-foot distance, and most beautifully, is a challenge to experiment!

Recent outstanding recordings for the audio engineers: (*Number of records in parenthesis*)

Stravinsky, Dumbarton Oaks Concerto. Instrumental group conducted by Stravinsky. .... **Keynote Manuscript Series** (3 vinylite)

Vivaldi, Concerto Grosso in D minor. Group conducted by Alexander Schneider; Ralph Kirkpatrick, harpsichord. .... **Keynote Manuscript Series** (1 vinylite)

These two recordings, available soon, are first in notable new series. Very fine, European-style recording, plastic pressings, guaranteed to 12,000 cycles. Very live, brilliant effect. Over-all mike, no accentuation. Engineering data will be printed in each album. Others soon to follow.

Ravel, Daphnis and Chloe Suites #1 and 2. Paris Conservatory Orch. Charles Munch......... **London Decca EDA 29** (3)

Two impressionistic, brillant pieces in all-French recording. Typical "ffrr" acous- excessively live (though this is still ...er of taste). Brilliant effects here.

..., Concerti Grossi opus 6. Busch ... Adolf Busch, conductor. .... **...lumbia M 685** (3 vols. 25 records)

...ing recording of small group of ... made to sound full and brilliant by proper liveness. These volumes available separately. Fine example of music ideal for recording purposes. Interesting balance effects between several instrumental groups in the music.

Poulenc, Sonata for Trumpet, Horn, Trombone. G. Mager, W. Valkenier, G. Mager. ...**Night Music NM-103** (1 plastic)

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Electronic Organs

[*from page* 19]

no different with two keys depressed than with one. The same criticism applies when attempting to play in octaves on one manual, using, say, eight and four-foot stops taken from the same rank of pipes.

The second disadvantage lies in the fact that a rank of pipes must be very carefully balanced in loudness over its entire range. This process is known as "voicing". A stop, at its best, is voiced differently for different uses and, particularly, for different registers. Therefore, when a stop is to be used for many purposes, its voicing must be a compromise of all of them.

We may conclude this portion of the article by stating that the pipe organ has attained a fairly high degree of perfection as a musical instrument. Its chief musical limitation is in the mechanical inertia of its action, which is an impediment to nuance, and to the execution of rapid musical passages.

Chief among its other disadvantages are its great expense, bulk and weight, the frequency with which it must be tuned and repaired (and the difficulties in doing both), and the amount of power which it consumes (50 kw is not uncommon for large organs).

[*To be continued*]

U. N. P-A Systems

[*from page* 29]

tunate to get uniform coverage without destroying the illusion. Because of the physical construction of the auditorium this is the only method of handling this loudspeaker problem, although the use of multiple loudspeakers is poor practice when physical conditions permit a single source. Tests conducted with a General Radio noise meter using a 400-cps warbled note showed a pattern of coverage throughout the auditorium to be within 2 db. All speakers are painted to harmonize with the background and the overall result is highly satisfactory.

All amplifiers, tubes, microphone

stands and loudspeakers were of RCA manufacture fabricated in Commercial Radio-Sound's New York plant. Everything but the power amplifiers for the loudspeakers were the RCA broadcast line of plug-in amplifiers. The construction of the two control consoles is unique in the fact that the sides and top of the console are totally removable permitting complete access to the interior for servicing and installation.

One other interesting feature is the use of master gain control for both the regular and emergency channel using a tandem balanced-ladder attenuator.

Musical Acoustics

[*from page* 37]

higher fidelity for those who cannot frequently attend instrumental concerts. Such is the curse of recorded and broadcast music. Although FM could do better it has not met its opportunity because of the above shortcomings associated with it.

But the author's most interesting observation was his dissatisfaction with music which lacked frequencies above his range of hearing; and the presence of which, because of resultant subjective difference tones, enriched the music for him.

In the consideration of high fidelity, high audio frequency reproduction, it seems that research should be conducted concerning the influence of supra-audible overtones upon the enrichment of musical qualities. It appears that such a study has been overlooked or neglected.

Corrections

In part I of "Musical Acoustics," in the June issue of AUDIO ENGINEERING, the following correction should be made in Table II, page 35: The fraction reading 15/8 between B and C should read 16/15.

Also, in Part I, page 36, right-hand column, seventeenth line from the bottom, the word "distinguished" should read "diminished."

In Part II in the July issue, due to a printer's error, the last line of the article was dropped from page 42. The last sentence should read: "From the bases of the 'rods of Corti' auditory nerves lead to the brain."

Preference Tests

[*from page* 12]

played on it (the result of poor transient response).

In spite of the last two paragraphs, recordings can be used without impairing results if they are sufficiently good. Chinn and Eisenberg showed that lacquer originals, properly recorded and reproduced, are suitable. Does this mean that the FM station's record library of the future will use direct recorded lacquer originals, incidentally? Whether transcriptions processed and pressed from these originals would be suitable, has not yet been proven. It would certainly demand the utmost care in the processing plant.

Phonograph records need to be chosen with great care. Depending on recording source, date, location, processor, we have the following variables not instantly pinned down by a glance at the label number; and subject to wide variation due to the world-wide character of the industry:

1. Studio acoustics
2. Microphone placement
3. Cutting head characteristics
4. System equalization as affected by the recordist's mood
5. Quality of processing
6. Stamper wear
7. Pressing stock and characteristics

To pick at random, a master cut in Birmingham concert hall by EMI in 1934, and repressed for the first time say in 1942 by Columbia, of which we might get the four-hundredth pressing in laminated process shellac from stamper 3 (source, mother 2) is a product whose *exact* characteristics could be defined by a specialist after some thought. It could not be defined by the non-specialist engineer, hence he could not even set up the appropriate equalization much less estimate whether the intermodulation distortion due to stamper wear was 5% or 25%. And if the next record played were recorded by Victor in New York in 1947 and pressed on vinylite, an entirely different equalization curve would be necessary. As Chinn has so ably pointed out, the phonograph record field is in a state of chaos, as regards recording characteristics.

Assuredly, rigorously controlled tests can be made with phonograph records, but not without the intimate supervision of a specialist, and even then with little right to generalize beyond the record field.

Finally, we come to the most variable element in the system, the loudspeaker. None of those reporting has given even a free field response curve on the speaker used, but such a curve would not be complete evidence, for we have the known fact that loudspeakers nominally alike in frequency response curves ma sound quite different. Many have attri uted this to excessively optimistic a vertising departments, but the tro is more fundamental. The writer re bers running curves on two two- units, and finding them alike quency response. Unfortunately tening to programs with th sounded like its curve (exceedingly uniform in response), whereas the other had strong peaks at 150 and 2500 cycles and was rather muddy-sounding everywhere else. We could talk glibly about differing crossover point, Doppler effect, breakup, and damping factor, but not pin the matter down quantitatively. It has recently been explained by an English study[12] which is well worth reading, but in this country we have only "calibrated" human ears to study transient response. It is evident that even a *complete* set of *conventional* steady state response curves does not *completely* define system performance.

Transients

Repeated reference has been made, above, to transient performance as a neglected factor. It can be said, very safely, that such neglect is prevalent throughout most engineering circles. So long as the engineer does not demand quantitative data on transient effects, the manufacturer will neglect this important but troublesome problem.

Even the response curve given in the Chinn and Eisenberg paper gives by its shape a suggestion of probable transient effects in the frequency region near cut-off—precisely the region which many observers found so objectionable. In short, steady state measurements are only half the story.

Incidentally, judging from some of the discussion of these papers, there is a serious misconception of motion picture practice, exemplified by mention of the "SMPE standard response curve". Actually, there is no "SMPE standard response curve", and we assume that the reference is actually to a standard of the Research Council of the Academy of Motion Picture Arts and Sciences, which was later reprinted in the Journal of the SMPE[19]. This has been cited as evidence of theater audiences' antipathy to high frequencies, since it shows strong high frequency attenuation.

In fact, its significance is rather different. The curves are electrical reproducing curves, measured as the amplifier electrical output (feeding a resistance load) when running a standard Research Council test film. These curves were adopted to standardize theater systems so that the recording studio could equalize for standard reproducing conditions. Whereas the NAB specified a standard transcription *recording* curve, [the] Research Council has specified *re[pr]oducing* curves for systems using va[rio]us specific loudspeakers. The motion [pictu]re studio is free to use whatever [...]ing equalization it deems will give [...] [re]sults. In short, the Research [...] standard ensures minimum film [...]d uniform reproducing condi[tions ...]e amount of compensating pre-equalization which the studio uses in recording is limited by the mechanics of the system, and also by the intermodulation distortion in release prints. To judge by the many SMPE articles, excessive intermodulation due to slipshod film processing control is more a matter of commercial carelessness than of chemical engineering limitations, and hence likely to be alleviated when laboratory competition becomes stronger.

There has been a general tendency to cite harmonic distortion figures of three to five percent as just recognizable. These seem to be based on a 1933 publication by Massa[18]. This is an exceedingly weak reed to lean on, for the tests covered only speech, and the amount of residual distortion of the amplifier alone was two per cent. In other words, the tests really showed that an *increase* from two to about five amplifier per cent would be just noticeable. The system had no true "undistorted" condition. The test technique reflects great credit on Massa in view of the then undeveloped state of the acoustic measuring art, but it is certainly outmoded fourteen years later. The few subsequent studies are equally subject to doubt.

Psychology

This is a consumer reaction problem, and we have to be careful that the dice are not unwittingly loaded by an unfortunate choice of words[14]. For example, see the words used to characterize the schools of thought: debunkers and idealists. The semanticist would say that these were loaded, biased words beyond a doubt. Remember also what happened when makers of men's toiletries asked men whether they would prefer *perfume* in a shaving lotion; the thundering verbal answer of *no* was entirely reversed by giving men a choice of perfumed and non-perfumed lotion. The word *perfume* has the wrong connotations, perhaps. We are entitled to know the exact words used in presenting the problem to the subjects.

A very basic question is likewise whether the method of choice correlates with actual use conditions. Do we set the tone control by a multiple choice vote? No, we turn the knob back and forth, listening to the sound, until a pleasing setting is found. We may change the setting later. The relation between cause and effect is always clear. Given an individual control in a small room, a chance to experiment, and some time, what would be the result? What is the learning factor, if any? This brings us to the next question, what are we really seeking to find?

[*Continued on page* 46]

In the midst of all the current furore there has been no attempt to get down to fundamentals and decide what it is most important to find. We might characterize the choice enigmatically as the confused reaction versus the considered reaction. So far all tests have been concerned with the confused reaction — a rapid representation of alternating ten or twenty second choices. This possibly correlates with the basis for the choice of a twenty dollar radio. Is it a good correlation with the process of choosing a console set, or with conditions of use? If we give the user a free chance to adapt to both choices for a week or two, what will his reaction be?

Which portion of the public is most important to us; or alternately, how should we weigh the results from the following:

a. The soap opera addict — interested in intelligible speech
b. The serious listener to music—whether the three Bs or boogie woogie
c. The background music devotee — desiring soft, muffled, definitely unobtrusive music.

Fatigue Factors

The question of immediate versus delayed reaction has one aspect which has been completely overlooked in all discussions: the fatigue factor. Why does a listener so often turn a set off after one or two programs? Why do so many listeners use weak, muffled music as a subdued background, but never really listen to a program? We have the artistic side, of course; but the writer believes the fault is often more basic: Listening to the *average* set is *very* fatiguing, and some sets are much worse than others.

The writer made a considerable number of experiments with fatigue in radio and record listening, and isolated some fatigue factors. In many cases the effect can be predicted quantitatively, but the remainder can be evaluated only on a relative basis, viz., A is little (or much) better than B as measured in a one (or five) hour test period. He made a curious discovery: One of our most successful low priced radio lines was distinguished by its low fatigue factor *as compared with its competitors' products.* How important was this in its success? We can only guess. Perhaps it gave skilful salesmanship just the extra lift needed to create a dominating position in the low priced field.

The fatigue factors just mentioned refer but lightly to the classical case (fatigue of the hearing organ itself[20]); they mainly govern fatigue of the central nervous system due to the nature of the impulses fed it by the hearing organs. In short, it is brain fatigue[22] rather than fatigue of the organ of Corti which concerns us here. A broad consideration of fatigue factors is beyond the scope of this article, but they include the following primary defects:

1. Harmonic and intermodulation distortion
2. Certain frequency-amplitude relation anomalies
3. Transient distortion
4. Noise
5. Acoustical nature of the space used

Some of these have been discussed by the writer[21] in relation to fatigue of hearing aid listeners, perhaps the only application in the last several years where reduced fatigue has gained considerable commercial recognition. Harmonic and intermodulation distortion heads the list only because it is often so bad as to mask other faults. The moment it is minimized, the other factors assume importance. It should also be remembered that "distortion" is not a dimensionless entity, that certain orders of harmonic distortion are much more offensive to the ear than others, even when the distortion meter readings are equal.

Hearing Aid Improvement

That these principles were of general application was shown in this field not too long ago. The writer was called on to overhaul a hearing aid line, and the problem was found to be not appearance, pride, or economy, but wearer fatigue. Wearers could grade out units of identical frequency response but different fatigue characteristics with an accuracy and consistency which was surprising. Production units were changed accordingly, with no change in visible appearance, in frequency response, in sales procedure, or sales personnel. Within six months sales had doubled, and both customers and dealers were much happier; all of this at a time when other manufacturers were having an acute loss of sales, were cutting prices, and were using extensive advertising campaigns.

Remember that about 50% of hard-of-hearing customers have hearing which is normal in every respect save acuity — that is, their hearing is like that of broadcast listeners except for a loss *which is substantially constant with respect to frequency.* These data are therefore 50% directly applicable to normal listeners.

The improved instrument had several interesting properties:

1. The average time of use increased from nine or ten hours per day to twelve to sixteen hours.

2. Wearers were able to use mo low and high frequency response w comfort (this could be adjusted by dealer on request but could not re be tampered with by the custo

3. Without being told of the users spontaneously commente proved "presence" and "easi ing".

4. Ordinary articulation testing methods would not show any improvement in intelligibility, probably because the articulation test period was too short. Perhaps twelve-hour test periods should have been used.

Remember that all these observations were made using the relatively narrow band of 150 to 4500 cycles, wherein the average engineer thinks the annoyance power of distortion to be less than in a broad band system. Remember also that there is a powerful urge to disregard fatigue and continue using a hearing aid — so if excess fatigue would cut use-time of an essential instrument by one-third, what does it do to the use of the ordinary radio which can be taken or left alone?

The writer recalls seeing the effect of replacing a number of home radios by commercial units of what were later found to be *relatively* lower fatigue designs. The daily hours of use approximately tripled, and the increase persisted long after the novelty effect had worn off.

Fatigue effects have also been observed in connection with 16 mm sound on film reproduction.

What Percentage Is Significant?

Remembering that we are unlikely to get a 100% vote for any probable single bandwidth, what should be done? Undesired excess bandwidth can be removed by the radio set's tone control. Missing kilocycles cannot be so restored. On this basis, what bandwidth would be necessary to satisfy what per cent of the listeners? To irritate our readers into creative thought we have utilized the Chinn and Eisenberg data for 5 and 8 kc., and split the remainder in a grossly implausible manner:

Upper Cutoff Frequency	% of Satisfied Listeners
5 kc.	40
8 "	80
10 "	90
12 "	99
15 "	99.99
20 "	100

Seriously, it is evident that to satisfy 80% of the listeners to speech and all types of music, an 8 kc. range would have to be *available* (which they could decrease as desired). If it becomes available throughout the country, the present vicious circle may be broken, and it may foster the increased production of better quality home radios.

Industry's Stake

Many parts of the electronic field ve a considerable stake in a correct swer to the questions propounded:

he Broadcast Network—By the time am has gone a thousand miles source over lines of present characteristics it is no longer quality. Would replacing the kc. lines by 8 kc. service, and reducing intermodulation and phase distortion, increase the *number* of listeners (at any given time) enough to pay for increased line costs? How far would such increased quality go in enabling networks to better resist inroads of the 8 to 10 kc. transcription?

The Phonograph Record Maker — Right now he is riding the crest of a boom, and even pressings from worn-out stampers, pressings with intermodulation distortion as great as 20 to 30%, can be sold. But looking more to the future, would it be worth while to improve care in processing to make the finished product sound more like the test pressing?

The Radio Set Manufacturer — A set with reduced fatigue factor costs no more to make than the present designs. A set with somewhat increased bandwidth as well as reduced fatigue factor would cost very little more. It has been profitable for one company to do a better job in the low-priced field. Would it pay others to do the same? A large number of sets have been advertised as "high fidelity"; almost none of them have actually been so. How profitable would it be to substitute a little performance (lower fatigue at least) for a lot of claim? Would it pay the NAB to subsidize set makers to the extent of say $1 per set (payable only for a low fatigue design), to increase the listening time?

The FM Broadcaster—Right now he has 15 kc. speech input equipment and transmitter, but usually 9 or 10 kc. transcription tables. Assuming that better FM sets could be sold, would it be worth his while to change to 15 kc. pickups, and to push the transcription makers for wider-band recordings? Or would it be better to push them for a higher average of pressing quality with the present bandwidth? Many transription pickups have poor quality from 7 to 10 kc. under certain conditions. Would it pay the station to install more modern units?

These have not been rhetorical questions; they face the industry now, and their answers may determine some profit and loss statements two years from now.

The average manufacturing organization can secure an answer for itself, limited to its own conditions, with moderate ease, but it would not be the universal or completely generalized answer. The hearing aid industry (in part) has found such a policy, applicable to itself, for example.

However, the completely generalized problem is too massive, too tied up with self interest. Particularly as regards the broadcasters, a decision which would enjoy broad acceptance is essential, in fact it is most important that the radio industry agree on its broad objectives. The technical facilities are easy to borrow, rent, or construct. What we require most, and which is hardest to find, is an organization with no adverse commercial commitment (ruling out some colleges), and *with a chief skilled in auditory opinion research.* It seems to the writer that this points unmistakably in the sole direction of Professor Harold Burris-Meyer.

Conclusions

For too long have we made reproducing devices which (as Burris-Meyer says) *remind* the listeners of music, but do not reproduce it. Also, for too long have we made sets of which only engineers would approve. Let us concentrate on designs which the public approves of. Let us be conscious of presence—the public is. Let us forswear "high fidelity" designs which reproduce only between 25 and 2500 cycles, or between 500 and 7000 cycles, or from 100 to 1000 and 3000 to 7000 cycles. Let us forswear dealing with loudspeaker manufacturers who offer only an interesting price—but have neither complete test equipment nor adequate quality control. Let us stop blaming "bad public taste" for the lack of success of inadequately engineered sets. Note that the public's dislike of false tones far overpowers its dislike of missing tones.

Let us eschew the easy generalization, the test which proceeds from an incorrect assumption to a foregone conclusion. Let us buckle down to a real study of the finer *dislikes* of the ear. *Let us adopt the principle that the true measure of the quality of an electro-acoustical system is the maximum band-width which the public finds acceptable.*

References

1. H. F. Olson, Frequency Range Preference for Speech and Music; Talk before Acoustical Society of America, May 9, 1947.
2. W. B. Snow, Audible Frequency Ranges of Music, Speech, and Noise; *JASA,* Vol. 3, No. 1, pp. 155-166, July 1931.
3. H. A. Chinn and Philip Eisenberg, Tonal Range and Sound-Intensity Preferences of Broadcast Listeners; *Proc. IRE,* Vol. 33, No. 9, pp. 571-581, Sept. 1945.
4. E. Massell, W. Van B. Roberts, C. C. Eaglesfield, B. Groom, H. A. Chinn and P. Eisenberg, Discussion of Chinn and Eisenberg Paper; *Proc. IRE,* Vol. 34, No. 10, pp. 757-761, Oct. 1946. L. V. Wells, T. Powell, same; *Proc. IRE,* Vol. 35, No. 4, p. 378, April 1947. James Moir, same, *Proc. IRE,* Vol. 35, No. 5, p. 495, May 1947.
5. N. D. Webster and F. C. McPeak, Experiments in Listening; *Electronics,* Vol. 20, No. 4, pp. 90-95, April 1947.
6. B. B. Bauer, Crystal Pickup Compensation Circuits; *Electronics,* Vol. 18, No. 11, p. 132, Nov. 1945.
7. George M. Nixon, Higher Fidelity in Sound Transmission and Reproduction; *JASA,* Vol. 17, No. 2, pp. 132-135, Oct. 1945.
8. Willis C. Beasley, Characteristics of Hearing Loss in Various Types of Deafness; *J. SMPE,* Vol. 35, pp. 59-85, July 1940.
9. J. C. Steinberg, H. C. Montgomery, M. B. Gardner; Results of the World's Fair Hearing Tests, *JASA,* Vol. 12, No. 2, pp. 291-301, Oct. 1940.
10. S. S. Stevens and Hallowell Davis, Hearing; pp. 209-217, Wiley (1938).
11. Harvey Fletcher and W. A. Munson, Loudness, Its Definition, Measurement and Calculation; *JASA,* Vol. 5, No. 2, p. 91, Oct. 1933.
12. D. E. L. Shorter, Loudspeaker Transient Response; *BBC Quarterly,* Vol. 1, No. 3, pp. 121-129, Oct. 1946.
13. J. P. Maxfield and W. J. Albersheim, An Acoustic Constant of Enclosed Spaces Correlatable with Their Apparent Liveness; *JASA,* Vol. 19, No. 1, pp. 71-79, Jan. 1947.
14. Elmo Roper, Survey Pitfalls; *Fortune,* Vol. 35, No. 4, p. 6, Apr. 1947.
15. D. F. Seacord, Room Noise at Subscribers' Telephone Locations; *JASA,* Vol. 12, No. 1, pp. 183-187, July 1940.
16. Daniel F. Hoth, Room Noise Spectra at Subscribers' Telephone Locations; *JASA,* Vol. 12, No. 4, pp. 499-504, Apr. 1941.
17. O. B. Hanson, Comments on High Fidelity; *Electronics,* Vol. 17, No. 8, pp. 386-387, Aug. 1944.
18. Frank Massa, Permissible Amplitude Distortion of Speech; *Proc. IRE,* Vol. 21, No. 5, pp. 682-689, May 1933.
19. Research Council, Academy of Motion Picture Arts and Sciences, Revised Standard Electrical Characteristics for Two-Way Reproducing Systems for Theaters; *J. SMPE,* Vol. 32, No. 2, pp. 213-216, Feb. 1939.
20. S. S. Stevens and Hallowell Davis, Hearing (Wiley, 1938); pp. 217-220.
21. C. J. LeBel, Hearing Aid Technic; *Electronic Industries,* Vol. 4, No. 1, pp. 104-106, 198-205, Jan. 1945.
22. A. C. Matthews, High Fidelity; RADIO, Vol. 29, No. 11, pp. 33-37, Nov. 1945.
23. Report on Dr. Harry F. Olson's Listener Preference Tests, AUDIO ENGINEERING, Vol. 31, No. 5, pp. 27, 43-44, June 1947.
24. N. M. Haynes, Listening Tests; *Electronics,* Vol. 20, No. 6, pp. 268, 270, June 1947.
25. Harvey Fletcher, Hearing, The Determining Factor for High-Fidelity Transmission; *Proc. IRE,* Vol. 30, No. 6, pp. 266-277, June 1942.

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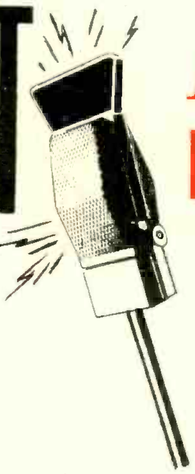

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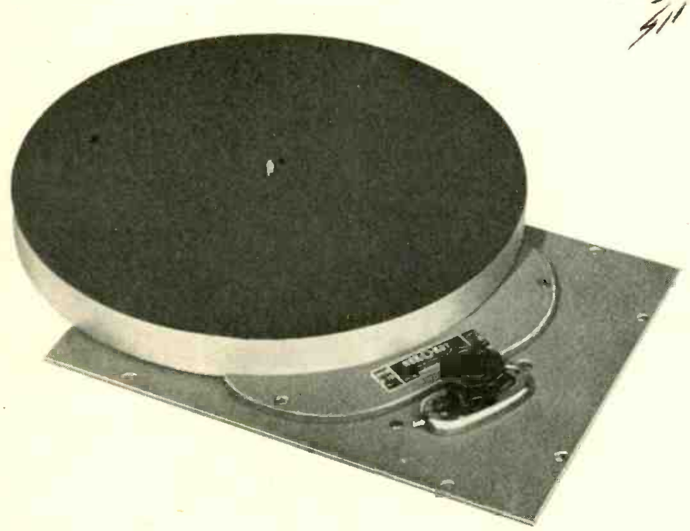

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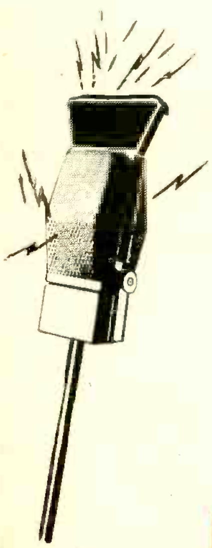

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