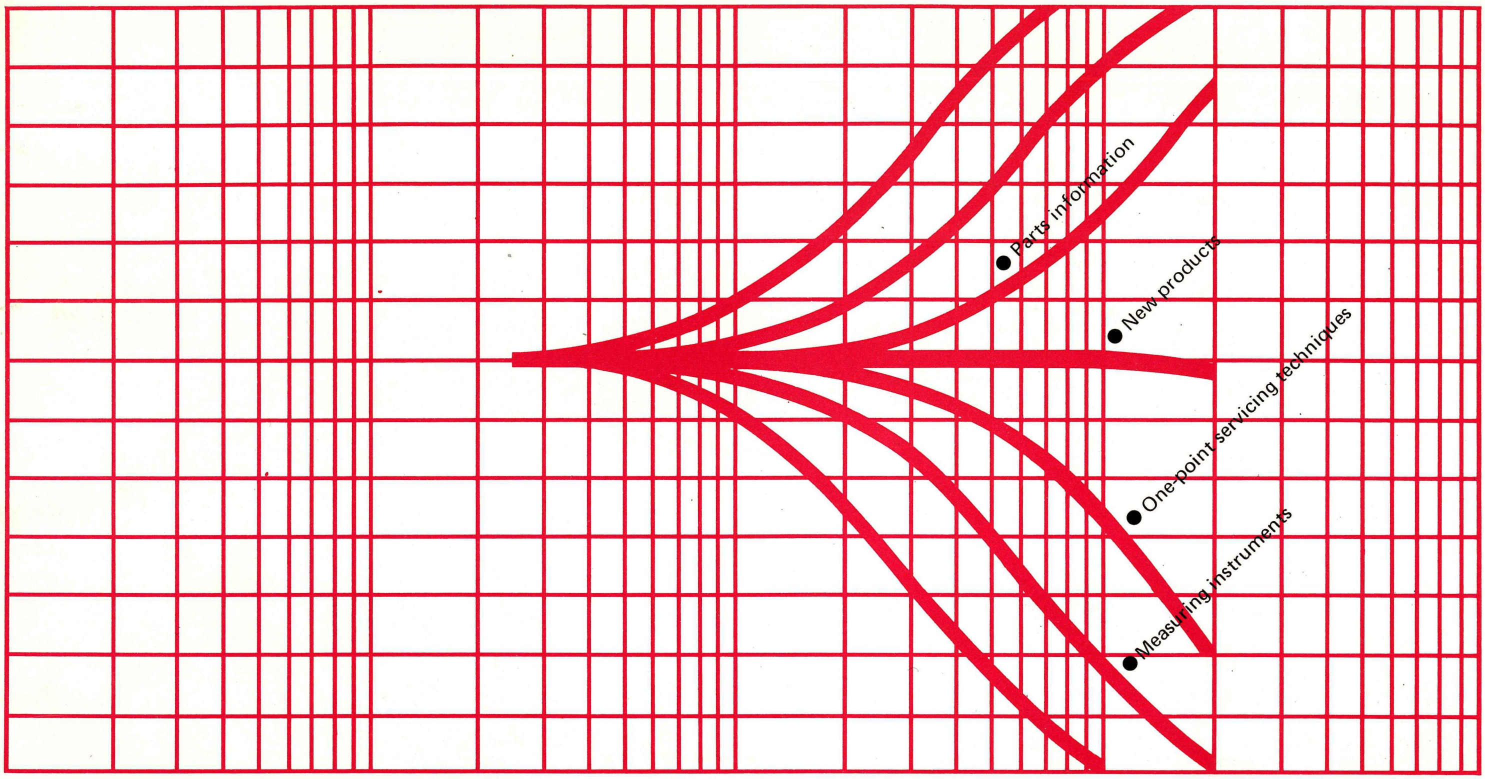
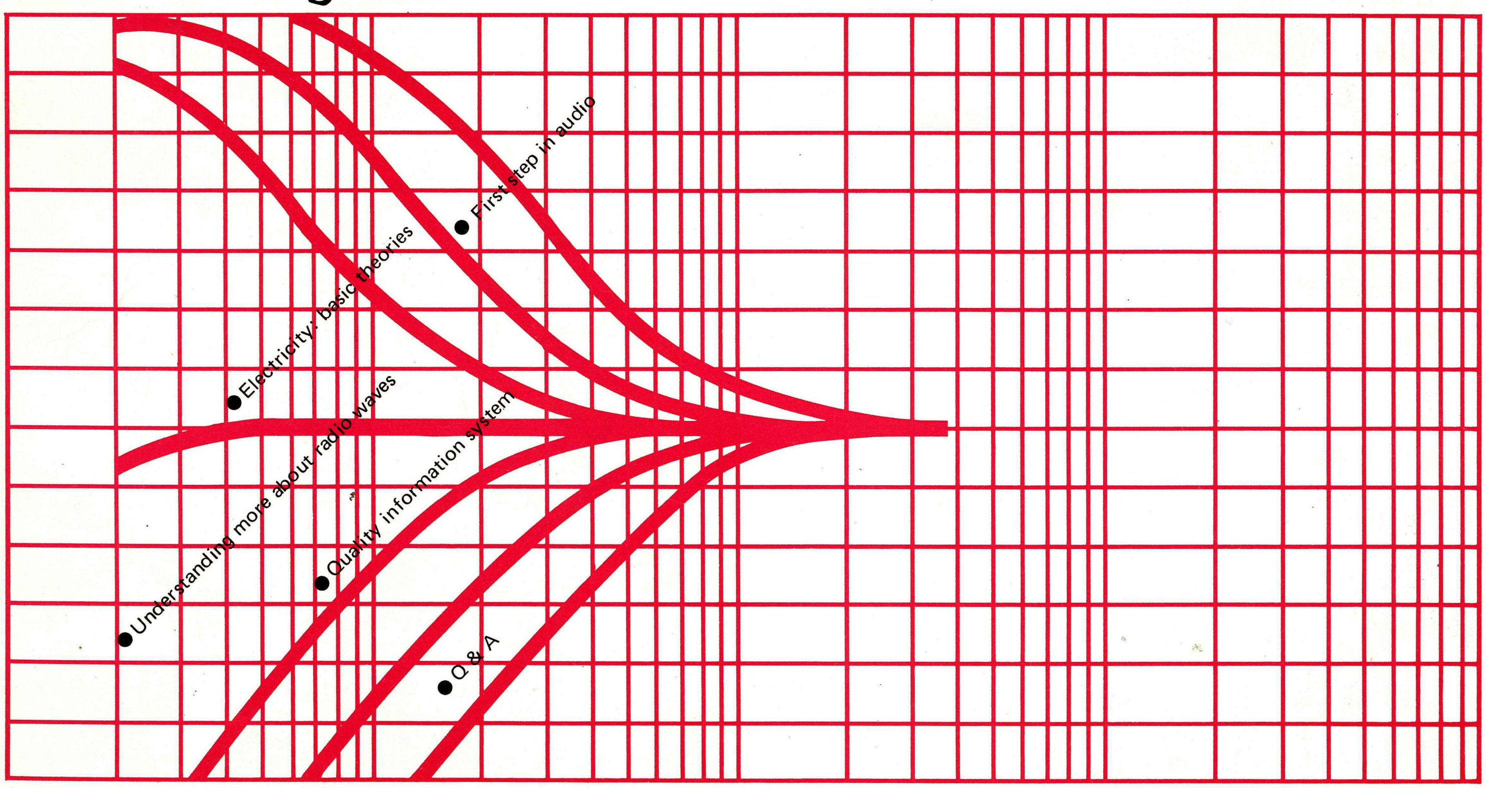


TUNING FORK

Audio service guide

No. 3



PIONEER®

CONTENTS

Parts Information (3)

Transformer and Coil

1. **Power Transformers**

Power transformers are found in various kinds of electronic equipment to change commercial electric power into the kind of power required by the equipment. They also serve as insulators to protect the secondary power source from the primary. The transformer itself is made of laminated iron sheets at its core, covered with the primary winding and then with the secondary winding (called N_1 and N_2 respectively). The core is laminated to reduce loss made by eddy current. When AC voltage (E_1) is applied to the primary winding, mutual induction causes an AC voltage to flow through the secondary winding. The relation between the relative voltages in the two windings varies according to the relative number of turns in the two windings:

$$E_1:E_2 = N_1:N_2$$

Therefore any secondary voltage value can be obtained for a given primary voltage by varying the ratio of turns between the two windings.

Power transformers fall into one of three groups, according to the structure of their iron cores:

1. EI core
2. Cut core
3. Ring core (toroidal)

1-1 EI Core

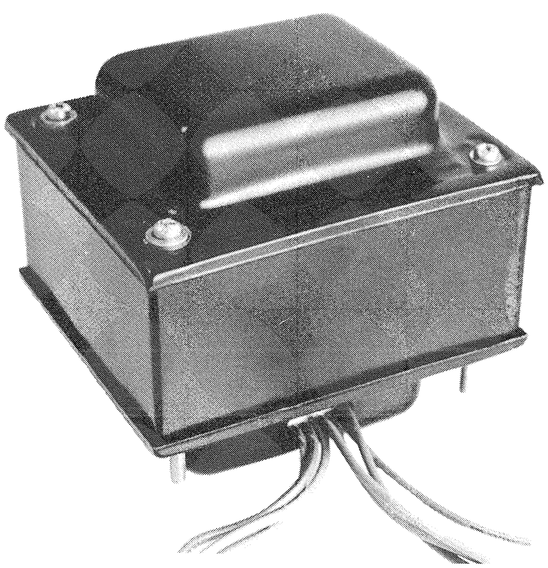


Photo 1

Presently, this is the most widely used type of transformer. It is well-suited to mass production, and because automatic winding machines can be used, the production costs are low. A drawback, however, is the amount of leakage flux caused by loss in magnetic energy.

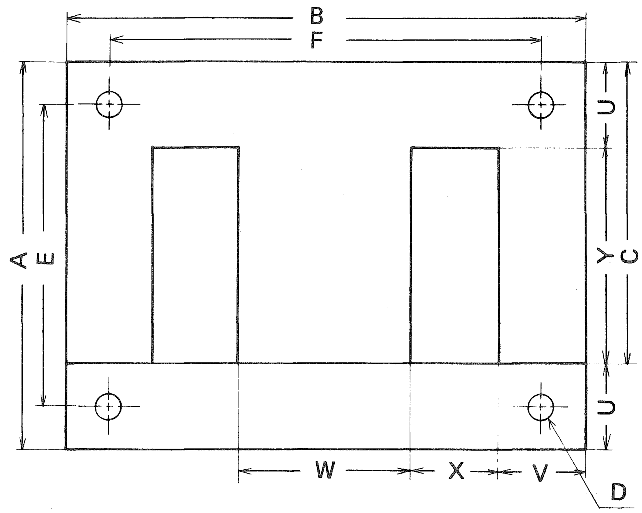


Fig. 1 The measurement of core

Type	A	B	C	D	E	F
EI−48	40	48	32	*	*	*
EI−54	45	54	36	*	*	*
EI−57	47.5	57	38	*	*	*
EI−60	50	60	40	*	*	*
EI−66	55	66	44	*	*	*
EI−76	68.5	76.2	50.8	5	50.8	64
EI−85	71.5	85.8	57.2	5	57.2	71
EI−96	80	96	64	6	64	79
EI−105	87.5	105	70	6	70	87
EI−114	95	114	76	7	76	95
EI−133	111	133.2	88.8	7	88.8	111

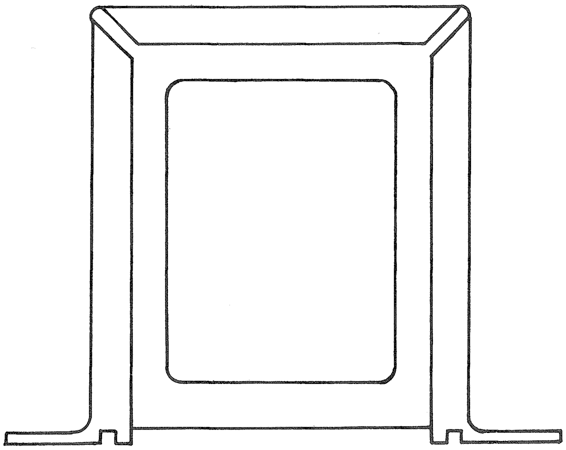
Unit : mm

$U = V = X = A - C \quad W = 2 \times U \quad Y = C - U$

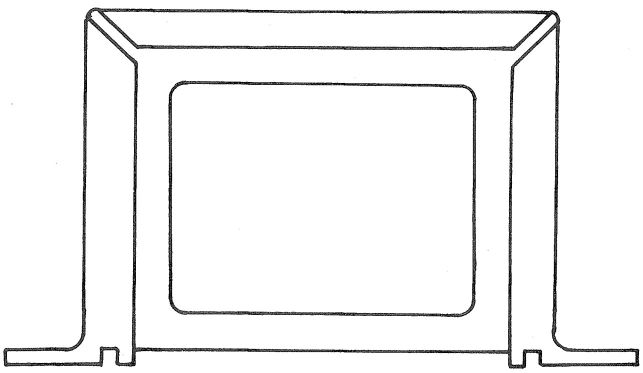
Table 1 The dimensions of cores

EI-cored transformers are classified by their measurements and their method of installation. Fig. 1 indicates the dimensions of the core as prescribed by the JIS (Japanese Industrial Standards). The measurement B becomes the name of the core, e.g., EI-48. Transformers of this type are also classified by the size of the core and its structure:

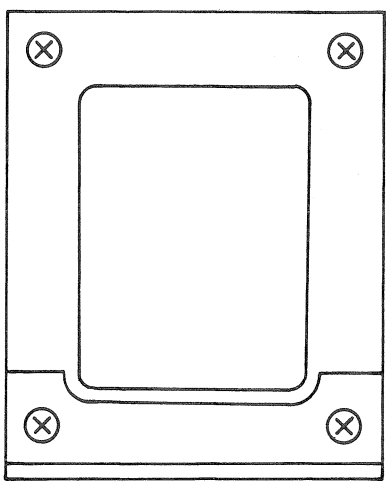
Band cover type (up to EI-66)
Angle type (over EI-66)
Laying type (over EI-66)



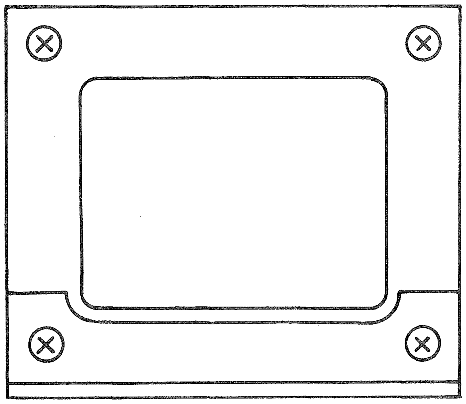
Band cover type (vertical)



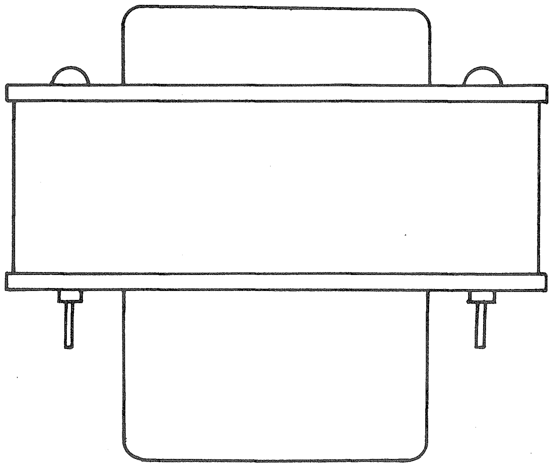
Band cover type (horizontal)



Angle type (vertical)



Angle type (horizontal)



Laying type

Fig. 2 The types of core

With regard to the magnetic material used for the core, Highlight or Orient is prescribed by the JIS standard (C-2552 and C-2553). Highlight is a core material made from a steel sheet of ferroalloy containing less than 3.5% silicon and then treated with heat and cold rolling alternately. It has no magnetic directivity. The surface of the steel is finely finished, and the volume factor is excellent. As an alternative, sheet steel containing from 3% to 3.5% silicon is rolled into plate one to two millimeters thick by a heat strip process and is then made into plate 0.3 to 0.35mm in thickness by alternate cold stripping and annealing. Magnetic directivity is applied by setting magnetic poles along the axis of the plate rolling. A core with magnetic directivity exhibits maximum magnetic properties in the direction of the strip, and becomes progressively weaker as the angle to the strip direction becomes greater (Fig. 3). Comparing this type of core to the first, the iron loss is smaller and relative magnetic permeability is larger.

1-2 Cut Core

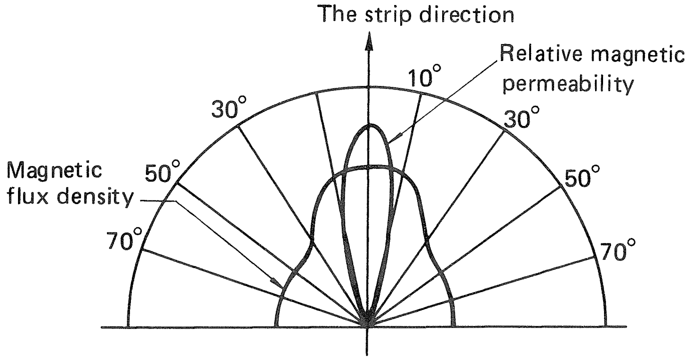


Fig. 3

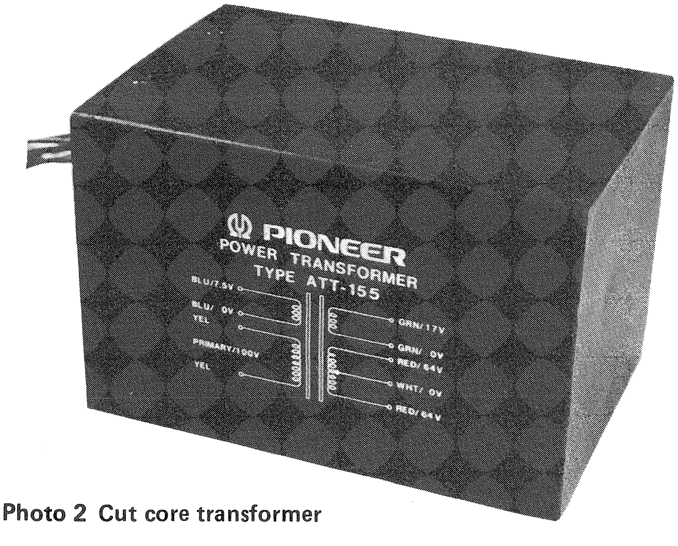


Photo 2 Cut core transformer

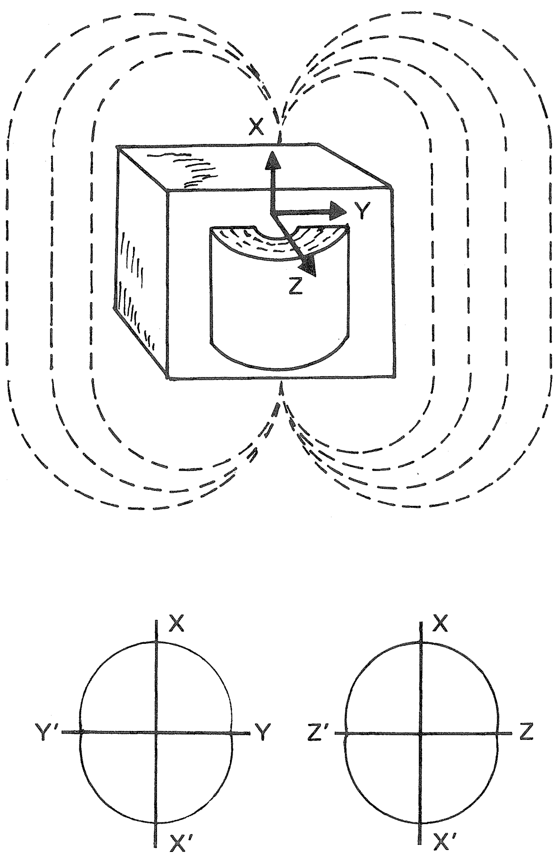


Fig. 4 The position of leakage flux

The structure of the cut core is unusual and is not widely used. However, the cut core type of transformer exhibits low leakage flux and is more efficient than EI cores and usually more compact as well. For these reasons they are often used where a high-class image is wanted or where EI cores are impractical for reasons of space. On the debit side of the ledger, cut core transformers don't lend themselves well to mass production, and so costs are high. For example, the cut faces are polished to mirror smoothness and each pair of faces must be precisely matched. Another production problem is that the windings must be balanced if the efficiency of the design is to be realized.

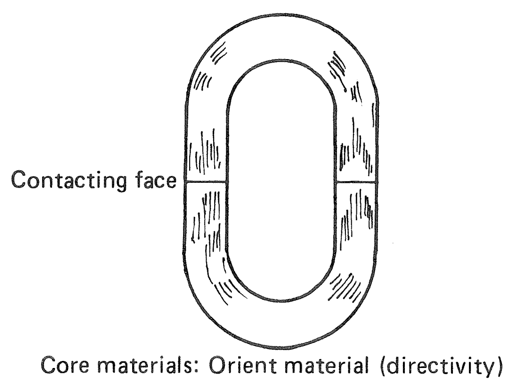


Fig. 5 Cut core

The shape of the cut core is shown in Fig. 5. The number in the name of the core refers to its maximum handling power. The following twenty cut cores are now standardized in the industry: CS-8, CS-16, CS-20, CS-25, CS-32, CS-40, CS-50, CS-63, CS-80, CS-100, CS-125, CS-160, CS-200, CS-320, CS-300, CS-400, CS-500, CS-800, CS-1000. The standard cut core transformer is generally equipped with a casing cover in which the transformer is fixed with a filling of epoxy resin or some other similar material.

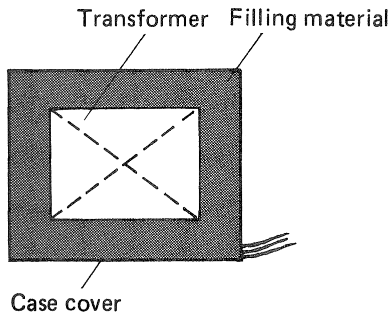


Fig. 6 The position of leakage flux (see Fig. 7)

1-3 Ring Core

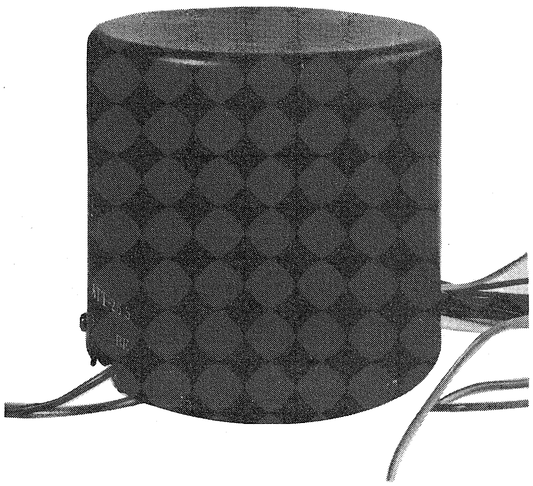


Photo 3

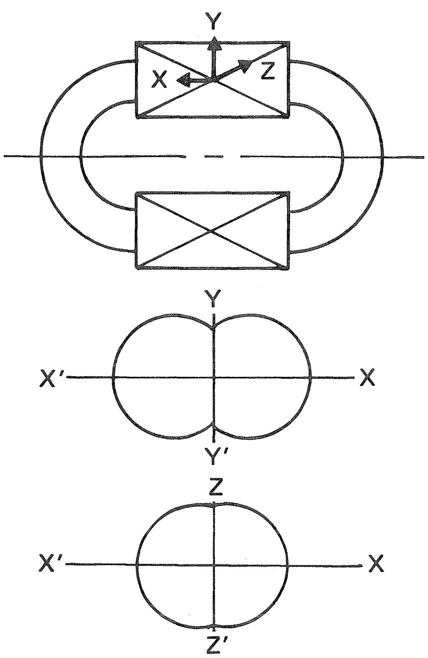
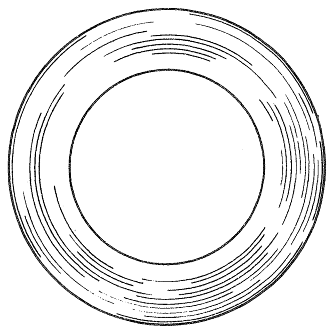


Fig. 7 The position of leakage flux

The ring core transformer, also known as a toroidal transformer, was once employed on a very limited number of electronic devices, but its application is widening with today's greater concern for resource conservation. The disadvantages of ring core transformers are these: there are more steps in the manufacturing process; transient current into the primary winding is greater; when combined with a rectifier circuit pulse noise is often induced, and "beat" often occurs because there is no escape for magnetostriction. The advantages of toroidal design, however, are many: ring core transformers are more efficient, having less leakage flux than either EI or cut core types; they are smaller, and voltage regulation is better. The unique advantages of toroidal transformers make them ideal in power amplifiers over 150W.



Core material: Orient material (directivity)

Fig. 8 Ring Core

Fig. 8 shows that ring cores have no contacting face, and since they don't have standardized names yet, they are specified strictly by measurement. Ring cores, like cut cores, are generally fixed with a filling that is poured into the space between the core and the casing cover, as shown in Fig. 9.

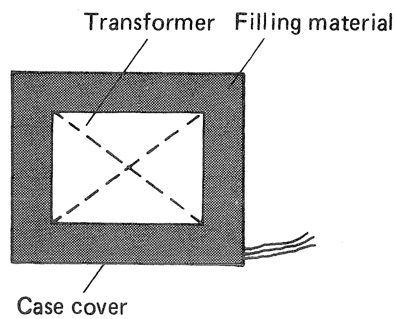


Fig. 9 The position of leakage flux

Fig. 10 shows the position where leakage flux is generated.

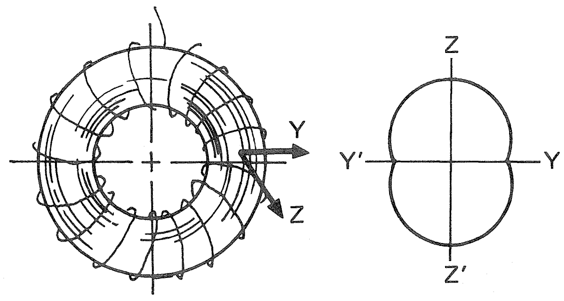


Fig. 10 The position of leakage flux

At this point we would like to go into more detail about how the merits of each of these transformer types is related to use in an actual power amplifier. The weight of the ring core is 10% less than that of the EI, and were it not for the casing cover it would be almost 50% lighter. In addition, it can be made 20–30% smaller in volume (refer to Photo 4 on the next page).

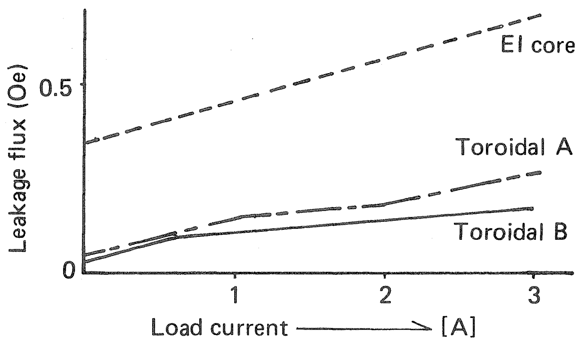


Fig. 11 Leakage flux

Leakage flux is reduced from 1/5 to 1/15 as indicated in Fig. 11, and voltage regulation is also improved. Because the ring has no contacting face, the cross-section of the magnetic circuit becomes smaller. This reduction permits fewer turns in both the primary and secondary windings so the thickness of the wire may be increased. A smaller loss factor in the magnetic circuit results in an efficiency gain of 5–10% and operating temperature remains lower.

1-4 Measures Taken to Reduce Leakage Flux

Leakage flux has an adverse effect on S/N of not only the set it is in, but if bad enough can raise the noise level of associated equipment as well. For this reason, some care has traditionally been taken to keep leakage flux to a minimum:

1. Winding the short ring
2. Winding the core band singly
3. Winding the core band doubly or triply
4. Use of an Orient core band
5. Increasing height and thickness
6. Use of an Orient core

1-5 Color Classification of Lead Wires

Transformer lead wires are color-coded as indicated in Fig. 12.

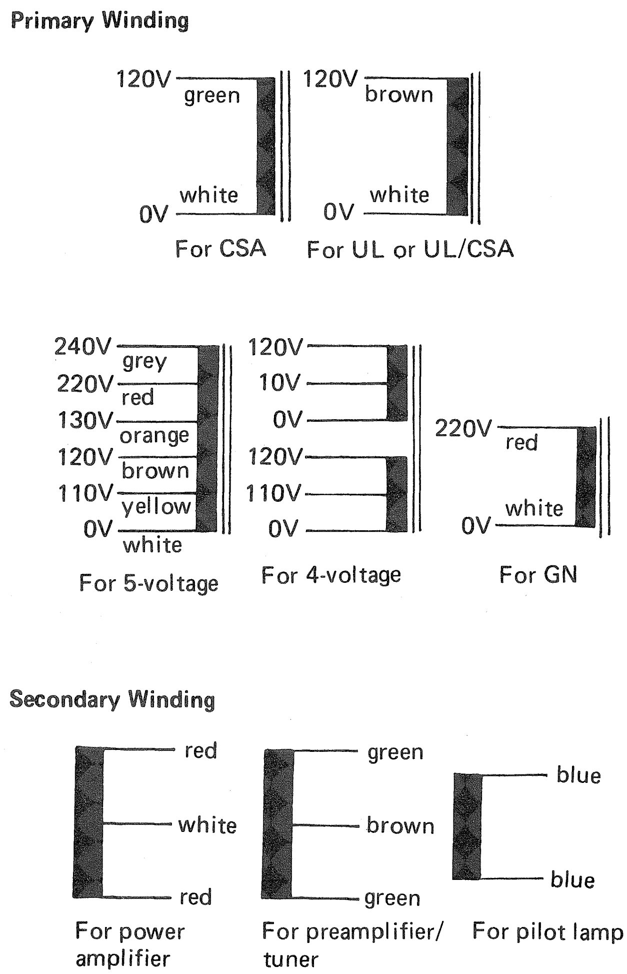


Fig. 12 Color classification of lead wires

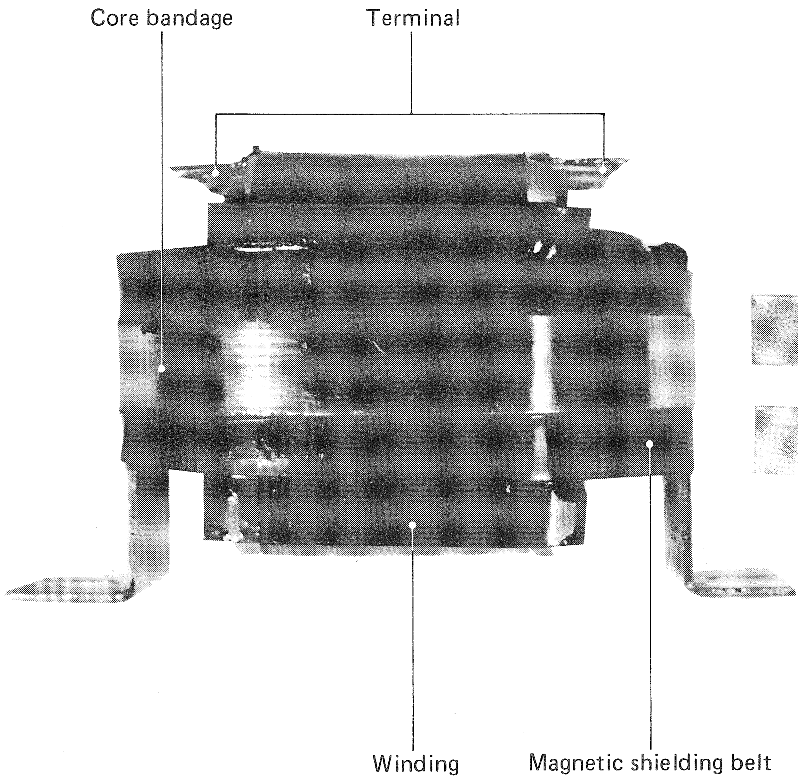


Photo 4 Cut core side view

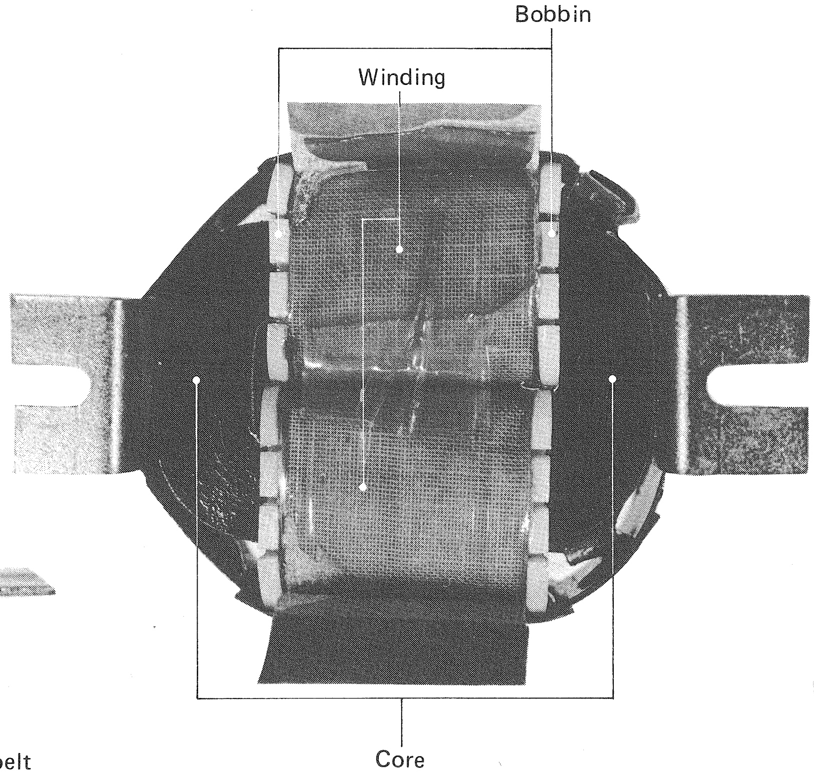


Photo 6 Cut core bottom view

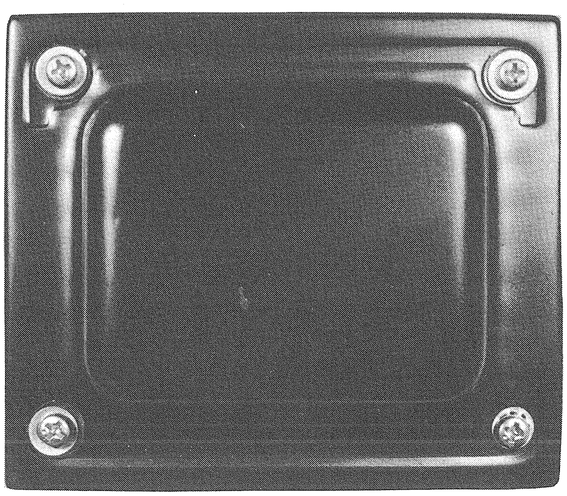


Photo 5 EI top view

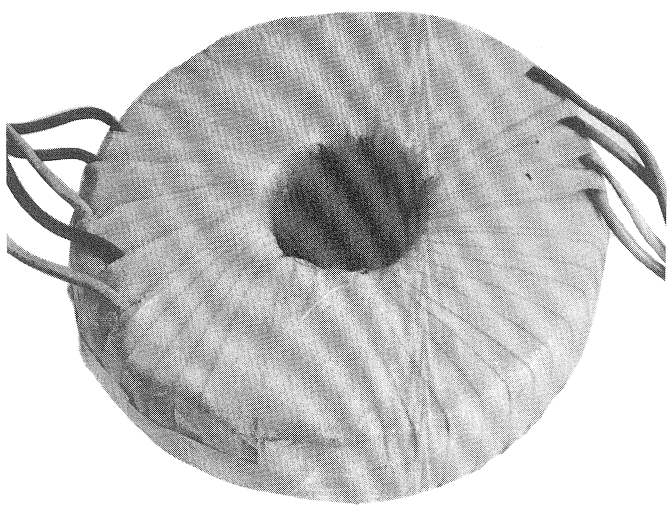
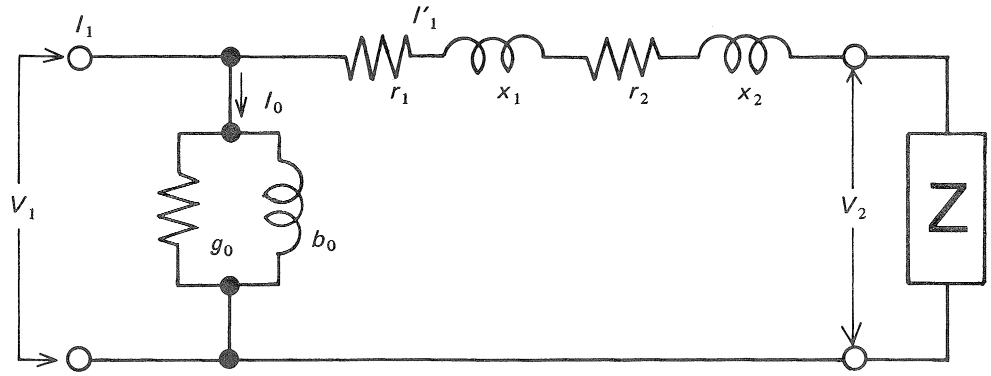


Photo 7 Ring core (uncovered)

1-6 Rush Current

Fig. 13 indicates the primary circuit equivalent of a power transformer. The current of each part is given by the following formulas:



r_1 : Impedance in the primary winding
x_1 : Leakage reactance in the primary winding
r_2 : Impedance in the secondary winding
x_2 : Leakage reactance in the secondary winding
$Y_0 = g_0 - jb_0$: Admittance in the excitation circuit
a : Turn ratio of a transformer

Fig. 13 The primary circuit equivalent of a transformer

The primary loaded circuit:

$$I'_1 = \frac{V_1}{(r_1 + jx_1) + a^2(r_2 + jx) + a^2 z}$$

Exciting current:

$$I_0 = Y_0 V_1 = (g_0 - jb_0)\, V_1$$

Primary current:

$$I_1 = I_0 + I_1 = Y_0 V_1 + \frac{V_1}{(r_1 + jX_1) + a^2(r_2 + jX) + a^2 Z}$$

The instantaneous current (I_1) which flows when voltage V_1 is applied is called rush current. The rush current in a transformer whose r_1 and r_2 are small, like those in a ring core, is proportionally larger. Therefore, some transformers, like those found in SPEC-2, and SX-1250, SX-1050, SX-1980, SX-1280, and SX-1080, are equipped with a special circuit that represses this rush current.

1-7 The Effect of "Beat" Caused by the Input Waveform

If the input waveform were a sine wave there would be no "beat." Bm (maximum magnetic flux density) of the transformer's core, nevertheless, is determined by the number of turns of the primary winding and the current. When the voltage exceeds the nominal norm during use, Bm is designed to restrain the on beat up to a point of 10% above the rated input, whether or not there is a load present at the secondary winding. "Beat" may occur if that limit is exceed-. ed. Also, in the event that the waveform is distorted for some reason, the transformer may beat because the alternation of current in the primary winding and the resulting magnetic flux may exceed the Bm of the core. Fig. 14 shows current waveforms in the primary winding.

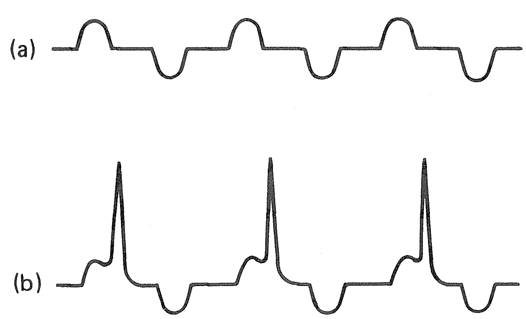


Fig. 14 Current waveforms in the primary winding

(a) is a regular current waveform. (b) is an example of a current waveform which will cause the transformer to beat. Amperage peaks are higher than in the regular waveform, causing the magnetic flux to exceed the Bm of the core, which in turn sets up a magnetostriction which causes the core to beat.

2. Coils and Transformers for Radio Frequency (RF)

2-1 A Variety of Coils and Transformers for RF

(1) Balun transformer

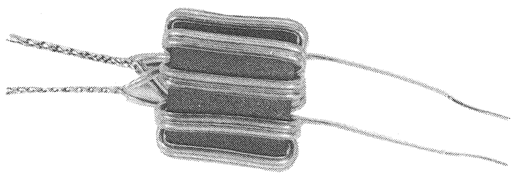


Photo 8

(2) RF tuning coil

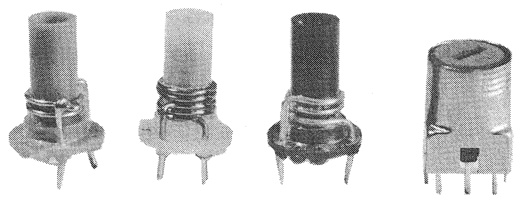


Photo 9

(3) Intermediate Frequency Transformer (I.F.T.)

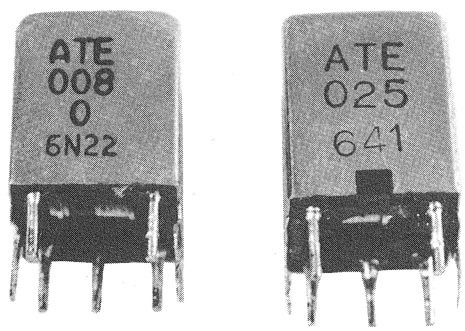


Photo 10

(4) Detection transformer

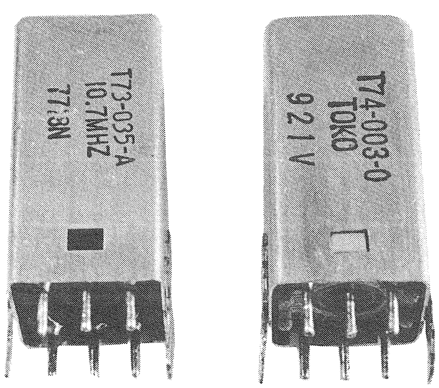
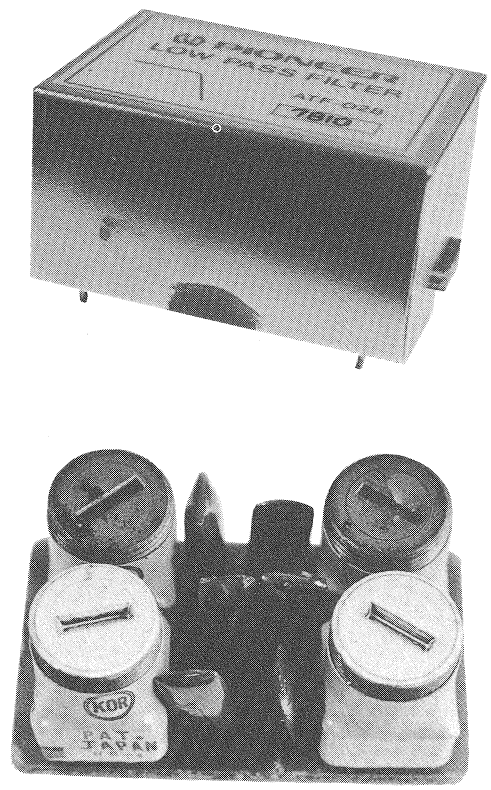
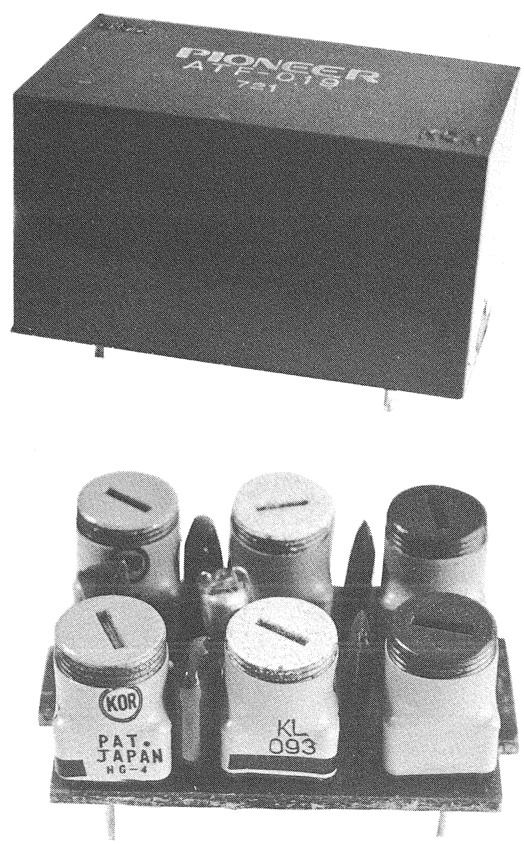


Photo 11

(5) Low-pass filter



2-pole type



3-pole type

Photo 12

(6) Other inductors

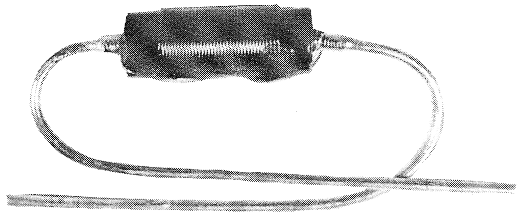


Photo 13 RF choke coil

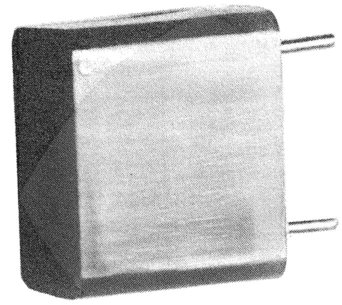
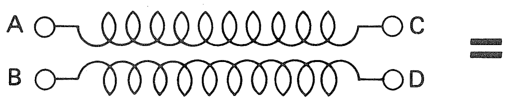


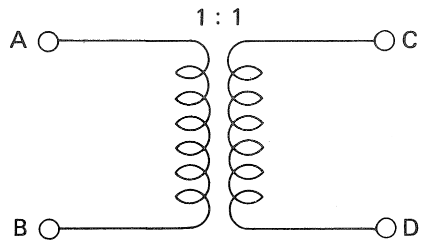
Photo 14 Micro inductors

2-2 Uses and Characteristics

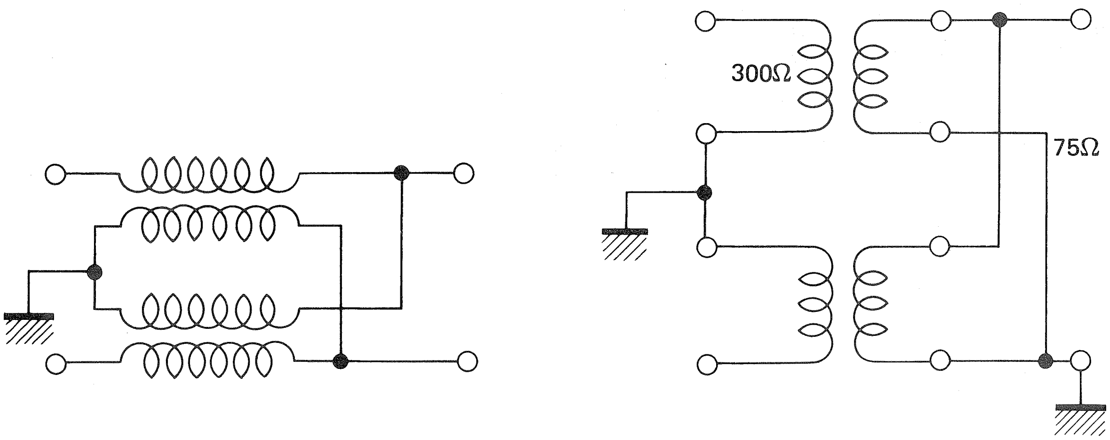
(1) Balun Transformers

A balun transformer converts the impedance of an antenna from a 300Ω balanced circuit to a 75Ω unbalanced circuit.





(a) An equivalent circuit of a bifilar winding



(b) An equivalent circuit of a balun transformer

Fig. 15

(a) Fig. 15 shows an equivalent circuit of a bifilar winding transformer. The impedance between A and B is designed to be 150Ω. Similarly the equivalent circuit of a balun transformer can be expressed as shown in Fig. 15 (B). The input side is a balanced circuit of 300Ω which is a total impedance for the two coils, and the output side is an unbalanced circuit of 75Ω consisting of parallel circuits of the two coils. Though it is possible for the front end of a tuner to be fed directly by a 300Ω line, usually a balun transformer is used to convert the impedance to 75Ω. Frequency range is from 70–120MHz, and the insertion loss is about 1dB.

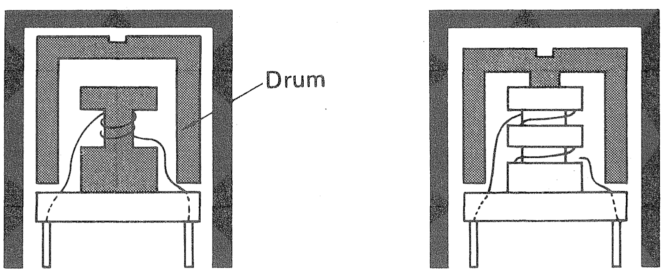
(2) Tuning Coils for High Frequency

Tuning coils for high frequency are used along with variable capacitors to tune in stations. Though there are a wide variety of shapes, colors, and winding configurations, these coils may be classified in two broad categories: those for high-frequency tuning circuits and those for local oscillator circuits (OSC coils). Both share a common characteristic. Ferrite cores are used in the RF coils, while aluminum cores are used in the OSC type. The ferrite core raises the Q-value of the tuned circuit, raising inductance and lowering the tuning frequency as the ferrite comes closer to the center of the coil. The aluminum coil

has a lower Q-value but its temperature characteristics make it ideal for use in oscillator circuits. As the aluminum center is drawn into the coil, Q-value lowers and tuning frequency becomes higher.
Some coils are equipped with an extra set of taps, called secondary windings, which are used for impedance-matching in the RF amplification or oscillator sections. The purpose of taps and secondary windings are the same, though secondary windings are a more convenient and less costly method.

(3) Intermediate Frequency (IF) Transformers

Intermediate frequency (IF) transformers were formerly used (in the era of discrete circuits up until about 1973) for increasing selectivity. But now that IC's dominate the scene along with ceramic filters, this type of transformer has been largely relegated to matching. The first stage IF transformer is placed within the front end of the tuner and plays an important role of impedance matching with the ceramic filter of the following stage and reduces stereo distortion as well. The usual shape is 10mm square.



Manufacturer A Manufacturer B

Fig. 16 The structure of IFT

Manufacturer A produces a transformer whose core has a secondary winding wound on top of the primary. The disadvantages of this design are that the distributed capacitance in both windings becomes large as does the variation in the coupling factor. Manufacturer B, on the other hand, produces an IF transformer with a slit bobbin on which the primary and secondary windings are separate, reducing distributed capacitance and allowing a reduced coupling factor.
Tuning capacitance can be from 22PF to 100PF to suit the purpose. As capacitance increases, Q_0 decreases because it is less affected by capacitance variations in external semiconductors. Therefore balance of these two factors is incorporated into the design of the transformer.

(4) Detector Transformers

A detector transformer is an ordinary double-tuning coil wound on a slit bobbin. As mentioned earlier, the advantage of this design is a reduction in the coupling factor between the two windings. It is suitable for detectors because of its strict differential-gain characteristic.

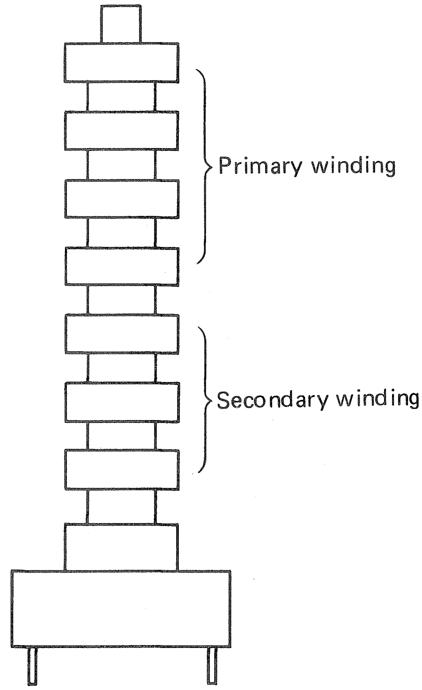
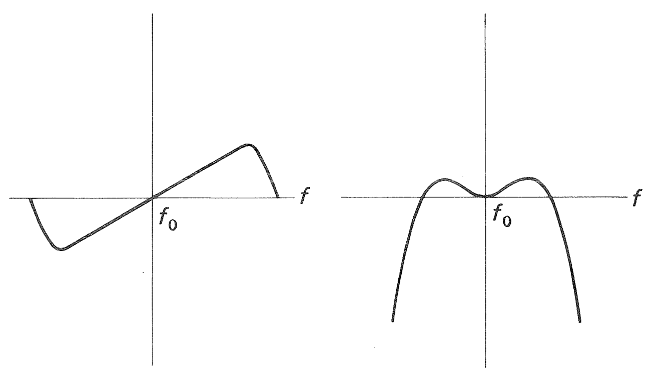


Fig. 17 Detection transformer

The detection-differential characteristic is similar to the group-delay characteristic in that non-linearity of the S-shaped curve of the detector can be magnified. This feature allows a relatively large change in detection output for a very small change in frequency. Because this relationship can be differentiating the S-curve against frequency, it is often called the differential gain characteristic. Like group delay, the detection-differential relationship should be constant throughout the band.



(a) S-curve characteristic (b) Dectection-differential characteristic

Fig. 18

(5) Low-Pass Filters

Low-pass filters are generally of two-pole or three-pole design. The purpose of the low-pass filter is to maintain flat frequency response up to 15kHz with sharp attenuation at 19kHz and 38kHz. As you probably know already, the FM signal is preemphasized at the station and then de-emphasized in the tuner. Correct de-emphasis is largely dependent on the flatness of the low-pass filter up to 15kHz. For this reason, the F-3's low-pass filter is designed to remain practically ruler-flat up to 15kHz with sharp dips at 23kHz and 38kHz. The 19kHz pilot signal is attenuated by another altogether, yielding greatly improved de-emphasis characteristics. The greater the number of poles, the more dips in frequency response there will be and the greater the amount of attenuation at each.

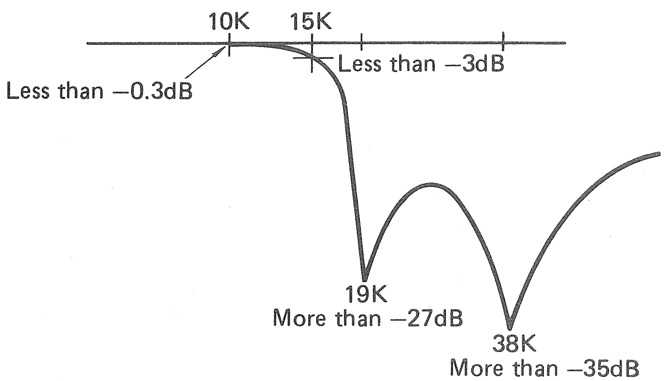


Fig. 19 Two-pole low-pass filter

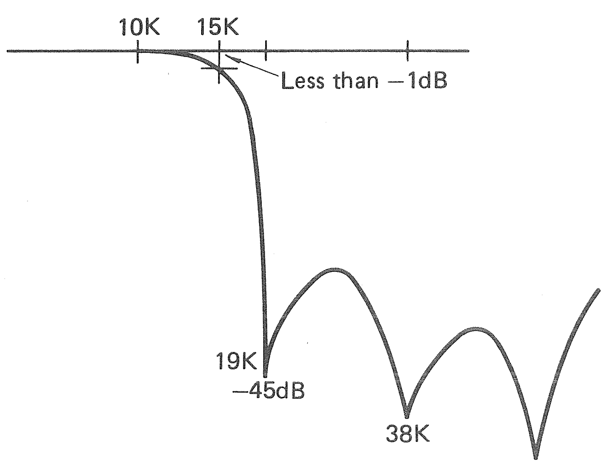


Fig. 20 Three-pole low-pass filter

3. Coils Used for Audio-Frequency Circuits

Audio-frequency circuit coils include inductors used in the output stages of power amplifiers (Photo 15) and coils used in speaker networks (Photo 17).

3-1 The Output Inductor

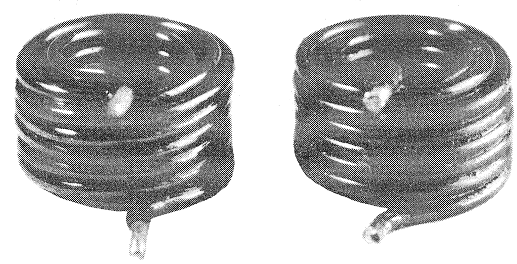


Photo 15

The load of a power amplifier is normally a speaker system, but it must be able to run without instability or oscillation no matter what the load. As Fig. 21 shows, the impedance characteristics of speaker systems are composed of inductance, capacitance, and resistance. Two-way or three-way speaker systems are composed of inductance, capacitance, with interaction by the impedance of the various drivers and the crossover network. As you may know, amplifiers employing negative feedback are likely to react adversely by inductance and capacitive elements normally found in such a speaker load. By placing an inductor in the output circuit, oscillation caused by capacitive loads can be avoided. For this reason, inductors are now found in the vast majority of amplifiers on the market today. The following are common inductor types:

T63-009 for general output (2.2μH)
ATH-003 for low output (1.2μH)
ATH-012 for high output (1.1μH)

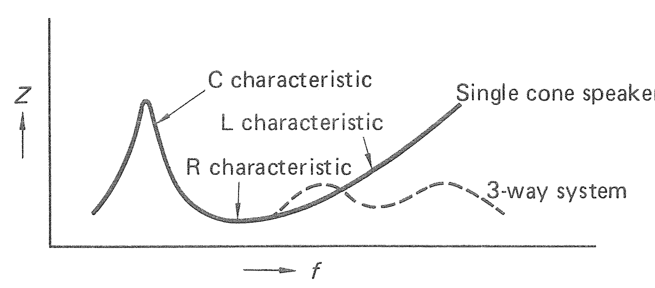


Fig. 21 Impedance characteristics of speaker systems

3-2 Speaker Network Coils

They are generally of two types: those with air cores and those with ferrite as a core. Choke coils with cores of silic steel sheet or permalloy are also used occasionally.

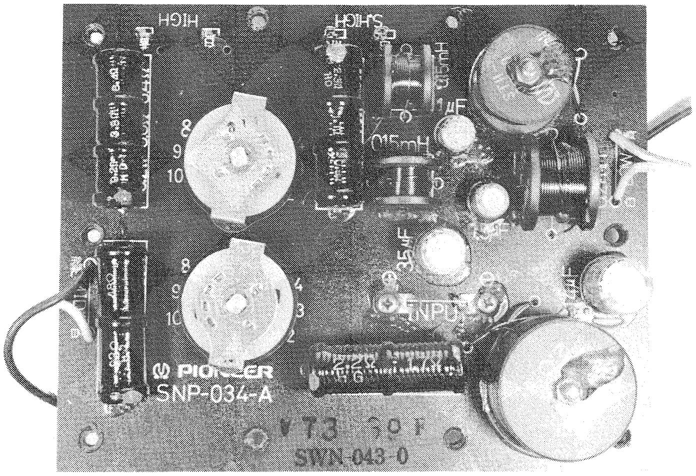


Photo 16

(1) Air Core Coils

Air core coils have no magnetic saturation against large input, harmonic distortion is infrequent, and inductance generally remains constant against input. On the minus side, the copper wire has to be thicker to reduce the amount of DC resistance, so the cost is high and the dimensions are apt to be large. Furthermore, if the coils are placed close together, mutual induction can occur, resulting in poor tone quality. The remedy for this is special attenuation at the time of installation.

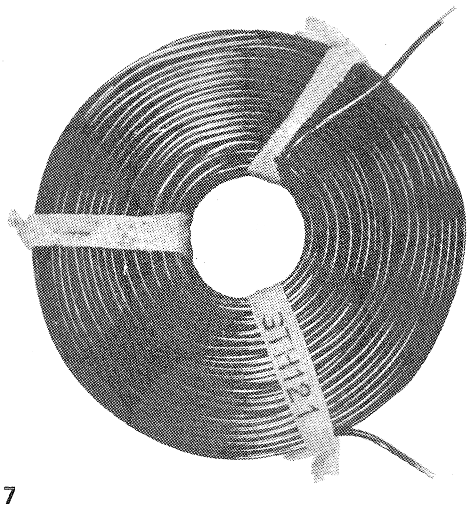


Photo 17

(2) Choke coils with ferrite core

The coils are of two types: the drum type shown in Photo 17 and the E type shown in Photo 18. Ferrite-cored choke coils offer low DC resistance and so can be made small. Installation is simple and they are easy to mass-produce. For these reasons, they are widely used. When used with low input, they exhibit excellent linearity and tone quality is normally good. However, as input power is increased the material from which the core is made becomes increasingly important in the maintainance of low distortion.

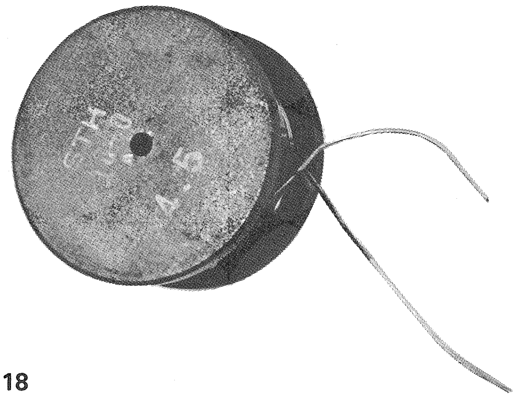


Photo 18

(3) Choke Coils with Cores of Silic Sheet

Choke coils with cores of silic sheet steel are rarely used any more. There are four types using this center, as shown in Fig. 22. The silic sheet steel of the core may have magnetic directivity or it may have none.

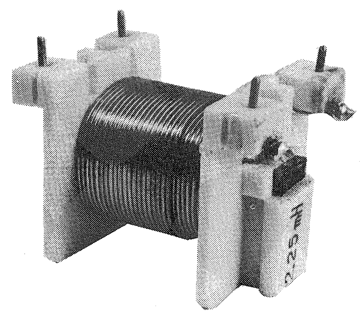


Photo 19

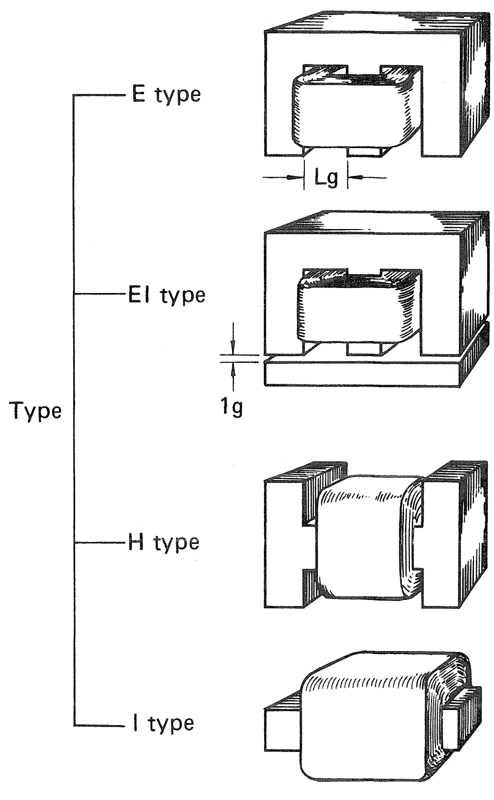


Fig. 22 Types of choke coils with cores of silic sheet steel

These chokes have the following characteristics as opposed to air core coils: they can be made compact, they have low eddy current distortion because the core is laminated, and distortion is low for even large inputs because the level of magnetic flux is high. However, unless a resin-solidifying treatment is used, resonance is likely to occur between the core and the coil. Another problem is that under some gapping conditions a closed magnetic circuit can occur causing saturation and a resulting lack of linearity against input.

(4) A Comparison of the Characteristics of Air Core Coils and Ferrite Core Coils

The Figs. 30 ~ 32 show frequency and distortion characteristics between an air core coil and two types of ferrite core coils. All three coils are 0.45mH. Fig. 23 shows the data on the air core coil, Fig. 24 shows the performance of a core of superior ferrite material, while Fig. 25 shows the performance of a ferrite core coil with inferior material. You can see for yourself that the materials and the construction of a coil have a great effect on its performance.

4. Coils Used in Tape Decks

Coils used in tape decks include peaking coils in recording equalizer circuits, trap coils to couple the recording amplifier to the bias circuit, oscillator transformers to generate AC current for the erase head and record bias, dummy coils to simulate the load of an erase head on the oscillator circuit of open-reel tape decks, and MPX filters used in the Dolby circuits of many cassette decks.

(1) Oscillator Transformers

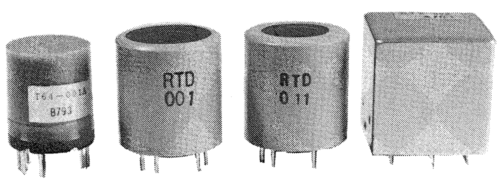


Photo 20

These are used in oscillator circuits to generate the necessary alternating current for the erase head and the bias circuit. Oscillating frequency is between 100 and 125kHz for open-reel decks and 85kHz for cassette decks. Secondary winding impedance is high to assure constant bias current and stable erasing properties. Because distortion in the oscillator circuit adversely affects the operation of the entire recording circuit, the core material is carefully selected and Q is kept high in order to keep the distortion figure as low as 0.5%. In some sets, shields called oscillator blocks are used to prevent the coils from interfering with reception by AM/FM tuners.

(2) Trap Coils

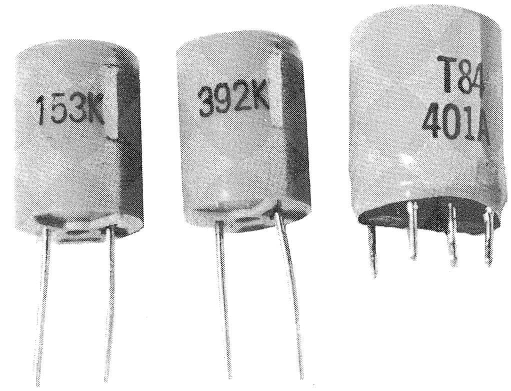


Photo 21

The high harmonic element of the oscillator transformer may set up a separate and unwanted oscillation in the recording amplifier or degrade its S/N. The flow between the bias circuit and the recording circuit's outputs and vice versa must be prevented to achieve a clean signal. Trap coils serve this purpose. The coils are tuned to resonate at the same frequency as the oscillator circuit, minimizing mutual interference.

(3) Peaking Coils

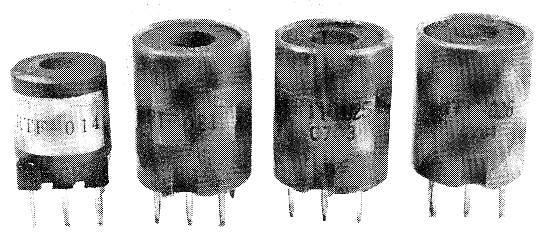


Photo 22

These are used in the LC networks of recording equalizers to make dramatic changes in the incoming signal, to conform to such standards as NAB, BTS, IEC, and others. An RC network is used in the playback circuit to de-process these signals back to flat response. Peaking coils are used because the changes in the input signal are so great that an ordinary RC network cannot be used.

(4) Dummy Coils

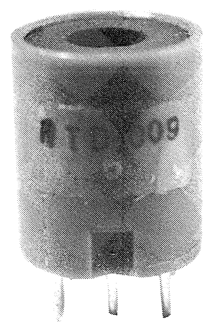


Photo 23

Tape decks now come in a variety of track configurations and often offer synchronized monitoring. In these cases, the erase head and the record head have to be disconnected from the oscillator circuit. The frequency of the oscillator, however, is in large part regulated by the inductance of the erase head and the oscillator coil and by its capacitance when connected in parallel with this inductance. Remove the erase head from the circuit, and the voltage and the frequency will both vary. These variations have an adverse effect on overall performance. To correct this, a dummy coil is inserted into the circuit whenever the erase head is disconnected to maintain proper oscillator load and frequency. In practice, the dummy coil and the oscillator impedance should be adjusted so that the frequency is precisely maintained when switching from the erase head to the dummy coil.

(5) MPX Filters

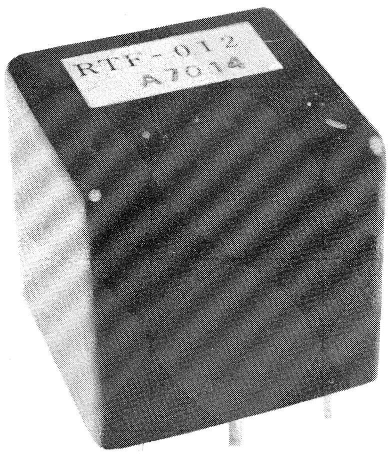


Photo 24

These are used in the record phase of Dolby processing. If a poor-quality stereo tuner is used to feed a signal to a cassette deck its stereo pilot signals of 19kHz and 38kHz will not be properly suppressed and will enter the Dolby circuit along with the audio signal. These two signals, the pilot and the carrier, can have a bad effect on Dolby decoding if they are not attenuated. The MPX filter serves this purpose.

(6) Headphone Transformers

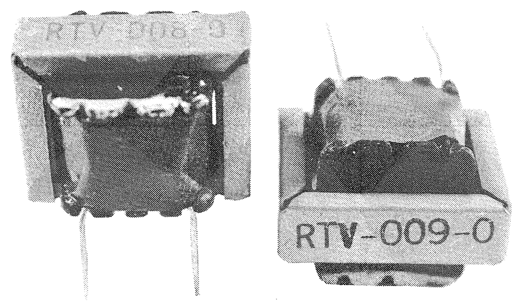


Photo 25

The headphone transformers are used mainly for the tape decks to match the headphone impedance to the output impedance of the headphone amplifier.

Headphone transformers are being superseded by IC's due to their lower cost, higher performance and convenience of use.

Fig. 30

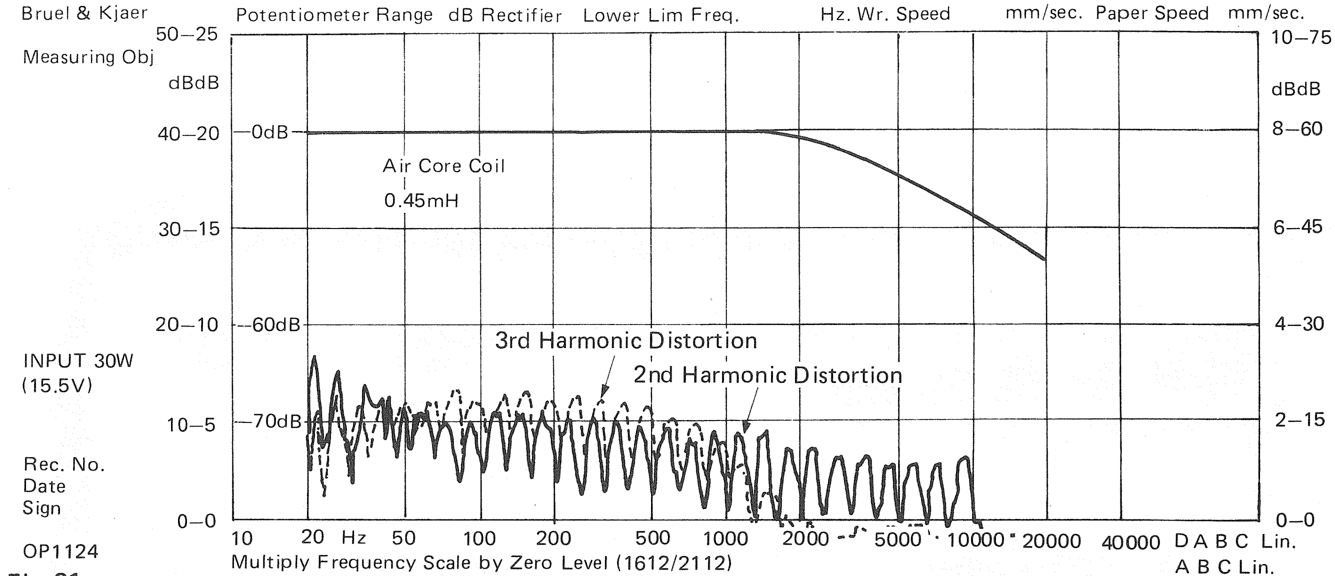


Fig. 31

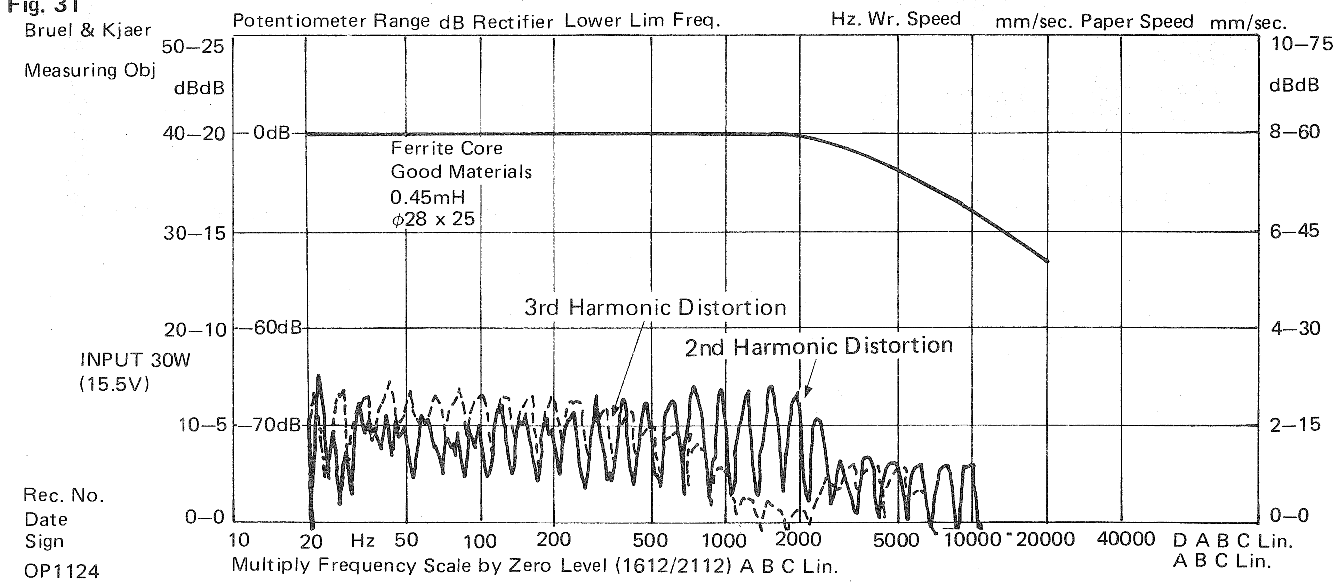
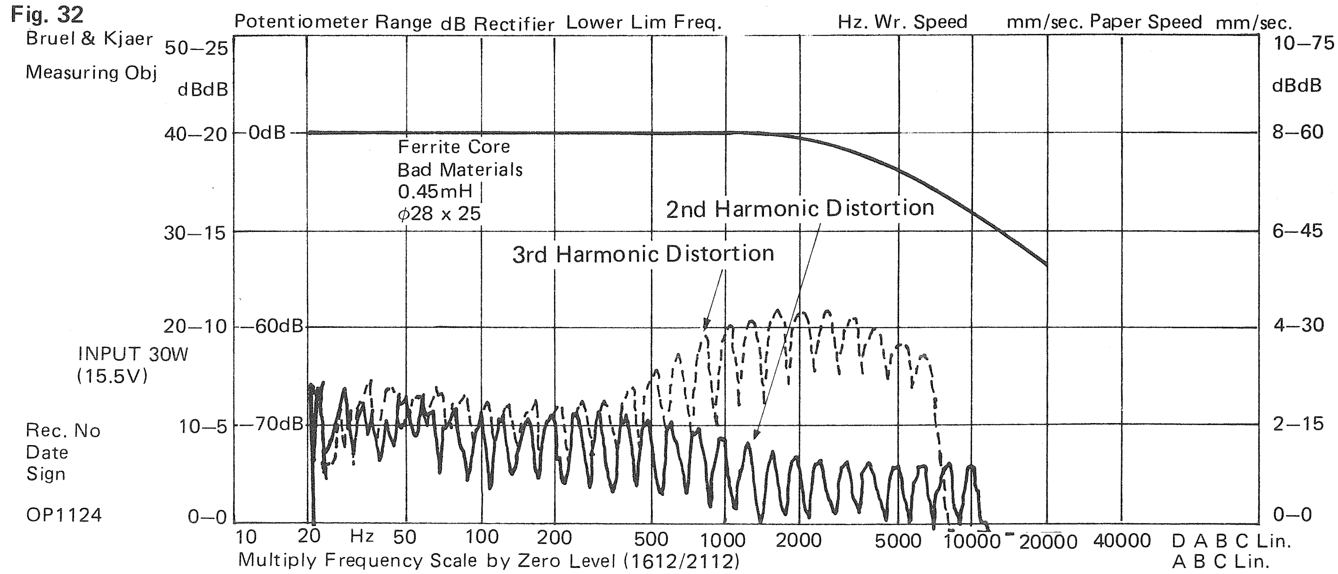
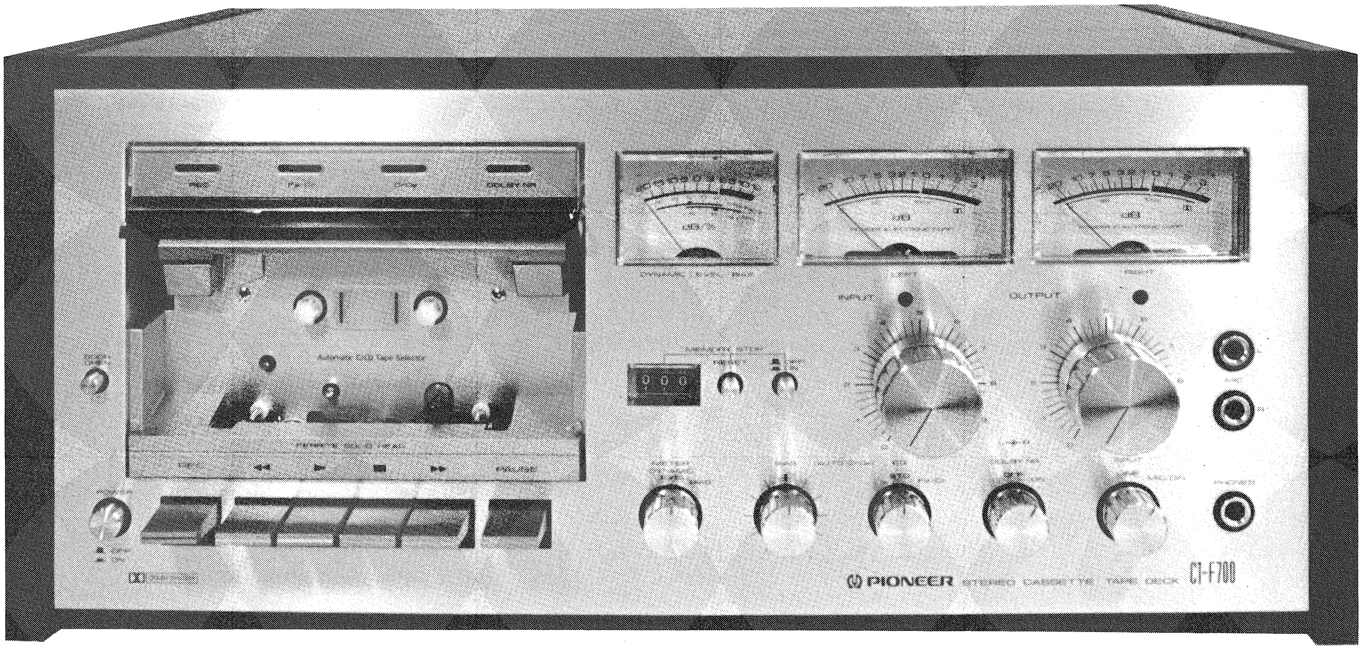


Fig. 32



New Products
CT-F700



More than 10 years have passed since the first introduction of cassette tape decks. Performance has greatly increased during that period and is now advancing to a level of sophistication which once belonged to middle-class reel-to-reel tape decks.
A fine example of this sophistication is the CT-F1000 three-head cassette tape deck with an efficient monitoring system during recording which was introduced by Pioneer last year.
CT-F1000 is now acknowledged as the top-ranked three-head casette deck. A new deck, CT-F700, is a two-head cassette deck which will also set the standard of performance in its class.
CT-F700 incorporates a number of features which are not found on other competitive models. One of those features is the system which monitors the dynamic level during recording through the use of a third meter as a recording bias adjustment system.

1. The dynamic range monitoring system.
As you may already know, setting the recording level when you record with a tape deck can be difficult at times, and even harder when recording through microphones.
Compared to reel-to-reel tape decks, the task is almost twice as difficult on a cassette tape deck.
These are the characteristics of cassette decks which contribute to the problem:

1) The width of the tape is narrow.
2) The magnetic coating of the tape is thin.
3) The speed of the tape is slow.

These facts result in the following:

1) The level of magnetic saturation is lower than in reel-to-reel decks.
2) The level of hiss is higher.
3) A greater number of high frequencies are dropped out.

In short, the dynamic range is narrower.
Fig. 1 shows the characteristics of cassette tape decks vs. reel-to-reel decks.

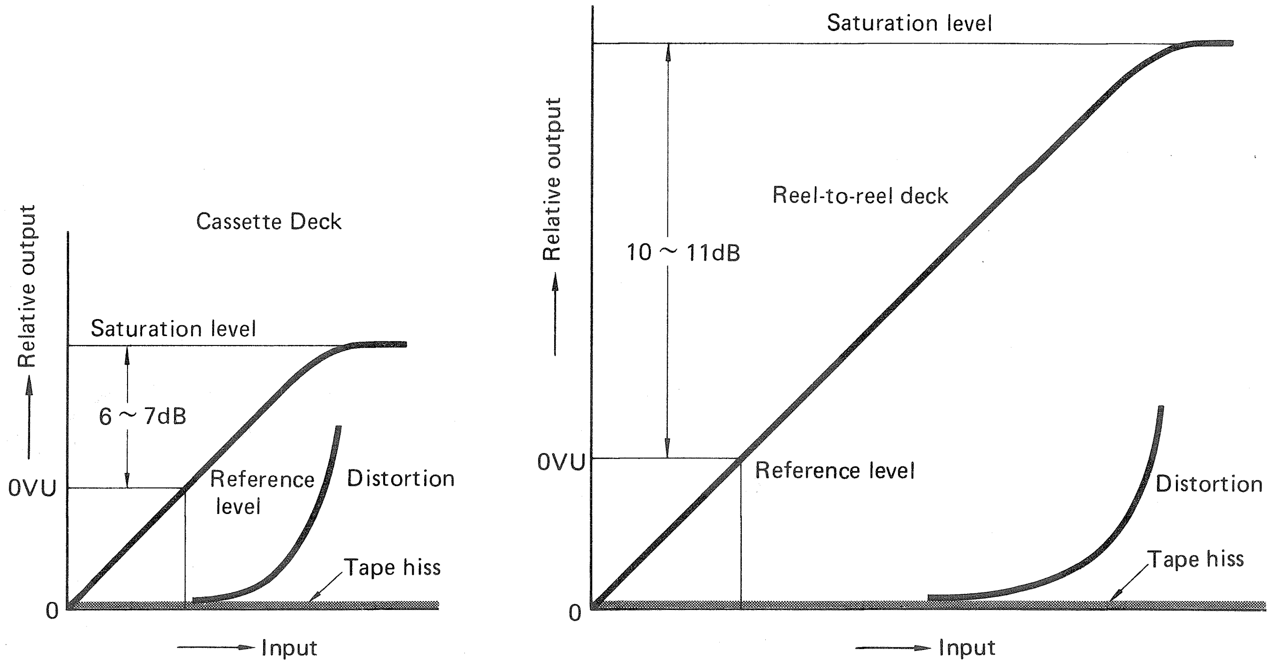


Fig. 1 Characteristics of input and output

Next let's consider the recording signal.
Fig. 2 shows a recording signal's wave form.

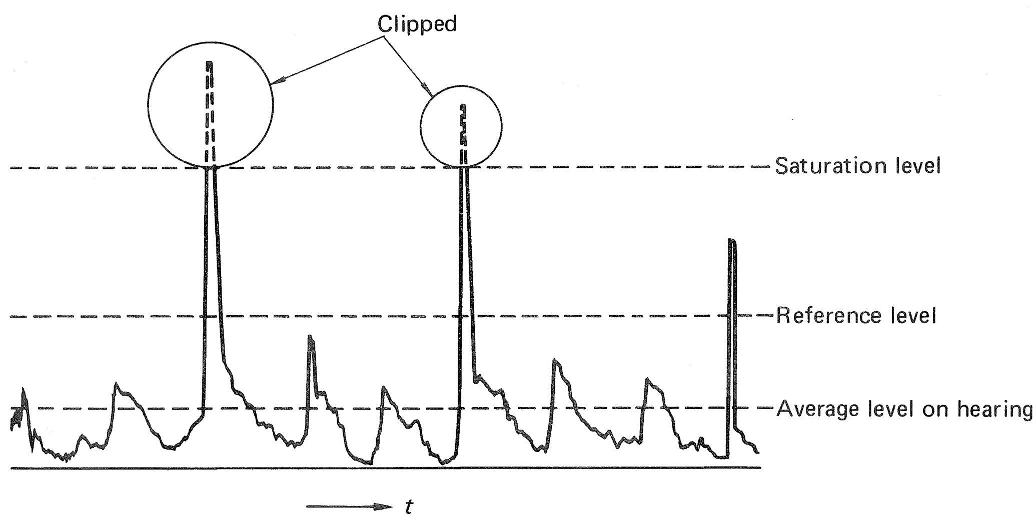


Fig. 2 The recording signal

Generally speaking, there is considerable difference between the peak and the average levels of the signal. When recording with the average level set too high (REC level setting too high), the signal will be clipped at its peak due to the magnetic saturation of the tape.
On the other hand, if the REC level setting is too low, S/N will be aggravated by tape hiss noise mentioned above although no saturation will occur at the peaks. In order to record signals that have a difference between peak and average levels without distortion and deterioration of S/N, it is necessary to know how high the peak level is.
The system to detect this peak level is found in the third meter.

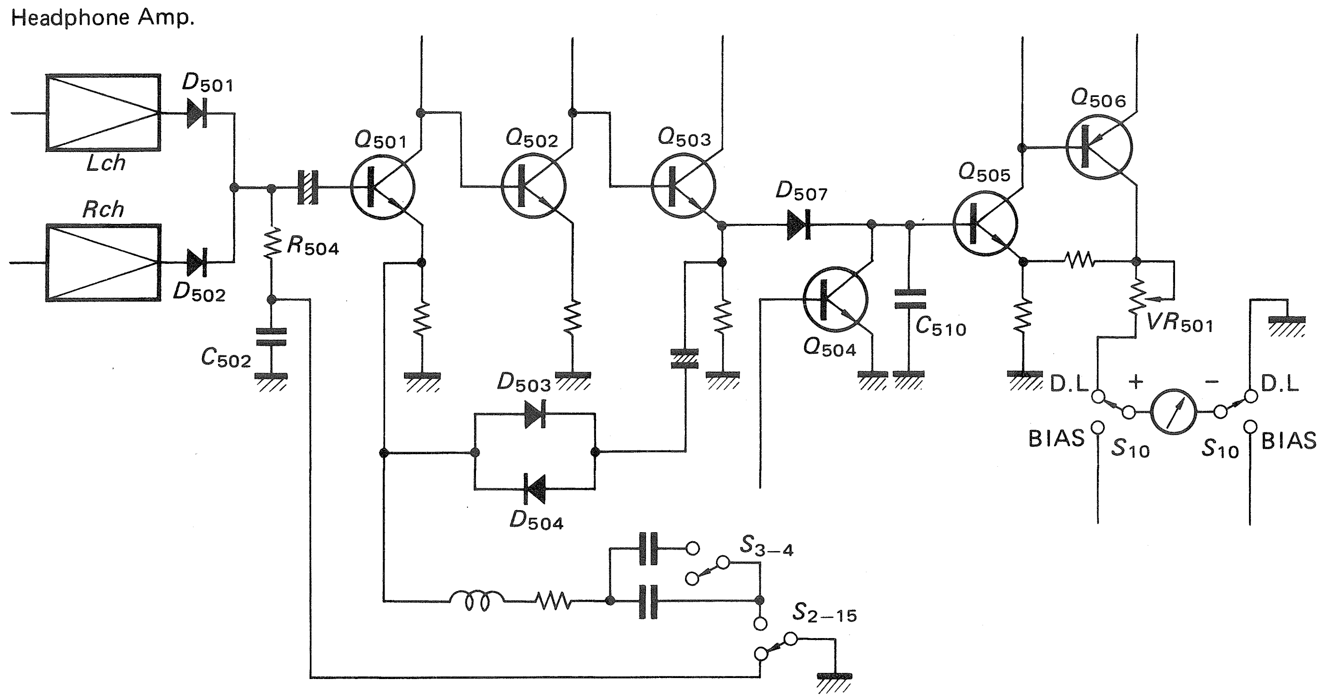


Fig. 3 Peak meter circuit

Fig. 3 shows the third meter drive circuit (peak-meter circuit). The Q_{501}, Q_{502} and Q_{503} form the three-stage direct-coupled amplifier and carry out logarithmically-compressed amplification because the D_{503} and D_{504} are inserted into the feedback loop from the emitter of Q_{503}.

The logarithmically-compressed amplified signal is rectified by the D_{507}, the peak is detected by the C_{510} and DC amplification is accomplished by the Q_{505} and Q_{506}.

Q_{504} is a muting transistor which prevents deflection of the meter needle when power is on.

As you can see in Fig. 4, this meter circuit has flat frequency response during playback, while it increases response at both high and low ends during recording.

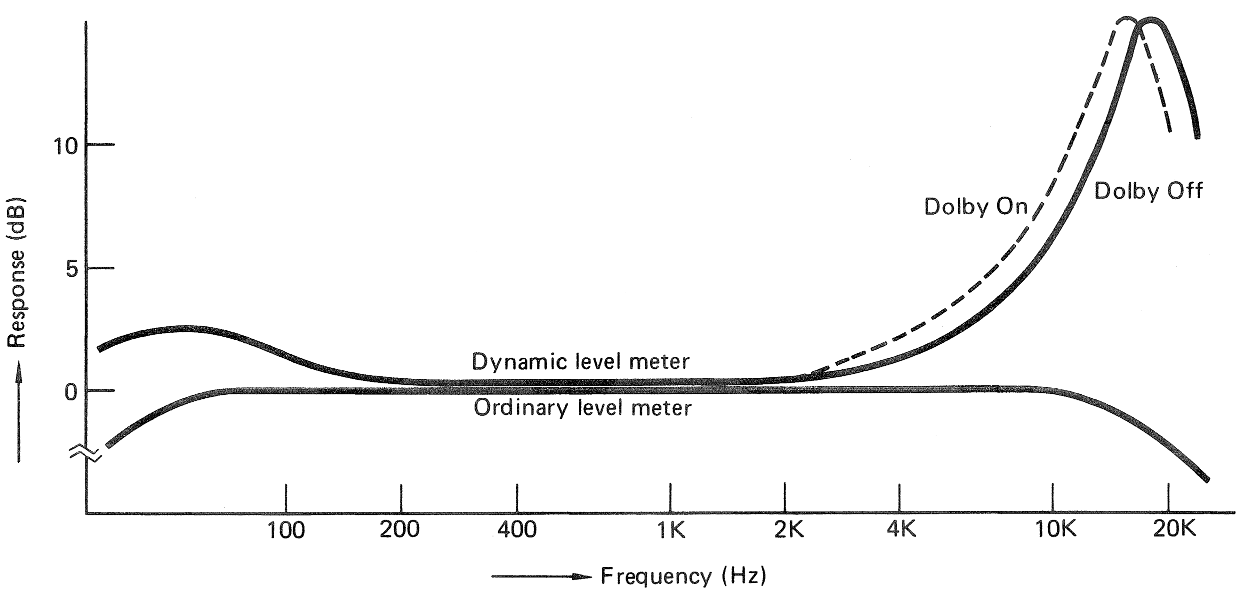


Fig. 4 Frequency characteristics of meter circuit

In general, while recording with a cassette tape deck, the following points can be made concerning frequency range of input signal and level setting:

1) At low frequencies, if the recording level is raised, the distortion increases (see Fig. 5).

2) At high frequencies, the saturation level of the magnetic tape decreases along with the increase of frequencies.

Accordingly, as the frequency of the recording signal rises, the signal saturates in the lower level and characteristics of input and output are aggravated (see Fig. 5).

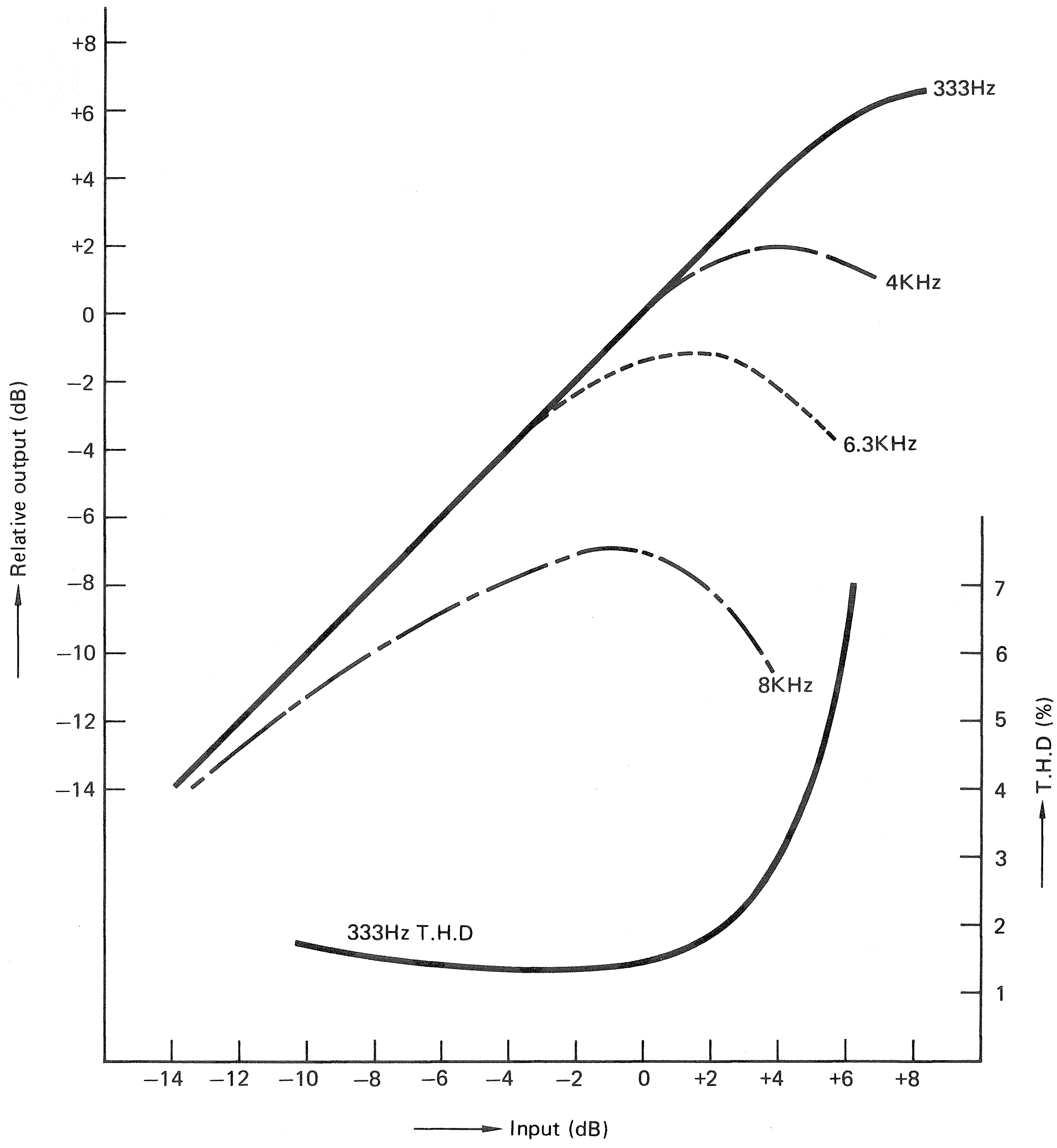


Fig. 5 Input and output, distortion characteristics of cassette tapes

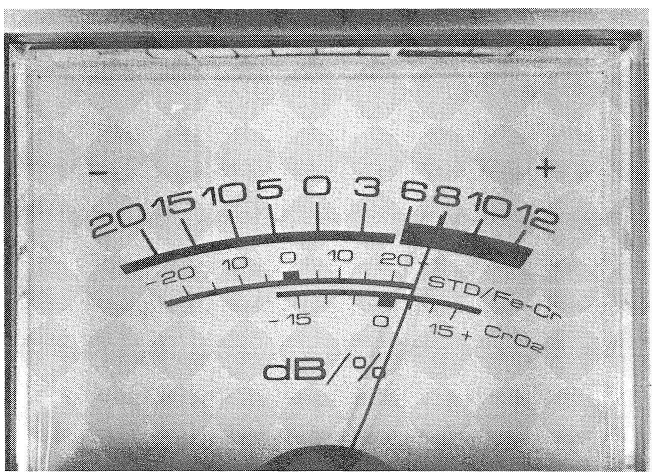
On the other hand, as level meters usually have flat frequency response characteristics, they cannot read frequency (components) of input source. Consequently, even though you set the recording level as you think fit by monitoring the level meters, distortion will occur as the output level decreases at high frequencies due to the magnetic saturation if the high-frequency spectrum of the source is strong.

Also, if the low-frequency spectrum of the source is strong, the distortion will sharply increase at low frequencies due to the low-frequency compensation of the recording equalizer.

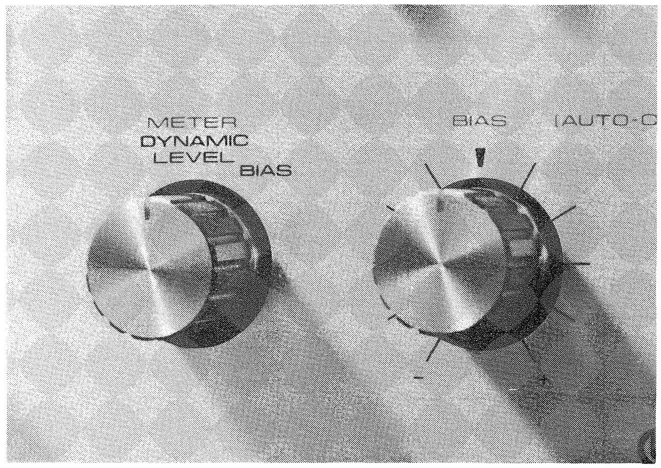
This is why the frequency characteristics of the third meter during recording appear as shown in Fig. 4.

Thus, in order to increase response a little in the lows and a lot in the highs, the level meter over-responds slightly in the lows compared with the middle range and by a considerably larger deflection in the highs. In the case of recording with Dolby On, this dynamic-level meter works even more effectively. Recording characteristics with Dolby On boost the high-frequency range. Therefore, when you set the recording level higher, the margin of high-frequency range signals to saturation level becomes small. In order to prevent this situation, the meter response in high frequencies is increased by approx. 2dB as compared to when Dolby is Off. It then becomes easy to set the recording level without distortion even if the high-frequency level is a little bit over.
The input signals of the dynamic-level meter are the indication of larger signals on the right or left channels.
Furthermore, the third meter works during playback as a peak level meter whose frequency characteristics are flat. The peak-level meter detects peaks of larger signals of both channels, shows the peak indication in dBs and can instantly respond to the pulsive sound. This is very effective for setting the output or for monitoring during dubbing from another tape deck.

How to set the recording level.
Turn the meter switch to "DYNAMIC LEVEL."
Set the input-level control at a level so that the deflection of the dynamic-level meter does not exceed +6dB, even though the VU deflection is within +3dB against the peak signal of the program source.
In some program sources (especially in the case of direct recording using microphones), levels of signals vary by a great extent, so be sure to constantly monitor the dynamic-level meter when recording.



2. **Recording bias adjustment system.**



CT-F700's other big feature is its recording bias adjusting system. The value of the bias added to the head when recording is chosen to maximize output and minimize distortion during playback, while taking into consideration "dropout" during the process.
Usually the point where the output is 0.5dB under the most sensitive point after increasing the bias from the most sensitive bias point is considered the best recording bias (see Fig. 6).
Fig. 6 shows the relation of bias, output and distortion of 333Hz signal, but what about the high frequency?

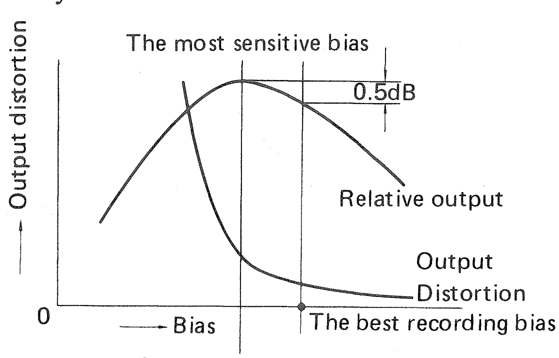


Fig. 6 Relation of the volume of bias, output and distortion

As you can see in Fig. 7, when the frequency of recording signal becomes higher, the relative output decreases to the signal of reference frequency (333Hz).

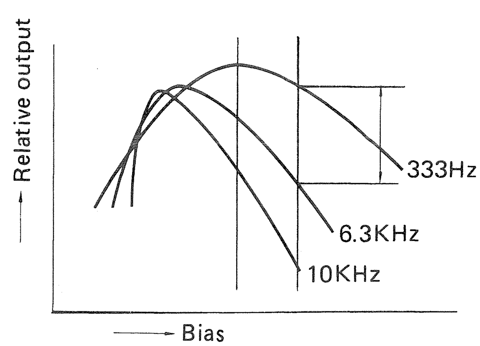


Fig. 7 Relationship between the volume of bias and output

The decreased volume of this relative output depends on the volume of bias.
How this fits into the relationship between the frequency of the recording signal and the relative playback output is shown in Fig. 8.

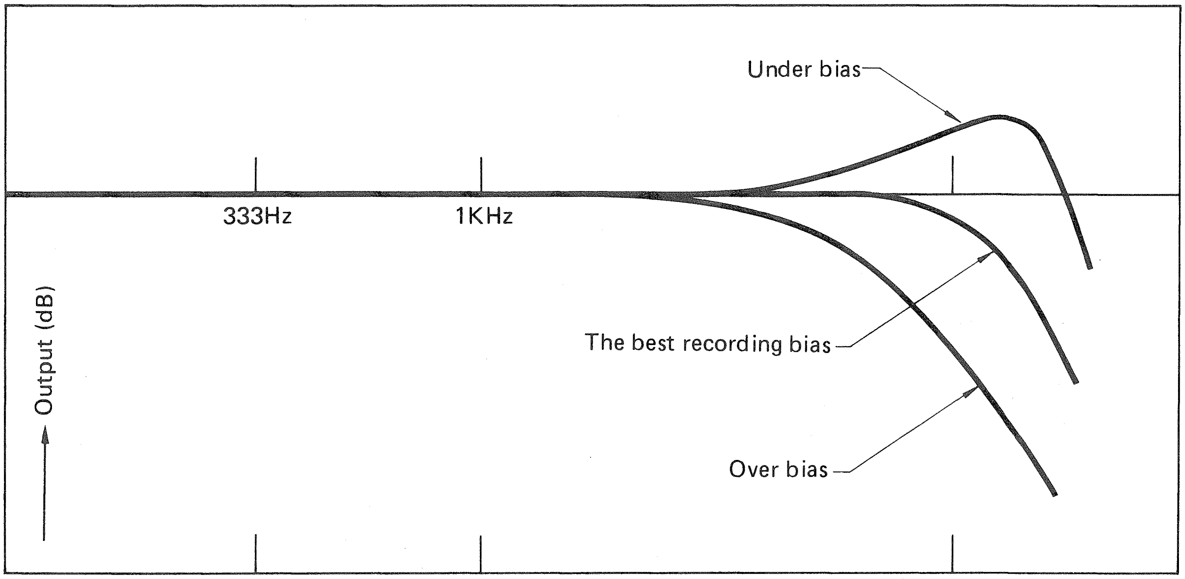


Fig. 8 The change of bias volume and the frequency characteristics
(Output shows overall characteristics)

Thus when bias is increased, the higher the frequency and the lower the level. On the other hand, when the bias is decreased, the level increases.
Moreover, the frequency characteristics of this bias volume depend on the kinds of tapes being used.

On CT-F700, the best bias volume can be set to accomodate any tape you use by continuously changing bias volume and reading the scale of the third meter simultaneously.

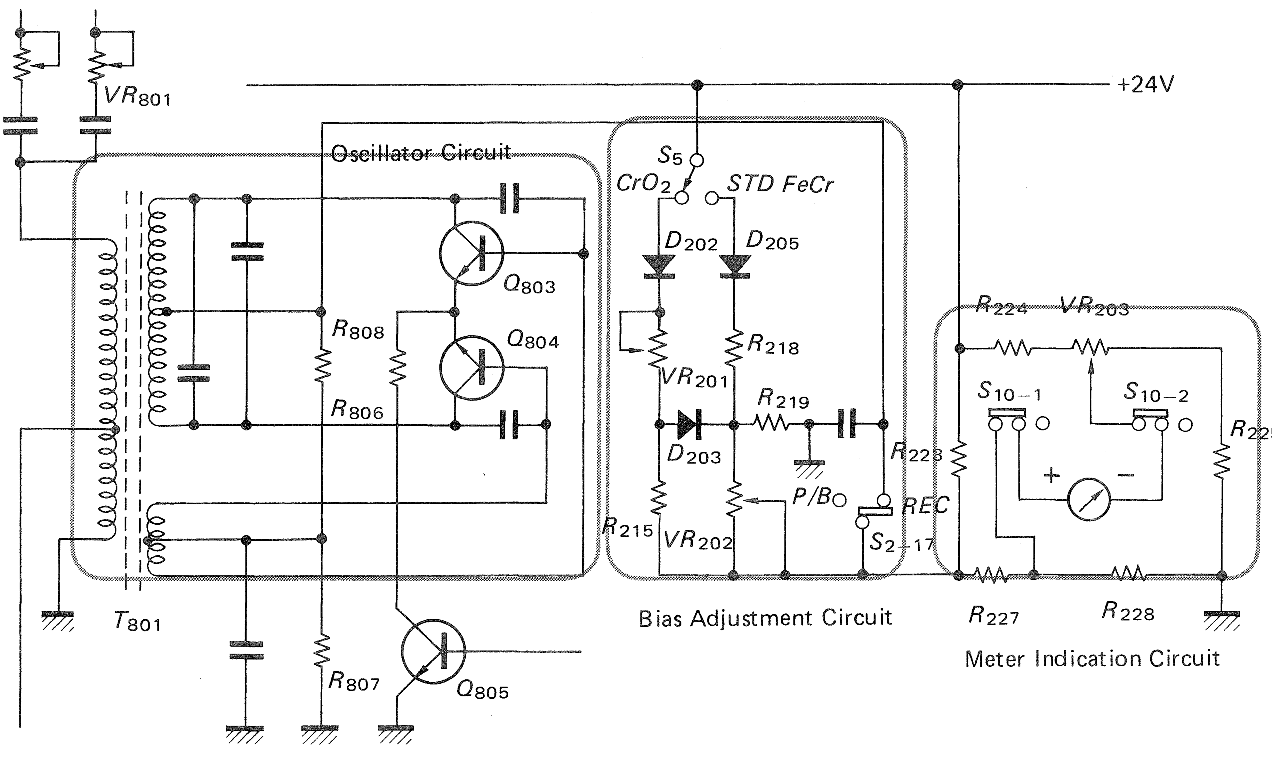


Fig. 9 Bias Adjustment Circuit

The bias adjustment is conducted at VR_{202} by changing the bias and collector voltage in the bias oscillation circuit.
The following table shows the voltage between Point B and the earth (grounding) at three points – minimum, center, maximum points – at VR_{202}.

	Min.	Center	Max.
STD., FeCr	30V	42V	56V
CrO_2	40V	49V	56V

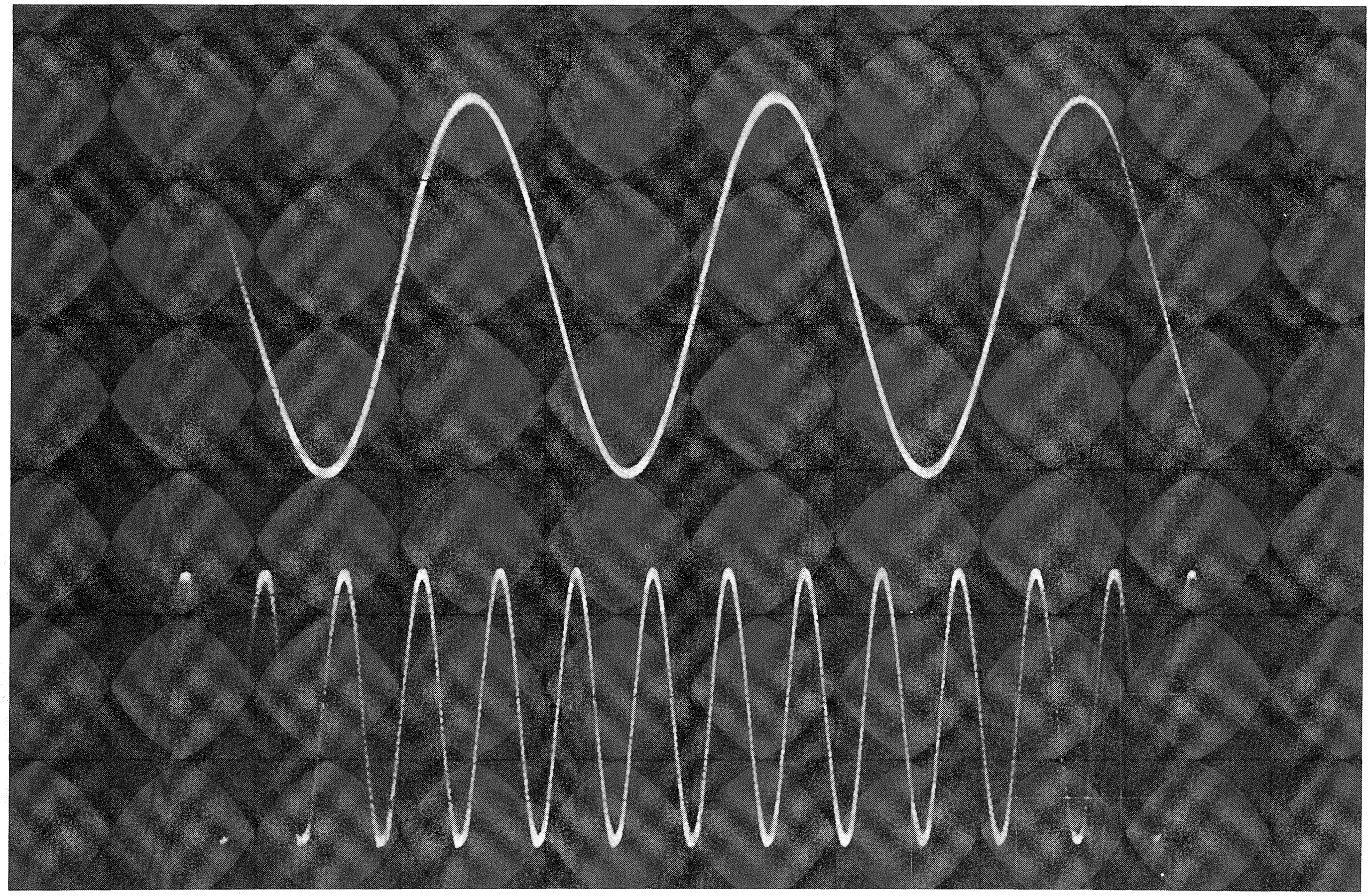
Table 1. Oscillation Voltage

Since the bias current is almost directly proportional to the voltage on point B, it can be changed within −29% to +33% when using STD or FeCr tape, and within −18% to +14% with CrO_2 tape.
The third meter reads the voltage on point A. A bridge circuit is incorporated in order to clearly show the indication of the meter.
Figures of the third meter scale (STD, FeCr and CrO_2) are indications for bias adjustment and have no units.
There are lots of kinds of tapes on the market. The bias control positions of leading brand tapes are shown in table 2.
When a tape is being used for the first time, set the bias control according to this table.
When you get accustomed to using this tape deck and are aware of tape characteristics, it is quite possible to actively utilize this system.
Consequently, according to program sources or your favorite source of sound, optimum recording quality is possible by adjusting the characteristics of the high end.

	Brand of tape	Bias control position (%)	EQ switch position
BASF	LH C-60	−15 (−10~−20)	STD
	LH C-90	−10 (−5~−15)	
	LN C-60	−20 (−10~−20)	
	LN C-90	−10 (−5~−15)	
	LH SUPER C-60	−15 (−10~−20)	
	LH SUPER C-90	−10 (−5~−15)	
AGFA	SUPER COLOR C-60	−10 (−5~−15)	
	SUPER COLOR C-90	−10 (−5~−15)	
	SUPER DYNAMIC C-60 +6	−10 (−5~−15)	
	SUPER DYNAMIC C-90- +6	− 5 (0~−10)	
SCOTCH	LH C-60, C-90	−20 (−10~−20)	
	CRYSTAL C-60, C-90	0 (−5~+5)	
	MASTER C-60, C-90	+ 5 (0~+10)	
TDK	D C-60, C-90	−15 (−10~−20)	
	SD C-60, C-90	−10 (−5~−15)	
	ED C-60, C-90	−10 (−5~−15)	
	AD C-60, C-90	+10 (+5~+15)	
MAXELL	LN C-60	−10 (−5~−15)	
	LN C-90	− 5 (0~−10)	
	UD C-60, C-90	+10 (+5~+15)	
	UD XLI C-60, C-90	0 (−5~+5)	
FUJI	FL C-60, C-90	− 15 (−10~−20)	
	FX C-60	0 (−5~+5)	
	FX C-90	+10 (+5~+15)	
	FX Jr C-60, C-90	+ 5 (0~+10)	
	FX DUO C-60, C-90	0 (+5~−10)	
SONY	LN C-60	−15 (−10~−10)	
	LN C-90	−10 (−5~−15)	
	HF C-60, C-90	0 (−5~+5)	
SONY	DUAD C-60	0 (−10~+10)	FeCr
	DUAD C-90	−10 (0~−15)	
BASF	FERROCHROM C-60	0 (−10~+10)	
	FERROCHROM C-90	−10 (−5~−15)	
SCOTCH	CLASSIC C-60, C-90	−15 (−5~−15)	
AGFA	CARAT C-60 CARAT C-90	0 (−10~+10)	
BASF	CHROME C-60	− 5 (0~−15)	CrO_2 (Chrome) Automatically selected
	CHROME C-90	−10 (−5~−15)	
SCOTCH	MASTER 70µs EQ C-60	0 (−5~+5)	
TDK	SA C-60, C-90 KR C-60, C-90	0 (−5~+5)	
MAXELL	C-60 CR, C-90 CR	−10 (−5~−15)	
	UD XLII C-60, C-90	−15 (−5~−15)	
FUJI	FC C-60	−15 (−5~−15)	
	FC C-90	−10 (−5~−15)	
SONY	CR C-60, CR C-90	+10 (+5~+13)	
AGFA	STEREO CHROM C-60 STEREO CHROM C-90	0 (−10~+10)	

Table 2. Bias control setting

Electricity: Basic Theories
Electric Power and Alternating Current



In previous issues (No. 1 and No. 2), we covered the basics of electricity–Ohm's law and its applications. In this issue we will begin to study the theory of alternating current, which is the basis of audio equipment systems. Let's begin with a discussion of electric power.

1. Electric Power

As you know from your own experiences, when you send an electric current through a conductor or resistor, heat is produced.

As Fig. 1 shows, when you connect a battery and a resistor, and create an electric current, the heat produced in the resistor over a certain period of time is proportional to the square of electric current and directly proportional to the resistance.

This relationship, called Joule's law, was discovered in 1841 by James P. Joule.

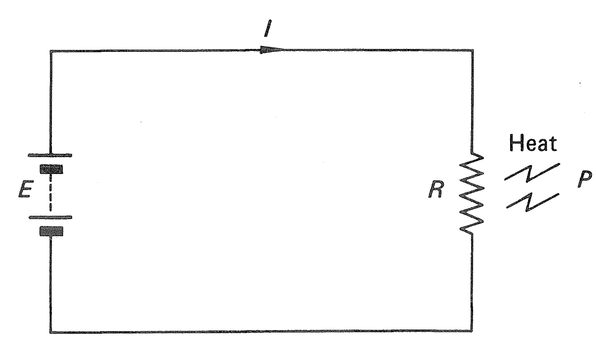


Fig. 1 Production of Joule heat

The relationship is shown by the following equation:

$$J = I^2 R \quad (\text{Unit: } J = \text{Joule}, I = \text{A}, R = \Omega) \ldots (1)$$

A Joule is an absolute unit of energy and equivalent to 0.239 cal. (A cal is a calorie. One cal is the amount of energy required to increase the temperature of 1g of pure water from 14.5°C to 15.5°C under 1 barometric pressure.)

As expressed in the above equation, if current and resistance are constant, the electric Joule heat is constant.

The ability to produce this Joule heat in the unit period (one second) is called electric power and its unit is expressed by "W" (watt). Electric power is usually represented by "P."

Accordingly, they can be shown in the following equation: $1W = 1J/s = 0.239cal/s$.

The relationship of watt, electric current and resistance is shown in the following way:

$$\left.\begin{aligned} P &= I^2R \\ &= EI \\ &= E^2/R \end{aligned}\right\} \text{(Please refer to Ohm's law)} \quad \ldots (2)$$

2. Sine Wave Alternating Current

In direct current (without relating to time passing) the voltage and electric current is unchangeable. On the contrary, along with time passed, regularly changing electric current is called alternating current. There are various kinds of alternating currents; however, let's study sine-wave alternating current which is widely used and is the basis for the theory of audio equipment.

1) What is Sine-Wave Alternating Current?

Audio signals are alternating signals and the figure which shows how many times they alternate in one second is called frequency. The unit is Hertz (Hz) and is generally expressed by "f." The time required for one cycle is called a "period"; its unit is "second" and is generally expressed by "T."

Accordingly, the following formula can be obtained:

$$f = \frac{1}{T} \quad \ldots\ldots (3)$$

As shown in Fig. 2, assume there is "E" (volume expressed by a length and a direction). The vector is revolving at a certain speed, "ω" (so-called angular velocity). Please see notes on page 30.

The end of this vector describes a circle whose radius is E, centered on the other end of the vector. When the former end of the vector is translated into a graph at time intervals starting from point 0, the wave shown at the right is obtained. This is the sine wave, which is the most basic wave of alternating current.

Now let's think about how the height of the wave and the amplitude changes with the lapse of time.

When the period of time has passed, the vector has revolved through the angle which is equivalent to $\omega \times t_1$. So e_1, the amplitude of the wave at that moment, t_1 (called an instantaneous value), is as follows:

$$e = E \sin \omega t \quad \ldots\ldots (4)$$

The unit of angular frequency (ω) is radian/second (rad/s) (See Note 2). 1 rad is 180°, so 90° is $\frac{\pi}{2}$ rad and 360° is 2π rad. Accordingly:

$$\omega = 2\pi f \quad \ldots\ldots (5)$$

When this relationship is put into the formula it appears as follows:

$$e = E \sin 2\pi f t. \quad \ldots\ldots (6)$$

Since the equation (6) is the most basic formula of characteristics and phenomena of electric circuits which we are going to study, please be sure to remember it.

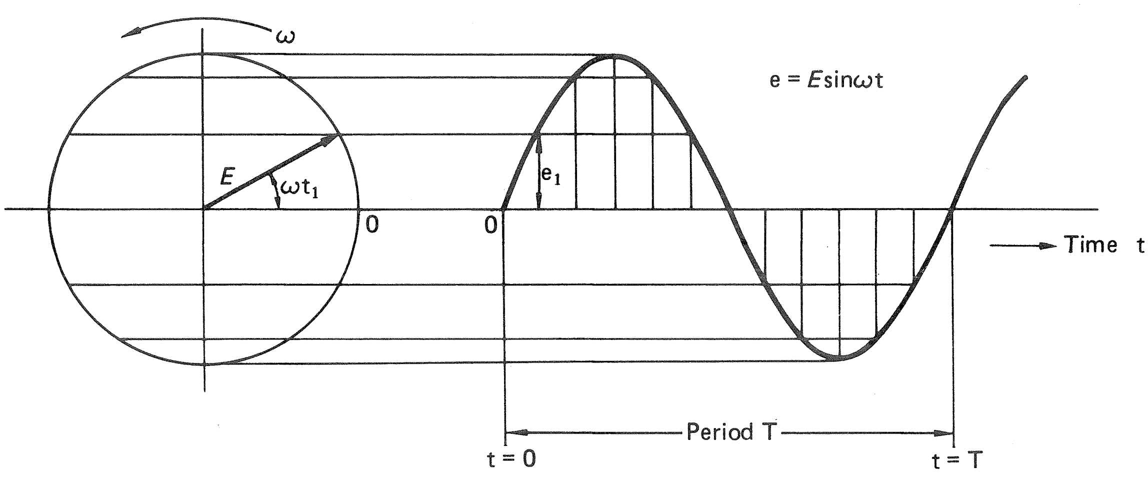


Fig. 2 Sine wave

2) The Expression of AC Voltage

Since the volume is unchangeable in the case of direct current, it is very easy to determine the voltage. However, if the volume and direction are always changing, as in the case of alternating current, a defined moment of voltage is indicated.

Usually in the case of sine wave alternating current, maximum value effective value and mean value are used depending on the case. The three values are also applied to electric current.

Maximum value is the maximum part of the sine wave in a half period and "E" in an equation. The unit is volt.

Let's think about the consuming electric power when AC current is fed into a resistor as shown in Fig. 3.

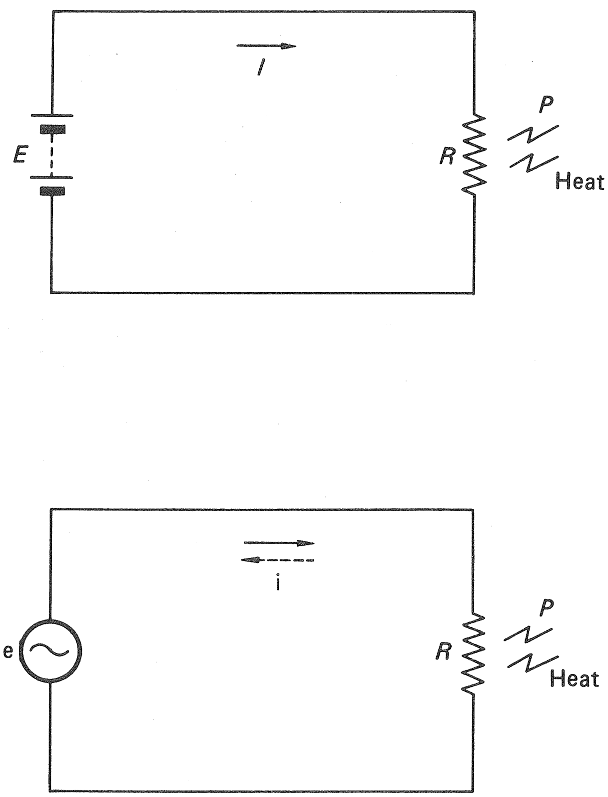


Fig. 3 Consuming electric power

Here, let us assume that voltage is $1V$ and current is $1A$. Electric power is voltage x current. In direct current the power can be easily determined as follows:

$1V \times 1A = 1W$.

However, in the case of alternating current, since voltage and current are changing according to time passing, electric power is also changing with time. Fig. 4 illustrates this.

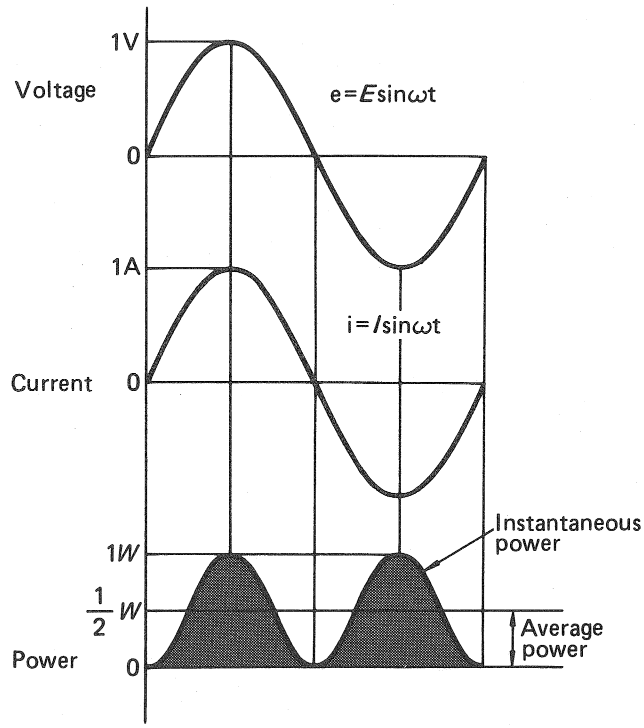


Fig. 4 Alternating current power

Actual consuming electric power of the resistor is simply $\frac{1}{2}$ of the product of $E \times I$.

Actually, to calculate electric power, it is inconvenient and illogical if results are different between direct and alternating currents.

Accordingly, the effective value, equivalent to the value of energy in direct current, is used for the voltage and current of alternating current.

Mathematically, the effective value is the value obtained after determining the square root of the average momentary value in one period. Therefore it is also called "root mean square" and the unit is expressed in abbreviation as Vrms and Arms. Consequently, effective value of sine wave is:

$\sqrt{\frac{1}{2}} = \frac{1}{\sqrt{2}} =$ 0.707 of maximum value.

Usually, described value of alternating electric voltage and current is the effective value of the sine wave and the units are simply expressed by "V" or "A."

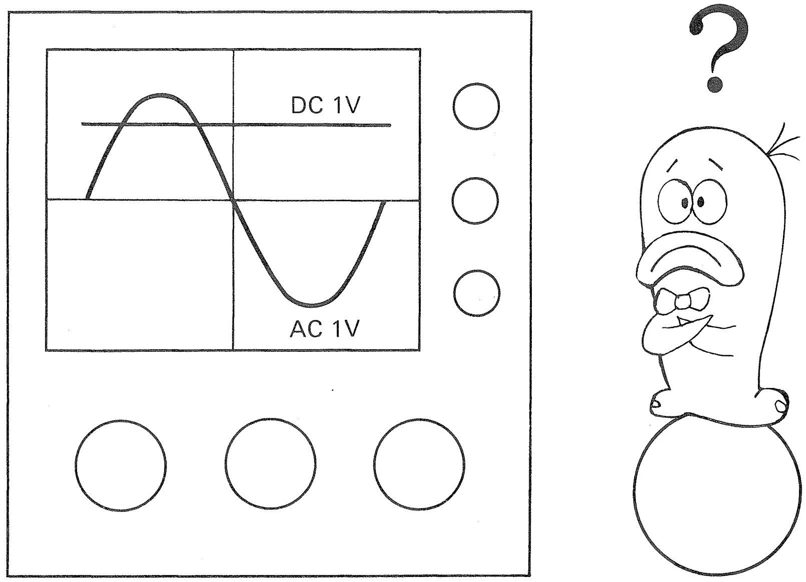


Fig. 5 Uh-huh!

3) Phase

The word "phase" is frequently used in the field of audio. Phase originally meant a difference or a relationship between a momentary point and a standard point of a periodical wave and is expressed by an angle, assuming that one cycle is 360° (this is called an electrical angle).

A phase relationship or difference of two or more voltages and/or electric currents is expressed as: phase is advanced or delayed. This is illustrated in Fig. 6.

Be sure to note that here the lateral axis represents a different angle from that of Fig. 2.

Vector E_A and E_B are revolving by angle velocity ω; however E_B starts after E_A, delaying by ϕ.

The changes of E_A and E_B are shown in sine waves A and B on the right of Fig. 6.

These two sine waves are expressed as follows:

$$e_A = E_A \sin \omega t = E \sin\omega t$$

$$e_B = E_B \sin\omega t = E \sin(\omega t - \phi)$$

"ϕ" represents the phase angle by which c_B is delayed to e_A.

On the other hand, taking e_B as a reference, e_A is advanced over e_B by ϕ. This can also be expressed as:

$$e_A = E \sin(\omega t + \phi)$$

$$e_B = E \sin\omega t$$

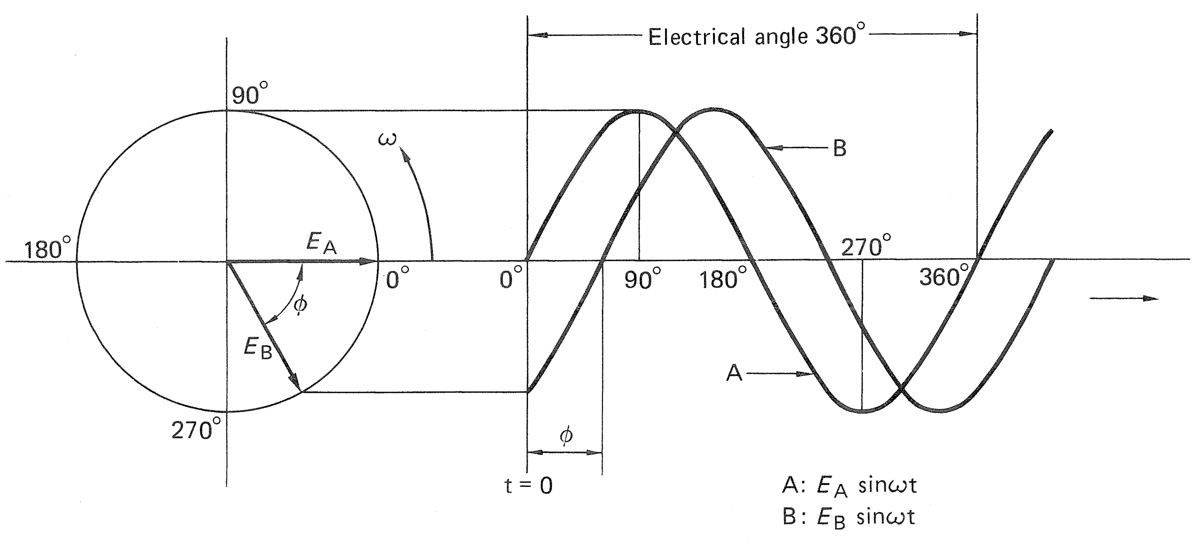


Fig. 6 Phase difference between two sine waves E1

4) Average Value and Peak Value

In addition to maximum value and RMS value, average value and peak value are in some cases utilized for sine-wave alternating current. Understanding these two values is a necessity when making measurements and using measuring instruments.

● Peak Value

The difference between the maximum and minimum value in a cycle, or the addition of the height of one peak and depth of another is called "Peak to Peak," $V_{p\text{-}p}$ for the voltage and $A_{p\text{-}p}$ for the current. Peak value is $2\sqrt{2}$ times the RMS value.

Thus, 1Vrms is equal to 2.828Vp-p (see Fig. 7).

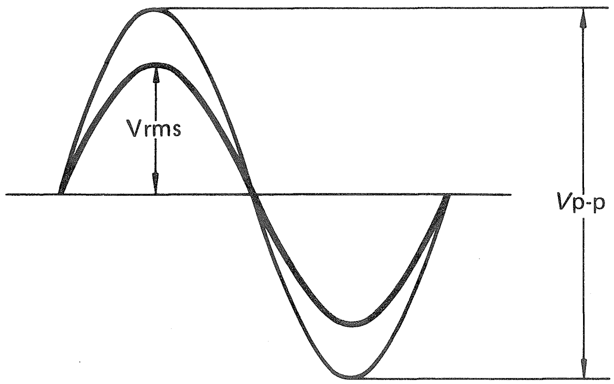


Fig. 7 RMS value and peak value

It is considerably important to know these values when you measure voltage by using an oscilloscope.

● Average Value

This is the average rate of voltage or current of various wave forms within a certain period of time or term. The average of one cycle is 0 in the case of a sine wave so the average value is determined as the average of the voltage or current of a half cycle.

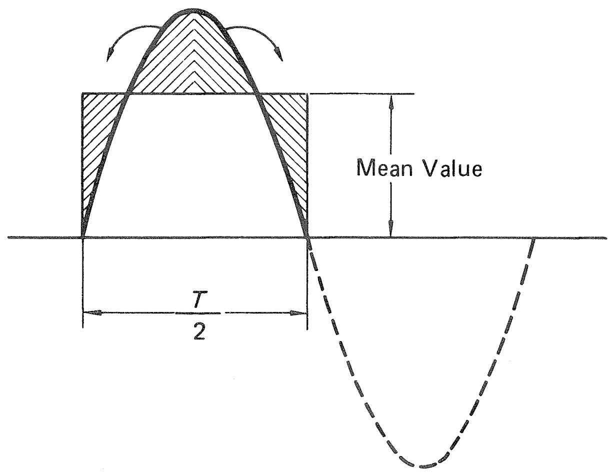


Fig. 8 The determination of the average value

The top oblique-lined area is used to fulfill the other oblique-lined areas (see Fig. 8).

Therefore, the average value of a sine wave is $\frac{2E}{\pi} = 0.637$ (E = maximum value)

It is essential to know the average value in order to understand the measurement principle of the voltmeter (multimeters and ACmV meters) with the moving-coil type ammeter. Please refer to the AC mV meter (pp. 31–36) of this manual.

Note 1)

Making the right-angled triangle's leg (a), the base (b), and the hypotenuse (c), as shown in the illustration, they are defined as follows:

$\frac{a}{c} = \sin\theta$ .... (1)

$\frac{b}{c} = \cos\theta$ .... (2)

$\frac{a}{b} = \tan\theta$ .... (3)

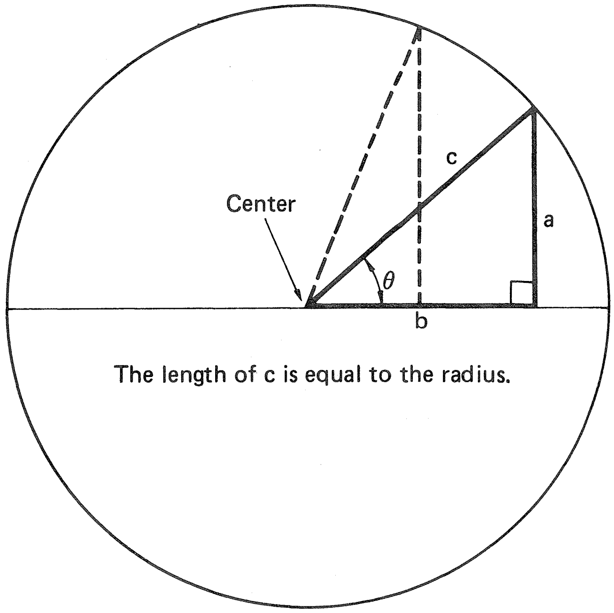
$a = c \sin\theta$ (from equation 1) .... (4)

You can get the value of (a) from equation 4 if (c) and θ are given. Making the angle θ_1 after vector starting from θt_1 seconds ago in Fig. 2, the equation becomes:

$\theta_1 = \omega t_1$

If this is adapted into equation 4, it becomes:

$e_1 = E\sin\omega t_1$



The length of c is equal to the radius.

Note 2)

If the length of arc ℓ equals r in circle radius r, you can call this center angle 1 radian toward this arc, and the unit is represented as rad.

The relationship between radian and degree is as follows:

the circumference is $2\pi r$, so:

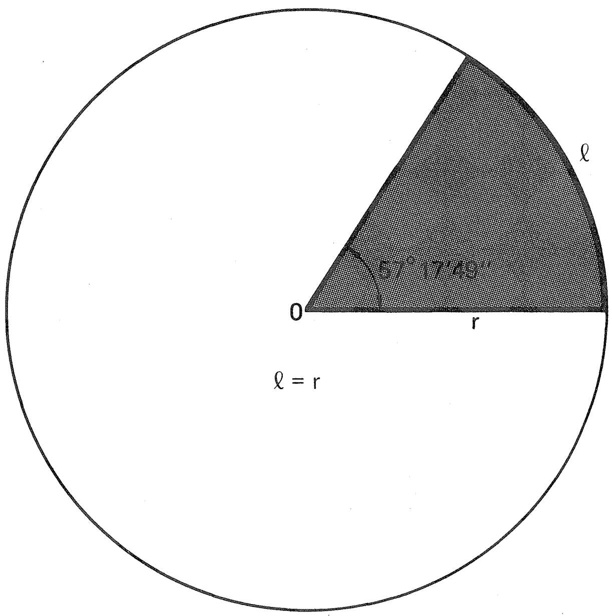
$$360° = \frac{2\pi r}{r}$$

$$= 2\pi \text{ (rad)}$$

and πrad becomes 180°. Furthermore:

$$\frac{180°}{\pi} = 57.3°$$

This is about 57° 17′49″



ℓ = r

The angular velocity means the speed per second in the circular movement, and the unit is represented as rad/sec. The angular velocity is denoted by the Greek letter "ω."

You then get the following expression:

$\omega = 2\pi f$ .... (5)

Measuring Instruments (3)
The AC Millivolt Meter (mV Meter)

The AC Millivolt Meter (mV Meter)

This time around we would like to discuss the AC millivolt meter, or mV meter for short.
Multimeters, as we have already mentioned, are used extensively by service engineers to check circuits. However, multimeters have the following disadvantages:

1. Incapable of measuring low voltages.
   When used with a high-sensitivity DC voltmeter, the multimeter can measure comparatively low DC voltages. But, when used as an AC voltmeter, the multimeter is incapable of measuring low AC voltages.
2. Low internal resistance (impedance).
   When used to measure high-impedance circuits, the multimeter loses its accuracy.
3. Comparatively poor frequency characteristics.
   Measuring frequencies below 30Hz results in heavy swinging of the pointer at the measured frequency, making accurate reading difficult. Measuring accuracy is also lost at high frequencies, due to the frequency characteristics of the internal circuitry and diodes in the meter itself. Depending on the grade of the multimeter, the acceptable upper-frequency limit is about 20kHz.

In order to overcome these disadvantages, AC mV meters are used to measure voltage in audio circuits which have minute voltage, high impedance and wide frequency characteristics. In addition, the signal levels in audio circuits are often given in decibels (dB) which require a high measuring accuracy.

1. Composition and Operating Principle of an AC mV Meter

Whereas an ordinary multimeter rectifies the AC input voltage directly by means of a full-wave rectifier circuit which drives a DC ammeter, the mV meter amplifies the input signal (AC voltage to be measured) and rectifies the amplified voltage which drives a DC ammeter (see Fig. 1).

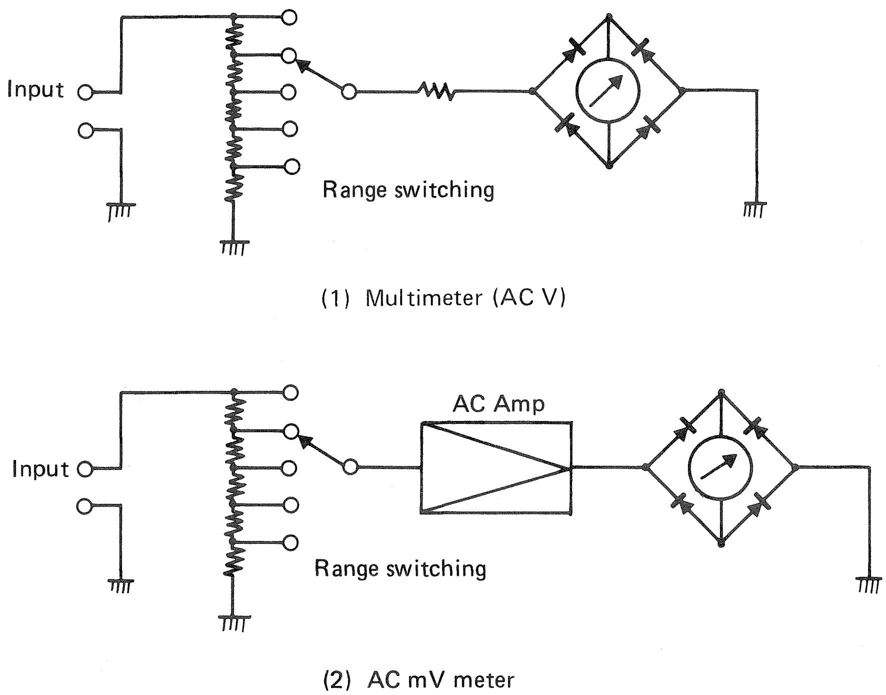


Fig. 1 Basic multimeter and mV meter circuit composition

Actual AC mV meters are, in fact, equipped with a buffer amplifier which provides very high input impedance, a protection circuit against overloading or breakdown when excessive inputs are applied, a 10dB step attenuator for easy handling, and a negative feedback circuit to ensure even scale distribution (see Fig. 2).

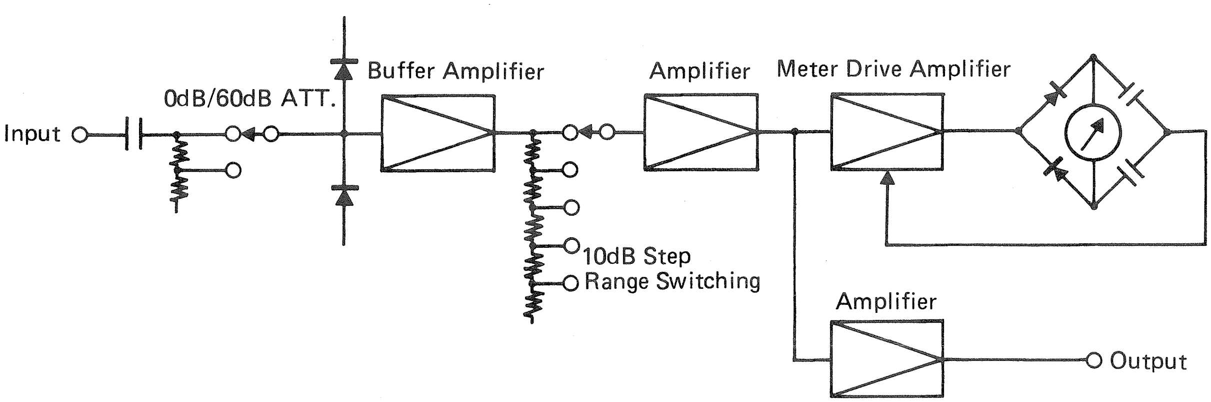


Fig. 2 Circuit composition of AC mV meter

Since rectified AC current flows through the meter, the mean (average value of the rectified ripple current) is indicated. The mean current of a half-cycle sine wave signal is equivalent to 0.637 times the peak value of the sine wave signal (see Fig. 3).

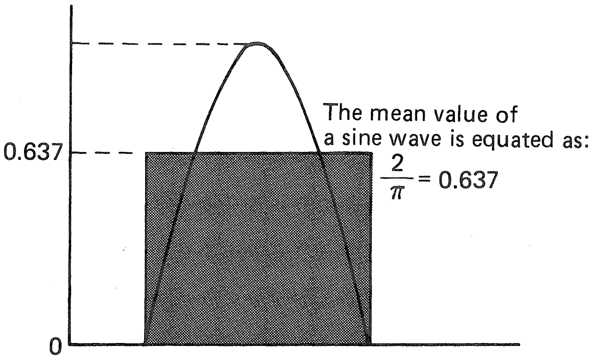


Fig. 3 Mean value of sine wave

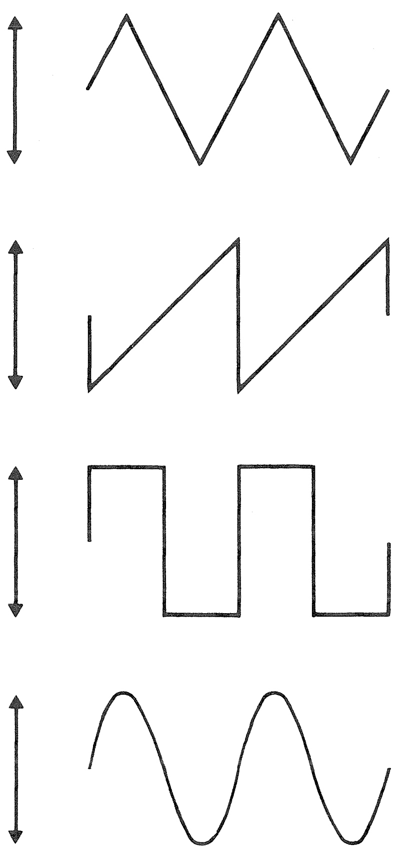
Unless otherwise specified, the voltage value of an AC current is that of the sine wave. Therefore mV meters also must indicate in RMS.
Meter scales are graduated to indicate root means square (RMS) values calculated from the mean value of the signal. At the time of calibration, a 1V rms sine wave signal is supplied to the meter input and the position the pointer indicates is marked 1V(rms).

Meter accuracy will be lost and error voltages indicated if the measured signal is anything other than a sine wave, i.e. square, triangle or sawtooth wave. Moreover, measurement errors will also result if the sine wave signal possesses harmonic distortion. However, in the actual measurement of audio circuits, the input signal (audio signal) is normally of low distortion, thus meter error due to harmonics is negligible.
Examples of signal waveform distortion and resultant indication errors are shown in Table 1.

Measured signal	RMS value(%)	Meter indication(%)
Basic wave at 100% amplitude	100	100
100% basic wave plus 10% 2nd harmonics	100.5	100
100% basic wave plus 20% 2nd harmonics	102	100 – 102
100% basic wave plus 30% 2nd harmonics	112	100 – 110
100% basic wave plus 10% 3rd harmonics	100.5	96 – 104
100% basic wave plus 20% 3rd harmonics	102	94 – 108
100% basic wave plus 30% 3rd harmonics	112	90 – 116

Table 1 Waveform distortion and indication error

Signal level indication and meter error for waveforms other than sine waves are shown in Fig. 4.



Waveforms	Mean value of 1Vp-p	RMS value of 1Vp-p	Meter indication	Error (%)
	0.25	0.29	0.96	3.8
	0.25	0.29	0.96	3.8
	0.5	0.5	1.11	11.0
	0.32	0.35	1.00	–

Fig. 4 Meter indication and error for various signals

2. Meter Types and Performances

Although there are various ways of classifying AC mV meter types, we have limited them to the following three for easier understanding:

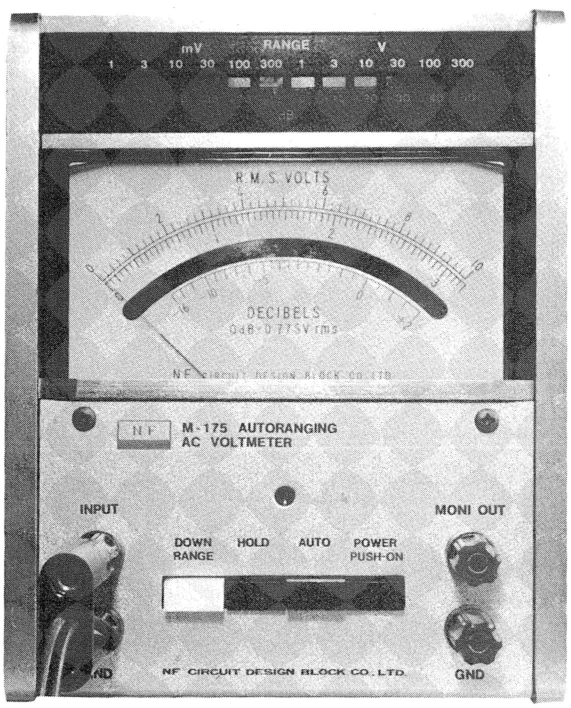
1) Ordinary 1-channel type
2) 2-channel type
3) Automatic range-switching type

The 1-channel mV meter is the most popular. It gives a full-scale voltage reading from 1mV to 1.5mV at the lowest range with an accuracy of ±3% and an operating frequency range from 5Hz to 1MHz. Certain high sensitivity mV meters give a full-scale reading as high as 50μV at the lowest range.

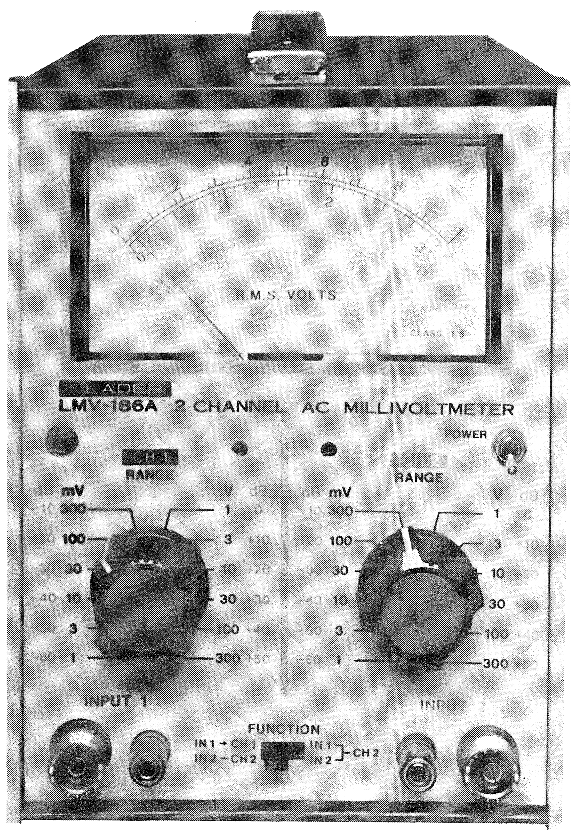
Photo 1 Various mV meters

(1) 1-Channel Type

(3) 2-Channel Type 1



(2) Automatic Range-Switching Type



(4) 2-Channel Type 2

Although some mV meters can be used up to 1,000 MHz, they are not intended for measuring audio circuits and have, therefore, been omitted here. With the 2-channel type, two 1-channel meters are combined into a single unit sharing a common scale. For practicality, a 2-channel meter is more convenient than two 1-channel units. It demonstrates its versatility when measuring and comparing the left and right signal levels of stereo amplifiers, or when both the input and output levels need to be measured at the same time. This makes the 2-channel meter an indispensable item on any service bench.
The automatic range-switching type mV meter, in addition to being so convenient, also offers the same performance as other types. Since the meter range is switched automatically, this type of meter is particularly useful when different voltage levels need to be measured repeatedly.
However, when the voltage to be measured is known, the automatic range-switching meter becomes rather troublesome since it automatically switches ranges as the input signals are interrupted. In which case, the operating mode should be switched to manual.
Specifications common to all these types of mV meters are listed below.

Measuring range
  Voltage 1mV to 300V 12 ranges
    1mV, 3mV, 10mV, 30mV, 100mV, 300mV, 1V, 3V, 10V, 30V, 100V, 300V
  dB range −60dB to +50dB, in 10dB steps
Measuring accuracy
  ±3% of full scale at 1kHz
Frequency characteristics
  5Hz to 1MHz, ±10% at 1kHz
Input impedance
  10MΩ (each range)
Input capacity
  50pF or less (1mV to 300mV range)
  35pF or less (1V to 300V)
Amplifier output voltage
  1V rms at full scale
Output impedance
  600Ω ±20%

As these specifications clearly indicate, mV meters offer higher performance and operating ease than multimeters for measuring AC voltages.

3. Operation and Precautions

3-1 Before measurement

1) Prior to turning the power switch ON, be sure to check that the pointer registers zero. If it doesn't, zero-adjust first.
2) Set the range-switching knob to the highest voltage range.
3) Wait approximately 10 seconds after switching the power ON before using.

3-2 AC voltage measurement

If the AC voltage to be measured is very low, or the circuit to be checked has a comparatively high impedance, an external induction voltage is likely to occur, in which case, a shielded cable or coaxial cable complying with the frequency of the signals to be measured should be used.
Ordinary parallel cable is adequate for measuring low-impedance circuits even if the signal frequency is high and the voltage is very low.
Measurement should be started from the highest voltage range so as not to overload the meter. If the pointer remains below the 30% level, the voltage range can be attenuated step by step. This operation is unnecessary with the automatic range-switching type meter.

3-3 dB measurement

Millivolt meters have both dB and Vrms scales. There is also a choice of dBm only or both dBm and dBV. Whereas dB representation gives a relative value, the dBm and dBV scales represent absolute levels. The dBm value represents the voltage at which 1mW of power is consumed in a 600-ohm circuit. Therefore, 0dBm equals 0.775V. Even if the scale is simply marked dB, it actually reads dBm. Meanwhile, 0dBV is equivalent to 1Vrms regardless of circuit impedance. For audio equipment, absolute values are normally dealt with regardless of the input/output impedance of the circuit under measurement. For this reason, the dBV value is more practical than the dBm, and it is frequently used for the adjustment of tape decks. Therefore, anyone contemplating buying an mV meter is recommended to select one with a dBV scale.

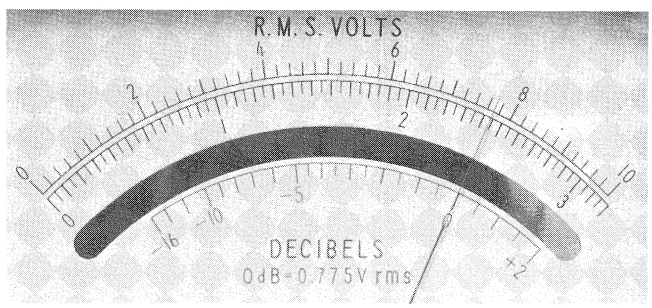


Photo 2 dBm scale

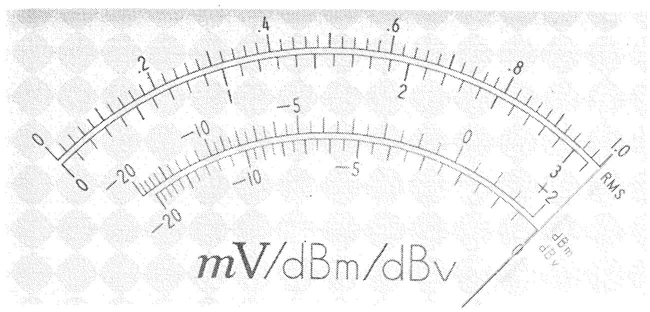


Photo 3 dBV scale

3-4 How to use the output terminals

All mV meters are normally equipped with output terminals. Which means that a signal proportional to the input signal can be obtained from these terminals via an internal amplifier. For example, with LEADER's LMV-181 and LMV-185 models, a 1Vrms output can be obtained at full scale regardless of the voltage range setting. Other manufacturers set the full-scale output voltage at 0.775V, which represents OdBm. Moreover, since the mV meter can be used as a flat amplifier, it also functions as a preamplifier. And in cases where waveforms of very small signals need to be determined using an oscilloscope, or when cartridge output levels are insufficient for a wow and flutter meter input reading, the mV meter makes these measurements possible.

3-5 Precautions when measuring

Avoid supplying any input voltage which exceeds the maximum for which the meter is designed. For example, if the mV meter has a maximum input voltage rating of 300V, then the allowable input voltage limit is 300V for sine wave signals and 840Vp-p for all others. Even though mV meters are provided with a protection circuit, excessively high input voltages may result in breakdown of the internal circuit elements for amplification or meter indication.

3-6 How to use the AC mV meter in daily service-bench operations

For daily servicing, the mV meter should not be used in place of a multimeter. Instead, it should be used whenever the signal level to be measured is beyond or close to the limit of a multimeter's capability. In other words, the two meters should be used as follows:

Multimeter — For general servicing such as DC voltage checks, AC power voltage checks, excluding signals, and quality checks of parts.

AC mV meter — Adjustment

Tuners and receivers: Stereo separation and signal meter adjustment, and recording level or output level checks.

Tape decks: Head height and azimuth adjustment, playback equalizer, recording bias, recording level, Dolby level, level meter and peak indicator adjustment.

Other measurements

In order to fully utilize the versatile functions of the AC mV meter, more than one is required per service bench. An ideal setup for a service bench is shown in Fig. 5. Also, refer to *Tuning Fork* No. 1, p. 28, STANDARD SERVICE BENCH for the setting-up procedure.

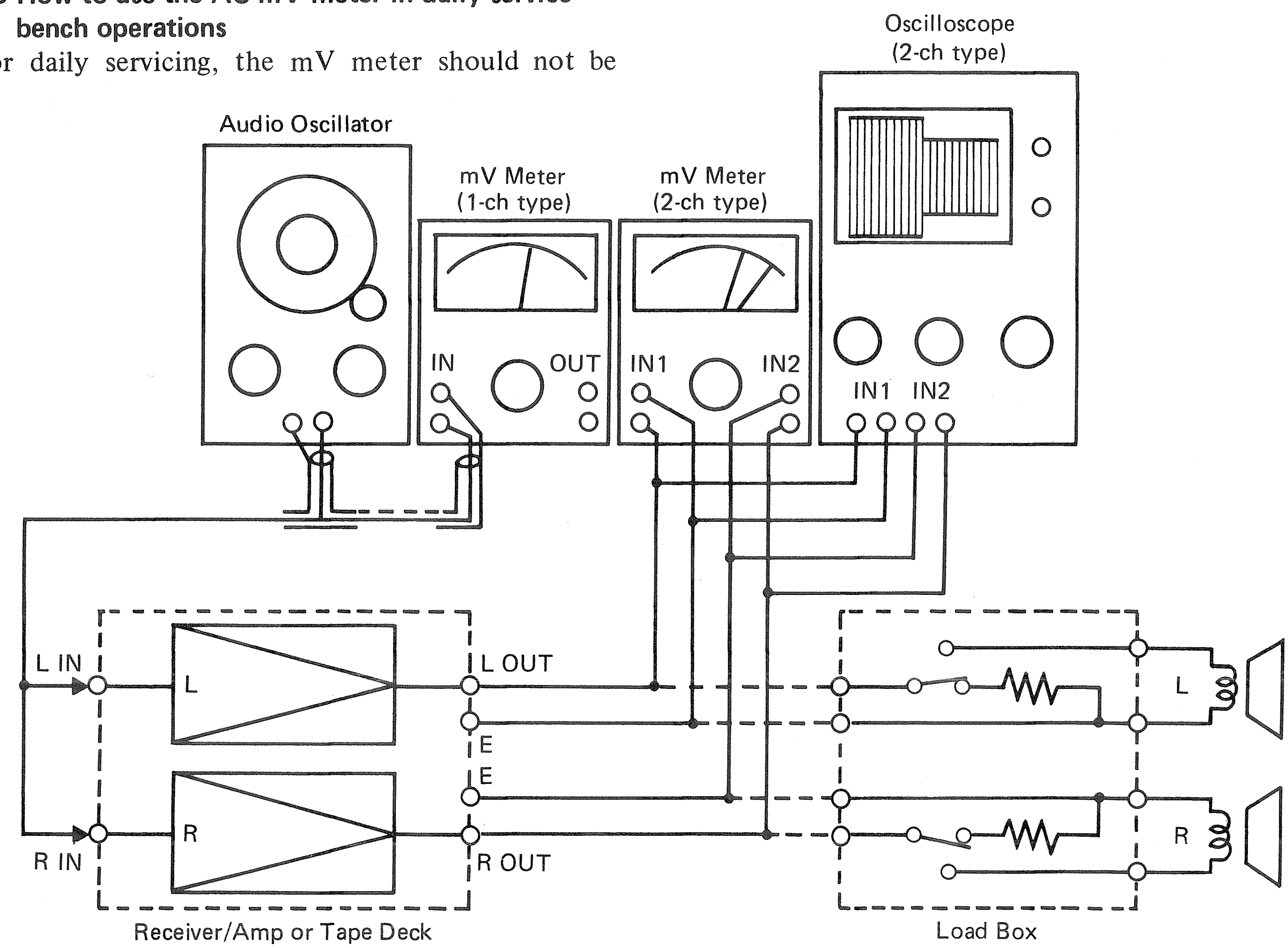


Fig. 5 Connections of mV meter and common measuring instruments

One-Point Servicing Techniques (3)

"AMP I" Countermeasures

Although we announced in the previous issue of Tuning Fork that we would be discussing IF & MPX adjustment, we will, in fact, talk about "Amp I" countermeasures instead. The reason for this change is threefold. Firstly, recent developments have resulted in higher-quality circuits and parts, making IF & MPX adjustment almost a thing of the past. Secondly, the parts which require adjustment are limited to certain models only, and these are clearly indicated in the respective service manuals. Finally, with the ever-increasing influx of electrical and electronic gadgetry in daily use, the number of noise- and radio-interference sources is increasing each year. This situation is causing concern among service engineers. So in answer to a number of reports of trouble in this area, we shall explain how to cope with hi-fi interference, its causes and remedies.

This is what amp interference sounds like

"Amp I"

Ideally, audio equipment should be immune to radio-frequency interference and unwanted external noise. In an effort to ensure that it is, many countermeasures are being taken in the design and manufacturing of the equipment. However, due to the high cost involved, it is impossible to eliminate all sources of interference. Radio-wave pollution is everywhere. In addition to numerous public broadcasting stations, there are many business and amateur stations as well as citizens' band users, all contributing to it.

The major causes of interference are excessively high-power transmission, spurious radiation, transmitters located too close to receivers and poor receiver protection. However, before the problem of interference can be tackled, the interests of all parties concerned—users, broadcasters and manufacturers—must be taken into consideration. Here we will discuss the various countermeasures currently being taken against external noise and interference. But first let us explain the meaning of "Amp I."

"Amp I" stands for Amplifier Interference. It is a kind of RF interference, and refers to RF energy transmitted at a high-enough magnitude to interfere with the operation of electronic equipment, in this case audio equipment.

The cause of this interference can come from any one of a variety of sources in which RF energy is detected. And the effects can be noticed in virtually every piece of audio equipment.

Although "Amp I" can be classified in various ways, such as tape deck or tuner interference, for the sake of simplicity we shall refer to everything collectively as "Amp I" here.

1. "Amp I." Its Main Causes and Entry Points

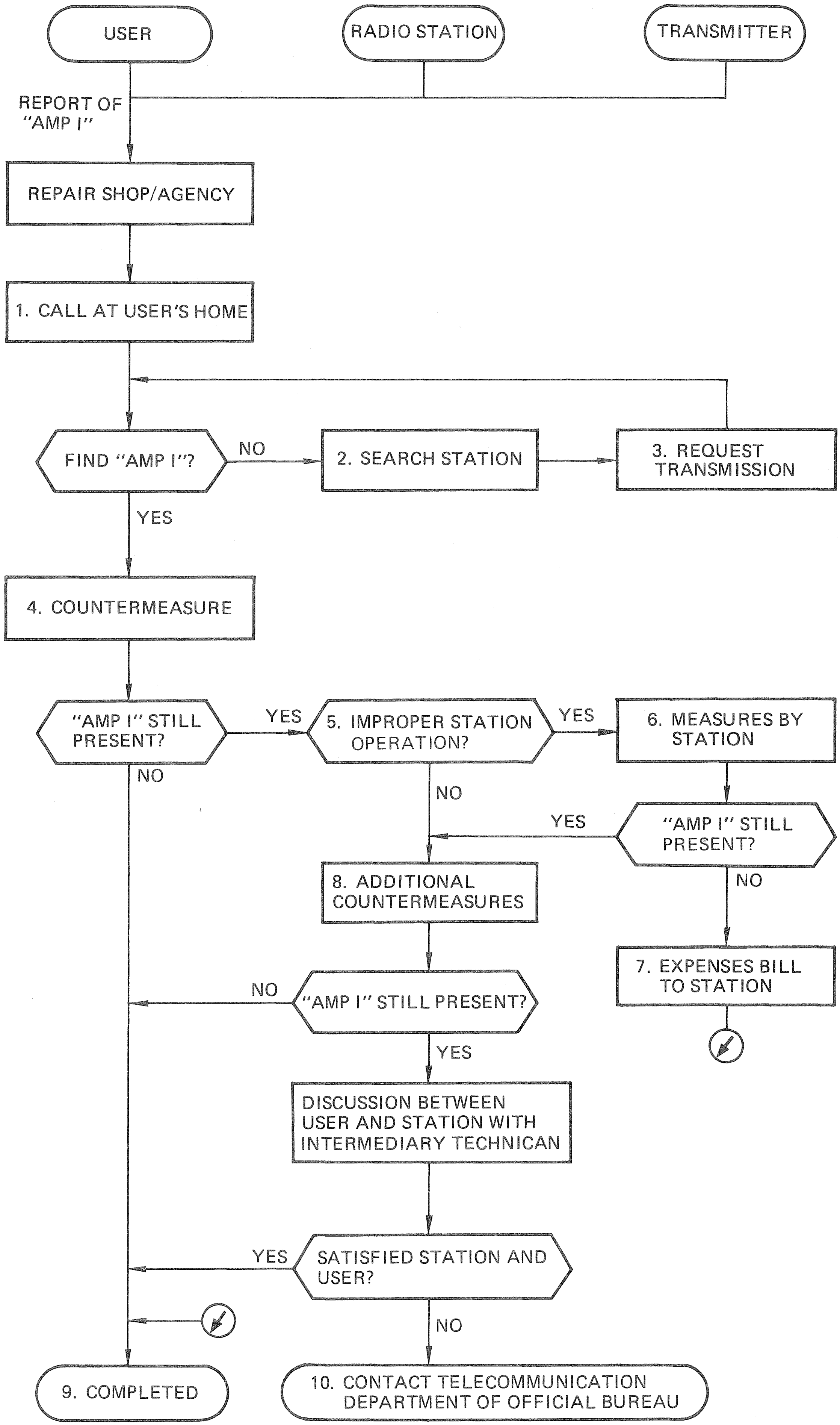
1.1 Main causes of "Amp I"

- Connecting cords used between components in an audio system may, due to their stray capacitance and inductance, act as resonant circuits for RF signals. When a broadcast frequency is picked up by the resonant circuit, the signal passes through the cords and enters the subsequent circuit, where it is detected and amplified by the various circuit components such as transistors and IC's.
- Powerful RF signals may enter directly into the circuit elements, bypassing the normal inputs. Phono cartridges, magnetic heads and circuits with non-linear amplitude characteristics are particularly susceptible to such interference.
- Incorrectly operated broadcasting equipment, or equipment that is incomplete or insufficiently protected against spurious radiation, causes RF signals to be induced into AC lines. This results in unwanted external noise and interference in audio equipment, TV's and other audio transducers in the immediate vicinity of homes using common lines.

1.2 Major "Amp I" entry points

- Turntable phono cartridges
- Magnetic heads of tape decks
- Connecting cords between audio equipment
- Primary power supply circuits
- Speaker cords
- Tuner antennas and connecting cords

2. Steps To Take When "Amp I" Occurs



Refer to the flow chart on the previous page.

Step 1. Call at user's home or business upon receipt of an "AMP I" report. If "AMP I" is detected in the system, proceed to Step 4 below.

Step 2. Should a check of the system fail to reveal any sign of "AMP I" it may indicate that the interfering station is not broadcasting. In this case, it will be necessary to visually check the vicinity for a possible source. It should be noted that interference most often comes from stations within a 100-meter radius of the user's home.

Step 3. When a possible source is found, explain the situation to the station personnel and request transmission for a period of time to permit troubleshooting.

Step 4. During the period of transmission, take any necessary countermeasures.

Step 5. If remedial measures prove ineffective, verify that the station is transmitting in accordance with prescribed operating procedures.

Step 6. If station management and transmission are not correct, such as broadcasting at excessive power or with spurious radiation, have the station operators take all appropriate measures necessary to correct the deficiency.

Step 7. If "AMP I" stops due to the station's corrective measures, bill the station for payment of travelling expenses and technical fees incurred as a result of the service call.

Step 8. Should you find nothing wrong with the station's operations, however, take the necessary actions according to the procedures in Steps 6 and 8 of the troubleshooting table.

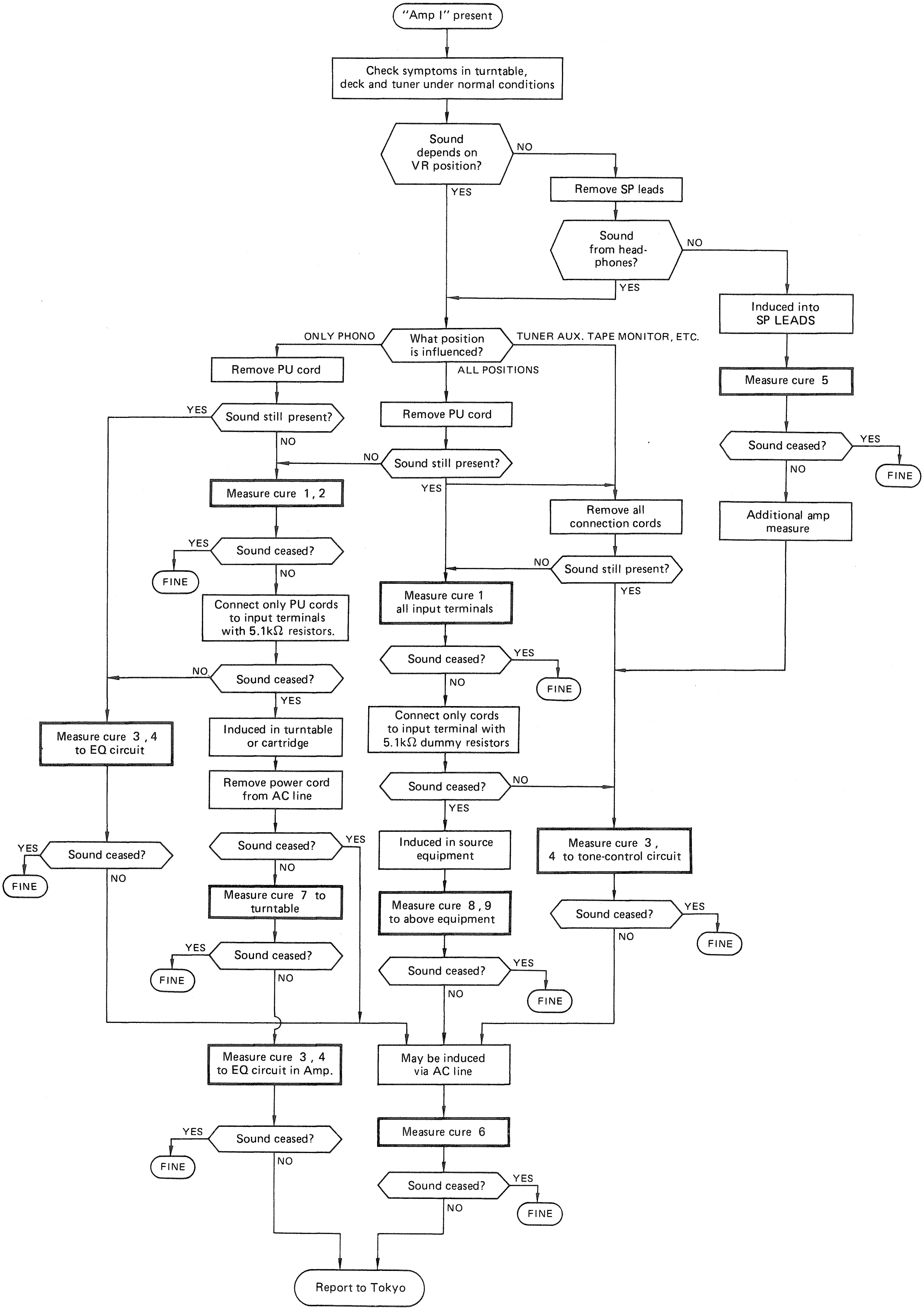
Step 9. After taking all remedial actions, your "AMP I" countermeasures are considered complete when the following have been achieved:

a) "AMP I" is no longer present.

b) The user accepts the measures taken as being sufficient.

c) The user and the interfering station reach a compromise with the service technician acting as a go-between; a compromise may constitute such action as a change of antenna location, a reduction of broadcasting power, the establishment of a time schedule for audio system and transmitter use, and so on.

3. "AMP I" Measurements and Flow Chart of Steps to be Taken



4. Remedial Actions

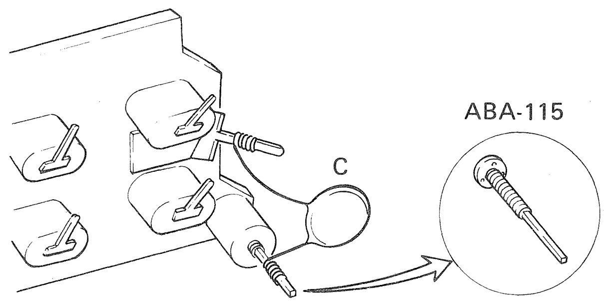
In many cases, the number of "AMP I" entry points in an amplifier can be many. Therefore, it is necessary to search out the most effective point for carrying out countermeasures.

If, during your search, you tamper with a part which is not related to "AMP I," put it back as it was lest it deteriorate performance.

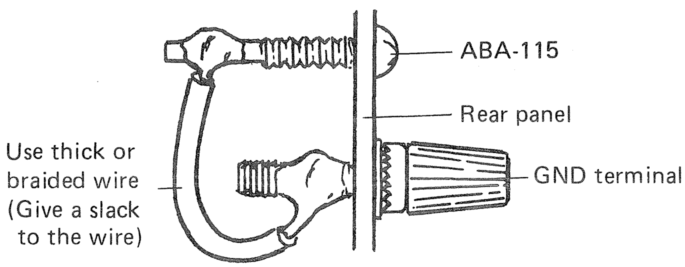
NOTE:

1. If the type of capacitors and resistors are not specified, use ceramic capacitors and 1/4W carbon-film resistors (RD).
2. Always keep to the values specified to prevent deterioration in performance.

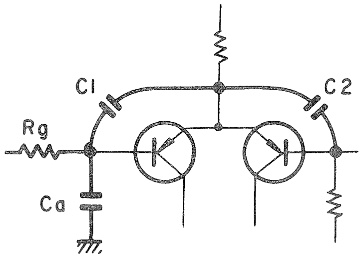
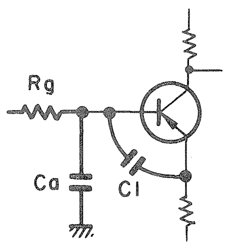
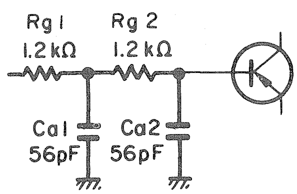
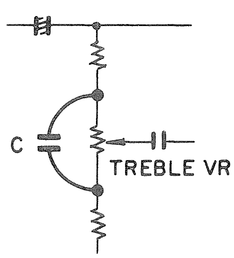
Entry Points: 1. Induced in the input terminal(s) or connecting wire(s)

How to detect	How to cure	Note
Refer to FLOW CHART.	(1) Connect a capacitor to the chassis and grounding point of input terminal.  ● Value: 0.01 ~ 0.047μF (CKDYF103Z50 ~ CKDYF473Z50) ● To ground the rear panel, replace the original screw with ABA-115 and solder one of the capacitor leads to it.	1. The length of the capacitor leads should be as short as possible. 2. ABA-115 should be fastened tightly.

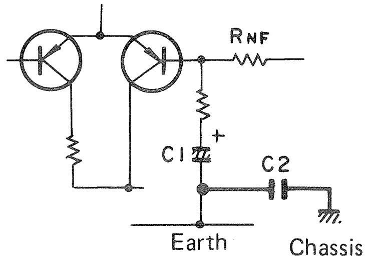
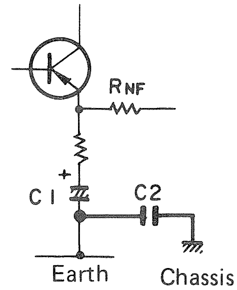
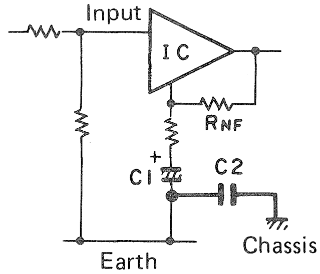
Entry Points: 2. Induced into the grounding wire of turntable output cord

How to detect	How to cure	Note
Connect only the grounding wire (black lead) to the GND terminal on the amplifier's rear panel.	(1) Add a new grounding point near the GND terminal using an ABA-115 screw and connect the terminal to the ABA-115 with a thick lead or braided wire.  	

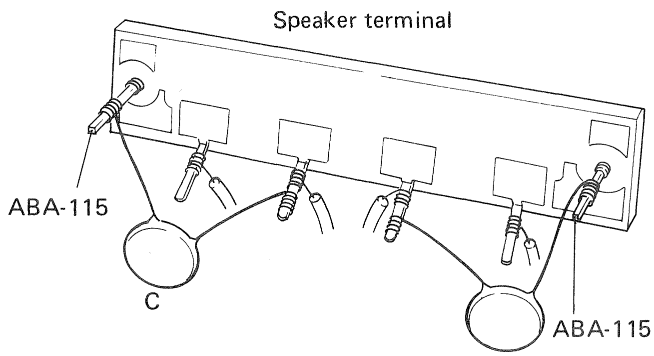
Entry Points: 3. Induced in the audio signal routes of the amplifier, such as the preamp stage and control circuits

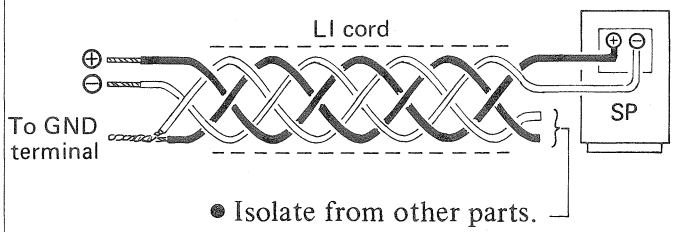
How to detect	How to cure		Note
Refer to FLOW CHART.	(1) Differential Amplifier  	a. Add capacitor(s) and/or replace C_1 and C_2 to achieve a capacitance of $C_1 = C_2 = 47 \sim 100pF$. (Most amplifiers employ C_1 originally) b. Replace Rg and Ca with larger value components, but not exceeding 2.2KΩ and 100pF, respectively.	
	(2) Conventional Amplifier Type 1  	a. Add a 47 ~ 100pF capacitor to C_1 between the emitter and the base. b. Replace Rg and Ca with larger-value components, but not exceeding 2.2KΩ and 100pF, respectively.	
	(3) Conventional Amplifier Type 2  	● Divide the filter consisting of Rg and Ca into two stages to get a sharper effect. The value of the parts should be: $2.2K\Omega = Rg_1 + Rg_2$ $100pF = Ca_1 + Ca_2$	
	(4) Insert a capacitor across the treble-control volume.  	● Capacitance value vs VR values should be: 470 ~ 680pF: 100KΩ up to 1000pF: 10KΩ	● This method only applies to tone-control circuits which employ B-type VR's (VAX or LUX type).

Entry Points: 4. Induced into the grounding (earth) loop in the amplifier

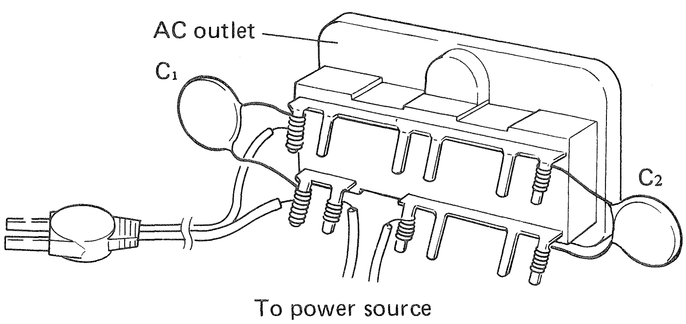
How to detect	How to cure	Note
Refer to FLOW CHART.	(1) Insert a capacitor between the earth line and the amplifier chassis.  a. Differential Amplifier  b. Conventional Amplifier Cure  c. IC-type Amplifier ● C_2: 0.047μF (CKDYF473Z50)	● C_2 should be connected to the nearest C_1 grounding lead and the shortest chassis earth point. Caution: If the power supply +B DC is for transistors grounded with a mylar or ceramic capacitor, this may cause oscillation. Therefore, in such a case, this method should not be used.

Entry Points: 5. Induced into the speaker leads

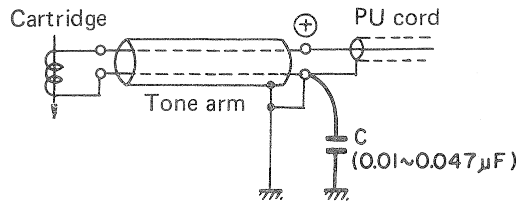
How to detect	How to cure	Note
Refer to FLOW CHART.	(1) Insert C into the minus (–) speaker terminal and amplifier chassis.  ● The capacitance of C should not exceed 0.001 ~ 0.047μF. ● To ground to the rear panel, replace the original screw with ABA-115 and solder one of the capacitor leads to it.	● The length of the capacitor C should be as short as possible. Caution: Confirm that amp is not oscillating.

How to detect	How to cure	Note
	(2) Replace the speaker cords with the cord JC-200.  ● Use LI cord and connect as shown above. Connect two wires to the GND terminal on the amplifier (GND terminal should be grounded completely by procedure No.2.)	
	(3) Try to bind, move wiring or adjust length of speaker cords to get the best condition.	

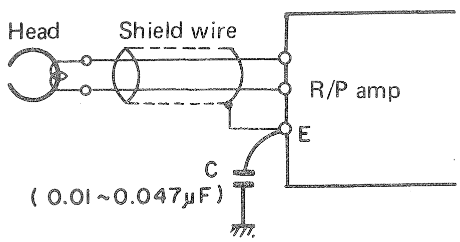
Entry Points: 6. Induced into the AC line

How to detect	How to cure	Note
Use an AC line filter.	(1) Insert a line-cross capacitor (C_1 or C_2) into the AC outlet terminals.  The respective capacitances should be: C_1 : ACG-003 (UL authorized capacitor 0.001μF) C_2 : ACG-001 (0.001μF) or ACG-003	An ACG-003 with insulator cover (AE279) should be used. Caution: Don't let capacitor leads touch others.
	(2) Insert an AC line filter into the AC line.	

Entry Points: 7. Induced in the phono cartridge or turntable lead wires

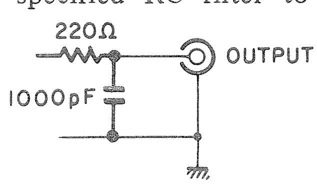
<table>
<tr><th>How to detect</th><th>How to cure</th><th>Note</th></tr>
<tr><td>"AMP I" is no longer audible when the head-shell or turntable output cord (PU cord) are disconnected.</td><td>(1) Connect the earth lead to the ground via the capacitor specified; employ the shortest route from the tonearm to the chassis.

</td><td>Never connect to the positive (+) lead.</td></tr>
</table>

Entry Points: 8. Induced in the tape deck

<table>
<tr><th>How to detect</th><th>How to cure</th><th>Note</th></tr>
<tr><td>● "AMP I" is no longer audible when the tape-deck output cord is disconnected.</td><td>(1) Remedy the tape deck between line in/line out grounding terminals and chassis as in procedure No. 1.</td><td></td></tr>
<tr><td>● "AMP I" is no longer audible when the tape-head leads are removed.</td><td>(1) Connect C, specified head lead wire, to the chassis.


● Replacing the head leads with shielded wire is sometimes effective, if shielded wire is not employed.</td><td>Caution:
Never connect a capacitor to the signal line.</td></tr>
</table>

Entry Points. 9. Induced in the tuner

<table>
<tr><th>How to detect</th><th>How to cure</th><th>Note</th></tr>
<tr><td>● Induced into the earth line of the tuner:
Even when the power switch is off, "AMP I" is sometimes present if only the grounding line of the output cord is connected to the amplifier.</td><td>(1) Connect the grounding terminal of the tuner output to the chassis, directly or via a 0.0047μF capacitor, using an ABA-115 screw.</td><td></td></tr>
<tr><td>● Induced into the signal line of the tuner:
One "AMP I" channel output cord of the tuner is removed.</td><td>(1) Connect a specified RC filter to the output terminal.

</td><td></td></tr>
</table>

First Step in Audio (2)

Specifications—2

	150 watts per ... 4 ohms from 20 hertz to 20,000 hertz with no more than 0.05% total harmonic distortion.
istortion: z)	No more than 0.05% (continuous rated power output) No more than 0.02% (60 watts per channel power output, 8 ohms) No more than 0.02% (1 watt per channel power output, 8 ohms)
Distortion: =4:1)	No more than 0.05% (continuous rated power output) No more than 0.02% (60 watts per channel power output, 8 ohms) No more than 0.02% (1 watt per channel power output, 8 ohms)
esponse:	5 to 100,000Hz +0dB, −1.5dB
vity/Impedance:	1V/50k ohms (POWER AMP. IN)
Speaker:	A, B, A+B
ctor: 0Hz, 8 ohms)	30
oise: -circuited A network)	100dB

Muting:	
FM TUNER SECTION	
Usable Sensitivity:	Mono; 9.8dBf (1.7μV)
50dB Quieting Sensitivity:	Mono; 14.2dBf (2.8μV) Stereo; 37dBf (39μV)
Signal-to-Noise Ratio (at 65dBf):	Mono; 80dB, Stereo; 71dB
(at 75dBf):	Stereo; 74dB
Distortion (at 65dBf): 100Hz:	0.1% (mono), 0.2% (stereo)
1kHz:	0.1% (mono), 0.15% (stereo)
6kHz:	0.1% (mono), 0.2% (stereo)
Frequency Response:	30 to 15,000Hz ±0.5dB
Capture Ratio:	1.0dB
Alternate Channel Selectivity:	80dB
Spurious Response Ratio:	100dB
Image Response Ratio:	90dB
IF Response Ratio:	100dB
AM Suppression Ratio:	55dB
Muting Threshold:	19.2dBf (5μV)
Stereo Separation:	50dB (1kHz), 35dB (30Hz to 15kHz)
Subcarrier Product Ratio:	65dB
SCA Rejection Ratio:	65dB
Antenna Input:	300 ohms balanced 75 ohms unbalanced
AM TUNER SECTION	
Sensitivity:	300μV/m (IHF, ferrite antenna) 15μV (IHF, ext. antenna)
Selectivity:	26dB
...Ratio:	50dB
	40dB

Radio Frequency

In the last issue, we learned about AF specifications. We hope you understand what the specifications mean and how to read and make use of them.

In this issue, we are going to learn about RF specifications.

While the AF section basically handles only audible frequencies, the RF section handles mainly radio frequencies, that is, 88–108MHz on FM, 525–1605kHz on AM (MW), in addition to AF.

Unlike amplifiers, tuners and receivers have many functions such as tuning, frequency conversion, detection, stereo demodulation in the case of FM stereo etc., as well as AF signal amplification.

Because of this and because the source of the tuner is the radio wave, there are more kinds of specifications in the tuner section.

Reception strongly depends on receiving conditions such as direction, distance and obstructions between the receiver and stations.

In a receiver, mutual interference or induction may occur between the units or blocks due to the characteristics of radio frequency signals. To compare the performance of tuners or receivers by specifications, they must be tested under equal conditions.

However, a standard test which is common throughout the world does not exist at present.

FM Section

Definition of Test Conditions

The terms mentioned in this chapter are defined below unless otherwise specified.

1. Standard Test Frequencies
   90MHz, 98MHz, 106MHz
2. Standard Modulation
   1kHz, 100% modulation (max. 75kHz deviation)
3. Stereo Modulation
   Unless otherwise specified, only L or R channel is given a modulating signal
4. Pilot Signal
   19kHz, 10% modulation (max. 7.5kHz deviation)

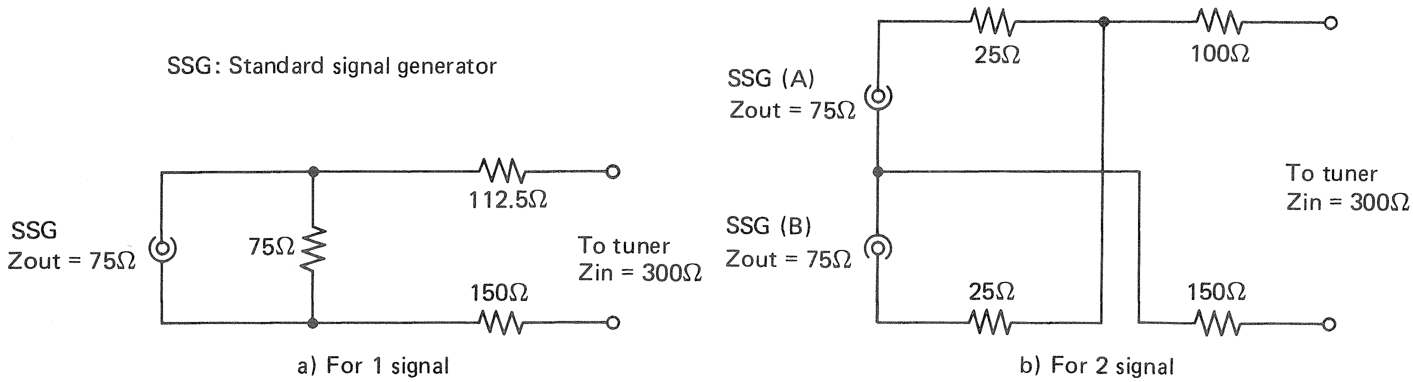


Fig. 1 FM dummy antenna

1. Usable Sensitivity

In general, sensitivity means an input level required to produce a stated output.

However, sensitivity in RF means an ability to receive weak radio waves as clearly as possible, and its figure does not necessarily correspond to gain.

usable sensitivity expresses an available power which provides AF output with –30dB (3.16%) distortion including noise.

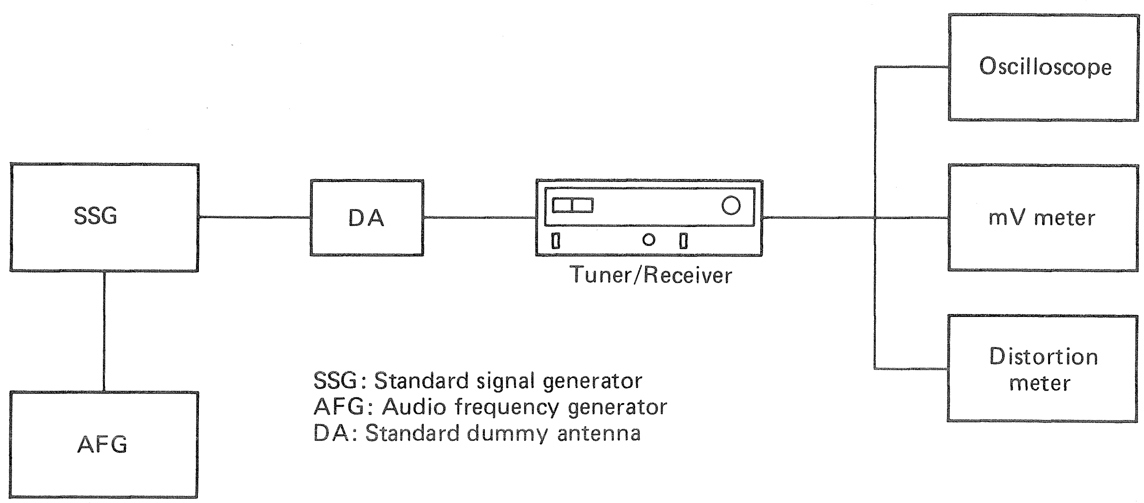


Fig. 2 Measuring usable sensitivity

Available power is expressed in dBf induced μV. 0dBf is the reference level at which a power of 1 femto watt (10^{-15} watt) is consumed at the antenna input. In the case where the antenna input impedance is 300 ohms, 0dBf corresponds to 0.55μV. (The reason why sensitivity indication was changed and how to make connections between dBf and μV values are described on page 24, *Tuning Fork* No. 1.) By using units of dBf, sensitivity of tuners or receivers can be compared directly irrespective of the antenna input impedance.

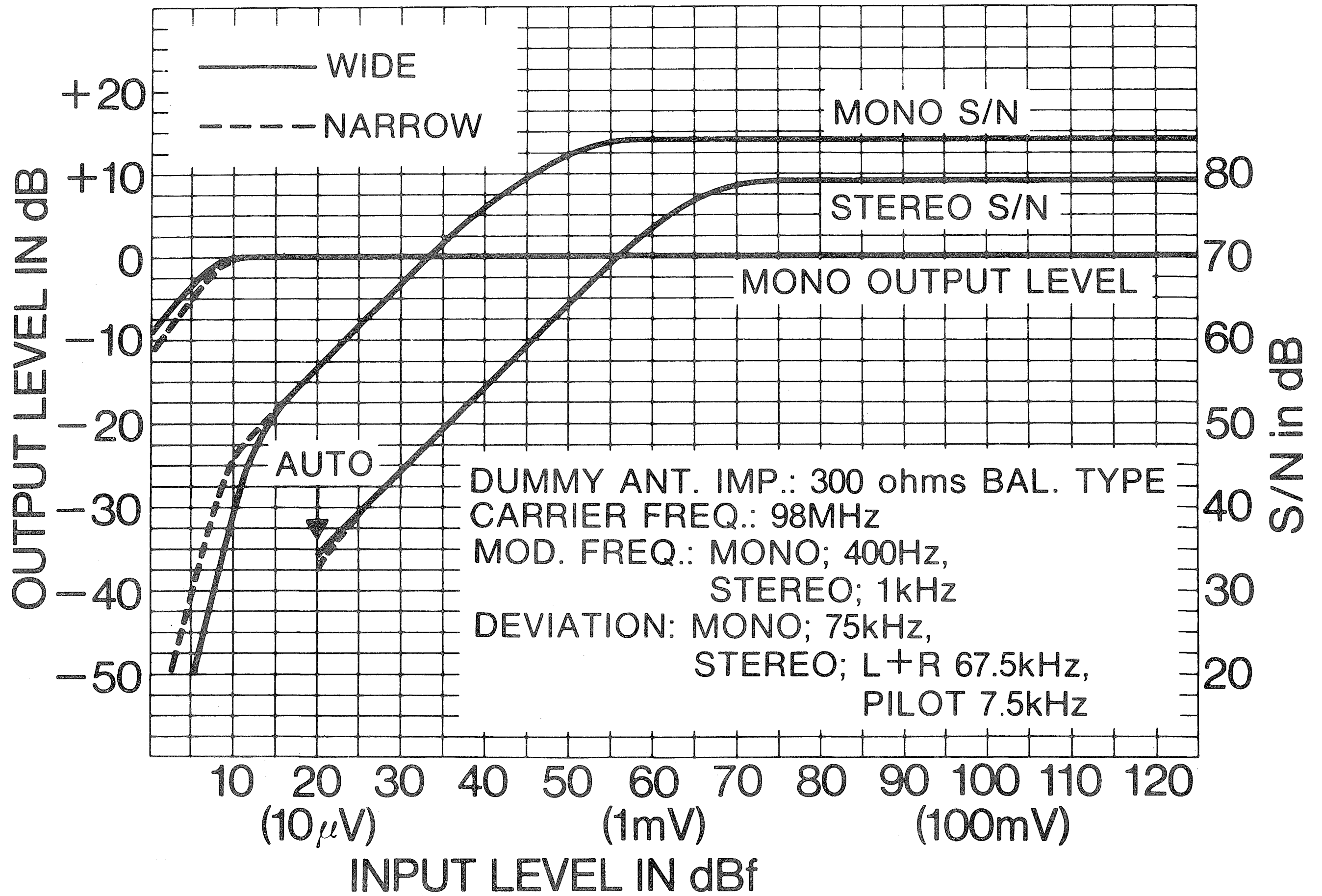


Fig. 3 Output level, S/N vs input level

2. 50dB Quieting Sensitivity

While usable sensitivity is an input power necessary to produce a specified output with −30dB distortion, 50dB quieting sensitivity is an input power necessary to produce an output signal of 50dB S/N ratio, which is more severe than usable sensitivity and is required for hi-fi listening. The measuring method is similar to that of usable sensitivity using a standard signal.

The measuring of signal level is made with standard modulation while the undesired signal, such as noise, is measured without modulation.
When the output difference between signal and noise becomes 50dB, its input signal level is expressed as available power.
It is specified for both stereo and monaural.

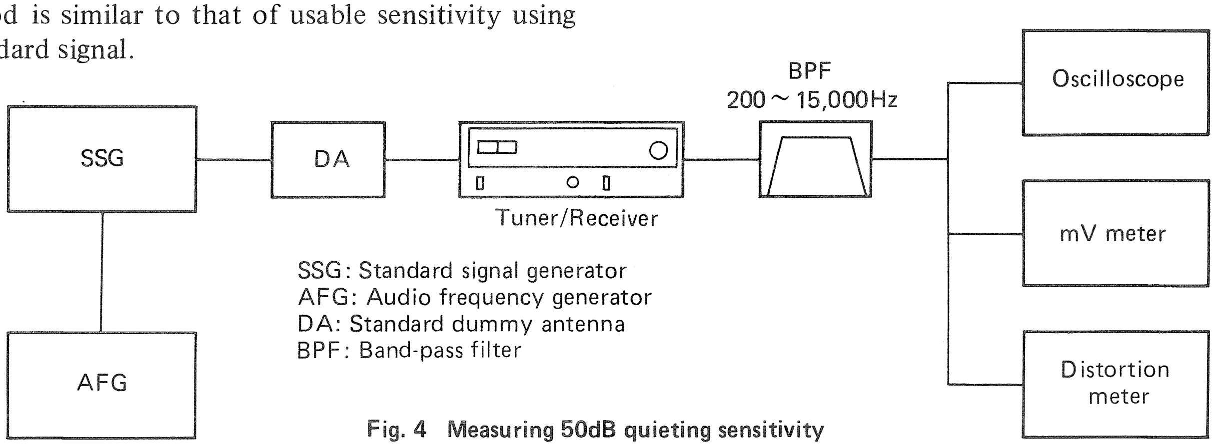


Fig. 4 Measuring 50dB quieting sensitivity

3. Signal-to-Noise Ratio at 65dBf and 75dBf

This is defined as the S/N ratio when an input signal of 65dBf or 75dBf is applied to the tuner.
This test is a measure of the S/N ratio attained with relatively strong input signals.
Measuring is performed as follows:

The outputs obtained with and without modulation are measured, and the resultant S/N ratio is expressed in dB for both stereo and monaural. The reason why the band-pass filter is inserted is to eliminate the hum and sub carrier component.

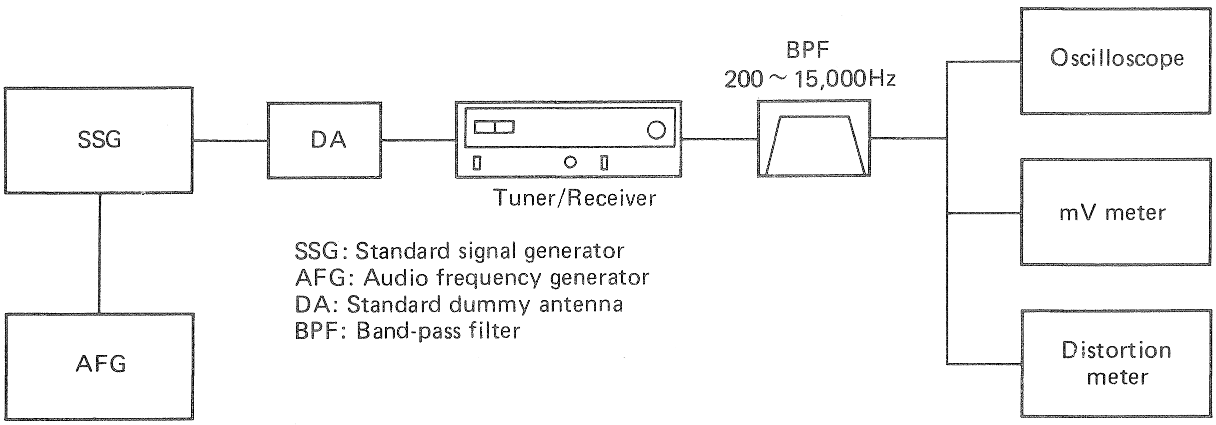


Fig. 5 Measuring S/N ratio at 65dBf

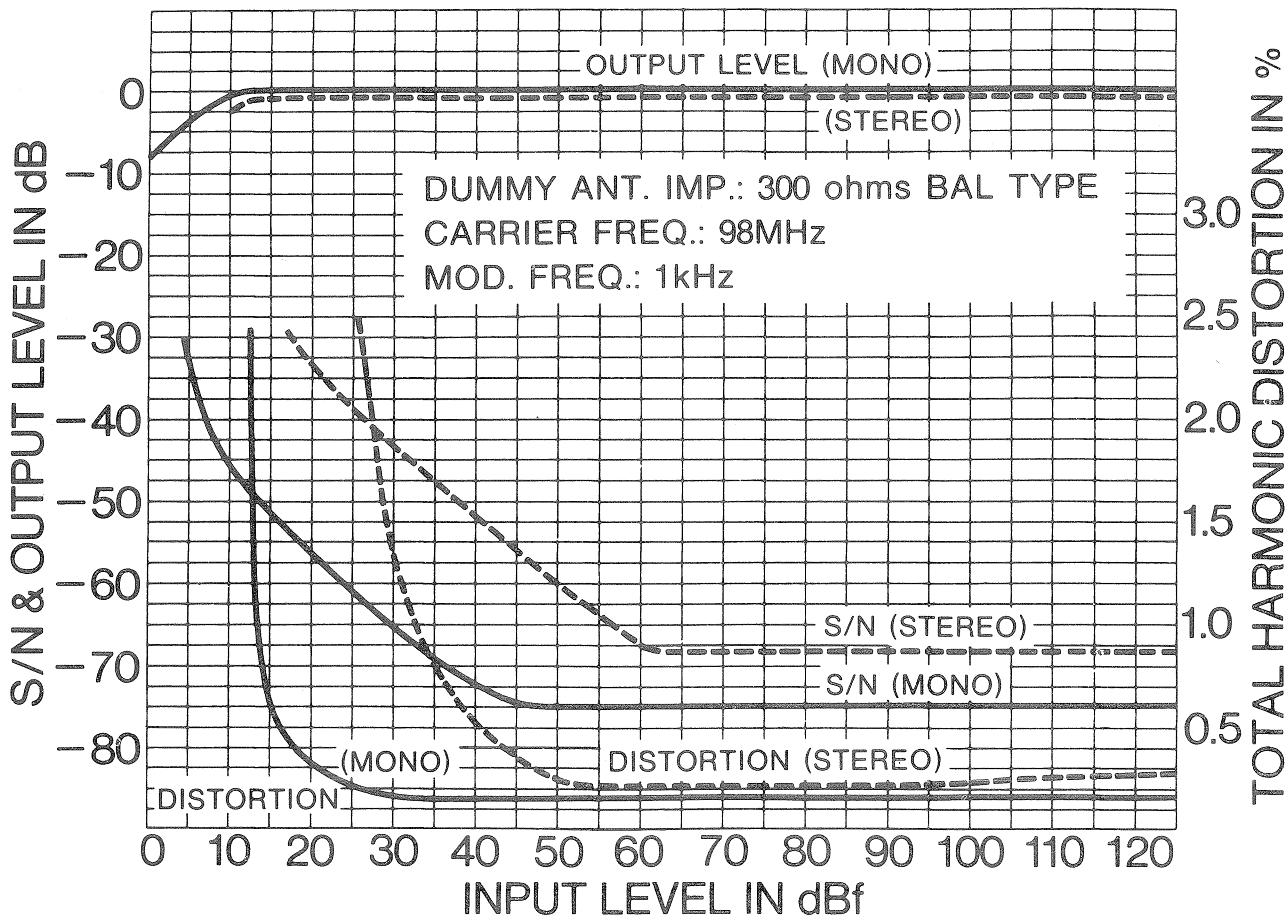


Fig. 6 Input vs output, S/N & distortion

4. Total Harmonic Distortion

This is to evaluate the linearity of the entire system, namely, from antenna input to AF output, of a receiver.

The definition of total harmonic distortion is exactly the same as that explained in AF Specifications in *Tuning Fork* No. 2.

When measuring the total harmonic distortion of a receiver, the conditions are specified as follows:

Input level	65dBf
Deviation	100% (±75kHz)
Modulation frequency	100Hz, 1kHz, 6kHz

Total harmonic distortion is expressed as a percentage for both stereo and monaural.

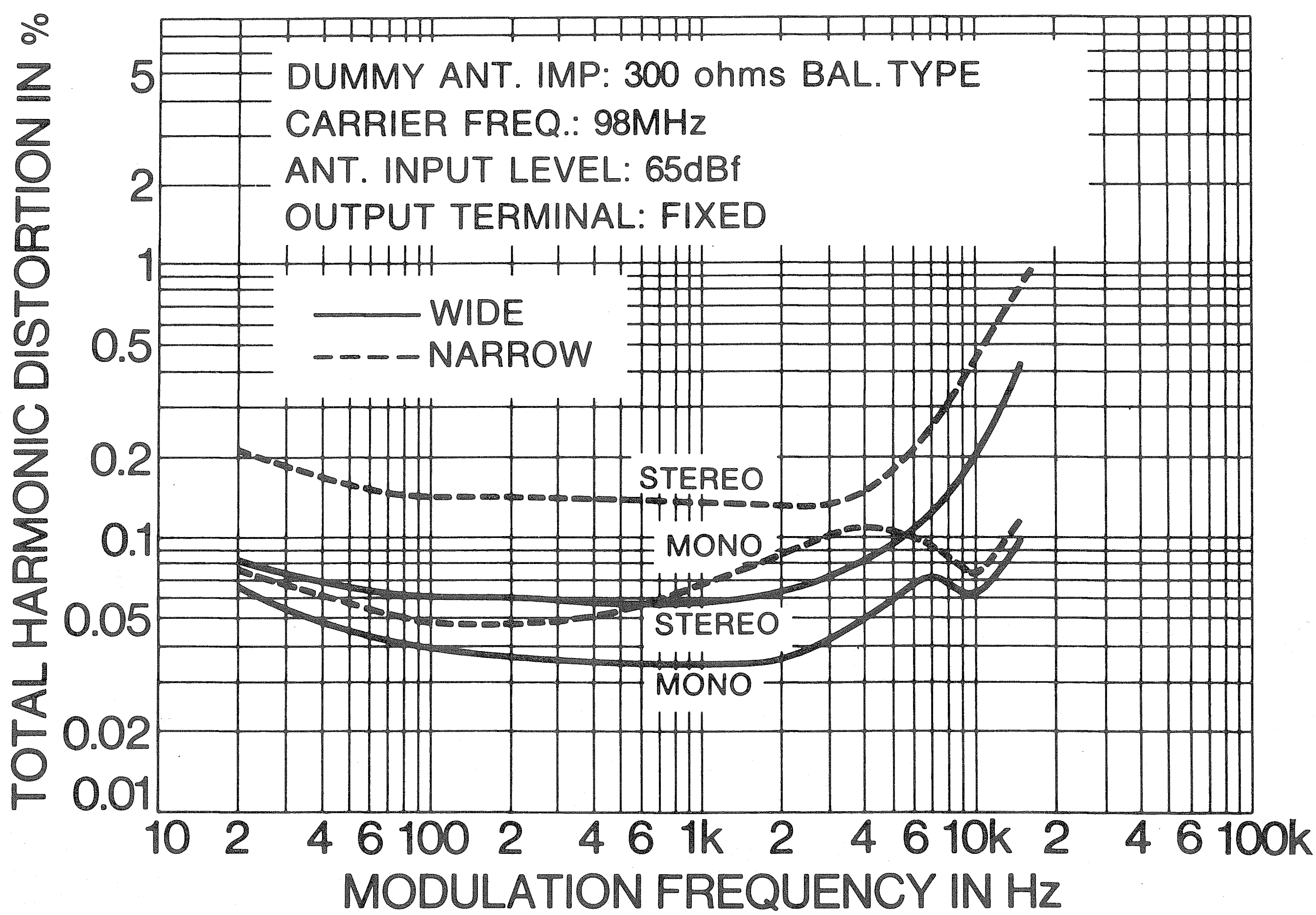


Fig. 7 Frequency vs total harmonic distortion

5. Frequency Response

The frequency response of a tuner is defined as the AF output amplitude response to modulating frequencies of standard amplitude in a specified range. The only difference from the Frequency Response used in AF is the frequency range. The AF output variation is observed while the modulation frequency is varied continuously from 30 to 15,000Hz. The rated frequency response shall be stated as: from 30Hz to 15,000Hz +XdB, −XdB.

6. Capture Ratio

In case two FM stations whose frequencies are the same exist in a given service area, only the station whose field strength is greater can be received when a receiver is tuned to that frequency.

This is the nature of FM broadcasts, and is called capture effect. Generally speaking, capture ratio is the index which shows the degree of capture effect, and it shows the minimum difference between the two antenna input signals necessary for the stronger signal to suppress the weaker one and to keep the S/N ratio at 30dB or more.

It is expressed in terms of dB by inducing a logarithm of the ratio between the two signals when an output with an S/N ratio of 30dB is produced. Therefore, the smaller the better.

As a matter of practice, two or more stations whose frequencies are the same will not be placed in the same service area.

Capture ratio, nevertheless, is an important factor when taking multipath reflections into consideration.

Multipath reflections severely affect reception if the tuner has an inferior capture ratio.

To measure capture ratio, two signal generators are required.

Both generators (A & B) are connected to the receiver through a dual dummy antenna and are set accurately to the same frequency (normally 98MHz) by zero beating or by countermeasurement.

The output of the unmodulated interfering signal generator B is initially set at zero output.

With a standard monophonic test modulation on the desired signal, generator A produces an input signal level of 65dBf and the controls are adjusted for standard test conditions. The output level of generator B is then increased until its interfering signal causes the receiver output to drop by 1dB; this output level of generator B is noted in dB. The output level of generator B is further increased until the receiver output falls by a total of 30dB, indicating that the interfering signal has substantially captured the channel; this output level of generator B is again noted in dB.

The difference in the two dB readings of generator B divided by two is defined as the capture ratio.

The smaller the spread between the output levels of generator B which causes the 1dB and 30dB drop in receiver output, the less vulnerable the receiver is to co-channel interference.

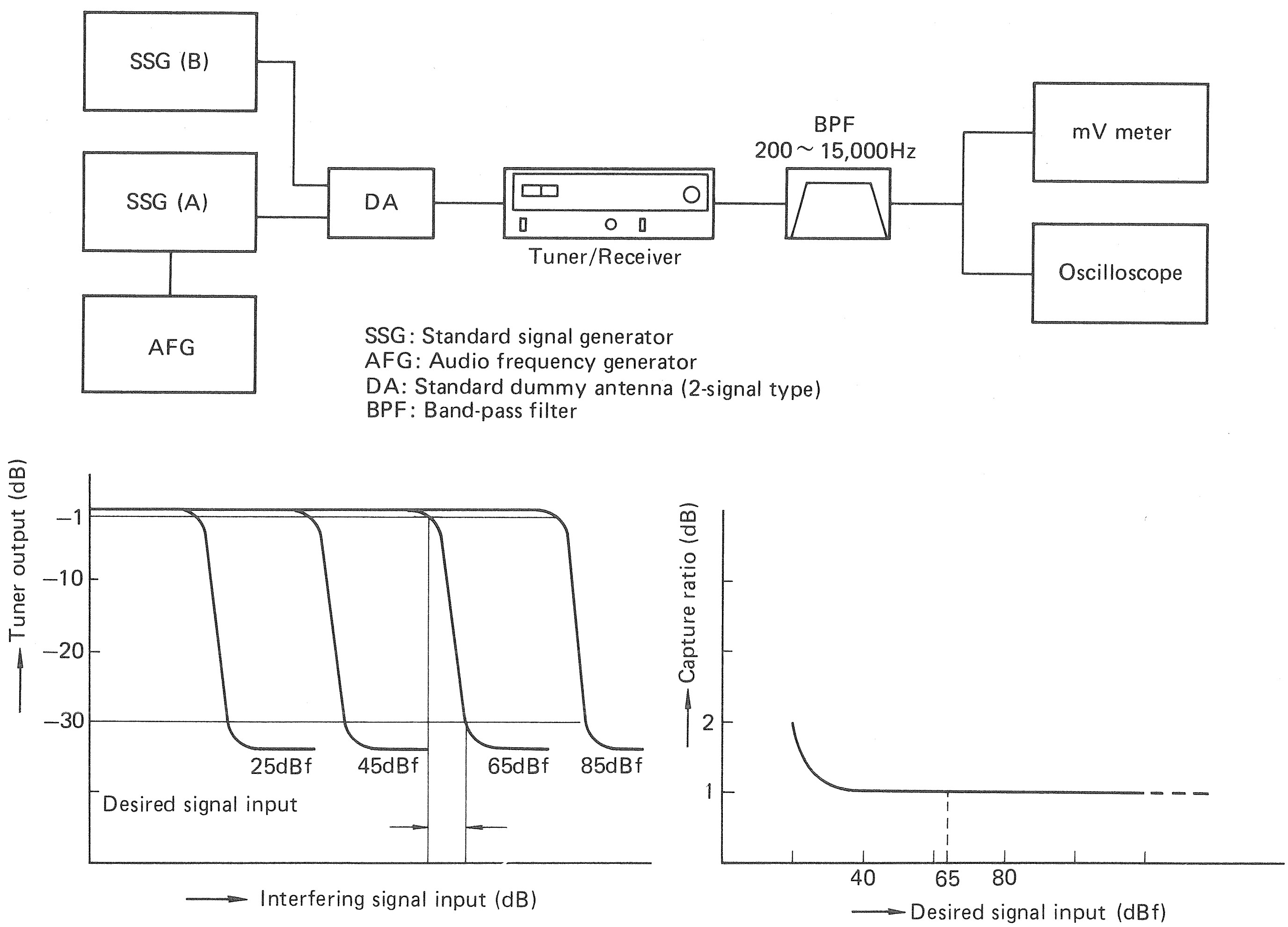


Fig. 8 Measuring capture ratio

7. Alternate Channel Selectivity

This represents a receiver's ability to receive a signal without interference in the presence of an interfering signal which is relatively close to the desired signal.

FM stations are placed at an interval of 200kHz (300kHz in Europe). In actual reception, the major interfering signals are broadcast carriers other than the desired one in the service area.

The measuring conditions are similar to those for the measurement of capture ratio, except that the interfering signal frequency B is different from the desired signal frequency A (separated by 400kHz). With generator B at zero output, standard monophonic modulation is applied to generator A and the input level is adjusted initially to 45dBf with all controls adjusted for standard test conditions. The modulation is removed from generator A, applied to generator B, and the output of generator B is increased until the interference produced is 30dB below the standard test output.

The ratio of the generator outputs is noted in dB.

The procedure is repeated with generator B offset a like amount on the opposite side of the desired signal.

The average of the two ratios in dB is the alternate channel selectivity. The ratio corresponding to a separation of two channels (400kHz) is called the alternate channel selectivity ratio.

For reference, the ratio corresponding to a separation of one channel (200kHz) is called the adjacent channel selectivity ratio.

In this measurement, a 1kHz band-pass filter is used at the output to reduce errors from noise.

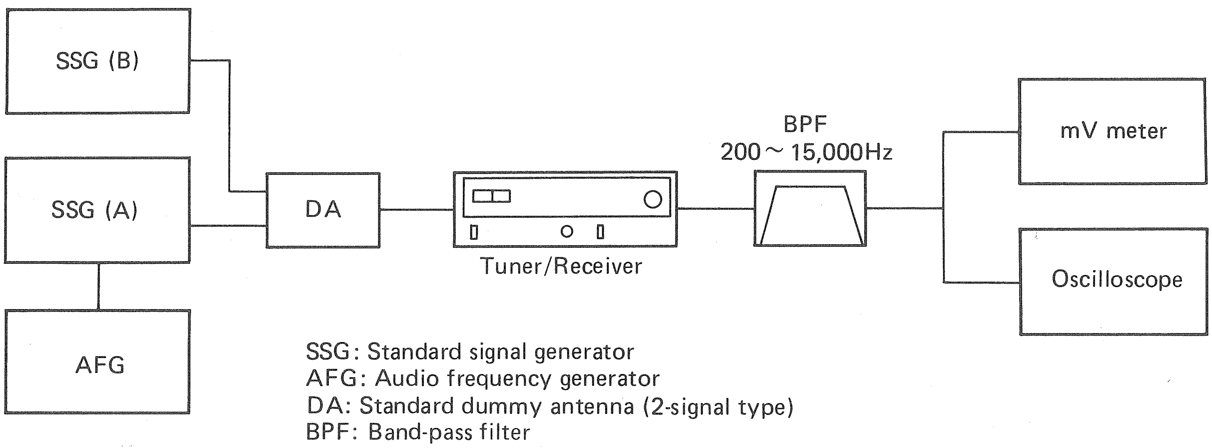


Fig. 9 Measuring alternate channel selectivity

8. Spurious Response

While capture ratio and alternate channel selectivity represent the resistance of a desired signal to interfering signals, the spurious response ratio represents the resistance of a desired signal to spurious signals other than broadcast transmissions.

Tuners employing the most popular superheterodyne system have a local oscillator to get an intermediate beat frequency of 10.7MHz by oscillating a signal 10.7MHz above or below the desired signal and mixing it with the received signal.

The relationship between their frequencies is as follows:

$f_s \pm f_0 = f_i$

However, f_i can be obtained by other frequencies as well, under the conditions shown below:

$n \times f_s \pm m \times f_0 = f_i$

m & n: Arbitrary positive integral number

f_s : Receiving signal (desired signal)

f_0 : Local-oscillator frequency

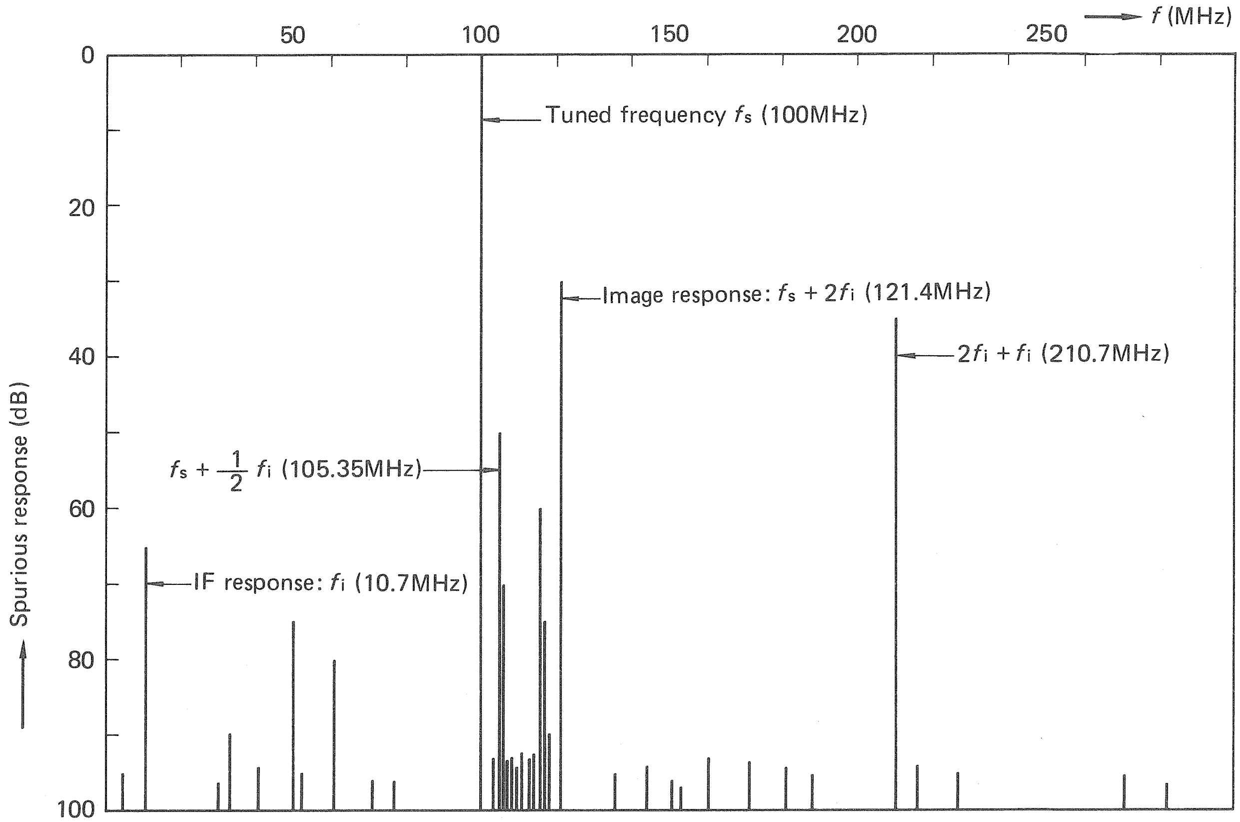
f_i : Intermediate frequency (10.7MHz)

This means that harmonics of the desired signal and the local oscillating signal may also cause a generation of f_i, and result in spurious response. A tuner is most vulnerable to interferences at the frequencies of spurious responses. In addition, any frequency received which causes a generation of f_i and its harmonics ($2f_i$, $3f_i$) can also interfere with the reception.

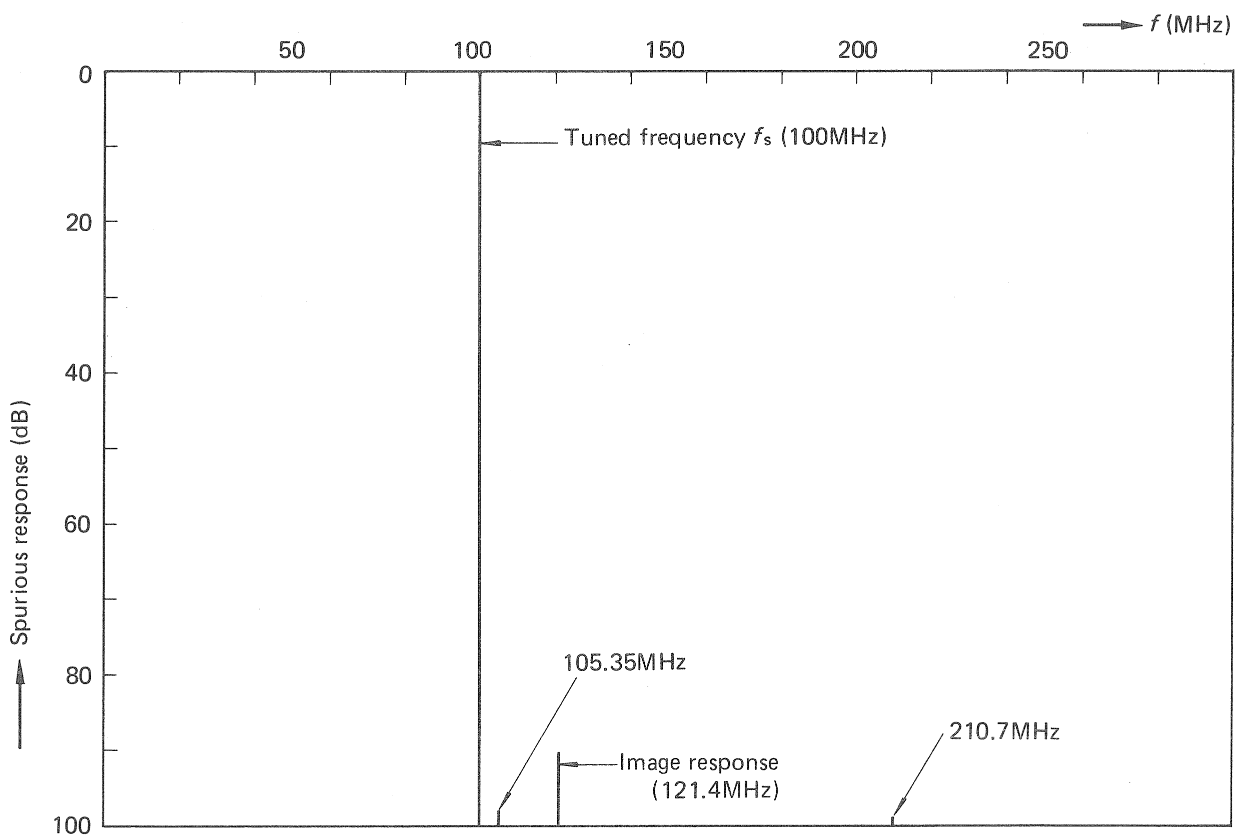
There are three kinds of spurious interferences.

1. Image Interference — $f_s \pm 2f_i$
2. IF Interference — f_i
3. So called Spurious Response — All other frequencies

Poorly designed oscillators can generate harmonics and RF amplifiers and mixers may also produce harmonics with non-linear characteristics, especially when receiving strong broadcast signals. Please see the difference shown in Fig. 10. (0dB refers to the response at the tuned frequency, 100MHz.)



a) Inferior designed



b) Well-designed tuners

Fig. 10 Spurious responses

Measuring method: The method of measuring spurious response ratio, image response ratio and IF response ratio is the same. Keeping the tuner tuned to the desired frequency, a test signal of the standard modulation is fed to the tuner and the output level is measured. Then, the test frequency is swept over a wide range to see whether the tuner responds to the input signal local (SG's output) is increased to the input where the tuner's output level becomes equal to the level of the desired signal (tuned signal) and the output level of the SG is recorded. The ratio of the level difference is expressed in dB.

Image response ratio refers to the difference in level between the desired signal and the image response.

IF response ratio refers to the difference in level between the desired signal and IF response.

Spurious response ratio refers to the highest response at other frequencies except the above two.

9. AM Suppression Ratio

Although it is ideal for FM tuners to demodulate FM signals only, FM demodulators also detect AM components in the signals a little. The AM may result from fading, multipath effects, airplane flutter, or a relatively narrow or misaligned receiver passband.

AM suppression ratio is an index of a tuner's ability to suppress response to AM components in the signal, and is defined in terms of the relative disturbance caused by AM components when the carrier is simultaneously amplitude and frequency modulated. That is:

AM suppression ratio =

$$20 \log \frac{\text{FM-caused output (100\% modulation, 1kHz)}}{\text{AM-caused output (30\% modulation, 400Hz)}} \text{(dB)}$$

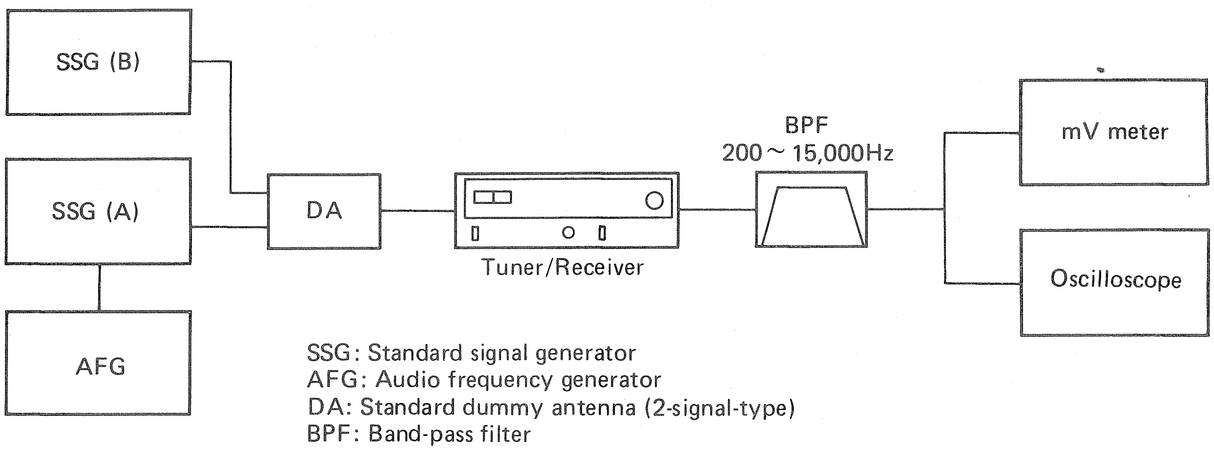


Fig. 11 Measuring spurious response ratio, image response ratio and IF response ratio

Measuring method: Using an SSG capable of FM and AM modulation, an FM modulated signal of 100%, 1kHz is applied to the tuner, and the output level is recorded.

Then FM modulation is removed, and an AM modulated signal of 400Hz, 30% is applied to the tuner. The output level is recorded again.

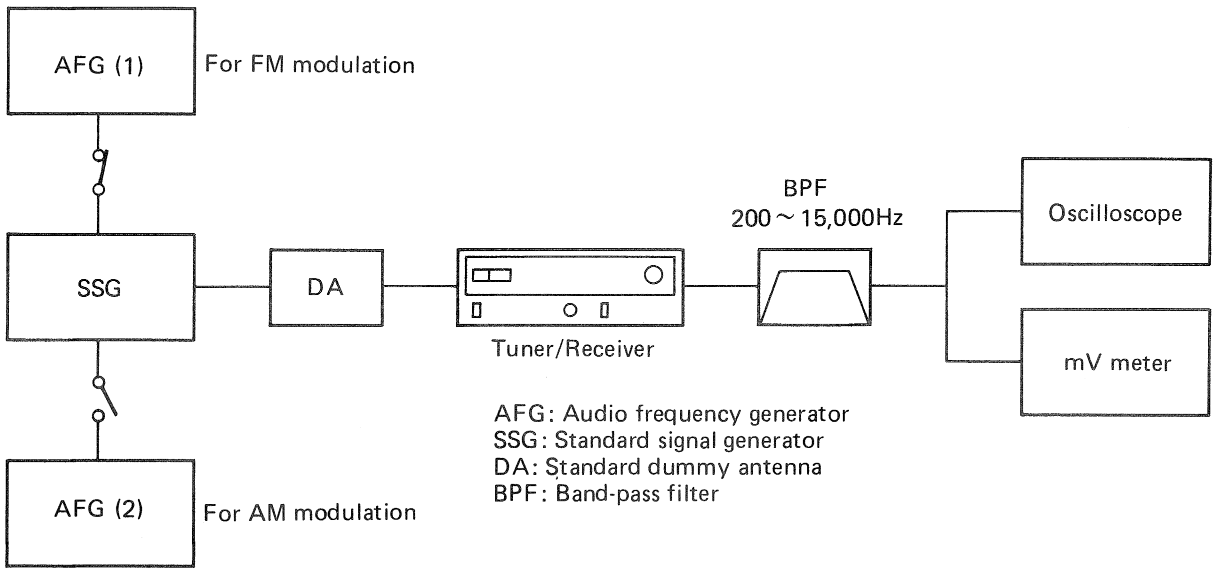


Fig. 12 Measuring AM suppression ratio

During this measurement, the volume control or output level control setting must not be changed.
The AM suppression ratio is given by the above formula.

10. Muting Threshold

While muting in an audio frequency block can be done only by reducing gain, FM muting is difficult when receiving low-level signals because of inter-carrier or interstation noise inherent to FM. In order to overcome this disadvantage, FM tuners have a device to cut FM output when the input signal is below a certain level. The operation is called FM muting. Measuring method: A standard signal is fed to the tuner from the SG. The signal level from the SG is decreased from a high level down to the point where the muting circuit activates and the level is recorded. Then, the level is increased from 0 up to the point where muting is released and the level is recorded.

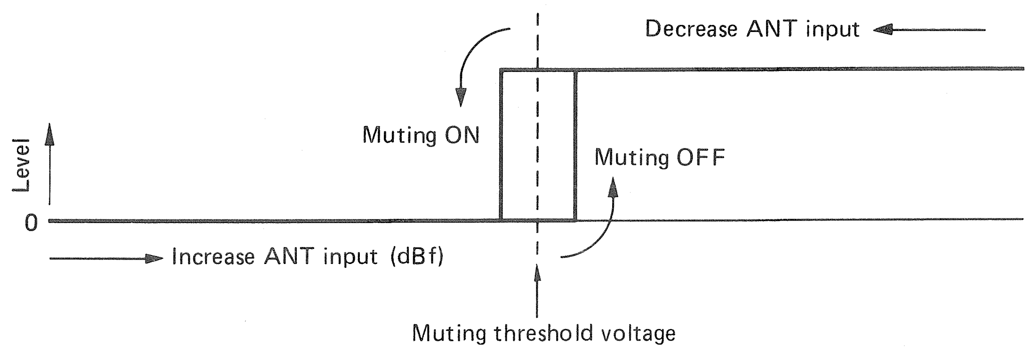


Fig. 13 Hysterisis loop of muting

You will find that the releasing level is higher than the activating one. The operation is called muting hysteresis. The mean value of the two power levels is called muting threshold and is expressed in dBf. Specification sheets for tuners equipped with a WIDE/NARROW switch show the mean value for both switch positions. Although the specification of the muting itself does not show performance, poorly designed tuners may have problems with a popping noise and unstable operation when the MUTING switch is pressed on and off.

11. Stereo Separation

With the FM stereo system, two audio signals, R & L, are carried on an RF signal. When receiving the RF signal, the two signals are extracted. Stereo separation is the degree to which two stereo signals are kept apart.

Measuring method: A standard stereo signal of 65dBf, 1kHz standard stereo modulation, is applied to only one channel of the tuner and the output levels of both channels are recorded. The separation is the level difference between the two which is expressed in dB. The range separation can be measured by varying the modulating frequency from 30Hz to 15kHz. The rated separation is measured at 1kHz and described in dB.

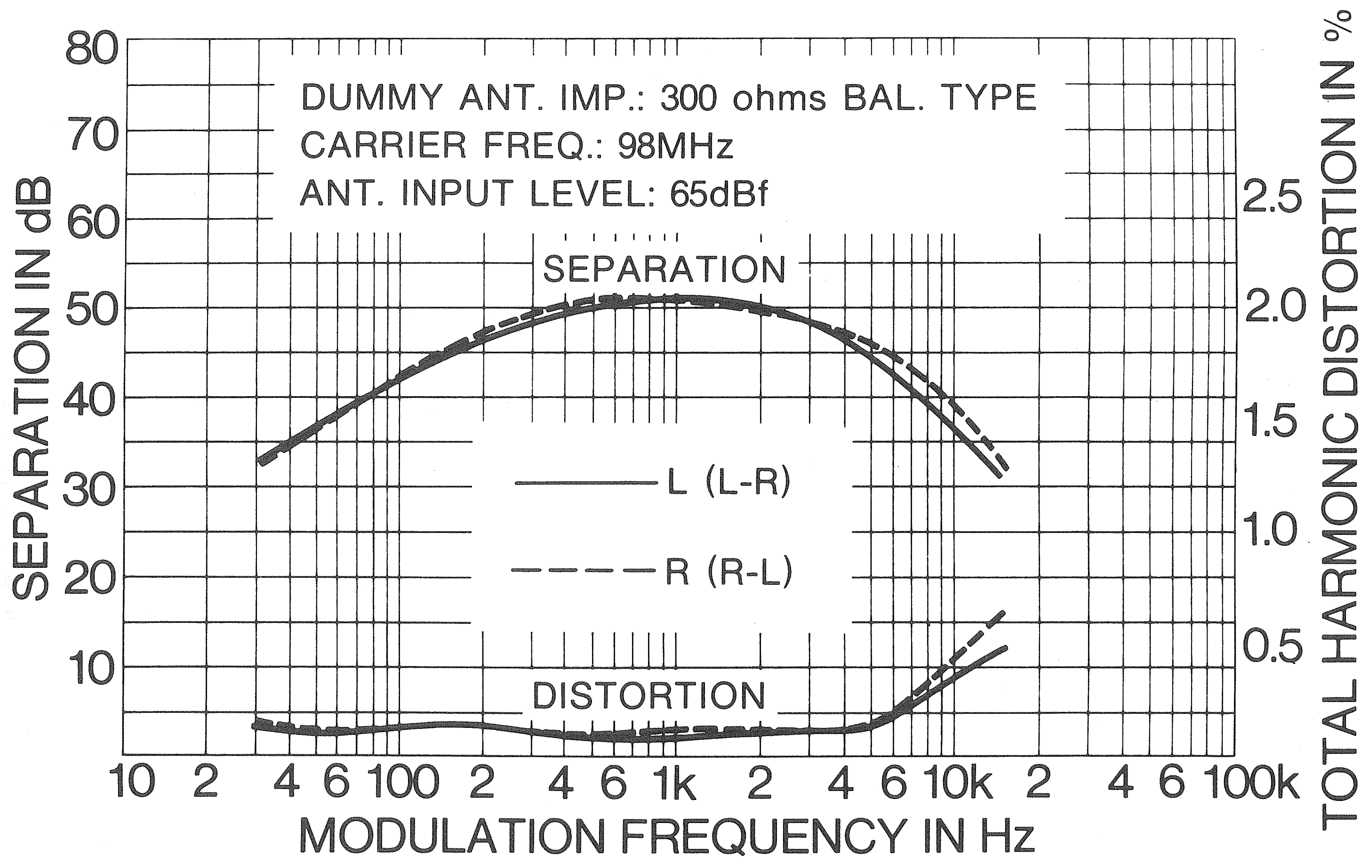


Fig. 14 FM stereo separation and distortion

12. Subcarrier Product Ratio

The subcarrier product ratio is defined as the content ratio of 38kHz found in stereo demodulation signals. The measuring method is that with the signal generator connected as usual and turned to the mean carrier frequency, a standard stereophonic test modulation is applied at 100% total system modulation. Output level is recorded with the tuner input level of 65dBf.

Next, all modulation except the 19kHz pilot signal is removed and the output level is once more recorded. The rated subcarrier product ratio is the difference between the two output readings taken above, expressed in dB. No filters are used for this measurement.

13. SCA Beat Response Ratio

SCA stands for Subsidiary Communication Authorization. On the second subchannel at 67kHz, a program other than FM stereo is being broadcast in some areas. The ratio specifies the degree of SCA's interference expressed in dB.

14. Antenna Impedance

All current FM receivers and tuners have two pairs of antenna terminals as shown below.

AM Section

AM Standard Signal

1. Standard input level
   74dBf with dummy antenna
   74dBf/m with loop stick antenna
2. Standard test modulation is an amplitude modulation at 400Hz with 30%.
3. Standard output
   The output signal obtained when the standard test signal is applied to the receiver. Fig. 15 shows the diagram for the measurement.

* Sensitivity

In general, it is defined as the minimum input signal level required to produce a specified output. However, sensitivity in RF specifies an ability to receive weak radio waves as clearly as possible, and its figure does not necessarily correspond to gain. When the gain is high and noise is low, the sensitivity becomes truly high.

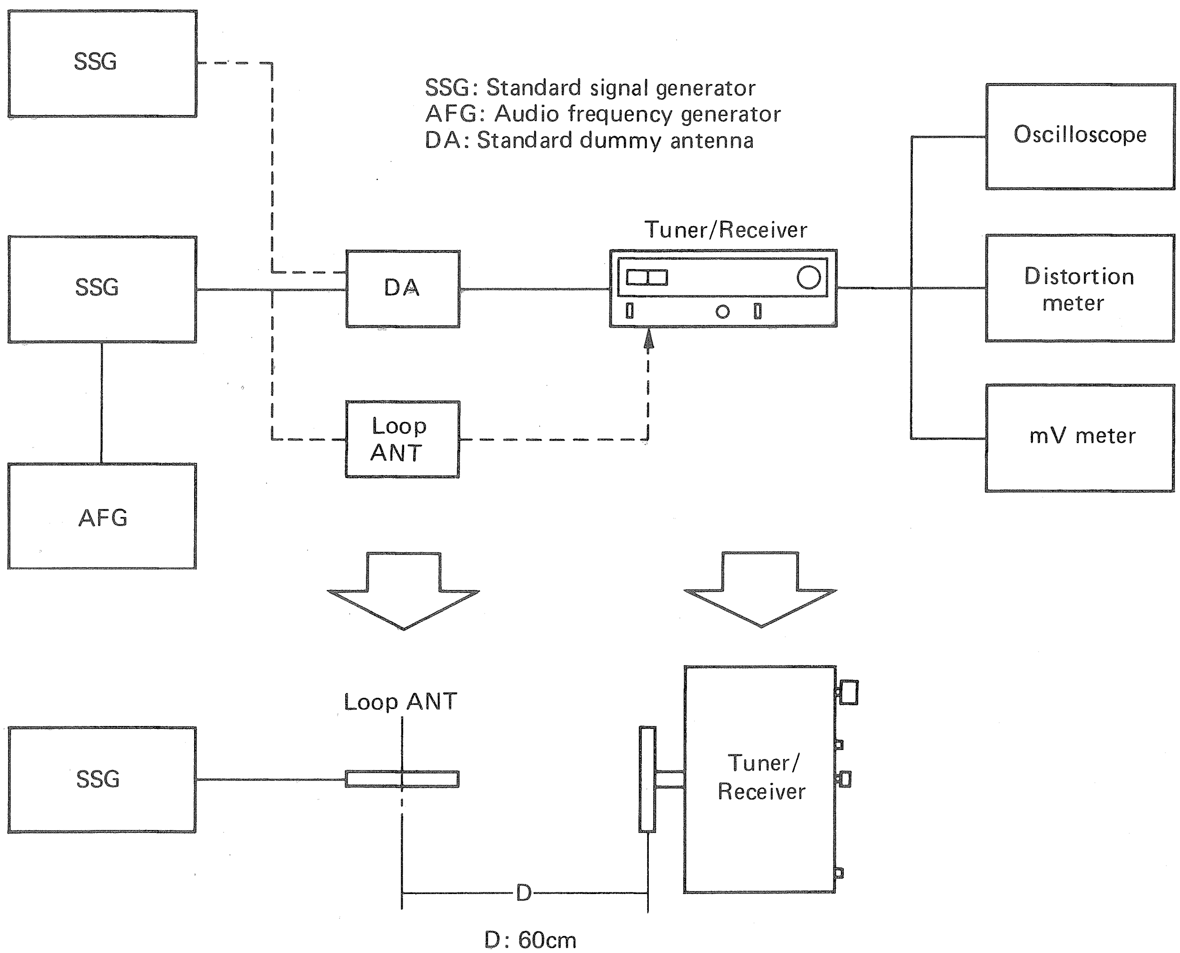


Fig. 15 Measuring AM specifications

1. Sensitivity

Although basically the same as that described in the FM section, the only difference is the minimum input signal necessary to make an output with a distortion rate of −20dB (10%), calculated by the following equation:

$$\frac{N + D}{S + N + D} = 0.1$$

N: noise components
D: other distortions
S: the desired signal

Tuners produced by Pioneer and other makers have a ferrite antenna for AM. Therefore for both sensitivities, one for antenna terminal and the other for ferrite antenna are specified.

2. Selectivity

The same as explained in FM excepting the measuring method.
Measuring method: The 1MHz standard test signal is applied to the AM antenna terminals from an SG, the tuner is tuned to 1MHz, and the input level is varied and the maximum sensitivity* is recorded. Then, the input frequency is deviated by 10kHz above or below 1MHz. Varying the input level, the maximum sensitivity is measured without turning the tuning knob. The difference in dB between the two input levels is the selectivity.

* Maximum sensitivity: The smallest signal input that produces a specified output without concerning S/N ratio.

3. Signal-to-Noise Ratio

The ratio of the magnitude of the signal to that of the noise expressed in dB.
Measuring method: The standard test signal (0% modulation) is applied to the tuner and the output levels of signal and noise are measured respectively.

4. Image Response Ratio

The definition and measuring method are the same as FM.

5. IF Response Ratio

The same as for FM except that the IF frequency is 455kHz. Measurement will be taken as follows: a test signal with standard modulation is applied to the receiver; the level of the test frequency and IF frequency are recorded when the maximum sensitivity is obtained; the level difference in dB is the IF response ratio.

6. Antenna

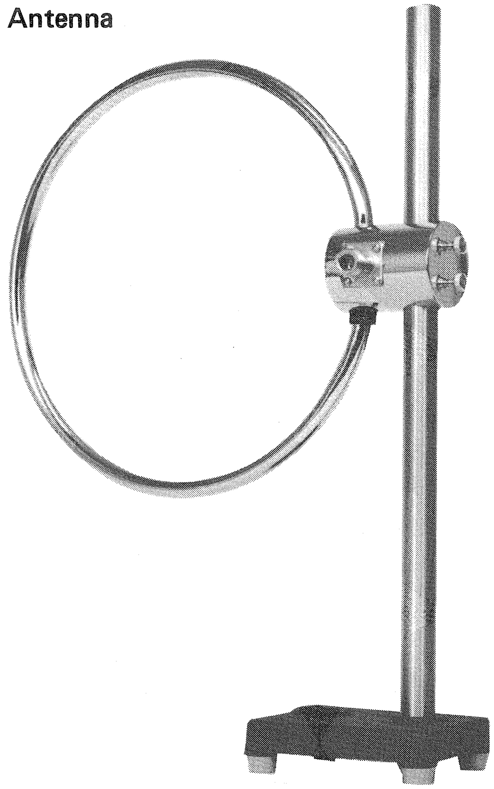
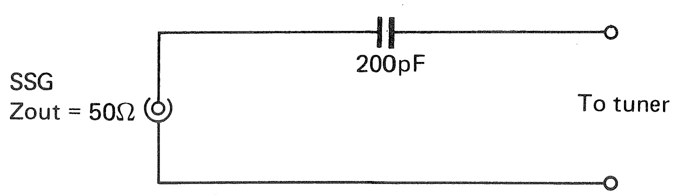


Photo 1 Standard loop antenna



SSG: Standard signal generator

Fig. 16 AM dummy antenna

Quality Information System (2)

In the last edition, Vol. 2, we had a look at the basic idea of quality and quality control. Here we are going to see the practical side of quality control which makes our service activity effective.

Outline of Quality Control

With rapid innovations, the improvement in audio equipment has been remarkable. Some top-grade equipment is even competing with measuring equipment in performance. But we cannot say that the higher the performance, the higher the quality. The most important factor in evaluating a product is its reliability, or freedom from defects. The purpose of quality control is to put reliable products on the market. The basic method of quality control requires two things: statistics as explained in the last issue, and the worker's will to aim for zero defects.

There are three elements of total quality:

Designed quality	Quality determined at the time of design: performance, durability, etc.
Manufacturing quality	Quality determined during manufacture: dispersion, workmanship, etc.
Field quality	Quality perceived by the market: consumers' trust and satisfaction.

Of these three, field quality is our greatest concern. Servicemen and salesmen as well as customers are happy if commodities are high in field quality and free from trouble, regardless of design quality and manufacturing quality. But there is a close relation between these three and they should not be considered independently. As field quality depends on the other two, to improve field quality it is necessary to maintain high design and manufacturing standards.
Refer to Fig. 1.

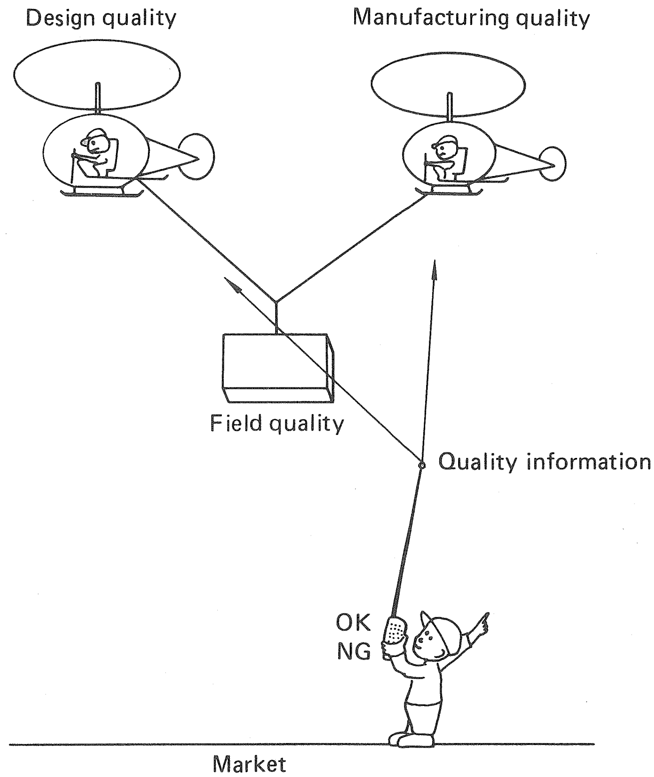


Fig. 1 Design quality, manufacturing quality and field quality

To improve overall quality, special care is being taken in the manufacturing process. Incoming inspection is being conducted on every part in addition to outgoing inspection. The inspection data is accumulated, analyzed, recorded and stored. If a problem is found in the data by analysis, it is fed back to the sections concerned for remedial action.

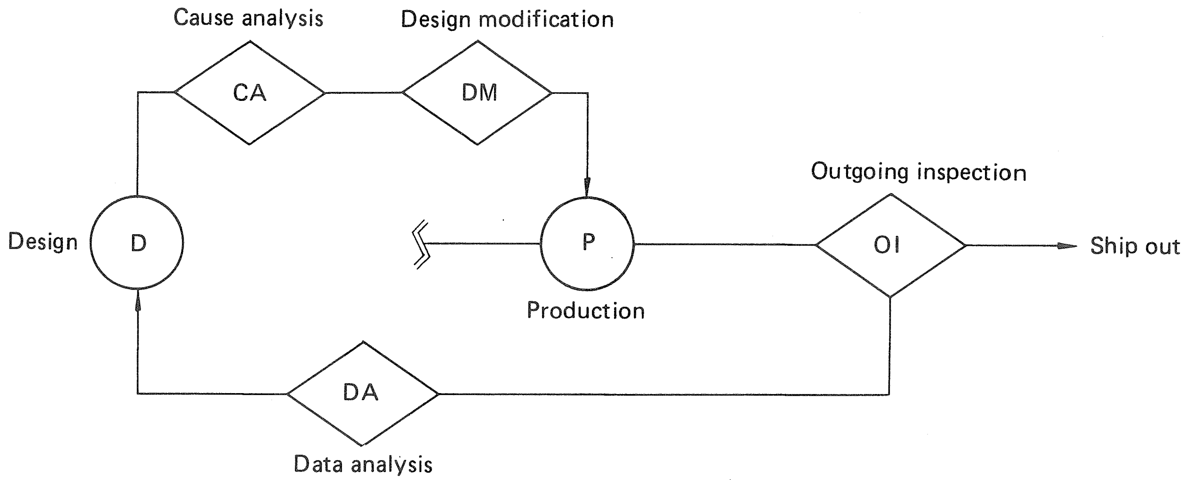


Fig. 2 Quality-information feedback and quality improvement

Fig. 2 shows an example of the process.

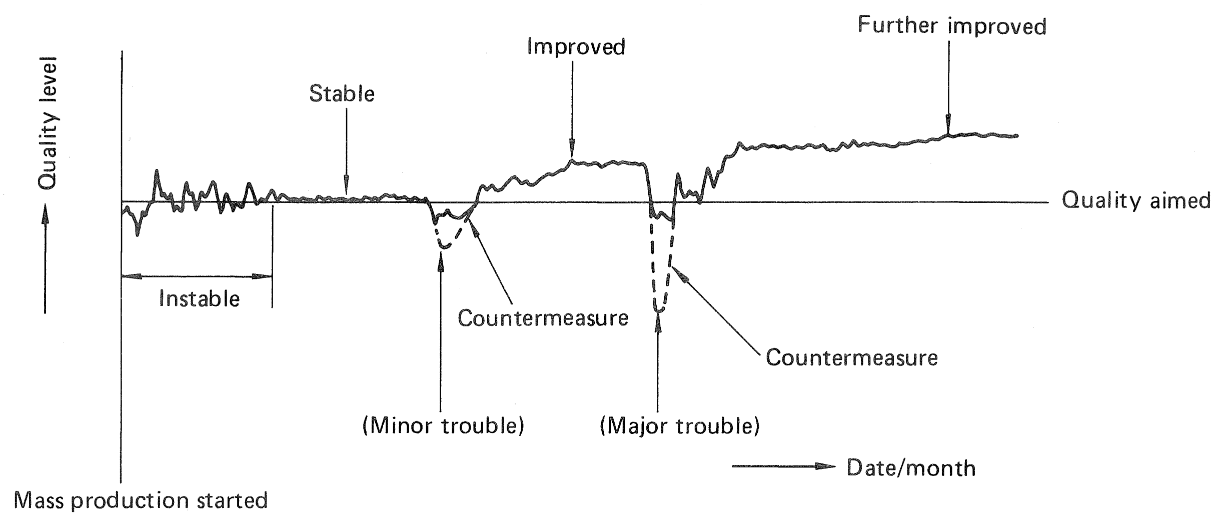


Fig. 3 Typical transition of quality level

This shows only the relation between the quality and time by day or month. Let us see how practical improvement is being made.

Practical Quality Control

Quality control is, in other words, the total effort necessary to assure outgoing quality from a customer's point of view. Needless to say, outgoing quality assurance cannot be conducted in one particular section. The people designing, manufacturing or inspecting a product are all responsible for quality. The main job of the Quality Control Section is to confirm the quality of products from the users' viewpoint by test, inspection, production evaluation, data analysis, and to give approval when the results satisfy the quality standard. This is called "outgoing quality assurance."

The second job is to prevent quality problems and their repetition, and to remedy them quickly.

With the above as a guide, we are conducting outgoing quality assurance and quality improvement operations. Outgoing quality assurance and quality improvement should not be considered separately. Outgoing quality assurance requires quality improvement and quality improvement needs the data generated by quality inspection.

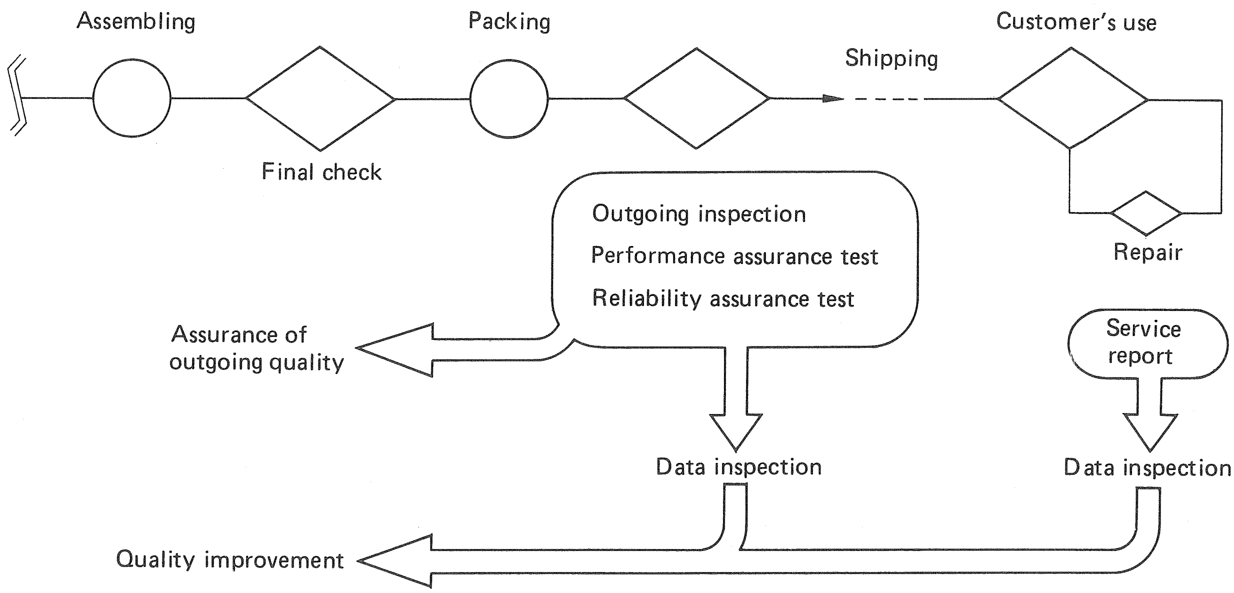
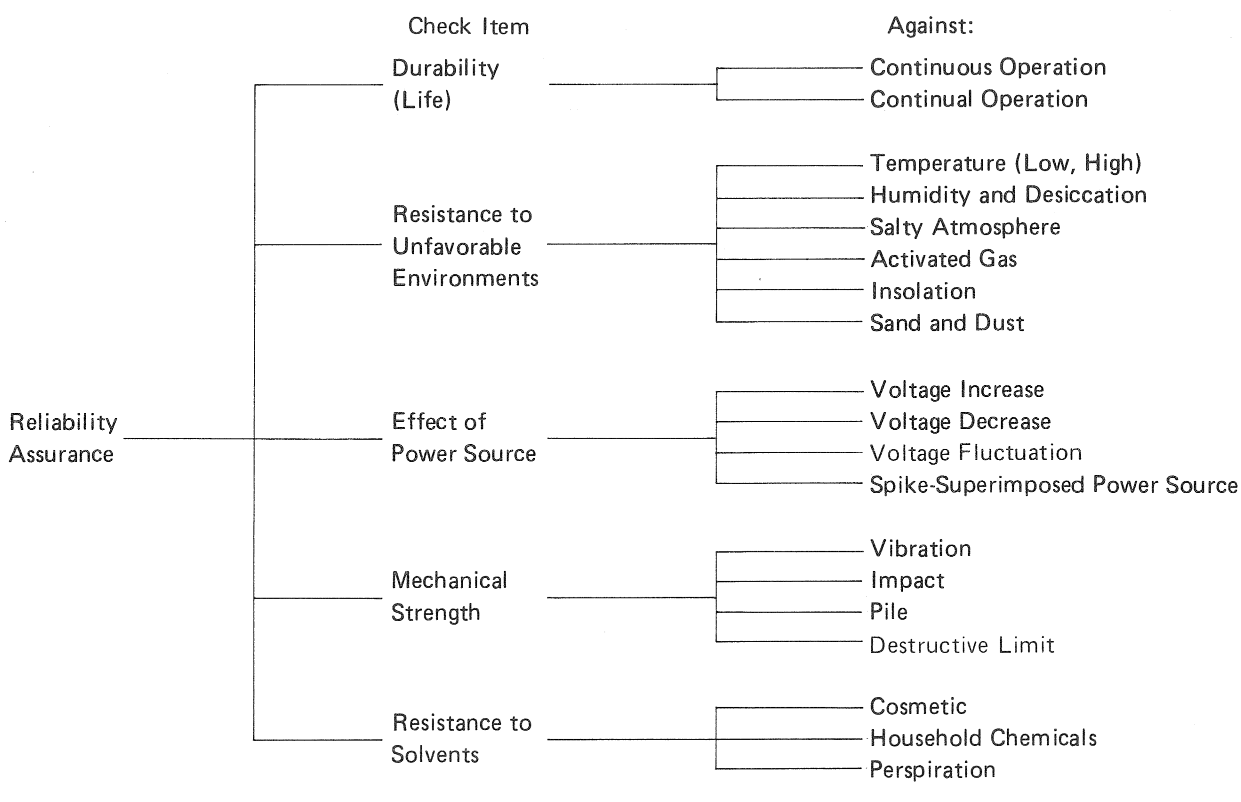


Fig. 4 Outgoing quality assurance and quality improvement

Checking is done on every unit at several checkpoints on the assembly lines and after completion, a final check is made. And before shipping out, outgoing inspection by sampling is done to a certain standard. A sampling check for operation, appearance and safety is conducted from the customers' point of view. We call this the "CA test" or "consumers' acceptance test." Judgement whether to pass a product lot or not is made using the results of the sampling inspection. Then the data is analyzed for quality improvement. While outgoing inspection is made on the basis of consumer acceptability, a performance-assurance test is made on every specification such as sensitivity, distortion, S/N frequency response, wow & flutter, etc. The performance is watched and quality is assured by continuous measurement. The first stage of product quality assurance is done with the results of the CA sampling test and the performance-assurance test. Reliability is another important factor in quality. The playing condition of audio equipment varies very much according to climate and care. A unit may be



played in hot, cold, dry, humid, or dusty places. It may be used in a hot spring resort, in a volcanic area where hazardous gas such as hydrogen sulfide, H_2SO_3, exists, on a salty sea coast, or in a dusty desert. Transportation conditions should also be considered. Humidity and temperature in a boat may sometimes rise very high. Vibration and shock may be intense on trains and trucks. Shock is common during loading and unloading. Piling up of products in warehouses puts high pressure on the bottom units. The product may be used intermittently or continuously, it may be left unused for a long time which will also cause trouble. To improve it in reliability, all of these conditions are considered. Reliability tests under hard conditions are being made as follows:

As explained before, quality should be considered not only at the shipping stage from the factory, but also after use by customers.

In other words, quality characteristics are important not only in the early stages but also with the passing of time.

This quality is called reliability.

Therefore, outgoing quality assurance must be considered quality control in a short-term sense together with reliability control. (When we say "Quality control" at Pioneer, it means both quality control and reliability control.)

In a narrow sense, quality control is performed by the QC division in the factory, but in a wider sense, it is related to all divisions: merchandise-planning, designing, manufacturing, inspection, sales, service, and then again back to planning and designing.

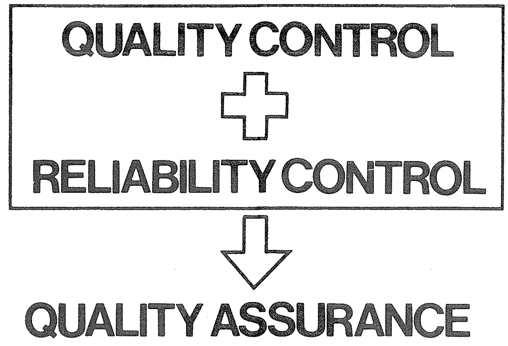


Fig. 5 What's quality assurance?

Quality control continues even after the sale.

This is especially important when troubles are concentrated in a particular product. When a product is used in various countries, not only do the environmental conditions differ, the people differ as well, so the assessment is also varied.

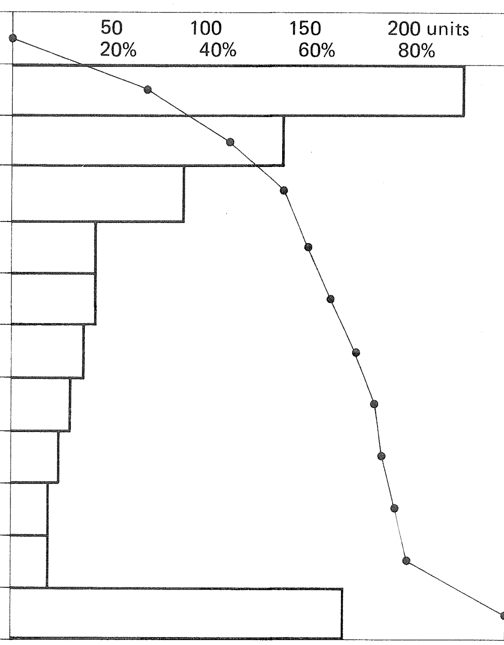
In fact, even if a product is not troublesome in place A or B, it can be troublesome in place C. The importance of quality information from the market is not limited to troubles that are concentrated in a particular product.

Efforts to improve all products are constantly required.

Please see Fig. 6.

This graph shows all the malfunctions reported by a certain country classifying them by symptoms.

(This is called the Pareto chart in QI terms)



PHENOMENON	Q'ty of defect	Defective rate	Accumulation
Sensing SW faulty	231	27.4%	27.4%
Head leads came off	140	16.6	44.0
Electrical adj faulty	92	10.9	54.9
Motor faulty	44	5.2	60.1
FF & REW not working	44	5.2	65.3
Dolby SW spring broken	39	4.6	69.9
Mechanical noise at PLAY	28	3.3	73.2
Wow	21	2.5	75.7
Fuse blown	17	2.0	77.7
P/R Head faulty	16	1.9	79.6
Others	170	20.4	100

Fig. 6 Pareto chart

The aim of quality improvement is to reduce malfunctions (occurrence of troubles). For that purpose, we have to know what to begin with. This chart tells us immediately what to do first.
For instance, failure can be reduced by 44% by removing defects in the sensing switch and head leads. And almost 55% of the failures can be reduced by removing the third cause, faulty adjustment.

Why is Quality Information Required From the Market?

By reading this far, you may already know the answer.
Quality control is based on actual quality information. That's the importance of quality information from the market. The basic thought behind quality assurance is to guarantee the function of a product.
Quality control is performed exactly for this reason. And the reason why "reliability," a single quality characteristic, is particularly stressed in this article is that it is most related to the market, and is the biggest concern for customers and us.
When considering the actual measures taken for quality assurance, the first step concerns the activities in the factory while the second step begins with collecting and analyzing data on product quality in the market.
All efforts to improve quality have to be verified as to whether they are valid or not.
In other words, it is requisite to understand the quality of the current models on the market, and to take action to solve a quality problem if it occurs. There are two things to do: repair defective products, and prevent recurrence of the trouble.
Repairs are a routine matter, but prevention of defect recurrence is no less important than repair.
In a narrow sense, prevention of trouble recurrence means to prevent a repaired product from developing faults again.
But in this chapter, it means actions taken to eliminate the real cause of the trouble at the production stage by first reporting the quality information of products on the market.
(This is called feedback of quality information.)
As explained above, when some kinds of information, such as model number, total defects, details of defects, number of respective defects etc. are reported, the problem of what is to be done becomes obvious.
A must for quality improvement and problem-solving is not rough-and-ready guesswork, but specific information based on facts.
This is why quality information from the market is required.
Here, the Pareto Chart is shown as an example of problem analysis. There are other methods used, such as the cause and defect diagram, histogram, distribution chart, control chart, etc..
These must be used properly in the analysis of every problem.
Presentation of the details of these methods are omitted since it is not the purpose of this article.

Then What Kind of Information is Required?

In any case, at least, model number, serial number, details, and causes of malfunctions, quantity of defects, and defect rate are always indispensable. For your information, please refer to the examples attached at the end. In the case of quality information, defective samples or photographs will be helpful in solving the problem.
In case of reporting malfunctions, please sum the results of monthly repair by category and model (only current models).
Your regular information is highly appreciated.

QI Processing System in Tokyo

With the above knowledge, quality data must be processed efficiently and effectively. Information such as QI and reports of malfunction flow as below.

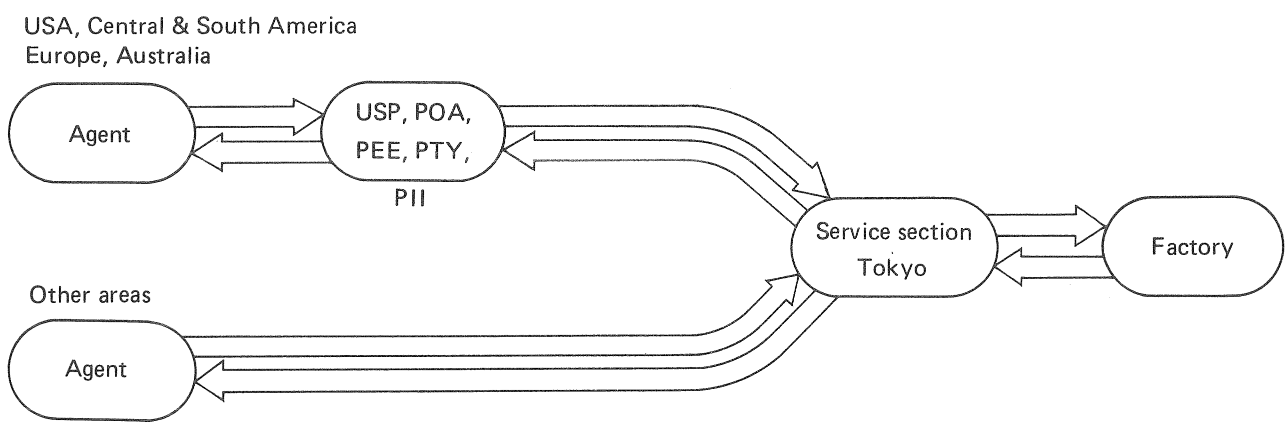
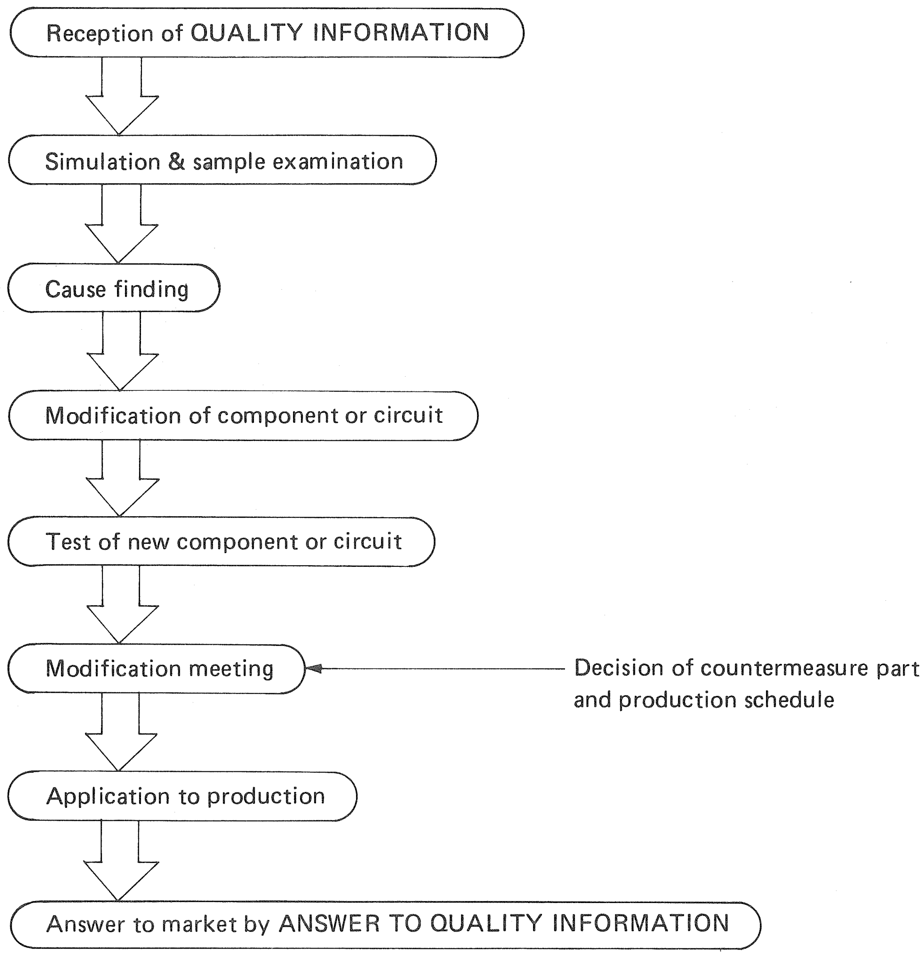


Fig. 7 Flow chart of data

QI's from USA, Europe, Australia and Central and South America are mailed to Tokyo via our subsidiaries and those from the other areas, directly.
In the Service Section, we check the data to see whether it is new, important, and complete and send it to the factories concerned. In case of QI, we make simulations, confirm the causes and recommend countermeasures.
Then, an answer to QI indicating countermeasures is sent to the parties concerned.
No answer is given to the report of malfunction. However, the report is important and your supplying it is highly appreciated.
Fig. 8 is the flow chart of the action taken after receiving QI.

We learned about quality, QC and QI in this series. Did you learn anything new? Some of you may be doubtful about helping us in the QC job. But, from a broad point of view, you will find it is important.

Quality improvement is profitable to consumers, and to distributors for increasing sales and decreasing service costs. Better design and materials will decrease production costs which is profitable to the consumers as well as to the distributors, and manufacturers, distributors, consumers: all the three parties need to help each other. We are all in the same boat in a sense. Let us do our best for a better future.



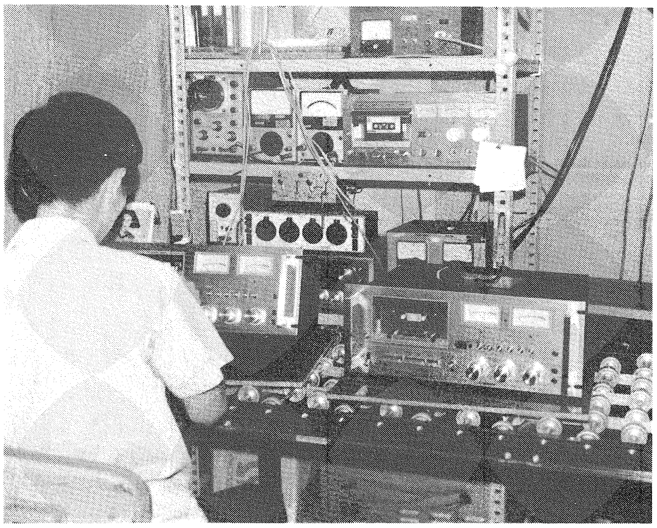


Photo 1
Outgoing inspection of tape decks.

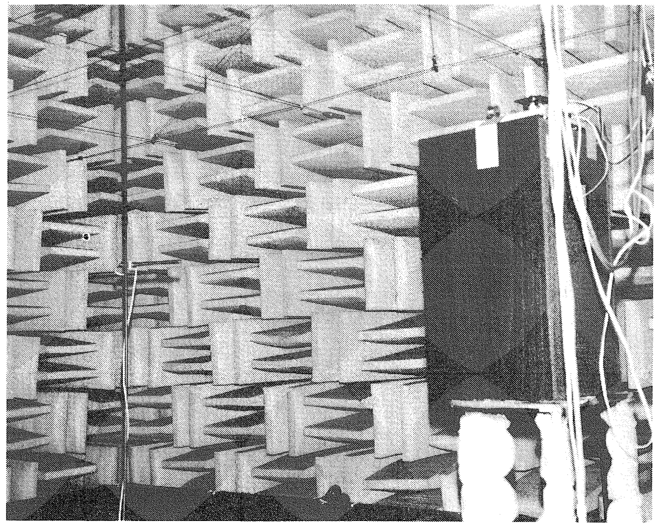


Photo 2
Performance evaluation test of a speaker system. All characteristics are measured in an anechoic room.

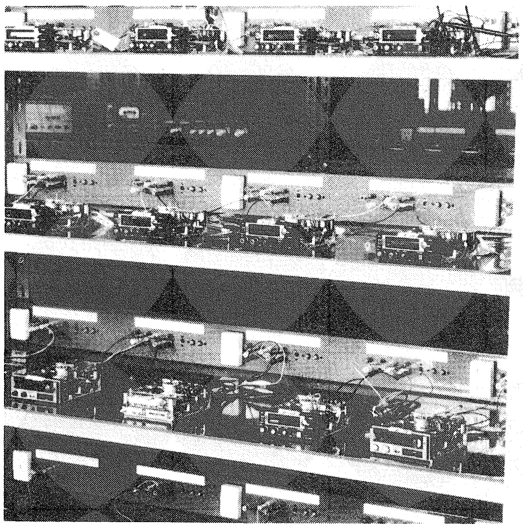


Photo 3
Continuous operation test for car stereos.

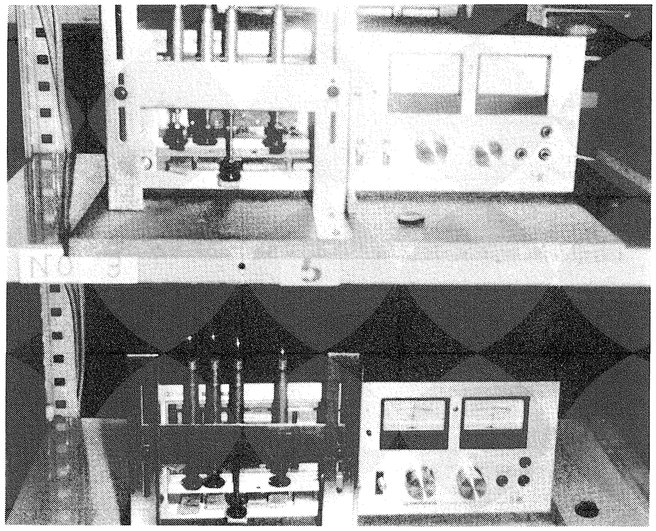


Photo 4
Continuous operation test for cassette decks.

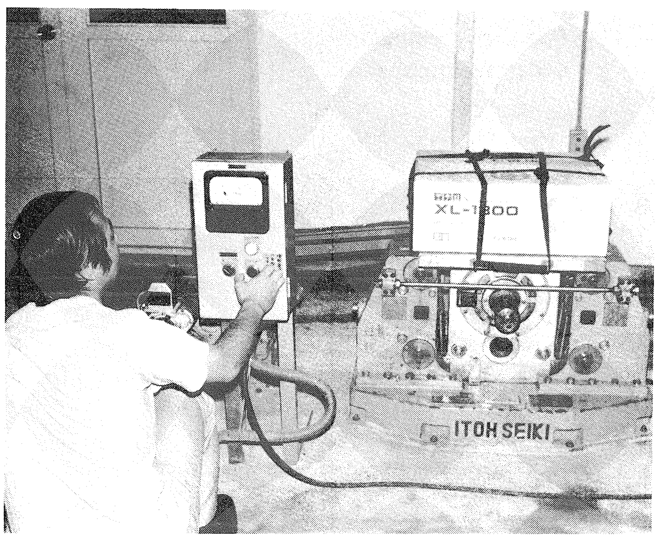


Photo 5
Vibration test for turntables.

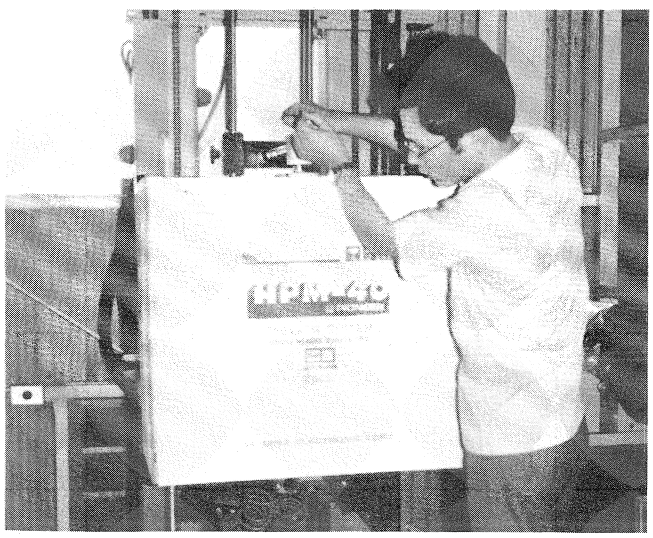


Photo 6
Drop test for speaker systems.

Photo 7
Temperature & humidity-controlled chamber for environment test.

Photo 8
We, the International Service Section, also do separate quality-checks when necessary.

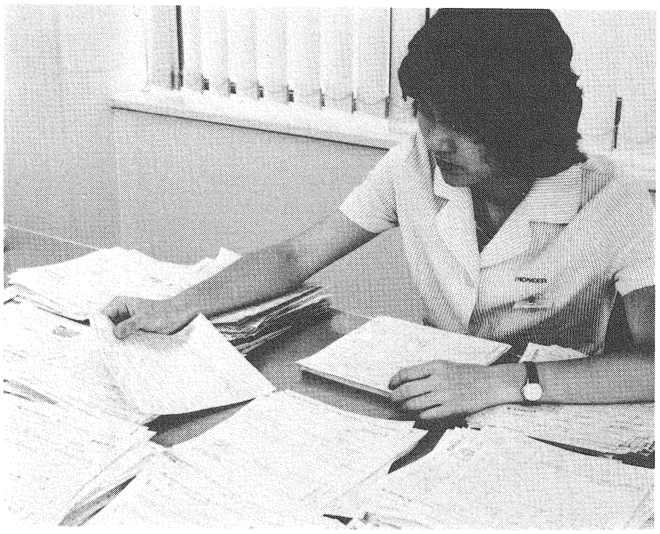


Photo 9
Data-summing of service reports sent from the field.

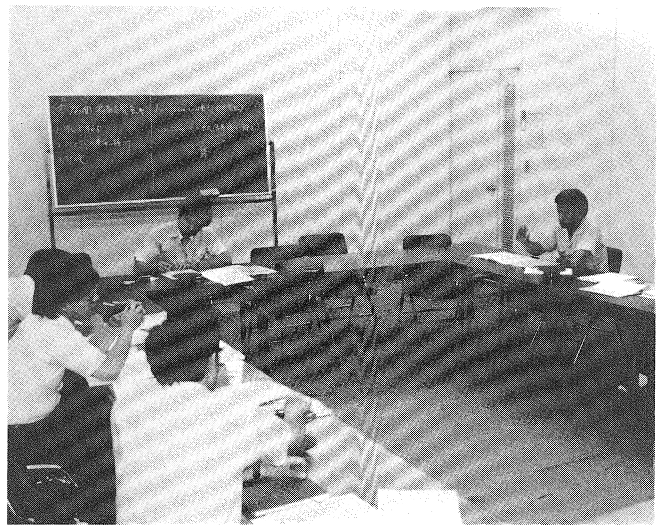


Photo 10
Regular QC meeting at the factory. Quality information from the field and factory is exchanged.

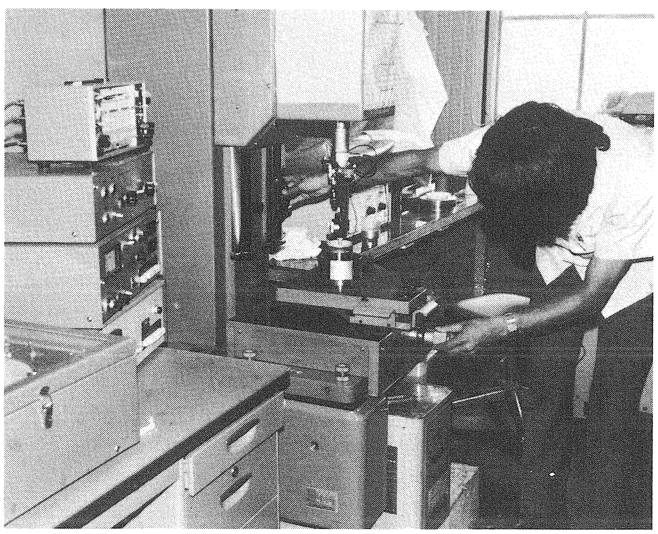


Photo 11
Finding the cause of trouble reported from the field. This photo shows the precise distortion measurement of the speaker's voice coil sent as a sample.

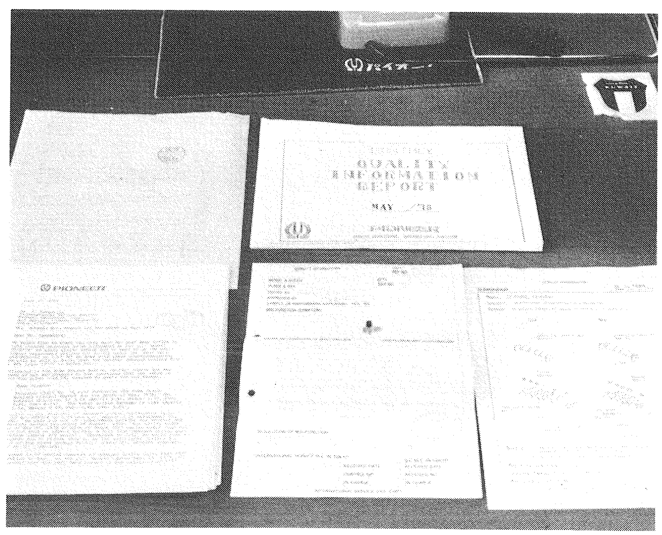


Photo 12
Various kinds of quality information incoming and outgoing.

QUALITY INFORMATION To Tokyo

DATE: 7-27-78

REF NO. NJ-809R

MODEL & SUFFIX RT-701/707	Q'TY: 1 SER NO. YB-3605750
PLACE & DEP. USP(NJ) Repair Dept.	
ISSUED BY Adolfo Reyes	
APPROVED BY	
SAMPLE OR PHOTOGRAPH APPENDING YES NO	

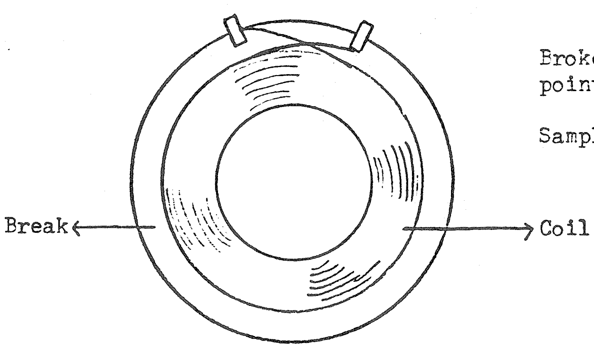
MALFUNCTION (SYMPTOM)

Sometimes the speed is too fast, some other times the speed is too slow and sometimes is OK.

NOTE: We have corrected a total of about 6 cases to present date, with the same problem and same cause.

CAUSE OR CONDITION OF MALFUNCTION (SUPPLEMENTARY ILLUSTRATION)

Upon inspection of circuitry, it was found that the motor generator coil was intermittently opening up. Further inspection of the motor coil revealed a break, of the coil wire at the terminal itself.



Broken at one of these terminals at this point.

Sample included.

RESOLUTION OF MALFUNCTION

By replacing entire motor.

INTERNATIONAL SERVICE DIV. IN TOKYO		Q.C SEC. IN TOKYO
断線傾向があるかどうかは不明	RECEIVED DATE: Aug. 8	RECEIVED DATE:
	CONTROL NO.: QD 8013	RECEIVED NO.:
	IN CHARGE: [illegible]	IN CHARGE:

C13

INTERNATIONAL SERVICE DEP. COPY

PIONEER

No. /

REPORT OF MALFUNCTION

☐ Discovered upon inspection
☑ Discovered upon repair service

DATE; July'78 PLACE; PEE(GMBH) ISSUED BY; J.Smith

MODEL NAME	SERIAL NUMBER	DETAIL OF MALFUNCTION	REPAIR ACTIVITY		
			ASS'Y NO.	REPLACED PARTS	DEFECTIVE PARTS NO. & CAUSE
SX-1050	XJ2909492	No sound		Replaced Q23, Q25, Q17, Q19, R61, R65, VR3 on power amp	
				(AWH-047) and R1,	R2 on protection ass'y (AWM-090).
	WK2902317	Left channel out		Replaced 2SB539C	x 2, 2SB536 x 1, 2SD381 x 1, 10 x 2.4
				and 7K x 1.	
SX-950 S	XE2904341	Cuts out		Replaced Q5 in P.S.	
	XB29033885	Sound cuts out		Replaced Q5.	
	WE2601548	Does not work		Replaced fuse	
	XC2902925	Phono right channel has noise		Replaced 2SC1313	on T2Q amp.
	WL2901769	Cuts out		Replaced 2SC1218,	Q5 on P.S.
	XB2901524S	Cuts off		Replaced trans. C 1211 (1318)	
	XD2903772	Cuts out intermittently		Replaced Q5, 2SC1318 on P.S.	
	XI2907225S	Cuts out		Replaced 2SC1318	on power supply
	XD2603257	Cuts off		Replaced C1318	
SX-850	XC2603156	Cuts out		Replaced Q5 in power supply	
	WE2600959	Cuts out		Replaced Q5 in power supply	
	XB2902613	Cuts out intermittently		Replaced Q1 + Q2	on AWR-101
	XB2903703	Check out		No problem	

Understanding More About Raidio Waves

In this issue, we will attempt to shed some light on the perplexing matter of how radio waves are transmitted through air. It would be simple if we could say that "because it is sent through air, it is a radio wave." But things aren't that simple. First, we must consider the characteristics of electromagnetic waves. It was once thought that three-dimensional space was made up of either air or a vacuum of ether, and that it was through this ether medium that electromagnetic waves were transmitted. However, the new school of thought is that the physical characteristics of space enable both light and electromagnetic waves to pass through it. Although this can be said of the space we know, there may still be some space through which radio waves will not pass. It is also generally thought that light travels in a straight line, and as the wavelength of electromagnetic waves becomes shorter, their characteristics, too, begin to resemble that of light. But we also know that light is bent by gravity. So, by determining the true characteristics of electromagnetic waves, we will be able to unravel the mysteries of space itself. And that time is fast approaching.

Do Radio Waves Travel in Straight Lines?

Since the speed of electromagnetic waves equals that of light, i.e. 300,000km per second, they pass around the earth seven and a half times every second. If radio waves were not refracted, reflected, diffracted or bent, this "fact" would not be true.

Just as light is refracted by a lens, so too are electromagnetic waves. (Light is a kind of electromagnetic wave.)

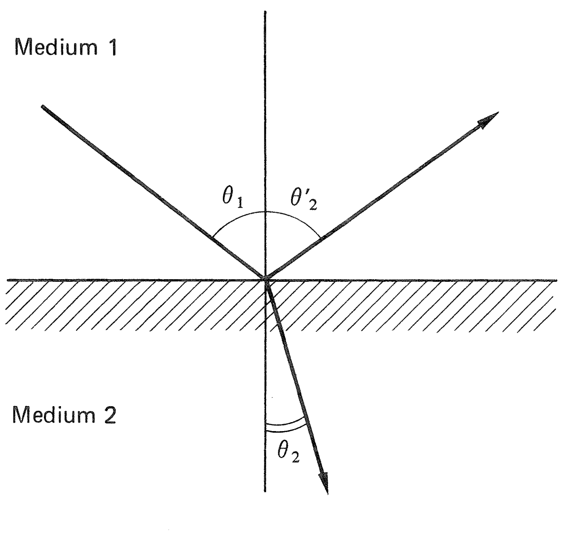


Fig. 1

A radio wave, which is a kind of electromagnetic wave, is also refracted by a lens.

What kind of conditions, then, reflect and refract electromagnetic waves? If you remember back to your science class at school, you will recall learning how radio waves travel in straight lines when passing through the same medium under constant conditions. However, this does not hold true when the medium changes, and reflection and refraction occur.

If you look at Fig. 1, you may recall the theory which states that the angle of incidence is equal to the angle of reflection. Here it is shown as $\theta_1 = \theta'_2$.

All radio wave energy, however, is not reflected. Part of it goes through another medium in the form of a refracted wave. The angle of this refracted wave is θ_2, and sin θ_1 divided by sin θ_2 is constant. If the new medium is a perfect conductor (with a resistance of 0), the refracted wave becomes zero and the incident wave is reflected 100%.

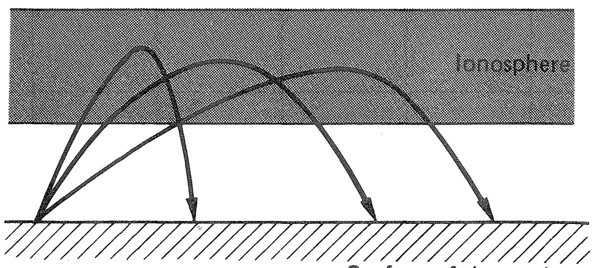
When the medium is a metal of almost perfect conductance, virtually all of the electromagnetic wave energy is reflected, and the ratio of energy absorbed by the medium (transmission factor) becomes low. As the conductivity decreases, or the resistance of the medium increases, the reflection factor drops, while the transmission factor rises. Which explains why wooden buildings are better for receiving radio waves than those made of ferro-concrete. These reflection and transmission factors are also related to wave frequencies. The higher the frequency becomes, the higher the reflection factor, with a corresponding drop in the transmission factor of the new medium after refraction. This is why AM reception is better than FM in ferro-concrete buildings. Referring back to Fig. 1, the relation of the angle of incidence to that of refraction is expressed as:

$$\frac{\sin \theta_1}{\sin \theta_2} = \frac{n_2}{n_1} = n_{12}$$

This is called "Snell's law." In the equation, n_1 and n_2 represent the refractive indices of medium 1 and 2 in a vacuum, while n_{12} represents the relative refractive indices of medium 2 against 1. Once you have determined the relative refractive indices, you will know how the radio wave is refracted after incidence. Using Snell's law, you can obtain the angle of refraction based on the angle of incidence.

Behavior of Electromagnetic Waves in the Ionosphere

The ionosphere is an area of space which has a thin atmosphere ionized by energy radiated from the sun, and its refractive index is less than 1. The refractive index is lower in the denser zone and gradually increases as the air gets more rarefied until it equals that of a vacuum. So what happens, then, when electromagnetic waves are beamed at the ionosphere? In point of fact, they are continuously refracted, there being little reflective component, back to the surface of the earth. For example, when a radio wave is sent at a certain angle, as shown in Fig. 2, the angle of refraction is bigger than that of incidence, as the refractive index of the ionosphere is less than 1, and the wave is refracted downward.



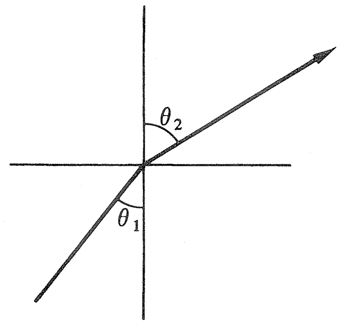


Fig. 2

The farther the radio wave penetrates into the ionosphere, the smaller the refractive index becomes. As a result, radio waves are continuously bent, forcing them to follow the curvature of the earth and finally descending to its surface. However, as you know, the higher the angle a radio wave is sent at, the farther into the ionosphere it will reach before it descends. Moreover, when the waves reach a certain height in the ionosphere, the refractive index starts to increase again and the refractive condition is reversed. This permits some of the radio waves to pass right on through the ionosphere. The frequency of these waves is called the "penetration frequency," and electromagnetic waves which have a higher frequency pass through the ionosphere and never return to earth.

Skin Effect on Conductors, Bone Effect on Dielectrics (Insulators)

When you hear the term "high-frequency alternating current" (radio waves in air), the expression "skin effect" automatically comes to mind. However since this expression is well known, we shall limit our explanation here to the "bone effect." This phenomenon was named by Dr. Hikosaburo Ataka and it is the reverse of skin effect. When a high-frequency alternating current is applied to a dielectric, the strength of the electric field is increased at its center. This energy is changed to heat in the dielectric and consumed as such. The center of the dielectric is also swelled by the heat (e.g. ebonite).
A dielectric, therefore, has the ability to change electromagnetic wave energy into heat. That is, a dielectric absorbs electromagnetic waves. As a result, some wave absorbers are actually made of dielectric material.
This is basically how electromagnetic waves behave. Now, let us consider what an electromagnetic wave is.

What is an Electromagnetic Wave?

Today, electromagnetic waves make telephone, radio, TV and many other forms of communication possible. But their existence was not confirmed in the same manner as other discoveries. Instead, they were theoretically predicted. The theorist was Maxwell of England, and he predicted their existence in 1865. Hertz of Germany is credited with actually generating an electromagnetic wave in 1887. But it was the famous Italian, Marconi, who invented the earthed antenna and succeeded in communicating by means of radio waves.
The following is a brief explanation of Maxwell's theory:

a) Magnetic field produced by an electric current flowing through a conductor

This occurs according to "Ampere's right-handed screw rule," which holds that a magnetic field takes the form of a circle, rotating in the direction of the electric current as it flows through a conductor. This is called a conduction current. The strength of the magnetic field decreases in inverse proportion to the square of the distance it is away from the electric current. This is called Biot-Savart's law.

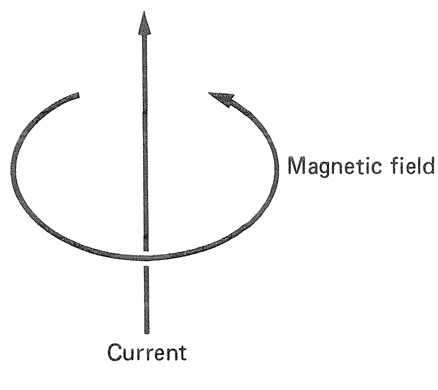


Fig. 3

b) Displacement current

In addition to conduction current, there is also convection current, which is produced by the electric charge generated when substances move, and displacement current (or dielectric flux current), which occurs when a dielectric is placed in an electric field. Convection current is that which is generated when flowing gaseous particles are charged (by an ionic charge, for example), or is current-collecting static electricity. However, it is displacement current that presents a problem for electromagnetic waves.

Expressed simply, displacement current is the electric current flowing in an insulator. As shown in Fig. 4, connecting batteries to the electrodes at each end of a capacitor immediately causes an electric current to flow, and then stop. This is because direct current will not flow through a capacitor. But connecting an alternating current source to the capacitor will cause the alternating current to flow in proportion to its frequency.

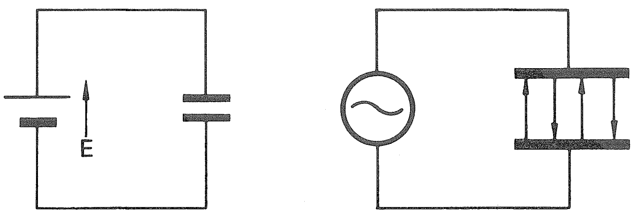


Fig. 4

Now, there is nothing surprising in the fact that alternating current flows through a capacitor. What is strange, however, is that insulators exist between the electrodes of a capacitor. And, as their name implies, they are supposed to insulate electric current.

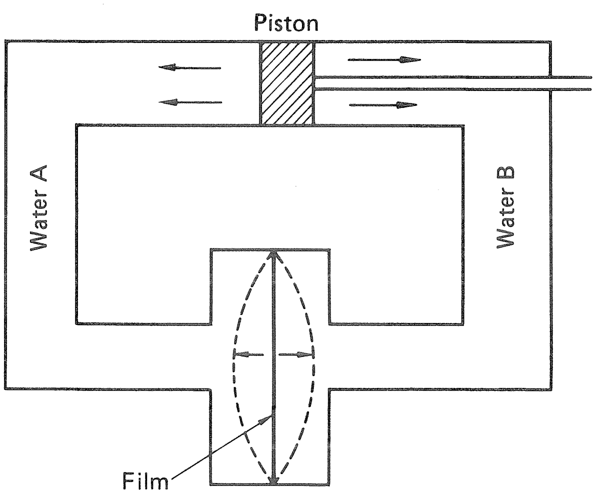


Fig. 5

This is a very interesting phenomenon which deserves some explanation. To do this, we will use the example shown in Fig. 5. Here, water A and water B are contained between a piston and a film, and the piston is made to move back and forth toward A and toward B. As the piston moves, water A or B is forced down, causing the film to swell on the left or right as the case may be. Looking at the movement of the water, it appears to be flowing up and down. However, each body of water is actually moving separately, being isolated by the piston and the film.

Comparing this with the electric current flow in a capacitor, the film represents the capacitor, while the water corresponds to the electric current. The piston plays the role of the alternating current source. Accordingly, even if the capacitor (the film) is insulated, when the alternating current source (the piston) is connected, or in this case made to move, electric current (the water) flows back and forth in an alternating manner. This is why the electric current flow in an insulator is referred to as displacement current.

c) Radiation of electromagnetic wave

A magnetic field occurs in circles around the current according to Ampere's right-handed screw rule. In addition, conduction current runs in the conductor when displacement current, which I explained above, flows in the insulator. Conversely, as the direction and the strength of displacement electric current changes, so the magnetic field which occurs around it also changes. According to Faraday's law on electromagnetic induction, when the magnetic field increases, an electric field occurs in the same direction to prevent an increase of magnetic field (the magnetic

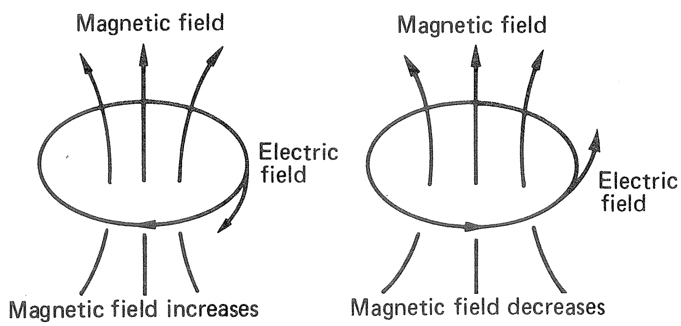


Fig. 6

field occurs at right angles to the electric field), and also when the magnetic field decreases, an electric field appears in the same direction to prevent the decrease in magnetic field. Furthermore when the electric field changes, a corresponding magnetic field appears at right angles to the electric field. Consequently "the change in the magnetic field induces an electric field and the change in the electric field induces a magnetic field and another electric field and yet another magnetic field, spreading in space like a linking chain at right angles." This chain of electric field and magnetic field, spreading into space, is radiating electromagnetic energy, namely radio waves.

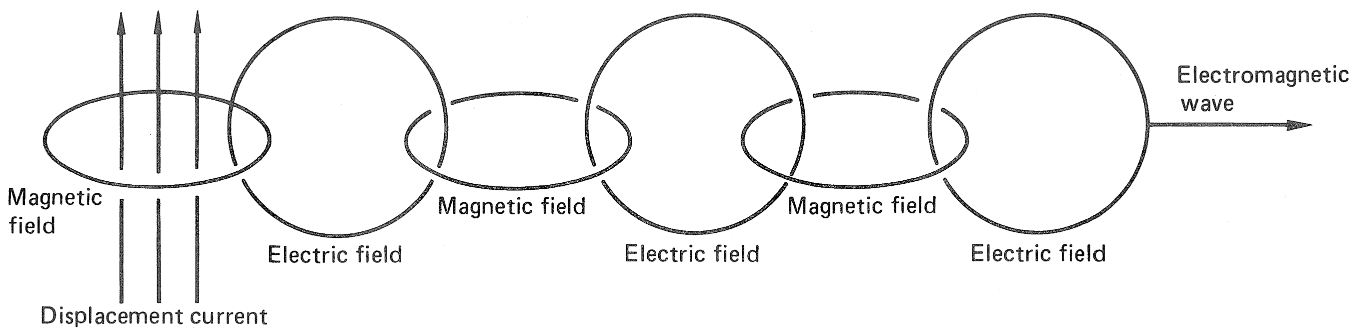


Fig. 7

d) Plane of polarization

Next, let us discuss earthed antennas and the planes of polarization. To begin with, as shown in Fig. 8 (1), when an AC voltage is applied to capacitor C, a displacement current flows in the capacitor and electric waves are radiated. Fig. 8 (2) shows a capacitor formed by a pair of conductors and air (air also being a kind of dielectric). Spreading the two conductors is shown in (3), while (4) shows an earth being used in place of one of the two conductors. In other words, if an alternating current is applied to a capacitor consisting of an earth and conductors, a displacement current will flow between the earth and one of the conductors. As a result, a radio wave is effectively radiated which can be sent a fairly long way. This kind of conductor is called an antenna, and when one of the two conductors is earthed, it is called an earthed antenna.

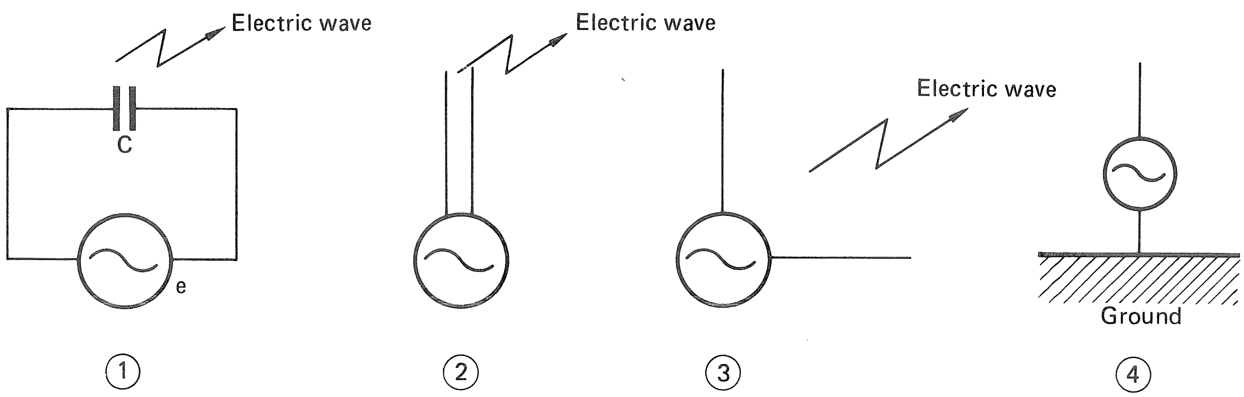
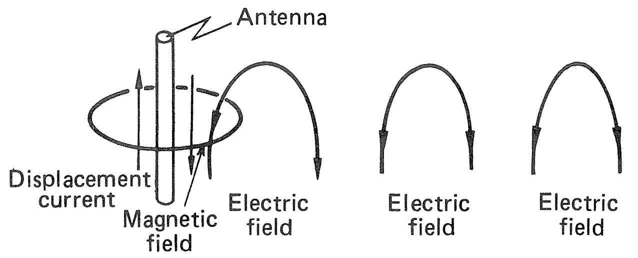


Fig. 8

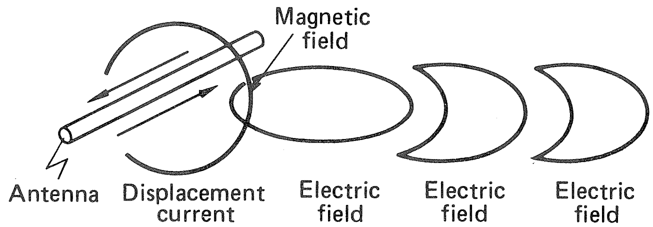
When the displacement current generated by an earthed antenna flows at right angles to the ground, a magnetic field occurs at right angles to that of the current. Moreover, an electric field also occurs at right angles to the magnetic field. Consequently, magnetic fields are always parallel to the ground, and electric fields are always at right angles to it. Thus, the relationship between the two remains constant. And the plane in which electric fields oscillate is called the plane of polarization. In this case, it is called a vertically polarized wave.

When the displacement current flows parallel with the ground, the electric field also oscillates parallel with it, too. In this case, it is called a horizontally polarized wave. For example, a medium-wave AM broadcast uses vertically polarized waves, while an FM broadcast on the VHF band uses horizontally polarized waves.

In the upcoming issue, we will talk about the propagation of radio waves.



Vertically polarized waves



Horizontally polarized waves

Fig. 9

Loudness and frequency differential threshold of the human ear

When a sound having a certain uniform loudness is boosted or attenuated, the human ear's sensitivity to the change is referred to as its *loudness threshold.* The loudness threshold represented by changes in sound volume is shown in Fig. 1. As the chart indicates, when the sound pressure level is under 50dB, the loudness threshold level rises as the sound pressure level and frequency get lower. When the sound pressure level becomes more than 50dB, the loudness threshold remains constant, regardless of the frequency, and stays under 1dB throughout the entire audio range. At a loudness threshold of 4kHz, sensitivity is a small 0.2dB.

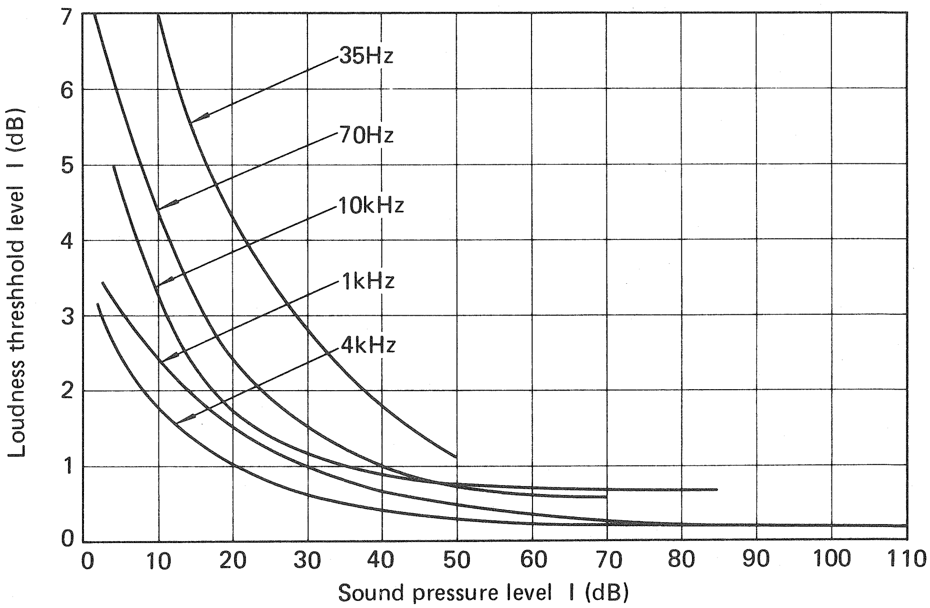


Fig. 1 Loudness threshhold level

On the other hand, when a sound of a certain uniform frequency (f) is shifted by ±(△f), the recognizable difference is referred to as the frequency differential threshold. The frequency differential threshold of a pure tone, represented by △f/f, is shown in Fig. 2. As you can see, when the frequency is below 500Hz, the threshold increases as the frequency gets lower, and when the frequency is higher than 500Hz, it becomes constant. In case the frequency is 1kHz and the sound pressure level is 60dB, △f/f is approximately 0.003, and even a frequency difference of 3Hz is detectable.

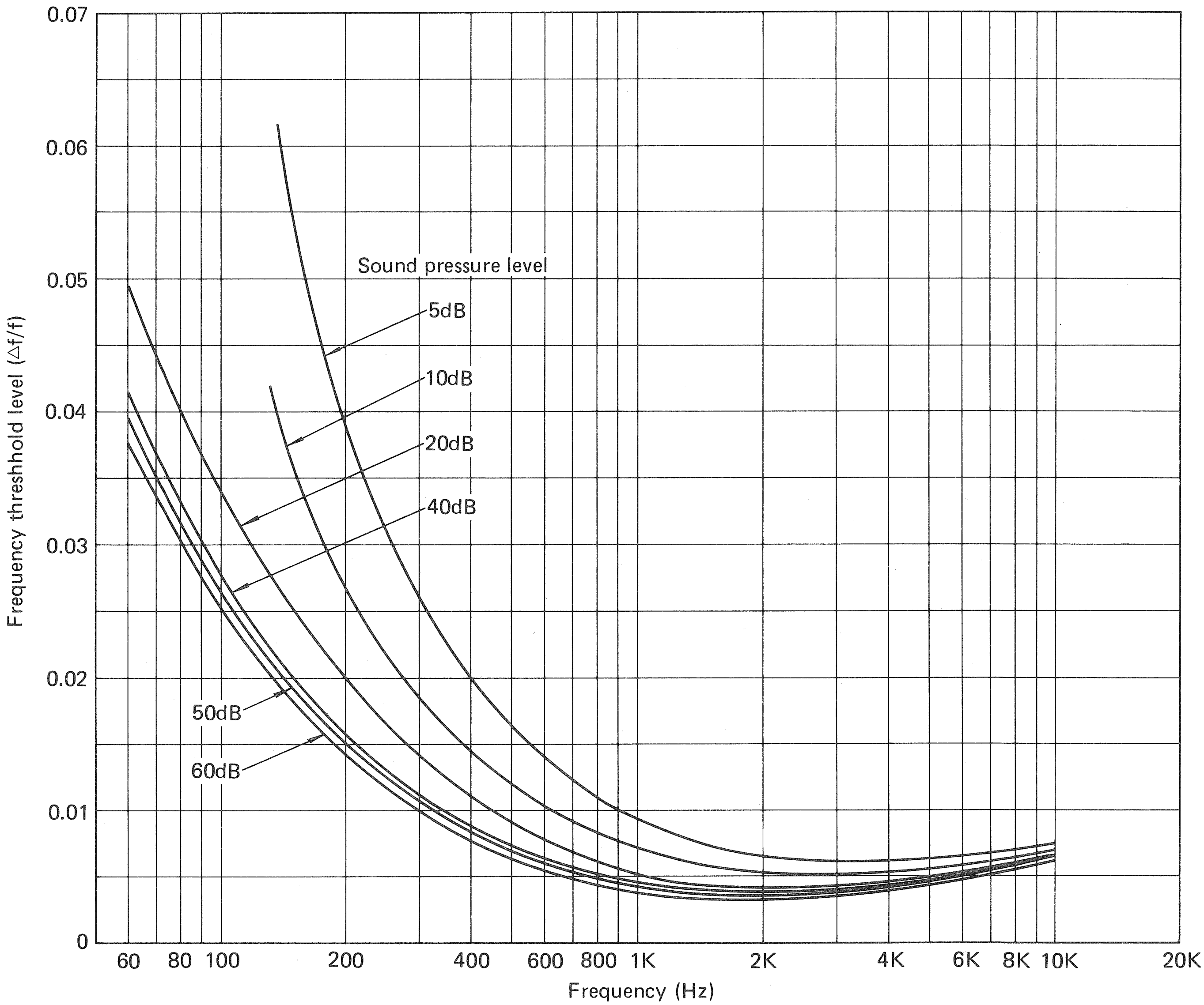


Fig. 2 Frequency threshhold level of pure sounds

This, however, only applies to pure sounds. In the case of complex sounds, such as voices and music, the recognizable threshold level becomes higher, in which case, a sensitivity level of 0.0002μbar represents 0dB.

Q & A

I have been a music-lover for some time now, and have owned Pioneer equipment for several years. Until recently, I was using a PL-71 turntable, an SX-1010 receiver, and two CS-99A speakers. About three weeks ago we moved to a new place, and in a sudden rush of generosity I gave my son the SX-1010 and the two speakers. In place of them I bought the SPEC-1 and SPEC-2 with a pair of your new HPM-150 speakers.

According to the salesman where I bought the new equipment, the so-called NOMINAL INPUT POWER: 125 WATTS, indicated in speaker specification of the HPM-150 is equivalent to MAXIMUM INPUT POWER: 250 WATTS, which was in use before.

I want to say that I'm very satisfied with the sound, but somewhat puzzled by the fact that the maximum volume of my new equipment sounds about the same as my older equipment. Not that I listen to music at top volume, of course.

Specifically, the MAXIMUM INPUT POWER of the CS-99A is 100W and the MAXIMUM OUTPUT POWER of the SX-1010 is 100W/Channel RMS. The MAXIMUM OUTPUT POWER of the SPEC-2 is 2½ times greater than the SX-1010 so I feel that my new equipment should be giving me more sound than the old. Can you help me solve this puzzle?

Yours truly,

Wickham Smith Mt. Kisco, New York

Answer: We receive questions like this one all the time, and we suspect that the reason for the confusion is in large part due to consumers' inability to understand the meaning of speaker specifications.

A speaker, of course, is a transducer which changes electrical energy into acoustic energy by vibrating the air. When considering loudness, we have to take into account the conversion efficiency of the speaker as well as the input power.

In brochures, we see specs like "SENSITIVITY.... XXdB/W." This indicates the efficiency of the speaker. But to better understand the sensitivity of a speaker, let's look at the definition: Sensitivity is measured in an anechoic room under specified conditions as shown in Fig. 1. The microphone is placed at a distance of one meter on-axis from the speaker enclosure. The input voltage applied to the speaker is 1W, adjusted for the rated impedance of the speaker.

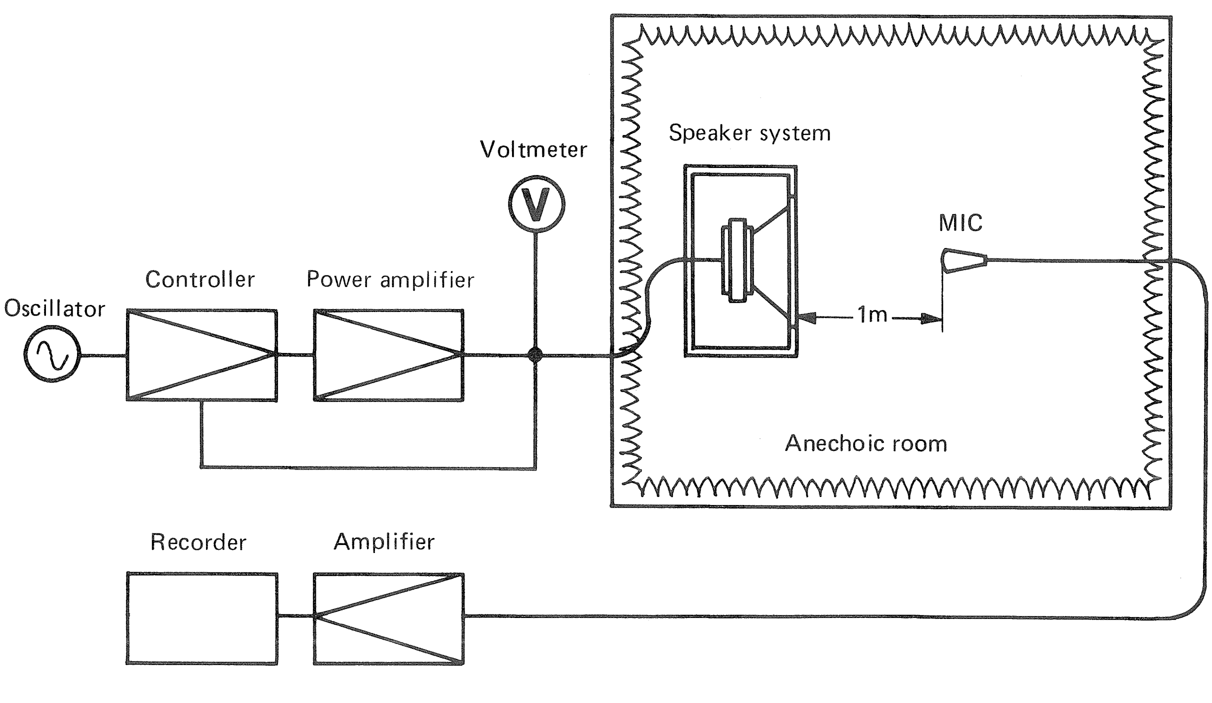


Fig. 1 The measurement of a speaker system

Under the conditions shown in Fig. 1, the sound pressure levels measured at 1 meter from the speaker are recorded with continuously varying frequencies (see Fig. 2).

The sound pressure is measured and expressed in dB, with a 0dB reference point being 0.0002μbar. Therefore:

$$S = 20 \log \frac{P}{0.0002\mu\text{bar}}$$

[S: Sound pressure level (dB)
P: Sound pressure (μbar = dyne/cm^2)]

This 0dB standard of 0.0002μbar is considered to be the minimum sound pressure level that man can hear, and so has been adopted as the increment of measurement.

From the chart showing frequency characteristics of output sound pressure measured using the process just described, we take measurements at four prescribed frequencies, average them, and then take the result for the published sensitivity specification. Therefore:

$$So = \frac{S_1 + S_2 + S_3 + S_4}{4}$$

[So: Output sound pressure level (dB)
$S_1 \sim S_4$: Sound pressure levels (dB) at 4 fixed-point frequencies]

The four fixed frequencies are usually 300Hz, 400Hz, 500Hz, and 600Hz (but if there is a peak or dip in frequency response at one or more of these points, they may be shifted a little to get a more uniform reading).

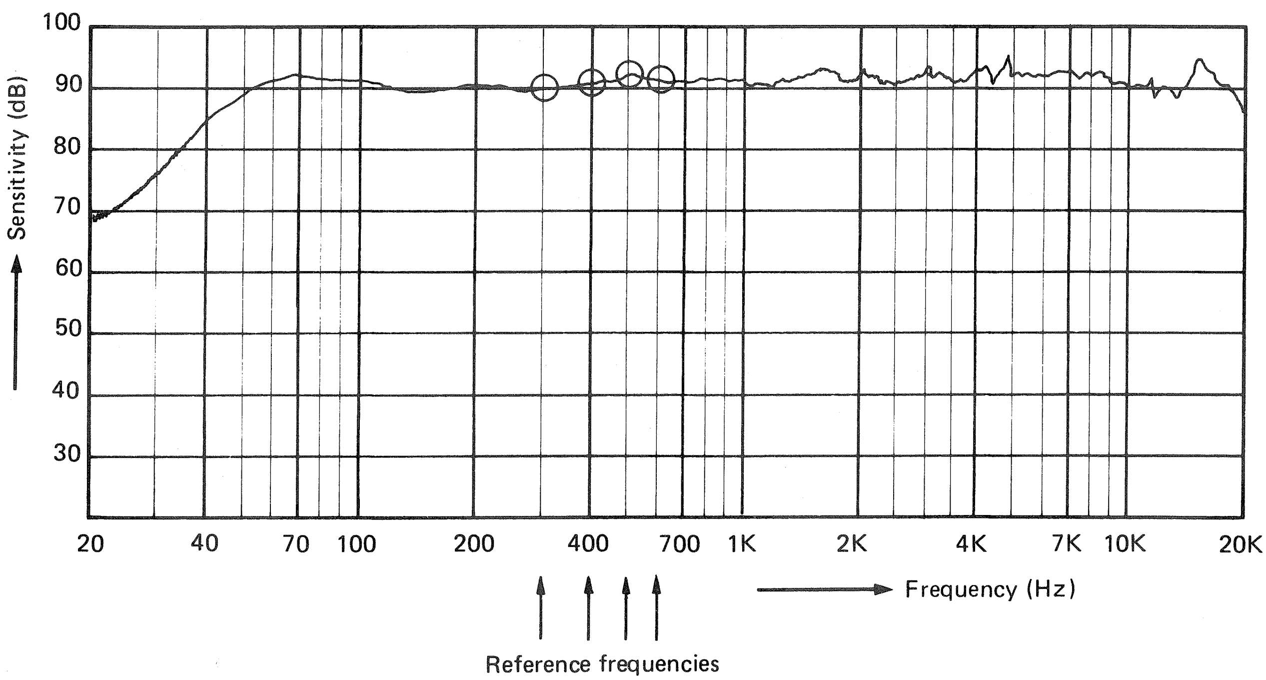


Fig. 2 Calculation of output sound pressure level

So taking what we now know about measuring sensitivity, let's calculate the sound pressure level at the maximum input power for each speaker system here.

Sound pressure obtained with a speaker is in proportion to the input power fed to the speaker. So the sound pressure obtained from maximum power, *Po* max (μbar), is:

Po max = $Po \times Pi$ max

[*Po*: Output sound pressure (μbar) at 1W input power

Pi max: Maximum input power (W)]

Accordingly, when the results are expressed in decibles (dB), the equation reads:

So max = $So + 10 \log Pi$ max

[*Po*: Output sound pressure level (dB) at 1W input power]

So the maximum sound pressure levels obtainable with the CS-99A and the HPM-150 are obtained as follows:

CS-99A: So max $= 97 + 10 \log 100$
$= 97 + 20 = 117$ (dB)

HPM-150: So max $= 92.5 + 10 \log 250$
$= 92.5 + 24 = 116.5$ (dB)

(Nominal Input Power, don't forget, is roughly equivalent to half the older Maximum Input Power, when referring to the HPM-150 i.e. 125W.)

Our calculations show that the difference between sound pressures obtainable at maximum input power is 0.5dB, virtually impossible to sense with human ears. That's the reason that Mr. Smith had such difficulty hearing any difference in sound levels between his two systems.

In closing, as a reference, the output sound pressure levels at 1W of input power are given below.

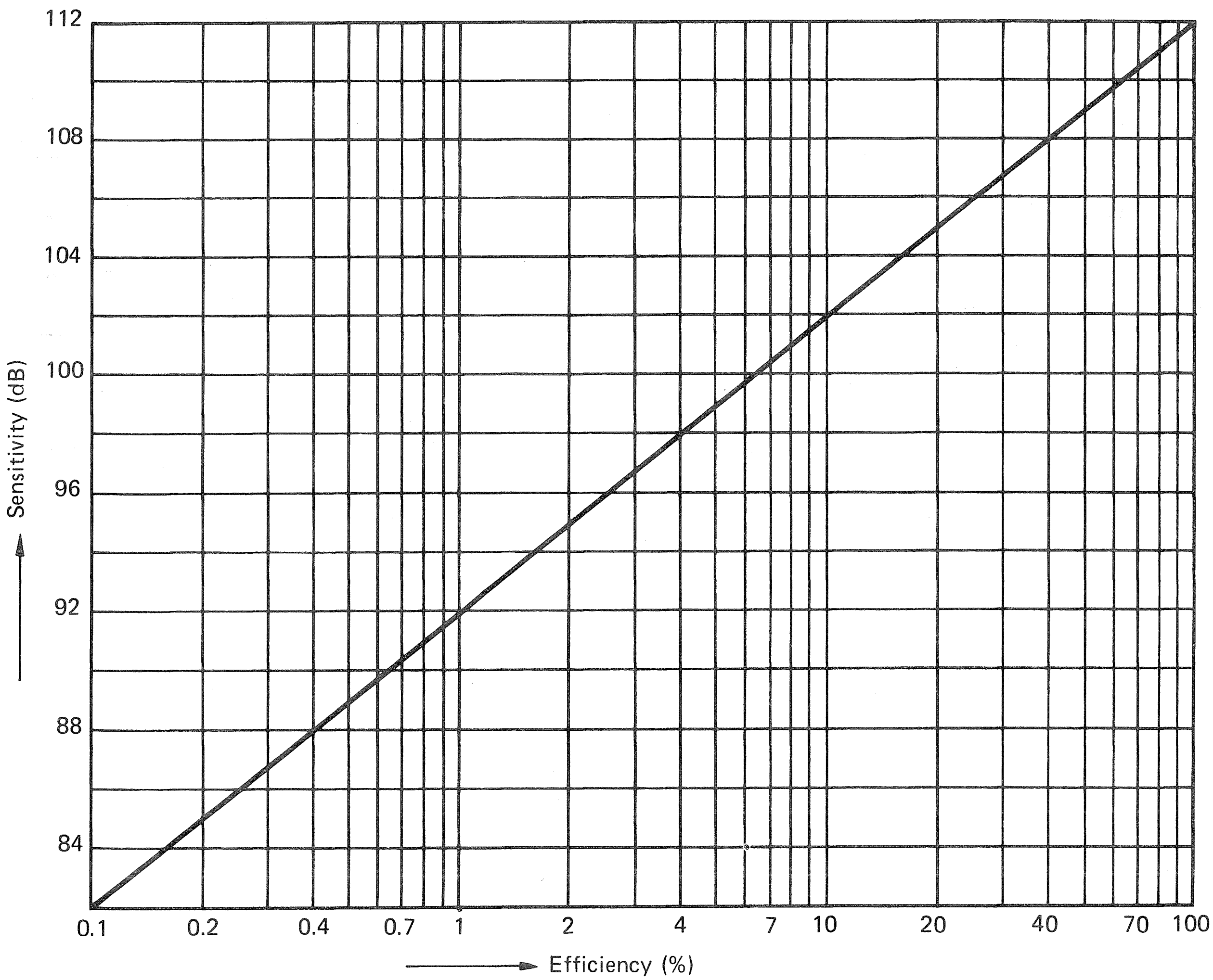


Fig. 3 Output sound pressure level at 1W input power and conversion efficiency

IN THE NEXT ISSUE

Editor's Note

Another year has almost passed and 1978 is quickly becoming just a memory. But it will be a good memory for those of us who have the pleasure of contributing to *Turning Fork*. At this time of year the Japanese people are hustling to get cleared away in the old year to bring in the new with a clean slate. From our office in Tokyo's Meguro district we can see snow-capped Mt. Fuji, and like all Japanese, we are very moved by the sight. We reflect back on the old year and plan for the new one as we hope you are doing–accompanied by the sound of your stereo systems.

We want to thank you for your response to this year's questionnaire, and we will make every effort to respond to the suggestions you have made.
Best of luck in the New Year.

T. Taguchi
H. Koike
A. Kogirima

Publisher
Ikki Nagashima

Service Section
Administration Department
International Division

PIONEER ELECTRONIC CORPORATION
4-1, Meguro 1-Chome, Meguro-ku, Tokyo 153, Japan

U.S. PIONEER ELECTRONICS CORPORATION
85 Oxford Drive, Moonachie, New Jersey 07074, U.S.A.
PIONEER ELECTRONIC (EUROPE) N.V.
Luithagen-Haven 9, 2030 Antwerp, Belgium
PIONEER MARKETING SERVICES PTY. LTD.
P.O. Box 317, Mordialloc, Victoria 3195, Australia
PIONEER ELECTRONICS OF AMERICA
1925E, Dominguez St., Long Beach, California 90810, U.S.A.

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