THE INTERNATIONAL TELEPHONE CONSULTATIVE COMMITTEE

(C. C. I. F.)

XVIIth PLENARY ASSEMBLY

GENEVA, 4-12 OCTOBER 1954

ANNEXES TO VOLUME IV

Quality of transmission
(documentation on methods of specification and measurement)
Telephone Equipment

Published by the
INTERNATIONAL TELECOMMUNICATION UNION
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TABLE OF CONTENTS OF THE BOOK OF ANNEXES TO VOLUME IV OF THE " GREEN BOOK "

ANNEX 1

MEAN RATING OF JUNCTIONS

A junction can be considered as a quadripole inserted between the impedance presented by the first trunk circuit seen through the components of the trunk switchboard or trunk automatic equipment and the impedance of the local telephone circuit (feeding bridge + subscriber's line + subscriber's apparatus).

For any given frequency, the loss introduced by such a line is represented by its " composite attenuation " * which is the sum of the image attenuation of the line itself and other terms representing all the effects due to reflexions introduced by the mismatch between the image impedance of the line and the impedance of terminations such as those defined above.

From work by the British Administration, the rating corresponding to reflexions can be represented by the arithmetic mean of the reflexion losses measured at the frequencies of 500, 1,000, 2,000 and 3,000 c/s.

On the other hand, the transmission performance rating of an unloaded line is given by its image attenuation at 1,500 c/s and such an attenuation is approximately equal to the arithmetic mean of the image attenuations at the four frequencies quoted above **.

Consequently, one can obtain directly the rating of the junction, comprising both the effect due to its image attenuation and that due to reflexions, by taking the arithmetic mean of the composite attenuations measured at the above mentioned four frequencies.

The impedance of local circuits is a very variable quantity so that one cannot define a unique quantity for the mean rating of a junction but only a mean value obtained by taking the arithmetic mean of several values of rating measured under several terminal conditions (see Document "C.C.I.F.—1952/1954—4th Study Group—Document No. 32 "—Annex).

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* In practice, insertion loss may be used instead of " composite attenuation ".

** The attenuation of a circuit in unloaded cable is proportional to the square root of the frequency. The frequencies 500, 1,000, 2,000 and 3,000 c/s are in the ratios 1, 2, 4, 6 and their square roots in the ratios 1, 1.41, 2, 2.45 of which the arithmetic mean is 1.72, i.e. approximately the square root of 3 ; therefore this mean corresponds approximately to a frequency of $3 \times 500 =$ 1,500 c/s.

For each type of junction (defined by the electrical characteristics of the line) the mean rating is approximately a linear function of the line length : this function can be *readily defined* when one has three or four values of rating. The function is of the form :

$$i = K \times L \qquad (1)$$

where

i = mean rating in decibels or nepers
L = length of the junction in kilometres
K = coefficient for each particular type of line in nepers per kilometre or decibels per kilometre.

To determine, once for all, the different values of the coefficient K one can measure the composite attenuation of three or four different lengths of each of the types of junction used in a particular network (actually represented by artificial lines); for this purpose one can use the technique described in the Document No. 32 referred to (see also Annex 2 to question No. 10 in Volume I *ter* of the *Yellow Book* of the C.C.I.F., page 400) and one of the methods for measuring composite attenuation described in Volume IV of the *White Book*, the *Book of Annexes* to Volume III of the *Green Book*, 2nd Part, Section 1.1.1.

The relationship (1) then enables the mean rating to be calculated for any length and any type of junction forming part of the national network considered.

ANNEX 2

SUMMARY OF A METHOD USED BY THE BRITISH TELEPHONE ADMINISTRATION TO DETERMINE RELATIVE TRANSMISSION PERFORMANCE RATINGS OF LOCAL SENDING AND RECEIVING SYSTEMS USING THE SAME TYPES OF MICROPHONE AND RECEIVER

1. The British Administration assesses the transmission performance of local telephone circuits by comparison with the performance of a specified working standard circuit. All ratings are expressed relative to this working standard. The working standard circuit has been chosen as representative of general practice and has reference equivalent values (for sending and receiving) equal to the limits permitted in the British network. This working standard circuit is defined by Figure 1 below and the description below this figure.

2. When certain basic data, as indicated below in Section 3, have been obtained, using representative components of the local telephone system, the assessments are calculated from these data and from objective (electrical) measurements. By the term " representative components " is meant components whose performance is known to be average among those in service and only assessments of relative ratings of circuits using such representative components are considered here. Changes of relative ratings due to variations of efficiency of components *of the same type* can be assessed by this method, but the method is not used for directly assessing relative ratings between circuits using different types of telephone transmitter or receiver.

For certain classes of objective comparison tests (e.g. when the circuits to be compared exhibit generally similar variations of response with frequency and do not involve sharp peaks of response or cut-off effects) a single valued rating is obtained from tests at the four frequencies 500, 1,000, 2,000 and 3,000 c/s. by taking the arithmetic mean of the ratio, expressed in db, measured at each of these frequencies. In what follows, specific indication of those tests to which this method is applied is given by the reference " 4-frequency rating ".

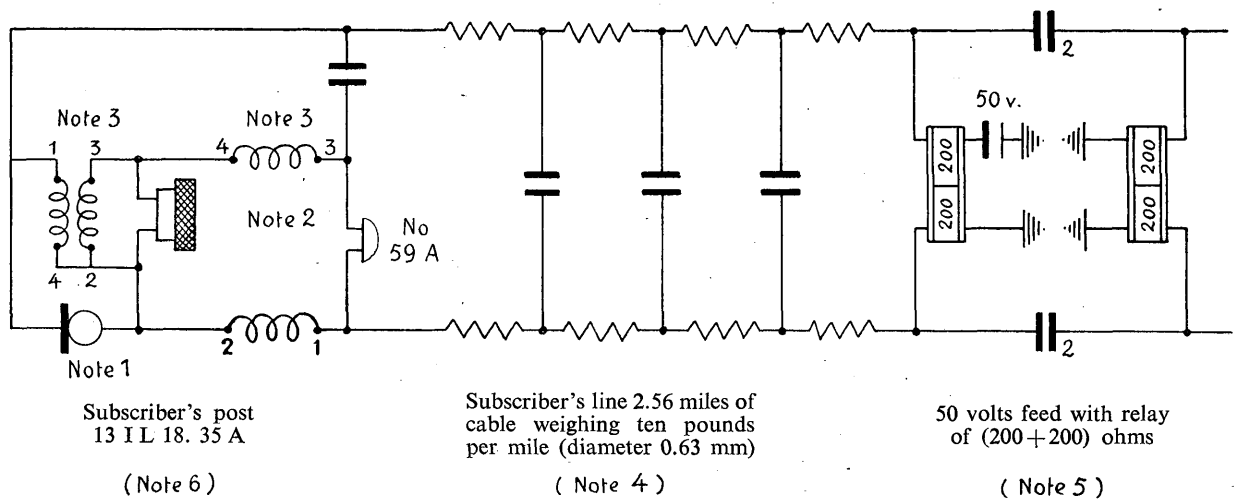


FIGURE 1. — *The circuit chosen to represent the lower transmission performance limit*

Note 1: Transmitter inset No. 13 fitted in TELE. 164 handset.
— Transmitter resistance (talking) = 66 ohms (see curve sheet).
— Transmitter feeding current = 40 mA.
— Transmitter rating (average in service) = + 1.5 db relative to Department's standard.

Note 2: Receiver inset I L fitted on TELE. 164 handset.
— Receiver rating (average of supplies) = 0 db relative to Department's standard.

Note 3: Coil induction No. 18 and transformer 35 A, both equal to Department's standard.

Note 4: Subscriber's line loop resistance 450 ohms.
— Capacitance : 0.075 μ F/mile.
— Attenuation 2.05 db/mile (average for actual cables) line current 55 mA.

Note 5: Circuit connected to average junction.

Note 6: Continuous spectrum noise, at both sending and receiving stations, weighted to have average room noise spectrum described by Hoth (*J.A.S.A.* April 1941) at a level of 60 db as measured on an American sound level meter. (American tentative standard specification Z 24.3 — 1936.)

The procedure outlined below refers in particular to Common Battery telephone circuits ; it can, however, also be applied to Local Battery circuits which are in fact simpler cases, since the feed current of the transmitter is independent of the line.

3. *Data required.*

3.1. — Relation between feed current in the transmitter and resistance of the transmitter (with speech).

To obtain this relation it is convenient to make use of a suitable steady sound instead of speech to energize the transmitter. Stopping the sound and slightly agitating the transmitter at intervals between measurements may be desirable to ensure uniformity of performance of the transmitter in a state simulating conditions of use.

3.2. — Relation between feed current in the transmitter and the generated e.m.f. due to speech.

Only relative values are required and these are conveniently obtained in decibels relative to the value at the feed current of the working standard circuit, using a simulating steady sound, as in 3.1.

3.3. — Relation between transmission performance rating and length (or resistance) of (unloaded) local line.

This rating is assessed (for each type of subscriber's line in use) as the attenuation at 1 600 c/s. for the length concerned.

3.4. — Variation of relative receiving rating due to change of side tone with 60 db room noise.

To obtain this relation, subjective testing, for example, by the A.E.N. technique, is necessary. Various ways of carrying out the tests are possible ; the method which has been used by the British Administration was as follows :—

An electrical circuit was set up which allowed the side-tone of the subscriber's sets (four in number) at the listening end of the articulation test circuit to be altered without change in the receiving path of the subscriber's set. This was accomplished by inserting (with proper precautions so as to present correct impedance values) a combination of attenuator and amplifier in the transmitter circuit of each subscriber's set. The sensitivity of the side-tone path could then be altered by known amounts both above and below the standard side-tone condition and A.E.N. measurement were made over a wide range of values of side-tone reference equivalent.

3.5. — Variation of relative sending rating due to change of side-tone.

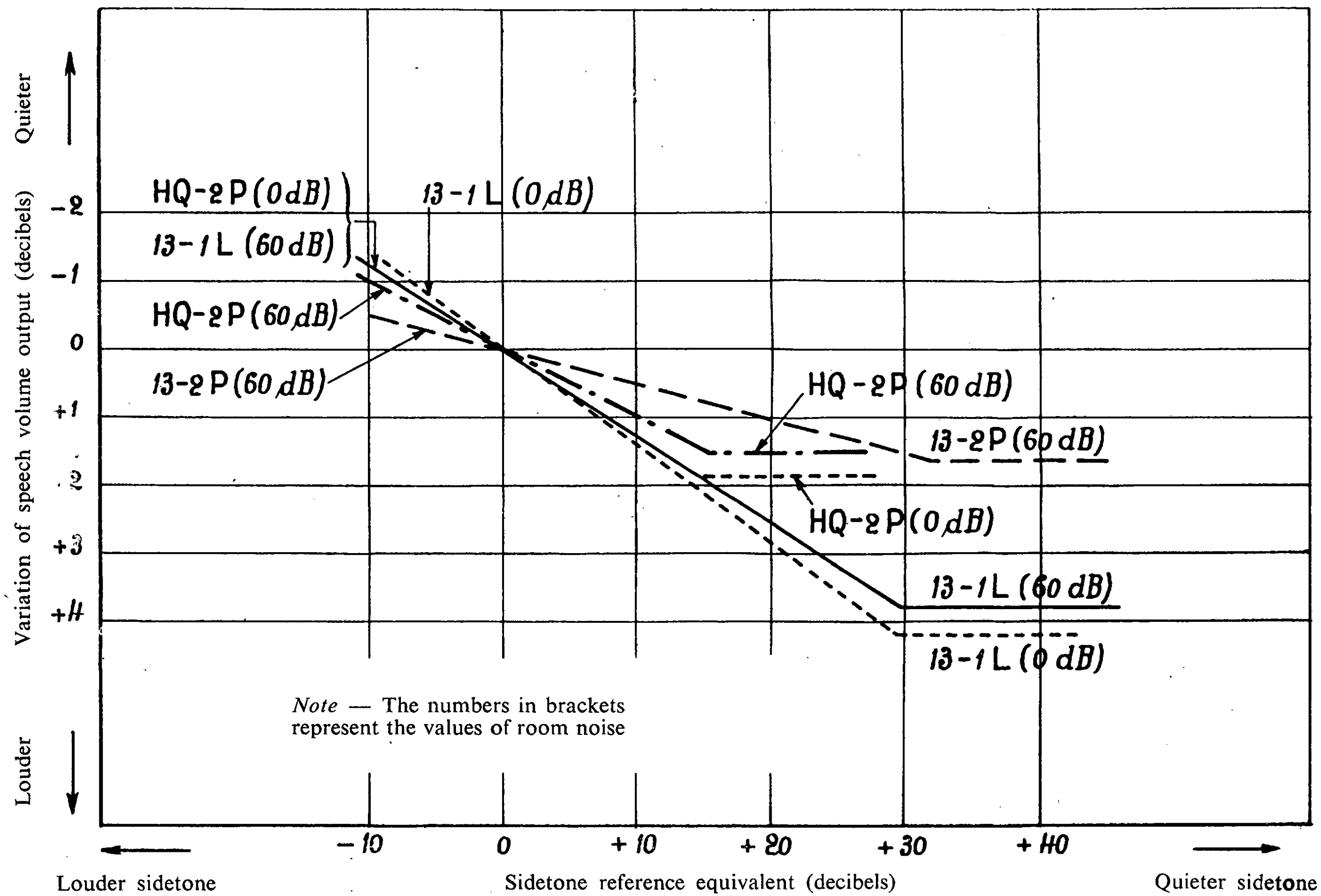


FIGURE 2. — *Variation of speech volume output as a function of sidetone reference equivalent*
(Information supplied by the British Telephone Administration)

Tests were made by means of a speech voltmeter (see Annex 12 of the Book of Annexes to Volume IV of the *Green Book*, p. 122) connected across the telephone circuit. Normal conversational speech was used and the side-tone sensitivity was varied by means of a combination of attenuator, amplifier and a separate receiver which replaced the receiver of the subscriber's set.

The effect can be measured only by tests with several subjects as speakers.

The curves in Figure 2 show the effects upon speech volume of the sidetone reference equivalent for a variety of telephone sets under conditions of silence and in the presence of 60 db room noise. These results apply for conditions of conversation substantially better than the tolerable transmission limit.

The microphones and receivers used were as follows :

(a) Transmitter Inset No. 13 (microphone standardized by the British Post Office).

(b) High-quality moving-coil microphone mounted on a normal handle (denoted by HQ in Fig. 2).

(c) Receiver Inset No. 1 L (old-type British Post Office receiver with a pronounced resonance).

(d) Receiver Inset No. 2 P (new-type British Post Office receiver having a level sensitivity-frequency characteristic).

Note. — On the other hand the Chile Telephone Company has given the C.C.I.F. numerical results concerning the effect of sending end sidetone on speech volume which are shown by the curves of Figure 3 below.

4. *Items of Calculation.*

It is convenient to list or tabulate separate items of calculation as follows :

A Length of subscriber's line,
B Feed current in transmitter,
C Feed current effect (db),
D Relative rating for loss of line (db),
E_1 Relative Electrical Rating of Telephone Circuit and Feeding Bridge for Sending (db),
E_2 Relative Electrical Rating of Telephone Circuit and Feeding Bridge for Receiving (db),
E_3 Relative Electrical Rating of Telephone Circuit for Sidetone (db),
F Relative Sidetone level (db),
G_1 Sidetone effect upon transmission performance rating for Sending (db),

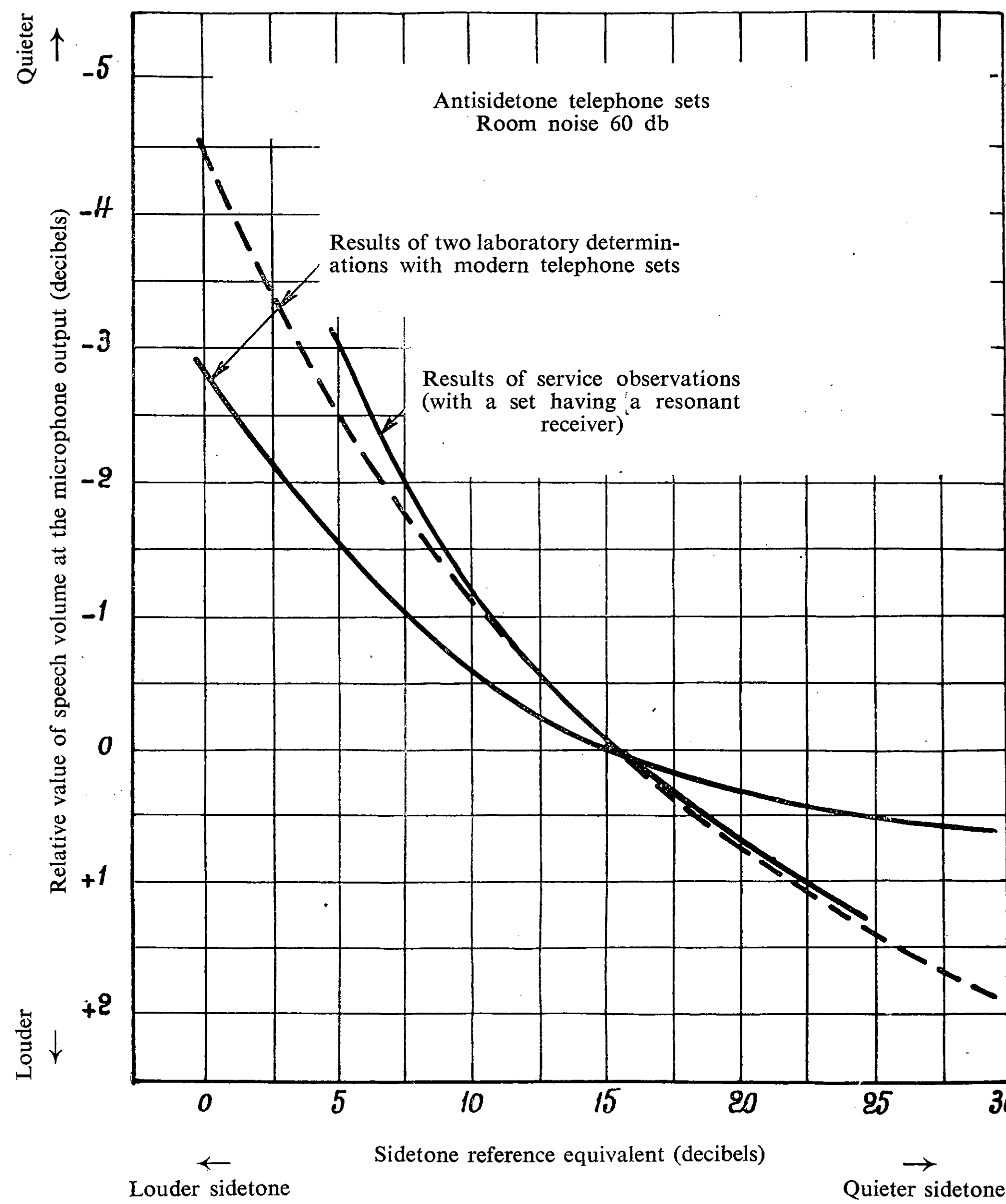


FIGURE 3. — *Variation of speech volume output as a function of sidetone reference equivalent*

(Information supplied by the Chile Telephone Company)

G_2 Sidetone effect upon transmission performance rating for Receiving (db),
H_1 Relative transmission performance rating for Sending (db),
H_2 Relative transmission performance rating for Receiving (db).

5. *Relative Transmission Performance Ratings for different subscriber's lines.*

Ratings are obtained relative to the chosen working standard circuit. Assessments are first made for variations only of the length of line. One item (the transmitter) has a resistance which varies with current; it is convenient therefore to choose values of current in the transmitter for which the calculations are made. Using the relation of 3.1 and knowledge of the supply voltage and resistances of other components in the feed circuit, the lengths of line to give each of these chosen values are calculated. Thus, for each value, items A and B can be entered up.

The main feed current effect is due to the variation of transmitter e.m.f. with feed current (3.2). A secondary effect, namely, the change of sending efficiency (for a constant transmitter e.m.f.) due to the change of transmitter resistance can be measured by electrical tests and, if appreciable, added to the main effect to complete Item C. The electrical tests can be made by substituting a resistor for the transmitter, injecting e.m.f. into the resistor and measuring p.d. at the connexion between subscriber's set and line, the circuit being closed by an impedance similar to that of a typical junction. The effect of any resistance value R of the resistor, relative to the working standard value R_o, is the difference in decibels between the ratios of p.d. to e.m.f. for R and for R_o (4-frequency rating).

It is generally sufficient to take the data from 3.3 directly for entry under Item D. This assumes negligible mis-match loss at the connexion between subscriber's set and line. In cases where this loss is not considered to be negligible, insertion loss tests with artificial line (4-frequency rating) can be made instead of calculation only of the line attenuation.

A value independent of the length and type of subscriber's line is used for the relative circuit ratings (Items E_1, E_2) since they would vary only on account of mis-match losses which are included, if necessary, in the rating of the line (Item D). In the present case changes of type of telephone and type of feeding bridge are not involved so these ratings are zero.

The relative circuit rating for side-tone (Item E_3) is the difference in decibels between the ratios of p.d. at the receiver terminals to e.m.f. in the transmitter

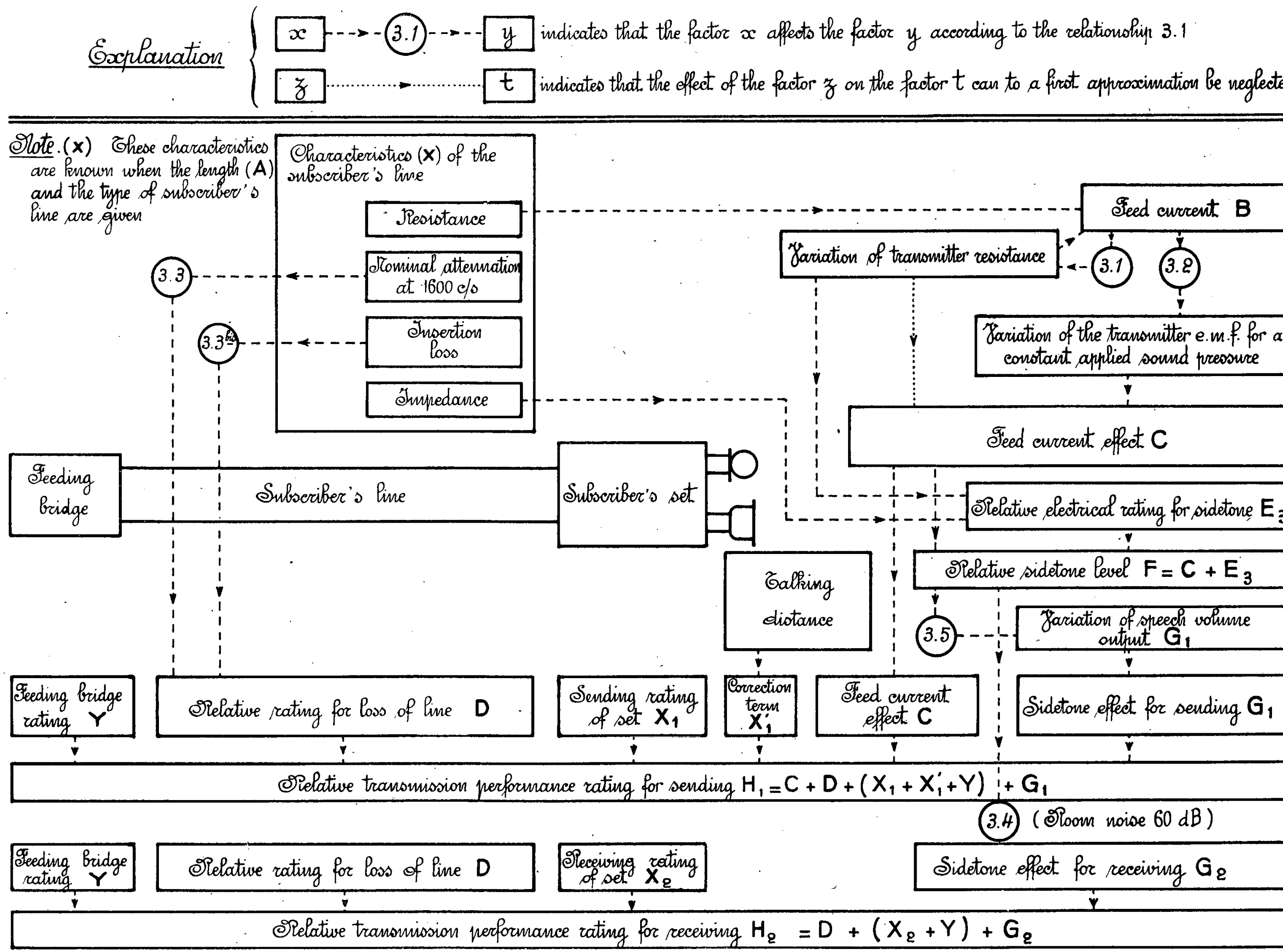


FIGURE 4. — *Table showing the factors which occur in calculations of relative transmission performance ratings*

for the length and type of subscriber's line concerned and the length and type of subscriber's line in the standard circuit. It can be measured by electrical test (4-frequency rating), the transmitter being replaced by resistors of appropriate values.

The relative side-tone level (Item F) is the sum of Items C and E_3.

The side-tone effect for sending (Item G_1) is obtained from Item F and the relation 3.5. Item G_2 is obtained from Item F and the relation 3.4.

The relative transmission performance rating for sending (H_1) is the sum of $C + D + E_1 + G_1$, and that for receiving (H_2) is the sum of $D + E_2 + G_2$. Thus, curves of relative sending and receiving ratings, relative to this working standard circuit can be plotted against subscriber's line length for each type of line considered, and the length of subscriber's line at which any given limit is reached can be read from the curves.

6. *Relative ratings when different types of telephone and/or feeding bridge are used.*

Only types of subscriber's set which have the same type of transmitter and receiver (i.e. to which the data listed in 3 above, are applicable) are considered. Any one combination of subscriber's set and feeding bridge which is different from the working standard can, in principle, have associated with it an arbitrarily chosen length and type of subscriber's line to constitute in effect a secondary standard. Relative ratings for the particular combination of subscriber's set and feeding bridge for different subscriber's lines, relative to this secondary standard, can be obtained (as described in Section 5 above). It remains to assess the relative rating of the secondary standard in terms of the working standard.

The procedure for so doing is again based on the items listed in Section 4. If the length of subscriber's line of the secondary standard is chosen to give the same feed current in the transmitter as in the working standard circuit, Items A, B and C are determined. Item D is found as mentioned in Section 5 above.

The relative circuit ratings, for sending and for receiving, each consist of two components, one for the subscriber's set and one for the feeding bridge. For purposes of description they are here designated, respectively, X_1 and Y for sending and X_2 and Y for receiving. Thus,

$$E_1 = X_1 + Y$$

and

$$E_2 = X_2 + Y$$

X_1 is the difference in decibels between the ratios of p.d. at the connexion between subscriber's set and line to e.m.f. in the transmitter for the secondary standard circuit. X_2 is the difference in decibels between the ratios of p.d. at the receiver terminals to e.m.f. in the line for the secondary standard and the working standard circuit.

These quantities can be measured by inserting a known e.m.f. and measuring the p.d. at the appropriate points in the circuit, the transmitter being replaced by a resistor of appropriate value (4-frequency rating). If the nominal speaking distance from the transmitter is different for the secondary standard and the working standard subscriber's sets, an allowance, based on tests with speech at the two distances, should be added to X_1. Y is the difference in decibels between the insertion loss of the bridge in the secondary standard and that in the working standard circuit, when the circuit is connected to an impedance chosen as being similar to that of a typical junction (4-frequency rating).

The relative circuit rating for side-tone (E_3) is the difference in decibels between the ratios of p.d. at the receiver terminals to e.m.f. in the transmitter for the secondary standard and the working standard circuit. It can also be measured by electrical test (4-frequency rating), the transmitter being replaced by a resistor of appropriate value.

The relative side-tone level (F) is found, as in Section 4, from the relationship $F = C$ and E_3, and the items G_1, G_2, H_1 and H_2 are also obtained exactly as in Section 5.

ANNEX 3

IMPAIRMENT DUE TO CIRCUIT NOISE

Noise on a telephone connexion appearing in the receiver of a subscriber's set reduces the efficacy with which the subscribers can converse. In the case of long-distance connexions, a large proportion of this noise arises from the trunk circuits. The effect of room noise present at the receiving end can also be taken into consideration (see Annex 4 below).

Articulation tests conducted at the C.C.I.F. Laboratory have confirmed that Table 1 below can be used provisionally in Europe to determine the transmission impairment (on a complete telephone connexion between subscribers) due to the presence (on this connexion) of various levels of noise arising from the trunk circuit and corresponding to a psophometric electromotive force measured (with the old psophometer for commercial telephone circuits specified by the C.C.I.F.) at the terminal trunk exchange of this trunk circuit closed with a 600 ohms non-reactive resistor, a matching transformer being inserted if necessary *. This table applies for the usual values of reference equivalent specified by the C.C.I.F. (see Volume IV of the *Green Book*, Section 1.1). It corresponds to the most interesting case to be considered, i.e. where a cable circuit is extended by an aerial open wire circuit exposed to induction from neighbouring power lines and where the total reference equivalent of the telephone connexion is in general high. In the values of impairment due to circuit noise given in the table below :

A. The effect has been included of a slight amount of circuit noise arising from the local subscriber's line of a large town,

* The XIVth Plenary Assembly of the C.C.I.F. specified a new psophometer weighting curve for commercial telephone circuits (see the *Green Book*, Volume IV, Section 3.2.3). The relationship between the psophometric voltage measured at the circuit terminals in the trunk exchange with the new psophometer and the psophometric electromotive force measured at the same point with the old psophometer will eventually be given by the C.C.I.F.

B. The presence is assumed at the receiving end of the telephone connexion of a room noise of average level corresponding to 50 dB above 2.10^{-4} dynes per square centimetre at 1,000 c/s.

TABLE 1

Psophometric electromotive force measured with the old C.C.I.F. psophometer for commercial telephone circuits and expressed in millivolts	Transmission impairment due to circuit noise and expressed in decibels
$<$ 2.5	0
2.5 to 4.0	1
4.0 to 5.5	2
5.5 to 7.0	3
7.0 to 8.5	4
$>$ 8.5	5

Note 1. — Table 1 above gives the " transmission impairment " of a trunk telephone connexion due to the presence of a certain level of circuit noise, this level being determined by a reading on the old C.C.I.F. psophometer for commercial telephone circuits, connected at the end of the trunk circuit. This " transmission impairment " is the additional loss it is necessary to insert in the reference circuit (without noise) to obtain the same results (from the point of view of the telephone service) in both cases. In general, the question of the practical use of " transmission impairments due to circuit noise " is treated in an analogous manner to that of " transmission impairments due to limitation of the band of frequencies effectively transmitted " (see below), i.e. they are used in practice in steps of 1 decibel. As it is usually impossible to predict the level of circuit noise which will exist on a circuit until it is set up, the magnitude of existing circuit noises is first measured with the psophometer and a " transmission impairment due to circuit noise " (expressed in decibels) is allocated to the circuit, according to the table above.

Note 2. — When a circuit is used in transit service, it must of course be satisfactory from the points of view of echo, of crosstalk and of stability (singing) and at the same time it must not contribute more than is permissible to the total reference equivalent of the whole telephone connexion. If the type of international circuit it is proposed to use for a certain connexion provides a " transmission impairment due to limitation of the band of frequencies effectively transmitted " or a " transmission impairment due to circuit noise " greater than 0 decibels, the value of these impairments must be deducted from the desired value of reference equivalent to obtain the loss (at 800 c/s) at which the international circuit considered must be set up for transit service. It is this last value of loss at 800 c/s which must be considered to determine whether this particular type of international circuit will be satisfactory from the points of view of echo, of crosstalk and of singing, as indicated in Volume III of the *Green Book*, First Part, Section **1**.2

In the United States, until the last few years, circuit noise was measured with a psophometer termed the " American Npise Meter " connected to the end of the trunk circuit closed in 600 ohms (the impedances of trunk circuits are always made 600 ohms by terminating transformers). This psophometer was furnished with an " A Network ") (conforming with the " Table of Weighting Factors " of the old C.C.I.F. psophometer for commercial telephone circuits), followed by a " B Network " (which reproduces the distortions of typical lines and equipment, encountered in the Bell System, between the terminals of the trunk circuit and the telephone receiver of the listening subscriber). Futthermore, in the United States, the psophometric readings were expressed in decibels with respect to the reference noise of 10^{-12} watt at 1,000 c/s. A "Noise Rating " and a " Noise Transmission Impairment " given respectively by the first and second columns of Table 2 are then allocated to a trunk circuit on which a certain amount of circuit noise (third column) has been measured.

TABLE 2

Noise Rating	Noise Transmission Impairment in decibels	Circuit noise measured at the end of the trunk circuit with the old American psophometer furnished with Networks A + B and expressed in decibels above the " Reference Noise " (10^{-12} watt at 1 000 c/s)
N 0	0	0 - 29
N 1	1	29 - 32
N 2	2	32 - 35
N 3	3	35 - 38
N 4	4	38 - 40
N 5	5	40 - 42
N 6	6	42 - 43
N 7	7	> 43

Note. — Psophometric readings which fall at the limits of the steps given in Table 2 above should correspond to the Noise Rating immediately below; e.g. a reading on the psophometer of 29 decibels would correspond to the rating N 0—a reading of 32 decibels to the rating N 1, etc.... As an example of the application of this Table 2, a psophometric reading of 33 decibels made at the end of a trunk circuit would attribute to the circuit the Noise Rating N 2 and a Noise Transmission Impairment equal to 2 decibels.

During a transition period, the relationship given in Table 2 between the impairment due to circuit noise and the circuit noise measured at the end of the trunk circuit has been used both for noise measured with the old psophometer and for noise measured with the new American psophometer for commercial telephone circuits. At the same time the values of noise measured with the new American psophometer furnished with the filter network for measurements at the end of a trunk circuit were expressed in " adjusted decibels " (in short dba). For measurements of circuit noise made at the end of a trunk circuit with the American psophometer furnished with the old filter network, the number of " adjusted decibels " is numerically equal to the number of decibels above " reference noise " (10^{-12} watt at 1,000 c/s). For measurements made at the end of a circuit with the new American psophometer, the number of " adjusted decibels " is equal to the number of decibels above the same reference noise, less 5 decibels. As explained in *Engineering Report* No. 45 of the Joint Subcommittee of the Edison Electric Institute and the Bell System (of which the *French translation* forms the subject of the document " C.C.I.F. 1947/1948—1st Study Group—Document No. 2 "), noise of which the values expressed as " adjusted decibels " are equal have approximately the same disturbing effect, and consequently produce almost the same " Noise Transmission Impairment ". Thus the relationship between the transmission impairment due to circuit noise and the value of the circuit noise, expressed in " adjusted decibels ", which has been employed in the United States of America during this transition period was the same as the relationship between the transmission impairment due to circuit noise and the circuit noise, expressed in decibels above " reference noise ", which was previously used and which is also given by Table 2 above.

As a result of recent tests concerning the relationship between the value of circuit noise (expressed in " adjusted decibels ") and the transmission impair-

TABLE 2 *bis*

Noise Rating	Noise Transmission Impairment in decibels (in short NTI)	Circuit noise measured at the end of a trunk circuit with the new American psophometer * and expressed in " adjusted decibels " (dba)
N 0	0	0 - 29
N 0.5	0.5	29.1 - 32
N 1	1	32.1 - 34
N 2	2	34.1 - 36
N 3	3	36.1 - 37.5
N 4	4	37.6 - 39
N 5	5	39.1 - 40
N 6	6	40.1 - 41.5
N 7	7	41.6 - 42.5
N 8	8	42.6 - 43.5
N 9	9	43.6 - 44.5
N 10	10	$>$44.5

* The frequency characteristic of the filter network of the new American psophometer is identical to that of the new C.C.I.F. psophometer for commercial telephone circuits (see the *Green Book*, Volume IV, Section 3.2.3), except that in the United States the reference frequency (corresponding by convention with the weighting coefficient, 0 decibel) is 1,000 c/s instead of 800 c/s.

Note. — In this table a new " Noise Rating " N 0.5 has been introduced ; this appeared desirable to take account of the fact that the interval of values of circuit noise (in " adjusted decibels ") between the values corresponding to transmission impairments of 0 and 1 decibel is larger than was case in the old Table 2.

ment due to circuit noise, on connexions between telephone sets of the American Type F 1 A, Table 2 *bis* is at present used in the United States stead of Table 2.

In the United States, when a trunk circuit is to be set up, psophometric measurements are made at each end; the greater of the two readings thus obtained serves to determine the " Noise Rating " to be allocated to this circuit. To facilitate the work of maintenance and testing staff, the card giving the summary specification of this circuit is marked with the " Noise Rating " as well as the " Noise Transmission Impairment ". (Such a " summary specification " gives the pair number which applies to the conductors in the cable, or the positions of the pole-arms carrying the conductors of the open overhead wires—the length of repeater sections—the type and gain of each repeater—the settings of regulating devices, etc.... in short, all the circuit particulars which are important for testing and maintenance.)

When two trunk circuits of different types are permanently connected together, the noise on this combined connexion is measured with the psophometer, the " Noise Rating " to be allocated to the connexion is thus determined directly rather than by having recourse to calculation.

It is nevertheless often desirable to calculate the " Noise Transmission Impairment " for the combination of a trunk circuit comprising two or more trunk circuits ; as an example, in the Bell System, the method followed to obtain the resultant noise consists essentially : (1) to refer to a single point (for example the end of the trunk connexion) the various noise components considered individually which exist on the various circuits which make up the connexions ; (2) to then add according to a quadratic law (square root of the sum of the squares), the voltages corresponding to these various noise components referred to the single point.

Take, as an example, two circuits AB and BC each having a loss of 9 decibels and interconnected by means of a cord circuit repeater at B giving a gain of 6 decibels. Suppose that a measurement of 34 " adjusted decibels " is given on the new American psophometer connected at B on the end of the circuit AB considered alone, and that 34 " adjusted decibels " is also measured on this psophometer at C on the end of the circuit BC considered alone.

It is necessary to first refer to the point C (the end of the connexion) the noise of 34 " adjusted decibels " measured at B at the end of the circuit AB, and for this to allow for the gain (+6 decibels) of the cord circuit repeater at B and the loss (−9 decibels) of the circuit BC ; the noise of the circuit AB referred to the point C is thus obtained : $34+6-9 = 31$ " adjusted decibels ".

It remains to add, at the point C, according to a quadratic law, the two noise components of 34 and 31 " adjusted decibels ".

The same number of decibels is obtained whether the voltages corresponding to the various noises are added according to a quadratic law or the corresponding power ratios (relative to an arbitrary reference power) are added according to a linear law.

Now power ratios of 2,512 and 1,259 correspond respectively to the numbers 34 and 31 decibels. The result of adding the voltages according to a quadratic law corresponds to a reading of 35.76 " adjusted decibels " on the American psophometer, so that we have $2{,}512+1{,}259 = 3{,}771$ and $10 \log_{10} 3{,}771 = 35.76$ decibels.

Consequently the two noise levels of 34 " adjusted decibels " measured at the ends of the circuits AB and BC considered separately produce a resultant noise of 35.76 " adjusted decibels " at the end of the connexion AB+BC *. Then Table 2 bis above gives, for such a resultant noise of 35.76 " adjusted decibels ", a " Noise Transmission Impairment " of 2 decibels to be added to the loss at 1,000 c/s of the combination of the two interconnected circuits.

Because of the methods followed in allocating nominal gains to the various repeaters, it has in general been found impractical to take account of noise by deducting the " Noise Transmission Impairment " from the desired loss of the circuit (as indicated in the notes relating to " Transmission Impairment due to limitation of the band of frequencies effectively transmitted " (see the *Green Book*, Vol. IV, Section VI.3.2, pages 138-143)) ; the reason for this is that the signal to noise ratio is, in general, independent of the loss of the circuit, in other words, an improvement in loss of a circuit tends to increase the " Noise Transmission Impairment " allocated to that circuit and this would largely compensate for the reduction in the value of loss.

In Europe, where circuit noise is measured by means of a psophometer for commercial telephone circuits in accordance with the C.C.I.F. recommendation and connected across a 600 ohms resistor terminating the trunk circuit (a matching transformer being inserted if necessary), it is recommended that the procedure for determining the " Transmission Impairment due to circuit noise " should be similar to that used in the United States and described above but, provisionally, Table 1 above should be used which gives the " Transmission Impairment due to circuit noise " (in decibels) as a function of the " psophometric electromotive force measured with the old C.C.I.F. psophometer for commercial telephone circuits " (in millivolts).

Combination of a " Transmission Impairment due to limitation of the band of frequencies effectively transmitted " with a " Transmission Impairment due to circuit noise ". — The methods described above give the practice followed in the United States for determining transmission impairments due either to circuit noise or to limitation of the band of frequencies effectively transmitted (either for a single trunk circuit or for a connexion comprising several interconnected trunk circuits). To obtain the " Effective Transmission Loss " of such a single circuit or of such a combined connexion, the usual practice consists of adding directly : (1) the loss at 1,000 c/s ; (2) the transmission impairment due to limitation of the band of frequencies effectively transmitted ; (3) the transmission impairment due to circuit noise, determined as indicated above.

* In practice, in the Bell System, curves or tables are used which give directly the results of such an addition of voltages according to a quadratic law.

ANNEX 4

TRANSMISSION IMPAIRMENT DUE TO ROOM NOISE

The quantitative effect of room noise on telephone transmission quality has been determined in the United States by laboratory judgement tests and has been expressed as curves giving for each type of telephone set, the room noise impairment (in short — RNI) as a function of the level of the room noise, measured with the American Sound Level Meter (see Annex 17 of this volume).

Zero impairment corresponds to a room noise level of 50 db measured with the American Sound Level Meter (the reference zero corresponds to an acoustic pressure of 2.10^{-4} dyne/cm^2 of a free progressive wave at 1,000 c/s).

Curve 1, " 1946 F 1 A " of Figure 1 is the curve at present used by the Bell System in the United States giving the room noise impairment as a function of the room noise level, measured with the American Sound Level Meter. Curve 2, " 1931 A.S.T." used previously, corresponds to antisidetone telephone sets use in 1931 in the Bell System.

In the Bell System the room noise impairment expressed in decibels is determined from this curve corresponding to the mean value of room noise determined from a series of measurements made in the area served by a given telephone exchange. In planning the subscribers' lines in this area, this number of decibels is simply added to the other component losses and it is checked that the sum obtained is equal to the specified overall loss.

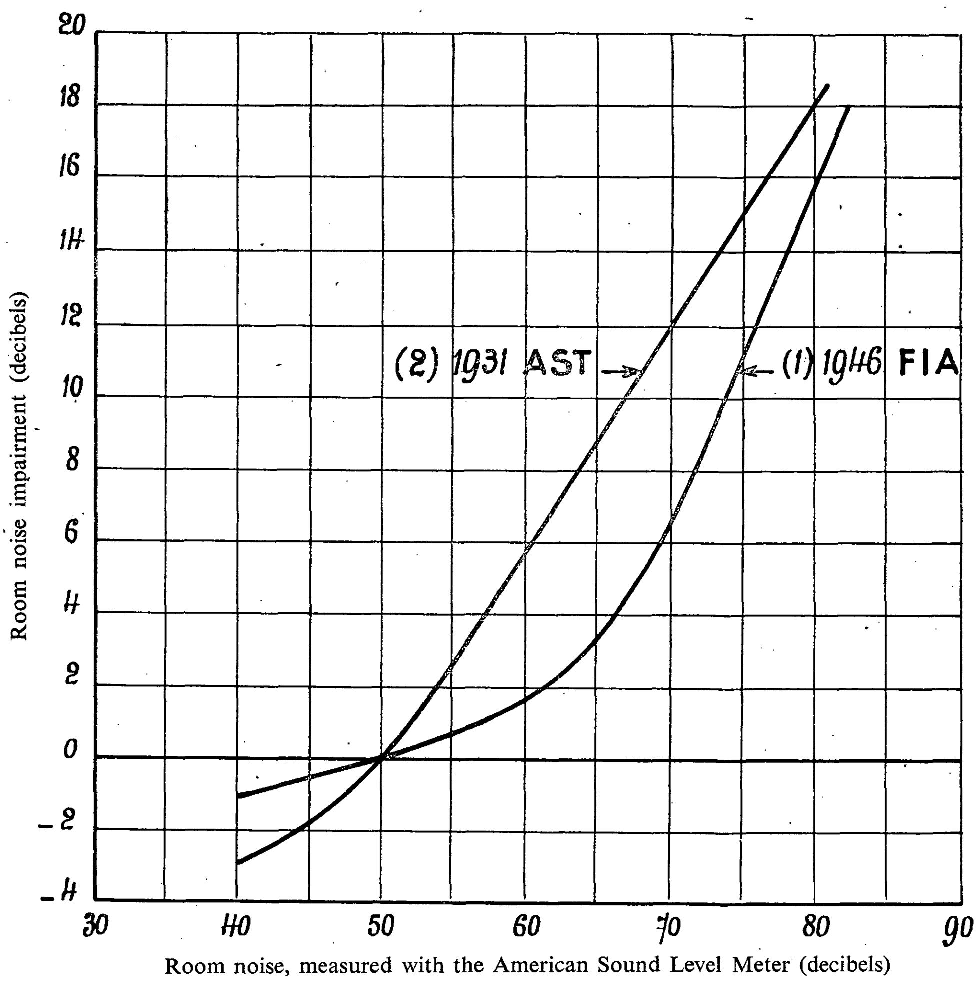


FIGURE 1. — *Curves used in the United States to give room noise impairment; comparison of the new curve obtained in 1946 with the old 1931 curve.*

ANNEX 5

ABSOLUTE CALIBRATION OF REFERENCE SYSTEMS AT THE C.C.I.F. LABORATORY

I. ABSOLUTE CALIBRATION PROCEDURE FOR THE S.F.E.R.T.

II. ABSOLUTE CALIBRATION PROCEDURE FOR THE A.R.A.E.N.

I. ABSOLUTE CALIBRATION PROCEDURE FOR THE S.F.E.R.T. AT THE C.C.I.F. LABORATORY

1. General principles of electro-acoustic calibration

Each condenser microphone used with the S.F.E.R.T. undergoes calibration in a closed chamber filled with hydrogen one wall of which is the microphone diaphram (see Figure 2). An acoustical wave is produced thermally, inside this chamber, by use of a thermophone.

The receivers are calibrated on an artificial ear, specified for use only with S.F.E.R.T., and which comprises a S.F.E.R.T. microphone and a closed coupler 6 cm^3 in volume filled with hydrogen at atmospheric pressure.

The C.C.I.F. Laboratory limits itself to checking the stability with time of the microphones and receivers.

The complete absolute calibration procedure of the S.F.E.R.T. is carried out, in principle, only twice a year. Nevertheless when a large number of reference equivalent determinations have been made, it is desirable to check the absolute calibration of the standard condenser microphone and the standard receiver used for the voice-ear tests and which in principle remain the same ones for a fairly long period. In general measurement of this microphone with a single thermophone and of the receiver with a single microphone is sufficient to check the constancy of the standard items.

The complete calibration procedure of the S.F.E.R.T. consists of 5 parts :

1. Calibration of the condenser microphones.
2. Calibration of the moving coil receivers.
3. Calibration of the sending system amplifier.
4. Calibration of the receiving system amplifier.
5. Noise measurement.

The method to be applied for calculating and drawing the calibration curves is shown in the *Black Book* (pp. 22 to 48) (specification of the S.F.E.R.T., supplied by the American Telephone and Telegraph Co.) held at the C.C.I.F. Laboratory. Furthermore, to determine the " normal adjustment " from the results of the calibration procedure, the rules given in the American Memorandum MM 2 768 of 16th March, 1931 are followed. This memorandum also gives various data on the absolute calibrations.

2. Calibration of condenser microphones by the thermophone method

2.1. General remarks on the theory of the thermophone

The European Master Reference System (S.F.E.R.T.) is defined for each frequency by a certain ratio of volts per dyne/cm^2 for the sending system and of dynes/cm^2 per volt for the receiving system.

Determination of these ratios is conducted by means of a special calibration of which the technique is based upon use of the thermophone in a closed chamber. The thermophone is a hot wire receiver which acts as an electro-acoustic transducer by virtue of thermal energy. It generally consists of an extremely thin metal ribbon (two stretched gold leaves 1/10th of a micron in thickness) of which the mass is sufficiently small to have a very small thermal capacity.

When this ribbon, of resistance R, carries an alternating current of pulsatance $\omega = 2\pi f$, by the Joule effect it undergoes temperature variations which are of the form :

$$T = kRI^2 \sin^2 \omega t, \tag{1}$$

which can be re-written :

$$T = kRI^2 \left(\frac{1 - \cos 2\omega t}{2}\right) \tag{2}$$

The gas particles close to the gold leaves are periodically varied in temperature by conduction. In expanding and contracting they form a piston of gas and under the action of this piston the mass of gas filling the remainder of the cavity is also periodically compressed and rarified. The fluctuations in gas pressure give rise to an acoustic pressure of the same frequency on the microphone diaphragm.

In practice it is necessary to fill this cavity with a light gas (hydrogen) in order that the essential dimensions of the microphone cavity shall remain small compared with the wavelength at the measuring frequency ; in this way the equivalent dimensions of this cavity are reduced by about three or four times.

In equation (2) above a double frequency term appears ; the temperature variation thus produces in the atmosphere close to the thermophone an acoustic wave of double the frequency of the electric current. If it is desired to remove this inconvenience, it is necessary to have recourse to an arrangement analogous to that used in the ordinary telephone receiver and to superpose a direct current I_c on the alternating current I_a. This polarising direct current plays the part of the unidirectional magnetic field of the ordinary receiver or of the magnitising current in the case of the electro-magnetic receiver. Under these new conditions the expression for T becomes :

$$T = kR\,(I_c + I_a \sin \omega t)^2$$
$$= kRI_c^2 + kR\,I_a^2 - kR\,\frac{I_a^2}{2} \cos 2\,\omega t + 2\,kR\,I_c\,I_a \sin \omega t \qquad (2)$$

It can be shown that if I_c and I_a are appropriately chosen the acoustical output can be greater than in the first case (without direct current). As I_a is always small compared with I_c, the second expression only is of interest by virtue of the fact that it multiplies the sensitivity in the ratio $\frac{I_a}{I_c}$ and retains the fundamental frequency by reducing the double frequency term to a relatively small and negligible value compared with that of the fundamental. In practice I_c is equal to 0.5 A and I_a is of the order of a few milliamps.

Although the thermophone is an instrument of poor precision when used under undefined acoustical conditions, the theory shows on the other hand and experience confirms that, in a chamber of dimensions very small compared with the wavelength of the sound in question, the pressure produced by the thermal effect is calculable to a very good approximation. This theory assumes, as stated, that the thermal element is located in a chamber whose linear dimensions are small in comparison with the wavelength of the sound waves, and in which the variations in pressure are transmitted instantaneously from the thermal element to all walls of the chamber which operates as a true manometer. The microphone being calibrated is on the other hand one of the walls of this chamber. It is thus acted upon by a precisely defined pressure. In order to render the chamber acoustically as small as possible and to remove the resonant frequency to beyond the useful range it is filled with hydrogen, a gas in which the speed of propagation of sound is, of course, four times as high as in ordinary air.

Operation of the thermophone.

The operation of the thermophone is studied theoretically by considering on the one hand the propagation of heat in the gaseous medium and the thermodynamic phenomena which accompany it and, on the other, the production of heat by the gold leaves and the thermodynamic equilibrium which is set up in the coupling chamber.

The calculations which lead to the expression given above and yield the curve of pressure, p, in the thermophone chamber have been derived from the following principal hypotheses :

(a) the thermophone is situated for calibration of microphones in a closed chamber of small volume,

(b) the walls of the chamber are assumed perfectly rigid,

(c) the linear dimensions of the chamber are small in comparison with the wavelength of the sound,

(d) the wavelength of the thermal diffusion wave arising from the thermophone heat source is small in comparison with the distance between the gold leaves and the walls,

(e) the heat loss from the thermophone by radiation and conduction is negligible.

Table 1 gives the value of p as a function of frequency for the standard thermophones at a direct current of 0.5 A.

$$p = \frac{239\, RI\, [1-3.95\, f^{-\frac{1}{2}}+7.80\, f^{-1}]^{\frac{1}{2}}\, [1-0.543\, f^{-\frac{1}{2}}+0.147\, f^{-1}]^{\frac{1}{2}}}{f\, [0.0053\, f+0.331\, f^{\frac{1}{2}}+10{,}33]^{\frac{1}{2}}}$$

Figure 1 gives the pressure in decibels produced by the thermophone relative to 1 500 dynes/cm.²

$$N = 20 \log_{10} \frac{p}{1{,}500}$$

TABLE 1

Pressure developed by the thermophone in the normal S.F.E.R.T. chamber

Wente formula. Numerical application

f.	P.	$\log_{10}P$.	$20 \times \log_{10}P$ or decibels relative to 1 dyne/cm²
25	1922.0	3.28635	65.72700
49	1021.0	3.00887	60.17740
81	635.8	2.80333	56.06660
100	517.1	2.71359	54.27180
225	228.7	2.35927	47.18540
400	123.9	2.09307	41.86140
900	49.9	1.69787	33.95740
1,600	25.4	1.40543	28.10860
2,500	14.7	1.16668	23.33360
3,600	9.25	0.96636	19.32720
4,900	6.08	0.79436	15.88720
6,400	4.38	0.64136	12.82720
8,100	3.35	0.50543	10.10860
10,000	2.41	0.38232	7.84640
16,900	1.17	0.07063	1.41060

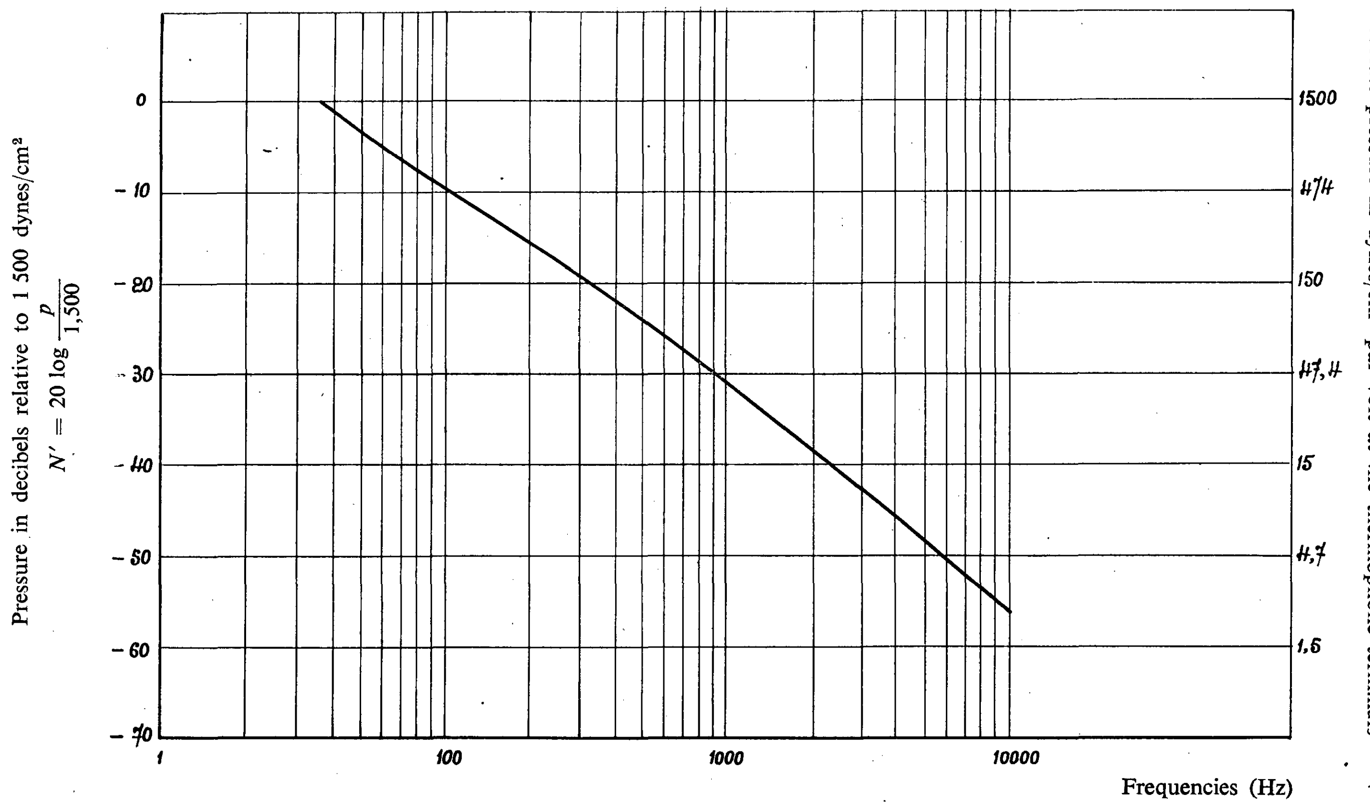


FIGURE 1. — *Characteristic of standard thermophones for a current of 0.5 ampere*

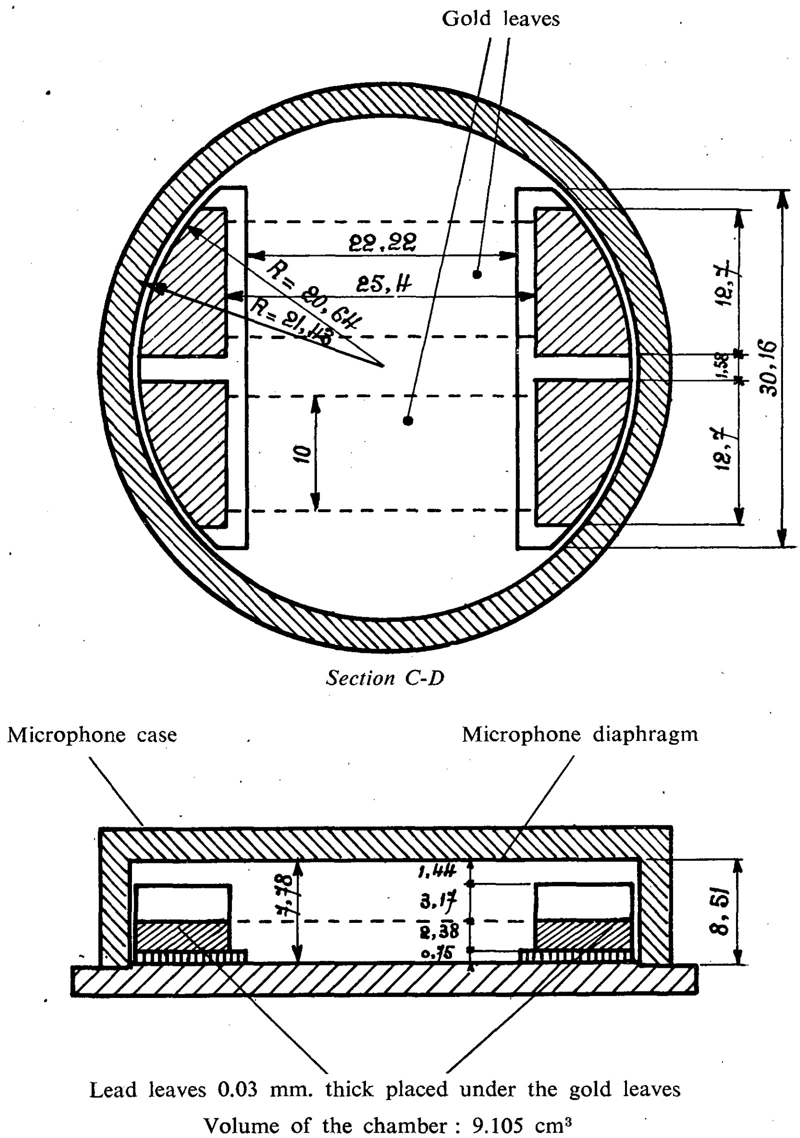


FIGURE 2. — *Schematic diagram of the thermophone*

3. Calibration of a condenser microphone

To calibrate a S.F.E.R.T. condenser microphone it is coupled acoustically to the mounting of the thermophone by means of a special coupler (see Figure 3).

The annular base of the microphone is lightly smeared with vaseline to ensure a good seal to the coupling chamber. The effective volume of the calibration chamber is 9.105 cm^3.

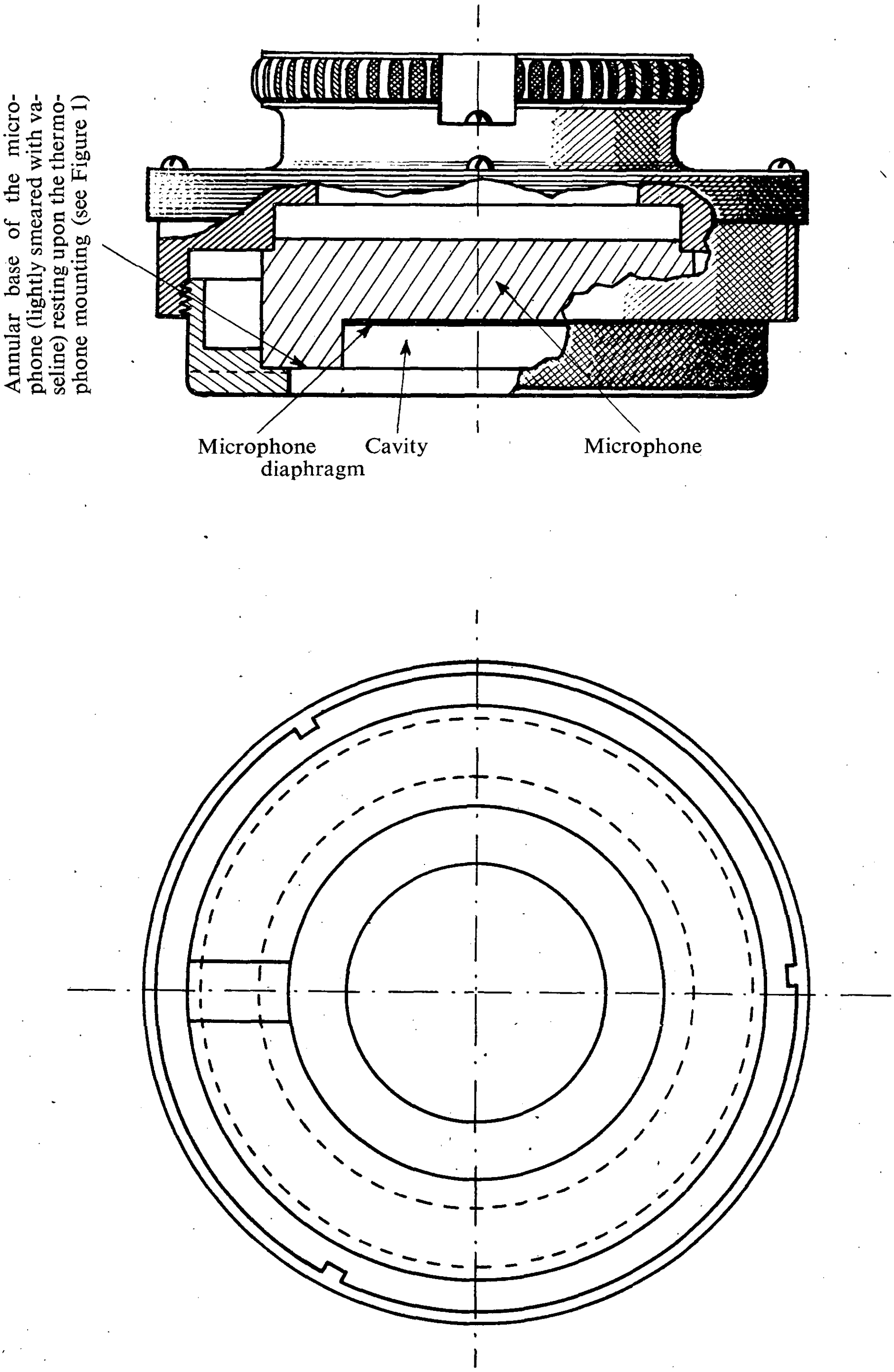


FIGURE 3. — *Special coupler to allow the S.F.E.R.T. condenser microphone to be placed on the thermophone mounting*

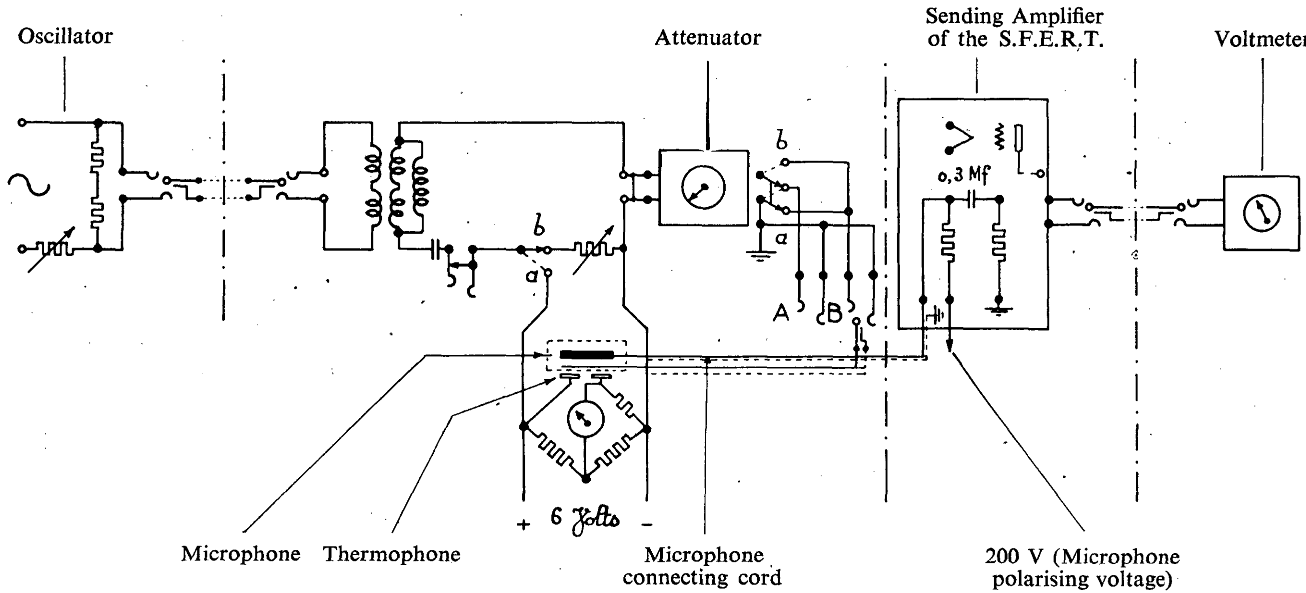


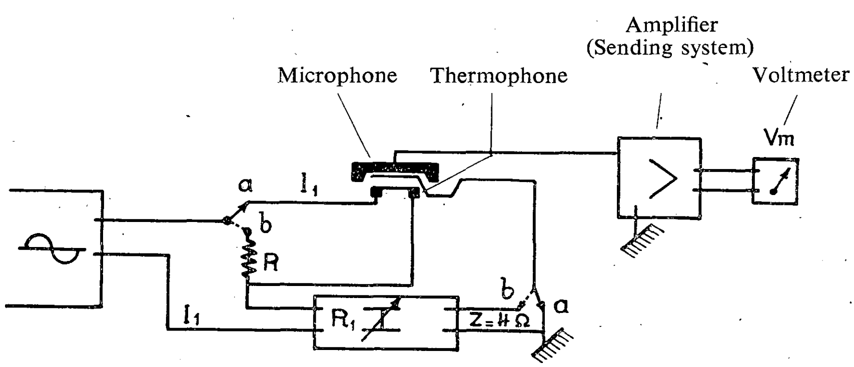
FIGURE 4. — *Absolute calibration of a microphone*

The checking and adjustment of the hydrogen filling of the coupling chamber is carried out by means of special devices fitted with pressure gauges. Before filling the coupling chamber the resistance of the gold leaves is measured in air with direct current at 0.5 amperes. This value is obtained by controls on the thermophone bay. During the filling the resistance value of the gold leaves is checked and it decreases progressively to a constant value about 10 % smaller than the value in air. When this limiting value is reached the hydrogen will have completely replaced the air which was in the thermophone chamber.

The diagram of electrical connections is given in Figure 4. This figure shows the thermophone connected in one arm of a Wheatstone bridge which is continuously supplied from a 6 volt source with a current adjusted to 0.5 amperes.

3.1 PRINCIPLE OF CALIBRATING A CONDENSER MICROPHONE BY THE THERMOPHONE METHOD

Schematic diagram of the measurement



Let :

E_T be the microphone sensitivity (in volts per dyne/cm²),
I_1 be the alternating current through the thermophone and at the input of the attenuator,
P be the pressure produced by the thermophone per ohm per ampere of alternating current through it,
V_M be the voltage at the voltmeter terminals,
A be the voltage gain of the amplifier,
R_1 be the input impedance of the attenuator (R_1 = 15.1 ohms),
N be the attenuator setting in decibels,
R be the resistance of the thermophone,

1. We have, on putting the switches in the " a " position, the value of voltage V_M read on the voltmeter, which is equal to :

$$I_1 \times R \times P \times E_T \times A$$

2. And, on putting the switches in the " b " position and adjusting the attenuator to the value N which produces the same reading on the voltmeter (the thermophone is replaced by an equivalent resistor) :

$$V_M = I_1 R_1 \times 10^{-0.05\,N} \times A$$

(it is known that : $N = 20 \log \frac{U_1}{U_2}$ whence : $U_2 = U_1 \times 10^{-0.05\,N}$)

By equating the two expressions for V_M in both positions 1 and 2, we have

$$E_T = \frac{R_1 \times 10^{\,-0.05N}}{R \times P} \tag{1}$$

The value of P, at each frequency, is given by the Wente formula

Putting : $$N' = 20 \log \frac{P_2}{P},$$

P_2 being an arbitrarily chosen number taken here as 1,500 dynes/cm^2.

Therefore $$\frac{P_2}{P} = 10^{-0.05N'} \text{ and } P = P_2 \times 10^{-0.05N'}$$

The curve of Figure 1 gives the values of N' such that, for each frequency, we have the relationship :

$$N' = \log \frac{1{,}500}{P}$$

Replacing P by this value in equation (1).

The expression for capital E_T becomes :

$$E_T = \frac{R_1 \times 10^{0.05\,(N'-N)}}{R \times P_2}$$

Putting : $$\frac{R_1}{R \times P_2} = M$$

with R_1 (15.1 ohms) and P_2 (1,500 dynes/cm^2) which are fixed values, M therefore depends upon the resistance of the thermophone used.

We can write :

$$N'' = 20 \log \frac{M}{K} \quad \text{or} \quad \frac{M}{K} = 10^{0.05N''}$$

K being a number arbitrarily chosen.

Making $K = 1$: therefore $M = 10^{0.05\,N''}$

Finally the expression for the sensitivity of the microphone becomes

$$E_T = 10^{0.05[N' - (N + N'')]},$$

Putting $N' - (N + N'') = N_c$ the sensitivity of the microphone is $10^{0.05 N_c}$, in which, N is the value of attenuation required to obtain the original deflection of the voltmeter.

N' depends on the pressure produced by the thermophone. This pressure is the same for the four S.F.E.R.T. thermophones but depends on frequency (see the curve of Figure 1).

N'' depends on the thermophone resistance (N'' is independent of frequency but varies with each thermophone) and is calculated for each of the four thermophones before the calibration measurements.

3.2 Interpretation of the results

The sensitivity of a S.F.E.R.T. microphone is expressed in decibels relative to 1 volt per dyne/cm² and is the mean value found with four thermophones which appear to be in working order.

A microphone is considered satisfactory if its sensitivity (arithmetic mean of the values obtained for 20 frequencies between 200 and 3,600 c/s) is within the limits 51 db ± 2 db below 1 volt per dyne/cm² and when the value at 5,000 c/s is more than 1 db and less than 4 db from the value measured at 1,000 c/s.

The C.C.I.F. Laboratory possesses a literature giving the rules to be followed in carrying out such calibrations. These rules are contained in Memorandum MM 2768 of March 1931.

As an example Table 2 below gives the values of sensitivity of a condenser microphone measured by the thermophone method.

4. Calibration of an electrodynamic receiver

After completing the calibration of the 4 microphones with each of the 4 thermophones, the procedure is completed by the calibration of the 4 moving coil receivers with the aid of each of the condenser microphones.

The S.F.E.R.T. receiver is coupled acoustically to a S.F.E.R.T. microphone by means of a special coupler (see Figure 6).

In this way an artificial ear is obtained specified solely for the calibration of these receivers.

The total volume of the calibration chamber is about 29.8 cm³ made up as follows :

12.3 cm³ representing the volume of the microphone cavity,
6.0 cm³ for the volume of the coupler,
11.5 cm³ for the volume contained between the plane of the base of the receiver and the receiver diaphragm.

This chamber is filled with hydrogen at atmospheric pressure.

The thermophone is not used in these measurements, and is covered up by the microphone coupler carrying the block which replaces the microphone.

TABLE 2

Calibration of S.F.E.R.T. condenser microphone by the thermophone method

Thermophone No. 2 — Resistance 3.82 ohms — Microphone No. 3978

Hertz	N' (db)	N'' (db)	$N'+N''$ (db)	N (db)	Sensitivity of microphone (db relative to 1 volt per dyne/cm²) N_c
100	+ 9.2	—51.6	−42.4	7.2	−49.6
200	+15.0	»	−36.6	13.7	−50.3
300	+18.7	»	−32.9	17.5	−50.4
400	+21.6	»	−30.0	20.3	−50.3
500	+23.5	»	−28.1	22.7	−50.8
600	+25.2	»	−26.4	24.3	−50.7
700	+26.7	»	−24.9	26.3	−50.9
800	+28.0	»	−23.6	27.3	−50.9
900	+29.6	»	−22.0	28.7	−50.7
1,000	+30.4	»	−21.2	29.2	**−50.4**
1,100	+31.4	»	−20.2	30.3	−50.5
1,200	+32.4	»	−19.2	31.3	−50.5
1,300	+33.4	»	−18.2	32.0	−50.2
1,500	+34.4	»	−17.2	33.6	−50.8
1,800	+36.5	»	−15.1	35.2	−50.3
2,100	+38.0	»	−13.6	37.0	−50.6
2,400	+39.4	»	−12.2	38.4	−50.6
2,700	+41.0	»	−10.6	39.3	−49.9
3,000	+42.0	»	− 9.6	40.3	−49.9
3,300	+43.0	»	− 8.6	41.3	−49.3
3,600	+44.2	»	− 7.4	42.0	−49.4
4,000	+45.0	»	− 6.6	42.9	−49.5
4,500	+46.4	»	− 5.2	44.1	−49.3
5,000	+47.7	»	− 3.9	45.5	**−49.4**
5,500	+48.6	»	− 3.0	46.9	−49.9
6,000	+49.5	»	− 2.1	48.8	−50.9
7,000	+51.0	»	− 0.6	53.0	−52.4
8,000	+52.0	»	+ 1.0	56.9	−55.9
10,000	+55.9	»	+ 4.3	63.9	−59.6
Mean sensitivity (in the frequency band 200 to 3,600 c/s).					—50.4 db relative to 1 volt per dyne/cm²

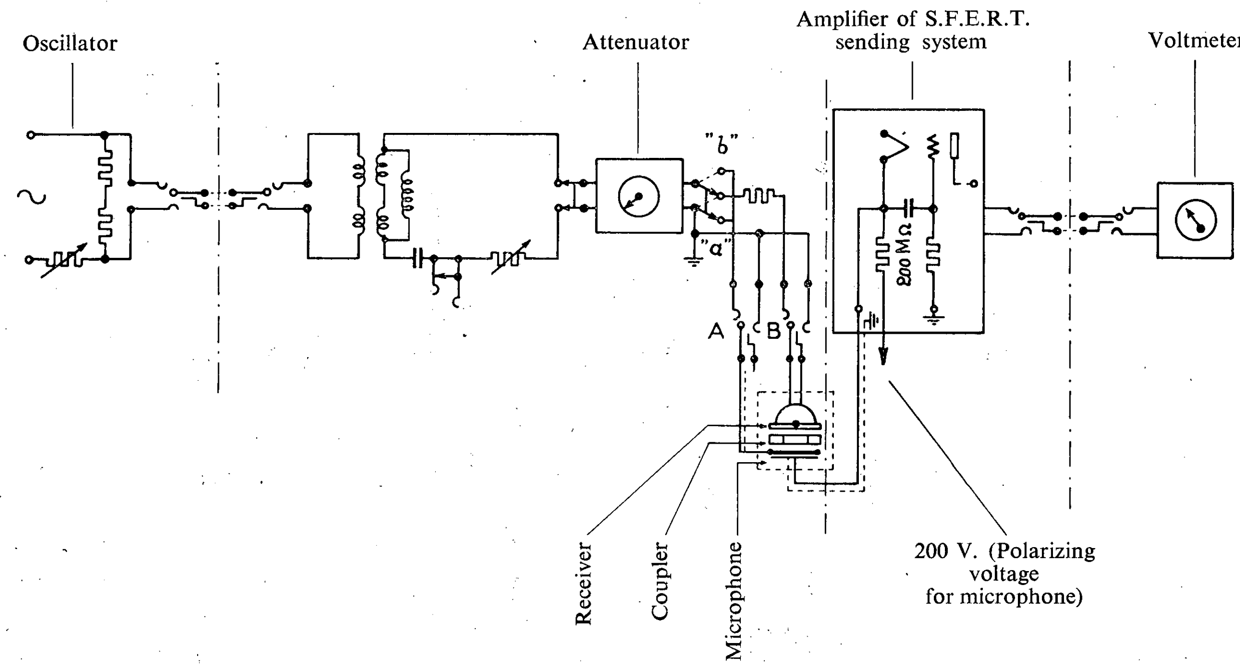


FIGURE 5. — *Absolute calibration of a receiver*

Nevertheless it is necessary to make sure, after putting the block into place, that the assembly is free from gas leaks, not only because any slight lack of sealing of the thermophone chamber will adversely affect the receiver calibration, but also because this precaution facilitates final checking for possible gas leaks when the receiver is place into position for calibration.

The schematic diagram of electrical connexions for such measurements is given in Figure 5.

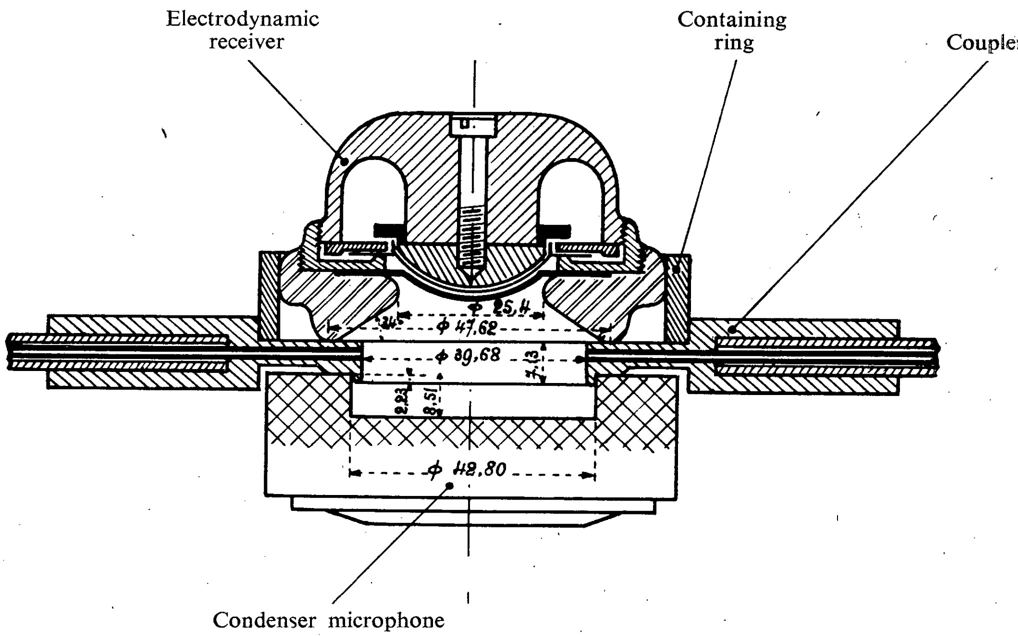
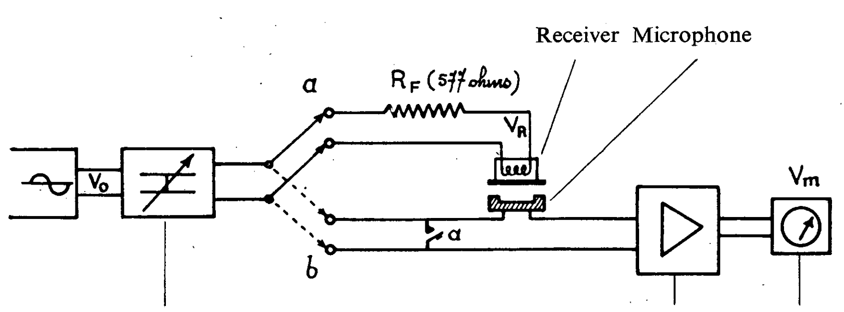


FIGURE 6. — *Artificial ear specified for calibrating the S.F.E.R.T. receivers*
Calibration chamber for an electrodynamic receiver

4.1 PRINCIPLE OF CALIBRATING A RECEIVER

Diagram showing the measurement principle



Let :

V_M be the voltage at the voltmeter terminals,
V_o be the voltage at the input terminals of the attenuator,
V_R be the voltage at the receiver terminals, $V_R = V_o \times 10^{-0.05\,N_a} \times \dfrac{R_R}{R_R+R_F}$.
R_R be the receiver impedance (20 ohms for the S.F.E.R.T. receivers) and that of its cord,
R_F be the fixed impedance of 577 ohms (used with the 4 S.F.E.R.T. receivers, the attenuator must be connected to an impedance of at least 100 ohms),
P'_R be the receiver sensitivity (in dynes per cm^2 per volt),
E_T be the microphone sensitivity (in volts per dyne/cm^2),
A be the voltage gain of the amplifier,
N_a be the attenuator setting with the switches in the " a " position,
N_b be the attenuator setting with the switches in the " b " position.

1. Putting the switches in the " a " position and setting the attenuator to any value N_a (in general $N_a = 5$ db), we have the value V_m read on the voltmeter equal to :

$$V_R \times P'_R \times E_T \times A = V_o \times 10^{-0.05\,N_a} \times \frac{R_R}{R_R+R_F} \times P'_R \times E_T \times A$$

2. Putting the switches in the " b " position and setting the attenuator to a value N_b so as to obtain the same indication on the voltmeter, we have :

$$V_M = V_o \times 10^{-0.05\,N_b} \times A$$

whence the receiver sensitivity P'_R is given by :

$$P'_R = \frac{(R_R+R_F) \times 10^{0.05\,(N_a-N_b)}}{R_R \times E_T} \qquad (1)$$

It is known that :

$$E_T = 10^{0.05\,[N'-(N+N'')]} \text{ (obtained in calibrating the microphone)}$$

where $E_T = 10^{0.05\,N_c}$ if we put : $N_c = N' - (N+N'')$.

Equation (1) becomes :

$$P'_R = \frac{(R_R+R_F) \times 10^{0.05\,(N_a-N_b-N_c)}}{R_R}$$

Putting :

$$\frac{R_R+R_F}{R_R} = S$$

We can write :

$$N_d = 20 \log \frac{S}{K}$$

K being an arbitrarily chosen number.

TABLE 3

Calibration of a S.F.E.R.T. Receiver

Microphone No. 3978 — Value of factor N_a = 5 decibels — Receiver No. 23

c/s	Microphone sensitivity (N_c)	Factor to take account of the receiver impedance (N_d)	Sum of the quantities $N_a+N_c+N_d$	N_b	Receiver sensitivity (db relative to 1 dyne/cm² per volt)
100	−49.6	−28.9	−83.5	+58.6	24.9
200	−50.3	−28.9	−84.2	+59.0	25.2
300	−50.4	−28.9	−84.3	+59.1	25.2
400	−50.3	−28.9	−84.2	+59.3	24.9
500	−50.8	−28.9	−84.7	+59.3	25.4
600	−50.7	−28.8	−84.5	+59.1	25.4
700	−50.9	−28.8	−84.7	+59.3	25.4
800	−50.9	−28.8	−84.7	+59.5	25.2
900	−50.7	−28.7	−84.4	+59.8	24.6
1,000	−50.4	−28.7	−84.1	+59.5	24.6
1,100	−50.5	−28.7	−84.2	+59.2	25.0
1,200	−50.5	−28.7	−84.2	+58.9	25.3
1,300	−50.2	−28.7	−83.9	+59.2	24.7
1,500	−50.8	−28.6	−84.4	+58.6	25.8
1,800	−50.3	−28.6	−83.9	+58.6	25.3
2,100	−50.6	−28.5	−84.1	+58.6	25.5
2,400	−50.6	−28.5	−84.1	+59.2	24.9
2,700	−49.9	−28.4	−83.3	+59.7	23.6
3,000	−49.9	−28.4	−83.3	+60.3	23.0
3,300	−49.3	−28.4	−82.7	+60.6	22.1
3,600	−49.4	−28.3	−82.7	+60.6	22.1
4,000	−49.5	−28.3	−82.8	+60.8	22.0
4,500	−49.3	−28.2	−82.5	+59.8	22.7
5,000	−49.4	−28.0	−82.4	+58.3	24.1
5,500	−49.9	−27.7	−82.6	+56.0	26.6
6,000	−50.9	−27.5	−83.4	+56.4	27.0
7,000	−52.4	−27.3	−84.7	+59.9	24.8
8,000	−55.9	−27.1	−88.0	+61.2	26.8
10,000	−59.6	−27.1	−91.7	+72.3	19.4
Mean sensitivity (in the frequency band 200-3,600 c/s)					+24.7 db relative to 1 dyne/cm² per volt

Making $K = 1$.

$$N_d = 20 \log S \qquad \text{or} \qquad S = 10^{0.05\, N_d}$$

Therefore :

$$P'_R = 10^{0.05\,(N_a - N_b - N_c + N_d)}$$

Note. — The microphone calibration : $E_T = 10^{0.05 N_c}$ always gives, with the bases actually chosen, a negative number for N_c. Considering only the modulus $|N_c|$ of N_c, we will have :

$$P'_R = 10^{0.05\,(N_a - N_b + (-N_c) + N_d)}$$

or

$$N_a + N_c + N_d - N_b = 20 \log P'_R$$

N_a is, in general, chosen as 5 db.
N_c depends upon the microphone used (and varies with frequency).
N_d depends upon the receiver impedance (and varies with frequency).

Note. — If the receiver being measured is greater in impedance than 100 ohms, the impedance R_F is not used, and the correction factor N_d dissappears.

4.2 Interpretation of the results

The sensitivity of a S.F.E.R.T. receiver is expressed in decibels relative to a base of 1 dyne/cm^2 per volt, and is the mean of the values obtained with the four microphones available.

A receiver is considered satisfactory if its sensitivity (arithmetic mean of the values obtained for 20 frequencies between 200 and 3,600 c/s), lies within the limits 26 db ± 3 db above 1 dyne/cm^2 per volt.

The C.C.I.F. Laboratory has documentation available giving the rules to be followed in carrying out such calibrations.

As an example, Table 3 (Page 40) gives the sensitivity values for a S.F.E.R.T. Receiver.

5. Calibration of sending system amplifier

This calibration consists of measuring the gain of the sending system amplifier under its normal conditions of use, into the non-reactive artificial lines ($Z = 600$ ohms). The principle of the measurement is shown schematically in Figure 7.

The output impedance of the attenuator (4 ohms) is small compared with the input impedance of the amplifier (200 megohms) and with that of the voltmeter, and the switch A-B does not alter the voltage at the attenuator output. The same applies to the voltage at the output of the sending system amplifier. The switch mentioned therefore allows this output voltage U_1 to be compared with the e.m.f. injected in series with the microphone.

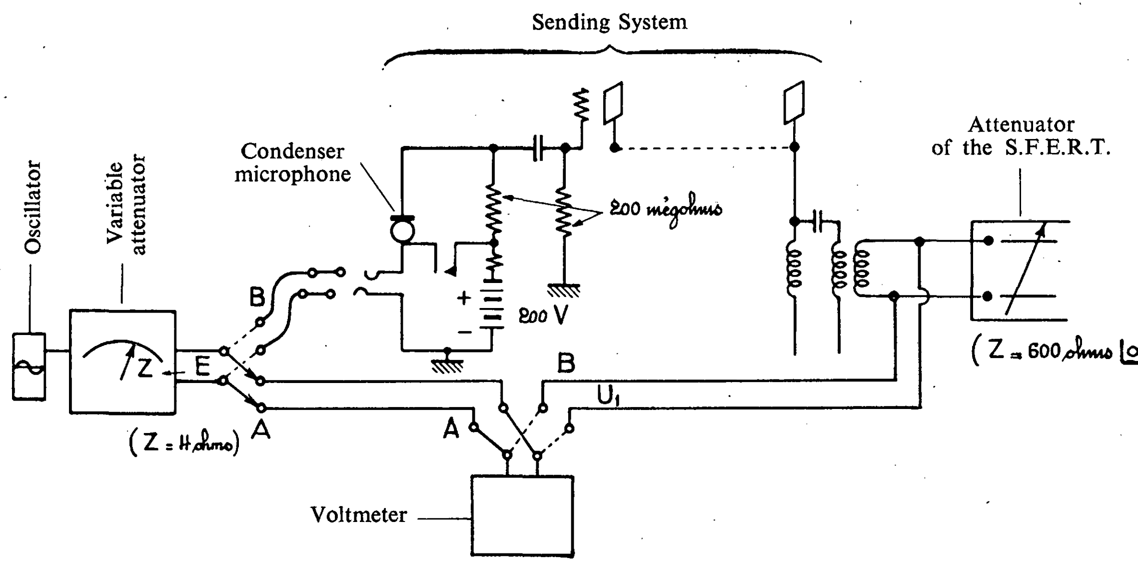

Sending System
Condenser
microphone
200 mégohms
Attenuator
of the S.F.E.R.T.
Oscillator
Variable
attenuator
200 V
B
E
Z
A
B
U1
A
Voltmeter

FIGURE 7

The measurement consists of two parts :

(a) The switches are put in the A position.

The attenuator dial is set to the value N_1 (in general 30 decibels) and the oscillator is adjusted so as to obtain a convenient deflection on the voltmeter.

(b) The switches are put in the B position.

The attenuator is adjusted to obtain the same deflection of the voltmeter as before. The difference between the new attenuation N_2 and the original N_1 measures the gain of the amplifier.

The gain of the S.F.E.R.T. sending system amplifier is practically independent of frequency (in the band from 100-7,000 c/s) ; its nominal value is 19.0 db at 1,000 c/s (its controls being set to zero) ; at 10,000 c/s its value is 18.0 db.

For " normal adjustment " of the S.F.E.R.T. sending system account is taken, in adjusting the sending amplifier, of the individual sensitivity of the microphone so as to obtain the values which define the " normal adjustment " of the S.F.E.R.T. sending system (see Volume IV of the *Green Book*, section IV.3.1.2).

6. Calibration of receiving system amplifier

This calibration consists of measuring the gain of the amplifier under its normal conditions of use, i.e. terminated in the impedance of a receiver.

The schematic diagram is given in Figure 8.

The operations are the same as in measuring the sending and amplifier.

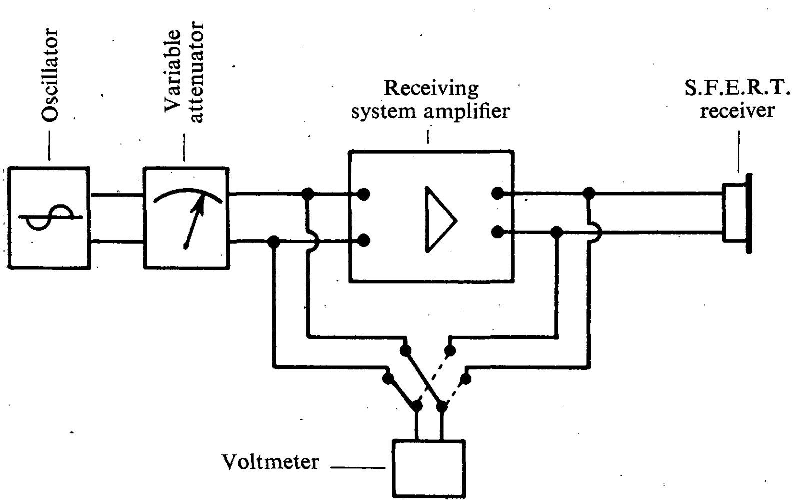


FIGURE 8

The gain of the S.F.E.R.T. receiving system amplifier is practically independent of frequency (in the band from 100 to 7,000 c/s) ; its nominal value is −2.0 db at 1,000 c/s (its dials being set to the zero position) ; at 10,000 c/s its value is −2.8 db.

For " normal adjustment " of the S.F.E.R.T. receiving system account is taken, in adjusting the receiving amplifier, of the individual sensitivity of the receivers so as to obtain the values which define the " normal adjustment " of the S.F.E.R.T. receiving system (see Volume IV of the *Green Book* Section IV.3.1.2).

7. Normal adjustment of the S.F.E.R.T.

The theoretical sensitivity of the S.F.E.R.T. is defined at each frequency (from 100 to 5,000 c/s) for the sending system on the one hand for the receiving system on the other in Volume IV of the *Green Book* Section IV.3.1.2.

The mean sensitivity of the sending system taking account of the mean sensitivity of the microphone used is adjusted to − 31.5 db relative to 1 volt per dyne/cm^2.

From the mean value of sensitivity of a given microphone (in the frequency band from 200 to 3,600 c/s) and of the nominal gain of the sending system amplifier (measured value, Section 5), corrections are determined to be made to the gain of the amplifier (from the nominal gain) to obtain the " normal adjustment " of the S.F.E.R.T. sending system (as indicated in the table below).

Mean sensitivity of microphone used 1	Mean gain of sending system amplifier 2	Normal adjustment of S.F.E.R.T. sending system 3	Sensitivity of S.F.E.R.T. sending system (1) + (2)	Correction to be made to the gain of the sending system amplifier
−50.4 db	+19.0 db	−31.5 db	−31.4 db	−0.1 db

The mean sensitivity of the receiving system taking account of the mean sensitivity of the receiver used is +24.2 db relative to 1 dyne/cm^2 per volt.

In the same way, to determine the " normal adjustment " of the S.F.E.R.T. receiving system, account is taken of the mean value of sensitivity of the receiver used (in the frequency band from 200 to 3,600 c/s) and of the nominal gain (measured value) of the receiving amplifier. The corrections made to the gain adjustment of the amplifier (from the nominal gain) are obtained as shown in the table below.

Mean sensitivity of receiver used 1	Mean gain of receiving system amplifier 2	Normal adjustment of S.F.E.R.T. receiving system 3	Sensitivity of S.F.E.R.T. receiving system (1) + (2)	Correction to be made to the gain of the receiving system amplifier
+24.7 db	−2.0 db	+24.2 db	+22.7 db	+1.5 db

The sensitivity of the complete S.F.E.R.T., without distortion, and with 0 db in the junction attenuator, which defines the " normal adjustment of the complete S.F.E.R.T." is -7.3 db relative to 1 dyne/cm^2 per dyne/cm^2.

In a reference equivalent determination, for sending or for receiving, the junction attenuator of the S.F.E.R.T. is set at 24 db. So the total sensitivity of the complete S.F.E.R.T. with no other supplementary attenuation is -31.3 db relative to 1 dyne/cm^2 per dyne/cm^2.

Note. — The sending and receiving systems of the S.F.E.R.T. are equipped with distortion networks which change the sensitivity-frequency characteristics of these systems to those simulating commercial telephone systems of a very old type.

The C.C.I.F. Laboratory continues, as in the past, to make calibration measurements under these conditions but nevertheless the S.F.E.R.T. is never used in this way for measuring reference equivalent.

8. Measurement of noise arising in the separate parts of the S.F.E.R.T.

In order to ensure that the noise produced in the components of the S.F.E.R.T. is lower than a given acceptable limit, the masking effect which this noice causes to various sinusoidal tones is determined.

Briefly, the method consists of injecting a sinusoidal voltage in series with the condenser microphone associated with the S.F.E.R.T. sending system. The tests are made with and without the condenser microphone polarising voltage. When the polarising voltage is applied to the microphone the latter must be shielded from acoustical vibrations by a special cover. In both cases the gain of the S.F.E.R.T. sending and receiving system amplifiers is adjusted to its maximum and all the attenuation is removed from the junction of the S.F.E.R.T.

An observer then listens on the moving coil receiver of the S.F.E.R.T. and if the level of the noise is lower than the permitted maximum limit he will hear the sinusoidal tone. If the noise masks the sinusoidal tone an attempt is made to reduce the noise in the system ; it is assumed that mistakes in wiring or connecting together the components of the system have been previously eliminated, and that the observer has normal hearing. These tests are made under silent conditions (listening cabinet).

The system can be considered satisfactory from the point of view of noise if the tone begins to be heard with a setting of the junction attenuation greater than 60 decibels at 100 c/s and greater than 80 decibels at each of the other frequencies ; these figures represent the mean of the results obtained with five observers.

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P. Chavasse. — *Acoustical and Telephonometric Measurements* (1943).

P. Riety. — A review of the theory of the gold-leaf thermophone (*Annales des Télécommunications*, French administration), 1955, Volume 10, No. 7-8.

II. ABSOLUTE CALIBRATION PROCEDURE FOR THE A.R.A.E.N. AT THE C.C.I.F. LABORATORY

1. General principles of electro-acoustic calibration

Each of the microphones used with the A.R.A.E.N. has been subjected to a free field calibration in an anechoic room and each of the telephone receivers has been calibrated on an artificial ear.

The C.C.I.F. Laboratory confines itself to verifying the stability of the microphones with time by periodically carrying out their calibration with respect to the sound pressure applied to their diaphragms under specified conditions. This calibration is effected in two parts.

1. The absolute acoustical calibration is carried out on a probe microphone by means of a stationary wave resonance tube and a Rayleigh Disk. The theory of this calibration is given in § 2 below. Measurement of the voltage developed at the microphone terminals thus enables the sound pressure at the tip of the probe microphone to be determined. Otherwise the probe microphone gives negligible disturbance in the acoustical field into which it is introduced. Consequently the probe microphone, so calibrated, allows of the absolute measurement of a sound pressure.

2. The microphone and receiver of the A.R.A.E.N. are calibrated under specified conditions as follows.

(a) For calibration, an A.R.A.E.N. microphone is acoustically coupled by means of a closed coupler (see Figure 3) to a telephone receiver used as a sound source and fed from a variable frequency oscillator. The tip of a calibrated probe microphone is introduced into the cavity as shown. The voltage at the terminals of the A.R.A.E.N. microphone is measured when a sound pressure, measured by means of the probe microphone, is applied to it. This determines the sensitivity of the A.R.A.E.N. microphone under the particular conditions of measurement for each of the measured frequencies.

(b) For calibration, an A.R.A.E.N. receiver is fixed on an artificial ear the acoustical impedance of which corresponds approximately to the mean of human ears and which contains a probe microphone permitting the sound pressure to be measured at a fixed point in the artificial ear cavity. On applying a fixed voltage at a given frequency to the receiver terminals and measuring the sound pressure produced in the artificial ear cavity, the sensitivity of the receiver at this frequency can be deduced.

Note. — Research Report No. 13,200 of the British Telephone Administration contains some theoretical notes on these calibrations and the results of the first calibrations made after the installation of the A.R.A.E.N. at the C.C.I.F. Laboratories in Geneva.

2. Theory of calibration of microphones with a stationary wave tube and a Rayleigh disk *

2.1 Theory of the Rayleigh disk

It is known that a thin circular disk suspended in a fluid excited at a horizontal velocity V, is influenced by a torque of moment M given by the equation

$$M = \frac{1}{6} \rho \, d^3 \, V^2 \sin 2\theta \tag{1}$$

where ρ is the specific gravity of the fluid,
d the diameter of the disk,
V the velocity of the fluid,
θ the angle between the direction of displacement of the fluid and a normal to the disk.

This expression has been derived theoretically by König, by approximating the disk to a very flat ellipsoid.

The direction of this torque is independent of the sign of V, consequently, if the displacement of the fluid is alternating, equation (1) remains true with V denoting the r.m.s. value of the velocity.

In practice the position of rest of the disk (when the fluid is stationary) is arranged to correspond to $\theta = 45°$: if φ is the angular displacement

$$\varphi \text{ (radians)} = \theta - \frac{\pi}{4} \tag{2}$$

If K is the moment of torsion of the disk suspension we have

$$M = K\varphi = \frac{1}{6} \rho \, d^3 \, V^2 \tag{3}$$

φ being sufficiently small for $\dfrac{\sin \varphi}{\varphi}$ to be taken as unity.

From equation (3)

$$V = \sqrt{\frac{6\,K}{\rho \, d^3}} \sqrt{\varphi} \tag{4}$$

Observing the displacement δ at a distance l of a light beam reflected by the Rayleigh disk used as a mirror, we have

$$\delta = 2 \, \varphi \, l$$

and

$$V = \sqrt{\frac{3\,K}{\rho \, l \, d^3}} \sqrt{\delta} \tag{5}$$

The moment of torsion K is determined in advance for the suspension thread by suspending, at the end to which is afterwards fixed the Rayleigh disk, a disk

* After W. West, *Acoustical Engineering* (Pitman & Sons, London, 1932).

of known moment of inertia J about its vertical axis. The period T and the logarithmic decrement D of the free oscillations of this disk are observed, and we have

$$K = \frac{J}{T^2} (4 \pi^2 + D^2) \tag{6}$$

Equation (5) then gives directly the velocity V as a function of the observed displacement.

2.2 Use of a stationary wave resonance tube

Various methods have been suggested to protect the Rayleigh disk from the effects of draughts which interfere with the measurement.

In the A.R.A.E.N. electro-acoustic measuring equipment the following method is used, the probe microphone to be calibrated is placed at one end of a long tube ; the sound source, placed at the other end of the tube, is a telephone receiver fed with alternating current by an adjustable frequency oscillator.

This frequency is adjusted so as to produce stationary waves with antinodes at the two ends of the tube, while at the middle of the tube, where the Rayleigh disk is suspended, a velocity maximum exists (i.e. a pressure node) ; the existence of such a condition is recognised because simultaneous maxima are obtained for the current I at the microphone output and the deflection of the Rayleigh disk. The corresponding wavelength is

$$\lambda = \frac{2 L}{n} \tag{7}$$

where L is the length of the resonance tube and, n, any odd number. The lowest frequency which can be used corresponds to a wavelength of $\lambda_1 = 2L$. The highest frequency is limited by the appearance of transverse waves which would upset the standing waves.

With the equipment and tube used at the C.C.I.F. Laboratory this upper limit occurs at about 6,500 to 7,000 c/s.

The theory of the Rayleigh disk enables the velocity V in the middle of the tube to be determined by equation (5) above as a function of the measured deflection. The pressure P at the ends of the tube are obtained from this by the equation

$$P = \rho c V \tag{8}$$

where ρ is the specific gravity of the fluid (in this case air).

c is the velocity of propagation of sound in air (in other words, ρc is the acoustical impedance of air). (It is well known that this equation applies to pressure and velocity at the same point of a plane progressive wave. It can be shown that it applies in the present case by considering the stationary wave as the resultant of two superposed progressive waves travelling in opposite directions).

I is measured with a milliammeter ; knowing the ratio I/P the sensitivity of the microphone can be calculated. It is seen that it is a matter of a calibration with respect to the pressure at the microphone diaphragm (or at the end of the probe, if a probe microphone is involved), as in the thermophone method or compensation method, and not with respect to the pressure which would exist, in the absence of the microphone, at the point at which it is placed, as in free field calibration methods.

3. Practical method for calibrating a probe microphone with the Rayleigh disk

Before beginning probe microphone calibration measurements, it is necessary to adjust the standing wave resonant tube to a suitable length and to place the Rayleigh disk carefully in its mounting at the centre of the tube. The disk should then be rotated about 45° (see § 2 above) ; in this way a light spot is obtained on a graduated transparent scale. Small displacements from the disk position corresponding to an angle of 45° introduce a negligible error in the calibration and the graduated scale can be slightly displaced laterally so that the light spot appears at the 0 graduation of this scale.

Figures 1 and 2 give the arrangements used for such a measurement. Special switches enable the electrical connections to be set up successively.

For each calibration frequency chosen, it is first necessary to find the nearest resonant frequency by use of the various lengths of resonance tube. Resonance is detected either by movement of the light spot itself or by the deflection of the voltmeter needle. Both indications should show the resonance point simultaneously by a maximum deflection ; if not there is every reason to suspect a leak or an error in setting up the resonance tube.

3.1 Measurement of output voltage of the probe microphone when a pressure is applied. (See Figure 1)

Having set up the electrical connections by means of the switches, the oscillator output voltage is adjusted to obtain the " normal deflection " of the spot and at the same time the sending amplifier is adjusted so as to obtain a convenient deflection x on the voltmeter ; the " normal deflection " is obtained from the constants of the Rayleigh disk which have been previously determined.

For example if, for a given disk, we have the following relationship between the acoustic pressure and the deflection of the spot : $P = 12.7\sqrt{\delta}$ dynes/cm².

Taking a value of pressure P of 50 dynes/cm², i.e. a value convenient for calculation, the value of 15.5 cm. is obtained for δ. This deflection of 15.5 cm. for the spot with respect to the zero of the graduated scale is called " normal deflection " and is used for measurements made with this disk.

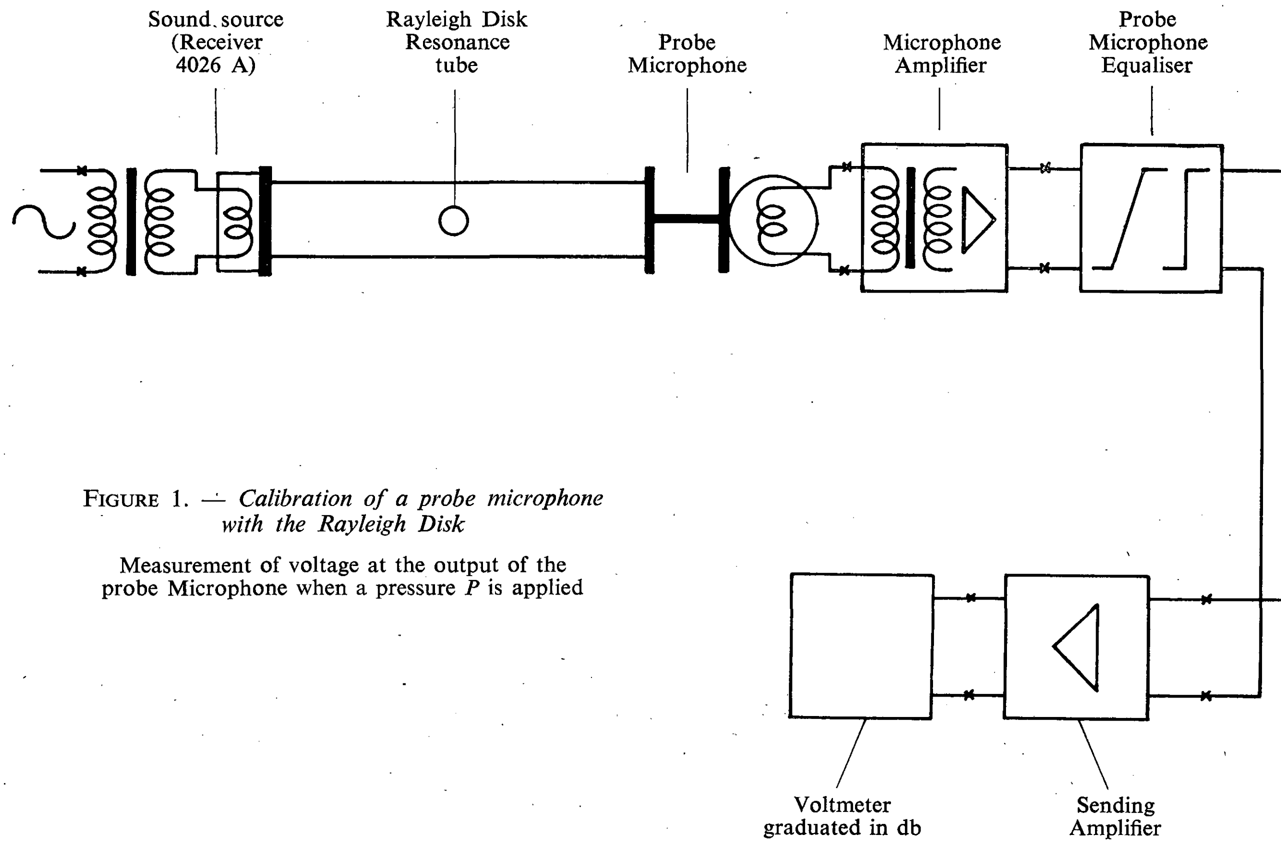

FIGURE 1. — *Calibration of a probe microphone with the Rayleigh Disk*

Measurement of voltage at the output of the probe Microphone when a pressure P is applied

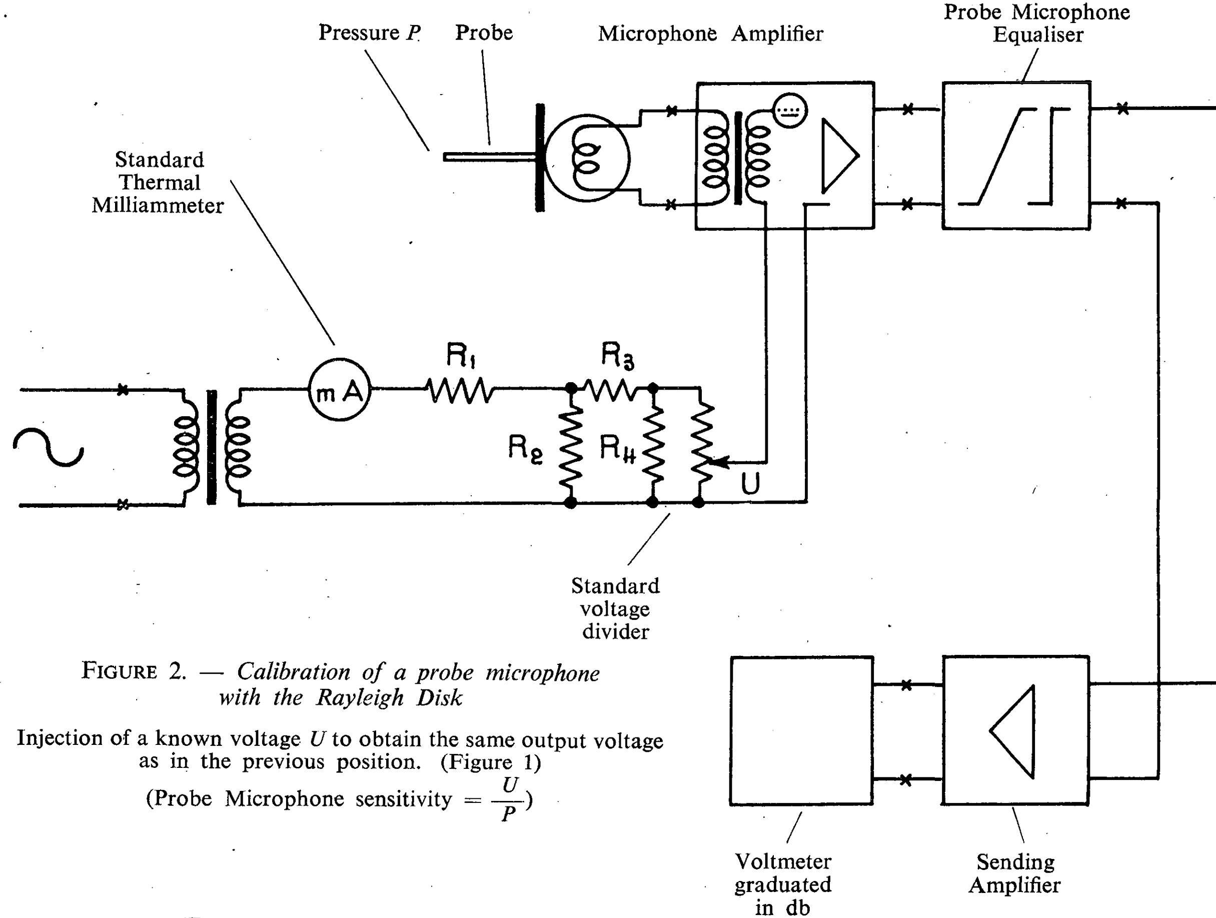

FIGURE 2. — *Calibration of a probe microphone with the Rayleigh Disk*

Injection of a known voltage U to obtain the same output voltage as in the previous position. (Figure 1)

(Probe Microphone sensitivity $= \frac{U}{P}$)

3.2 Determination of the voltage corresponding to the deflection " x " on the voltmeter (see Figure 2)

In this measuring position a known voltage, U, adjusted so as to obtain the same deflection " x " on the voltmeter is substituted for the e.m.f. developed by the probe microphone due to the acoustic pressure P at the end of the stationary wave resonance tube. This calibrated voltage is obtained, as shown in Figure 2, from the value of current shown by a standard thermal milliammeter. For example in the C.C.I.F. Laboratory, when the thermal milliammeter is adjusted to the 15 milliamps reading (marked in red on the scale) this corresponds to a voltage of 10 millivolts at the input of the voltage divider. The voltage divider is adjusted until the deflection " x " is obtained on the voltmeter ; the voltage divider dials are graduated directly in voltage.

From these values the sensitivity of the probe microphone can be determined as shown in the table below :

Frequency	Injected voltage U	Microphone sensitivity		Remarks
		U/P ($P = 50$ dynes/cm^2)	$20 \log_{10} U/P$	
c/s	mV	mV per dyne/cm^2	db relative to 1mV per dyne/cm^2	Constant acoustic pressure
154	0.84	0.0168	−35.5 or −95.5 db relative to 1 V per dyne/cm^2	$P = 12.7\sqrt{\delta}$ (for the disk chosen) $P = 50$ dynes/cm^2

The measurements are repeated for odd multiples of the fundamental resonance frequency (for which the length of the tube is equal to one half wave-length). If the minimum length has been chosen for the stationary wave tube, the fundamental frequency is 154 c/s (the example in the table). The measurement frequencies will be about 450, 750... up to 7,000 c/s. To obtain a greater number of results at the low frequencies it is necessary to lengthen the tube to its maximum extent to obtain integral multiples of the fundamental resonance frequency which in this case is about 80 c/s.

4. Calibration of an electrodynamic microphone of the A.R.A.E.N.

4.1 General

This measurement is essentially designed to verify the stability of the moving coil microphones of the A.R.A.E.N. Research Report No. 13,200 of the British Administration gives a method of calculating the absolute sensitivity of the microphone *under the particular conditions of measurement.*

The microphone to be calibrated is placed on a closed coupler formed by a small cylindrical cavity closed at one end by a moving coil receiver which serves as a sound source. At the other end is placed the microphone to be calibrated.

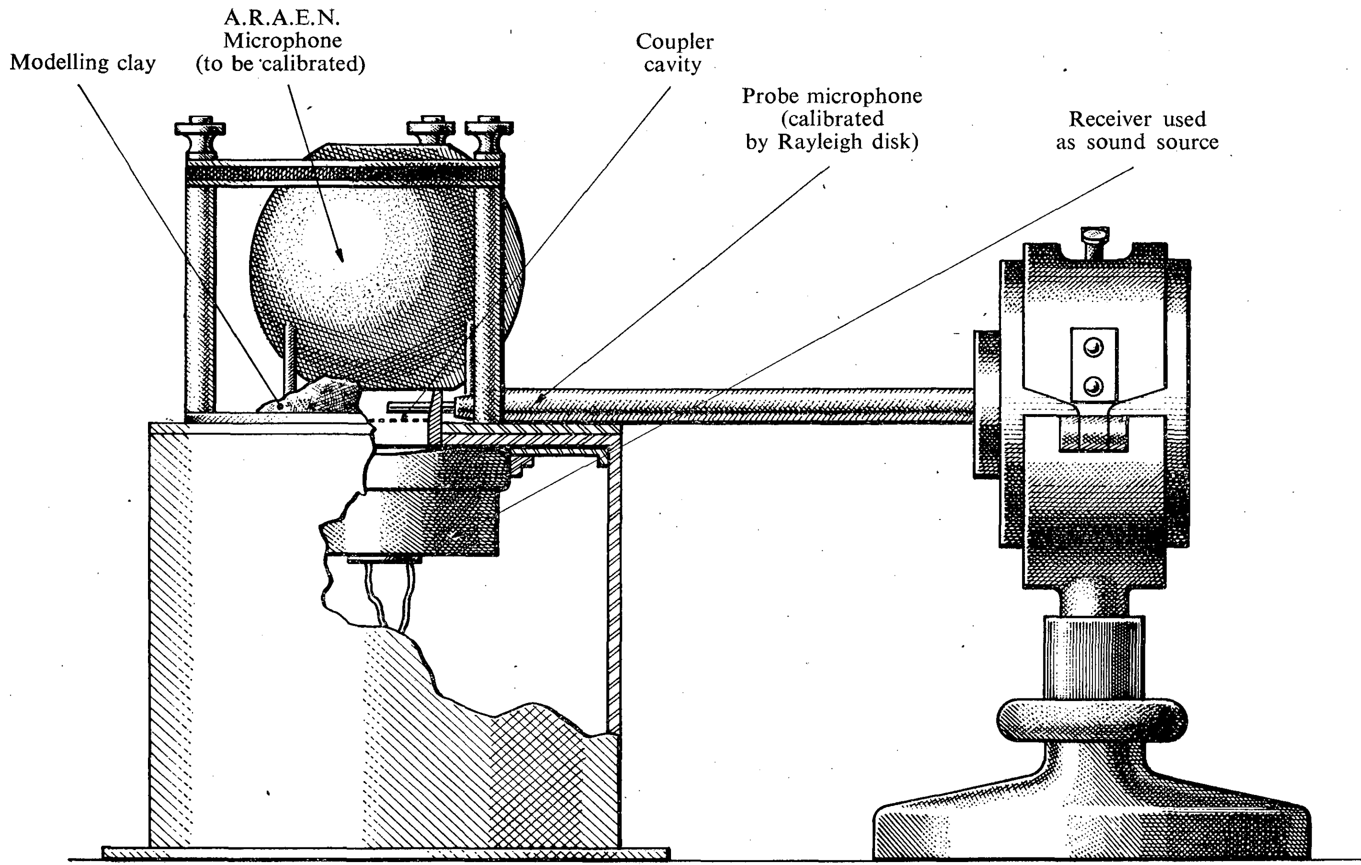

FIGURE 3. — *Closed coupler for calibration of moving coil microphones*

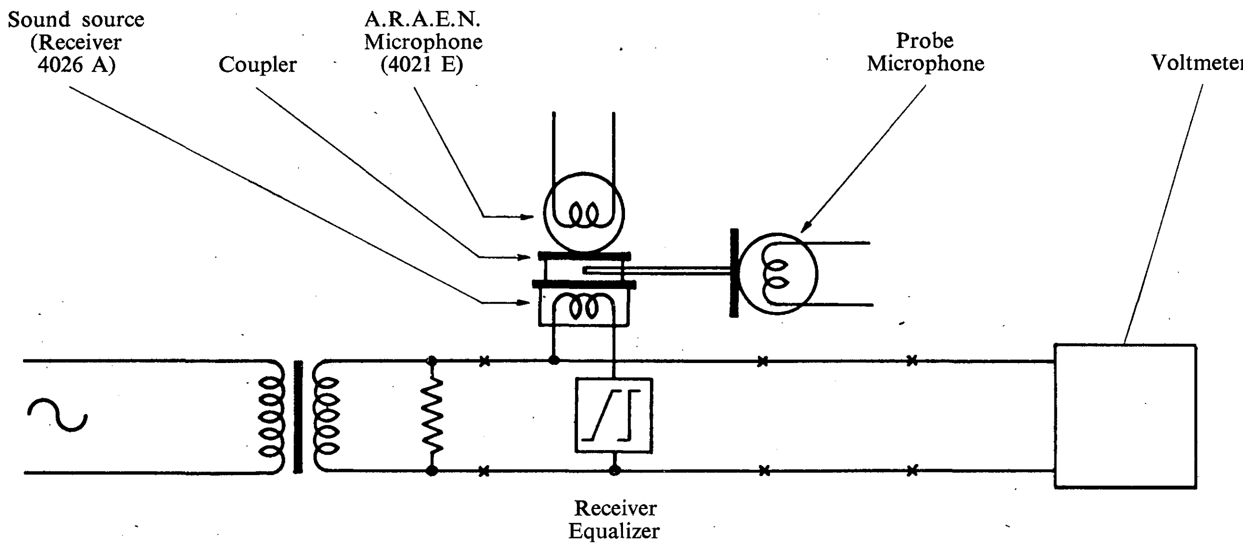

FIGURE 4. — *Calibration of an A.R.A.E.N. microphone*

Check of the voltage applied to the receiver used as the sound source

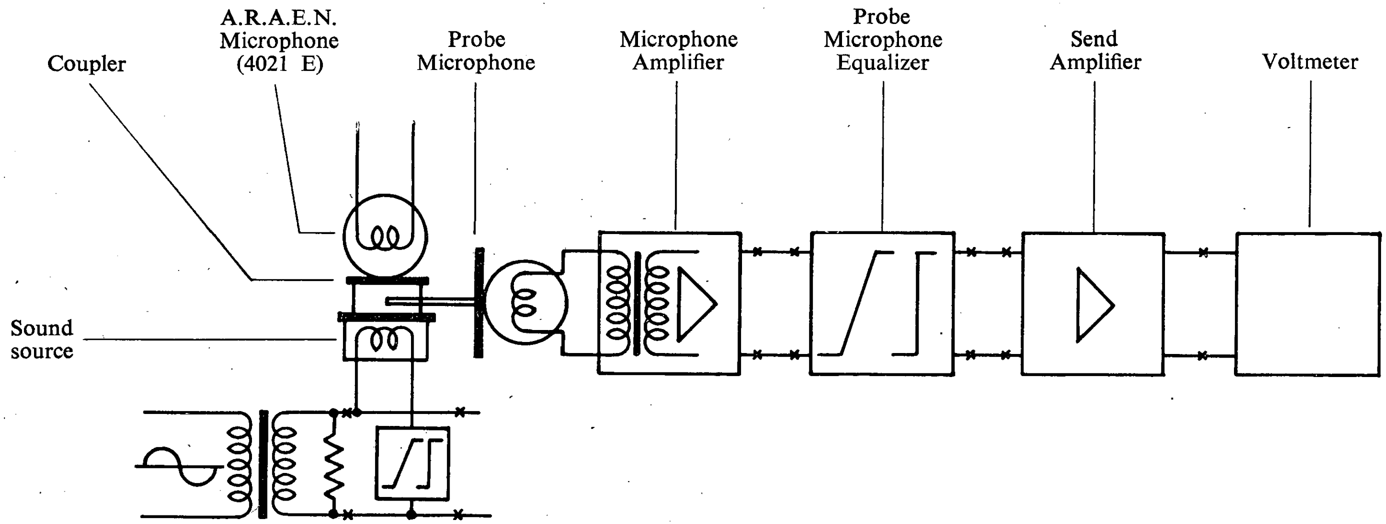

FIGURE 5. — *Calibration of an A.R.A.E.N. microphone*

Measurement of the output voltage of the probe microphone

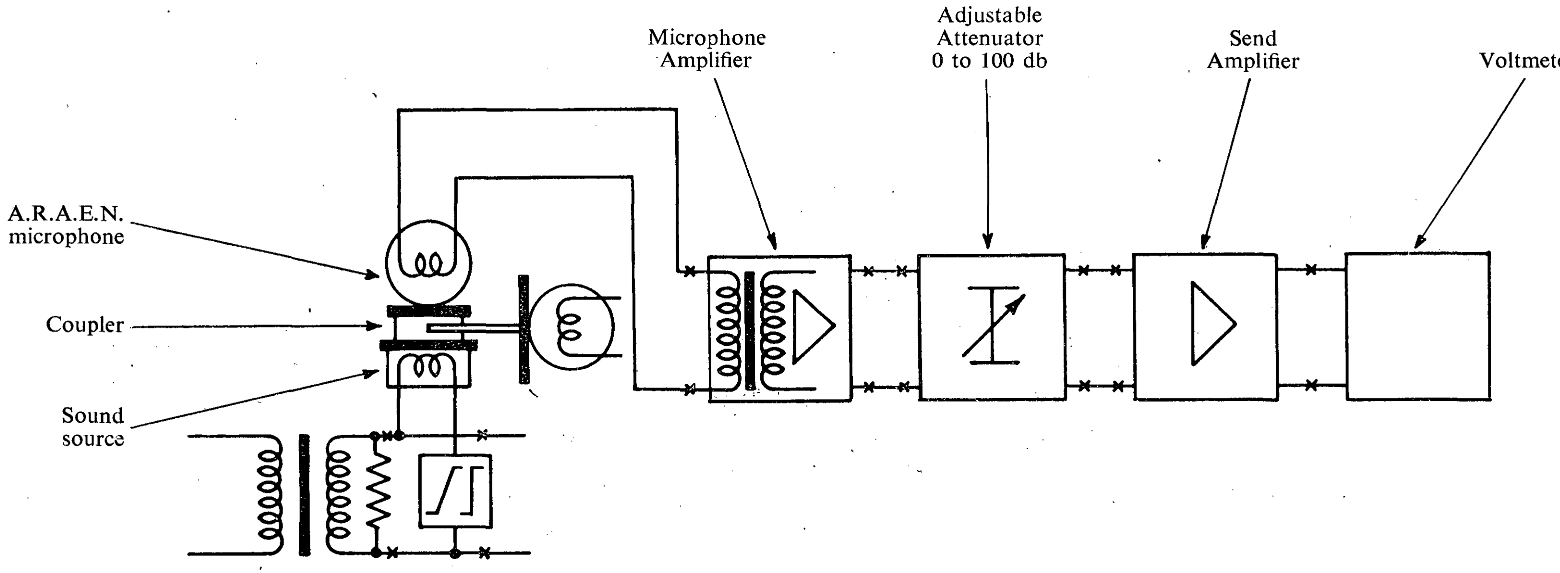

FIGURE 6. — *Calibration of an A.R.A.E.N. microphone*

Measurement of the output voltage of the A.R.A.E.N. microphone

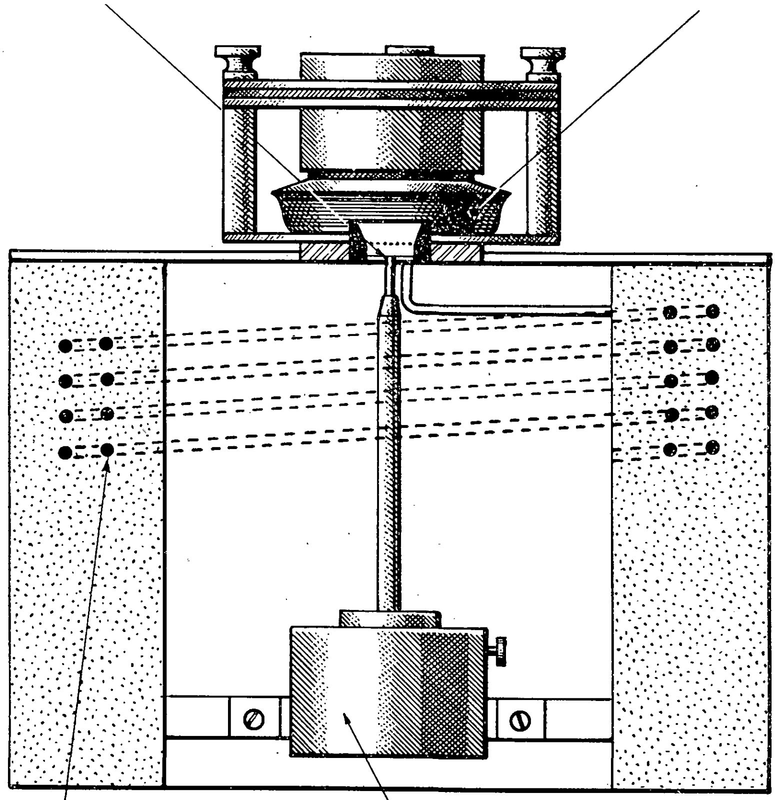

FIGURE 7. — *Artificial Ear for the calibration of moving coil receivers*

The tip of a probe microphone is introduced into this cavity via an orifice provided and used to measure the sound pressure at a fixed point ; the position of this is precisely defined with respect to its proximity to the diaphragm of the microphone to be calibrated. (Figure 3 shows the position of the microphones and receiver in the closed coupler).

4.2 CHECK OF THE VOLTAGE APPLIED TO THE RECEIVER USED AS A SOUND SOURCE (See Figure 4)

In this measuring condition the voltmeter indicates the voltage at the receiver terminals. It is not strictly necessary with a linear receiver to set up this value, but it allows the measuring conditions to be specified and reproduced. Generally a voltage of -15 db relative to 1 volt is adopted at a frequency of 1,000 c/s.

4.3 MEASUREMENT OF THE OUTPUT VOLTAGE OF THE MICROPHONE (See Figure 5)

In this measuring position the voltmeter is placed at the output of the probe microphone and without changing the output voltage of the oscillator the sending amplifier gain is adjusted so as to obtain a convenient deflection " y " of the voltmeter.

4.4 MEASUREMENT OF THE OUTPUT VOLTAGE OF THE A.R.A.E.N. MICROPHONE (See Figure 6)

In this measuring position, the attenuator (which varies between 0 and 100 db) is adjusted until the voltmeter indicates exactly the same deflection " y ". If the sensitivity of the probe microphone, measured as indicated in § 3, is added to the value which is read on the attenuator a magnitude is obtained which is a function of the sensitivity of the A.R.A.E.N. microphone *under these particular conditions of measurement,* which must remain constant throughout all the series of measurements and which also serves as a criterion of stability of the A.R.A.E.N. microphone (see also § 6).

5. Calibration of an electrodynamic receiver of the A.R.A.E.N.

5.1 GENERAL

The calibration of an A.R.A.E.N. electrodynamic receiver is obtained by placing the receiver on an artificial ear the acoustical impedance of which represents approximately the mean value of those obtained on human ears. This artificial ear consists essentially of a cylindrical cavity of 3 cm^3 volume terminated at the lower end by a spiralled tube representing the acoustical impedance (see the description of this artificial ear in Annex 9 of the Book of Annexes to Volume IV of the *Green Book.*

The sound pressure developed by the receiver at a fixed point in the artificial ear cavity is measured by means of a calibrated probe microphone.

Figure 7 shows this artificial ear with the receiver to be calibrated and likewise the probe microphone in the measuring position.

5.2 MEASUREMENT OF THE VOLTAGE APPLIED TO THE TERMINALS OF THE RECEIVER TO BE CALIBRATED (Figure 8, Position 1).

Here the voltage at the terminals of the receiver is measured and adjusted to a convenient value which in the case of the C.C.I.F. Laboratory is fixed at −15 db relative to 1 volt.

5.3 MEASUREMENT OF THE VOLTAGE AT THE OUTPUT OF THE SYSTEM FORMED BY THE PROBE MICROPHONE AND ITS AMPLIFIER (Figure 8, Position 2).

The deflection of the voltmeter connected at the output of the amplifier is brought to a convenient value towards the centre of the scale. This value is noted.

5.4 ADJUSTMENT OF THE INJECTION VOLTAGE WHICH IS SUBSTITUTED FOR THE E.M.F. PRODUCED BY THE PROBE MICROPHONE (see Figure 8, Position 3)

The voltmeter is placed at the input of a potential divider and the level is adjusted to give −20 db relative to 1 volt. This is the terminal voltage of the injection system.

5.5 ADJUSTMENT OF THE INJECTION VOLTAGE APPLIED TO THE PROBE MICROPHONE CIRCUIT (see Figure 8, Position 4)

In this measurement the dials of the potential divider are adjusted so as to obtain the same deflection of the voltmeter as that obtained in Position 2.

Measurements are generally at a series of increasing frequencies starting from 80 c/s and going up to 7,000 c/s.

5.6 INTERPRETATION AND DISCUSSION OF RESULTS

The following table shows typical results. The sensitivity calculated from these results corresponds to the case of a constant voltage calibration (−15 db relative to 1 volt, measured at the plug of the receiver) the sound pressure being measured at the bottom of the artificial ear cavity.

Frequency	Voltage	Injection	Sensitivity of probe microphone	Sound pressure	Sensitivity of receiver
c/s	mV	db relative to 1 mV	db relative to 1 mV/dyne/cm^2	db relative to 1 dyne/cm^2	db relative to 1 dyne/cm^2/volt
80	0.392	8.2	−35.5 (From the measurement described in § 3.2 above)	+27.3	+47.3

Note. — In practice, when the input voltage to the potential divider is adjusted, it is advantageous for precision in reading the voltmeter to use a level of −15 db because this indication is at the centre of the scale whilst the −20 db division

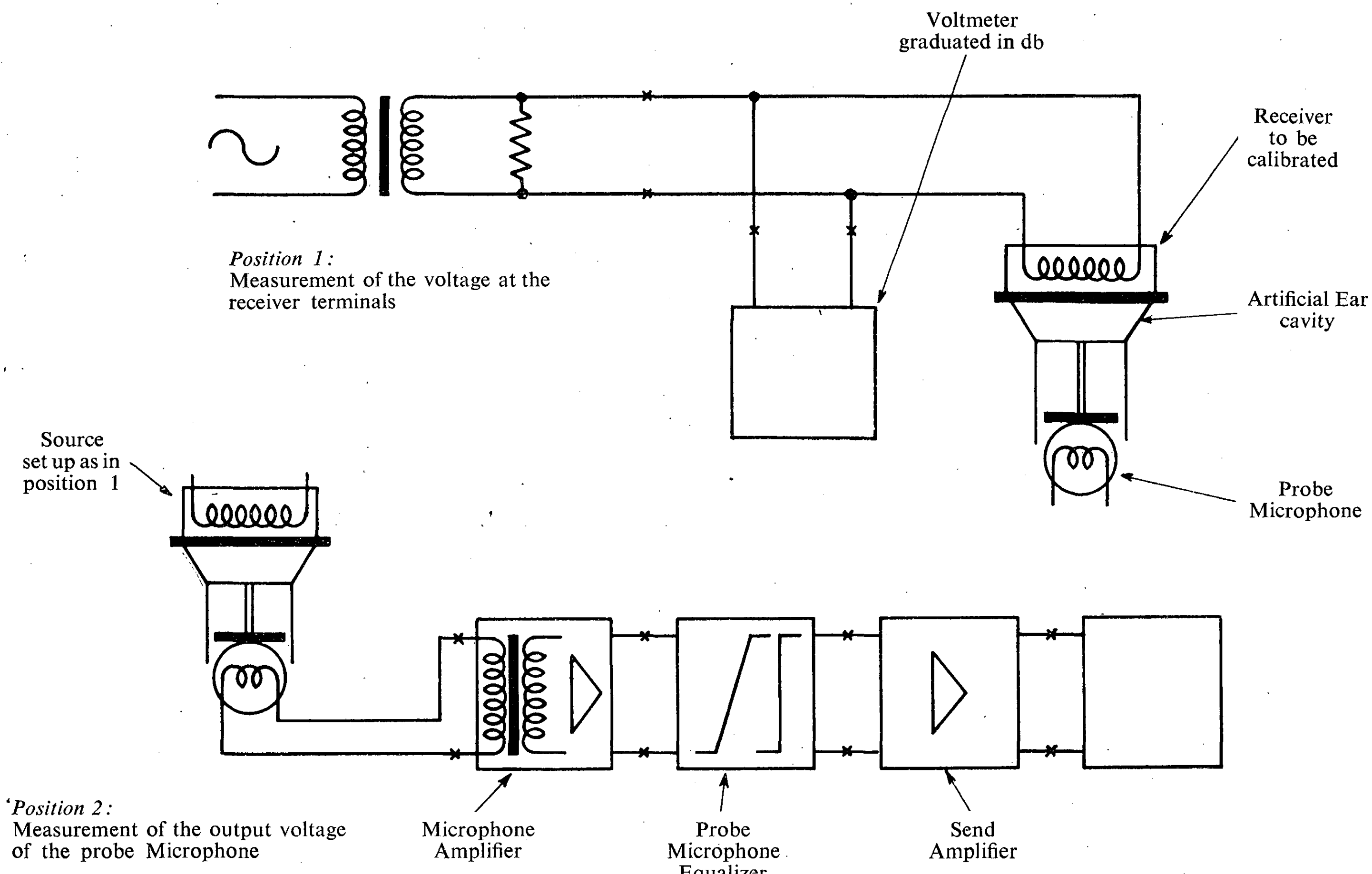

FIGURE 8. — *Calibration of an A.R.A.E.N. receiver*

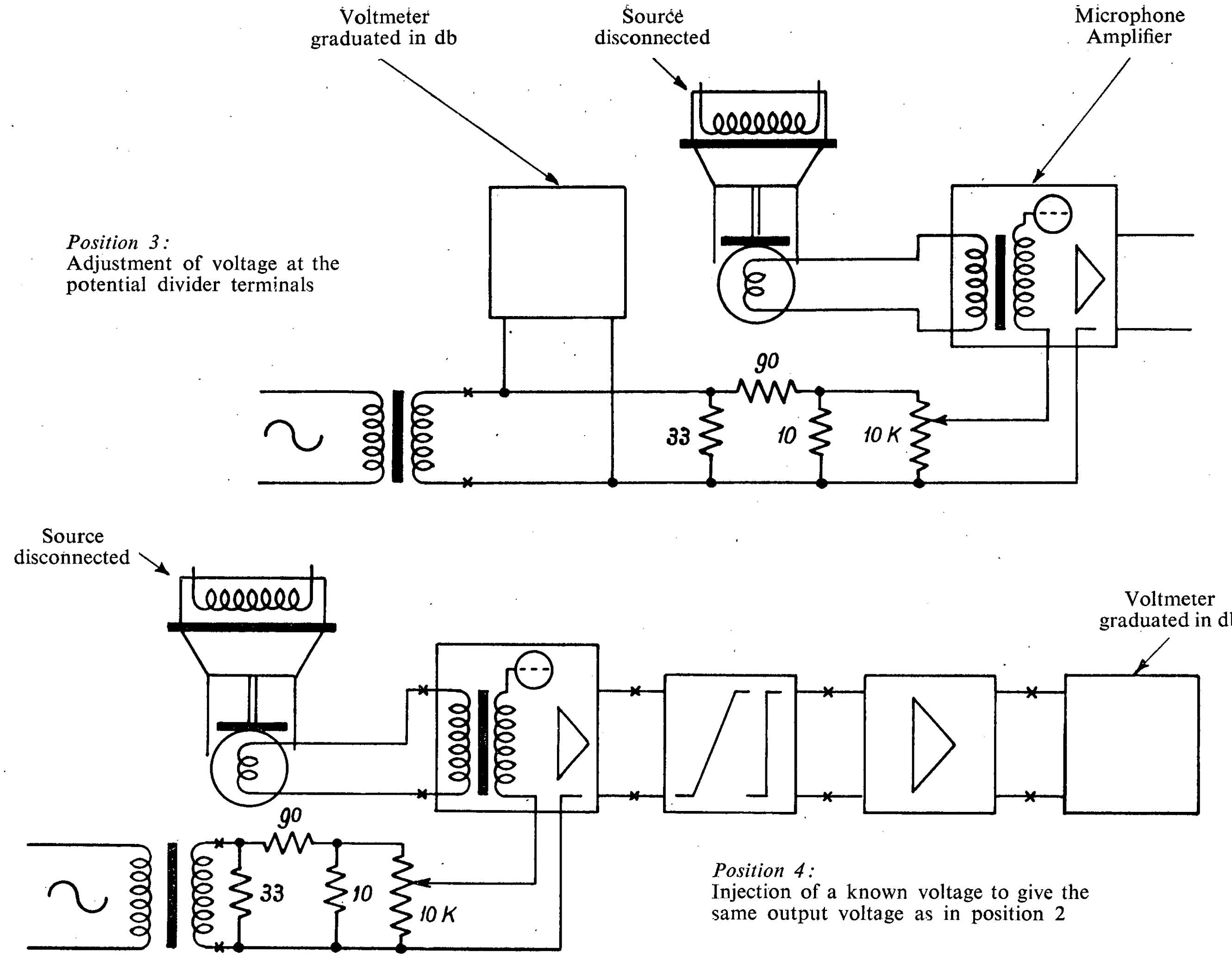

FIGURE 8 (continued). — *Calibration of an A.R.A.E.N. receiver* is not very clear on account of the curved scale shape. In this case it is necessary to apply a corresponding correction to the sensitivity value of the receiver.

6. Adjustment of gain of the A.R.A.E.N. send amplifier

Measurements described in Research Report No. 13,180 of the British Post Office have shown that a speech pressure of 1 dyne/cm^2 exists at a point 13.25 inches from the lips of a talker speaking at Reference Vocal Level for the A.R.A.E.N. (for a definition of the term speech pressure see the report mentioned). The A.R.A.E.N. is provided with a voltmeter for the purpose of controlling the vocal level of the talking operator. This voltmeter is connected across the input of the A.R.A.E.N. junction and the overall gain is so adjusted that when an operator speaks at the reference vocal level for the A.R.A.E.N. the speech voltage at the input to the junction is 1 volt. The absolute sensitivity of each of the microphones used with the A.R.A.E.N. varies slightly, consequently when one microphone is changed for another the gain has to be adjusted accordingly.

The sensitivity/frequency of the microphones (measured in a free acoustic field) is approximately flat with a 5 db peak at 450 c/s but the equalizer network placed after the microphone amplifier largely compensates for this. After allowing for this compensation there is a variation of the order of ± 0.5 db in the frequency range 100 to 1,000 c/s and a slightly larger variation outside of this range. To take full account of these variations, it is necessary to take some mean value for the sensitivity ; this value represents the microphone sensitivity over the speech frequency band which has the largest energy content as it is this band which determines the voltmeter reading. A study of the spectrum of the human voice shows that the greatest power is in the range 100 to 900 c/s. Consequently it is in this frequency band that the sensitivity of the microphone should be considered. As the sensitivity/frequency characteristic of the microphone plus equalizer is substantially flat in this region it is reasonable to take the arithmetic mean of the sensitivities of the microphone at 100, 300 and 900 c/s. This mean figure can then be used to calculate the gain required at the sending end of the A.R.A.E.N. to give 1 speech volt at the input of the junction when a speech pressure of 1 dyne/cm^2 is applied to the microphone. The normal gain of the microphone amplifier plus send amplifier (gain defined as

$$20 \log_{10} \frac{\text{Output voltage measured across a 600 ohm termination}}{\text{Input e.m.f. from a 20 ohm generator source}}).$$

is equal to 89 decibels, and if this gain is compared with the gain necessary for the specific microphone, the setting of the send amplifier can be obtained.

(The controls for this adjustment are calibrated in db relative to normal gain.)

The gain adjustment of the send amplifier is reduced to a series of routine operations which is carried out in three parts.

1. Take the free field sensitivity/frequency of the microphone in question and subtract the insertion loss of the microphone equalizer at 100, 300 and 900 c/s.

2. Take the mean of the three values of corrected sensitivity so determined.

3. Express this mean sensitivity in decibels relative to 1 volt per dyne/cm^2 and add algebraically 89 db (the sum of the microphone amplifier and send amplifier gains). (The mean sensitivity is in the order of -90 db relative to 1 volt per dyne/cm^2.) If this sum is zero the normal amplifier gain setting is correct. If the sum is positive the amplifier gain must be reduced by the amount of the sum ; if it is negative the amplifier gain must be increased by the magnitude of the sum.

Each microphone is supplied with a free field sensitivity/frequency characteristic which can be used for the method described above. The periodic tests made on the microphone will however provide a sensitivity/frequency characteristic determined on the arbitrary closed coupler. The sensitivity/frequency characteristic as determined on the coupler is allowed to change by 1 db before a microphone is rejected, but the send amplifier gain setting must be altered if the sensitivity changes by 0.2 db. Although the sensitivity determined on the coupler cannot be used directly for calculating the gain setting it can be used to modify the result obtained from the free field response.

If a coupler sensitivity/frequency characteristic is taken at the same time as the original free field characteristic, any change in sensitivity during the life of the microphone will be shown by differences between later coupler characteristics and the original characteristic. If the mean of the sensitivities at 100, 300 and 900 c/s is taken for the original characteristic and for each of the following determinations, the value of any change of the mean sensitivity can be determined.

This value must then be used to correct the original gain setting calculated from the free field sensitivity. (A decrease in sensitivity requiring an increase in gain and vice versa), This method assumes that for any change in free field sensitivity there is a similar change in the sensitivity as measured with the coupler ; this hypothesis is justified for the small variations considered here.

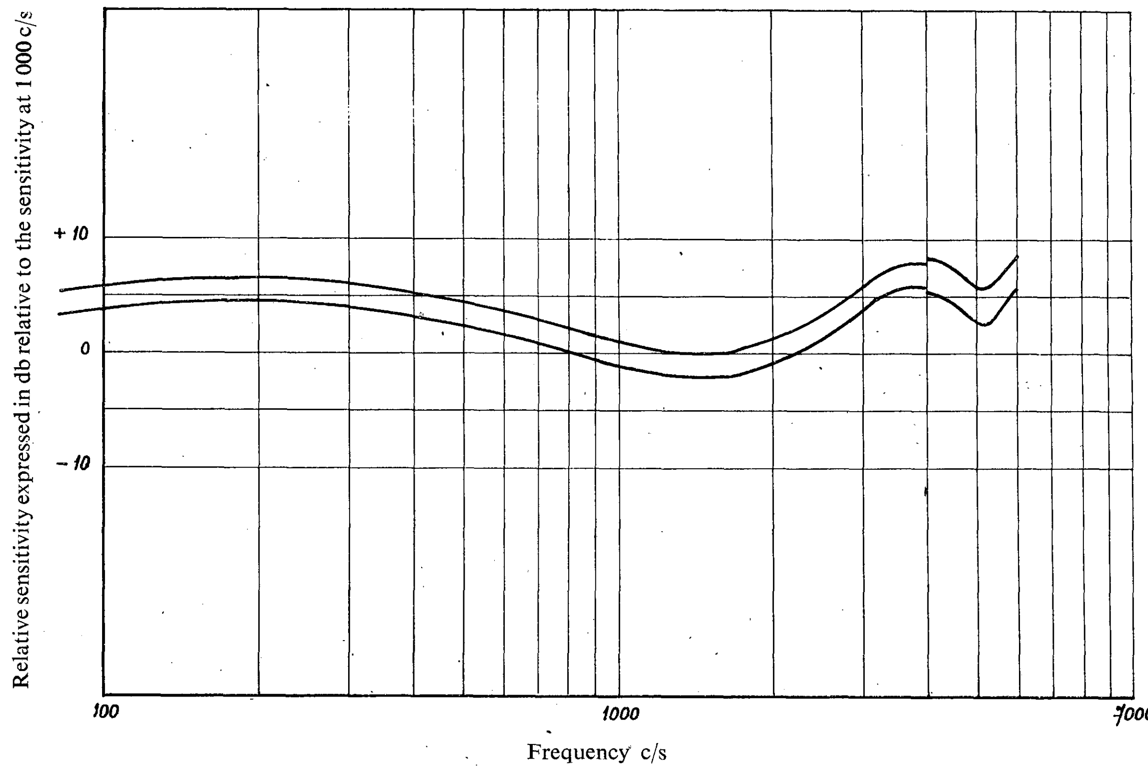

FIGURE 9. — *Curves giving the limits between which the " sensitivity/frequency " of the A.R.A.E.N. receivers, measured under the conditions described in section 5, must lie — Receivers No.* 4026 *A*

7. Adjustment of gain of the A.R.A.E.N. receive amplifier

The receiving system of the A.R.A.E.N. is terminated by four receivers of the same type (Standard Telephone and Cable, Type 4026 A) connected in series. Their relative sensitivity/frequency characteristics and absolute sensitivities must be kept within close limits. Furthermore the *gain of the receive amplifier is adjusted to a fixed value of 23 db.* The total attenuation of the electrical circuit of the receiving system of the A.R.A.E.N. between the input terminals of the receive attenuation equalizer and the terminals of any one receiver is 19.5 db at 1,000 c/s and takes account of the supplementary attenuation of 12 db due to the division by four of the output p.d. of the receiver impedance matching transformer (the four receivers are in series).

The sensitivity/frequency characteristics of the four receivers must be within the fixed limits (see Figure 9) ; if this is not so, the receivers which are outside these limits are returned to the British Administration.

If the sensitivity/frequency characteristics are within the fixed limits, the sensitivity of a single receiver only needs to be considered, and the mean of its sensitivivites at 100, 300, 1,000 and 2,000 c/s must be determined. This mean value must be equal to +43.7 db relative to 1 dyne/cm^2 per volt. In order to obtain this value so that all the receivers have the same sensitivity special attenuators, variable in steps of 0.25 db, are matched to each receiver.

8. Normal adjustment of the A.R.A.E.N.

The theoretical sensitivity of the A.R.A.E.N. is defined in Vol. IV of the *Green Book*, 3rd Part, § 3.1.3.B5.

The C.C.I.F. laboratory takes into account at the time of adjustment of sending system gain the differences between the individual sensitivity/frequency characteristics of the microphones. These differences are determined from the results of calibrations made periodically.

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ANNEX 6

METHOD STANDARDIZED IN THE UNITED STATES FOR THE CALIBRATION OF TELEPHONE RECEIVERS ON A COUPLER

(Extracts from the American Standards Association specification Z.24.9.1949)

1. Scope and purpose

1.1 The purpose of this standard is to describe a practical and reproducible method of evaluating the performance characteristics of an earphone by means of physical measurements of the earphone in conjunction with a standard terminating volume known as the " coupler ".

1.2 The method is adequate for controlling the characteristics over the frequency range most useful for speech, i.e. 300 to 5,000 cycles per second. Limitations of this method are discussed in 3.2.1, Type 1 Coupler.

1.3 This standard specifies a number of couplers, each of which is suitable for a certain type of earphone. No one of these couplers in suitable for all of the different types. Test laboratories are expected to select the coupler which is most suitable for each particular instrument in order that their results may be comparable with those obtained for other instruments of the same general type but of different manufacture.

1.4 The selection of the pressure microphone to be used for measuring the sound pressure in the coupler will depend upon the magnitude of the sound pressure, the range of frequency to be covered by the test, and the acoustical impedance of the diaphragm. Suitable pressure microphones are describer in the American Standard Specification for Laboratory Standard Pressure Microphones, Z24.8.1949. The basic research leading to the establishment of the couplers in this standard was done with the Type L microphone of that standard. The use of other standard microphones might lead to differences in the measured results and when intercomparison of data is desired these differences should be established for a particular design of earphone. The method for the calibration of the microphones is described in the American Standard Method for the Pressure Calibration of Laboratory Standard Pressure Microphones, Z.24.4.1949.

1.5 Drawings and test procedures, as well as other pertinent information, have been included as an aid to designers in setting up the tests.

2. Definitions

2.1 Letter symbols

For convenience the letter symbols used in the text and figures of this specification are summarized below. The symbols for voltages, currents and pressures represent absolute r.m.s. values. All electrical units are based on the absolute system.

η_a	= attenuator reading $\left(20 \log_{10} \frac{E}{E_1}\right)$
D_n	= harmonic distortion $\left(20 \log_{10} \frac{p_{nf}}{p_f}\right)$
E	= voltage of generating source
E_1	= voltage across R_2
E_e	= voltage across earphone
I_e	= current flowing in earphone
E_m	= open-circuit voltage generated by microphone
P	= available power in watts
L_p	= earphone sound pressure level in db
η_p	= available power response with a resistive source
p	= sound pressure in coupler as measured by microphone
p_f	= sound pressure in coupler of fundamental frequency
p_{nf}	= sound in coupler of n^{th} harmonic of p_f
R_o	= resistance of generating source ; also preferred source resistance
R_1, R_2, R_3	= measuring circuit resistances
η_v	= applied voltage response
Z_e	= impedance of earphone ($Z_e = R_e+jX_e$)
r_m	= pressure response of laboratory standard pressure microphone

2.2 and 2.3 (these paragraphs deal with other types of electro-acoustic transducers then telephone receivers).

2.4 Handset

A handset comprises an earphone and a microphone mounted on a common member which is normally held in the hand during use.

2.5 Electrical input

In this standard the electric input to an earphone is the voltage across the earphone or the power available from the source to which the earphone is connected.

2.6 Constant available power

A source of constant available power has a purely resistive internal impedance and has a constant open-circuit terminal impedance and has a constant open-circuit terminal voltage independent of frequency. An earphone is tested under conditions of constant available power if it is connected to such a source with or without a pure resistance in series between the earphone and the source.

Note. — In practice, the output circuit of the amplifier, oscillator, or transmission line supplying electric power to an earphone can usually be represented as an equivalent generator whose internal impedance is essentially resistive and is independent of frequency. Hence, a test procedure using the concept of constant available power to the earphone is of basic importance in yielding data which are indicative of the performance of the earphone.

A generator having a power capacity corresponding to an open-circuit voltage, E, and an internal resistance, R_0, both constant with frequency, is shown in Figure 1*a*. The available power is

$$\frac{E^2}{4R_0}$$

When the load impedance is a resistance equal to R_0 an impedance match exists and the maximum available power

$$\frac{E^2}{4R_0}$$

is dissipated in the load.

An earphone under test is connected as shown in Figure 1*b*.

When a generator with a constant voltage and a constant internal resistance is not available, suitable test conditions for this type of measurement can be provided by a generator whose voltage is maintained at a constant value, E, across its terminals in series with a resistance, R_0.

2.7 Coupler

An earphone coupler is a cavity of predetermined shape which is used for the testing of earphones. It is provided with a microphone for the measurement of pressure developed in the cavity.

2.8 Acoustical output

The acoustical output of an earphone is taken in this specification to be the sound pressure generated in a coupler.

Note. — Although it would be desirable also to express the output in terms of acoustical power delivered to the ear, the state of the art at this time is such that the constant of the ear not well enough known to permit such expression.

2.9 Preferred source resistance

The preferred source resistance is the generator resistance (R_o in Figure 1) specified by the designer for use in calibrating an earphone.

Note. — Normally this resistance is a compromise between a condition under which the maximum power is absorbed from the source and a condition under which the most uniform frequency characteristic is obtained.

2.10 Available power response

The available power response of an earphone at a given single frequency is defined as 10 times the logarithm to the base 10 of the ratio of the absolute value of the square of the r.m.s. sound pressure generated by an earphone in an appropriate coupler to the available electric power of the source. It is expressed in decibels relative to (1 dyne per cm²) ² per watt of available electric power.

The available power response is given by the equation :

$$\eta_p = 10 \log_{10} p^2 - 10 \log_{10} \frac{E_2}{4R_0} \qquad (1)$$

2.11 Applied voltage response

The applied voltage response of an earphone in a coupler at a given single frequency is defined as 20 times the logarithm to the base 10 of the absolute value of the ratio of the r.m.s. sound pressure generated to the r.m.s. voltage applied at the terminals of the earphone. The applied voltage response, $_v$, is expressed in decibels relative to 1 dyne par cm² per volt. The applied voltage response is given by the following equation :

$$\eta_v = 20 \log_{10} \frac{p}{E_e} \qquad (2)$$

Note. — The applied voltage response is useful in applications where a number of elements in a transmission system are being measured, individually or as a group, to obtain the overall response of a system and its components, including the earphone. A curve of the earphone impedance versus frequency should accompany the curve of voltage response versus frequency in order to specify completely the behavior of the earphone as a function of frequency. The applied voltage response and the available power response are related, as shown in the following equation :

$$\eta_p = \eta_v + 10 \log_{10} 4R_0 - 10 \log_{10} \left| 1 + \frac{R_0}{Z_o} \right|^2 \qquad (3)$$

2.12 Sound pressure level

The sound pressure level in decibels of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure of the sound to the reference pressure. In this specification the reference pressure is taken to be 2×10^{-4} dynes per cm². The sound pressure level in decibels may be expressed by the following equation :

$$L_p = 20 \log_{10} \frac{p}{0.0002} = 74.0 + 20 (\log_{10} p) \text{ db} \qquad (4)$$

2.13 Electrical impedance

The electrical impedance, at any given single frequency, of an earphone terminated in a coupler is the ratio of the voltage to the current measured at the input terminals. The value of the electrical impedance is a complex number and is expressed preferably in terms of magnitude and phase angle.

2.14 Frequency characteristics

The various frequency characteristics of an earphone express its electrical, mechanical, and acoustical properties as a function of frequency. "Available power-response frequency characteristic" is an example.

2.15 Output-input characteristic

The output-input characteristic of an earphone at any given single frequency expresses the square of the absolute value of the r.m.s. sound pressure produced in the coupler as a function of the available input power to the earphone. The input is specified in decibels above 1 watt and the pressure in decibels of sound pressure level.

2.16 Component harmonic distortion

The component harmonic distortion is 20 times the logarithm to the base 10 of the ratio of the absolute r.m.s. sound pressure of the harmonic frequency to the absolute r.m.s. sound pressure of the fundamental frequency. The input is a single frequency signal and may be specified as the available power or applied voltage.

The componant harmonic distortion in decibels is given by

$$D_n = 20 \log_{10} \frac{p_{nf}}{p_f} \tag{5}$$

3. Test Equipment

3.1 General

The test equipment needed for measuring the characteristics of the earphone by the coupler method is as follows :

- *(a)* a coupler to couple the earphone under test to a standard pressure microphone ;
- *(b)* a standard pressure microphone ;
- *(c)* electrical devices suitable for measuring electrical output, electrical impedance, and distortion ;
- *(d)* a variable frequency oscillator.

Primarily, couplers will be covered in this specification. Suitable standard pressure microphones are described in the American Standard Specification for Laboratory Standard Pressure Microphones, Z.24.8.1949, or any subsequent edition thereof approved by the American Standards Association, Incorporated. The electrical devices employed are well known so detailed specifications are not given in this standard.

3.2 Couplers

Three types of couplers are described. The first is suitable for determining the response of earphones which rest against the pinna ; the second is for use in testing insert-type earphones without the tips ; and the third is suitable for insert-type earphones with the tips attached. The material to be used in the construction of these couplers should be a hard non-magnetic substance such as brass or hard rubber. In this way the wall impedance is very high compared to the impedance of the air volume.

3.2.1 *Type-1 coupler.* — The outline dimensions of a coupler volume for the measurement of characteristics of earphones which press tightly against the pinna are shown in Figure 2.

The external construction of the coupler must be adapted to the particular receiver being tested. Figure 3 shows a suitable coupler for a typical receiver having a hard ear-cap. In this case, the inclined walls of the cavity are as one with the ear-cap (the C.C.I.F. reference artificial ear is of this type).

3.2.2 and 3.2.3 (These paragraphs describe couplers of type 2 and 3 which are not used with the telephone receivers).

4. Test procedure

4.1 General

Test procedures and basic circuits which are included in this specification are suitable for testing the following characteristics of earphones :

(a) available power response,
(b) applied voltage response,
(c) output-input characteristics,
(d) electrical impedance,
(e) component harmonic distortion.

The constants of the circuit components have been selected to permit the calibration of earphones having impedances within the range from 1 to 25,000 ohms. For high impedance earphones appropriate changes in the circuit constants will have to be made. In testing earphones for the characteristics listed above care should be exercised in selecting equipment which is linear in the complete voltage and pressure range encountered.

Note. — Any one of the frequency characteristics of an earphone may be obtained by measuring the above characteristics at a sufficient number of frequencies to establish a smooth curve, or a family of curves. For example, available power response frequency characteristic.

4.2 Available power response

The following instruments and circuit elements, as indicated in Figure 7, may be used in measuring the available power response :

(a) A variable frequency oscillator. For most purposes it is adequate for the frequencies at which tests are to be made that the harmonic content be less than 1 percent of the total output. It is desirable that the impedance of the oscillator plus any resistance in series with it be not less than 50 ohms ;

(b) A calibrated variable attenuator with a range of approximately 60 db. The smallest dial reading should be 0.1 db. If this is not the case, the output indicator mentioned under item 4.2 *(d)* should have 0.1 db

graduations. In order to minimize the difficulties of obtaining a sufficiently high voltage across R_1, the use of low impedance attenuators should be avoided. In general, attenuators having impedances ranging between 150 and 600 ohms are adequate. If necessary, the impedance of such attenuators can be increased by adding a series resistance ;

(c) A standard pressure microphone as specified in American Standard Specification for Laboratory Standard Pressure Microphones, Z.24.8.1949, or any subsequent edition thereof approved by the American Standards Association, Incorporated, and an associated microphone amplifier ;

(d) An amplifier equipped with a graduated indicator to measure r.m.s. a-c output and also equipped with an attenuator having a 40-db range

(e) A suitable a-c voltmeter. By proper switching means, the equipment listed under *(d)* can be used for this purpose ;

(f) An appropriate coupler ;

(g) A variable calibrated resistor, R_3, with a range from 1 to approximately 25,000 ohms ;

(h) Two resistances, R_1 and R_2.

Note. — In order to simplify the testing procedures, and to make the formula for the power response of the earphone a simple function of R_0, p_m, and η_a, the value of R_1 is restricted. These restrictions are such that the error in the formular for η_p given in 4.2.1.2 is not greater than 0.1 db. On this basis the value of R_1 should be selected to meet the following requirements :

(1) Be equal to or smaller than 0.01 times the input impedance of the attenuator. This will insure that the voltage across the input of the attenuator is equal to the generator source voltage E within 1 percent.

(2) Be equal to or smaller than 0.01 times the absolute value of the output impedance of the oscillator plus any resistance placed in series with it. This will insure that the reactance of the oscillator will not affect the current passing through the earphone under test by more than 0.25 percent.

(3) Be equal to or smaller than the preferred source resistance, R_0, for the earphone under test.

R_2 should be of such a value as to properly terminate the attenuator, and be placed in the low side of the microphone circuit. For convenience, the combined resistance of R_2 and the attenuator in parallel should have a negligibly small value compared to the sum of the microphone impedance and the input impedance of the microphone amplifier.

To improve the frequency characteristic or waveform of the oscillator, a suitable resistance (not shown) which is much larger than R_1 may be added directly in series with the oscillator to provide it with its proper load, but the a-c voltmeter is always connected directly across R_1.

Since the different characteristics of an earphone will not be exactly the same at all levels, the pressure at which it is tested should be definite and typical of the conditions of use. For general-purpose earphones the standard testing level shall be 20 dynes per cm². With the pressure response p_m of the microphone known in terms of the absolute values of the r.m.s. open-circuit voltage E_m and the r.m.s. pressure p applied to the diaphragm, the voltage E_m developed by the microphone at the standard level must be computed from the equation

$$20 \log_{10} E_m = p_m + 26 \text{ db above 1 volt} \tag{6}$$

4.2.1 *Procedure for determining the available power response.* — Before proceeding as outlined below, it will be advisable to check the calibration of the various attenuators used, as a function of frequency and of applied voltage.

4.2.1.1 *Step 1.* — Set the oscillator frequency at 1,000 cycles por second, place switch S in position A and adjust the oscillator output and the attenuator setting to obtain a voltage E_1 across R_2 equal to the voltage E_m comuted from equation (6). Observe the output indicator reading.

Step 2. — Adjust R_3 so that the sum of R_3 and the resistive component of R_1 together with the output impedance of the oscillator in parallel is equal to R_0. Then with switch S in position B, adjust the oscillator output to obtain the same indicator reading as in Step 1 above. This establishes at 1,000 cycles per second an acoustical pressure of 20 dynes per cm² in the coupler due to the earphone.

Note. — When the preferred source resistance R_0 is not known, it may be necessary to obtain the available power response frequency characteristic for a sufficient number of choices of R_3 to determine a preferred value.

Step 3. — With the oscillator unchanged, and with switch S in position C, note the voltage across R_1 which will be the generating source voltage E. In the procedure that follows, this voltage is kept constant with frequency and, therefore, the available power,

$$\frac{E^2}{4R_0}$$

als remains constant. The equipment is now set up to determine the available power response at any frequency.

Note. — If the preferred source resistance R_0, is such that R_3 is at least 100 times larger than R_1, then R_3 may be used instead of R_0 in computing the available power.

Step 4. — Set oscillator at desired frequency. Place switch S in position C and adjust oscillator output to give same voltage E as in Step 3.

Step 5. — With oscillator unchanged, place switch S in position B and observe output indicator reading.

Step 6. — With oscillator unchanged, place switch S in position A and adjust attenuator to give same output indicator readings as in Step 5. This gives the value of

$$\eta_a = 20 \log_{10} \frac{E}{E_1} \tag{7}$$

4.2.1.2 *Calculations.* — All information necessary to calculate the available power response at the frequency of Step 4 has now been obtained. Since the available power response is given from equation (1) since from the microphone calibration

$$p = E_1 \, 10^{-\frac{p_m}{20}}$$

and from the attenuator reading

$$E = E_1 \, 10^{\frac{\eta_a}{20}}$$

the available power response is given by

$$\eta_p = 10 \log_{10} 4R_0 - p_m - \eta_a \tag{8}$$

To obtain the available power response at other frequencies, the procedure starting at Step 4 is repeated for each of the desired frequencies.

4.3 Applied voltage response

The same test apparatus and same circuit arrangement as described in 4.2, Available Power Response, may be used for measuring the applied voltage response except that the variable resistor, R_3, is short-circuited.

4.3.1 *Procedure for determining applied voltage response*

4.3.1.1 *Step 1.* — This step is identical with Step 1 of 4.2.1.1.

Step 2. — Place switch S in position B. Adjust output oscillator to obtain same indicator reading as in Step 1. This establishes at 1 000 cycles per second an acoustical pressure of 20 dynes per cm² in the coupler due to the earphone. Note the voltage E_e across R_1 and, therefore, across the earphone. This voltage is kept constant with frequency.

Step 3. — Set oscillator at desired frequency. Retain switch S in position B and adjust oscillator output to give the same voltage across R_1 as in Step 2 above. Note the output indicator reading.

Step 4. — Place switch S in position A, and adjust the oscillator output to give the same voltage across R_1 as in Step 3. Adjust the attenuator to give the same output indicator reading as in Step 3. This gives the attenuator reading

$$\eta_a = 20 \log_{10} \frac{E_e}{E_1} \tag{9}$$

4.3.1.2 *Calculations.* — All information necessary to calculate the applied voltage response at the frequency of Step 3 has now been obtained. Since the applied voltage response is given by

$$\eta_v = 20 \log_{10} \frac{p}{E_c}$$

and since

$$p = E_1 10^{-\frac{p_m}{20}}$$

and

$$E_e = E_1 10^{\frac{\eta_a}{20}}$$

we have for the applied voltage response

$$\eta_v = -r_m - \eta_a$$

To obtain the applied voltage response at other frequencies, the procedure starting at Step 3 is repeated for each of the desired frequencies. The procedure outlined above can be simplified if R_1 can be made small in comparison to Z_e. If R_1 is equal to or smaller than 0.01 times Z_e at all frequencies of interest, the voltage across R_1 romains equal within 1 percent for switch positions A and B, thereby eliminating the need for adjusting the oscillator output.

4.4 Output-input characteristic

At any given frequency, the output-input characteristic can be measured with the same test apparatus and test circuit shown in Figure 7.

4.4.1 *Procedure for determining output-input characteristics*

4.4.1.1 *Step 1.* — Set the oscillator to the desired frequency. With the switch S in position C, apply a voltage E across R_1 and read its value by means of the a-c voltmeter. The available input power P, in terms of the source voltage E, and the preferred source resistance R_0 is

$$P = 10 \log_{10} \frac{E^2}{4R_0} \text{ db referred to 1 watt}$$

Step 2. — With the oscillator unchanged, and switch S in position B, observe the output indicator reading. Observe also the voltage across the earphone. This relates the earphone voltage to the available input power and is used in determining the maximum allowable voltage to be applied to the earphone when making impedance measurements.

Step 3. — With the oscillator still unchanged, and switch S in position A, adjust the attenuator to give the same indicator reading as in Step 2. This gives the value of

$$\eta_a = 20 \log_{10} \frac{E}{E_1}$$

4.4.4.1 *Calculations.* — The sound pressure level is calculated as follows

$$L_p = 74 + 20 \log_{10} p$$

since

$$p = E_1 10^{-\frac{p_m}{20}}$$

and

$$E_1 = E 10^{-\frac{\eta_a}{20}}$$

$$L_p = 20 \log E - \eta_a - p_m + 74 \qquad (11)$$

4.4.1.3 Repeat Steps 1, 2 and 3 with a suitable range of values of available input power. Generally, the information thus obtained is plotted using the values of the logarithm to the base 10 of the available input power, P, as abscissae, and the values of the sound pressure level, L_p, as ordinate. This curve, in addition to giving the approximate amount of undistorted sound pressure level L_p, which the earphone can generate, is useful in determining the maximum available power and voltage which may be applied to the earphone in making impedance measurements.

4.5 Electrical impedance

The following apparatus may be used for measuring the electrical impedance :

(a) A variable frequency oscillator,
(b) A suitable impedance bridge with a detector-amplifier and detector,
(c) An appropriate coupler to act as an acoustical termination for the earphone,
(d) An a-c voltmeter to measure the voltage across the earphone,
(e) A standard pressure microphone as specified in American Standard Specification for Laboratory Standard Pressure Microphones, Z.24.8.1949, or any subsequent edition thereof approved by the American Standards Association, Incorporated.

4.5.1 *Procedure for determining electrical impedance*

4.5.1.1 *Step 1.* — Determine by inspection of various output-input characteristics at different frequencies, the maximum voltage which may be applied to the earphone without serious distortion. With the earphone connected in the bridge arm as the unknown, adjust the oscillator output at a particular frequency to give a voltage the earphone not greater than the value determined above.

Step 2. — Remove the a-c voltmeter from the circuit and balance the bridge. This measurement determines the resistance and reactance and hence the impedance and phase angle of the earphone at the frequency at which the measurement is being made.

After balance is obtained, check the voltage across the earphone to insure that it is at the desired value.

4.6 Component harmonic distortion

For measuring the component harmonic distortion, replace the amplifier and output indicator of Figure 7 by a wave analyzer as shown in Figure 8. If the oscillator output is not sufficiently free of distortion to permit the measurement of harmonics in the earphone output to the desired degree of accuracy, add an adjustable low-pass filter to the measuring circuit as shown.

4.6.1 *Procedure for determining component harmonic distortion.* — The procedure to be employed in making harmonic distortion measurements is outlined in detail below. Depending on whether the component harmonic distortion is being measured in terms of the voltage across the earphone, or available power, the procedure indicated in 4.6.1.1 *(a)* or 4.6.1.1 *(b)* should be followed.

4.6.1.1 *Step 1 (a).* — If a harmonic measurement is to be made with a specific sinusoidal voltage applied to the terminals of the earphone, short-circuit R_3, set the oscillator at the required frequency, adjust the low-pass filter to pass the fundamental oscillator frequency and to suppress all of the harmonic frequencies, place switch S in position B and adjust the voltage E_e to the required value.

Step 1 (b). — If a harmonic measurement is to be made with a specific available power, P, from a source of specific impedance, adjust R_3 as in Step 2 of 4.2.1.1 Set the oscillator at the required frequency. Adjust the low-pass filter to pass the fundamental oscillator frequency and to suppress all of the harmonic frequencies. Then with switch S in position C set the voltage E to the value determined from the following expression :

$$E = 2\sqrt{PR_0} \tag{12}$$

Step 2. — With the low-pass filter unchanged and with the switch S in position B, tune the wave analyzer to the desired harmonic frequency nf (n = 1, 2, etc.) and note the output reading of the wave analyzer.

Step 3. — With the wave analyzer unchanged, the filter by passed, and switch S in position A, set the oscillator to the frequency nf and adjust the attenuator to give the same analyzer output reading as in Step 2. Determine the voltage E_1 across R_2 in terms of η_a and the voltage across R_1.

4.6.1.2 *Calculations.* — The pressure p_{nf} at the harmonic frequency *nf*, can now be calculated. It is given by

$$p_{nf} = \left(E_1\, 10^{-\frac{p_m}{20}}\right)_{nf}$$

where the subscript nf on the right-hand member of the equation indicates that E_1 and p_m are evaluated at the frequency nf. p_f, the pressure of the fundamental frequency (i.e., for n = 1), is given by

$$p_f = \left(E_1 \, 10^{-\frac{p_m}{20}}\right)_f$$

The component harmonic distortion is, therefore, equal to

$$D_n = \left(20 \log E_1 - p_m\right)_{nf} - \left(20 \log E_1 - p_m\right)_f \qquad (13)$$

4.6.1.3 *Determination of limitations of measuring circuit.* — The component harmonic distortion determined as outlined in 4.6.1.2 is actually the combined harmonic distortion of the earphone under test and the measuring system. Since this is the case it is essential that the harmonic distortion of the system be sufficiently low with respect to that of the earphone to permit the earphone distortion to be measured to the desired accuracy. If the distortion of the system is known, the limiting accuracy applicable to the measurement of earphone distortion of a given magnitude can be readily determined. The system distortion may be determined as follows :

With the oscillator frequency, the low-pass filter, voltage E and resistor R_3 adjusted as required in connection with a particular measurement, and with switch S in position Bm tune the wave analyzer to the fundamental oscillator frequency f and observe the reading on the indicator forming a part of the wave analyzer. Throw switch S to position A and adjust the attenuator to obtain an analyzer reading essentially equal to that obtained immediately above. Then tune the harmonic analyzer to each of the various harmonic frequencies nf of interest and observe the corresponding analyzer output readings. If the gain of the microphone amplifier, measured in terms of the ratio of the output reading to the voltage E_1, is not the same for all of the frequencies at which readings have been observed, correct the readings to the values which would have been obtained had the gain of the amplifier at each of the harmonic frequencies been the same as that at the fundamental oscillator frequency. From these data determine the harmonic distortion of the measuring system and the limits in which measurements of harmonic distortion of earphones can be made.

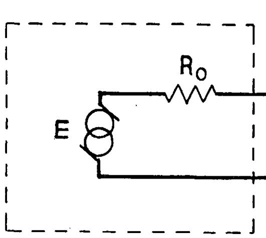


FIGURE 1*a*. — *Generator with available power* $\frac{E^2}{4R_0}$

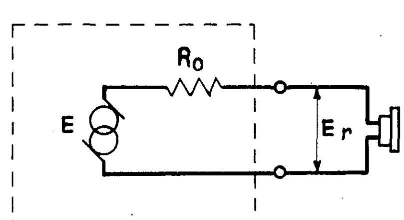


FIGURE 1*b*. — *Circuit for calibrating a receiver*

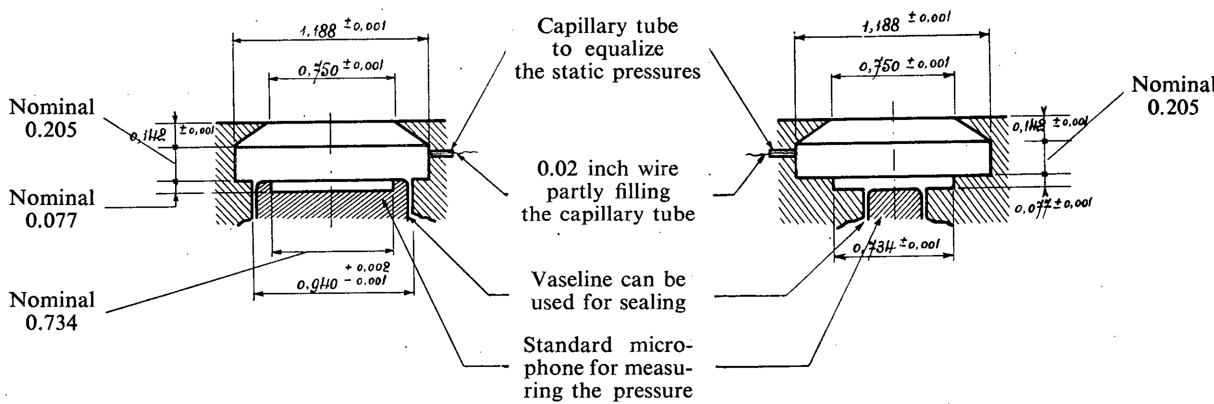
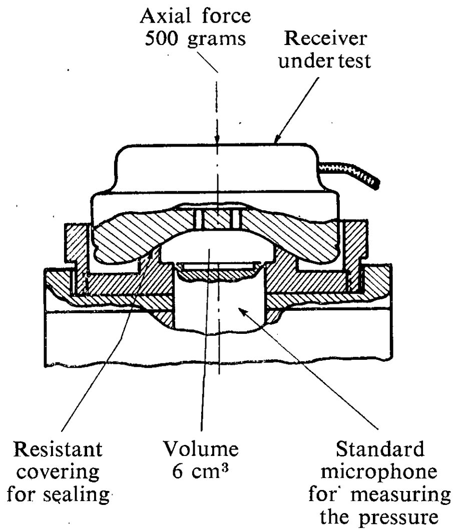


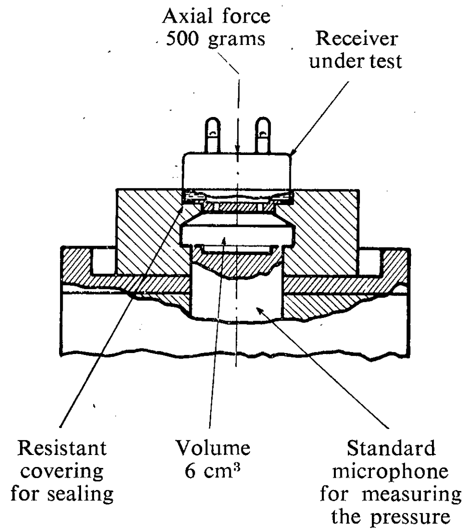
FIGURE 2a

FIGURE 2b

Notes 1. All dimensions in inches
2. All dimensions perpendicular to the axis are diameters

FIGURE 2. — *Coupler Type 1 for receivers applied to the ear cap*





Note — The capillary tube is not shown — See Figure 2

FIGURE 3. — *Coupler Type 1 modified for receivers with a hard ear cap*

FIGURE 4. — *Coupler Type 1 modified for small receivers*

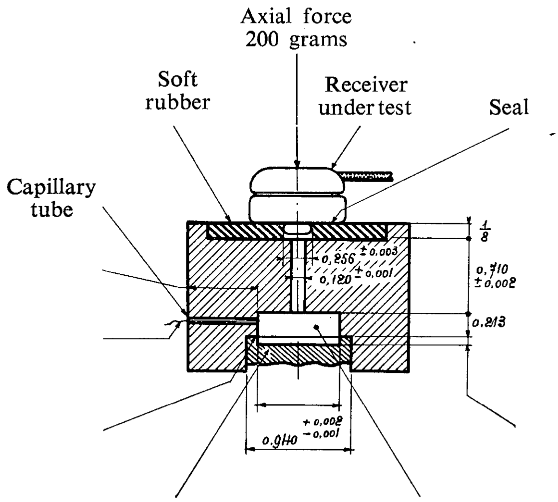


FIGURE 5. — *Coupler Type 2*

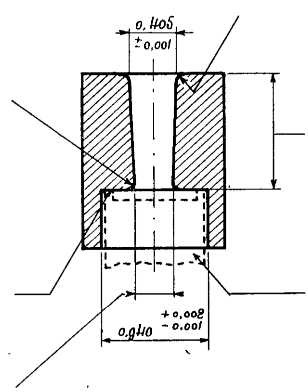


FIGURE 6. — *Coupler Type 3*

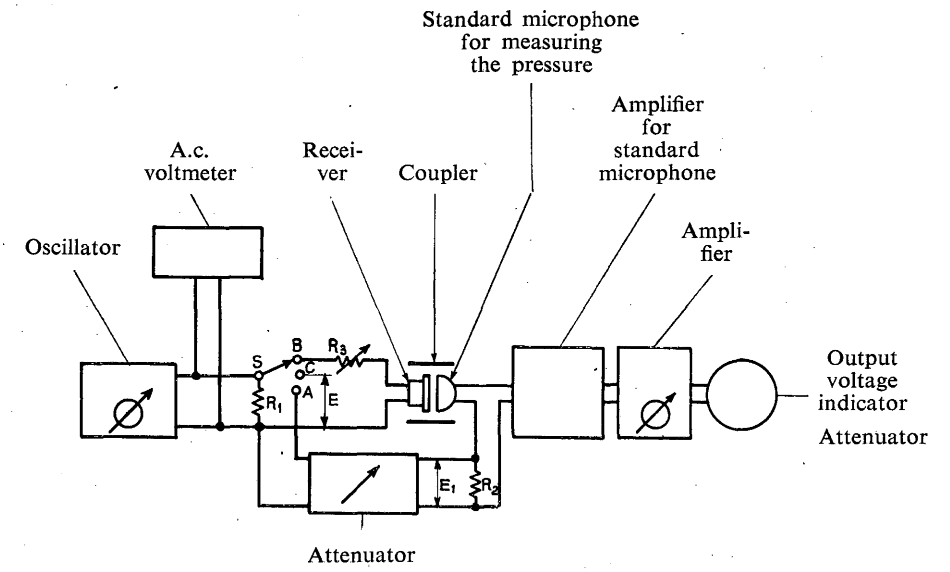


FIGURE 7. — *Circuit for measuring the power response, the voltage response and the transfer characteristic of a receiver*

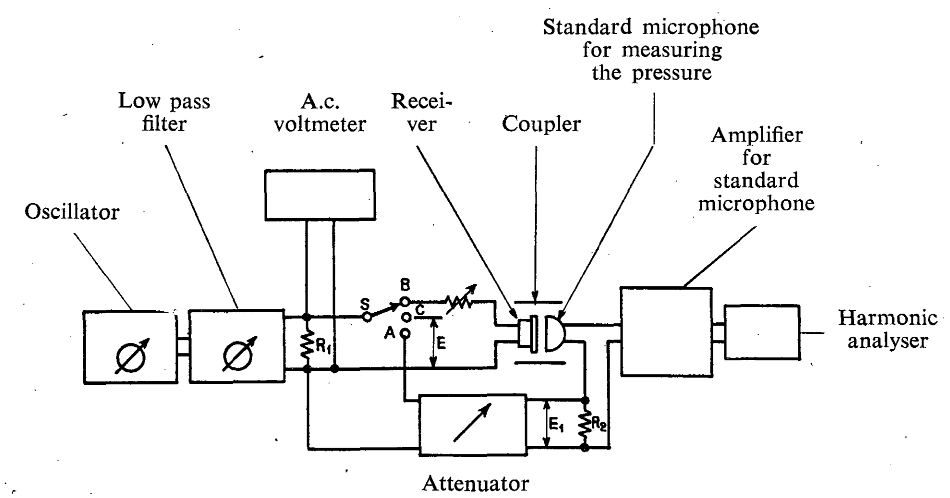


FIGURE 8. — *Circuit for measuring the harmonic distortion of a receiver*

ANNEX 7

ARTIFICIAL MOUTH, VOICE AND EAR USED BY THE FRENCH TELEPHONE ADMINISTRATION

The system which has been designed and constructed by the Centre National d'Etudes des Télécommunications (and an example of which has been given to the C.C.I.F. Laboratory) may not permit absolute measurements but at least enables a check to be made of the stability of the sensitivity or of the reference equivalent of a telephone connexion including the terminal subscribers' equipment. It is intended to replace, by objective equipment, the telephonometric operators who would normally conduct these measurements in a subjective manner. It is thus not intended for the absolute determination of the equivalent, although very encouraging progress has been made towards this goal, but only for the verification of its constancy. It comprises four essential parts :

1. An artificial mouth.
2. An artificial voice.
3. An artificial ear.
4. A logarithmically calibrated voltmeter.

The artificial mouth

The artificial mouth is an acoustical source which produces in front of the microphone a complex sound having the mean structure of the real voice and called an artifical voice. Furthermore the conditions of radiation correspond to those of the human mouth during a normal conversation at average level.

The model developed by the Centre d'Etudes des Télécommunications comprises a commercial permanent magnet electrodynamic loudspeaker with a 4'' diaphragm in conjunction with an acoustical network of channels in front of it. The cavity between the diaphragm and the end of the channels forms a compression chamber. The channels attenuate the loudspeaker resonances, reduce the eddies and concentrate the acoustic field towards the artificial mouth. The central opening of the diaphragm is closed with varnished cloth in the shape of a spherical cap giving the greatest rigidity with small mass. The termination and the central section which forms part of the arrangement of channels has been designed with regard to the acoustic field which is radiated. This arrangement to some extent imposes its own characteristics which feature renders the system less dependent upon the characteristics of the loudspeaker and allows the latter to be replaced.

The amplifier associated with the loudspeaker corrects the response curve of the latter by means of negative feedback with specially designed complex network. Figures 1 and 2 (transverse and longitudinal sections) give a view of the artificial head. Figures 3 and 4 show the variation of the acoustic field along the axis of the real mouth (full line) and of the artificial mouth (dotted line), together with their distributions around the head (with it being fed from the artificial voice). Very good results have been obtained for telephonometric measurements in comparison with mean values obtained with several operators.

The artificial voice

This equipment consists essentially of a current source with continuous and uniform spectrum, produced by a neon tube working in an unstable part of its characteristic ; it is polarized with direct current in series with a resistor and capacitor whose functions are to induce low frequency oscillations. This effect is supplemented by a simple filter circuit formed by the polarization of the tube and of the network of connections to the first stage of amplification associated with the tube. It is also possible to alter the general shape of the spectrum and to obtain at will, by operation of a key, a standardized form of mixed voice (male and female), or a uniform spectrum (white noise). Figure 5 gives the general circuit diagram of the equipment. It has been possible to simulate the curves published by Fletcher (Figure 6). An amplifier allows an output power of 6mW to be obtained in 600 ohms.

Telephonometric measurements conducted with this current source on telephone receivers having sensitivities varying from 2.3 db better to 23.1 db worse than the standard, and with very different response curves, have given results in agreement with those of normal telephonometric measurements. The individual variations between the two series of measurements are at the greatest 1.6 db which is no greater than the usual telephonometric variations.

To make the analogy between the artificial voice and the real voice more complete, a device has been added which to some extent adds to the continuous vowel sounds a relaxation modulation corresponding to consonants. This has been achieved by means of a thyratron the current of which passes through the primary of a transformer. The voltage produced by induction at the secondary terminals polarizes the grid of a valve amplifying the noise produced by the neon tube. The frequency of this modulation can be adjusted but is, for preference, set at 4 per second which is about the mean frequency for uttering consonants (for very slow speech, such as of a public speaker, 2 per second ; very fast speech, 8 per second). When measuring microphones the artificial voice is used in conjunction with an artificial mouth. There is provision for including an automatic interrupter giving 2 seconds interruptions every 6 or 7 seconds, so as to simulate the normal sequences of taking breaths and uttering continuous speech (variable from 3 to 13 seconds).

The artificial ear

The artificial ear is an assembly of acoustic impedances such that a telephone receiver placed against the outside surface (forming the opening of the ear) radiates in the same manner as if it were applied to an average real ear. In the case of the ear developed by the acoustics laboratory of the C.N.E.T., direct comparisons have been made between a fairly large number of real ears (44) and the system representing the artificial ear. This has been done by taking several earphones in the plane of the base of which were fitted axially a very small peizo-electric microphone. The impedance network of the artificial ear has been adjusted so that the combined characteristics of the earphones associated with the small microphone were practically identical when the earphone was applied close to the average real ear and against the artificial ear. Figure 7 gives the sectional plan view of the final arrangement.

The impedance of the auditory canal is represented by the volume contained by the cylindrical part of the coupler (1.2 cm^3). A leather cap provides in conjunction with the upper part of the coupler an additional volume of 2 cm^3.

The leaks necessary at low frequencies are provided by a channel 0.5 mm dia. and 1 cm long. The agreement of results obtained with the artificial ear and with real ears is shown by Figure 8 which gives a mean and extreme curves of sensitivity for a receiver related to the 44 real ears and to the artificial ear.

The eardrum is represented by the diaphragm of an electrostatic microphone. The whole is symmetrical around the axis of the figure. For calibration the receiver rests on the leather cap. Very satisfactory agreement has been obtained during telephonometric comparisons between sound received by this ear (successively associated with different receivers) and sounds received by the ear of an average operator through a receiver connected to a condenser microphone. This arrangement is so as to include the neuro-subjective appreciation of intensity of the brain. There is not yet any apparatus which can reliably replace the latter ; in the absence of this an amplifier voltmeter with logarithmic scale has been connected ; it consists of a stabilized linear amplifier followed by a special indicating instrument with pole pieces shaped so as to obtain a deflexion of the needle proportional to the logarithm of the current measured. It permits a direct reading over a range of 30 db. In addition an attenuator incorporated in the amplifier permits its gain to be varied continuously over a range of 50 db.

The three components whose principles have already been described have circuits and characteristics shown below. The relevant constructional diagrams are given on pages 86-92.

Note 1. — In addition to the above artificial mouth, voice and ear, the French Telephone Administration has available various analogous items of equipment, manufactured commercially, amongst which are the following :

1. An artificial voice having the same essential characteristics as the above voice but enabling other spectra to be obtained in addition and used for electro-acoustical or electrical measurements.

2. An assembly comprising an artificial mouth, voice and ear together with a measuring voltmeter, considerably simplified and lightened as compared with the equipment previously described. This portable equipment requires no external power supply ; it is primarily for maintenance of telephone sets in subscribers' homes.

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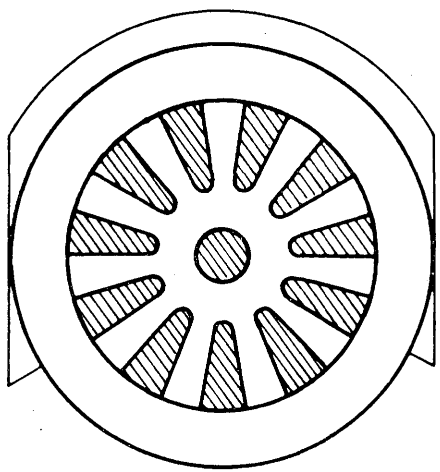


Section *a-b*

FIGURE 1

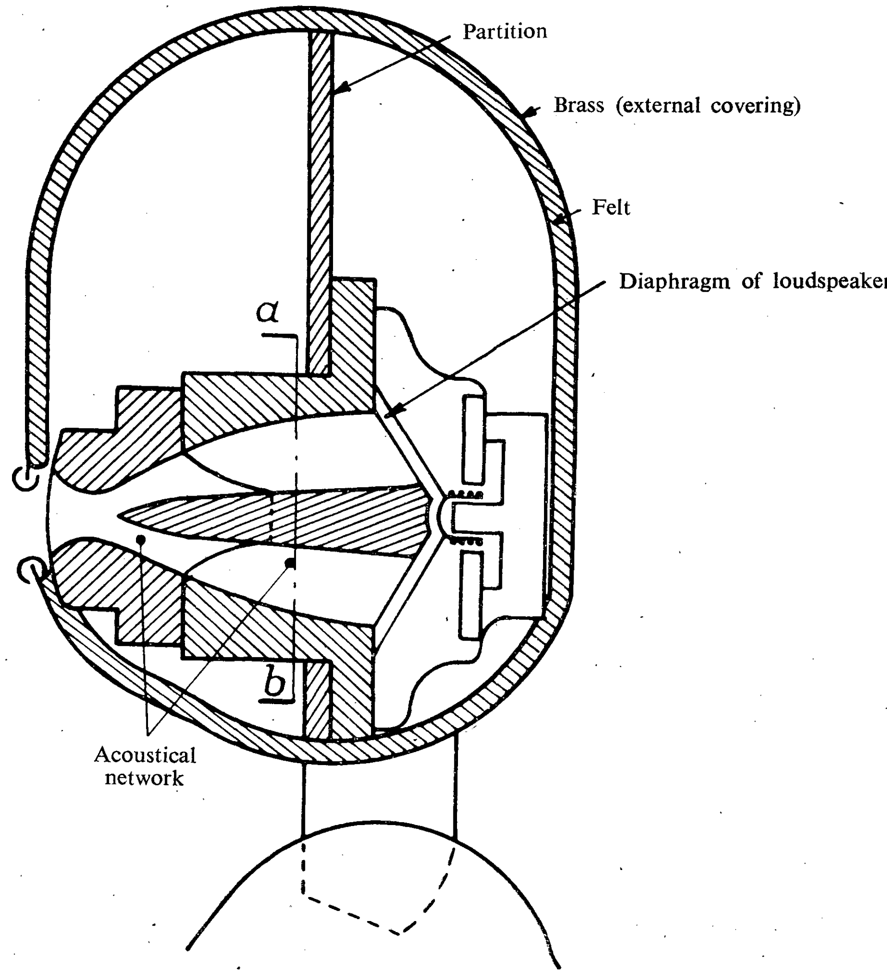


FIGURE 2

Artificial mouth

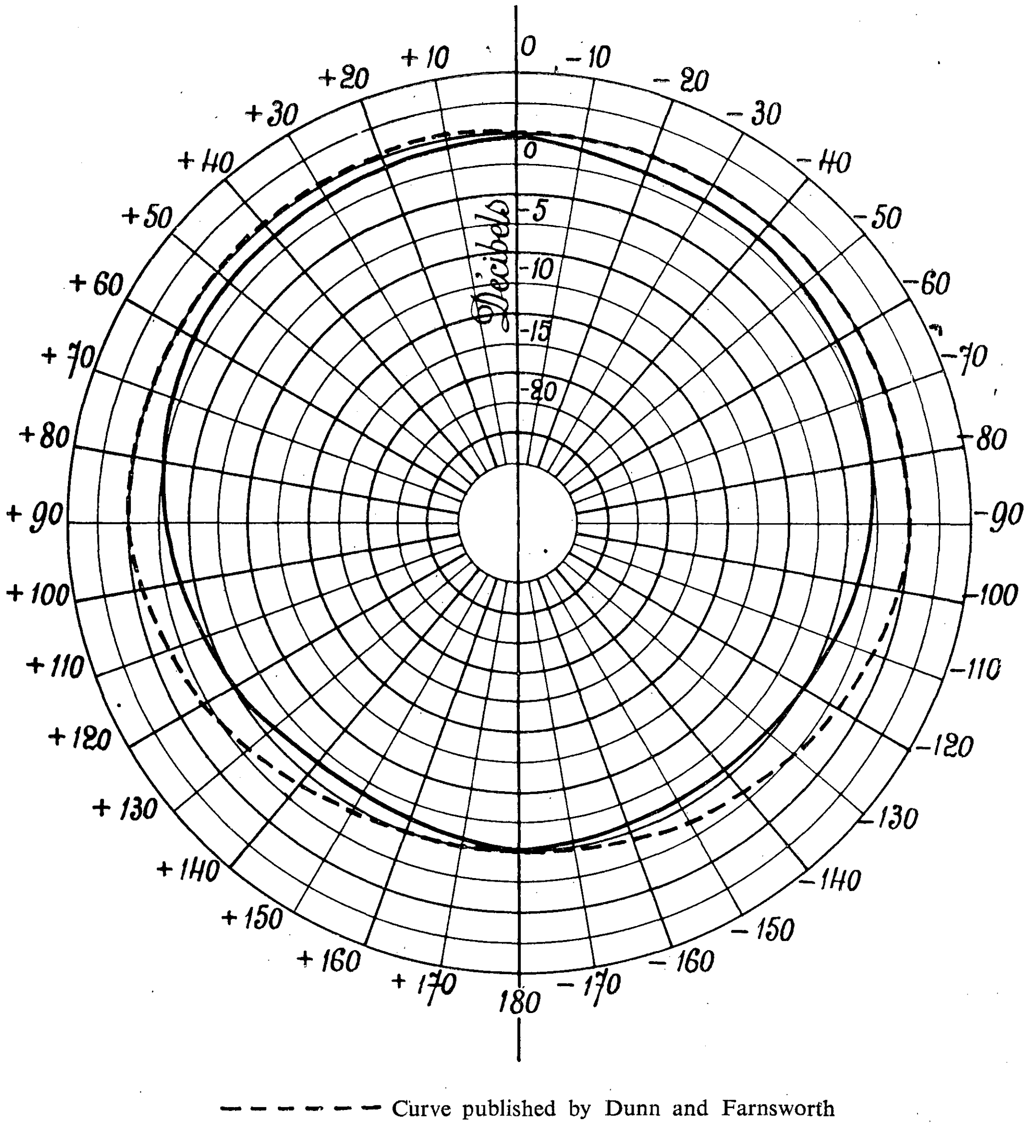


FIGURE 3. — *Artificial mouth fed from the C.N.E.T. artificial voice. Curve of directivity. Curve published by Dunn and Farnsworth*

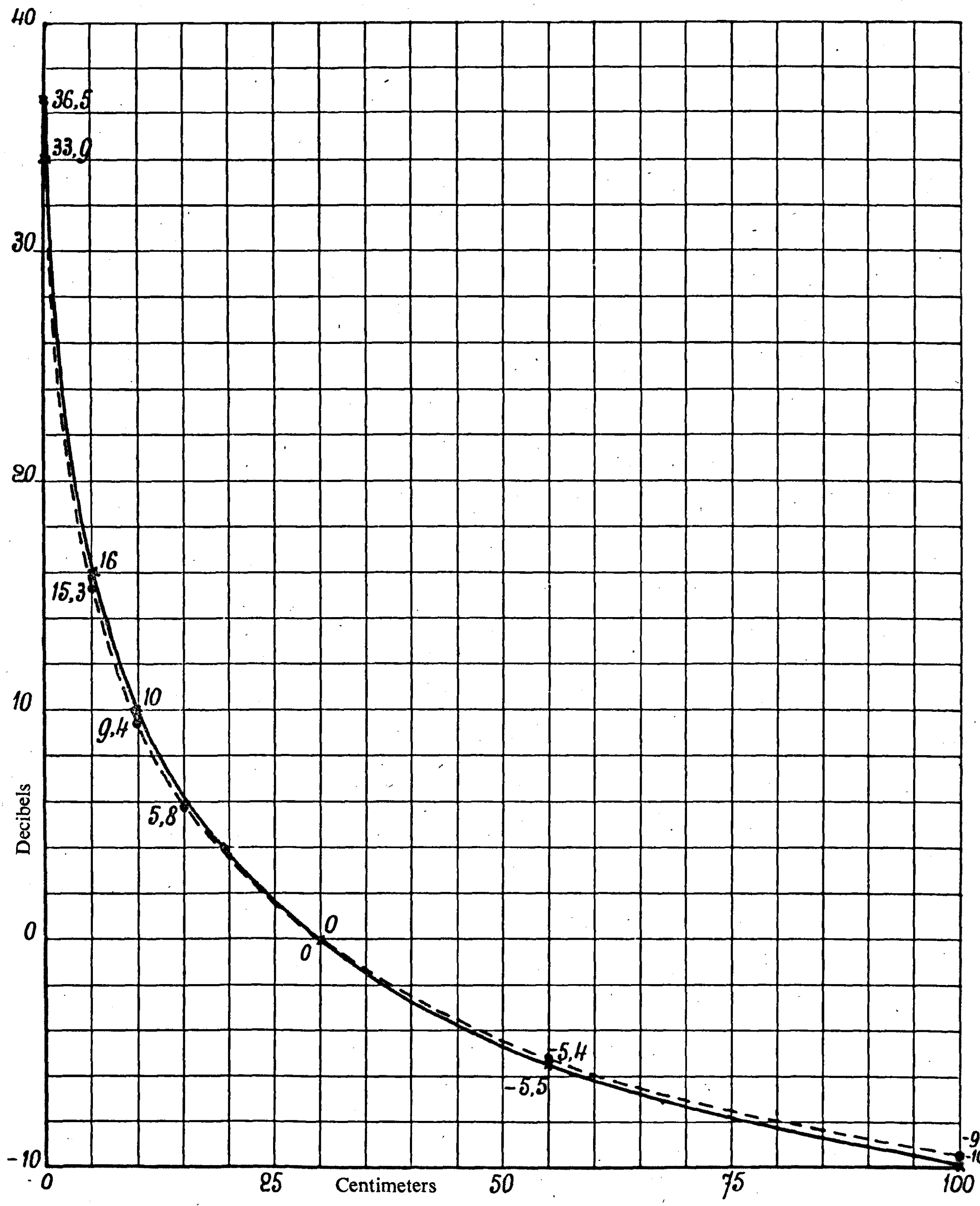
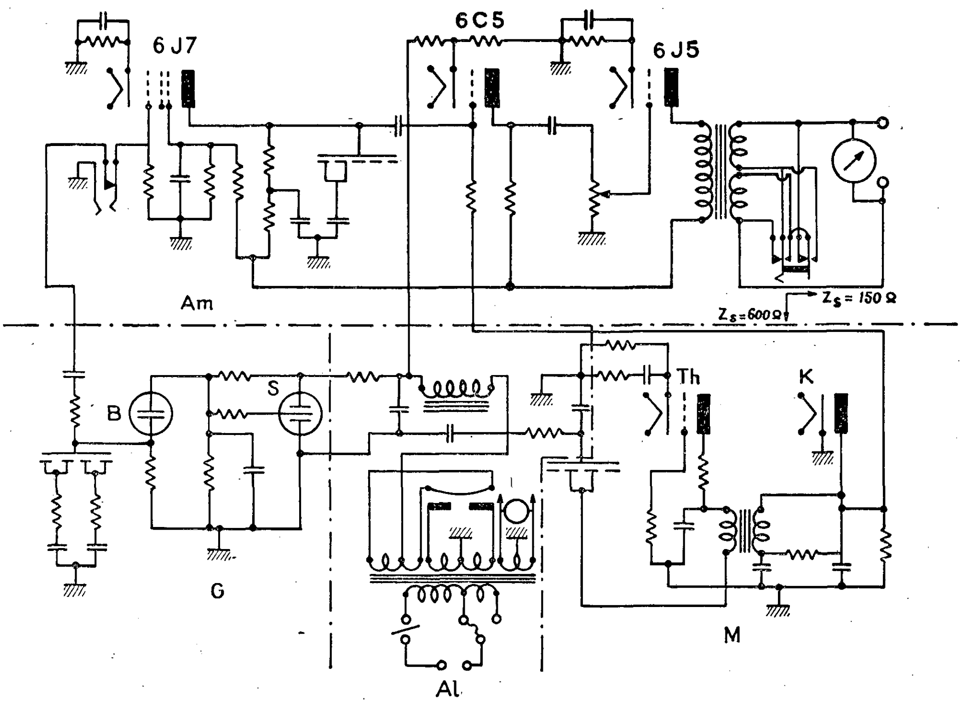
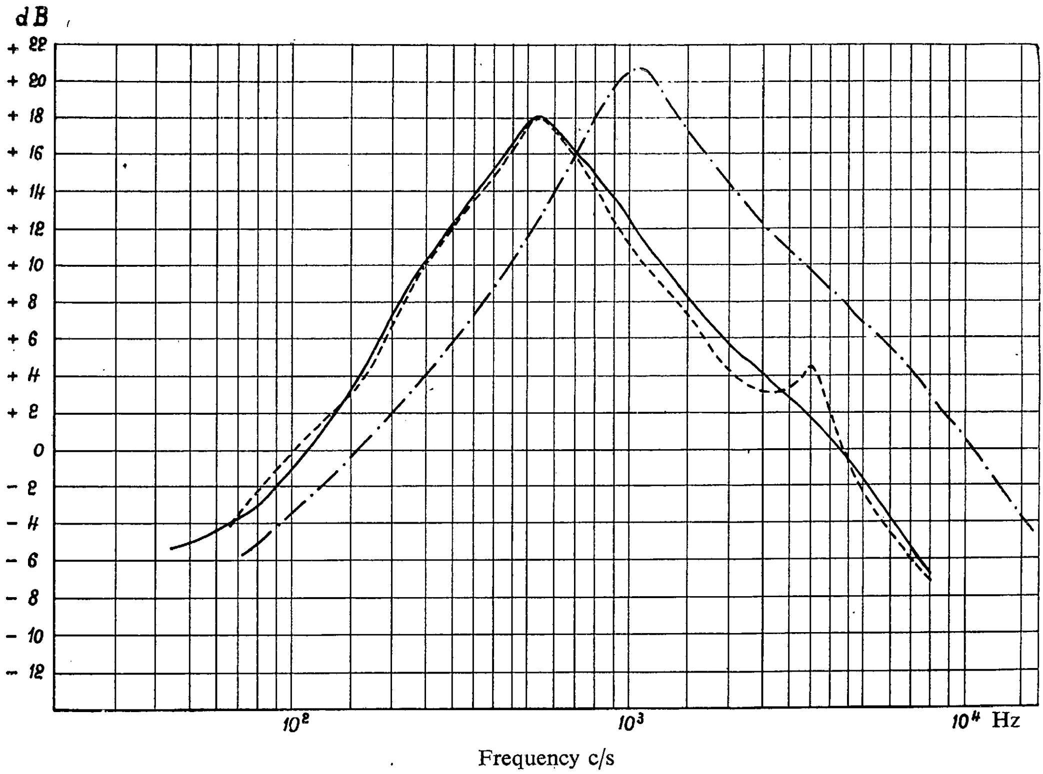


FIGURE 4. — *Artificial mouth fed from the artificial voice (male voice). Decrease in pressure along the axis*



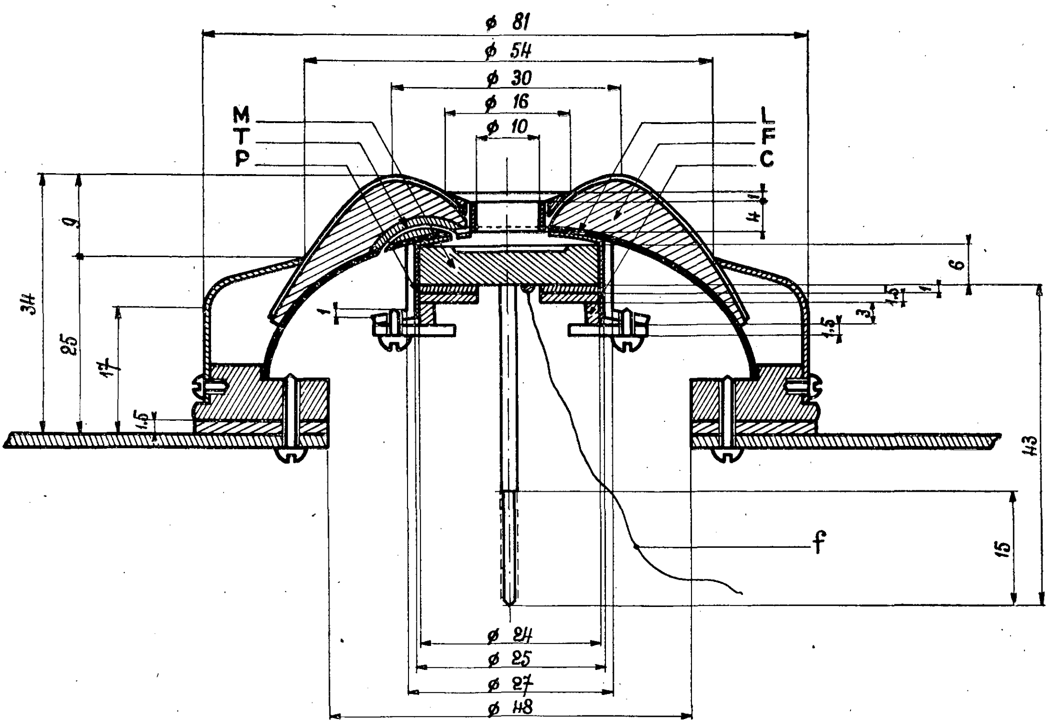
M = Modulator
Amp = Amplifier
V = Voltmeter
S = Stabilizor
K = Kenotron
G = Generator
Al = Power Supply
B = Noise source
Th = Thyratron

FIGURE 5. — *Artificial voice*



Male voice
Female voice
Curve given by Dunn and Farnsworth

FIGURE 6. — *Artifical voice. Spectrum of the output*



M = Electrostatic microphone
T = Tube internal dia 0.1 mm
P = Paper + varnished bakelite
L = Brass

F = Felt cup
G = Rubber
f = Earth wire

FIGURE 7. — *Artificial ear*

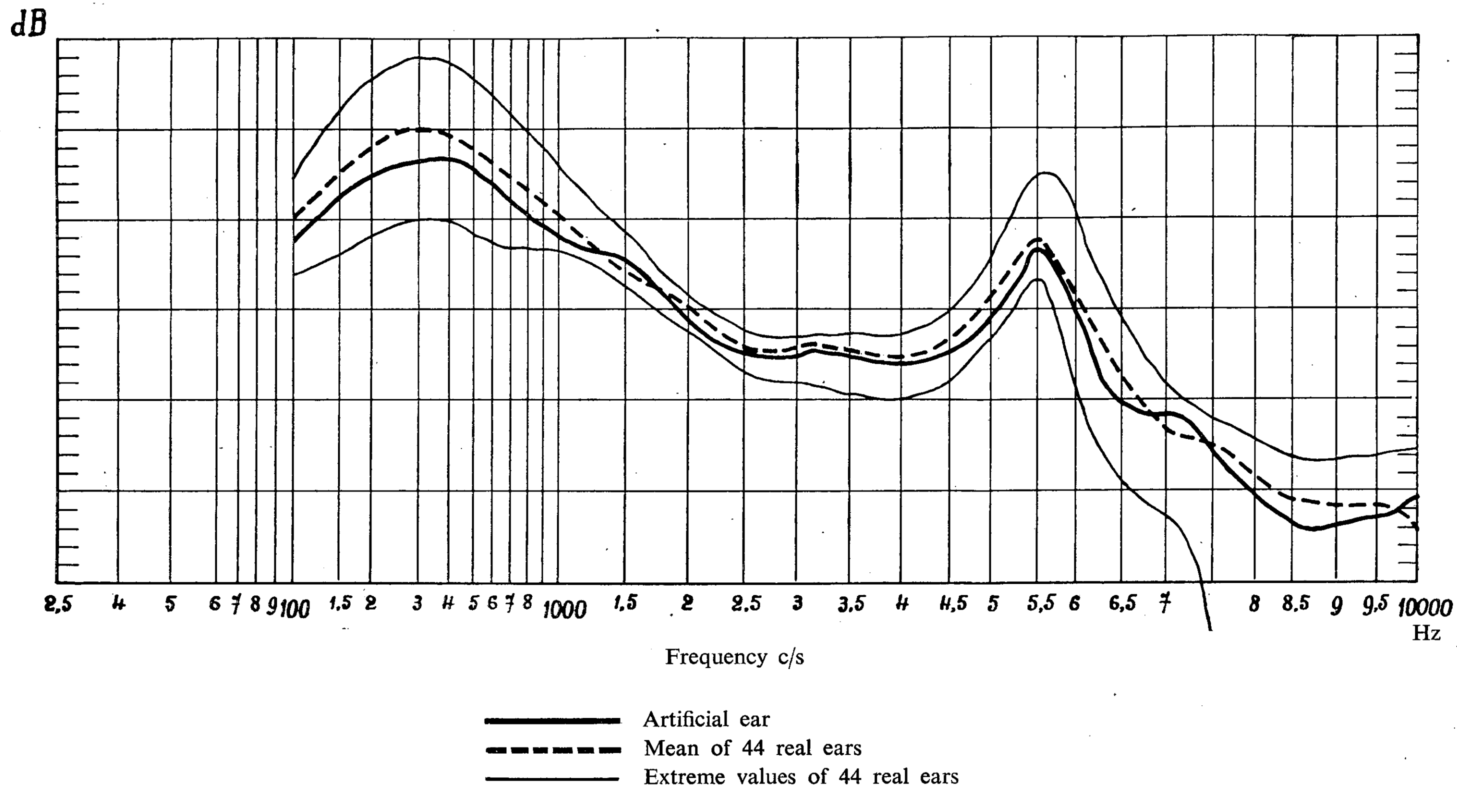

dB
2,5 4 5 6 7 8 9 100 1,5 2 3 4 5 6 7 8 1000 1,5 2 2,5 3 3,5 4 4,5 5 5,5 6 6,5 7 8 8,5 9 9,5 10000
Hz
Frequency c/s
Artificial ear
Mean of 44 real ears
Extreme values of 44 real ears

FIGURE 8

ANNEX 8

ARTIFICIAL MOUTH AND EAR USED BY THE TELEPHONE ADMINISTRATION OF THE FEDERAL GERMAN REPUBLIC

1. *The artificial mouth*

The artificial mouth consists of an amplifier associated with a loudspeaker. Figure 1 shows a section of the permanent magnet electrodynamic system, acoustical network and sound output opening. The acoustical part forms the artificial mouth proper. It is also provided with a ring, representing the lips and termed the " artificial lips ", which fixes the correct position of the apparatus being measured with respect to the sound source which represents the artificial voice. The diaphragm in the form of a spherical cap gives the greatest rigidity essential for continuous use over long periods.

The linear distortions of the loudspeaker are compensated by an electrical network within the associated amplifier (Figure 2). When using the equipment for measuring reference equivalents, this compensation is only necessary so far as it affects the measured results. For this reason it is sufficient, for measurements with the S.F.E.R.T. microphone, for the " sensitivity-frequency " characteristics to be constant within $\pm$ 3 db from 200 to 4 000 c/s. Non-linear distortion is less than 3 %. The absolute sensitivity of the artificial mouth (amplifier + loud speaker) is 37.6 dynes/cm^2 per volt when using, for measurement, the S.F.E.R.T. microphone in the position, with respect to the lips of the artificial mouth, fixed by the C.C.I.F. The artificial mouth is also capable of producing an acoustic pressure greater than 9 db without non-linear distortion. Correction of the distortion produced by the increase in acoustic pressure due to the S.F.E.R.T. microphone is provided in the amplifier of the artificial mouth. For measurements in a free acoustic field, this correction can be removed by means of a switch.

The small aperture loudspeaker radiates the artificial voice corresponding to the real voice as a spherical wave. The acoustic pressure decreases along the axis in the same manner (Figure 3) as in the case of a real mouth ; in the plane of the ring representing the lips, the source of the artificial voice corresponds to that of the real voice.

2. *The artificial ear*

The purpose of the artificial ear is to represent the input impedance of the real ear when using a telephone receiver applied to the ear. In order that this condition shall be fulfilled for earcaps of different types, the surface of the artificial ear must be shaped in an appropriate manner. For frequencies up to about 1 000 c/s the volume of the pressure chamber determines the acoustic resistance sufficiently. It has been determined that a volume of about 3 cm^3 for the artificial ear corresponds well with that of the real ear. At higher frequencies the shape and acoustic attenuation of the pressure chamber also plays a role in representing the acoustic resistance of the ear.

Figure 4 shows a sectional view of the damped cavity artificial ear used in the equipment for the objective measurement of reference equivalent. The surface is conical in order to permit a perfect fit for the various telephone receiver earcaps. The cavity consists of a cylindrical tube at the end of which the acoustic pressure is measured with a small condenser microphone. To represent the leaks in the cavity of the real ear, 16 attenuation channels of different lengths are situated at the end.

The artificial mouth and artificial ear have been so arranged as to be used in the equipment for the objective measurement of reference equivalent of the Federal German Administration (see Annex 21 below). The principle of measurement and the construction of the apparatus enable all the measurements to be made according to the C.C.I.F. recommendation particularly measurements of reference equivalent relative to the S.F.E.R.T. The results of measurements are given by direct readings. Furthermore a graphic level recorder provides an immediate visible indicator of the "sensitivity-frequency" characteristic.

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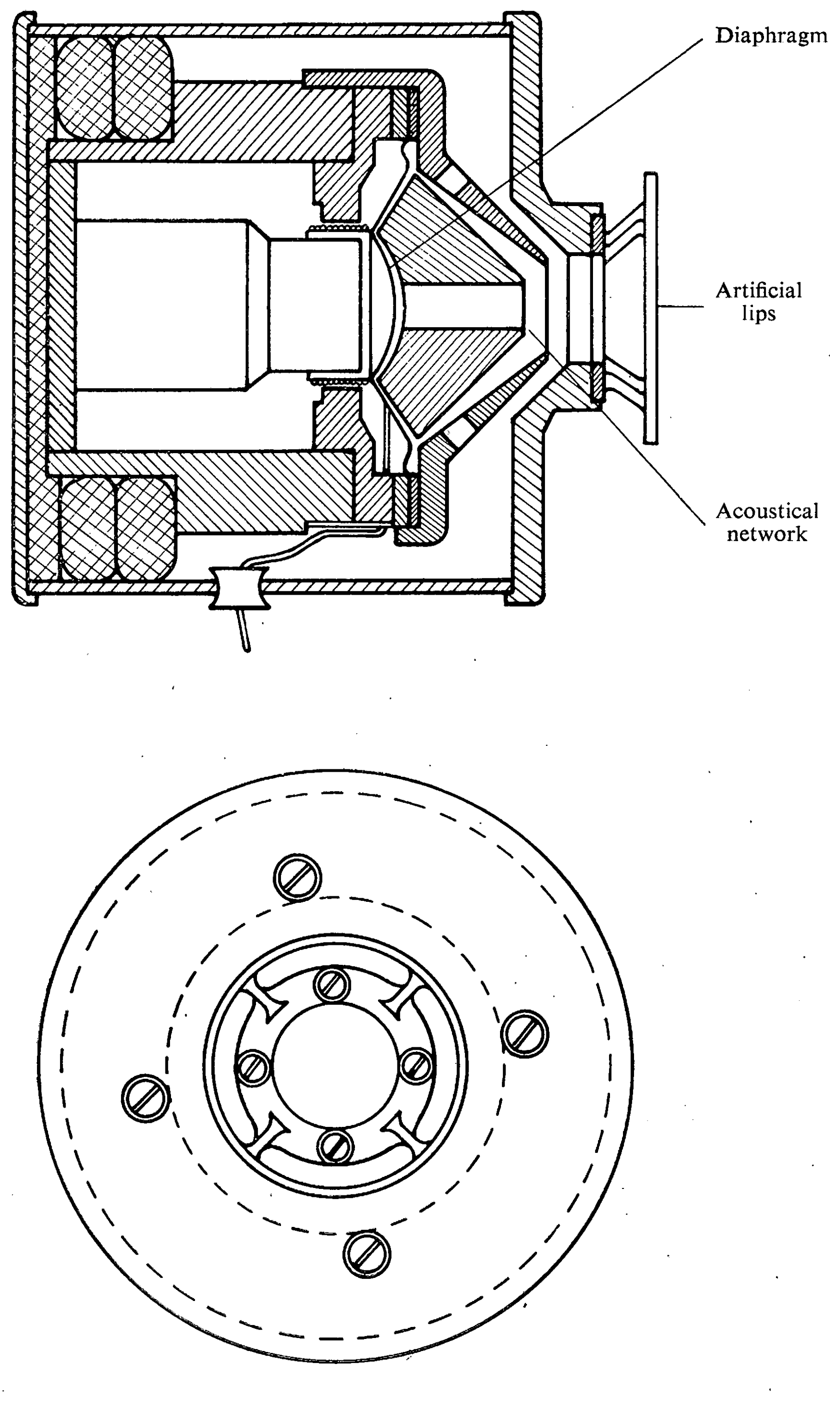


FIGURE 1. — *Artificial mouth*

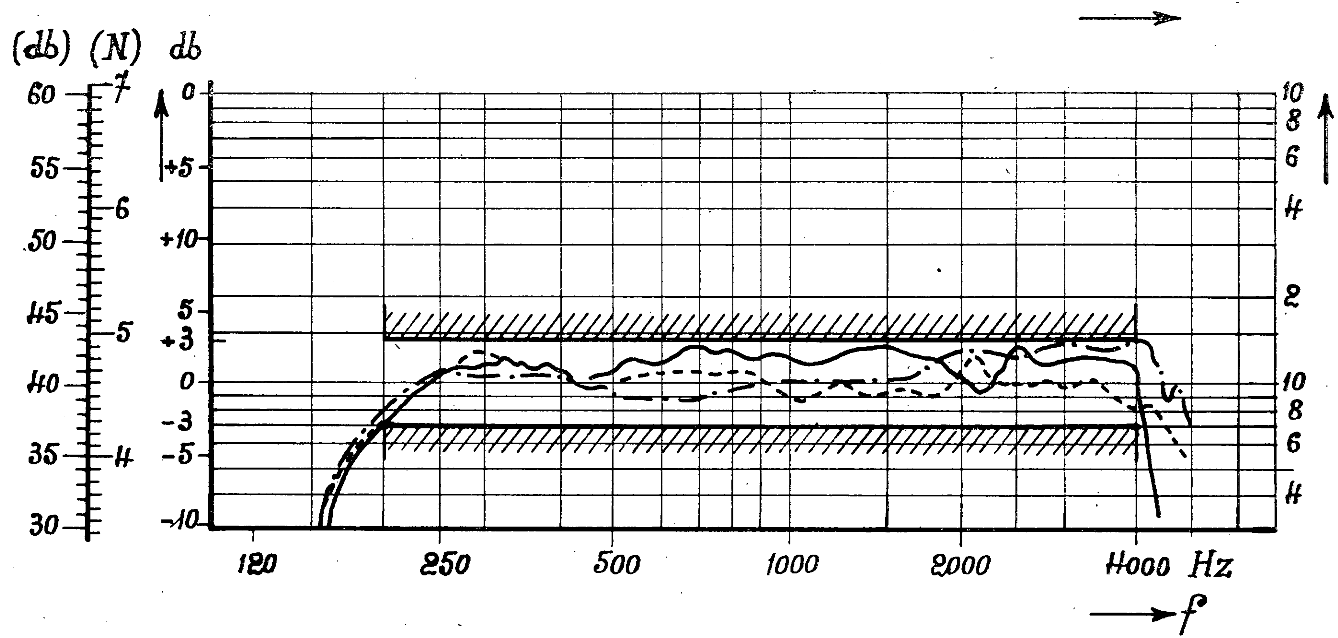


FIGURE 2. — " *Sensitivity-frequency* " *characteristic of the artificial mouth, relative to the S.F.E.R.T. microphone, for 3 examples*

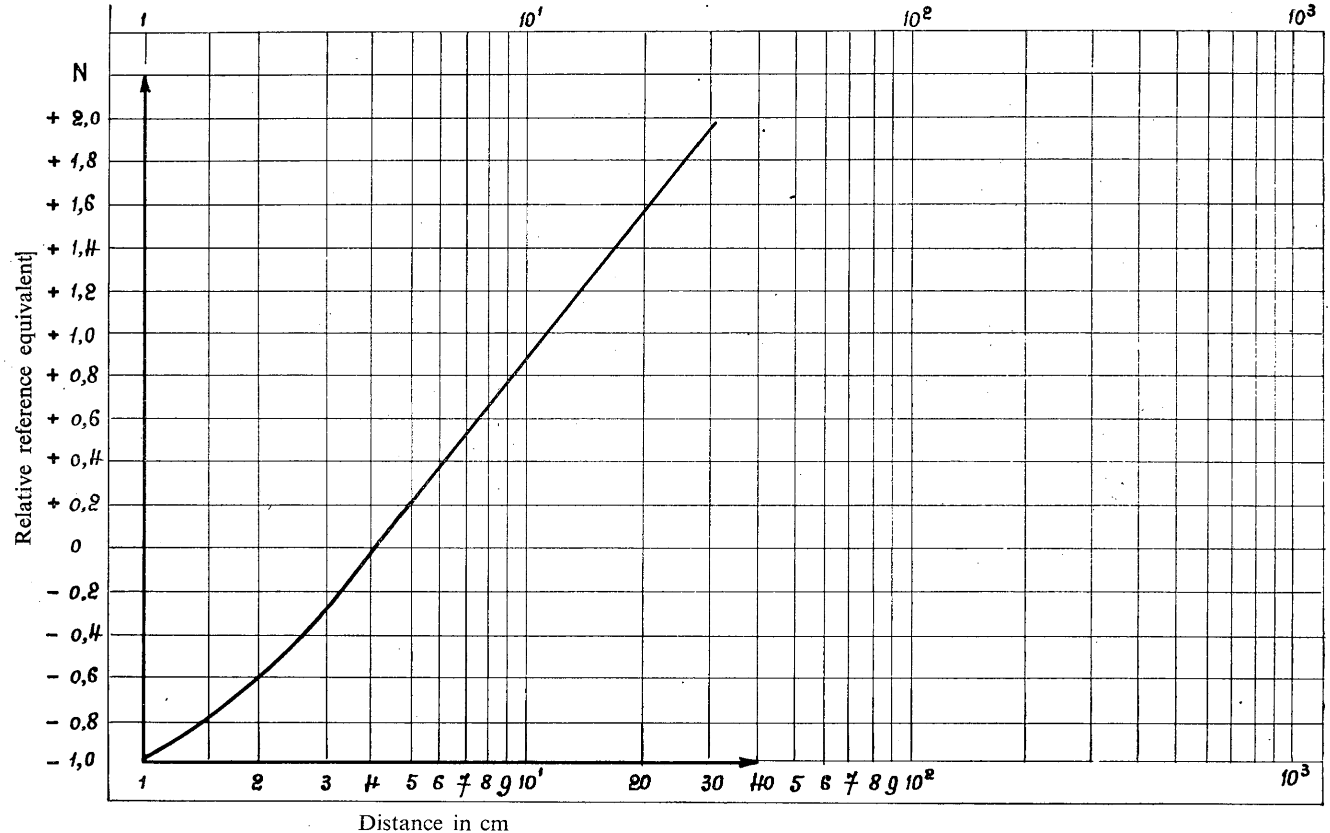
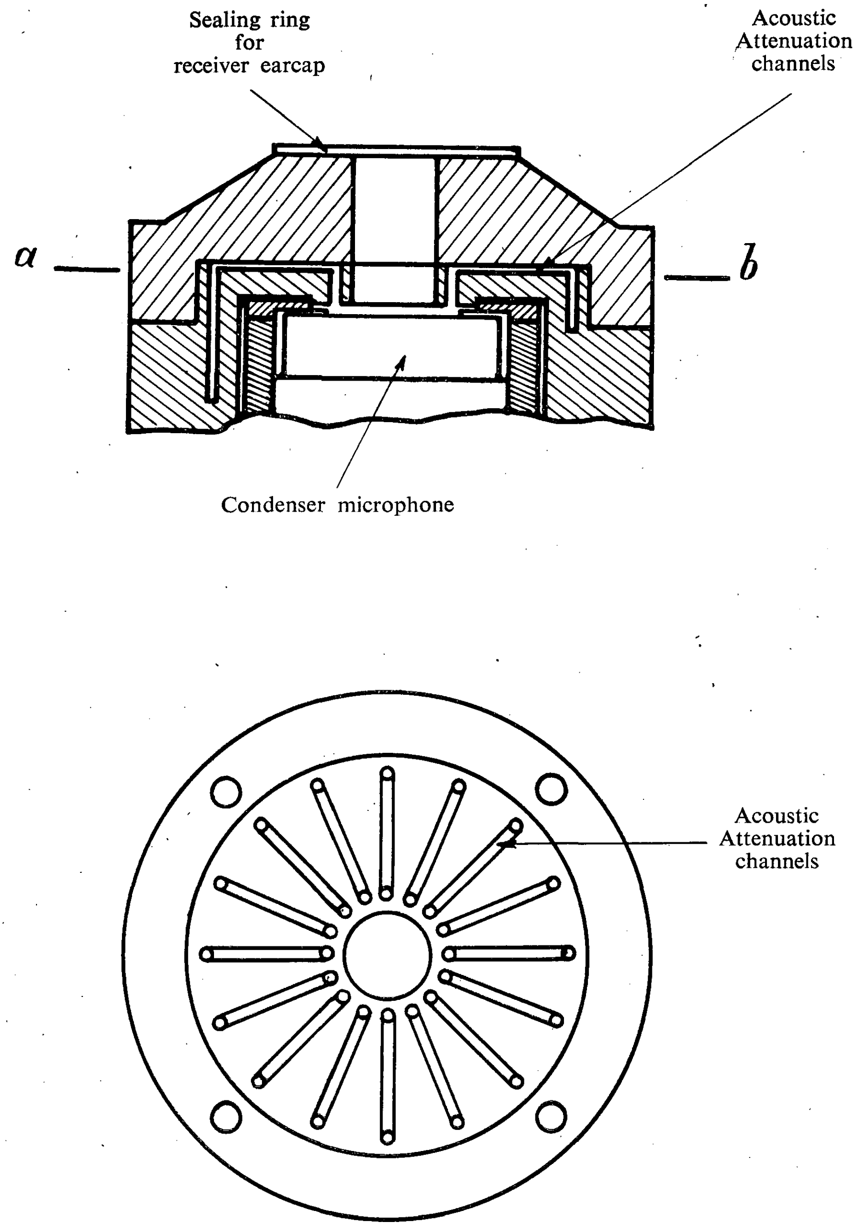


FIGURE 3. — *Variation, as a function of distance, of the acoustic pressure on the axis of the artificial mouth*



Section *a-b*

FIGURE 4. — *Artificial Ear*

ANNEX 9

ARTIFICIAL EAR USED BY THE BRITISH TELEPHONE ADMINISTRATION

1. General

The artificial ear used by the British Telephone Administration for calibration of external receivers is designed to present an acoustical impedance equivalent to that of a human ear [1] when a perfect seal is maintained by the receiver at the ear. The simulation of an acoustical leak between the receiver and ear, which exists in practice, is not made on account of its variability and dependence on the type of receiver used. It has been found that by using the impedance match as defined, comparisons between the sensitivities of similar forms of receivers made with the artificial ear conform closely to the same comparisons made on real ears. [2] [3].

The artificial ear, which conforms to British Standard 2042 : 1953, consists essentially of a volume reactance, in the form of a cavity, combined with an acoustical resistance. The receiver is coupled to the cavity and the sound pressure developed for a known excitation of the receiver is measured by means of a calibrated probe microphone.

2. Construction

The assembly of the essential elements of the artificial ear is shown in Figure 1.

2.1 *Cavity and Resistance Tubes.* — The cavity is made in brass to avoid magnetic coupling and has the shape of a frustum of a right circular cone of depth 1.0 cm and end diameters 1.1 and 2.5 cms. The acoustical resistance element, in the form of two copper tubes each of 0.475 cms internal diameter and 450 cms length, leads out from the smaller end of the cavity. In order to realize the resistive impedance required, each tube contains several strands of wool graded in length and arranged so that the density of the wool progressively increases towards the ends of the tubes remote from the coupler. The number of strands depends upon the quality of wool ; for the apparatus supplied for use with the A.R.A.E.N. seven strands of three-ply wool are used of lengths 25, 50, 75, 100, 125, 150 and 180 cms.

Provision is made for the tip of a probe microphone to be inserted at the shallow end of the cavity as shown in Figure 1.

2.2 *Probe Microphones.* — The essential properties of the probe microphone are that it must cause negligible disturbance of the sound field and in addition present a relatively high acoustical impedance to the small cavity and shunt resistance tubes.

These conditions are fulfilled in the fine bore probe microphone Figure 2, which has an external probe diameter of 0.125 cms and an impedance at the tip of about 9 000 acoustical ohms largely resistive.

The basis of the microphone is a moving coil unit housed in a heavy steel case to give protection from external magnetic fields and noise sources. A brass cover shaped to the profile of the microphone diaphragm and spaced about 0.03 cms from it couples the fine air clearance so obtained to the narrow bore probe.

In the model used for the A.R.A.E.N. testing equipment a small amount of lambs' wool is introduced at each end of the tube to damp the natural resonances of the tube and so produce a substantially constant sensitivity/frequency characteristic in the range 70 c/s. — 3 000 c/s. Above 3 000 c/s the sensitivity falls smoothly as shown in Figure 3. Since these microphones were introduced experiments have shown [2] that it is possible to obtain a relatively constant sensitivity over a much wider frequency range (Figure 3 dotted curve) by suitable choice of coupler dimensions i.e. tube termination, without introducing any form of damping into the tube.

2.3 *Assembly.* — The tubes are coiled on a 20 cms diameter and housed in a square wooden box. A second inner cylindrical compartment of the box is used to take the probe microphone. With this arrangement it is possible to bury the tubes in sand and so reduce interference from extraneous noise and bench vibrations to negligible proportions and at the same time allow for easy access to the probe microphone.

3. Receiver termination

For receivers with shallow earcaps of the type used with most commercial telephone circuits, the earcap is sealed directly to the top of the artificial ear coupler. In the case of the A.R.A.E.N. receiver with rubber earpad, which encloses a large air volume on the human ear, agreement with real ear calibrations is obtained when the earpad is sealed on a flat metal plate mounted flush with hte top surface of the artificial ear coupler.

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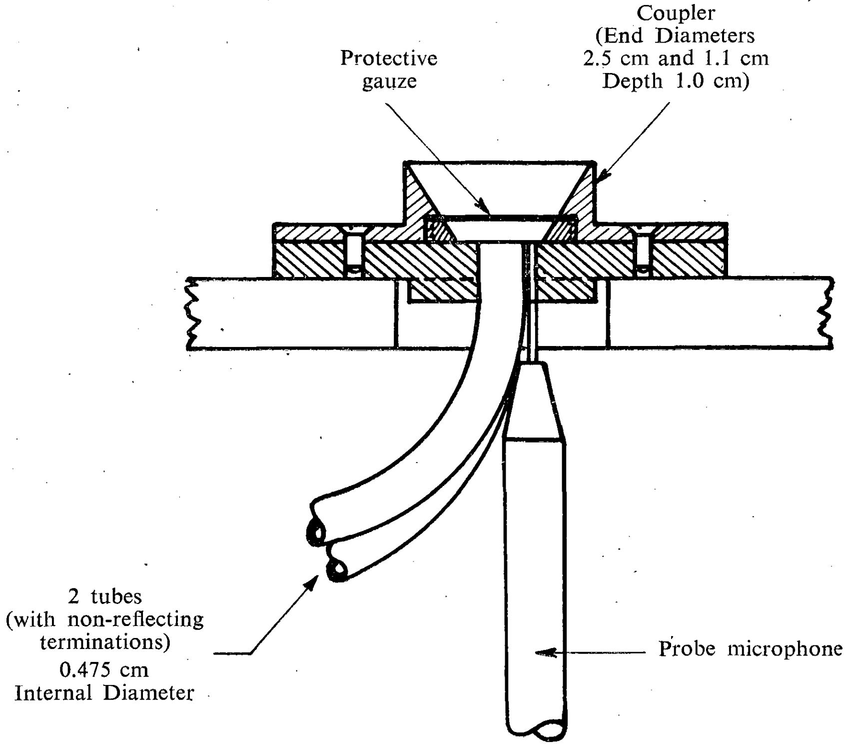


Figure 1. — *Artificial ear (Assembly of acoustical elements)*

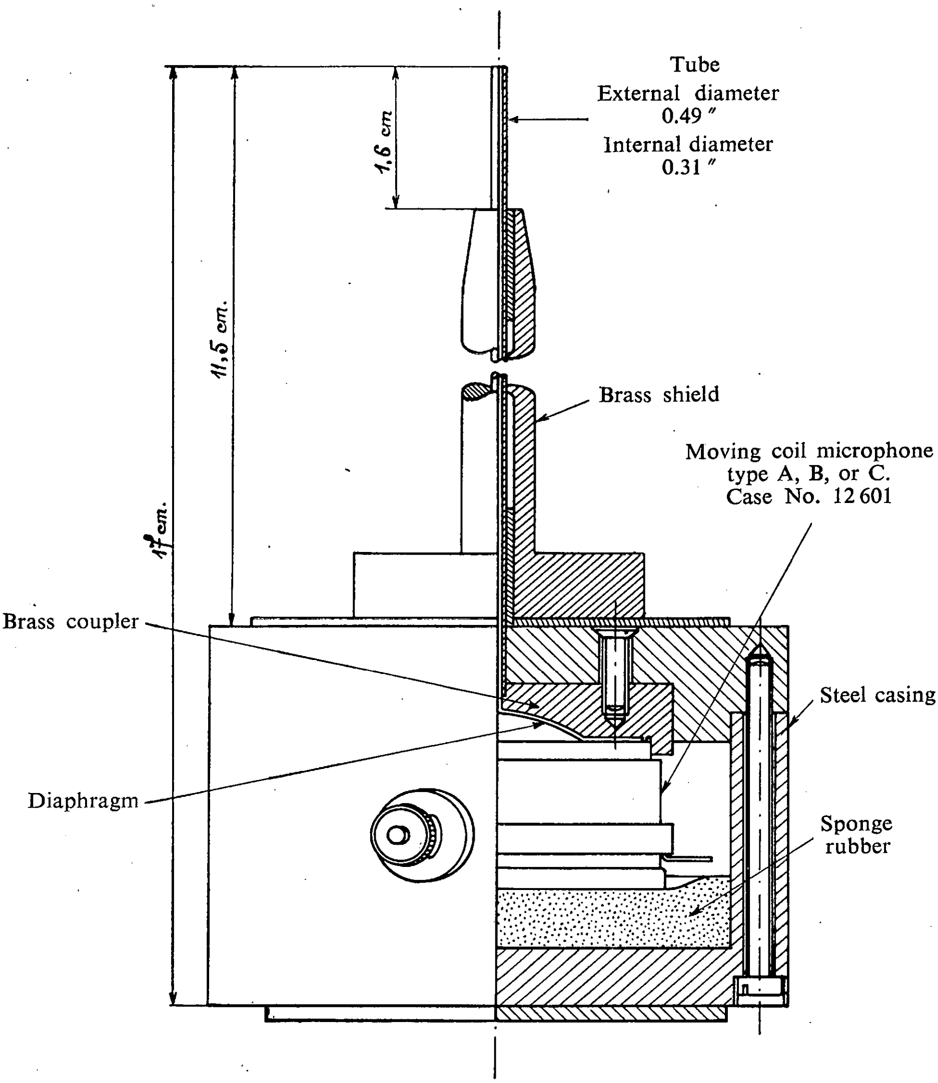


FIGURE 2. — *Probe tube microphone*

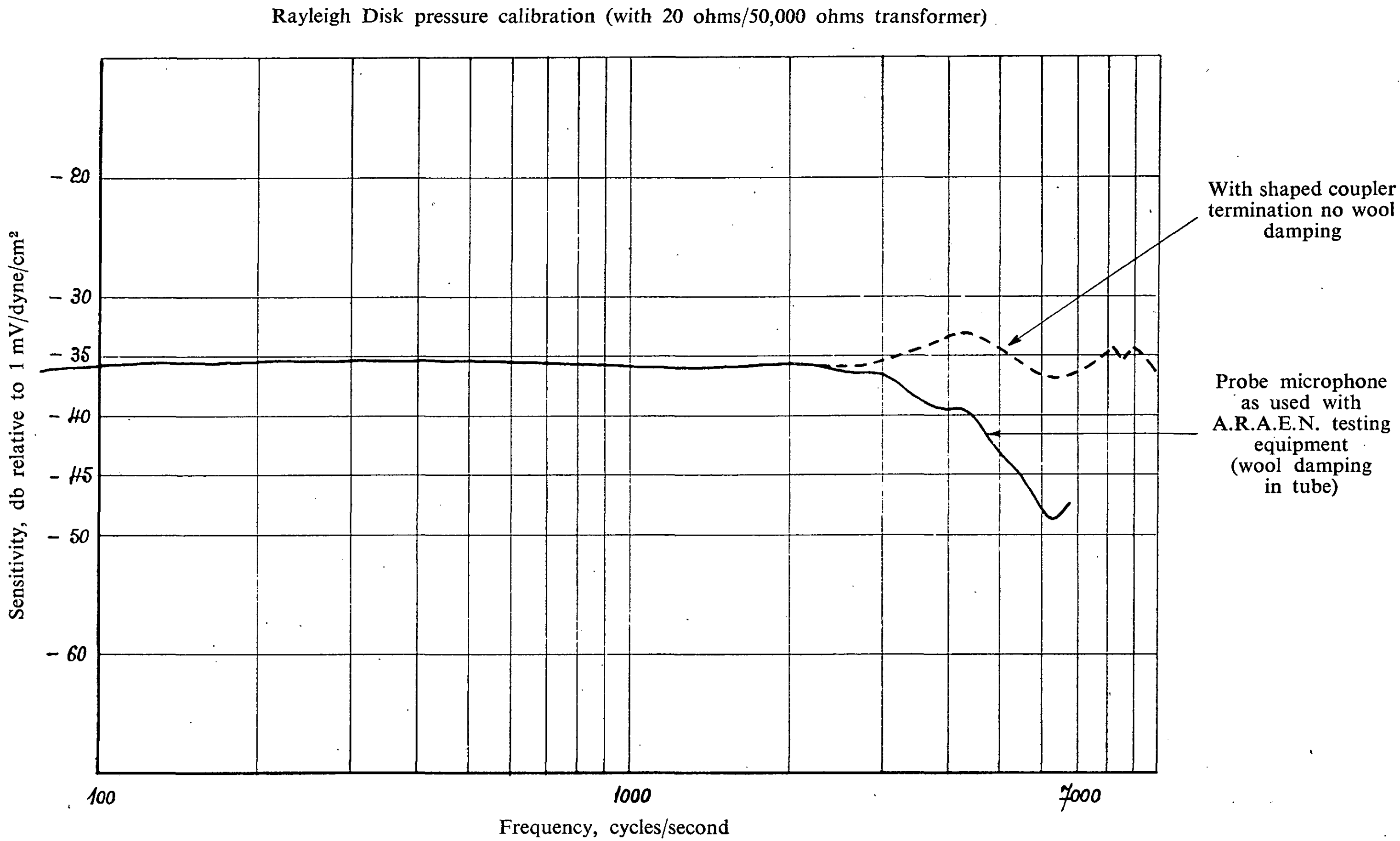


FIGURE 3. — *Sensitivity-frequency characteristic of a typical probe-tube microphone*

ANNEX 10

ARTIFICIAL MOUTH AND EAR USED BY THE SWISS TELEPHONE ADMINISTRATION

1. Artificial Mouth

The artificial mouth is to create an acoustical pressure on a microphone used close to the sound source. The distribution of pressure around the artificial mouth, particularly on its axis, must correspond as much as possible with that around the human head. The frequency characteristic (measured on the axis of the head) must be as flat as possible in the band of frequencies relevant to speech (100... 5 000 c/s). The distortion factor must be small even with high acoustic pressures.

The artificial mouth shown in Figure 1 which operates like a dynamic loudspeaker has already been described in a previous publication [1].

The generator system consists of a powerful permanent magnet and a 10 watt loudspeaker moving coil. The diaphragm of the artificial mouth comprising 2 cones is of aluminium 0.5 mm thick. At its outside edge the diaphragm is held in position by a leather suspension stretched tightly between two metal rings attached to the magnet. The moving coil is centred by two crossed and stretched springs of 0.2 mm dia. steel wire. The magnet, diaphragm, centering arrangement and metal rings form a unit. This unit is held by screws to a circular casting carrying the artificial mouth proper. The purpose of this is to obtain the desired distribution of acoustic pressure in front of the mouth. The distance between the diaphragm and the artificial mouth is only 1.5 mm in order to have the smallest possible volume and so reach a high limiting frequency. The output aperture determined empirically is a funnel opening steeply to avoid too great a concentration at high frequencies and presents no plane surface so as to reduce the formation of stationary waves between the microphone being measured and the mouth.

Figures 2 to 5 show the characteristics of the artificial mouth.

2. Artificial Ear

An artificial ear intended to measure receivers must reproduce as exactly as possible the acoustic impedance of the real ear within the relevant frequency

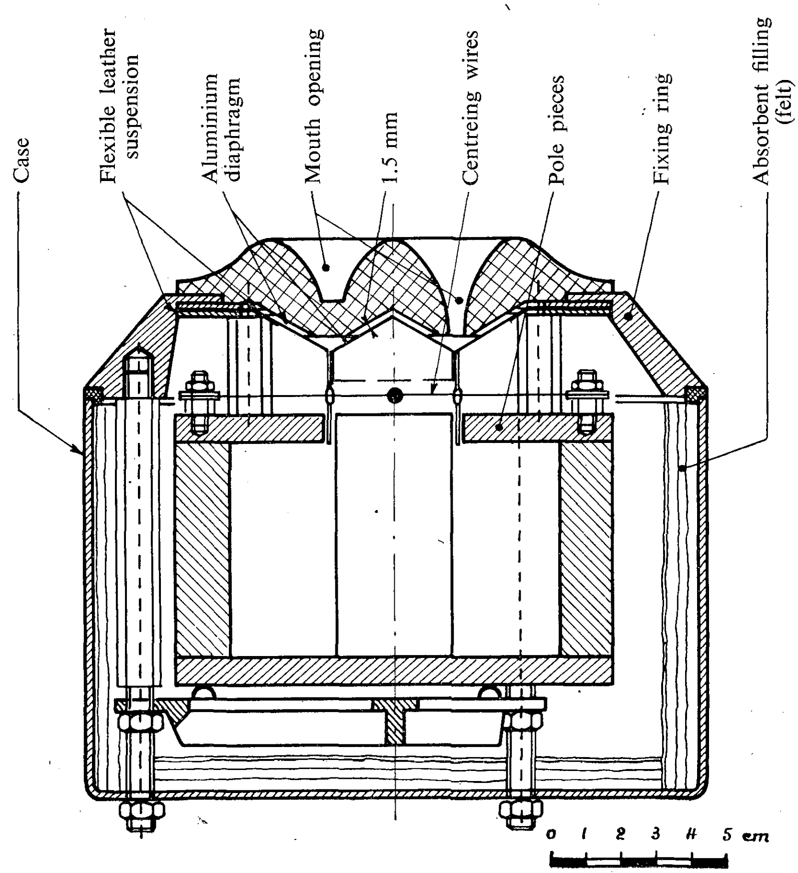


FIGURE 1. — *Artificial Mouth*

CHARACTERISTICS OF THE ARTIFICIAL MOUTH

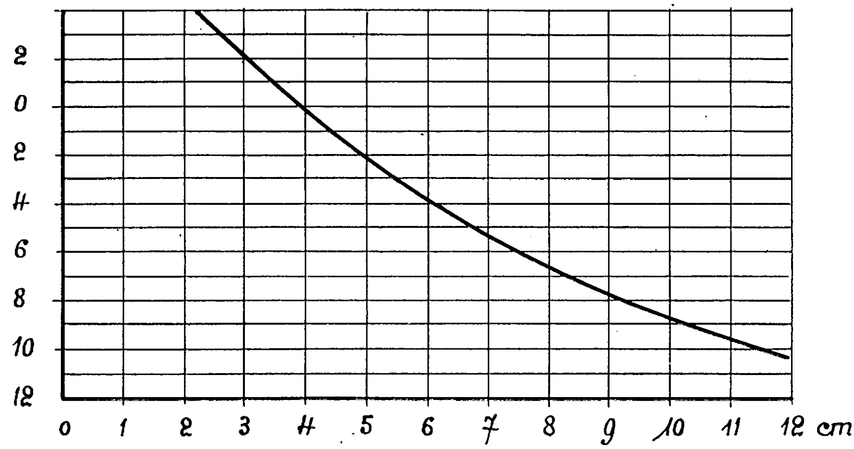


FIGURE 2. — *Mean decrease of acoustic pressure as a function of the distance along the axis of the artificial mouth*

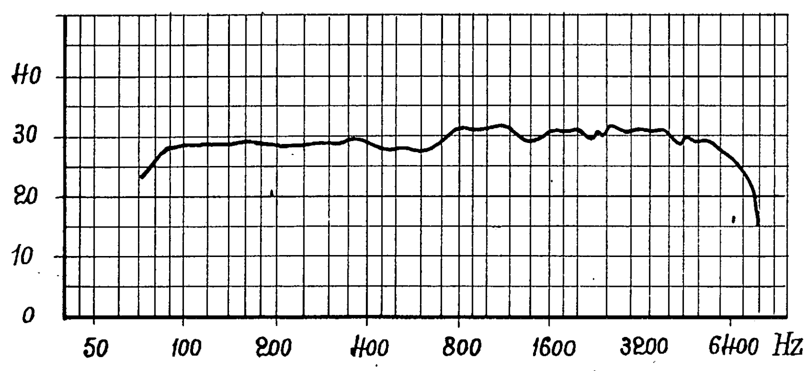


FIGURE 3. — *Response curve of the artificial mouth, measured through its correcting filter*

CHARACTERISTICS OF THE ARTIFICIAL MOUTH

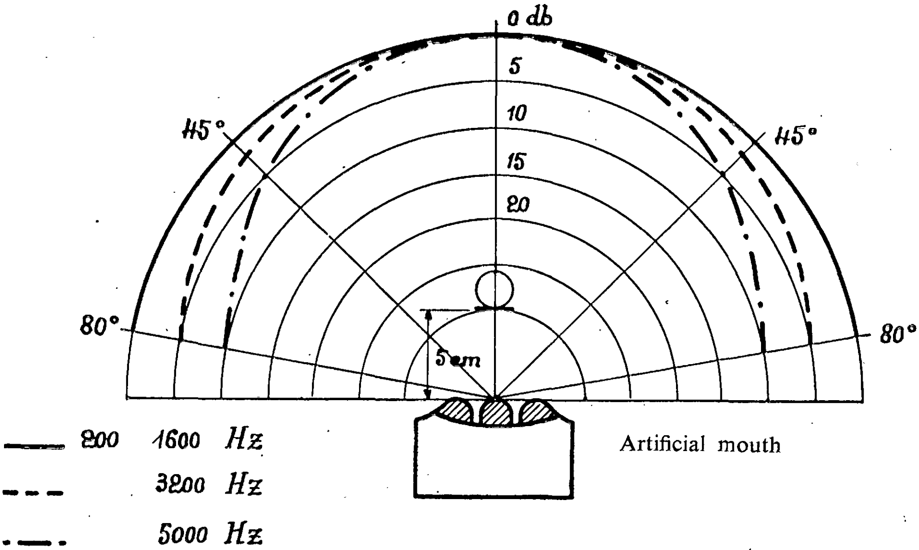


FIGURE 4. — *Polar diagram*

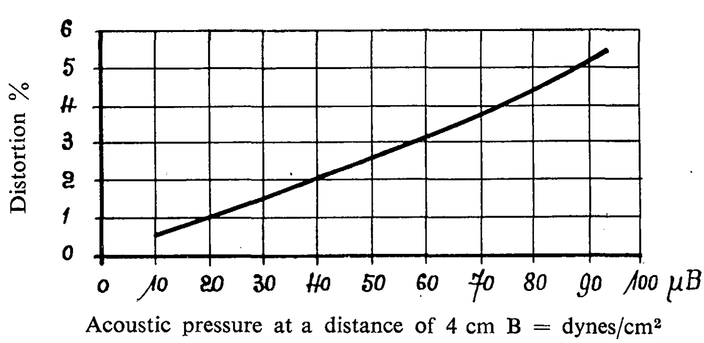


FIGURE 5. — *Distortion at 100 c/s*

band. The acoustic pressure measured at the centre of the artificial ear must correspond exactly to the pressure produced on the real ear at a defined point. The ear must be stable and permit reproducible measurements.

The introduction of a provisional reference artificial ear recommended by the C.C.I.F. is the first step toward the satisfactory standardization of artificial ears. Figure 6 shows our version of this ear combined with a preamplifier.

In the case of receivers with a wide frequency range and small mechanical impedance, a better reproduction of the acoustic impedance of the real ear is desirable. For this purpose the construction principle has been adopted of an artificial ear described by Weber [2], of which the acoustical characteristics have proved themselves. This artificial ear " P.T.T. Mod. 51 " (Figure 7) comprises two cavities (1.5 cm^3 between the condenser microphone and the opening of the ear and a coupled cavity of 1.4 cm^3) connected by a strongly damped channel. For frequencies above 1 000 c/s a higher acoustic impedance is thus obtained than that corresponding to the total volume of both cavities. Furthermore the coupling channel also provides the necessary acoustic resistance. The external shape of the artificial ear is dictated by the necessity of obtaining measured results comparable with those of the real ear.

Figure 8 gives a view of the complete measuring equipment. The handset is put into position by means of a centreing arrangement (visible on the left of the sketch), which is first placed on the centreing guides in place of the artificial ear.

The microphone is clamped by twisting a knob on the side of the box, the centreing device is raised, and the artificial ear, which weighs about 500 gr., is placed on the handset and provides just the prescribed pressure. There is sufficient play between the centreing guides and the artificial car mounting to ensure a good seal between the ear and the receiver earcap which can be made to fit by tilting slightly.

For measuring normal receivers a special capsule holder is available which is centred automatically when placed in position and does not need to be clamped.

3. Measurement of " sensitivity-frequency " characteristics of carbon microphones

It is difficult to make a precise and reproducible measurement of the " sensitivity-frequency " characteristic of a carbon microphone because the disposition of the carbon granules is not absolutely defined and can vary appreciably according to the nature of the acoustical excitation. It is for this reason that the present known measuring methods often give very different results.

To obtain the " sensitivity-frequency " characteristics of carbon microphones a heterodyne oscillator is used which the frequency can be made to vary by means of an automatic mechanism whilst the output voltage remains constant over the whole frequency range. This method does not impose very strict requirements on the " sensitivity-frequency " characteristic of the artificial mouth.

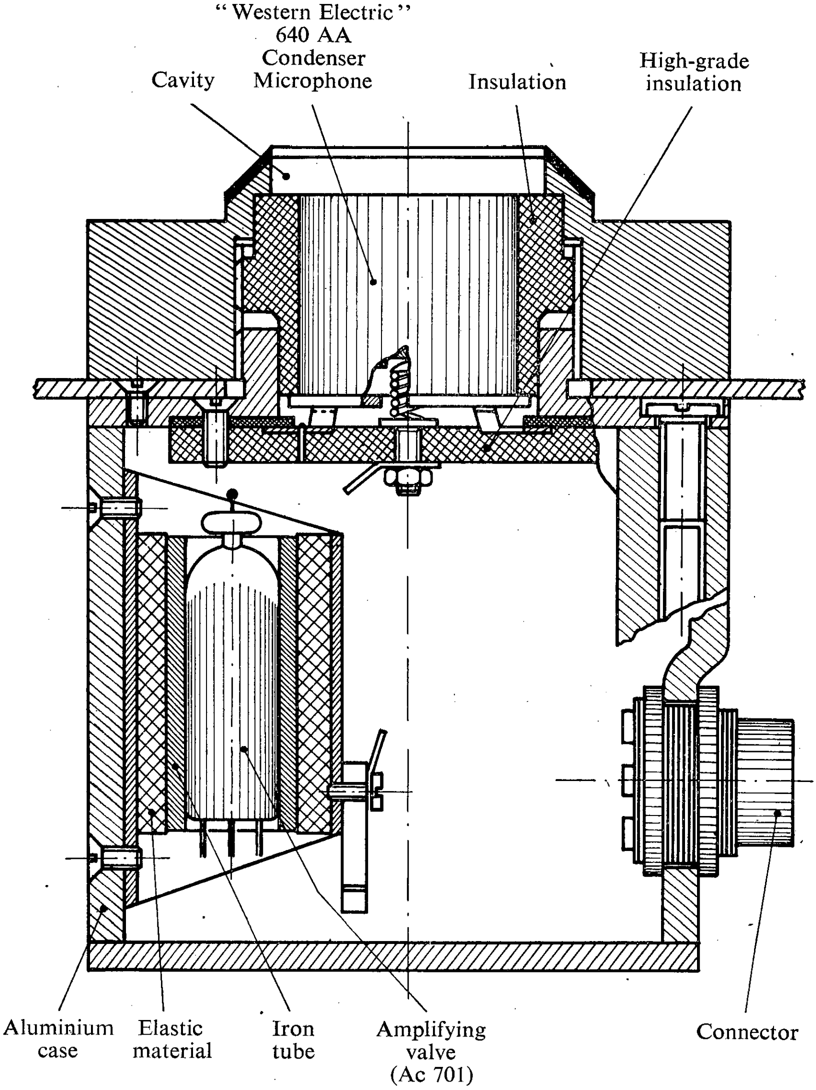
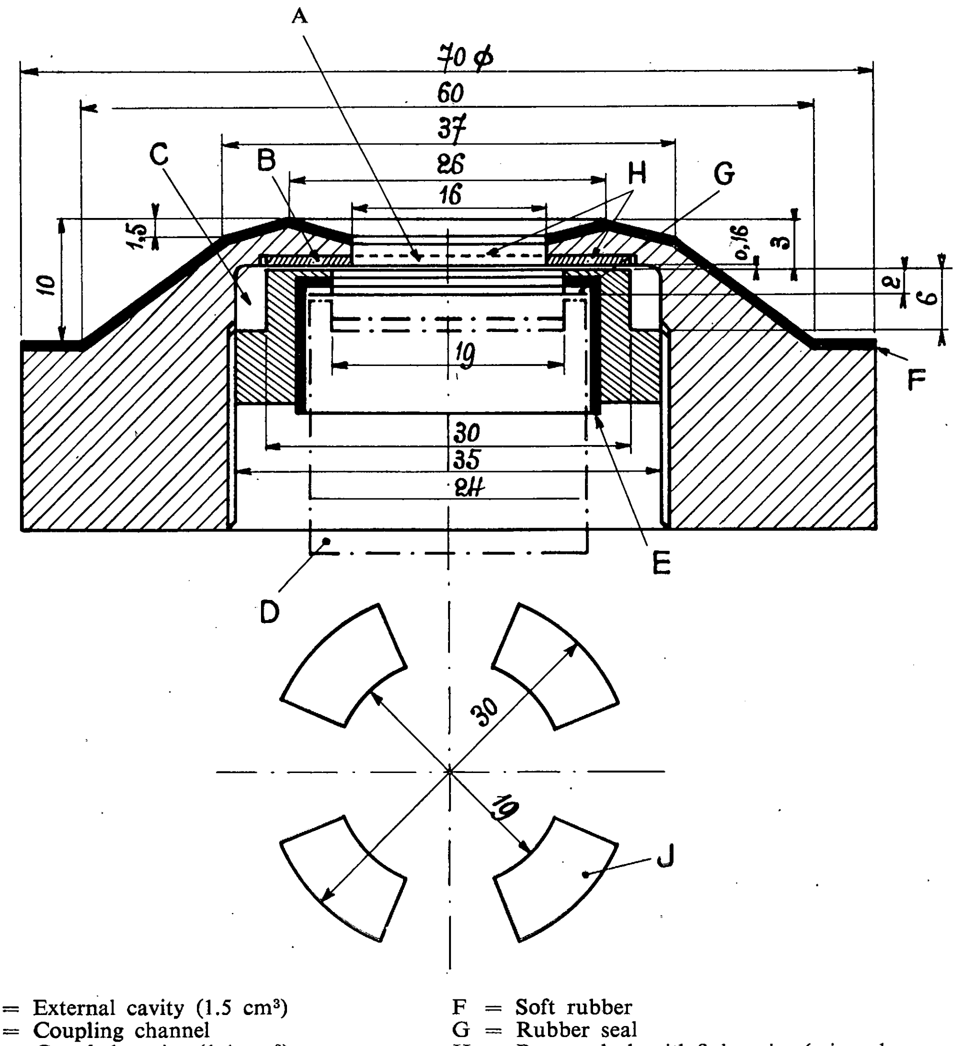


FIGURE 6. — *Artificial ear* (section)
(with the coupler of the provisional reference artificial ear recommended by the C.C.I.F.)



A = External cavity (1.5 cm³)
B = Coupling channel
C = Coupled cavity (1.4 cm³)
D = " Western Electric " 640 AA Microphone
E = Insulation
F = Soft rubber
G = Rubber seal
H = Porous cloth with fixing ring (microphone protection)
J = Partial filling of coupling channel

FIGURE 7. — *Section of the P.T.T. Model 51 artificial ear*

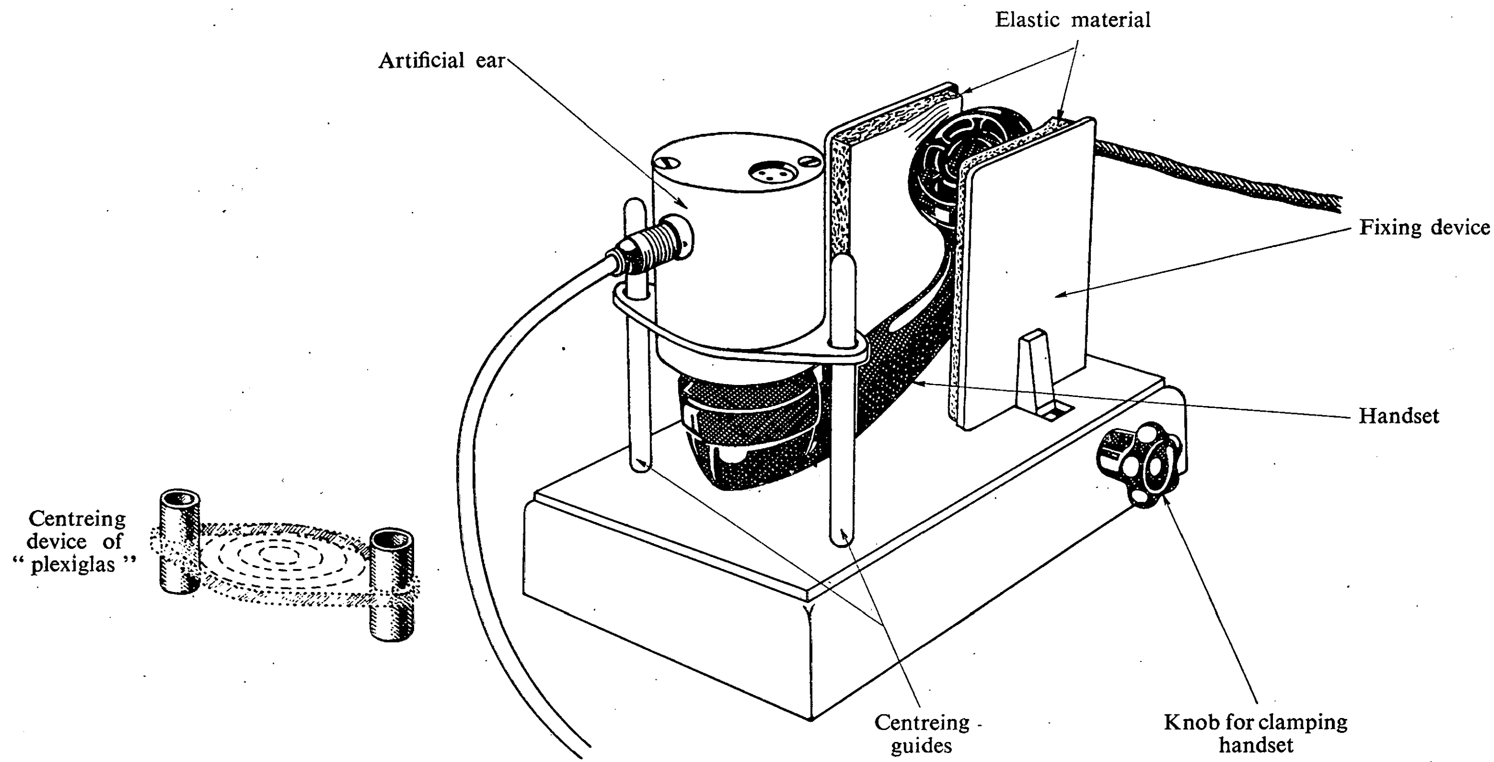


FIGURE 8. — *Arrangement for measurement of telephone receivers*

At any instant there is only a single tone and so the pressure can be set up by means of a standard microphone and maintained constant with an automatically regulated amplifier.

A distinction is made between a slow sweep and a rhythmical frequency variation according to the time necessary to traverse a given frequency band.

To measure the frequency characteristic with a slow sweep a frequency generator is used built according to the C.C.I.F. directions. The frequency band from 30 to 10 000 c/s is swept automatically in 212 seconds. When the sweep is slow a mechanical recording instrument can be used. The disadvantage of the slow frequency sweep is that the carbon granules of the microphone have time to settle. This results in the microphone packing and completely losing its sensitivity.

This packing of microphones does not happen when normally excited by speech. To test carbon microphones under conditions which reproduce as closely as possible those encountered in service (speech is composed of a mixture of frequencies), the transmission of a pure tone is interrupted every 2 seconds by the transmission of a background noise (for about 1 second) which prevents the microphone packing. Tests may be made with a background noise of which the power spectrum is continuous or according to the distribution of speech. In either case quite reproducible measurements are obtained. Although the background noise is transmitted only very briefly each time, measurement is assisted and approximation is obtained to conditions encountered in the transmission of speech.

Figure 9 shows the schematic diagram of a measuring apparatus which has given every satisfaction. With the aid of an artificial ear measurements can also be made of the sidetone attenuation of a subscriber's set.

The switching from pure tone to background noise is performed automatically by the arrangement for varying the oscillator.

As it is essential for the acoustic pressure to be as constant as possible, an amplifier with automatic regulation is used which equalizes the frequency characteristic of the artificial mouth. The schematic diagram of the amplifier with automatic regulation is shown in Figure 10.

The artificial mouth is described in Section 1.

As shown in Figure 9, the microphone is measured in a normal circuit with the usual subscriber's set and telephone exchange feeding bridges. The termination impedance is 600 ohms.

Figure 11 shows, as an example, the results obtained with a bad quality microphone. Without transmission of background noise (Curve *a*) this microphone was not stable enough and the measurements were not reproducible. With the periodical transmission of a background noise (Curve *b*) quite reproducible results were obtained.

By this method the frequency characteristic is only recorded fragmentarily but the curve can be completed by hand with sufficient accuracy. This disadvantage can be avoided by making a second measurement with the pure tone and background noise interchanged.

The level of background noise is also recorded by the graphic level recorder and should for an ideal microphone produce always the same deflexion but any

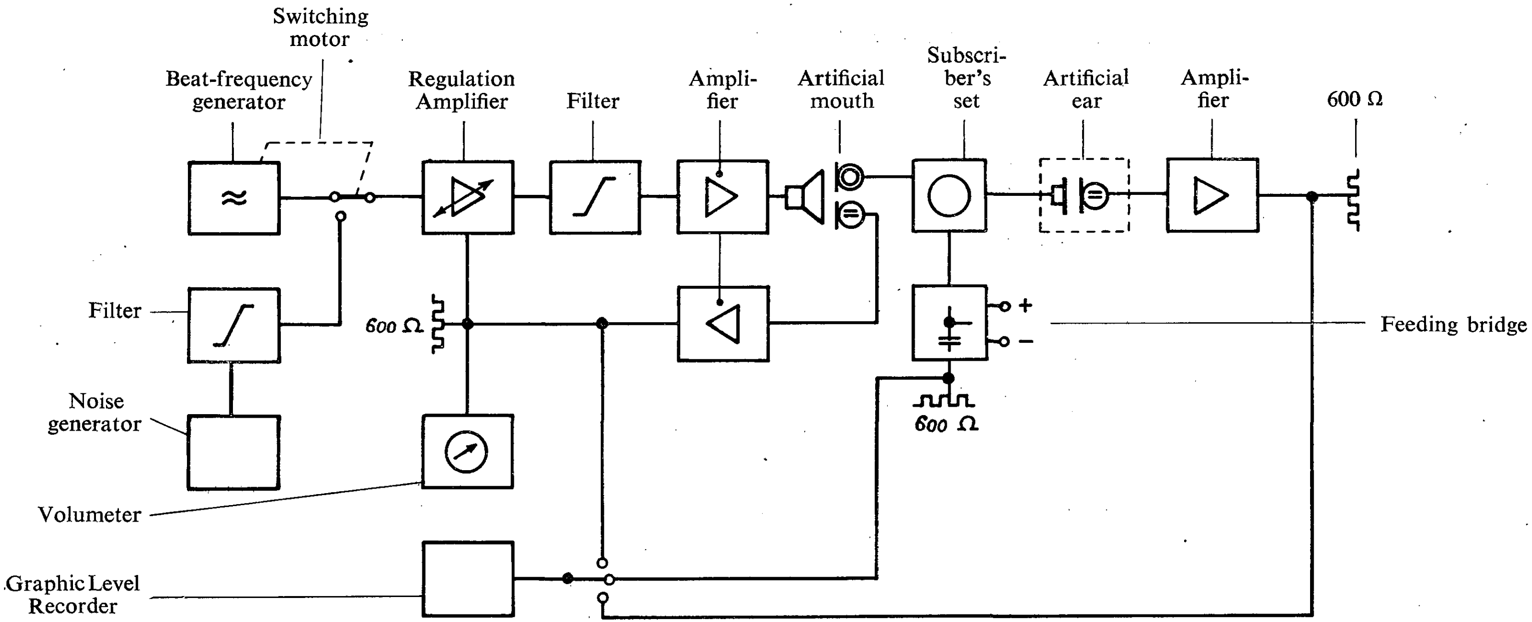


FIGURE 9. — *Circuit for sending and sidetone measurement, by means of a generator in accordance with the C.C.I.F. specifications and a noise generator*

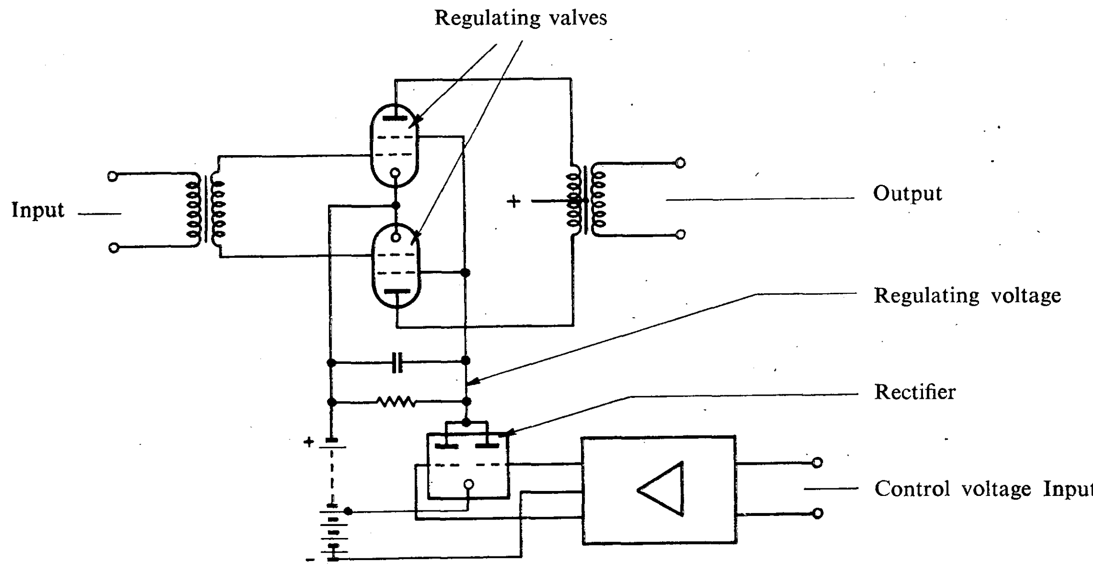


FIGURE 10. — *Schematic diagram of the principle of the regulating amplifier*

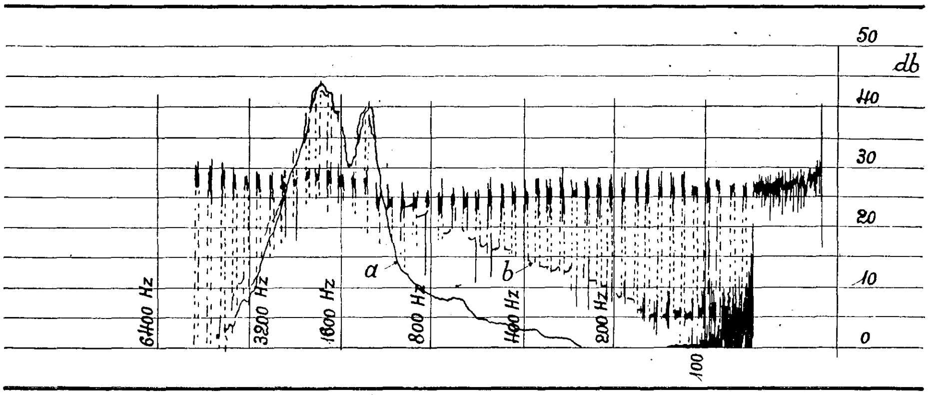


FIGURE 11. — *Response curve of a bad quality microphone*
Curve *a* obtained with pure tone
Curve *b* obtained with pure tone and background noise

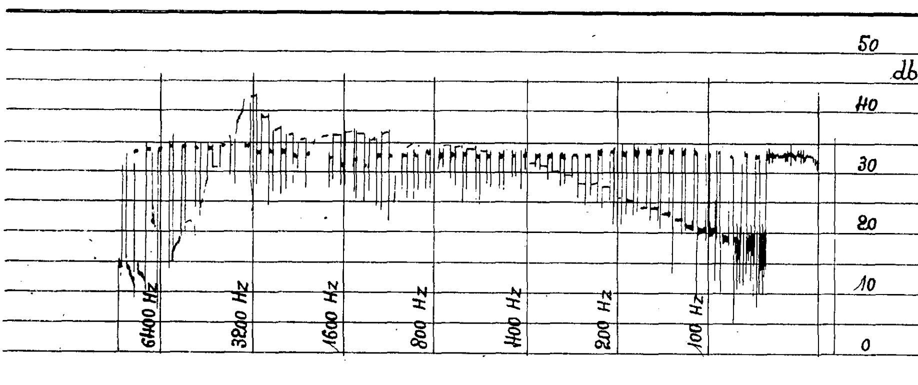


FIGURE 12. — *Response curve of a good quality microphone measured with pure tone and background noise*

irregularities present in the level recorded show up any instability of the microphone.

Figure 12 shows results obtained with a good quality microphone.

The method described above is used especially when there is need of recording the frequency characteristic for documentary reproduction.

For measurements which require no special reproduction (checking, preliminary tests, etc.) the measuring apparatus shown in Figure 13 is available. The heterodyne oscillator sweeps the frequency band in both directions periodically about once per second.

The amplifier with automatic regulation has been adjusted as a result of tests so that the regulation operates in about 0.05 seconds and the degree of regulation is 3.3%. The lowest measuring frequency which is undistorted and remains stable is approximately 200 c/s.

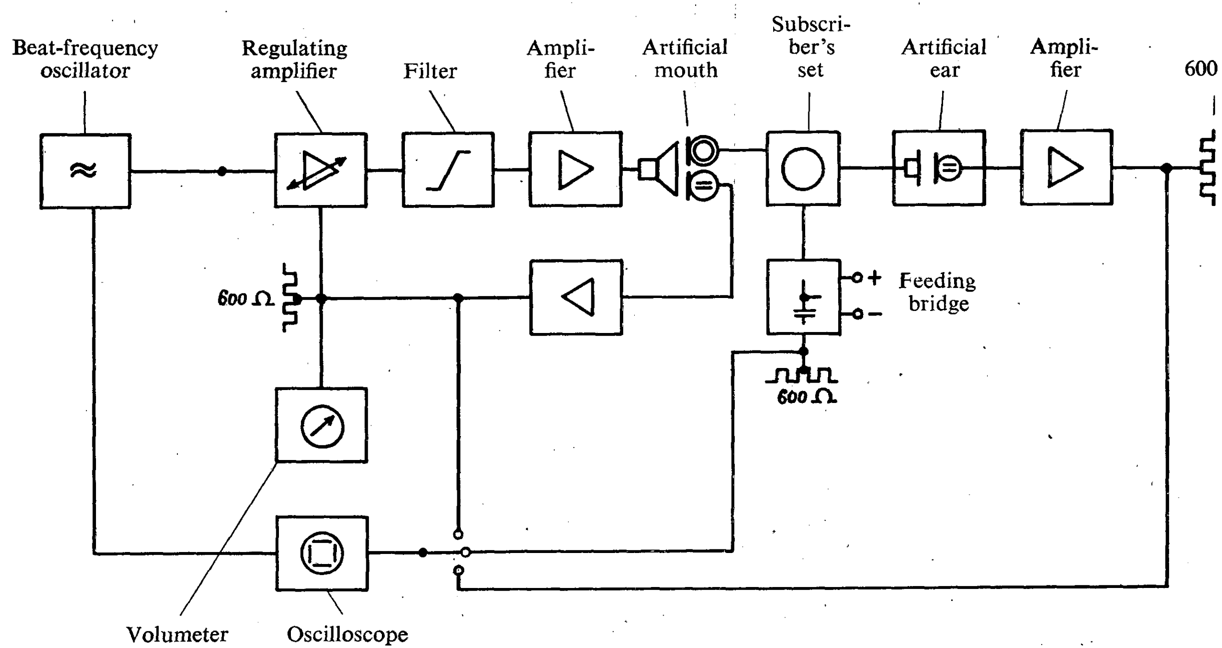


FIGURE 13. — *Circuit for sending and sidetone measurement by means of a rhythmic oscillator*

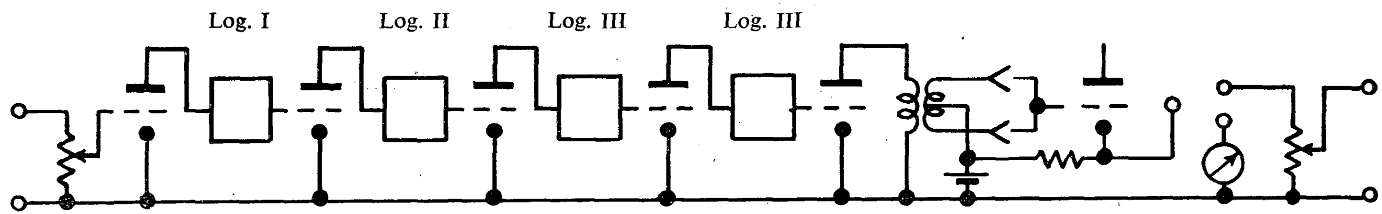


FIGURE 14. — *Schematic diagram of logarithmic amplifier*

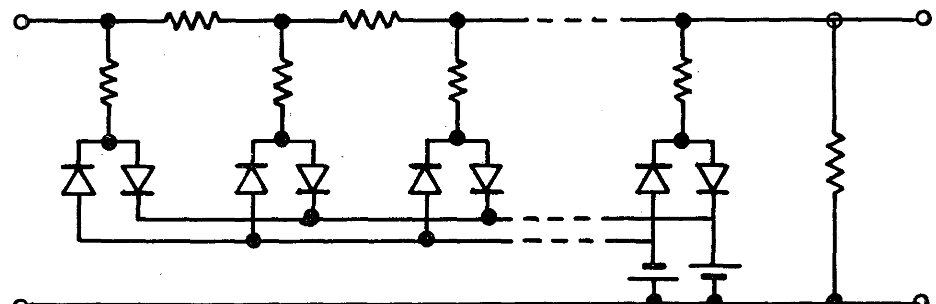


FIGURE 15. — *Coupling circuit (combination of linear elements)*

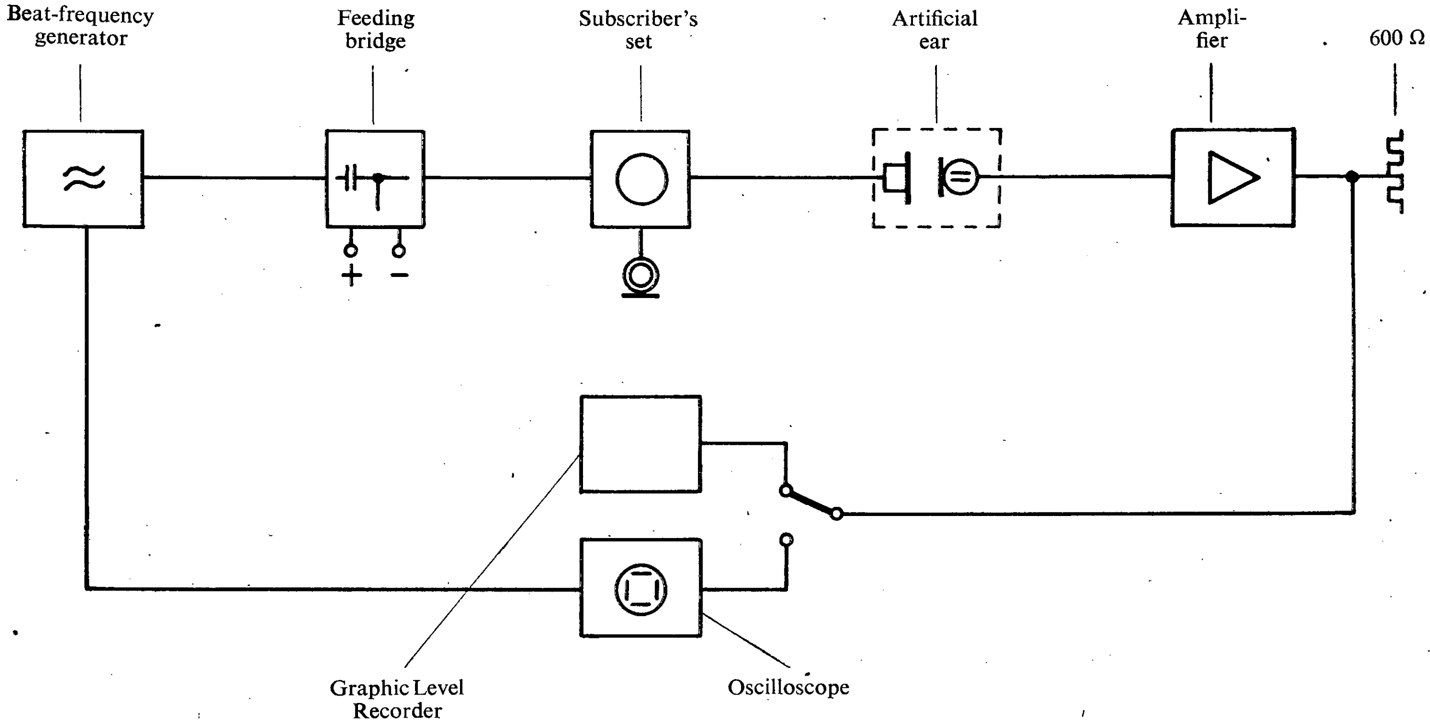


FIGURE 16. — *Circuit for receiving measurement*

The writing speed of a graphic level recorder is not high enough and so the frequency characteristic is displayed on the screen of a cathode ray oscilloscope, This equipment has a logarithmic frequency scale on the abscissa axis and a linear or nearly logarithmic scale of levels on the ordinate axis.

If an absolutely logarithmic scale of levels is required a logarithmic amplifier is inserted before the cathode ray oscilloscope equipment.

Unlike usual logarithmic amplifiers which require time to operate, the amplifier used gives an instantaneous logarithmic gain. At any instant the output voltage corresponds to the logarithm of the input voltage. Large and rapid variations can thus be studied accurately such as occur during microphone measurements by rhythmical frequency variation. In the amplifier in question the logarithmic function is obtained by approximation using linear elements. Between the various stages of amplification are included elements consisting of linear networks comprising resistors and crystal diodes with appropriate bias voltages. Figure 14 shows the diagram of the complete amplifier and Figure 15 the arrangement of the coupling circuits.

The electrical characteristics of the logarithmic amplifier are as follows.

Extent of amplification without input potentiometer	50 db
Precision of logarithmic indication	±0.3 db
Calibrated input potentiometer, 10 steps of . . .	5 db
Input resistance	100 kohms
Voltage with input potentiometer	0.7 mV to 70 V
Output voltage rectified for cathode ray oscillograph	0 to 5.5 V
Frequency range	30 to 50,000 c/s

4. Measurement of frequency characteristics of telephone receivers

Measurement of the frequency characteristic of a telephone receiver is performed according to the diagram shown in Figure 16. The receiver is inserted for measurement in the normal circuit of a telephone set.

As a source of voltage a heterodyne oscillator is used driven by a motor. The frequency variation is either in accordance with the C.C.I.F. recommendations or is appreciably faster (about 1 sweep per second) depending on whether a graphic level recorder or oscilloscope is used as the measuring device. The artificial ear is described in Section 2.

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ANNEX 11

S.F.E.R.T. VOLUME INDICATOR

A Volume Indicator *, to measure the volume on a telephone circuit (wire or radio) during a continuous conversation, should satisfy the following conditions :

Integration time. — To measure the volume at a point on a circuit during continuous conversation, an instrument is used of which the integration time (time from closing the circuit until the correct reading is given) is 200 milliseconds, which corresponds to the mean duration of a logatom.

The integration time is defined as the minimum time for which a sinusoidal alternating voltage must be applied to the terminals of the instrument for the needle of the indicating device to reach, within 0.2 nepers or about 2 decibels, the deflexion which would have been obtained had the same voltage been applied indefinitely.

Time to return to zero. — The time for the needle of the indicating device to return to zero should be adjusted to the same value as the integration time, i.e. 200 milliseconds.

Graduation. — The instrument should be graduated in transmission units (nepers or decibels) and should indicate directly the relative value of the volume in transmission units measured relative to " reference volume " (see below), which bears a definite relationship to the " normal volume for telephonometric measurements " which is defined in the *Green Book*, Volume IV, Section 3.1.2.B.

Use and calibration. — So far as the method of using such equipment for measuring volume on a telephone circuit concerned, it is necessary :

— to calibrate the equipment ;
— to measure the volume.

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* The volume indicator used in the C.C.I.F. Laboratory for telephonometric measurements, in conjunction with the European Master Reference Telephone System (S.F.E.R.T.) is by convention termed " The Volume Indicator " in C.C.I.F. documents ; it forms an integral part of the S.F.E.R.T.

A calibration line is marked on the scale of the indicating instrument. The equipment is connected to a source of 800 or 1,000 c/s alternating current, adjusted to develop a power of 6.0 mW in 600 ohms resistance, and the " transmission units dial " is set to " 0 " corresponding to " reference volume " *. Calibration then consists of adjusting the " calibration dial " so that the needle of the indicating instrument is at the calibration mark.

To make a volume measurement at a point on a circuit when the instrument has been so adjusted, it is connected in parallel across the circuit and the " transmission units dial " is adjusted so that the needle reaches the calibration mark about once every three seconds. The number of units read on the transmission units dial gives the volume at the point of the circuit considered, relative to " reference volume ".

To facilitate measurements, it is possible to have, in addition to the graduations of the transmission units dial, a certain number of divisions on either side of the mark on the indicating instrument graduated in transmission units.

Overshoot. — So far as overshoot of the indicating instrument is concerned, the needle must not exceed the final deflexion by more than 1 decibel when a sinusoidal alternating voltage is applied to the terminals of the equipment.

Frequency Characteristics. — The frequency characteristic of the equipment must be level within ± 0.1 nepers or about ± 1 decibel in the frequency range from 200 to 3,500 c/s for equipment intended for use on commercial telephone circuits and, in addition, within ± 0.2 neper or about ± 2 decibels in the supplementary ranges from 30 to 200 c/s and from 3,500 to 7,000 c/s for equipment intended for use on radio broadcasting circuits.

Law of addition. — The equipment must be designed so that different frequency components are added together as far as possible according to a quadratic law.

Impedance. — The input impedance of the equipment must not be lower than 10,000 ohms. The equipment must be balanced, i.e. the deflexion must be the same whatever the polarity.

* " Reference volume " of the " Volume Indicator " is then the volume of speech sounds which corresponds to the zero division of the volume indicator, this zero division having been reached when calibrating the volume indicator with a voltage applied at 800 or 1,000 c/s corresponding to a power of 6.0 mW in 600 ohms pure resistance.

ANNEX 12

A.R.A.E.N. VOLUME METER OR SPEECH VOLTMETER

This equipment complies with C.C.I.F. specification (see specifications given in Annex 11) except so far as integration time, time to return to zero, and reference point for calibration are concerned. The integration time and time return to zero specified by the C.C.I.F. are 200 milliseconds, whilst for the A.R.A.E.N. volumemeter these are approximately 100 milliseconds. Experience gained with this volumemeter, having a shorter integration time, has shown that it is quite satisfactory for controlling the speech level in articulation tests. If it is necessary to have the longer integration time specified by the C.C.I.F. it is possible to change the milliammeters used with this equipment.

The specification of the A.R.A.E.N. volume meter is as follows :

Integration time: 100 milliseconds.
Time to return to zero: 100 milliseconds.

Graduation: in decibels. — The scale of the indicating instrument is marked at 1 decibel intervals from −8 to +3 decibels relative to the reference point defined below.

Calibration: — The equipment is calibrated to indicate the number of decibels relative to 1 V at 1,000 c/s. The sensitivity controls enable 0 db on the indicating instrument to be obtained for voltage levels in the range from −50 to + 30 decibels relative to 1 V at 1,000 c/s.

Use. (a) *To measure speech levels during continuous conversation.* — The volume meter is connected to the point where it is desired to measure the volume and the sensitivity controls are adjusted so that the needle of the indicating instrument reaches the reference point (0 decibel mark on this instrument) approximately once every three seconds.

(b) *To determine the vocal level of the talker in an articulation test.* — The volume meter is connected to the A.R.A.E.N. at a point where the speech volume would correspond to 1 V. The sensitivity controls are set at 0 so that the equipment is adjusted to indicate its reference volume. The talker then pronounces the logatoms (in an appropriate phrase) so that each preliminary syllable of the carrier phrase causes the needle of the indicating instrument to deflect to the reference position (0 decibels on the scale of this instrument).

Overshoot. — On application of a sinusoidal voltage, the needle must not exceed it final value by more than 0.1 db.

Frequency characteristic. — The sensitivity of the equipment for sinusoidal waves at any frequency from 40 to 12,000 c/s is the same as at 1,000 c/s with the following tolerances :

40 to 12,000 c/s	$\pm$ 2 db
150 to 6,000 c/s	$\pm$ 0.5 db

Law of addition. — The law of addition of components at different frequencies is quadratic over a range of $\pm$ 8 db about the reference point (0 db) of the indicating instrument.

Impedance. — The input impedance of the equipment is greater than 10,000 ohms over the whole useful frequency band.

ANNEX 13

VOLUME METER STANDARDIZED IN THE UNITED STATES, TERMED VU-METER

(From Standard C.16.5.1942 "American recommended practice for volume measurements of electrical speech and program waves", adopted 6th November 1942 by the American Standards Association)

Introduction

Radio broadcasting, of which the fundamental requirement is the transmission of speech or music either by metallic circuits or by radio, has raised the question of measuring the electrical signals used in such transmission. Measurements made by different radio broadcasting companies with instruments of different designs have shown considerable disagreement.

The instrument described in the following standard has been designed particularly for radio broadcasting and for the telephone circuits which provide the interconnexion between the broadcasting transmitting stations. It gives sufficiently good correlation between measurements made under normal conditions of use and its dynamic characteristics allow the needle of the instrument to be read quicker and more precisely than do previous types of volume meter. The following standard relates to the volume indicator and its method of use.

General

This standard applies to the methods of and a device for measuring the dynamic magnitude of complex audio-frequency electrical waves such as occur in speech and music.

The measurement of the complex and nonperiodic waves encountered in electrical communication cannot be expressed in simple fashion in the ordinary electrical terms of current, voltage, and power. The concept of "volume" furnishes a practical method of great utility to the communications engineer for assigning a numerical value to the magnitude of electrical speech and program waves.

Volumes are read by noting the more extreme meter deflections of a device known as a volume indicator. Since the response of the meter of such an instrument to the rapidly varying waves is greatly dependent upon its dynamic characteristics, a standard for volume measurements must, therefore, include a specification of these characteristics.

It has hitherto been the custom to express measured values of volume as a number of decibels above or below one of several different arbitrary reference levels. The present standard uses a new term " volume units " (vu in short), to express the volume in terms of the number of decibels above or below a particular reference level specified in this standard.

Definitions

The following definitions relate to the use of the terms below so far as they apply specifically to the quantities and the instruments used to measure volume.

Volume. — This term is used to denote the magnitude of the electrical waves corresponding to the transmission of speech or music. It is the reading of an instrument termed the " American volume meter ", defined below, which has specified dynamic and other characteristics, and which is calibrated and read in a prescribed manner.

Volume meter. — A device for the indication of volume. A volume meter conforming to the American standard must have the characteristics indicated in the paragraph below entitled " Specification of the American volume meter ".

" *Volume unit* " (v.u., customarily written in small letters). — This expression is used to denote the numerical value of the volume. The volume in " volume units " is numerically equal to the relative magnitude of the waves considered, expressed in decibels above or below the " American reference volume * ", defined below.

The term vu should not be used to express results of measurements of complex waves made with devices having characteristics differing from those of the standard American volume meter.

American reference volume. — This is the basis of the system of measuring volume. The " American reference volume " is the magnitude of the electrical waves, corresponding to the transmission of speech or music, which gives a reading of zero volume units on a volume meter whose characteristics and method of reading are described in the present standard, and which is calibrated so as to give the reading zero volume units for a continuously applied sinusoidal wave, of frequency 1,000 c/s and dissipating 1 milliwatt in 600 ohms *.

Reference deflexion. — The deflexion of the needle of the measuring instrument corresponding to the point on the graduated instrument scale at which, or near which, it is normally intended to make readings.

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* It will be noted that the " American reference volume " differs from " reference volume " defined by the C.C.I.F. (see Annex 11 " S.F.E.R.T. volume indicator ").

Specification of the American volume meter

Volume is measured by means of a volume meter. This device must conform to the following specifications and must be used in the manner described below.

Component parts. — A volume meter consists of at least two parts :

(a) A meter.
(b) An attenuator.

Dynamic Characteristics. — If a sinusoidal voltage between 35 and 10,000 cycles per second, of such amplitude as to give reference deflection under steady-state conditions is suddenly applied, the meter pointer shall reach 99 per cent of reference deflection in 0.3 seconds, $\pm$ 10 per cent, and shall then overswing reference deflection by at least 1.0 per cent and not more than 1.5 per cent. The time required for the meter pointer to reach its position of rest on the removal of the sinusoidal voltage shall not be greatly different from the time of response.

Response-Versus-Frequency Characteristic. — The response of the volume meter shall not depart from that at 1,000 c/s by more than 0.2 decibel, between 35 and 10,000 nor more than 0.5 decibels, between 25 and 16,000.

Response to Complex Waves. — The response to complex waves of such amplitude as to give reference deflection when read, as described below, shall be that equivalent to the response with a direct-current meter and a rectifier, the exponent of whose characteristic is 1.2 $\pm$ 0.2.

Reversibility. — The response when measuring unsymmetrical waves must be independent of the poling of the volume meter. Such a characteristic may be obtained by the use of a direct-current meter in conjunction with a full-wave rectifier.

Graduation of Meter Scale. — The point of reference deflection shall be definitely indicated in some suitable manner. The remainder of the scale shall be graduated in vu above and below reference deflection. (See also paragraph " Scale marking " below).

Attenuator. — The attenuator is normally adjustable and its control should be graduated in volume units.

Calibration. — The measuring instrument of a correctly calibrated volume meter with its attenuator set at 0 vu will give reference deflection when connected to a source of a sinusoidal voltage adjusted to develop 1 milliwatt in a resistance of 600 ohms, or with the attenuator set at n vu when the calibrating voltage is adjusted to develop a power n decibels above 1 milliwatt.

Method of Reading Volume Meter. — The reading of the measuring instrument is determined by the greatest deflections occurring in a period of about a minute for program waves, or a shorter period (e.g., 5 to 10 seconds) for message telephone speech waves, excluding not more than one or two occasional deflections of unusual amplitude.

The volume meter is usually connected across the circuit at a point where the impedance is 600 ohms and the attenuator is adjusted until the deflections, read as described above, just reach the scale point corresponding to reference deflection. The volume in vu is determined by the markings on the attenuator at the setting thus obtained. If for any reason the deflections reach some other scale point than that corresponding to reference deflection, the volume is given by the algebraic addition of the attenuator setting and the actual deflection as read on the meter scale.

When the impedance of the circuit at the point at which the instrument is connected differs from 600 ohms, the volume indicated must be corrected to correspond to this difference in impedance by the following relationship :

Correction (to be added algebraically) in vu =

$$10 \log_{10} \frac{600}{|Z|},$$

where $|Z|$ = magnitude of actual impedance.

Good Engineering Practice

The following items are not fundamental to this standard but are matters of good engineering practice.

Impedance. — The volume meter is normally used as a bridging instrument and when so used its impedance must be sufficiently high so as not to influence unduly the waves in the circuit with which it is used. It is good practice to make the value of impedance not less than 7,500 ohms for use on a 600 ohms circuit.

Harmonic Distortion. — The root-mean-square value of the harmonic distortion produced when the volume meter is bridged across a resistive circuit impedance through which a sinusoidal wave between 25 and 8,000 cycles per second is being transmitted should not exceed 0.2 per cent of the fundamental.

Ability to withstand overload. — Because of the great variation in amplitude which this volume meter may encounter, it must be capable of withstanding greater overloads than is required of current-measuring instruments. A specification often used requires that the volume meter should be capable of withstanding without injury or effect on calibration, a momentary overload of ten times the voltage corresponding to reference deflection, and a continuous overload of five times that voltage.

Scale marking. — The point of reference deflexion should be located within a sector between 2/3 and 3/4 of full scale. In addition to the vu scale, a 0-to-100 scale proportioned to voltage should be provided, the 100 point coinciding with reference deflexion. Samples of the two types of scales in general use are shown in Figures 1 and 2.

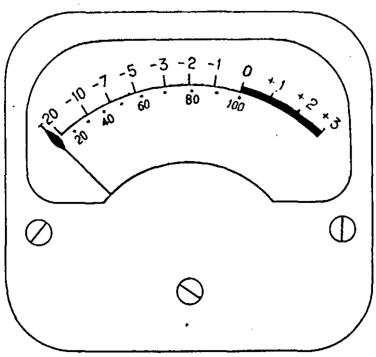

FIGURE 1

This scale emphasises the graduation in vu.

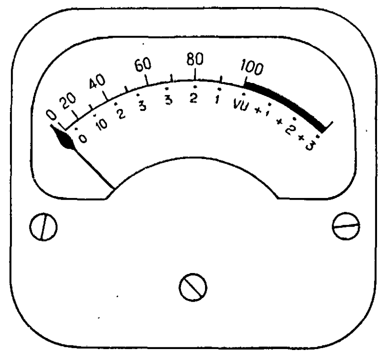

FIGURE 2

This scale emphasises the graduation 0-to-100 and is used to indicate the percentage utilization of circuits and equipment. It is used in general by the principal broadcasting companies.

ANNEX 14

MODULATION METER USED BY THE BRITISH BROADCASTING CORPORATION

Description

The modulation meter used by the British Broadcasting Corporation (Peak Programme Meter) is an instrument designed to register the peak amplitudes reached by complex waveforms which occur in broadcast speech and music. It uses a $3^1/_4''$ diameter pointer-type meter with a simple scale engraved in white on a black background. The scale is logarithmic, covering 26 db in 6 steps, and the registration time is relatively quick, while the decay is quite slow.

The type of indicator was chosen to assist the eye to make rapid and accurate readings at a glance, and the simple white-on-black scale tends to reduce eye strain. Although the instrument is essentially a peak voltmeter the registration time is deliberately curtailed to a value below that readily obtainable for the following reason. The ability of the ear to appreciate distortion due to overload in a broadcasting system depends upon the duration of the peak amplitudes involved ; therefore the restriction of very short duration peaks to an amplitude below the overload point is not only unnecessary but result in a lower general level of modulation than the economic use of the system dictates.

The decay time of the instrument is made slow enough to enable the eye to observe without strain the peak reading reached, yet it is fast enough to enable subsequent peaks of a lower value to be registered accurately.

Circuit

The meter instrument is driven by an amplifier consisting of a buffer stage with a high input impedance, a full-wave diode peak rectifier, and a variable-mu pentode valve which provides the logarithmic scale.

The charging and discharging time constants of the peak rectifier are 2.5 milliseconds and 1 second respectively to give the performance specification below, and the correct scale shape is adjusted by variation of the electrode potentials of the variable-mu valve.

The meter itself is a fast operating instrument of low inertia with its rest position at the right-hand end of the scale ; this is because the " no-signal " output from the variable-mu valve corresponds to full-scale deflexion which therefore deflects the needle to the left. The scale consists of seven divisions numbered " 1 " to " 7 ", each division representing 4 db except the lowest which is 6 db.

Performances and tests

The time of rise of the instrument is such that the peak amplitude of a square pulse of 4 milliseconds duration is registered as 80% of its absolute value, while the time of fall is such that the needle falls from " 7 " to " 1 " (i.e. 26 db) in 3.0 seconds $\pm$ 0.5 seconds. The rise time is checked by means of the voltage pulse which occurs across a 600 ohms resistor when a condenser of 5 microfarads is discharged through it. The condenser is charged to the peak voltage of steady tone which would deflect the meter to " 6 " and the pulse must deflect the meter to " 4 " $\pm$ 1 db.

The meter instrument itself is required to deflect to 97% of full-scale when a condenser of 10 microfarads is charged in series with a 100,000 ohms resistor and the meter from a voltage which, with the condenser short-circuited, would produce full-scale deflexion. Moreover when this voltage is suddenly applied with the condenser short-circuited the meter needle must not overshoot full-scale deflexion by more than 5%.

Use

The Peak Programme Meter is generally used to monitor programme of " zero " volume in a 600 ohms circuit. This implies that pure steady tone at 0.775 volts shall correspond to 40% modulation of the system and the programme meter is adjusted to read " 4 " under these conditions. A reading " 6 " therefore corresponds to 100% modulation so that peaks 20 db or more below this level are readily seen and overloads of 4 db are measurable.

ANNEX 15

MAXIMUM AMPLITUDE INDICATOR TYPE U 21 USED IN THE FEDERAL GERMAN REPUBLIC

Application

The maximum amplitude indicator described below is suitable for monitoring the peak voltage during a radio broadcast program. For this reason it is particularly useful where overloading of transmission equipment must be avoided or where the dynamic range of the transmission is to be controlled, e.g. in studios, at the input and output ends of a radio transmission circuit, or at certain intermediate points and at the input to a radio transmitter.

General characteristics

The maximum amplitude indicator has a very short integration time so as to be capable of following the amplitude of a rapidly increasing voltage. The time for the needle to return to zero is relatively large so that very brief voltage peaks can be easily observed. The indicating instrument is furnished with an illuminated spot and consequently has a short rise time. Several secondary indicating instruments and a recorder can be connected. The scale is arranged to be approximately logarithmic and calibrated in nepers or decibels. An additional scale is calibrated in percentage (100% corresponds to the highest voltage which can be used). The frequency range is somewhat larger than that recommended by the C.C.I.F. for normal radio broadcast transmission circuits. The input impedance is so high relative to 600 ohms that the error due to connecting the maximum amplitude indicator across a circuit can be neglected.

Specified values

Frequency range	30 to 15,000 c/s
Input impedance	greater than 30 000 ohms
Dynamic range	1 : 300
corresponding to	0.3 to 100%
(An additional range from 100% to 180% for use as an overload range, is marked in red).	

Input voltage for 100% (corresponding to 8 or 32 milliwatts in 60 ohms)	2.2 or 4.4 volts r.m.s.,* switchable
Variation of reading with frequency, at 100%	less than ±10% relative to the value at 800 c/s
Rectifier characteristic	linear
Integration time for sudden application of a sinusoidal voltage at 100%	
to reach 80%	about 5 milliseconds
to reach 90%	about 10 milliseconds
Rise time	
to reach 90%	about 80 milliseconds
Time to return from 100% to 10% on sudden disconnection of a sinusoidal voltage	about 1 or 2 seconds switchable
Overshoot	Maximum 10%
Power Supply	40 to 60 c/s, 110 or 220 volts, about 55 watts.

* Some administrations have specified other values for the voltage at the control points ; in Germany there are other maximum amplitude indicators which are referred, for example, to input voltages of 1.55 or 3.1.

ANNEX 16

COMPARISON TESTS OF DIFFERENT TYPES OF VOLUME METERS CARRIED OUT BY THE BRITISH TELEPHONE ADMINISTRATION

The British Post Office has made some measurements comparing the readings of some different types of volume meters, used as speech voltmeters, on the same passages of recorded conversational speech.

For this purpose an apparatus, termed the Speech Voltmeter Type 4 (SV4), was constructed with the object of eliminating the subjective assessment of the reading of a meter, the needle of which is continually moving. This was done by stimulating the mechanical properties of the moving parts of a speech voltmeter by an analogous electrical circuit such that the instantaneous value of the (rectified) voltage is proportional to the corresponding instantaneous deflexion of the needle of the meter which is replaced by the electrical circuit.

An electronic counting device is added to record the number of occasions on which each of ten present levels is exceeded by the (rectified) voltage. From these data an arbitrary but precise specification of the " level " of the speech can be derived. The one used for the purposes of these tests was as follows :

The level at which the maximum number of " level crossings " occurs corresponds approximately to the " long-time " r.m.s. value but is ill defined. The level above this at which half the maximum number of " level crossings " occurs is, however, well defined. A level 3 db higher than this one is taken as the reading. The figure of 3 db was previously arrived at empirically by comparison with readings by a team of experienced observers, on a British Post Office Speech Voltmeter Type 3 (identical to the A.R.A.E.N. volume meter).

The S.V. 4 was made so that different and known characteristics could be inserted in respect of :

(a) The law of rectification (law of addition of components of different frequencies).

(b) The integration time (or rise time).

(c) The overswing.

TABLE I

Relative readings of Speech Voltmeters

Item	Type of meter measurement	Law of Rectification (Note 1)	99% Rise time (mS)	Integration time (mS)	Overswing (per cent)	Relative Reading (db.)
A	Speech Voltmeter, Type 3 (S. V. 3) (identical to A.R.A.E.N. volume meter).	2	230	110 (approx.)	2.5	0
B	S. V. 3, changed only in respect of law of rectification	1	230	110 (approx.)	2.5	−2.8
C	v.u. –meter.	1.0	300	120 (approx.)	1.0 to 1.5 (note 2)	−3.5
D	v.u. –meter,	1.4	300	120 (approx.)	1.0 to 1.5 (note 2)	−2.2
E	C.C.I.F. Volume Indicator.	2	650 (approx.)	200 *	0	−1.4
F	C.C.I.F. Volume Indicator.	2	410 (approx.)	200 *	5	−2.4
G	Level exceeded for 1% of time on continuous speech.	—	—	—	—	+5.8
H	B.B.C. Peak Programme Meter.	1	—	—	—	+6.8
J	Thermal meter read on long utterances.	2	12,400 (approx.)	2700	0	−4.5
K	" Long time " rms value (conditions speech).	—	—	—	—	−6.9

Note 1. — The index n in the formula $V_{(out)} = V_{(in)}{}^{n}$ (for each half wave).

Note 2. — No appreciable difference was noticed within this range.

* *Note by the C.C.I.F. Secretariat.* — For this type of volume meter, the " integration time " is defined by the C.C.I.F. as the " minimum time for which a sinusoidal alternating voltage must be applied to the terminals of the instrument for the needle of the indicating device to reach, within 0.2 neper or 2 about 2 decibels, the deflexion which would have been obtained had the same voltage been applied indefinitely ". A logarithmic interval of 2 decibels corresponds to a percentage of 79.5% and an interval of 0.2 nepers to a percentage of 82%.

The results obtained by comparison of different types of speech voltmeter for speech currents from a high-quality microphone are shown in Table I as readings relative to the Speech Voltmeter Type 3. Except for items H, J and K, all the results refer to measurements made with a SV4 adjusted to the characteristics stated in the table. Some other observations have also been made of relative readings, by observers, of actual meters of the types stated. The results agree with those in the table within about ± 1 db, which is less than the variation likely in reading a level by observing a continually moving needle.

The effects of changing the law of rectification in a speech voltmeter, otherwise similar to the Speech Voltmeter Type 3, are shown in Table II for speech both from a high-quality microphone and from a British commercial telephone set (with Transmitter Inset No. 13). The results are referred to square-law rectification ($n = 2$) as datum.

TABLE II

Effect of law of rectification (relative to square-law)

	Law of rectification, $V_{(out)} = V_{(in)}{}^n$	
	$n = 1.5$	$n = 1.0$
Commercial telephone	No measurement made	−3.6 db
High-quality	−1.1 db	−2.8 db

The influence of overswing and rise time (here defined as the time which elapses between the application of a steady voltage to the meter and a deflection on the meter equal to 99% of the final deflection) is shown in Table III, being expressed relative to the value corresponding to an overswing of zero and rise time of 1.0 seconds.

TABLE III

Rise time (secs)	Overswing (per cent)		
	0	1	5
0.25	+2.6 db	+2.6 db	−2.4 db
0.5	+2.0	+0.3	+0.5
1.0	0	−0.8	−1.2

The following general conclusions have been reached for speech voltmeters operating within the ranges of values of the parameters which were covered by these tests.

(1) Effects due to changes of the law of rectification are substantially independent of the dynamic characteristics of the meter, but they do depend on the type of circuit (i.e. whether commercial or high-quality).

(2) Effects due to changes of rise time and of overswing are mutually dependent but are independent of whether the circuit is commercial or high-quality.

(3) Variation of overswing in the range 1 to 5 per cent produces negligible effect.

ANNEX 17

APPARATUS STANDARDIZED IN THE UNITED STATES FOR THE OBJECTIVE MEASUREMENT OF ROOM NOISE

(Standard Z.24.3.1944 "American Standard Sound Level Meters for Measurement of Noise and other Sounds", approved 28th July 1944 by the American Standards Association)

Scope and Purpose

Scope. — The purpose of a set of standards for sound level meters is to bring about a condition such that, if a given noise is measured with any meter meeting the design objective, the result will be the same as that which would be obtained with any other similarly designed meter, and will approximate the loudness level which would be obtained by the more elaborate ear-balance method described in the American Standard for Noise Measurement Z.24.2.1942, or latest revision thereof.

Purpose. — At present it is impractical to build sound level meters that exactly meet the design objectives. Therefore certain tolerances in the response-frequency characteristics of the meter have been specified. In many cases differences of practical importance may be expected between the results obtained with different meters unless corrections are applied which take into account deviations of individual meters from the design objective.

Due to the complexities in the human hearing mechanism which cannot, at present, be adequately simulated in a sound level meter, it is expected that in many instances the approximations to loudness level will not be close, although differences in loudness level may be more closely approximated. The discrepancies involved can be determined more satisfactorily by comparison of meter readings and ear-balance loudness levels after a certain degree of uniformity has been attained among the meters in use by investigators in the field of sound measurements. It is, therefore, recommended that the standards for sound level meters established herein be accepted until further work and experience reveal advantages to be gained by revision.

Since the microphones used at present in sound level meters have directional characteristics, it will ordinarily be desirable, when sound measurements are made, to average the results of readings taken with the microphone turned in several different directions, inclusing particularly those directions from which the sound appears to come. If this is done and the standards given below are met, differences between the results obtained with meters of different manufacture will be minimized.

It is recognized that, in special cases, certain non-standard characteristics may be desirable, and it is not intended that these standards preclude the use of special characteristics for special problems. It is believed, however, that these standards will aid in promoting uniformity in meter measurements of various types of sounds and will thus be of value in the sound measurements field.

Definitions

Scale. — A sound level meter shall have a decibel scale. The quantity measured by sound level meters is weighted sound pressure level and shall be referred to as " sound level " ; e.g., a sound level meter reading of 60 decibels shall be referred to as " 60 decibel sound level ", or " sound level of 60 decibels ".

Response-frequency characteristics. — The over-all free-field response-frequency characteristics of a sound level meter shall be that shown in Figure 1. Curves A and B are the 40 and 70 decibel equal loudness contours, respectively, each modified by the differences between random and normal free-field thresholds.* Curve C is a flat response.

Sound level measurements. — When reporting the results of sound level measurements the response-frequency employed should always be given.

The tolerable deviations from the standard response-frequency characteristics at different frequencies are shown by the dashed curves on Figure 1 and in tabular form in Table II. Appendix D. To determine whether or not the tolerances

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* American Standard for Noise Measurement, Z.24.2.1942, or latest revision thereof; L. J. Sivian and S. D. White " On Minimum Audible Sound Fields." *Journal of the Acoustical Society of America,* Vol. IV, No. 4, April 1933.

are exceeded, the procedure outlined in Appendix D is to be followed. The response-frequency characteristic of a sound level meter shall lie within the above tolerances for any setting of the gain controls with which the response-frequency characteristic is intended to be used and for the range of operating and ambient temperatures within which the meter is to be used.

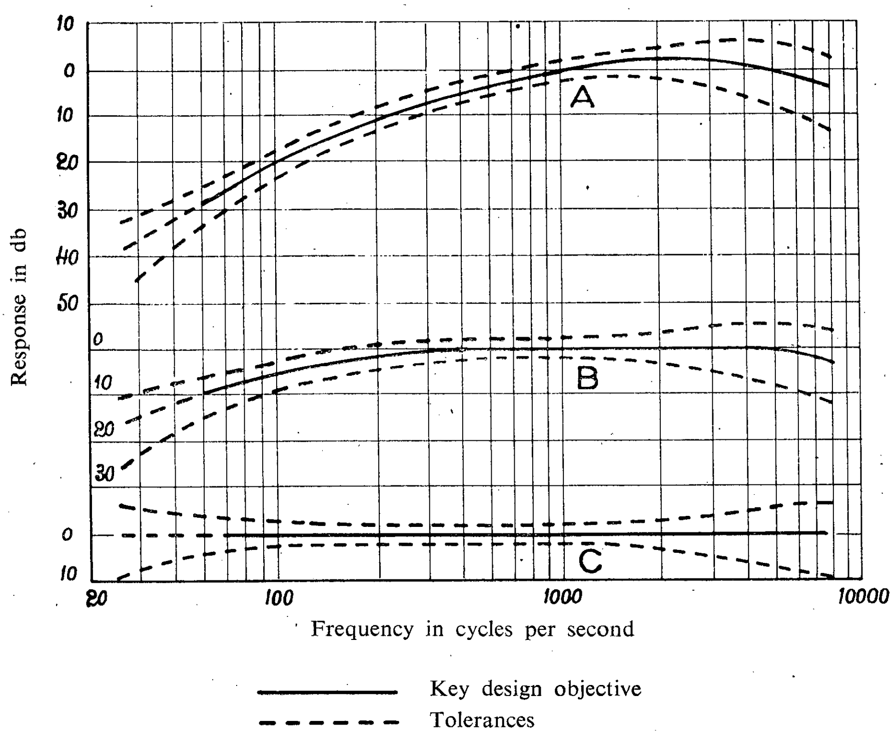

FIGURE 1. — *Over-all free-field response-frequency characteristics and tolerances for sound level meters*

Free-field response. — The over-all free-field response at a given frequency shall be the average response-frequency characteristic for free waves at various angles of incidence up to 90 degrees from the normal, assuming all angles of incidence within this region equally probable. The method of averaging is described in Appendix A.

The difference between the normal-incidence response and that for any other direction of incidence shall not exceed 5 decibels at any frequency up to 1,000 cycles and 20 decibels at any frequency between 1,000 and 3,000 cycles.

Reference point. — The reference point of the decibel scale incorporated in sound level meters shall be the reference sound pressure of 0.0002 dynes per square centimeter at 1,000 cycles.

Notes. — Some differences will usually exist between the standard response-frequency characteristics and the response-frequency characteristics of an individual meter. In order that the primary calibration will result in a given noise of a general type producing substantially the same meter readings in meters whose response-frequency characteristics vary in different ways from the standard curves, the procedure outlined in Appendix C is to be used.

Any possible deviations in calibration of a sound level meter due to the effects of variable ambient and operating temperatures shall be specified.

Rule of combination. — The characteristic of the sound level meter shall be such that the meter will indicate the square root of the sum of the squares of the weighted sound pressures of the different single frequency components of the complex wave.

Note. — For most sound level meters, this rule will not be exactly followed for all types of input waves. Tolerable deviations from the rule and tests to determine these deviations from the rule and tests to determine these deviations are given in Appendix B.

Dynamic characteristic of indicating instrument. — The dynamic characteristic of the indicating instrument forming a part of a sound level meter shall be such that the following over-all dynamic characteristics will be met by the meter.

(a) The deflection of the indicating instrument for a constant 1,000 cycle sinusoidal input to the sound level meter shall be equalled by the maximum deflection of the indicating instrument for an input to the sound level meter consisting of a pulse of 1,000 cycle power which has the same magnitude as the constant input and a time of duration lying between 0.2 and 0.25 seconds.

(b) The deflection of the indicating instrument for a constant 1,000 cycle sinusoidal input to the sound level meter shall not be exceeded by more than 1.0 decibels by the maximum deflection of the indicating instrument obtained upon the sudden application of the constant input.

(c) The deflection of the indicating instrument for a constant 1,000 cycle sinusoidal input to the sound level meter shall be at least 1 decibel greater than the maximum deflection of the indicating instrument for an input to the sound level meter consisting of a 1,000 cycle pulse which has the same magnitude as the constant input and a time of duration of 0.1 seconds.

These characteristics should hold for any part of the scale of the indicating instrument.

In cases where it is not desirable to observe rapid fluctuations in level, a slow acting indicating instrument instead of the one described above, can be used for convenience to obtain an average result. On steady sounds the reading of this indicating instrument should be the same as that of the rapidly acting instrument.

APPENDIX A

Response-frequency characteristics of sound level meters

Meter reading. — When the microphone forming a part of a sound level meter responds differently to sound waves arriving with different angles of incidence, the reading of the meter will depend upon the directional characteristics of the sound waves at the point of measurement. In more or less diffuse sound fields, sound level meters having microphones with different directional properties will give different readings in they have been previously adjusted to give equal readings in the, same normal incidence sound field. These differences may be minimized by :

(a) limiting the maximum difference between the normal-incidence response and that at any other angle, and

(b) using a method of specifying the response-frequency characteristic which takes into account the probable angles of incidence of the more important components of the sound. The maximum tolerable differences in response, and a method of specifying the over-all free-field response, are given in the definition above ; the following paragraphs give the details of the method of determining the response.

Method of determining response. — The over-all free-field response-frequency characteristic of a sound level meter as defined above may be computed from responses obtained to sound waves approaching the microphones at different angles within the specified region. The computation consists of averaging the microphone outputs for plane waves of a given frequency arriving successively from all directions within the given region, assuming that all such directions of arrival are equally probable. Since, in general, the variation of microphone output with angle of incidence is not easily expressible mathematically, it will be found practicable to use finite steps in the averaging process. An expression which has been found to give satisfactory results with condenser or dynamic microphones is

$$P = 0.034\ P_0 + 0.259\ P_{30} + 0.448\ P_{60} + 0.259\ P_{90} \qquad (1)$$

where P = average power output of microphone ;
P_0 = power output from normal-incidence wave ;
P_{30} = power output from wave 30 degrees to normal ;
P_{60} = power output from wave 60 degrees to normal ;
P_{90} = power output from wave 90 degrees to normal.

The following is given as an example of the manner in which this expression is to be used. Measure the microphone response to free progressive waves at 0, 30, 60 and 90 degrees from normal incidence. Suppose that the following data are obtained.

Frequency	Angle	Measured power output for free-field Input of 60 decibels above reference pressure
4000 Cycles	0°	3.00 micro-microwatts
	30°	1.9 —
	60°	0.75 —
	90°	0.38 —

Substituting in Formula 1,

$$P = 0.034 \times 3 + 0.259 \times 1.9 + 0.448 \times 0.75 + 0.259 \times 0.38 = 1.03$$

Hence the average output power at this frequency for an input of 60 decibels above reference pressure is 1.03 micro-microwatts. (This $1.03/3 = 0.34$ of the output power for normal incidence, so that the over-all free-field response at 4,000 cycles, as defined in 2.4, is 10 log. $0.34 = -4.6$ decibels referred to the normal-incidence response).

Formula 1 is derived as follows :

Assume that the microphone is symmetrical about an axis normal to the plane of the diaphragm, so that its output is constant for all directions of approach of sound waves which make an angle x with this axis. Make the approximation that the output is constant at a value P_0 for all values of x lying between 0 and 15 degrees, that it is constant at another value P_{30} for all values of x lying between 15 and 45 degrees, and so on.

Imagine a hemisphere, and a cone having its axis perpendicular to, and its vertex at the center of, the great circle on the hemisphere and having a vertex angle of 2×15 degrees. Since all directions of approach of the sound waves within the hemisphere are equally likely, the chance that a given sound wave will approach at some direction for which x lies between 0 and 15 degrees is the ratio of the area cut from the surface of the hemisphere by the cone to the total area of the hemisphere ; this is found to be 0.034. Hence the contribution of the directions comprised within this cone to the output power is 0.034 P_0. Similarly, the contribution of the directions lying between this cone and a cone with a vertex angle of 2×45 degrees is found to be 0.259 P_{30} ; and so on. The total output is the sum of the contributions from all directions of incidence considered ; this leads to Formula 1.

If the assumption of symmetry of response about the axis normal to the plane of the diaphragm cannot be made, a number of measurements should be made, each representing the response for a certain area of the hemisphere in front of the microphone. The average of the powers indicated by these readings, compared with the power indicated by the reading on the normal to the diaphragm will provide a means of determining the difference between the over-all free-field response-frequency characteristic, as defined in 2.4, and the normal incidence response, providing all areas for which readings are taken are about equal in magnitude.

APPENDIX B

Tests for root mean square addition

Circuit arrangements. — Make circuit arrangements so that two single frequencies with negligibly small harmonics may be introduced into the sound level meter input simultaneously and at the same time be controlled independently. The wave form of the frequencies should be purely sinusoidal and ideally they should be introduced acoustically, although if it is found preferable, because of distortion caused by loudspeaker or receiver characteristics, sinusoidal waves may be introduced electrically.

The frequencies must be in non-harmonic relation to each other and separated sufficiently so that the indicating instrument will not follow the beats. Choosing the frequencies above 800 cycles will aid in making the effects of oscillator harmonics negligibly small. Adjust the magnitude of each frequency so that it alone produces a mid-scale deflection of the indicating instrument. When adjusting one of the single frequencies the impedance relations between this source, the sound level meter input and the other source must be identical in every respect with the relations that exist when both sources are applied. As stated above, when adjusting one of the single frequencies the other must make no contribution to the deflection, though the impedance that it presents to the other frequency source and to the meter must be the same as before. When each frequency has been adjusted so that it alone produces a mid-scale deflection, both are put on together, and both are attenuated by the same amount until the mid-scale deflection is again obtained. This amount of attenuation should be 3 ± 0.5 decibels.

The experiment should be repeated at other points on the scale, including full scale.

Among the sounds of which measurements may frequently be desired will be steady peaked waves, and sounds of such short duration that the indicating instrument will not follow them. For sounds such as these, the addition property of the rectifier should extend somewhat beyond the region normally brought into play when a full-scale deflection is produced by a sine wave. Also, the amplifier should have some spare carrying capacity for these varieties of sound. Hence it is necessary to make tests to ensure that the amplifier and rectifier characteristics are such that addition of components holds within certain limits for these types of sounds. To determine the adequacy of the meter in this regard, make the following test :

Insert a resistance network between the indicating instrument terminals and the points to which these are normally connected, so that the same resistance is presented to the rectifier as before, but so that, for a given voltage applied to the rectifier, less current flows through the indicating instrument than before. This resistance network should be such that its insertion causes a change in indicating instrument reading of about 7 or 8 decibels.

In the same manner as before make circuit arrangements so that two single frequencies in non-harmonic relation to each other and separated sufficiently so that the indicating instrument will not follow the beats, may be introduced into the sound level meter input simultaneously and at the same time be controlled independently. Without the resistance network, adjust the magnitude of each frequency so that it alone produces a full-scale deflection of the indicating instrument. Insert the resistance network and increase the magnitude of each single frequency by 3 decibels. Note the indicating-instrument deflection from either of the two single-frequency sources. (The deflection from each should be the same). Put on both single frequencies simultaneously and attenuate both equally until the same deflection is again obtained. This amount of attenuation should be 3 ± 1 decibels.

Repeat the experiment except that after inserting the resistance network, increase the magnitude of each frequency by 6 instead of 3 decibels. Note the deflections from each single frequency (the deflections should be the same) and then put both on, simultaneously, attenuating both, equally, until the deflection obtained with either, alone, is again obtained. In this case the attenuation should be 3 ± 1.5 decibels.

These limits should be met for all positions of the calibrating and measuring gain controls.

APPENDIX C

Calibration adjustment of individual sound level meter

An individual sound level meter having a response-frequency characteristic differing from the design objective shall be so calibrated that its reading on noise of a general character will agree substantially with the reading obtained with a meter having the design objective characteristic. The process consists of computing for a specified noise the weighted reading of, first, a meter having the design objective characteristic and, second, that of the individual meter. These readings are then made equal by suitably adjusting the calibration of the individual meter.

The adjustment depends not only upon the individual and design objective response-frequency characteristics but also upon the frequency distribution of the noise which is specified. For this purpose there has been chosen the spectrum having the relative frequency distribution shown on Column 2 of Table 1. This represents the average, in 100 cycle bands, of the spectra of a variety of noises. This spectrum and the design objective response (Column 3), both expressed in decibels relative to the 1,000 cycle level, are then added algebraically. The resultant sums (Column 4) represent the contribution to the meter reading in decibels in each frequency band. The weighted pressures squared corresponding to each of these decibel figures are then determined (Column 5) and totaled to give the Sum of Weighted Pressures Squared relative to that in the band at 1,000 cycles. Since it is apparent that the total sum would not be greatly affected by frequencies above 3,000 cycles, for the chosen noise, the summation is purposely stopped at this frequency.

By going through these same processes substituting in Column 3 the frequency response of the individual meter (plotted so that it passes through 0 decibel at 1 000 cycles) another value for the Sum of Weighted Pressures Squared is obtained. If this value is the same as that obtained from the computations using the design objective response, the individual meter should be calibrated to give a reading of 0 for reference sound pressure at 1,000 cycles. If the two sums are not equal, the individual meter should be calibrated so that the reference pressure at 1,000 cycles produces a reading of X decibels, where

$$X = 10 \log_{10} \frac{\text{Sum of Weighted Pressures Squared, Design Objective Meter}}{\text{Sum of Weighted Pressures Squared, Individual Meter}}$$

The value can, of course, be either positive or negative—positive, when the sum of the weighted pressures squared for the individual meter is less than the corresponding sum for the design objective meter ; negative, when the individual meter has the greater sum.

A similar process should be used to calibrate the meter for use with the " B " and " C " weighting.

TABLE I

(1)	(2)	(3)	(4)	(5)
Frequency	Relative Pressure Level in db in 100 Cycle Bands of General Type of Noise db	Design Objective Response (Curve A) db	Sum db	Weighted Pressures Squared *
100	+ 9.7	−19.3	− 9.6	0.11
200	+ 8.2	−11.2	− 3.0	0.50
300	+ 6.9	− 7.4	− 0.5	0.89
400	+ 5.7	− 5.1	+ 0.6	1.15
500	+ 4.5	− 3.7	+ 0.8	1.20
600	+ 3.4	− 2.5	+ 0.9	1.23
700	+ 2.3	− 1.6	+ 0.7	1.18
800	+ 1.5	− 0.9	+ 0.6	1.15
900	+ 0.7	− 0.3	+ 0.4	1.10
1 000	0	0	0	1.00
1 100	− 0.8	+ 0.5	− 0.3	0.93
1 200	− 1.5	+ 0.9	− 0.6	0.87
1 300	− 2.3	+ 1.1	− 1.2	0.76
1 400	− 2.9	+ 1.2	− 1.7	0.68
1 500	− 3.6	+ 1.5	− 2.1	0.62
1 600	− 4.2	+ 1.6	− 2.6	0.55
1 700	− 4.9	+ 1.7	− 3.2	0.48
1 800	− 5.5	+ 1.8	− 3.7	0.43
1 900	− 6.2	+ 1.9	− 4.3	0.37
2 000	− 6.8	+ 2.0	− 4.8	0.33
2 100	− 7.5	+ 2.0	− 5.5	0.28
2 200	− 8.1	+ 2.0	− 6.1	0.24
2 300	− 8.7	+ 2.0	− 6.7	0.21
2 400	− 9.3	+ 2.0	− 7.3	0.19
2 500	− 9.8	+ 2.0	− 7.8	0.17
2 600	−10.3	+ 2.0	− 8.3	0.15
2 700	−11.0	+ 2.0	− 9.0	0.13
2 800	−11.5	+ 2.0	− 9.5	0.11
2 900	−12.0	+ 1.9	−10.1	0.10
3 000	−12.5	+ 1.8	−10.7	0.08
		Sum of Weighted Pressures Squared		17.19

* Weighted Pressures Squared = $\text{Antilogarithm}_{10} \frac{\text{db}}{10}$.

APPENDIX D

Tolerances in response-frequency characteristic

To determine whether the frequency response of an individual meter falls within the tolerances, the individual response, shifted by X decibels, as determined for the noise of general character specified in Appendix C, should be super-imposed on the design,

TABLE II

Frequency (cps)	Design Objective Response (db) Relative to Response at 1000 cps			Tolerances (db) Curves A, B, C
	Curve A	Curve B	Curve C	
25	−39.5	−16.2	0	+6.0 −9.5
60	−26.2	− 9.0	0	±3.0
100	−19.3	− 5.7	0	±3.0
200	−11.2	− 2.2	0	±2.5
300	− 7.4	− 1.0	0	±2.0
400	− 5.1	− 0.6	0	±2.0
500	− 3.7	− 0.4	0	±2.0
600	− 2.5	− 0.3	0	±2.0
700	− 1.6	− 0.1	0	±2.0
800	− 0.9	0	0	±2.0
900	− 0.3	0	0	±2.0
1 000	0	0	0	±2.0
1 100	+ 0.5	0	0	±2.0
1 200	+ 0.9	0	0	+2.0 −2.5
1 300	+ 1.1	0	0	+2.0 −2.5
1 400	+ 1.2	0	0	+2.0 −3.0
1 500	+ 1.5	0	0	+2.0 −3.0
1 600	+ 1.6	0	0	+2.5 −3.0
1 700	+ 1.7	0	0	+2.5 −3.5
1 800	+ 1.8	0	0	+2.5 −3.5
1 900	+ 1.9	0	0	+3.0 −4.0
2 000	+ 2.0	0	0	+3.0 −4.0
2 100	+ 2.0	0	0	+3.0 −4.0
2 200	+ 2.0	0	0	+3.0 −4.0
2 300	+ 2.0	0	0	+3.5 −4.5
2 400	+ 2.0	0	0	+3.5 −4.5
2 500	+ 2.0	0	0	+3.5 −5.0
2 600	+ 2.0	0	0	+3.5 −5.0
2 700	+ 2.0	0	0	+3.5 −5.0
2 800	+ 2.0	0	0	+3.5 −5.0
2 900	+ 1.9	0	0	+4.0 −5.5
3 000	+ 1.8	0	0	+4.0 −5.5
3 500	+ 1.4	0	0	+4.5 −6.5
4 000	+ 1.0	0	0	+5.0 −7.0
4 500	+ 0.5	− 0.2	0	+5.0 −7.5
5 000	0	− 0.5	0	+5.0 −7.5
6 000	− 1.0	− 1.1	0	+5.5 −8.0
7 000	− 2.0	− 1.9	0	+6.0 −9.0
8 000	− 3.0	− 2.7	0	+6.0 −9.5

objective response. In this shifted position, which is upward relative to the standard response if the sum of the weighted pressures squared for the individual meter is less than the corresponding sum for the design objective meter, the individual meter response should fall within the dashed lines shown on Figure 1 and within the tolerances shown in Table II. This procedure should be followed in examining the tolerances with the " B " and " C " weightings.

APPENDIX E

Correction of reading on specific noises

The quantity desired of a sound level measurement is the reading given by a meter which exactly meets the design objectives. An actual meter calibrated as described in Appendix C to read correctly on a " noise of general character " may not, however, read correctly on other noises having a different distribution of pressures with frequency. However, when meters are used for measuring a noise whose spectrum differs markedly from that of the spectrum used for calibrating the meter, the reading can be corrected closely to that given by a meter with the design objective response. For this purpose the user of the meter requires a knowledge of

1. the response-frequency characteristic of the individual meter after the adjustment, if any, made by the manufacturer following the procedure of Appendix C, and
2. the approximate spectrum of the noise that is to be measured.

The procedure for determining the correction to be made to the measured sound level involves a computation of the " Sum of the Weighted Pressures Squared " for

1. a meter having the design objective response, and
2. the individual meter, using its adjusted response.

The method is identical with that outlined in Appendix C with the exception that the computations are made with the spectrum of the particular noise in question instead of the spectrum of the " noise of general character ".

The correction of C decibels to be applied to the sound meter reading is given by

$$C = 10 \log_{10} \left(\frac{S_1}{S_2}\right).$$

where S_1 is the sum of the weighted pressures squared of the particular noise, for the meter of design objective response, and S_2 is the corresponding sum for the actual response of the particular meter as adjusted by the manufacturer. If S_2 is less than S_1, the correction is positive and is added to the meter reading ; if S_2 is the larger, the correction is negative and the meter reading is consequently reduced.

ANNEX 18

RECIPROCITY METHOD FOR THE CALIBRATION OF CONDENSER MICROPHONES

Memorandum of January 22, 1947 of the American Telephone and Telegraph Company

1. Introduction

During the war it was found advantageous at several laboratories, including the Bell Telephone Laboratories, to adopt a system of calibration for condenser microphones, called the reciprocity method, which has supplanted the thermophone method previously in use. There were a number of reasons for this step, which need not be discussed here, except to mention that a very satisfactory agreement has been found between various laboratories, working independently, when round-robin comparisons are made.

A fundamental paper on the subject was published by R. K. Cook, *Journal of Research, National Bureau of Standards*, November 1940, pp. 489-505. It is not the intention of the present memorandum to cover the same ground, but rather to concentrate on an outline of the physical principles. This relates particularly to the derivation of the reciprocity relation, which is stated but not proven in the paper, and will supply some omitted steps in the application of the relation.

2. Outline of reciprocity method

In its simplest form, the reciprocity method yields the response curve for two microphones, I and II, simultaneously. One of these, say I, must be capable of working also as a source of sound. Another source, III, which may be another condenser microphone, is required in one of the steps.

The responses obtained are coupler responses, that is, the acoustic pressures are generated and applied to the microphones within a closed chamber which connects source to microphone. The reciprocity relation provides a method by which the pressure in the coupler can be computed in terms of easily measured physical quantities. To avoid propagation effects, such as standing waves, the dimensions of the coupler must be small with respect to the wave length of sound in the gas contained in the coupler.

The steps in the calibration are then :

(a) Using I as source and II as microphone the ratio of electrical output to electrical input is found. This ratio is proportional to the product of the responses of I and II.

(b) Using III as a constant source, the electrical outputs of I and II are found in succession. One of these is divided by the other, giving the quotient of the responses of I and II.

(c) Since the factor of proportionality in the first step can be calculated from the reciprocity relation, the results of *(a)* and *(b)* can be solved simultaneously to yield the separate responses of I and II.

3. Reciprocity relation

In a passive linear electrical network the reciprocity relation may be stated as follows : If an impedanceless generator in branch 1 produces a certain current in branch 2, then the same current will be produced in branch 1 if the generator is moved to branch 2. In an electro-acoustic transducer the quantities reciprocally related are not so immediately evident to the intuition. They will be found below by considering a condenser microphone, first, as a source, and, second, as an absorber of sound.

The objective in the first case is to find the change in the volume (cubic contents) of the condenser when a small increment of potential difference is applied to its plates; this is worked out in Section 3.1. In the second case, the inquiry relates to the amount of electrical charge caused to flow from the condenser when a small increment of acoustic pressure is applied to the diaphragm; this is discussed in Section 3.2. The results of these sections demonstrate the reciprocal relation. They are then restated in terms of the response of the instrument as a microphone (Section 3.3). The remaining sections describe the principle of the calibration.

It will be evident from the character of the approximations made below that all variations considered, of potential difference, charge, capacitance, displacement etc., are restricted to such small values that the condenser microphone can be considered a linear device. In these circumstances, a natural and concise method of approach is the use of generalized energy relations, in terms of differential equations, to yield the equations of motion of the diaphragm. Another method of approach is that of the electroacoustic analogies to electric networks; this of course, is the same method in different form. The relations may also be derived directly from the formulas describing the capacity and the eletrostatic attraction between the plates of a parallel plate condenser. This is the method chosen; the algebra is more tedious, but the nature of the physical considerations may be more apparent.

3.1 *As source*

Referring to Figure 1 *a*, attached, consider that a popularizing battery E is applied to one plate of the microphone, and that the capacity of the microphone is C. If the separation of the parallel plates of the condenser is D, the electrostatic force per unit area between the plates is

$$F = \frac{E^2}{8\pi\, D^2} \text{ (dynes/cm}^2\text{)}$$

Next suppose the potential difference between the plates to increase by e. This will increase the force and cause the diaphragm to move closer to the back plate. The decreased separation in turn increases the force still more. The movement will continue until stopped by the elastic contraints of the diaphragm. In equilibrium, the displacement will be d_1 centimeters, where :

$$F + f_1 = \frac{(E + e)^2}{8\pi\, (D - d_1)^2}$$

The increase in force, f_1 is :

$$f_1 = \frac{(E + e)^2}{8\pi\, (D - d_1)^2} - F = \frac{E^2}{8\pi\, D^2} \left[\frac{\left(1 + \frac{e}{E}\right)^2}{\left(1 - \frac{d_1}{D}\right)^2} - 1 \right]$$

Now the total force over the whole area of the diaphragm, A, is Af_1 dynes. This is consistent with a displacement of d_1 centimeters, when

$$d_1 = KAf_1$$

the constant K being defined as the amount by which the diaphragm yields per dyne applied. Substituting for f_1 :

$$\frac{d_1}{KA} = \frac{E^2}{8\pi\, D^2} \left[\frac{(1 + e/E)^2}{(1 - d_1/D)^2} - 1 \right]$$

This is to be solved for d_1. Expanding the quantities in the bracket, dividing and neglecting terms of orders higher than the first in e/E and d_1/D :

$$\frac{d_1}{KA} = \frac{E^2}{8\pi\, D^2} \left[2\frac{e}{E} + 2\frac{d_1}{D} \right]$$

Rearranging terms :

$$d_1 = \frac{KAEe}{4\pi\, D^2 \left(1 - \dfrac{KAE^2}{4\pi\, D^3}\right)} \text{ cm}$$

Before the application of e the volume of the condenser was AD cubic centimeters. The application of e changes the volume by a small amount, v, where

$$v = Ad_1$$

Or,

$$v = \frac{KA^2 E}{4\pi\, D^2 \left(1 - \dfrac{KAE^2}{4\pi\, D^3}\right)} e \text{ cm}^3$$

The capacitance (static value) of the condenser, neglecting edge effects is :

$$C = \frac{A}{4\pi\, D} \text{ e. s. u.}$$

So that :

$$v = \frac{4\pi\, KC^2 E}{1 - \dfrac{KCE^2}{D^2}} e \text{ cm}^3$$

3.2 *As microphone*

Referring to Figure 1 *b*, attached, consider that a plane progressive acoustic wave strikes the diaphragm, causing a small increment of pressure, p. The diaphragm moves, and as the separation between the plates decreases the electrostatic force of attraction from the polarizing battery increases. The combination of acoustic pressure and electrostatic attraction pulls the diaphragm toward the back plate until the elastic constraints of the diaphragm stop further motion. The distance of travel is then d_2 centimeters, where

$$d_2 = KA\,(p + f_2)$$

and K is the same compliance as in Section 3.1.

(The fact that the diaphragm is clamped at its edges and thus cannot move piston-wise may raise some doubts at this point. The mode of flexure is, of course, much more complicated than a piston. But if the force is applied uniformly over the diaphragm in both the electrostatic and the acoustic cases, the resulting motion is the same, and K may be considered the same, even though K is a function of the location of the point on the diaphragm at which the displacement is measured. However, d as used in the formulas is a kind of average value, justified by being small in comparison with D. The factor K is a function of the acoustic loading as well as the mechanical properties of the diaphragm.)

When the instrument is used as a microphone the increase in electrostatic attraction, f_2, is found from :

$$F + f_2 = \frac{E^2}{8\pi (D - d_2)^2}$$

since the polarizing potential is held constant. Approximating as before :

$$f_2 = \frac{E^2 d_2}{4\pi D^3} \text{ dynes}$$

d_2 is then obtained by solving :

$$d_2 = KA\left(p + \frac{E^2 d_2}{4\pi D^3}\right)$$

giving :

$$d_2 = \frac{KA\,p}{1 - \dfrac{KAE^2}{4\pi D^3}}$$

This displacement increases the capacitance of the condenser, since the plates are nearer together. The capacitance changes from

$$C = \frac{A}{4\pi D}$$

to

$$C + C_0 = \frac{A}{4\pi (D - d_2)} = \frac{A}{4\pi D\left(1 - \dfrac{d_2}{D}\right)}$$

$$= C\left[1 + \frac{d_2}{D} + \left(\frac{d_2}{D}\right)^2 + \ldots\ldots\right]$$

Or, approximating :

$$C_0 = C\frac{d_2}{D}$$

The change in capacitance causes a change in electrical charge from $Q = CE$ to

$$Q + q = (C + C_o)\,E$$

The change in charge is then :

$$q = C_o\,E$$

Substituting from the preceding formulas :

$$q = \frac{CE}{D} d_2$$

$$q = \frac{KACE}{D\left(1 - \dfrac{KAE^2}{4\pi D^3}\right)} p$$

Or, using $A = 4\pi DC$

$$q = \frac{4\pi KC^2E}{1 - \dfrac{KCE^2}{D^2}} p \quad \text{e. s. u.}$$

3.3 *Reciprocal relations in terms of response as microphone*

The two previous sections ended with the following relations

$$v = \frac{4\pi KC^2 E}{1 - \dfrac{KCE^2}{D^2}} e$$

$$q = \frac{4\pi KC^2 E}{1 - \dfrac{KCE^2}{D^2}} p$$

The coefficients are identical. Therefore the quantities between which reciprocity exists for a condenser microphone are -

As source,

e as cause; v as effect;

As absorber,

p as cause; q as effect;

Or more generally :

If, as absorber

$$q = tp$$

Then, as source

$$v = te$$

where the constant, t, is the same in both cases.

The next step is to restate the reciprocal relations in terms of the response of the condenser microphone when it is used as a microphone. This amounts to finding the value of t in terms of response. In this form it will be found that the compliance, K, which might be awkward to handle directly, disappears into a redefinition of the capacitance of the condenser. The latter becomes, as would be expected intuitively, the dynamic capacitance.

The response of a condenser microphone is defined as

$$p = \frac{\text{open circuit change in potential difference}}{\text{applied change in pressure}}$$

The concept of « open circuit » potential difference also requires definition. It is assumed that the polarizing battery is applied through a resistance so high that it may be considered infinite. Under such circumstances no change can flow. When an applied acoustic pressure causes the diaphragm to move so that the capacitance increases from C to $C + C_o'$, the potential difference must decrease from E to $E = e'$, to be consistent with the constant charge, Q. The amount of the change, e', is defined as the « open circuit » potential difference caused by the sound wave. Before the sound wave is applied :

$$E = \frac{Q}{C}$$

After it is applied :

$$E - e' = \frac{Q}{C + C_o'} = \frac{EC}{C + C_o'}$$
$$= E\,(1 - C_o'/C + \ldots\ldots.)$$

Approximating,

$$e' = \frac{EC_o'}{C}$$

The response is therefore

$$r = \frac{e'}{p} = \frac{EC_o'}{pC}$$

The value of C_o' is not quite the same as C_o, which is the change in capacitance when the instrument is used as a microphone, but the relation between them can be found. Let the displacement of the diaphragm under the conditions of definition "open circuit" potential difference be d_3 centimeters. C_o' is related to this displacement by

$$C + C_o' = \frac{A}{4\pi\,(D - d_3)} = \frac{C}{1 - \dfrac{d_3}{D}}$$

Or, approximating as before :

$$C_o' = C\,\frac{d_3}{D}$$

In this case the displacement is caused solely by the acoustic pressure p, since the charge remains constant, the potential decreases and there is no aiding increase from the electrostatic force. That is,

$$d_3 = KA\,p$$

When the instrument is used as a microphone the displacement is larger, since it is aided by the increase in electrostatic force. The displacement, d_2, was found in Section 3.2 to be :

$$d_2 = \frac{KA\,p}{1 - \dfrac{KAE^2}{4\pi\,D^3}}$$

Hence :

$$d_3 = \left(1 - \frac{KAE^2}{4\pi\,D^3}\right) d_2$$

The corresponding change in capacitance, when used as a microphone was, from Section 3.2 :

$$C_o = C\frac{d_2}{D}$$

Hence :

$$C_o' = \frac{d_3}{d_2}\,C_o$$

The open circuit response can therefore be written as :

$$p = \frac{EC_o'}{p\,C} = \frac{EC_o}{p\,C} \times \frac{d_3}{d_2}$$

But when the instrument is used as a microphone the charge which flows in response to the pressure p is :

$$q = EC_o$$

Thus the open circuit response can be stated in terms of the charge which flows when used as a microphone :

$$p = \frac{q}{p\,C}\,\frac{d_3}{d_2}$$

Rearranging terms, and inserting the value of d_3/d_2 from above :

$$q = \frac{r\,C\,p}{1 - \dfrac{KAE^2}{4\pi\,D^3}}$$

Or, using $A = 4\ \pi\ DC$, as before,

$$q = \frac{p\,C\,p}{1 - \dfrac{KCE^2}{D^2}}$$

The expression in the denominator arose from including in the foregoing discussion the increment of electrostatic attraction which resulted from the movement of the diaphragm. It is convenient accordingly to redefine the capacitance of the microphone as

$$C' = \frac{C}{1 - (KCE^2/D^2)}$$

In these terms

$$q = r\,C'\,p \qquad \text{e. s. u.}$$

Having this relation between q and p, for the instrument as a microphone, the reciprocity relation immediately gives the following formula when the instrument is used as a source of sound :

$$v = p\,C'e \quad \text{cm}^3$$

Up to this point the derivations have been given in terms of slow changes in otherwise static conditions. The considerations are compatible, however, with derivation on a dynamic basis, and henceforth the quantities p, e, v, etc., can be considered as complex quantities $p\ e^{j\omega t}$, involving frequency. The quantity K, being an expression of the mechanical properties of the diaphragm, including its stiffness and damping, will depend on frequency, and therefore the dynamic capacitance C' will indicate the value measured at the frequency considered.

We may now proceed to complete the steps in the calibration.

4. Pressure produced in a closed coupler

Referring to Figure 2, attached, consider that a microphone is terminated by a closed chamber, having cubic contents V and containing gas at a static, pressure B, usually atmospheric pressure. The application of a small additional potential difference causes a volume change of v cubic centimeters, where v is given above.

The volume change causes a pressure change, p, in the coupler. The volumes and pressures are related by the gas laws as follows :

$$(B + p)\,(V + v)^\gamma = BV^\gamma$$

where γ is the ratio of the specific heat at constant pressure to the specific heat at constant volume. In the case of air, $\gamma = 1.41$. The form of the relation assumes the volume change to take place adiabatically, without gain or loss of heat. From this relation :

$$p = \frac{BV^{\gamma}}{(V+v)^{\gamma}} - B = B\left[\frac{1}{\left(1+\frac{v}{V}\right)^{\gamma}} - 1\right]$$

$$= B\left[1 - \gamma\frac{v}{V} + \text{terms in } \left(\frac{v}{V}\right)^2 \text{ and higher powers } \left(\frac{v}{V}\right) - 1\right]$$

Neglecting $\frac{v^2}{V^2}$ and higher powers $p = -\frac{\gamma B}{V} v$. And since $v = pC' e$

$$p = -\frac{\gamma B p C'}{V} e$$

where p is the response as a microphone.

5. Product of two responses

The preceding result permits the determination of the product of the responses of two microphones. Referring to Figure 3, attached, consider that microphone I is used as a source of sound and is coupled to microphone II by a chamber of cubic contents V. An additional potential difference e'_1, applied to I causes a pressure p in the coupler. This causes an open circuit potential difference, e'_2 at the terminals of II. By the definition of response $e'_2 = r_2 p$.

But

$$p = -\frac{\gamma B p_1 C'_1}{V} e'_1$$

So that

$$e_2' = -\frac{\gamma BC'_1}{V} p_1 p_2 e'_1 \text{ c.g.s. units}$$

or

$$p_1 p_2 = -\frac{V}{\gamma BC'_1} \frac{e_2'}{e'_1} \text{ c.g.s. units}$$

6. Quotient of two responses

Referring to Figure 4, attached, the same source, III, is used to drive first microphone I and then microphone II, using the same coupler and the same applied e.m.f., e_3. The quotient of the responses is given by the quotient of the outputs

$$\frac{p_1}{p_2} = \frac{e_1}{e_2}$$

7. Responses in C.G.S. units

The individual responses are now derived from the product and quotient, giving

$$\left.\begin{aligned} p_1 &= \sqrt{-\frac{V}{\gamma\, BC'_1} \times \frac{e'_2}{e'_1} \times \frac{e_1}{e_2}} \\ p_2 &= \sqrt{-\frac{V}{\gamma\, BC'_1} \times \frac{e'_2}{e'_1} \times \frac{e_2}{e_1}} \end{aligned}\right\} \text{c.g.s. units}$$

(The negative sign is of interest only if the phases of the e.m.f.,s are measured).

8. Responses in usual units

The preceding formulas have been in C.G.S. units throughout. The customary units for expressing responses are hybrid, the electrical units being in the practical system and the mechanical units (length, force, etc.) in the C.G.S. system. The dimensions of a response on this basis are volts per dyne per square centimeter. If the number expressing a response in C.G.S. electrostatic units is p_0, then the number expressing the same response in hybrid units, p, is

$$p = \frac{c}{10^8}\, p_0$$

where c = ratio of electromagnetic to electrostatic unit of charge or $c \simeq 3.10^{10}$.

Likewise, if a number designating a capacitance in C.G.S. electrostatic units is C_o, then the number designating the same capacitance in farads, C, is

$$C = \frac{10^9}{c^2}\, C_o \ \text{farads}$$

The formulas in Section 7 then become, in hybrid units,

$$p_1 = \frac{c}{10^8} \sqrt{-\frac{V}{\gamma\, BC'_1} \times \frac{10^9}{c^2} \times \frac{e'_2}{e'_1} \times \frac{e_1}{e_2}}$$

or

$$p_1 = \sqrt{-\frac{10^{-7}\, V}{\gamma\, BC'_1} \times \frac{e'_2}{e'_1} \times \frac{e_1}{e_2}} \ \text{volts/dyne/cm}^2$$

and

$$p_2 = \sqrt{-\frac{10^{-7}\, V}{\gamma\, BC'_1} \times \frac{e'_2}{e'_1} \times \frac{e_2}{e_1}} \ \text{volts/dyne/cm}^2$$

To illustrate steps in reciprocity calibration of condenser microphone

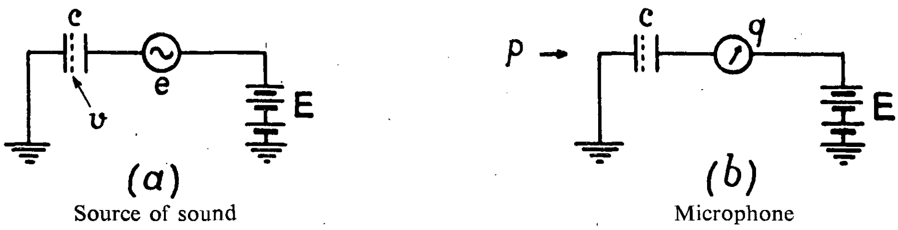

Figure 1. — *Reciprocity relations*

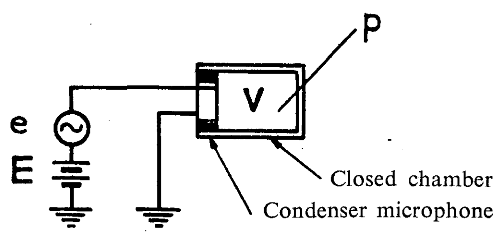

Figure 2. — *Pressure produced in closed chamber*

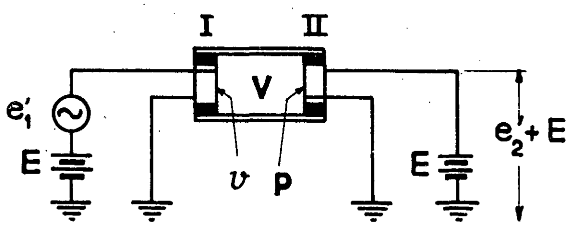

Figure 3. — *Product of two responses*

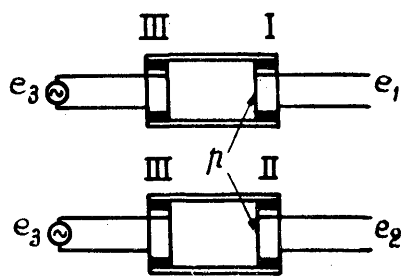

Figure 4. — *Quotient of two responses*

It may be noted also that the formula of Section 3.3 relating volume change to e.m.f. becomes in hybrid units

$$v = 10^7\, p\, C'\, e \quad \text{cm}^3$$

This function in this form is used as the starting point in the paper by R.K. Cook, referred to above.

9. Conclusion

The formulas for the responses of the microphones indicate that the accuracy of the results depends on measurement of ratios of voltages, the cubic contents of the coupler, a capacitance, the atmospheric or other gas pressure and knowledge of the gas constant, y. These can all be obtained with straight-forward methods. There are, however, other physical considerations, which are indicated to some extent by the various approximations made in the development of the formulas. In particular, the relative diaphragm motions, voltage changes, pressure changes, etc., must be so small that the systems can be considered linear. Also detailed physical reactions have been omitted, such as the fact that when connected by a coupler the diaphragms of both microphones move. These constitute limitations on the experimental conditions which are discussed in the paper referred to. In some situations, still other corrections are made which have not as yet been published. The practical working circumstances of most condenser microphones are, however, well-suited to the requirements of the method.

ANNEX 19

DISPLAY OF THE FREQUENCY CHARACTERISTIC OF A MICROPHONE WITH THE AID OF AN AUDIOGRAPH

The frequency characteristic of a microphone can be displayed by means of an audiograph. This equipement comprises an oscillator and a recording device coupled together. For automatic recording the variable condenser which controls the frequency of the oscillator is coupled directly to the recording drum driven by the motor M_1. The output voltage of the oscillator is at the frequency indicated by the pen which draws the characteristic on the recording drum. (The oscillator is also provided with a manual control independent of the recording mechanism, so that it can be used as a separate instrument. On putting the switch, situated on the panel in front of the audiograph, in the " Recording " or " Non-recording " position, the recording drum can be made to operate or the manual control used.)

When the recording drum is used, the oscillator frequency varies continuously from 30 to 10,000 c/s. The output impedance of the oscillator can be set as desired at 50, 200, 500 or 5,000 ohms. The input impedance of the recorder is 10,000 megohms.

The principle of operation of the recorder (see Figure 1). — The input voltage to the recorder is applied, by means of a potentiometer having a range of 40 decibels, to an attenuator A_1 with a fixed impedance of 50,000 ohms and of which the attenuation can vary in steps of half a decibel over a range of 80 such steps. The output voltage of this attenuator is rectified and the resultant current is made equal to a constant direct current by means of a system of relays which controls the direction of rotation of a motor M_2 which moves the pen of the recorder and the same time varies the attenuation of A_1. If for example the applied input voltage increases, the current I_1 becomes greater than I_2 and the motor turns in such a direction that the recorder pen shows a higher value on the graduated paper carried by the recording drum and the attenuation of A_1 increases. I_1 then decreases until it becomes less than I_2 and the direction of

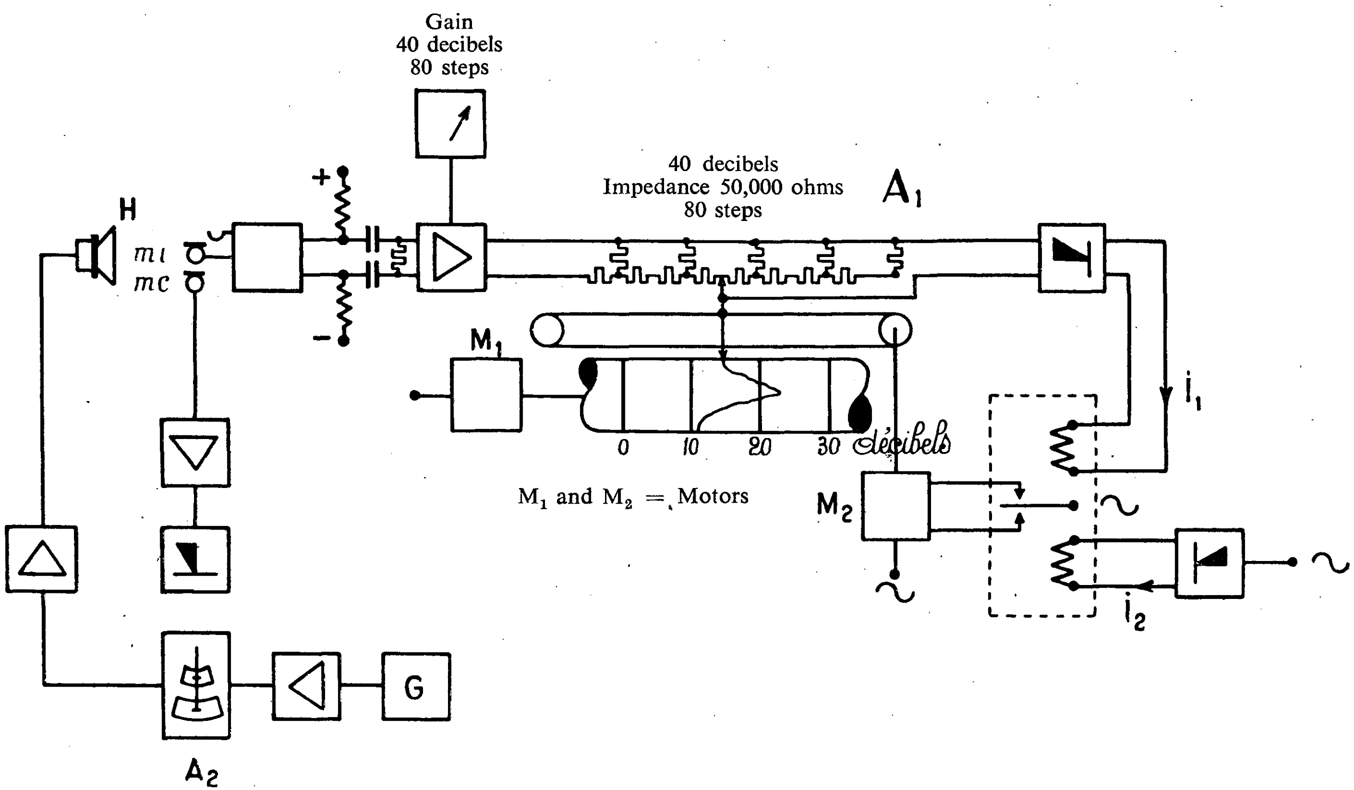

FIGURE 1. — *Schematic diagram of the audiograph*

rotation of the motor M_2 changes. In consequence the recording pen traces a zig-zag course of amplitude equal to half a decibel of which the mean curve is the desired characteristic.

Maintenance of a constant sound pressure. — The output voltage of the oscillator G is applied to the loudspeaker H by means of a voltage amplifier, a special attenuating device A_2 and a power amplifier. This loudspeaker is located in a silence cabinet, free of echo, with a standard microphone *mc* and the microphone under test *mi*. These two microphones are placed side by side at a point on the axis of symmetry of the loudspeaker, 60 cm. in front of the baffle. The standard microphone makes it possible to obtain a constant sound pressure in the following way.

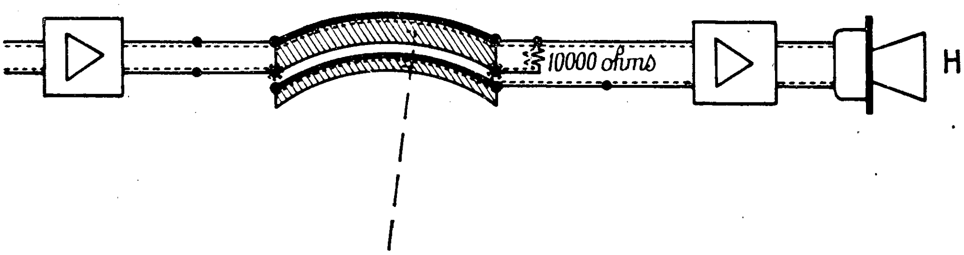

FIGURE 2. — *Details of the attenuating device* A_2 *which stabilizes the sound pressure produced by the loudspeaker*

The output voltage of the standard microphone is amplified and rectified and the resulting direct current fed to a milliammeter, which is part of the attenuating device A_2. The needle of this instrument is provided with a platinum arc which dips into two separate vessels filled with ordinary water (see Figure 2). The input and output terminals are connected so that the device functions as a high resistance potentiometer. A deflexion of the milliammeter reduces the output voltage. If, for example, the sound pressure increases, the milliammeter deflexion increases and the sound intensity produced by the loudspeaker in consequence diminishes. This stabilizing process produces a sound pressure from the loudspeaker, at the point of measurement, which depends upon the frequency characteristic of the standard microphone. A piezoelectric microphone practically free from attenuation distortion is used as a standard and so a perfectly constant sound pressure is obtained. A normal value for this sound pressure is 10 dynes per square centimeter at the point of measurement under free field conditions. The sound pressure stabilizing device described above operates satisfactorily in the frequency band 100 to 4,500 c/s.

ANNEX 20

NON-LINEAR DISTORTION OF CARBON MICROPHONES

(Contribution by the Telephone Administration of the Federal German Republic)

The Federal German Administration has in the past undertaken detailed studies on the question of non-linear distortion of telephone microphones *. As a result of these studies it was found that two factors give rise to non-linear distortion.

1. The stiffness of a carbon granule depends on the amplitude of vibration and
2. The contact resistance of carbon granules depends upon the contact pressure.

The variation of stiffness as a function of the amplitude of vibration is an important factor determining the non-linear effect of carbon microphones. It gives rise to harmonics and in some cases, to sub-harmonics. The main harmonic appears very strongly at a half to a third of the resonant frequency. When the fundamental frequency reaches a certain value it is also possible for sub-harmonics to appear and this effect is particularly marked if the frequency is equal to about twice the resonant frequency.

Furthermore the variation of stiffness as a function of amplitude results, as the amplitude increases, in a large reduction at this frequency and a change in sensitivity of the microphone, i.e. the sensitivity of the microphone first increases with the acoustic pressure and then decreases as the acoustic pressure becomes still higher.

When the non-linear distortion coefficient of a microphone is measured as a function of displacement of the diaphragm, it is found that this is approximately the same at different frequencies if large amplitudes of displacement are assumed

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* K. Braun, *T.F.T.*, Vol. 27 (1938), pp. 395-404. — K. Braun, *T.F.T.*, Vol. 28 (1939), pp. 115-120.

for the diaphragm. This shows that the non-linear effect of the microphone is essentially determined by the diaphragm displacement.

If two tones are applied simultaneously to the microphone instead of a single tone, it produces not only harmonics but also a spectrum of sum and difference tones. The amplitudes of the difference and sum tones depend upon the interval between the frequencies of the two applied tones. The first difference tone appears most strongly when the difference in frequency between the two exciting tones is about equal to the resonant frequency.

In addition to the appearance of this spectrum, there occurs also a variation in sensitivity for the two tones as a function of the amplitudes of the two fundamental components, i.e. the sensitivity for one tone changes when a second tone is applied.

Non-linear distortion can be neglected in a good quality telephone receiver in comparison with that of a microphone. For this reason attention can be confined to the non-linear distortion characteristics of the microphone only.

Meanwhile the Telephone Administration of the German Federal Republic has continued its study of carbon microphones. From these studies it is found that, when tones are applied to a microphone, noise voltages are generated in addition to the harmonics and difference tones. These additional voltages can be observed with a cathode ray oscilloscope if the response curve of a microphone excited by pure tones is presented. With unstable microphones, the response curve does not remain steady; on the contrary it appears in a variable manner and floats about. If one listens at the same time, it is apparent that the instability has a very degrading effect upon the quality of the microphone. An arrangement consisting of a cathode ray oscillograph is not very convenient for obtaining numerical results.

The method of measuring modulation products indicated by K. Braun lends itself to this purpose. A tone of frequency varying from 200 to 4,000 c/s is applied to the microphone capsule to replace the speech spectrum and a measurement is made, at the output of a high-pass filter cutting off at 5,000 c/s, of the non-linear components and noise voltages which appear outside the frequency range of the variable frequency tone and additional excitation. By this method using an appropriate voltmeter, e.g. a psophometer, the instability of the carbon microphone which often appears in a spontaneous manner can easily be observed. This method is particularly convenient when a large number of microphones have to be measured.

Measurements show that, in general, the modulation products become greater as the sensitivity of the carbon microphone increases. To obtain a ratio of the alternating voltage from the microphone to the modulation product voltage, it is preferable to show the modulation product coefficient which is the difference between the logarithmic values of these two voltages. This quantity can vary between about 17 and 34 decibels, according to the quality of manufacture and the amount of use. It the sensitivity of the microphone does not vary too much, say within the permitted tolerances of sending reference equivalent, it is in general sufficient to determine a maximum value for the modulation product voltage. From the results of measurements verified by subjective tests a value of about 20 mV can be allowed for the modulation product voltage. This value is arrived at by measuring the voltage at the output of a high-pass filter cutting off at 5,000 c/s.

The Telephone Administration of the German Federal Republic can, based on these studies, give the following answers :

(a) Non-linear distortion, by itself, is not a satisfactory indication of instability of telephone microphone characteristics because it does not take account of the noise in service of the microphone. To determine the complete distortion, i.e. the non-linear distortion and the instability distortion, it is recommended that the modulation product voltage should be measured.

(b) The modulation product voltage can be measured while a tone is applied varying between 200 and 2,400 c/s, to replace the speech spectrum, by measuring part of the additional waves produced, through an appropriate filter, for example high-pass whose cut-off frequency is 5,000 c/s.

(c) Modulation product voltages up to about 20 mV do not seriously affect the transmission quality of a carbon microphone.

APPENDIX

Results of tests conducted by the Telephone Administration of the Federal German Republic on non-linear distortion in subscriber's telephone equipment

The method employed in these tests is represented schematically in Figure 1.

A sound of frequency variable between 200 and 4,000 c/s is applied to the microphone capsule and the voltage due to distortion products, which appear outside the band of frequencies of the variable frequency sound, is measured at the output of a high-pass filter of cut-off frequency 500 c/s (see Figure 2). Figure 3 shows the variation of the voltage due to modulation products as a function of the number of conversations for two different types of microphone. For comparison this figure also shows for one type of microphone the noise voltage in the quiet condition, which is much smaller than the voltage due to modulation products. Figure 4 shows the statistical distribution of voltage due to modulation products for almost new microphones and for microphones which have been used a great deal.

The investigation has been broadened to enable the noise while in use to be measured in the form of a voltage due to modulation products in the band of frequencies transmitted by the microphone as a function of the excitation frequency. For this purpose, the microphone, without mouthpiece, was excited by a sinusoidal sound of about 11 dynes par square centimetre. To eliminate the exciting frequency f and the harmonics produced in the microphone, a bandpass filter has been inserted at the output of the feeding bridge of the telephone equipment, the band of the filter being such as to transmit only those components of the noise voltage between f and $2f$. This voltage was measured as distortion and referred to a bandwidth of 400 c/s. Figure 5 shows the results obtained for two different microphones. For comparison the results obtained with a bad microphone (Microphone Capsule 1) have also been included. The full curve shows the "sensitivity-frequency" characteristic of the microphone; this shows the useful voltage produced when the microphone is excited with a constant acoustical pressure independent of frequency. The dotted curve shows the voltage due to modulation produced by a sinusoidal sound. It has fundamentally the same shape as that of the " sensitivity-frequency " characteristic but the peaks and troughs are much greater. Near the frequency of resonance the ratio of useful voltage to voltage of modulation products produced by a sinusoidal sound can be very small if the microphone is a bad one.. Noise can then represent a serious degradation. For this reason very marked resonances when carbon microphones are used can have an adverse effect not only upon the " sensitivity-frequency " characteristic but also on that of the noise voltage.

For comparison the noise under quiet conditions has been measured in the whole frequency band of the instrument used for measurement of distortion voltage. The noise voltage under quiet conditions varies, according to the quality of the carbon microphone, between a lower and an upper value, and so the two values have been recorded : the ratio of these two values becomes smaller as the microphone is more stable. Furthermore the

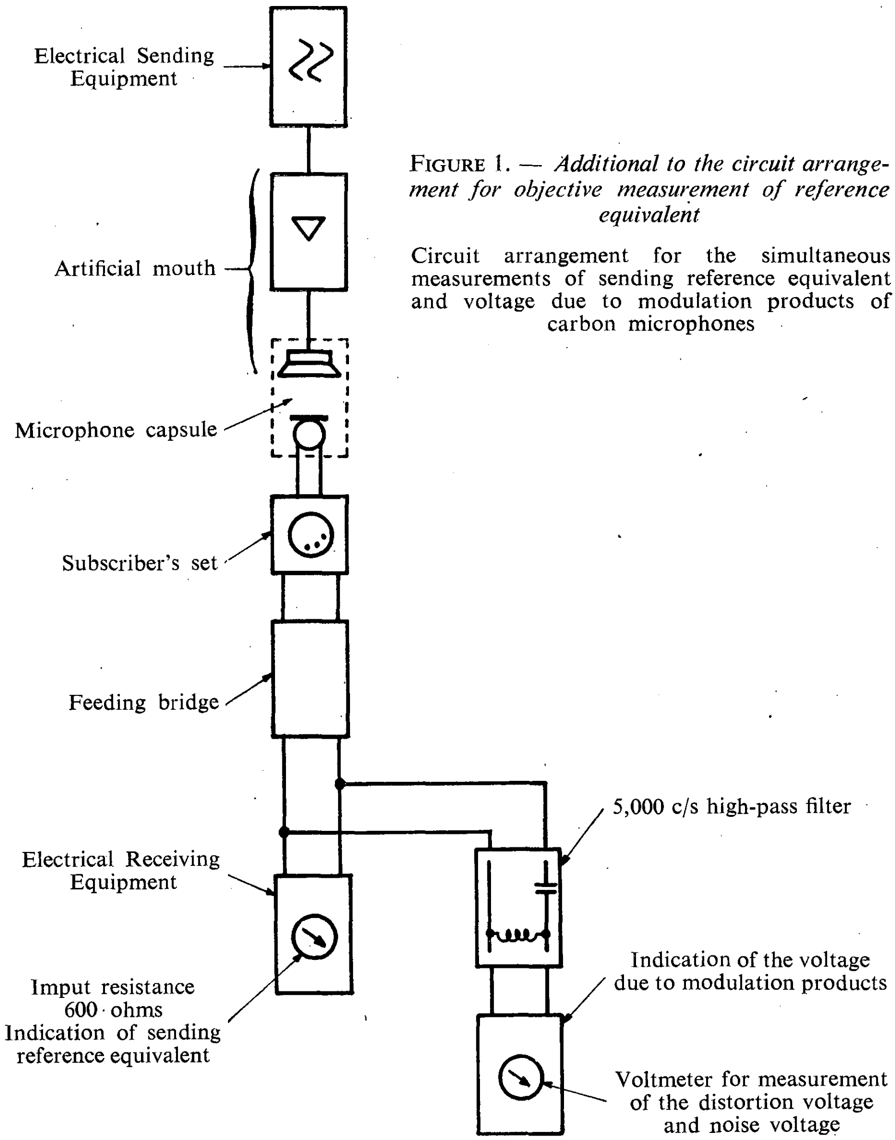


FIGURE 1. — *Additional to the circuit arrangement for objective measurement of reference equivalent*

Circuit arrangement for the simultaneous measurements of sending reference equivalent and voltage due to modulation products of carbon microphones voltage above 5,000 c/s due to modulation products has also been recorded, measured with a sound of frequency variable between 200 and 4,000 c/s. For the two microphones, the ratio of the maximum noise voltage under quiet conditions and the voltage due to modulation products is about 20 decibels. The voltage due to modulation products can therefore be used as a criterion of noise voltage under quiet conditions.

The components of the noise voltage under working conditions which fall in the band of frequencies transmitted by the microphone is a controlling factor in loss of articulation. To enable this part to be determined the components of the voltages due to modulation in the band of frequencies 300 to 3,400 c/s and in the band of frequencies above 5,000 c/s have been measured for the maximum noise voltage under quiet conditions, which is approximately equal to the voltage due to modulation products. The ratio of these two voltages depends upon the shape of the noise spectrum. The more unstable the microphone the more the lower frequencies are represented. In general the voltage due to modulation products for the frequency band 300 to 3,400 c/s is about 1.3 to 4 times greater than the voltage due to modulation products above 5,000 c/s, but it can have still more unfavourable values.

Nevertheless, when a sound is applied to a microphone, the ratio of useful voltage to voltage due to modulation products above 5,000 c/s is of the order 17 to 34 decibels and so the ratio of speech to noise voltage under working conditions can, in the case of bad microphones, be less than 17 decibels which would considerably reduce the articulation. For carbon microphones which have been used a great deal, the loss of articulation can consequently be partly due also to the noise voltage under working conditions.

It is therefore important to determine the noise voltage under working conditions of carbon microphones and to set maximum values for it. The method using the voltage due to modulation products which is described above is particularly suitable for this purpose.

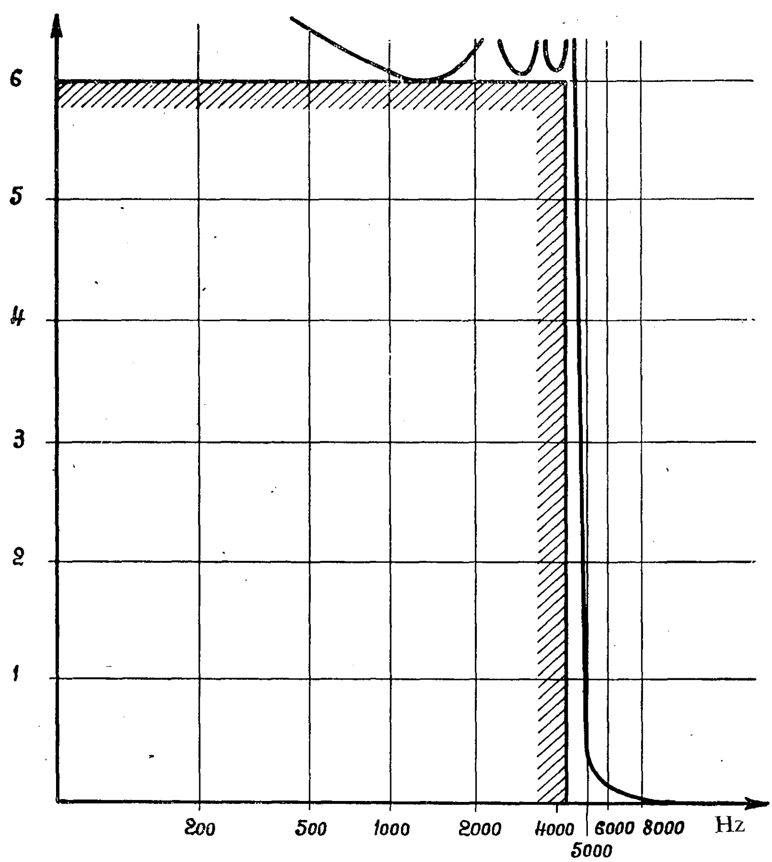


FIGURE 2. — *Composite attenuation of 5,000 c/s high-pass filter*

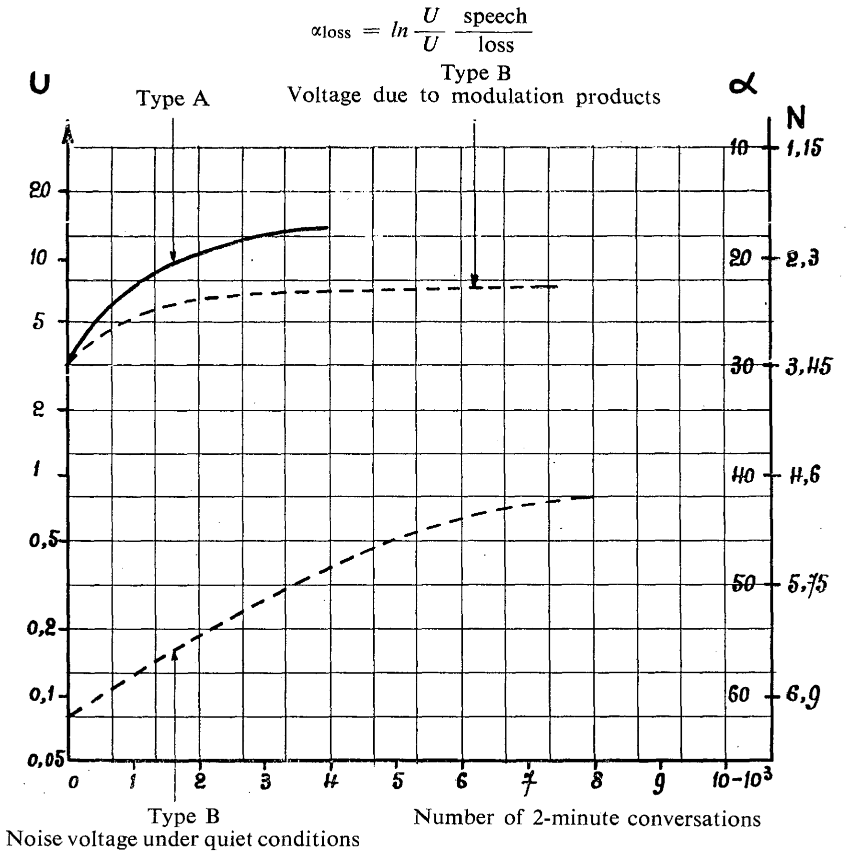


FIGURE 3. — *Voltage due to modulation products and noise voltage under quiet conditions*

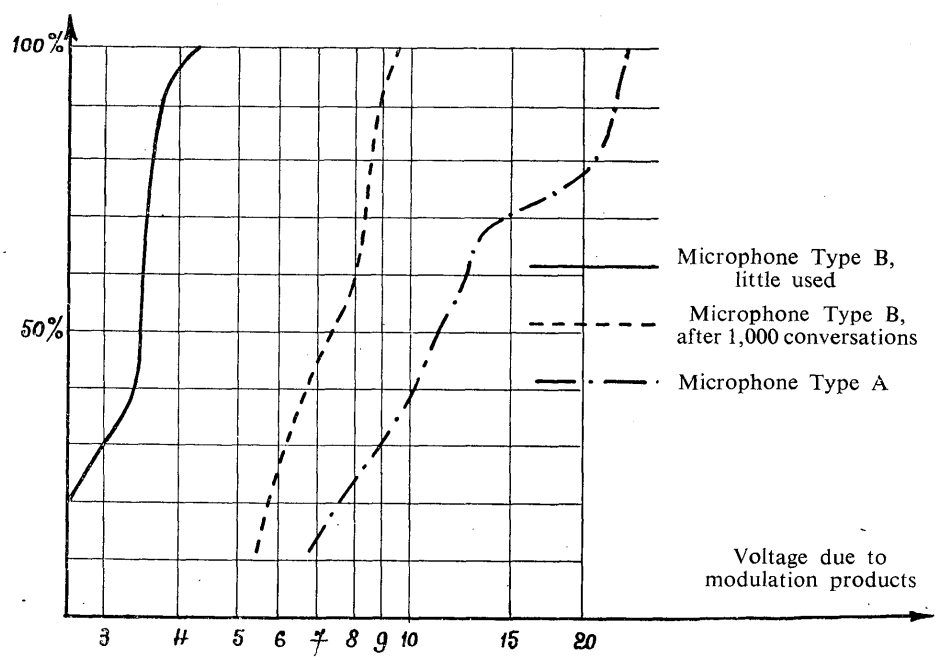
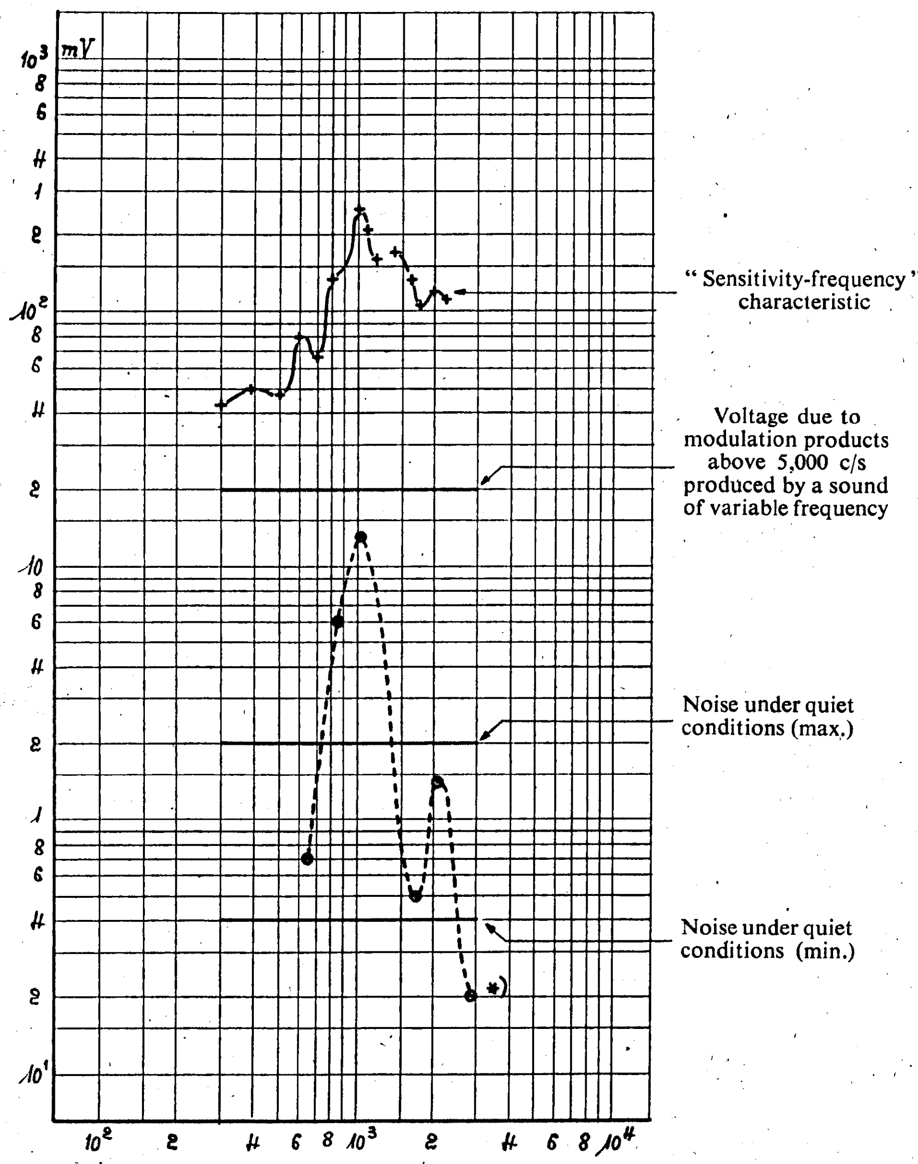
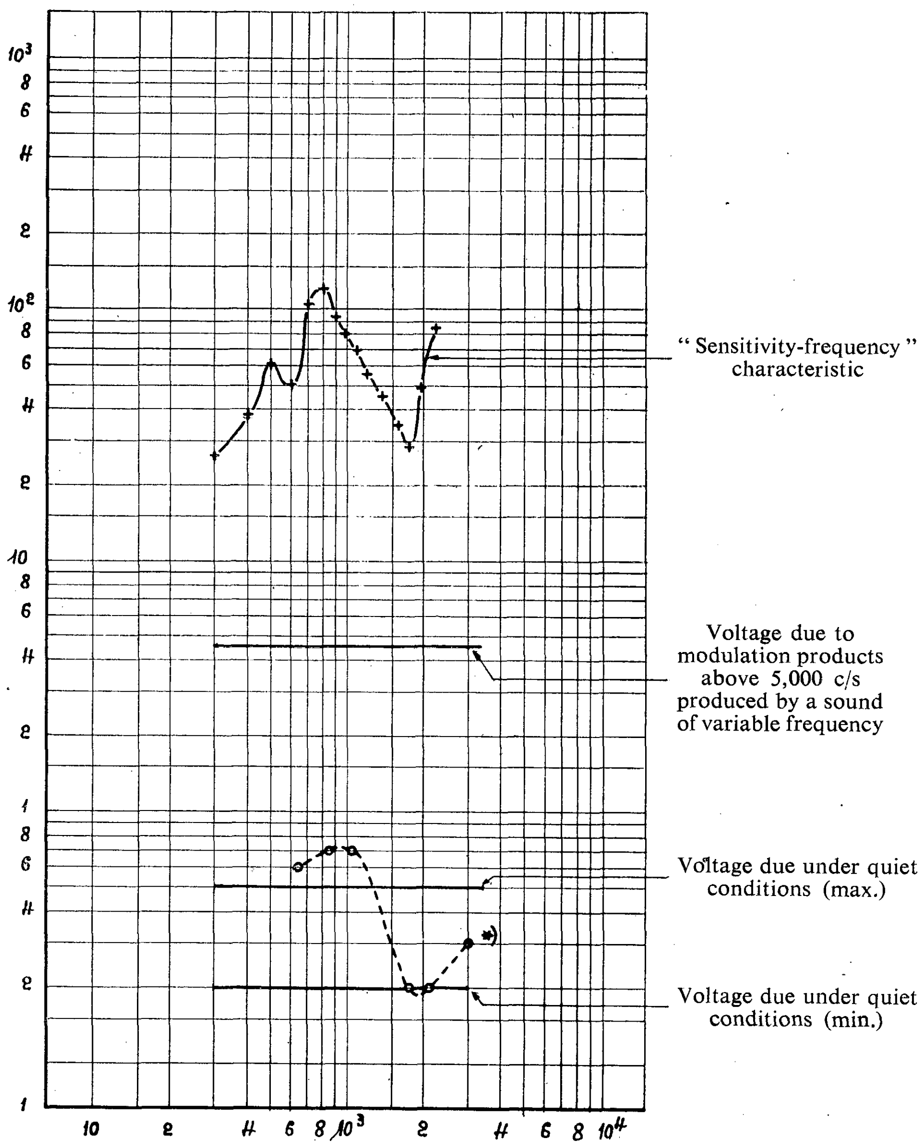


FIGURE 4. — *Statistical distributions curve of the voltage due to modulations products of telephone microphones, measured at the output of the equipment*



* Tenfold increase of the voltage due to modulation products produced by a sinusoidal sound, referred to a bandwidth of 400 c/s between the sound and its octave.

FIGURE 5 *a.* — *Microphone capsule 1*



* Tenfold increase of the voltage due to modulation products produced by a sinusoidal sound, referred to a bandwidth of 400 c/s between the sound and its octave.

FIGURE 5 *b.* — *Microphone capsule 2*

ANNEX 21

APPARATUS FOR THE OBJECTIVE MEASUREMENT OF REFERENCE EQUIVALENTS USED BY THE TELEPHONE ADMINISTRATION OF THE FEDERAL GERMAN REPUBLIC

This apparatus enables objective measurement of reference equivalents (1) to be made. For these measurements, the natural organs of speech and hearing are replaced by an artificial mouth and an artificial ear.

The method of objective measurement can be applied in a relatively simple manner, if warbling sinusoidal tone is used as an artificial voice, in place of a mixture of frequencies, i.e. instead of producing the components of the sound spectrum simultaneously, they are produced one after the other on the measuring transducer and on the indicating instrument.

Thus, the r.m.s. value relating to the speech pressure can be obtained by measuring successively the r.m.s. values for the various frequencies of the warble tone with the aid of an indicating instrument of long integration time. Replacing the human mouth and ear, an artificial mouth and ear are provided which faithfully reproduce the acoustical properties of the human organs of speech and hearing.

The apparatus for the objective measurement of reference equivalents (2) is composed of four fundamental parts :

1. The electrical source, used as a speech generator,
2. The electrical receiving system, giving on r.m.s. indication,
3. The artificial mouth, as an acoustical source,
4. The artificial ear, for acoustical reception.

1. *The electrical source* produces a warble-tone which, in one second, sweeps the frequencies from 200 to 4,000 c/s, to and fro. The output voltage at the terminals is 285 mV. independent of frequency, with an internal resistance of 600 ohms which corresponds to an absolute voltage level of-8.6 db, i.e., to the mean speech voltage at the output of the telephone transmission reference system. To check the " amplitude-amplitude " distortion of a transducer, eg. a carbon

microphone, under test, the output voltage can be varied ± 8.6 db. The variation in amplitude of the warble tone, as a function of time, corresponds to the sound spectrum of the human voice.

2. *The electrical receiving system* evaluates each voltage element associated with a frequency component, according to the r.m.s. law. The reading is made directly in nepers, on the scale of the receiving instrument. The range of measurament can be changed in steps of 1 neper (8.6 db) by inserting or removing fixed attenuations.

3. *The artificial mouth* consists of a very stable loudspeaker with a small cone. At a distance of about 1 cm from the cone is a ring, representing the lips from which point the variation (as a function of distance) of the radiated sound pressure corresponds approximately to that of the human mouth.

The power which drives the loudspeaker is supplied by a special amplifier, which also incorporates the equalizers. The sensitivity-frequency characteristic is flat within about ± 0.35 nepers (± 3 db) from 200 to 4,000 c/s, when using the microphone of the telephone transmission reference system to determine this characteristic. The mean sensitivity of the artificial mouth, with amplifier, is 37.6 dynes/cm^2 per volt, so that, for the normal sending voltage of 285 mV. applied to the 600 ohm input impedance of th amplifier, a sound pressure of the order of 11 dynes/cm^2 (10.75 exactly) appears at the diaphragm of the microphone of the telephone transmission reference system.

To enable the exact position of a handset mouthpiece to be adjusted relative to the artificial mouth, the earpiece is positioned according to dimensions prescribed by the C.C.I.F., for a normal head. It is possible in position A, to adjust the spacing for reference equivalents to American head, correspond to the dimensions of an average and, in position E, the spacing corresponding to the dimensions of an average European head for A.R.A.E.N. values may be measured (see C.C.I.F. *Green book*, Volume IV, pages 146-161).

For the rapid checking of microphone capsules a quick changing device is provided to take the place of the mouth piece on the handset.

4. *The artificial ear* consists of a cavity representing the external human ear, a measuring condenser microphone and a microphone amplifier. The cavity represents the input impedance of the human ear when a telephone receiver is applied to it. In order that this condition may be fulfilled for earcaps of different types, the surface of the artificial ear must be shaped in an appropriate manner. For frequencies up to about 1,000 c/s it is sufficient to fix the acoustical resistance by the volume of the pressure chamber. It has been established that a volume in the region of 3 cm^3 for the artificial ear corresponds well with that of the human ear. At higher frequencies, the shape and the acoustical attenuation of the pressure chamber also plays a part in representing the acoustical resistance of the ear.

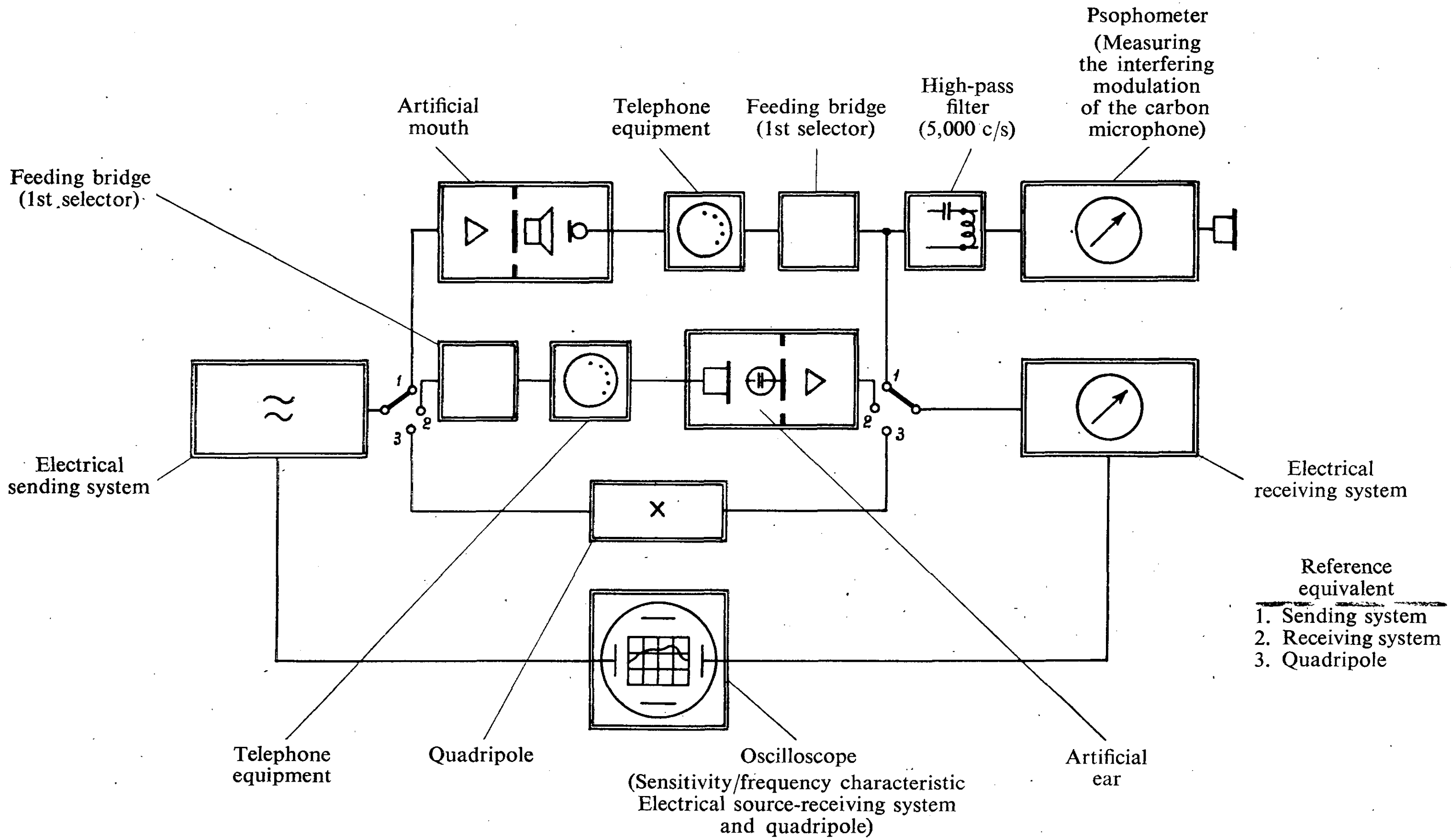
The contact surface is conical to ensure a perfect fit for the different earcaps of telephone receivers. The cavity is made of a cylindrical tube at the end of which the sound pressure is measured with a small condenser microphone, corresponding to the ear drum. To represent the losses, 16 attenuation channels, of different lengths, are situated at the end. The sensitivity of the condenser microphone with its microphone amplifier is 26.6 millivolts per dyne/cm^2, when the amplifier is terminated with a resistance of 600 ohms. It is practically independent of frequency in the range 200 to 4,000 c/s.

To test receiver capsules, a special device is used, which includes the normal earcap as well as the cavity. The complete holder of the measuring microphone can be easily withdrawn and replaced, for checking the artificial mouth, by the normal cavity or mouthpiece of the telephone transmission reference system.

To check microphone and receiver capsules, a central control panel is provided, which represents the average circuits of the subscribers' sets and telephone exchanges, and connects, in a control panel, the four basic pieces of apparatus vith the exchange circuits and subscriber's equipment and the switch for making the various measurements and calibrations. The supply voltages necessary, between 24 and 60 V, are provided by a power unit via a selector. The supply voltage and feeding current are indicated by an instrument on the control panel. These readings enable the resistance of the carbon microphone to be determined. The stability of receiver capsules can be checked by a reversal of polarity of the supply voltage. In this case the diaphragm receives a magnetic schok.

Carbon microphones are not stable and generate considerable non-linear distortion and microphonic noise which impairs the transmission quality. The importance of these disturbances depends on the construction of the microphone, on the quality of the carbon granules, and on the quality of manufacture. In consequence, care must be taken to keep these unstable distortions and microphone noise within admissable limits. For this purpose, an indication of the " modulation product voltage " has been introduced. By this method the frequencies above 5,000 c/s produced by the warble-tone, are shown, at the same time as the reference equivalent is measured by a suitable psophometric voltmeter, after passing through a high-pass filter which eliminates the warble-tone. The voltmeter reading is then a measure of the distortion and the microphone noise.

Furthermore, it is also possible to connect an *oscilloscope* showing the "sensitivity-frequency" characteristic of the microphone or receiver. These supplementary measurements can be made at the same as the objective measurement of the reference equivalent. In this manner, it is possible to check all the factors which can reduce the articulation and so make the A.E.N. worse. However, the principal point remains the objective measurement of the reference equivalent. In comparison with the subjective measurement of reference equivalents, the duration of the measurement is reduced to a few seconds, and the inaccuracy of the measurement to ± 0.1 nepers (± 1 db). The meter readings therefore, are in close agreement with the values obtained by subjective measurement. The following schematic diagram shows the arrangement of the apparatus for the objective measurement of reference equivalents.



Schematic diagram of apparatus for the objective measurement of reference equivalents

Method of measurement

To measure the sending reference equivalent, the two switches shown in the diagram are set to Position 1. In this position the electrical source is coupled to the artificial mouth. The telephone microphone is modulated by the artificial voice and, by means of the electrical receiving system, the voltage produced at the output of the feeding bridge is measured, this giving the sending reference equivalent. At the same time, the voltage of the modulation products is measured at the output of the 5,000 c/s high-pass filter, by means of the psophometer.

To measure the receiving reference equivalent, the two switches on the diagram are set to Position 2.

The receiver of the telephone equipment is excited electrically by the feeding bridge and the sound pressure thus produced in the artificial ear is measured with the aid of the measuring condenser microphone and the electrical receiving system, this giving the receiving reference equivalent.

With the two switches in Position 3, the reference equivalent of a quadripole may be measured. Obviously, it is also possible to measure the reference equivalent of a complete connexion by modulating the telephone microphone at the beginning of the connexion, by the artificial voice, and measuring the received sound pressure at the far end of the connexion, by means of the artificial ear. Nevertheless, such measurements are only to be considered for fundamental research.

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ANNEX 22

APPARATUS FOR THE OBJECTIVE MEASUREMENT OF REFERENCE EQUIVALENTS USED BY THE SWISS TELEPHONE ADMINISTRATION

The Swiss Telephone Administration determines reference equivalents only from objective measurements.

It can be shown that the human ear responds in a general manner to the various frequency components of speech according to square root law. This statement enables relatively simple measuring methods to be employed and these are described below without entering into any theoretical considerations.

1. Determination of reference equivalent from the response curve of the transmission system

The frequency characteristic of the transmission system is measured according to a method mentioned in Annex 10 of the Book of Annexes to Volume IV of the *Green Book* and presented as a graphical construction as shown in Figure 1. The amplitude axis of the graph takes account of the square root law, whilst the frequency axis emphasises by a corresponding expansion the frequency bands mainly occupied by speech. The frequency characteristic must be drawn according to an existing calibration system or according to a level fixed theoretically, in such a manner that measured values of reference equivalent are effectively related to those of the S.F.E.R.T. It is necessary to measure carbon microphones under the same conditions of acoustic pressure and position as those of the S.F.E.R.T.

2. Direct measurement of reference equivalent by means of the reference system

Direct measurement is made in a manner similar to the graphical method. The microphone and receiver capsules are inserted in a measuring circuit according to Figures 13 and 16 of Annex 10 of the Book of Annexes to Volume IV of the *Green Book*. The source is a heterodyne oscillator, whose output voltage is independent of frequency and which, driven by a motor, sweeps the frequency band 200 to 5,000 c/s to and fro about once per second. The rotary condenser plates have a special shape so as to obtain the frequency scale indicated in the abscissa axis of the graph.

The apparatus for measuring reference equivalent, which assesses the acoustic intensity in accordance with the ear, is connected in parallel with the oscilloscope (Figures 13 and 16 of Annex 10 of the Book of Annexes to Volume IV of the *Green Book*); The indicating instrument receives a current proportional to the square root of the input voltage of the equipment. The instrument has a time constant relatively large compared with the period of the warble tone and allows a direct reading of the reference equivalent on the scale graduated in nepers.

3. Large scale checking of carbon microphones and receivers

For routine checking of receivers and carbon microphones, a test equipment is available, specially designed for this purpose, but which permits only our normal capsules to be measured.

Figures 2 and 3 show respectively the schematic diagram and a sketch of the apparatus by means of which the following measurements can be made :

— Reference equivalent of the microphone and receiver capsules relating to our subscriber's sets (measured according to the principle described in Section 2 above),

— Check of the frequency characteristic, the variations of which must lie within the limits of a mask placed in front of the cathode ray tube,

— Check of the centreing of the armature of receiver capsules. For this the receiver is fed with a sinusoidal voltage at 500 c/s. Through the receiver superposed on this voltage is passed a direct current of which the polarity can be reversed. A subjective listening test shows if the distortion factor is small and if the centreing of the diaphragm actuating system is satisfactory,

— Measurement of the direct current resistance of carbon microphones,

— Measurement of the modulation products. The microphones are submitted to an acoustic field of 4,000 to 300 c/s. The modulation products are measured after a high-pass filter (cut-off frequency 5,000 c/s); it provides a measurement of spurious voltages produced under working conditions,

— Check of the condition of the carbon granules of microphones. The microphone is placed in a holding device provided with a cover which somewhat reduces the room noise reaching it. The capsule is fed normally but receives no sound. The holding device carrying the capsule is the rotated by hand. The resulting displacement of the carbon granules generates a noise voltage in the microphone which can light a neon lamp. When the capsule is rotated the lamp generally lights; but with a good microphone, it should go out as it comes to rest. This relatively simple test permits the elemination of microphones (particularly used microphones) which have a tendency to generate noise.

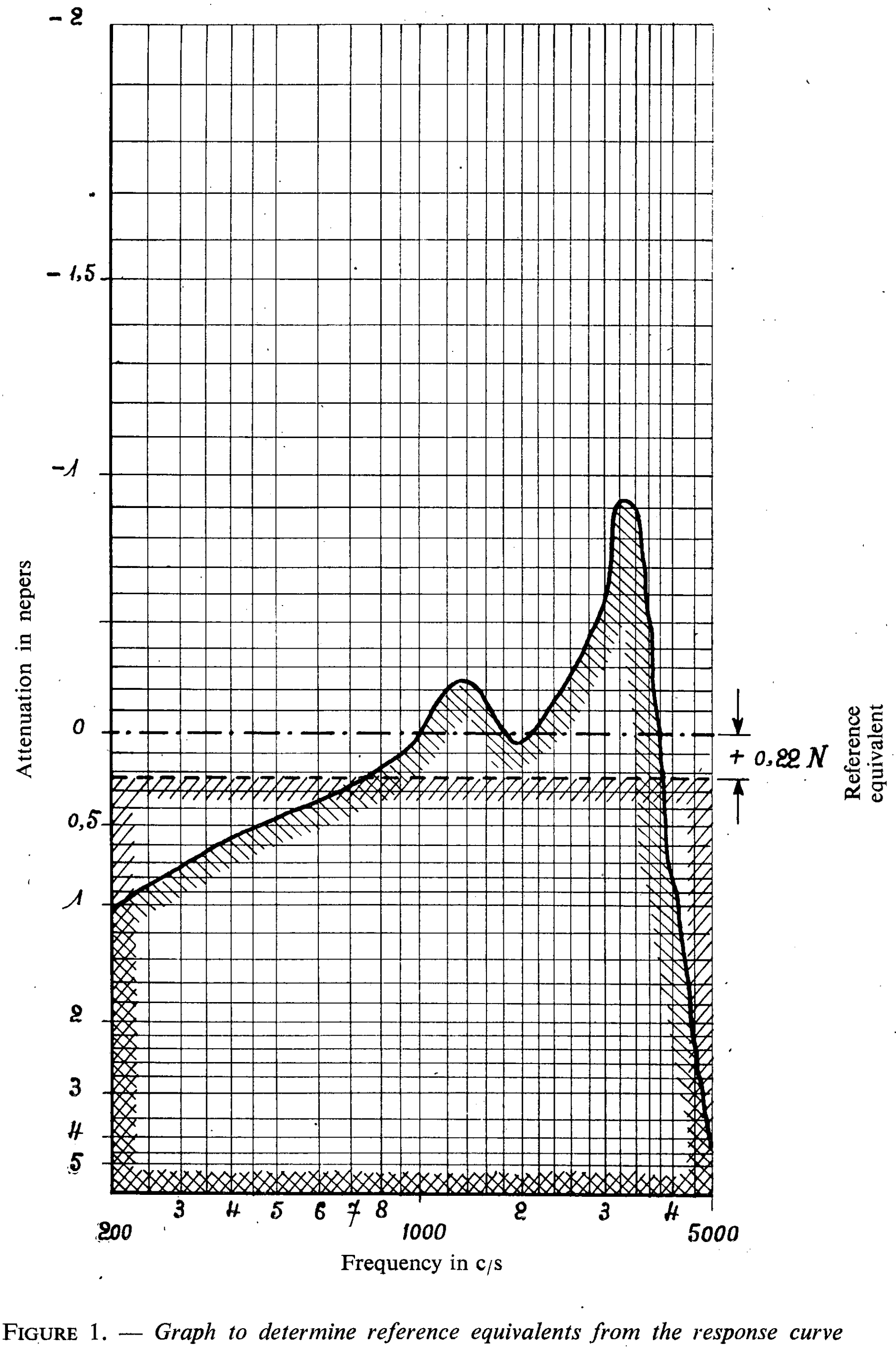

FIGURE 1. — *Graph to determine reference equivalents from the response curve*

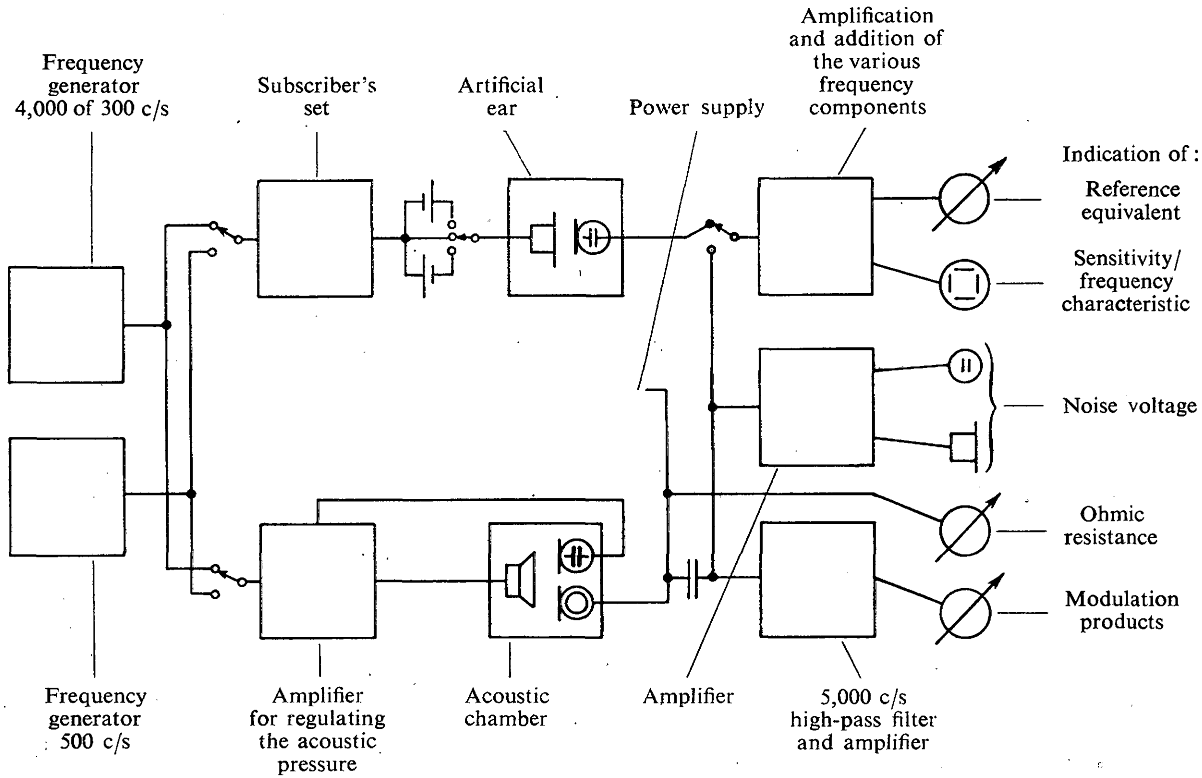

FIGURE 2. — *Schematic diagram of the apparatus for checking microphones and receivers used by the Swiss Administration*

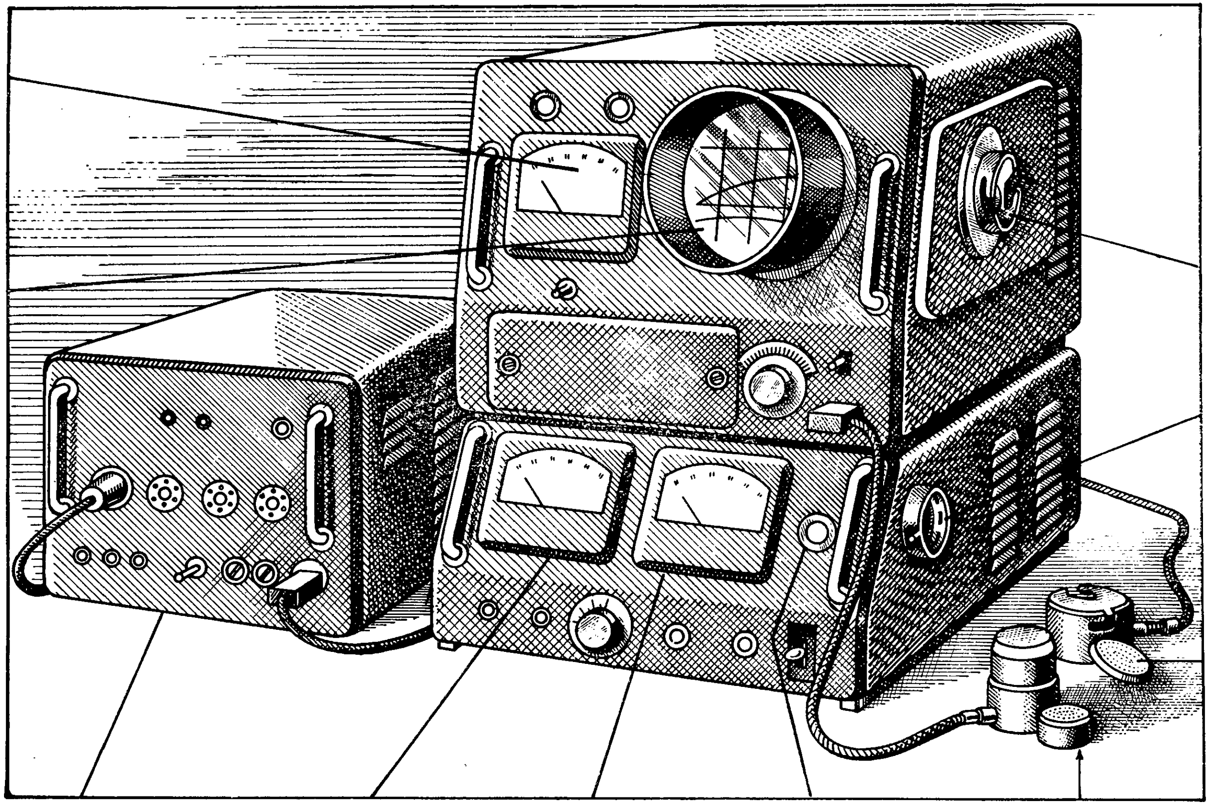

FIGURE 3. — *Apparatus to check microphone and receiver capsules used by the Swiss Administration*

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ANNEX 23

IMMEDIATE APPRECIATION TESTING METHOD

1. History of the " immediate appreciation " testing method

The " immediate appreciation " testing method was first described by W. H. Grinsted (1) * of Siemens Bros., Woolwich, England, in January, 1937. In the tests described a number of short, unrelated sentences read from a newspaper were transmitted over the test circuit, and listeners were asked to record the number of sentences " immediately appreciated ", i.e. believed to be understood without having to review the meaning of the sentence after reception.

The final result of the measurements was not a relative transmission performance rating as is now used, but a statistical table showing the percentage of telephone connexions in actual service which could be expected to have an immediate appreciation score better than a given value. Measurements were made with various values of room noise at the listening end and with varying values of attenuation in the junction line. Using a volume meter, speakers adjusted their speaking level to give, as nearly as possible, the " normal volume for telephonometric measurements " of the C.C.I.F., and therefore sidetone effects at the speaking end had no effect on the results of the test.

A further paper by J. R. Hughs (2) in November, 1938 described the testing technique in greater detain and introduced an improved method of statistical evaluation of the test results.

A further development of the technique was described by J. R. Hughs (3) in December, 1942. In the modified technique, the speaker's talking level was not controlled and room noise was introduced at the talking end. Measurements were made systematically with a number of combinations of values of speaking end and listening end room noise. The " natural speaking level " of each speaker taking part in the test was measured separately and allowed for. The result obtained

* The figures in brackets refer to the bibliography below.

from the measurements, after allowing for the statistical distribution, under normal telephone service conditions, of room noise levels and subscriber's talking level, is a graph of junction line attenuation versus percentage of total calls made in normal service which can be expected to enjoy an immediate appreciation score exceeding a given value.

2. Principle of the immediate appreciation testing method *

The principle on which the method is based can be set out as follows :

Transmission is satisfactory when the two subscribers can converse without misunderstanding or strain, i.e., when they can give the whole of their attention to the *matter* under discussion with practically no diversion to the *means* by which they are carrying on the discussion. The satisfactoriness of the transmission is measured by the degree to which this holds good.

During 1933 to 1936 experiments have been made in Siemens Brother's laboratories on a method of transmission testing which aims at such a measurement, and we have now worked out a technique which gives reasonably consistent results. This method is based upon fundamental ideas of transmission by telephone which, although perhaps not new, may need some explanation.

The object of a telephone connection is to enable two parties to exchange ideas by means of speech. The ideas start in the mind of the speaker and have to be appreciated in the mind of the listener. The quality of transmission is good or bad according to the ease with which this exchange of ideas from the *mind* of one party to the *mind* of the other is achieved. During the conversation the minds of both parties are occupied with the matter which they are discussing; all other mental processes which are concerned in the presentation of the sensations and ideas to the mind are, or should be, sub-conscious.

Fundamentally, therefore, the quality of the transmission should be judged by the smoothness and unobtrusiveness of the processes :

(a) Of converting the ideas of the speaker into words;

(b) Of converting what the listener hears into ideas and their presentation to his mind.

All questions of volume, attenuation, noise, articulation, intelligibility, sidetone and the like are subservient to this.

Anything which distracts the attention of the speaker, and so hinders him from putting his thought into words, affects the quality of transmission. So

* Extract from the article by W. H. Grinsted, cited as Reference (1) in the bibliography below.

does anything which tends to prevent him from using the intensity, the inflections and the accents which he would ordinarily employ in expressing his ideas.

Similarly, the freedom of the mind of the listener to deal with the matter which is being discussed is interfered with, if concious attention is necessary from time to time to understand the received speach, or if cross-talk or intermittent noises on the circuit distract his attention.

The process of formulating words and even whole phrases to express ideas on the part of the speaker is a process of habit and is normally carried out without conscious attention. It can be interfered with if some reaction such as heavy sidetone or echo attracts his attention, or if he finds he has consciously to raise his voice in order to make himself heard. In such cases he is conscious of the interference. It can also be interfered with sub-consciously if he hears his own voice as sidetone more loudly than he hears it in normal face-to-face conversation. It is part of the speech-habit to regulate the loudness of one's voice by the extent to which one can hear it above surrounding noises.

Similarly, with hearing, the process of the translation of sounds entering the ear into ideas presented to the mind is a process of habit. We do not consciously identify each speech sound from its characteristic frequency bands, piece them together into words and the words into phrases. Although, as Collard has shown, satisfactory quantitative results are given by theories based upon this process, it is not a conscious process but one of habit, in which normally we recognise a word or even a phrase all at once " in a flash ".

It is by the direct formation of these habits of speech and hearing that a baby learns its native language. It acquires the ability to form and to recognise words and phrases directly without deliberate synthesis. Everyone who has learned a foreign language is familiar with the difference between the conscious effort of the early stages and the fluency which comes only when the processes of speech and hearing in the new language have become habits.

The mind is capable of distributing attention to a limited extent; for example, we may be debating a subject while walking, yet steer clear of obstacles. If, however, the number or difficulty of the obstacles increases, attention may be completely distracted from the matter in hand. This distraction gives rise to a feeling of irritation, slight at first but tending to be cumulative. It seems that the attention can be successfully distributed only when the one object involves what is almost a sub-conscious or automatic operation and so requires very little attention. As soon as attention is paid consciously (i.e., by an effort of the will) to what is otherwise automatic, we lose the thread of the main mental processes. The same occurs if attention is diverted by some outside influence.

If, for example, the transmission is of such a quality that here and there the listener misses a word and has to consider from the context what it might have been, his attention is diverted from the main matter.

The essential criterion for assessing the satisfactoriness of telephone transmission is, therefore, the extent to which conscious effort is absent from the transmission of ideas between the speaker and the listener.

Transmission comparisons should be made under conditions not far removed from those of normal operation. It is unreasonable to increase attenuation, noise or distortion so far that the limit of intelligibility is reached. It is equally unreasonable to make use of nonsense syllables which must be recognised without the assistance normally afforded by meaning and context. But, granted that the conditions should be similar to those in normal operation, in terms of which factor should the comparison be made?

What should be the criterion by which the " satisfactoriness " of the transmission is judged, and in what terms should it be measured?

Measurements are easier and more definite if one goes to the limit of intelligibility or uses nonsense syllables (logatoms), but neither represents conditions similar to normal operation. The latest schemes used in the United States avoid this mistake. One is " judgment observations ", i.e. personal preference as the criterion, equalised by adjustment of attenuation as the measure. This scheme is open to the objection that different persons have astonishingly different reasons for " preferring " one circuit to another, some of them not at all dependent upon essential characteristics of transmission. Another method which might be preferable is that of observing actual conversation and counting the number of cases in which the listener asks the speaker to repeat a word or phrase, using this " repetition rate " as the criterion, again employing adjustment of attenuation as the measure. This method might conceivably be useful for determining the plant standards of an administration but it would require a large volume of actual traffic and a very long time before it yielded a result. It is therefore, not suitable for comparisons in a manufacturer's laboratory.

In the method about to be described, the quality of the transmission over a telephone connection is measured by noting the number of times that the listener's attention is diverted by the need for conscious effort to understand a word or phrase. The method is highly subjective, but this is not a disadvantage, since any method of correctly assessing the quality of telephone transmission must inevitably be subjective, if it is to take into account some of the most important factors involved.

It is not difficult to know when attention has to be given consciously. Words and phrases that are satisfactorily received spring at once into the mind, and we have no doubt at all about them. Recognition is full and complete. The idea conveyed is immediately appreciated. On the other hand, when a word is not successfully received, we have a feeling of doubt and detect in ourselves an effort to find associations which will help to identify it. We find that the reception of the main flow of ideas being transmitted is held up while we apply the sound received to that part of our store of knowledge associated with the context to see if there is any response. It is only when a satisfactory response which fits the sound and the context arrives that we recognise the word. Very slight and elementary introspection is sufficient to enable us to detect this effort for every word or phrase not satisfactorily received.

The technique which has been found most satisfactory for putting these ideas into practice is briefly as follows. A list of sentences is prepared by the person supervising the test. These sentences are such as occur in ordinary telephone conversation and are extracted from the news paragraphs of a daily newspaper, all having a length of from six to ten words and usually embodying only a single phrase. The circuit to be tested is set up and the conditions (room noise, etc.) applied. The sentences are spoken over the circuit in as natural a tone as possible and using as nearly as possible " normal volume for telephonometric measurements " defined by the C.C.I.F. The listener listens normally, but notes at once any need for conscious effort to apprehend what is said. Sentences " immediately appreciated ", i.e., received without conscious effort, count as successes; those for which effort is detected are reckoned as failures. The percentage of the former is used as the measure of " satisfactoriness ".

We find that reliable averages can be obtained for one given circuit condition with permutations of three persons acting as speaker and listener in about 120 sentences. When the figures so obtained for varying circuit conditions are plotted, the points lie on a reasonably smooth curve.

3. Method used in 1938 for conducting immediate appreciation tests *

The basis of immediate appreciation testing is to observe, during the transmission of matter of a conversational nature, not the number of complete failures, but the number of occasions when strain occurs, i.e., the occasions when appreciation is not immediate. In practice the speaker reads sentences, chosen from a newspaper article, to a listener whose duty it is to recognise and record the occasions when any sort of mental effort or deduction is required for the appreciation of the meaning of the sentence heard. Obviously the subject matter and phraseology should be of a type conforming with the listener's previous mental experience and intelligence. It has been confirmed by experience that only slight introspection on the part of the listener is required to detect the occasions when appreciation is not immediate.

* Extract from the article cited as Reference (2) in the bibliography below.

The practical procedure is as follows :—

A crew of three is employed. There is no peculiar significance in the choice of this number, but it has been found large enough to give reliable and consistent results, while still small enough to avoid making the tests long and cumbersome.

For speaking and listening all the six permutations of the members of the crew two at a time are employed and, for each individual test, 20 sentences are read out. This gives a total, for each set of applied conditions (junction length, noise level, etc.), of 120 sentences. Experience has shown that this is adequate for consistency.

In order to obtain a reasonably constant literary style and also to simplify the task of extracting the sentences from the newspaper, the same paper (but, of course, not the same issue of it) is always used. It has been found that the slow and tedious process of preparing sentences in advance of the reading can, after a certain amount of practice, be omitted. The crew quickly acquires agility in reading out sentences of the required length and type, without preparation other than previously reading the article through.

It is desirable, on the score of consistency, that the sentences should be of a standardised length of between 6 and 10 words, and that they should convey a single idea. For example, the newspaper article might run :—

" Powerful cranes are hoisting into the vessel sections of the boilers and engines, which are being built up by an army of skilled engineers "... but the speaker would without hesitation read modified sentences such as — " Powerful cranes are hoisting sections of the boilers ", followed, perhaps by :— " The engines are being built by skilled engineers ".

The sentences are read in the speaker's natural tone of voice, but the loudness is carefully controlled to the normal volume for C.C.I.F. telephonometric tests*. This is ensured by the continuous use of a volume meter throughout the tests.

The listener is in a cabinet in which noise can be introduced at a specified level. The noise used is of a miscellaneous type obtained from a microphone which is hung in the rafters of a noisy machine shop. For consistency during the tests the noise level is maintained constant with the aid of an objective noise-meter. Prior to the tests, the noise-meter was calibrated by subjective measurements made by the Davis method**.

When the noise level has been adjusted to the desired value and the junction adjusted to the required length, the six readings, each of 20 sentences, are carried out.

As the listener receives each sentence he decides whether or not he has appreciated its meaning without doubt, strain or the necessity for mental deduction.

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* Normal volume for telephonometric tests has been defined by the C.C.I.F. (see *Green Book*, volume IV, § **3.1.2.B**).

** This is a subjective method using a 1000 c.p.s. tuning fork of known decay factor. The results are obtained directly in phons. The method was described in Nature, No. 3141, Vol. 125 of January 11th, 1930.

If he is satisfied that he experienced no such difficulty he classes the reception of that sentence as a success. The percentage of sentences so received as successes is the " immediate appreciation score " under the particular conditions considered. This process is repeated for as many other sets of conditions as may be necessary. It has been found that five noise levels are adequate, the values used at present being " silence ", 52, 60, 68 and 76 phons. For each of these noise levels the junction length is increased, usually in steps of 5 db, from a value which, with the systems in use and under the particular noise level being applied, will undoubtedly give 100%, immediate appreciation, to a value which reduces that percentage to 50% or 60%.

During the test, the scores of each member of the crew are recorded by him in his own personal file and he is unaware of the results being obtained by the other two. The value of junction length at which the test starts and stops is decided by the person supervising the test.

With the exception of the noise level at the listening end and the length of the connecting junction, both of which are known variables, all the conditions of the test are standardised as far as possible. Definite precautions of the following kind, for example, are taken in the interests of consistency :—

(a) The position of the listener in the listening cabinet is standardised so as to avoid variations due to unequal noise distribution.

(b) The distance from the speaker's lips to the transmitter mouthpiece is maintained constant. In the case of insets in hand-microtelephones the " normal conversation distance " in accordance with the C.C.I.F. recommendations is used.

(c) The transmitters of both the sending and receiving telephones are shaken before each 20 sentences are read out, but they are not mechanically disturbed during the actual reading. (The condition of the transmitter at the listening end is important since it governs the extent of sidetone interference.)

(d) Sentences over which the speaker stumbles are ignored by the listener.

(e) No sentence is ever repeated.

The results obtained by the three operators are averaged and plotted in the form shown in Figure 1 and then redrawn as Figure 2. The statistical analysis then follows.

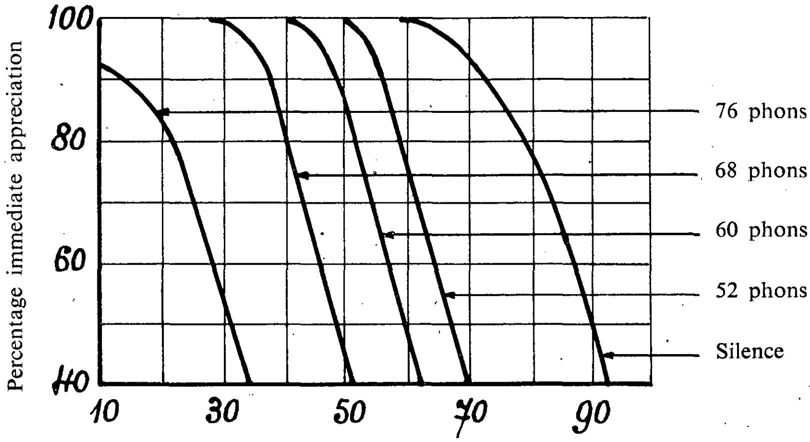


FIGURE 1. — *Variation of percentage immediate appreciation with overall reference equivalent, for various values of room noise at the listening end*

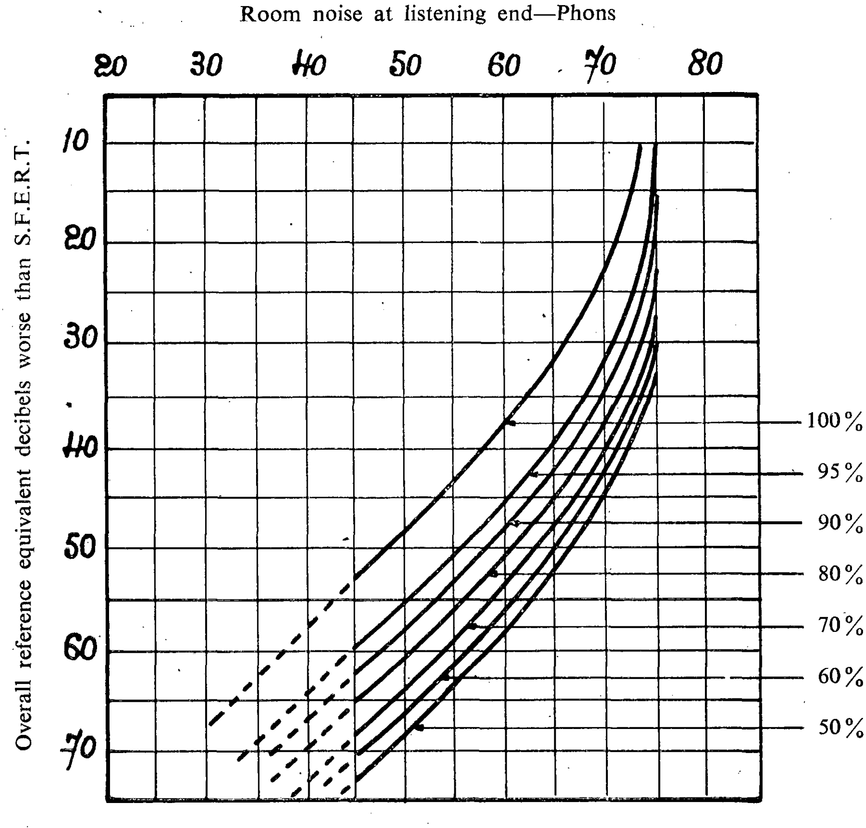


FIGURE 2. — " *Equi-immediate appreciation* " *contours obtained by replotting Figure 1*

It will be remembered from the previous article (1)* that the basis of the method is a statistical survey which combines graphically the estimated statistical distributions of room noise level and speaker's volume level**.

Figure 3 is reproduced from the previous article and shows the completed statistical chart on which have been plotted the " equi-immediate-appreciation " contours of Figure 2.

For the manner in which this chart is constructed the reader is referred to the previous article; it is sufficient here, to remember that the number which occurs in any one square is the product of the probability, expressed as a number per 100, of the occurrence of the noise level lying between the limits represented by the vertical sides of the square, multiplied by that of the speaker's volume level lying between the limits represented by the horizontal sides of the square. Consequently, since the noise level and the speaker's volume level are independent, the number in any one square is the probability, expressed as a number per 10,000, of the simultaneous occurrence of the noise level and speaker's volume level which lie between the limits set by the sides of the square.

It will be remembered that the position of the contours on the chart is determined by the assumed overall reference equivalent of the transmission standard. The assumption is made that the scale of reference equivalent in Figure 2 and the scale of speaker's volume level in Figure 3 are in the same units. Then if it is desired to study the transmission on the assumption of a transmission standard of 25 db worse than S.F.E.R.T., the contours of Figure 2 are placed in such a vertical position on Figure 3 that 25 db on the former's scale coincides with zero (that is normal volume for telephonometrie measurements) on the latter's scale. In Figure 3, as shown, this has been done.

The total of all numbers lying in squares above and to the left of the 100 per cent. contour is the number of cases in 10,000 which can be expected to enjoy 100 per cent. immediate appreciation. To study how this number varies as the standard of transmission changes, it is necessary to slide the contour up or down by the required amount, and again total all the numbers above and to the left of it. It can be seen that this process, although satisfactory, is tedious, particularly

* The passage is not reproduced in the extract from this article appearing in 2 of the present annex.

** These distributions, as explained in the previous article, were built up from such scanty information as was available at that time.

Since then, however, in a paper by A. H. Inglis read before the American Institute of Electrical Engineers, the results of a survey of noise levels at " telephone stations " have been published. They showed that the average noise level was about 50 phons and the standard deviation approximately 12. Even remembering that there possibly exist differences in the technique of noise measurement as between this American survey and the results upon which our own estimated distribution was based, the closeness of agreement between these figures and our corresponding figures of 50 phons and a standard deviation of 11, is reassuring.

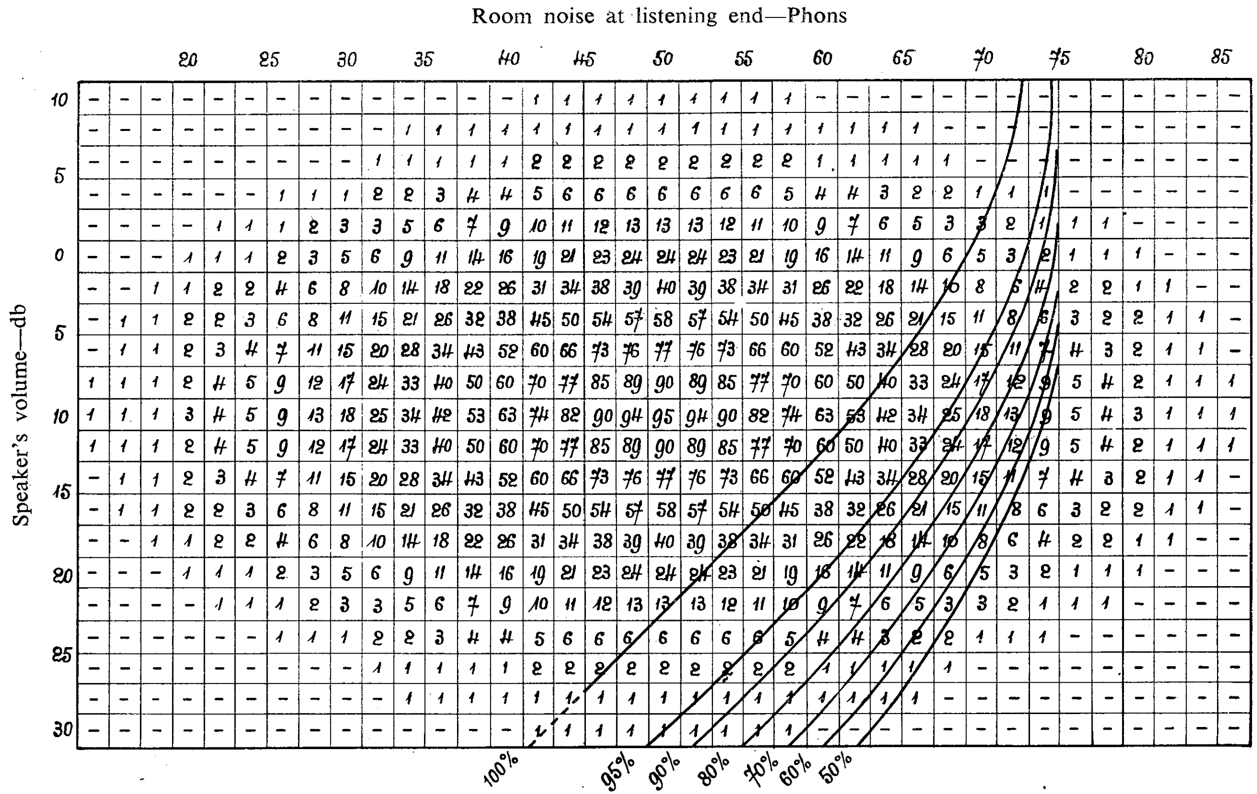


FIGURE 3. — " *Equi-immediate appreciation* " *contours of Figure 2 superimposed on the original statistical chart*

as many of the squares are only partly above and to the left of the line and an estimate must be made in each case of the proportion of the number in such a square which is to be included. The diagram moreover suffers from the fact that it is not pictorial. By this is meant that the approximate result is not readily apparent from the position or shape of the contour. Since the publication of the last article, a simpler technique has been developed.

Figure 4 shows the chart in the form in which it is now used and with the 100 per cent and 95 per cent. contours of Figure 3 drawn on it in full lines. The arrangement of this new chart is such that, instead of totalling numbers in squares above and to the left of the contour, it is only necessary to measure the total area above and to the left of the contour. This process, which can be done by means of a planimeter, is far simpler and much quicker.

In order to make the simplified operation possible, the new chart differs from the original in that the distance between say, the vertical lines representing 50 phons and 52 phons is proportional to the probability of the noise level lying between those two values. This applies similarly to all the lines on the chart, horizontal as well as vertical. The size of the chart is 100 units square and the area of the chart is therefore 10,000 square units.

The area of any one rectangle in the chart, being the product of the lengths of its sides, is therefore the product of two probabilities and so is the probability of simultaneous occurrence of the values lying between the limits set by the sides of the rectangle. In other words the area of a particular rectangle, in " square units ", will be found to equal the number which occurred in the corresponding square of the old chart.

When it is desired to total all the numbers above and to the left of a particular contour, it is now only necessary to measure the complete area of the space above and to the left of that contour. Naturally, owing to the non-linear scale, it is no longer possible simply to slide the contours up the chart as the transmission standard is changed. Instead, the curve must be replotted in each new position, and, of course, it changes shape as well as position. In Figure 4 the dotted contour is the 100 per cent. line replotted on a basis of a transmission standard of 35 db instead of 25 db worse than S.F.E.R.T.

The new method not only saves a very great deal of time but, also, the diagrams become fully pictorial. That is to say, one can see, at a glance, the approximate percentage of the chart area which lies above and to the left of the contour.

In the tests which have been carried out recently, we have standardised our comparisons on a 95% immediate appreciation contour. It is in general uneconomic to design anything to perfection (i.e., 100%) and moreover, in this case, it is not easy accurately to read off the " noise level—reference equivalent" co-ordinates at 100% immediate appreciation from curves drawn as in Figure 1.

The choice of 95 per cent. overcomes this difficulty and yet is probably a sufficient standard for high-quality commercial speech. There is, nevertheless, no finality in the choice of that figure.

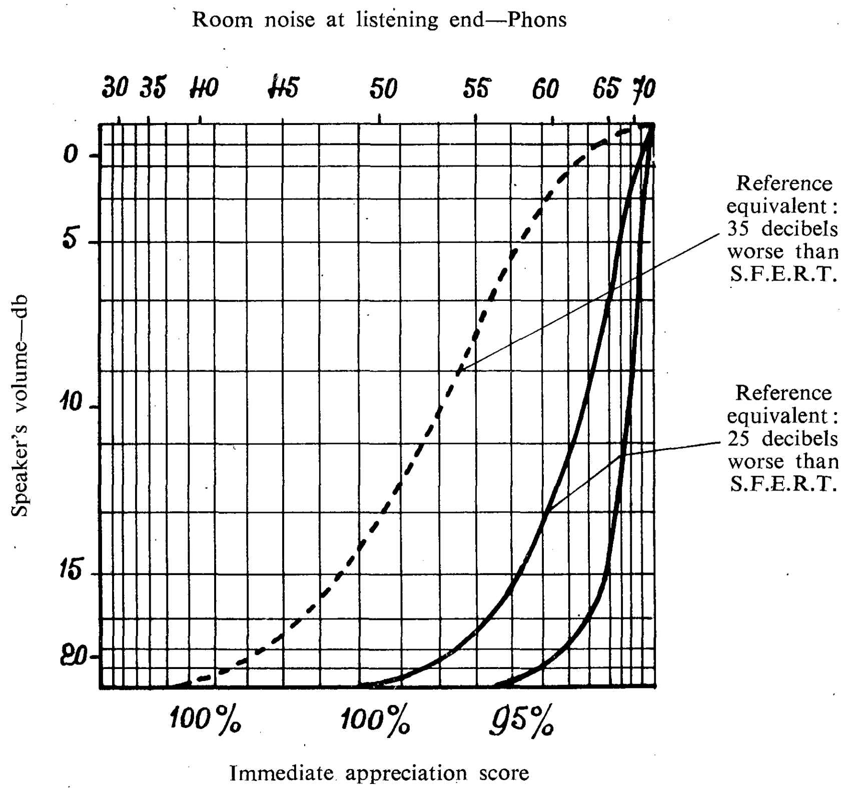
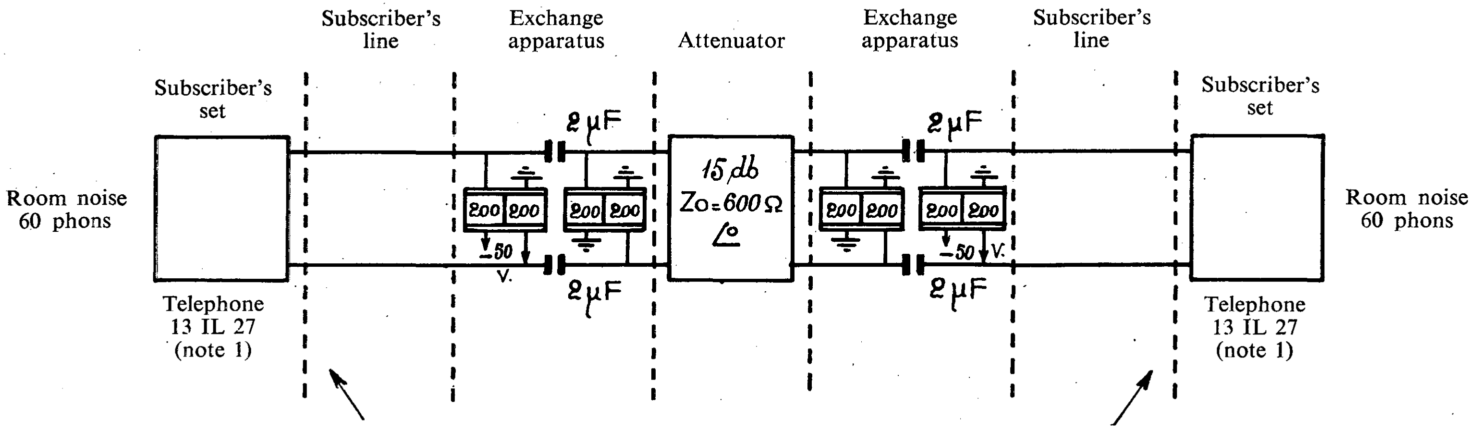


FIGURE 4. — " *Equi-immediate appreciation* " *contours from Figure 3 drawn on the new form of statistical chart*

4. Immediate appreciation test procedure used in 1948-1949 by the Australian Telephone Administration

While the basic principles described above have been adopted in tests in the Laboratory of the Australian Telephone Administration, the objective is normally the measurement of transmission performance rating of one communication circuit relative to a working reference circuit. Tests are therefore usually



2.84 miles of 10 lb cable loop resistance 500 ohms
Resistance per mile 176
Capacitance per mile 0.072 μF

Note 1. — Telephone 13.IL. 27 refers to handset type telephone having the following transmission components:

Transmitter inset No. 13;
Receiver type IL;
Anti-sidetone induction coil No. 27

FIGURE 5. — *Adopted by the Australian Telephone Administration*

carried out with a fixed room noise level (60 db). This room noise is of the continuous spectrum type, generated by amplifying the Johnson noise present in a resistor at room temperature. The noise level is measured by means of an American (General Radio) Sound Level Meter, using the 70 db weighting network.

The speakers' talking level is allowed to find its normal level under the particular conditions of the test and sidetone effects all allowed full play. As normal routine measurements are of transmission performance rating, involving a comparison test, it is not necessary to know the absolute value of the talking level. In tests of a more absolute nature, a sufficiently large number of speakers is used to ensure that the mean of all speakers' levels approximates to the mean value observed in service. Grinsted (1) deduces from published data that the distribution of service speaking levels is roughly that of the statistical " normal " or gaussian distribution and has a standard deviation of 6 db. The standard deviation of the mean of the talking levels of a group of N speakers selected at random is therefore approximately $\frac{6}{\sqrt{N}}$ db.

In the most recent tests by the laboratory of the Australian Telephone Administration, a junction splitting amplifier is used at the receiving end of the junction line, represented by the 15 db attenuator in Figure 5, so that up to four listeners may listen simultaneously. The gain of this amplifier is adjusted to be 0.0 decibel and identical exchange feeding bridges and subscribers line circuits can be connected to each output channel. Thus all listeners receive simultaneously the same speech volume level and are subject to the same sidetone conditions as apply for one listener when the splitting amplifier is not used.

The use of a multi-channel listening circuit reduces the time required to make a test and as a result increases the reliability of comparison tests. The reliability of comparison tests of the receiving ratings of two different local line circuits or subscribers' instruments may be improved by connecting each instrument to a separate receiving channel and listening to a common speaker, the listeners being interchanged systematically from instrument to instrument.

5. Statistical analysis of test results

5.1 " *Pure chance* " *variations*

Actual test results show that, with practical telephone systems, one hundred percent immediate appreciation of a group of, say, one hundred sentences is recorded only on infrequent occasions, and that the transmission performance rating of a given system is based on observations taken when there are a certain proportion of sentences not immediately appreciated.

The factors which cause a failure of immediate appreciation to be recorded are factors which vary in an unpredictable manner, e.g. variations in speaking level, difficulty of sentences, the random nature of continuous spectrum noise and variability of sensitivity of carbon granule telephone transmitters. As a result of this, the mean immediate appreciation scores obtained from a number of replications will not be the same for each replication.

The distribution of the various replication means about the overall mean, due to purely random factors, will be that of the binominal distribution considered in elementary statistical theory. (See, for example, Reference 4.) It is therefore possible, on the assumption that the probability of failure to receive a sentence remains the same from sentence to sentence, to predict the standard error, i.e. the standard deviation of replication means, resulting from such factors. This is given by the following expression :

$$S = \sqrt{\frac{p(100-p)}{n}}$$

where S is the standard error (root mean square deviation of replication mean from the overall mean),

n is the number of sentences in each replication, and

p is percentage "immediate appreciation" obtained over an extended period.

Under the usual test conditions, when n is large, the distribution of group means is very nearly gaussian and this fact may be used to estimate the confidence limits to be placed on the results of a test in which only random factors influence the result.

For example, in the tests described by Grinsted, 120 sentences were read for each set of test conditions. In a typical case where the average immediate appreciation score was 95 %, the standard error of the 120 sentence mean estimated from the above expression is 2.0 % and the 95 % confidence limits based on gaussian distribution are approximately 91 % and 99 %.

5.2 *Other variations*

In addition to variations due to minor unassignable causes, there are factors due to the speaking and hearing characteristics of crew members which, on the average, tend to remain constant or change slowly from test to test. For example, one crew member may naturally speak at a level a few decibels higher than the other crew members. Such factors do not necessarily affect the accuracy with which two communication systems may be compared, but they do increase the variance of the immediate appreciation scores obtained on a group of sentences. If the overall mean square deviation of individual observations from the general means is S_o (see Reference 5), we can write :

$$S_o = S_1 + S_2 + S_3 + \ldots\ldots + S_r$$

where S_1, S_2, etc..., are the mean square deviations contributed by certain assignable causes, and S_r is the residual mean square deviation contributed from unknown sources, and which, in a well conducted test, will be a function of the standard error due to pure chance dealt with in the previous section.

The statistical technique of determining S_1, S_2, S_r, etc..., from which the corresponding variances V_1, V_2, V_r, may be calculated, is known as the analysis of variance and is described in standard texts on statistical theory.

In a comparison test, the residual variance is normally the only one which determines the inaccuracy of the comparison. Since this may be estimated beforehand from the binomial distribution it is possible to calculate the minimum amount of testing work necessary to carry out a given test. For example, suppose that two communication systems are to be compared under conditions which cause the immediate appreciation to be 90% and that the uncertainty in the overall means is not to exceed $\pm 1\%$. If the 95% confidence limits to the overall mean are taken as 89% and 90%, then the standard error of the overall mean must not exceed 0.5%. Using the relation

$$S = \sqrt{\frac{p\,(100-p)}{n}}$$

we put $S = 0.5$ and $p = 90$ and calculate n, the total number of sentences to be used in the test on each system. It is found that $n = 3600$ sentences, and this is the *minimum* number of sentences to be spoken over each system if the overall mean is to have the required degree of precision. The testing rate, including rest periods, is about 300 to 400 sentences per hour, and in this particular case, a total testing time, assuming no multiple listening arrangements, is 18 to 24 hours.

6. Application of immediate appreciation test to check subscribers' local line limits

As an example of the application of the immediate appreciation test technique, a series of measurements on the Overall Transmission Performance Working Standard Circuit adopted by the Australian Telephone Administration, will be described. This test was originally planned as an absolute measurement and not as a comparison measurement, i.e. the final objective was the production of an immediate appreciation versus junction attenuation graph, and for this reason a large crew was used. The test also provided a basis for the study of the immediate appreciation technique as very little previous experience of this type of test had been gained.

6.1 *Selection of operators*

As it was decided to use untrained and untried of subjects in these tests, preliminary checks were made to detect those having abnormal speaking or hearing characteristics.

6.1.1 *Speaking tests.* — A telephone circuit using 300 type C.B. telephone sets was set up between two rooms, using continuous spectrum room noise (60 db sound level) in both rooms. Provision was made for simultaneous reception at the listening end by 4 persons, selected members of the Telephone Instrument Division.

Each speaker read a list of 100 English words, which were written down by the listeners. The average word articulation score was 62.0 % (mean of 277 observations) and those speakers for whom the listeners obtained an average score of less than 40 % were rejected as unsuitable. The 40 % rejection level was determined after a few tests had been made, and was largely an arbitrary figure.

6.1.2 *Listening tests.* — Two sets of listening tests were carried out. The first set followed the same procedure as for the speaking test, except that the person being tested formed one of the listening crew of four. At a later date an audiometer was set up using laboratory components and a calibrated moving coil receiver; with this apparatus the hearing loss of crew members was measured. Despite the fact that all crew members gave satisfactory results in the speech tests, it was found that high frequency deafness was quite pronounced among the older men.

6.2 *Organisation of crews*

At the time the studies reported herein were undertaken, no satisfactory apparatus was available for multiple listening and the observers available for the tests were split up into crews of three. The test programme provided for one crew member speaking, another listening, and the third resting at any time. Larger crews would have meant a larger number of observers disengaged at any time and the cost of the tests would have been increased accordingly; such considerations outweighed the advantages which would have been obtained as a result of the larger number of crew combinations (speaker-listener) which result from the use of a larger crew.

The selection of crews was made so as to cause the least possible disorganisation to the other section of the Laboratories from which the individual observers were drawn. In all, 11 different crews of 3 participated in the tests. Only 6 crews produced sufficient results to enable their use in this Report.

Attemps were made to organise equal numbers of male and female crews but this was not possible due to the rapid replacement rate experienced with female employees, insufficient data was obtained from any such crew to enable satisfactory comparison of the results obtained by it with the results obtained by the male crews. No female crews produced sufficient results for use in this Report.

The tests extended over a period of 10 months, from September 1948 to July 1949. Because of difficulties of organisation, this period is much too long, and in any future tests of this kind, the maximum possible use of multiple listening arrangements will be made.

6.3 *Test procedure*

The proposed " Overall Transmission Performance Working Standard Circuit " shown on Figure 5 was set up in two rooms connected by a junction line into which a 600 ohm variable attenuator was connected. Networks representing the limiting lengths of 20 lb, 10 lb and 6½ lb subscribers' cable were made up and connected into the circuit as required.

Each speaker read a group of 20 sentences from the sentence list published by Fletcher and Steinberg (7). All of the six possible combinations of speaker and listener were used and the tests were repeated with a number of different valeurs of junction line attenuation, set up in random order.

6.4 *Test results*

The overall averages are plotted as graphs of immediate appreciation versus junction attenuation in Figure 6. The graphs show that there is little difference in the transmission performance of telephone connections complying with the Overall Transmission Performance Working Standard laid down in the Transmission Engineering Instructions but having different sizes of subscribers' cable of the lengths given. The deviations which do appear on the graphs of Figure 6 are mostly at junction attenuations exceeding 25 db, well in excess of the maximum of 15 db set down in the Overall Transmission Performance Working of the Australian Telephone Administration.

6.5 *Statistical analysis of the results*

A detailed analysis of the results given for the 6½ and 10 lb cables for a junction attenuation of 32 db has been made using the analysis of variance technique. The individual scores obtained by each crew combination, which form the basic data for this calculation, are given in Tables 1 and 2. Each score in

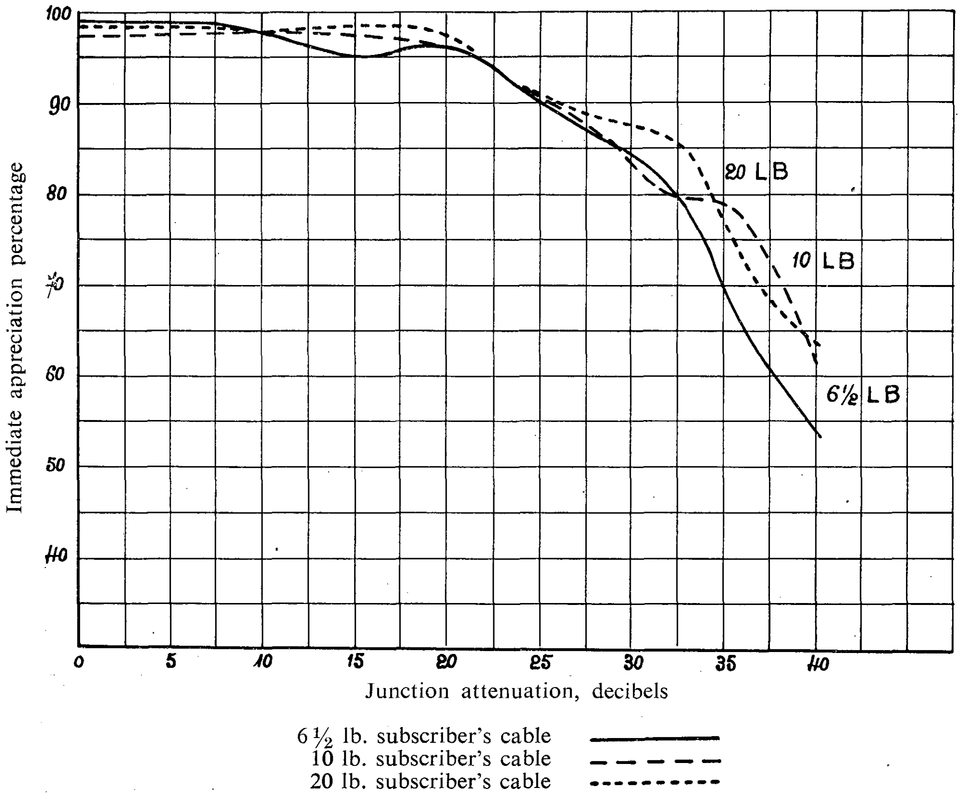


FIGURE 6. — *Comparison of subscriber's line limits*

these tables gives the number of sentences immediately appreciated out of a total of 20 read. The results of the analysis of variance are given in Table 3.

In Table 3 the variance figures given have been obtained from the mean square deviations (from the overall mean) due to the various sources listed, and the number of degrees of freedom. In the case of crew combinations, line conditions, and replications, the number of degrees of freedom is one less than the number of variables, e.g. there are 36 crew combinations listed of speaker and listener and hence 35 degrees of freedom in calculating the variance. The number of degrees of freedom for the interaction terms are the products of those for the two principal sources of variation concerned, and the number of degrees of freedom for the residual term is the product of those for all three principal sources of variation.

The " variance ratio " is the ratio of the variance due to the source listed to the residual variance. The residual variance is that due to all unassignable causes, as discussed in * 5 above. If the variance ratio is high, them it is fairly certain that the source of variation concerned is a significant one : if the variance ratio is low, then it is possible that the variance attributed to a particular source

TABLE 1

Details of immediate appreciation scores obtained at a junction attenuation of 32 db in checking subscribers' line limits for 10 lb cable

Crew No.	Speaker	Listener	Replication No. 1	2	3	4	5	Means
3	LWL	RWK	16	14	(0)	(0)	(9)	7.80
	LWL	LS	19	15	(0)	(0)	(9)	8.60
	RWK	LWL	20	19	(0)	(0)	(7)	9.20
	RWK	LS	17	18	(3)	(0)	(9)	9.40
	LS	LWL	20	20	(19)	(15)	(17)	18.20
	LS	RWK	17	18	(4)	(10)	(17)	13.20
4	RC	JF	15	16	17	17	19	16.80
	RC	FFH	13	20	13	9	20	15.00
	JF	RC	14	19	20	13	17	16.60
	JF	FFH	19	20	19	18	20	19.20
	FFH	RC	14	16	19	3	19	14.20
	FFH	JF	19	20	20	18	20	19.40
5	GCS	JWC	18	18	20	18	20	18.80
	GCS	AHL	14	20	20	16	18	17.60
	JWC	GCS	17	20	14	16	19	17.20
	JWC	AHL	20	20	19	18	19	19.20
	AHL	GCS	19	20	16	18	17	18.00
	AHL	JWC	18	20	20	19	18	19.00
7	JWS	ROR	19	17	15	10	17	15.60
	JWS	CVE	20	19	20	19	20	19.60
	ROR	JWS	18	17	18	19	19	18.20
	ROR	CVE	16	16	19	19	20	18.00
	CVE	JWS	18	17	20	20	20	19.00
	CVE	ROR	20	7	14	18	19	15.60
8	DG	JF	12	10	9	14	20	13.00
	DG	AHA	11	10	16	20	16	14.60
	JF	DG	16	14	9	17	19	15.00
	JF	AHA	20	16	18	20	20	18.80
	AHA	DG	12	17	20	19	20	17.60
	AHA	JF	15	17	19	20	19	18.00
10	BPP	NCH	11	18	14	20	20	16.60
	BPP	JTB	20	18	19	20	18	19.00
	NCH	BPP	19	6	8	16	20	13.80
	NCH	JTB	18	18	18	18	20	18.40
	JTB	BPP	17	18	6	20	8	13.80
	JTB	NCH	6	18	11	19	12	13.20
Means			16.583	16.833	14.333	14.889	17.250	15.978

TABLE 2

Details of immediate appreciation scores obtained at a junction attenuation of 32 db in checking subscribers' line limits for $^1/_2$ lb cable

Crew No.	Speaker	Listener	Replication No. 1	2	3	4	5	Means
3	LWL	RWK	2	3	(0)	(5)	(5)	2.00
	LWL	LS	7	12	(0)	(11)	(11)	8.20
	RWK	LWL	13	16	(16)	(17)	(17)	15.80
	RWK	LS	16	17	(16)	(14)	(14)	15.40
	LS	LWL	19	20	(16)	(20)	(20)	19.00
	LS	RWK	19	20	(6)	(19)	(19)	16.60
4	RC	JF	20	19	20	15	20	18.80
	RC	FFH	20	15	19	10	20	16.80
	JF	RC	19	17	20	20	18	18.80
	JF	FFH	20	20	19	20	20	19.80
	FFH	RC	20	19	20	20	20	19.80
	FFH	JF	20	20	19	20	20	19.80
5	GCS	JWC	16	20	19	12	19	17.20
	GCS	AHL	14	20	20	12	15	16.20
	JWC	GCS	18	18	18	19	14	17.40
	JWC	AHL	18	20	20	15	20	18.60
	AHL	GCS	11	18	18	17	19	16.60
	AHL	JWC	19	18	19	19	14	17.70
7	JWS	ROR	12	13	13	13	20	14.00
	JWS	CVE	19	18	14	17	18	17.20
	ROR	JWS	18	14	15	16	18	16.20
	ROR	CVE	20	16	16	17	17	17.20
	CVE	JWS	19	18	18	20	18	18.60
	CVE	ROR	20	8	12	16	19	15.00
8	DG	JF	4	12	18	15	12	12.20
	DG	AHA	13	12	20	20	20	17.00
	JF	DG	7	19	19	15	20	16.00
	JF	AHA	20	20	20	20	20	20.00
	AHA	DG	16	20	18	19	20	18.60
	AHA	JF	17	17	20	20	18	18.40
10	BPP	NCH	19	20	18	15	18	18.00
	BPP	JTB	18	19	16	15	19	17.40
	NCH	BPP	2	20	3	10	20	11.00
	NCH	JTB	11	19	12	17	20	15.80
	JTB	BPP	20	20	20	15	18	18.60
	JTB	NCH	19	20	8	16	17	16.00
	Means		15.694	17.139	15.694	16.111	17.694	16.467
				Overall mean Tables 1 and 2 =				16.223

TABLE 3

Results of analysis of variance of data in Tables 1 and 2

Test : Comparison of local line limits for subscribers $6\frac{1}{2}$ lb and 10 lb cable. Junction attenuation, 32 db.

Source of variation	Degree of freedom	Variance	Variance ratio	Degree of significance (See Note)
Crew combinations	35	94.7	5.25	Highly significant
Line condition (i.e. $6\frac{1}{2}$ or 10 lb)	1	21.2	1.175	Not significant
Replications	4	74.4	4.12	Highly significant
Interactions				
Replications and crew combinations	140	10.14	0.562	Not significant
Line conditions and crew combinations	35	16.60	0.920	Not significant
Line conditions and replications	4	14.58	0.808	Not significant
Residual	140	18.04	—	—

Note — Variance ratios greater than the 1% significance level are designated "highly significant"; those lying between the 5% and 1% levels "significant"; and those less than the 5% level, "not significant". The variance ratios corresponding to the difference significance levels depend upon the number of degrees of freedom; the actual values may be found from tables given in standard texts on statistics.

is a result of random sampling effects only. Statistical tables (e.g. those of Reference 4 and the graphs in Reference 6) show, for various degrees of freedom, the probability of a variance ratio being due to random sampling effects. The value of variance ratio, which for a given number of degrees of freedom can be expected to occur only once in a hundred times as a result of " chance ", or random sampling, is called the 1% significance level. If the variance ratio calculated for a given source of variation exceeds this value, that source of variation, for the purposes of this Report, is designated " highly significant ". Variance ratios lying between the 5% and 1% levels are designated " significant ", and variance ratios less than that corresponding to the 5% probability are designated " not significant " and are assumed to arise from the same sources as the residual variance.

In Table 3 the residual variance is 18.04 corresponding to a standard deviation of

$$\sqrt{18.04} = 4.25 \text{ or } \frac{4.25}{20} \times 100\% = 21.25\%$$

The value estimated from binomial theory (see § 5.1) for a probability of success

$$\frac{16.222}{20} = 0.8111 \text{ or } 8.75\%$$

Thus, in this case, the variability of the test results, after allowing for known sources of variation, is much greater than would be obtained from a simple test in which the probability of success remained constant. It is quite possible to obtain residual variances close to the theoretical values and an example is given in § 7.2 below.

The Analysis of Variance table shows that the variance due to crew combinations is highly significant and it is therefore concluded that the *absolute* values of the results of immediate appreciation test of the type described in this Report is of little value unless a fairly large crew is used. In the present case, 36 crew combinations were used and the variance due to crew combinations, from Table 3, is 94.7, corresponding to a standard deviation of

$$\sqrt{94.7} = 9.74 \text{ or } 48.7\%$$

The standard error in the overall mean as a result of crew combinations is

$$\frac{48.7}{\sqrt{35}} \text{ or } 8.1\%$$

To achieve, in this test, a standard error of 1% in the overall mean due to crew variations, the number of crew combinations would need to be increased to about 2370. It is of interest to determine whether the variance due to crew combinations arises from differences between speakers or from differences between listeners, or from both. This could only be determined accurately if each speaker spoke to all listeners, which was not so in the present case. However, by taking from Tables 1 and 2 the mean score for each observer as a speaker and as a listener, the speaker means are found to have a standard deviation of 15.7 percent and listener means a standard deviation of 13.8 per cent. Although this indicates that slightly more variation arises from speakers than from listeners, use of statistical tests shows that the difference between 13.8 and 15.7 per cent. is not significant and such a difference would occur due to chance variations 6 times in every 10 trials.

After crew combinations, the largest source of variation is in replications, i.e. in repetitions of the same test. It is of value to determine whether this arises from random variations or as a result of some systematic variation, such as might arise as a result of observers becoming accustomed to a certain type of distortion and obtaining higher scores at each trial. The crew totals for the first ten tests at a junction attenuation of 32 db are given in Table 4. The results of Crew 3

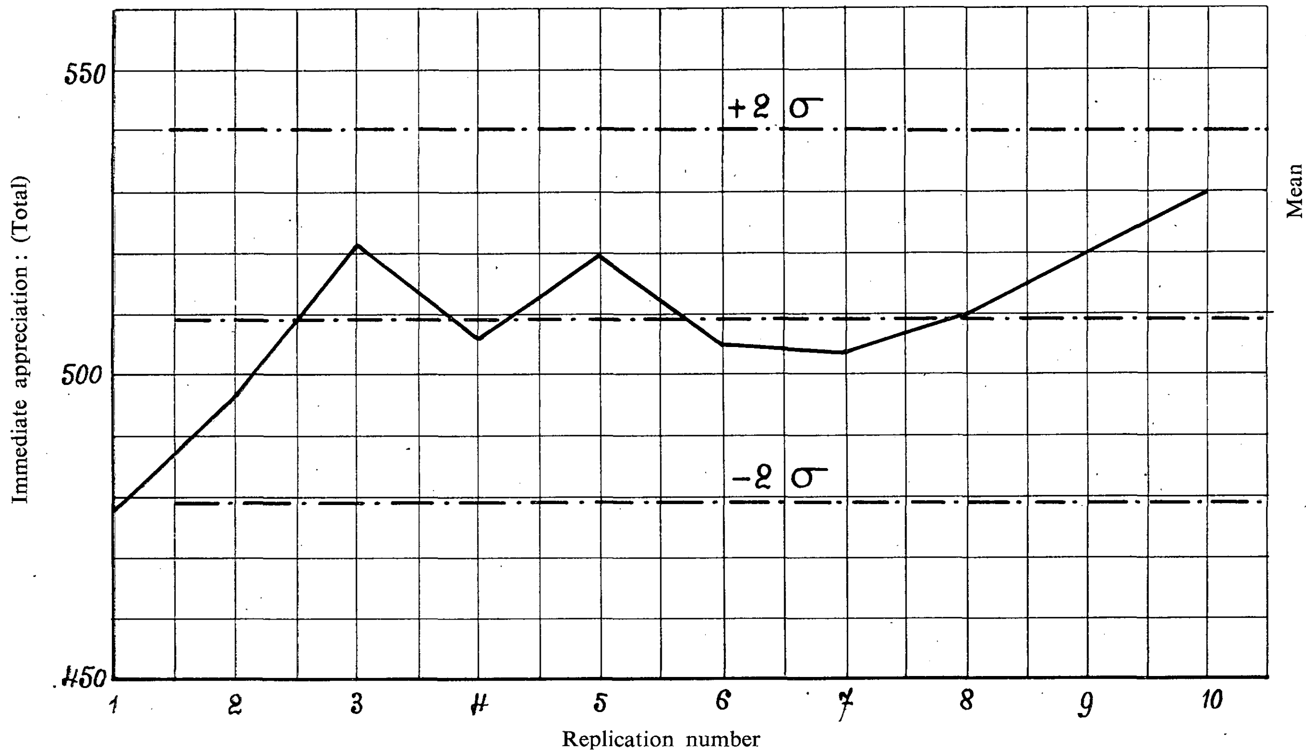


FIGURE 7. — *Total immediate appreciation score vs. Replication number*

TABLE 4

Crew totals for first ten tests at a junction attenuation of 32 db

Replication No.	Crew No. 4		5		7		8		10		Total
	Line	Total	Line	Total	Line	Total	Line	Total	Line	Total	
1	20 lb	94	6½ lb	96	10 lb	111	10 lb	86	10 lb	91	478
2	10 lb	94	20 lb	113	20 lb	106	6½ lb	77	20 lb	107	497
3	6½ lb	119	10 lb	106	6½ lb	108	6½ lb	100	6½ lb	89	522
4	10 lb	111	10 lb	118	10 lb	97	10 lb	84	10 lb	96	506
5	6½ ln	110	6½ lb	114	6½ lb	87	10 lb	91	6½ lb	118	520
6	6½ lb	117	6½ lb	114	6½ lb	88	10 lb	110	10 lb	76	505
7	10 lb	80	10 lb	109	10 lb	106	10 lb	96	10 lb	113	504
8	10 lb	108	10 lb	105	10 lb	105	6½ lb	115	6½ lb	77	510
9	10 lb	115	10 lb	111	6½ lb	98	6½ lb	109	6½ lb	88	521
10	6½ lb	105	6½ lb	94	6½ lb	110	6½ lb	110	6½ lb	112	531

Replication mean = 509.4 (= 84.9%)
Standard deviation = 15.2

are omitted since there were insufficient data available for the chosen junction attenuation of 32 db but the results for all three types of local line have been included. The overall replication means have been plotted in graphical form in Figure 7. It will be seen that a rise in immediate appreciation was experienced after the first test and there was also a gradual increase during the last tests. It appears, therefore, that there is some evidence of a "practice effect" but of rather different nature to that observed in syllable articulation tests (Reference 7).

7. Application of immediate appreciation test to compare two telephone instruments

7.1 *Test procedure*

In Research Laboratory Report No. 3,333 of the Australian Telephone Administration (Reference 8) comparative tests on two different telephonist's telephones were described. Portion of the results obtained in the speaking tests

TABLE 5

Comparison of Australian Telephone Administration telephonist's telephone and a telephonist's telephone of Belgian Manufacture

(See Research Laboratory Report No. 3333, of the Australian Telephone Administration)

Australian Telephone: No. of sentences "immediately appreciated" in a total of 20 read

Listener	Speaker					Listener totals
	N.L.	E.McI.	P.B.	D.W.	E.M.	
N.L.	(17.7)	20	19	20	17	93.7
E.McI.	18	(20)	20	19	16	93
P.B.	18	20	(20)	20	17	95
D.W.	17	20	19	(19)	15	90
E.M.	17	20	20	20	(16.3)	93.3
Speaker totals	87.7	100	98	98	81.3	465.0

Belgian Telephone

Listener	Speaker					Listener totals
	N.L.	E.McI.	P.B.	D.W.	E.M.	
N.L.	(19.75)	20	20	20	20	99.75
E.McI.	17	(18.1)	20	19	16	90.1
P.B.	19	20	(20)	19	19	97.0
D.W.	18	18	19	(17.1)	14	86.1
E.M.	17	16	20	16	(15.4)	15.4
Speaker totals	90.75	92.1	99.0	91.1	84.4	457.35

are given here to show that a more satisfactory test than that described in § 6 above may be made when multiple listening arrangements are provided. In these tests a splitting amplifier at the receiving end allowed four observers to listen simultaneously to each speaker. A crew of 5 could then be effectively used and a large number of results obtained in a comparatively short time. The crew used consisted of 2 men (E.McI. and D.W.) and three women (N.L., P.B., and E.M.).

7.2 *Test results*

The test results obtained for a junction attenuation of 15 db are given in Table 5. There is inevitably a series of " missing values " in a table of this kind, due to the fact that no observer can speak and listen simultaneously. The missing values in Table 5 have been estimated by means of the technique described in Reference 6 and are enclosed in brackets () to distinguish them from the actual test results. Calculation of the missing values is a necessary preliminary to the analysis of variance, the results of which are shown in Table 6. The most important result of the test is that there is an insignificant difference between the sending performances of the two telephones. It is of interest to note that there is a significant interaction between telephones and speakers, and between telephones and listeners, but no significant interaction between speakers and listeners. The greatest source of variation is the differences between speakers, and it is considered that this is mainly due to differences of speech volume. It would therefore appear to be advantageous to use controlled speaking volumes in tests where this action would not affect the final result, e.g. in cases where the effects of sidetone at the sending end are not to be included in the test.

TABLE 6

Comparison of Australian Telephone Administration telephonist's telephone and a telephonist's telephone of Belgian manufacture

Results of analysis of variance

Source of variation	Degrees of freedom	Variance	Variance ratio	Significance
Telephones	1	29.27	1.697	Not significant
Speakers	4	390.8	22.7	Highly significant
Listeners	4	159.3	9.25	Highly significant
Interactions				
Telephones and listeners	4	73.82	4.28	Significant
Telephones and listeners	4	82.44	4.78	Significant
Speakers and listeners	11	31.48	1.827	Not significant
Residual	11	17.42	—	—
Total	39			

N.-B. — Variances given above were calculated from the immediate appreciation scores given in Table 5, expressed as percentages.

The residual variance of 17.42 (calculated on percentage scores) compares favourably with the value (35.9) estimated from the binomial distribution (the corresponding standard deviations are 4.15% and 6.0% respectively), and it is therefore concluded that the test procedure is efficient; more so than in the test discussed previously, where the actual residual variance was approximately 2½ times the estimated value.

8. Calculation of relative transmission performance ratings from immediate appreciation test results

The definition of " transmission performance rating " (relative to the S.R.A.E.N.) appears in the *Green Book* of the C.C.I.F., Volume IV, § 1.2 *a)*. By departing from this definition, it is clear that, if immediate appreciation is taken as the criterion of transmission quality, the *relative* transmission performance rating can be obtained, relative to any reference system, by measuring the interval in decibels between the two points of intersection of a line of constant immediate appreciation with the curves giving the variation of immediate appreciation score, for these two systems, as a function of junction attenuation.

Examination of the shape of typical curves indicate that the relative transmission performance rating obtained in any given case depends upon the value of immediate appreciation chosen for the intercept. Grinstead (1) used a value of 95%; however, frequently cases arise in which an intercept at this level is unsatisfactory. Such a case exists when one or both curves do not reach a value of 95% at any value of junction attenuation. In other cases, the 95% value may be reached but the slope of one or both graphs may be very small at this level, and in such a case large errors in transmission performance rating may result from small errors in immediate appreciation scores.

On the other hand, the use of an intercept at a low level cannot be justified because the standard error of immediate appreciation scores increases as the immediate appreciation average falls (see § 5.1 above), and further, such a procedure may result in a rating which was only applicable when the transmission performance of the circuit was commercially unsatisfactory.

Due to the fact that the immediate appreciation scores obtained by one crew are not necessarily repeatable by any other crew, it is undesirable to choose an invariable value of immediate appreciation score for the evaluation of transmission performance rating. It may be suggested that the immediate appreciation score chosen may be at a level corresponding to, say, 90% of the maximum average value of immediate appreciation obtained during the test, but this is still open to the objection that the inferior system may not reach a sufficiently high level of performance to make an evaluation possible at the value of immediate appreciation so calculated.

To overcome these and other difficulties, it is suggested that the relative transmission performance rating might be calculated from the *area* enclosed between the immediate appreciation junction attenuation graphs of the test and reference systems, the rating so determined being dependent upon circuit performance over a range of junction attenuations.

Similar suggestions to this have been made by Pocock (9) in relation to articulation tests, with the object of obtaining better correlation between ratings determined by such tests and those determined by repetition rate tests. Although the method of determining relative transmission performance rating suggested above has the possible advantage of resulting in better correlation with ratings determined from repetition rate tests, it is suggested here mainly for the purpose of ensuring that ratings determined by subjective tests will be satisfactorily repeatable. A definite recommendation as to the method of calculating relative transmission performance ratings for general use from immediate appreciation scores cannot be made until further experience has been gained in subjective testing, particularly in repetition rate tests.

9. Conclusions

9.1 *General remarks on immediate appreciation tests*

The immediate appreciation test has the important advantage over articulation tests, so far as the Laboratories of the Australian Administration are concerned, that crew members require only a minimum of preliminary training. The checking and calculation of immediate appreciation scores is a relatively straightforward matter.

To obtain results which have useful *absolute* values, crews larger than can be assembled in those Laboratories are required, and hence future laboratory tests should be organized and carried out on a comparative basis only. The use of multiple listening arrangements such as have now been installed, makes it possible to complete tests of satisfactory precision in a much shorter time than was possible previously.

9.2 *Use of immediate appreciation test results for calculation of relative transmission performance ratings*

Various methods of calculating Relative Transmission Performance ratings from immediate appreciation test results have been discussed in this annex, but no satisfactory recommendations can be made at this stage. The problem in very similar to one being considered by the C.C.I.F., viz. the determination of A.E.N. ratings from articulation tests.

9.3 *Use of statistical techniques in connection with immediate appreciation tests*

It has been shown that statistical techniques enable an estimate to be made of the minimum amount of testing required to achieve a given result. The analysis of variance procedure is an invaluable tool in the evaluation of the test results.

9.4 *Transmission performance of the working standard reference circuit*

The " Overall Transmission Performance Working Standard Circuit " adopted by the Australian Telephone Administration in Transmission Engineering Instructions, General TS 2050, has been tested with networks representing the proposed limiting lengths of subscribers' cable in weight 6½, 10 and 20 pounds per mile. In confirmation of the transmission performance calculations, the immediate appreciation test results show that there is no significant difference in the performance of these three networks.

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ANNEX 24

METHOD USED BY THE BRITISH TELEPHONE ADMINISTRATION FOR THE SUBJECTIVE ASSESSMENT OF TELEPHONE TRANSMISSION QUALITY

1. Introduction

The British Telephone Network has been developed with the broad objective of providing most economically a quality of transmission performance which telephone users find satisfactory. Experience shows that circumstances can occur in which two circuits are being compared by ordinary telephone users and that one which yields the higher articulation scores (as assessed by a trained and highly skilled crew) is found less satisfactory than the other which yields lower articulation scores. That is, the correspondence between satisfaction of ordinary telephone users and the scores obtained in an articulation test is by no means perfect. Thus the validity of all ratings such as by articulation must be scrutinized in the light of assessments conducted under much less artificial conditions.

The British Telephone Administration does not, therefore, rely entirely upon highly artificial assessment methods such as loudness balancing (Reference Equivalent) or articulation (A.E.N.) but employs also several " free conversation " assessment techniques which require the performance by ordinary untrained users of several tasks closely related to normal use of the telephone. Indeed the undesirability of relying upon any one criterion has led to a " multi-criterion free conversation " technique.

The planning of the British Telephone Network will continue to be based on the most recent information on the performance of its plant under practical condition of use rather than under any arbitrary and highly artificial conditions. For the practical purposes of making relative assessments (e.g. for planning) of various types of local telephone circuits where the same handset (i.e. microphone and receiver) is used the method described in Annex 2 above in this volume is still found to be satisfactory.

2. Some experimental results by free conversation methods

The results are given below of some experiments conducted to determine the manner in which the effectiveness of a complete telephone communication falls off as physical degradation is introduced. The circuit used consisted of two local telephone circuits with limiting lengths of subscribers line (similar to those supplied by the British Post Office for the 8th, 9th and 10th series of Experiments at the C.C.I.F. Laboratory) used in the presence of 60 db (Hoth spectrum) room noise at each end. They were connected by a 600 ohms attenuator which was varied. The results are expressed as effectiveness (by several criteria) as a function of attenuator setting.

Two of the experiments used " free conversation " methods with untrained subjects while the third was an ordinary articulation tests using trained crew members. The first " free conversation " experiment A employed untrained, non-technical subjects chosen from outside the Dollis Hill Research Station. These consisted partly of ordinary subscribers and partly of staff from non-engineering departments of the Post Office (e.g. Savings Bank, Postal Departments, etc.). The telephonic ability of this population probably resembles quite closely that of the general public.

The second " free conversation " experiment employed untrained subjects B who work at Dollis Hill Research Station but who are otherwise unconnected with speech investigations. The telephone ability of this population appears from the results to be slightly superior to A.

2.1 *Observations of the performance of users*

A sample of 1732 from population A was employed in the first experiment.

Each pair of subjects was seated in the two cabinets and they were rung by an operator who told them that they were connected together and asked them to converse. Curve *a* of Figure 1 shows the proportion of pairs of subjects who were able to make contact with each other, if necessary after reassurance by the operator that they were indeed connected.

Also plotted for this population is Curve *b* which shows Message Rate Efficiency. After the freely conducted conversation the subjects were given a defined task to perform over the telephone circuit (actually the solution of a pictorial puzzle which required verbal cooperation). They were timed completing this task ; Message Rate Efficiency is the ratio of time they would have required had they used a perfect circuit, to, the time they actually required using the given circuit. The former time was estimated from further observations using the same pair of subjects to complete a similar puzzle over a direct air path.

The second experiment employing about 50 subjects from population B yielded percentage " no-mental-effort " (Curve *c*) and word articulation (Curve *d*). These observations were conducted after the subjects had each conducted a Message Rate test as described above. The " no-mental-effort " test consists of one subject reading a list of short sentences each conveying a simple idea while the other subject listens and indicate whether or not he has understood the meaning " without mental effort " i.e. without having to hesitate and think. The proportion of sentences understood without mental effort gives the " no-mental-effort " percentage.

The same subjects then performed an articulation test using English single-syllable words each in a carrier phrase. There was no control of vocal level or of the manner in which the handset was used.

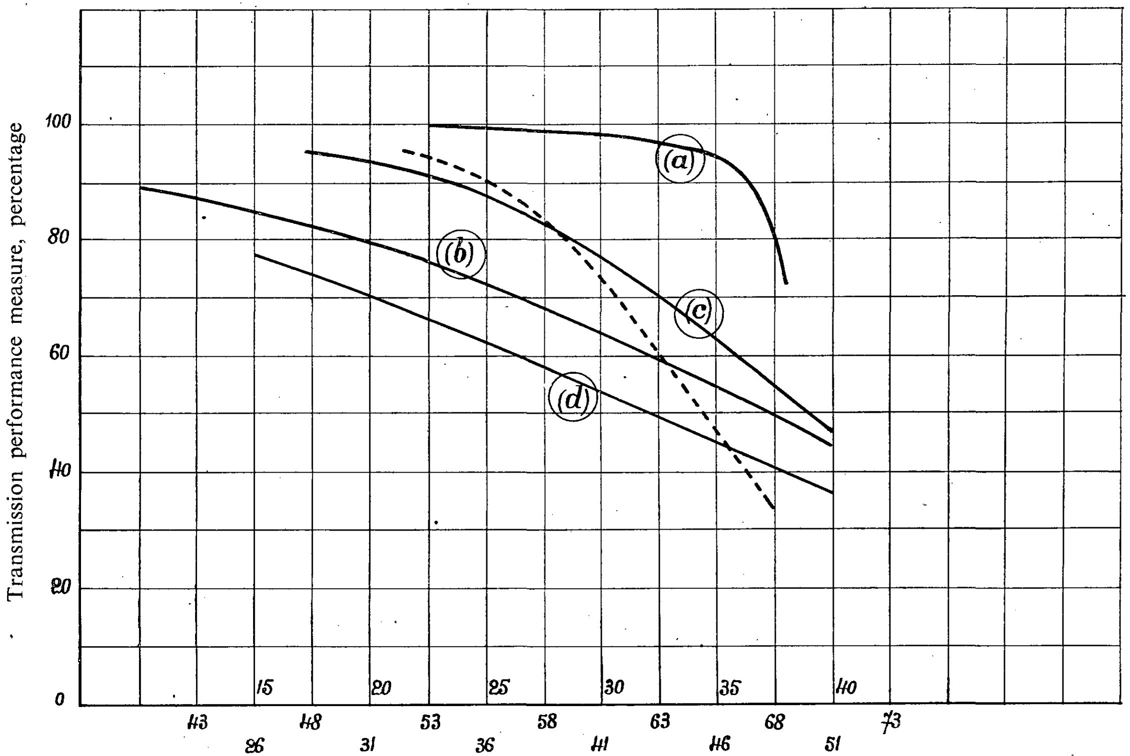
The dotted curve of Figure 1 shows the word articulation obtained in a third experiment in which a trained crew (population C) used a controlled vocal level and fixed lip position relative to the handset. This curve is generally much higher than the corresponding curve for untrained subjects but falls below it at the largest attenuation settings because the untrained subjects will increase their vocal levels and possibly talk closer to the mouthpiece as the circuit becomes more impaired thus to some extent compensating for the increasing degradation.

2.2 *Subjective opinion furnished by the users*

In addition to the objective observations of the performance of the users at the various tasks outlined in § 2.1 above, the subjects were asked to express their subjective opinions of the circuits they had just used.

In the second experiment using population B the subjects were required to classify the circuit over which they had performed the Message Rate task as GOOD, FAIR, POOR or BAD. These classifications are explained on Figure 2. These opinions were expressed by the subjects before they performed the " no-mental-effort " task. The results are given in Figure 2. The lowest curve shows

Circuit :—two British local telephone circuits in 60 db (Hoth) room noise; the circuits are connected together by an attenuator whose setting is given in the abscissa scale.



The lines plotted represent:

(*a*) Percentage able to make contact (non-technical subjects from outside Dollis Hill).

(*b*) Message rate efficiency (non-technical subjects from outside Dollis Hill).

(*c*) Percentage " no mental effort ". (untrained Dollis Hill subjects).

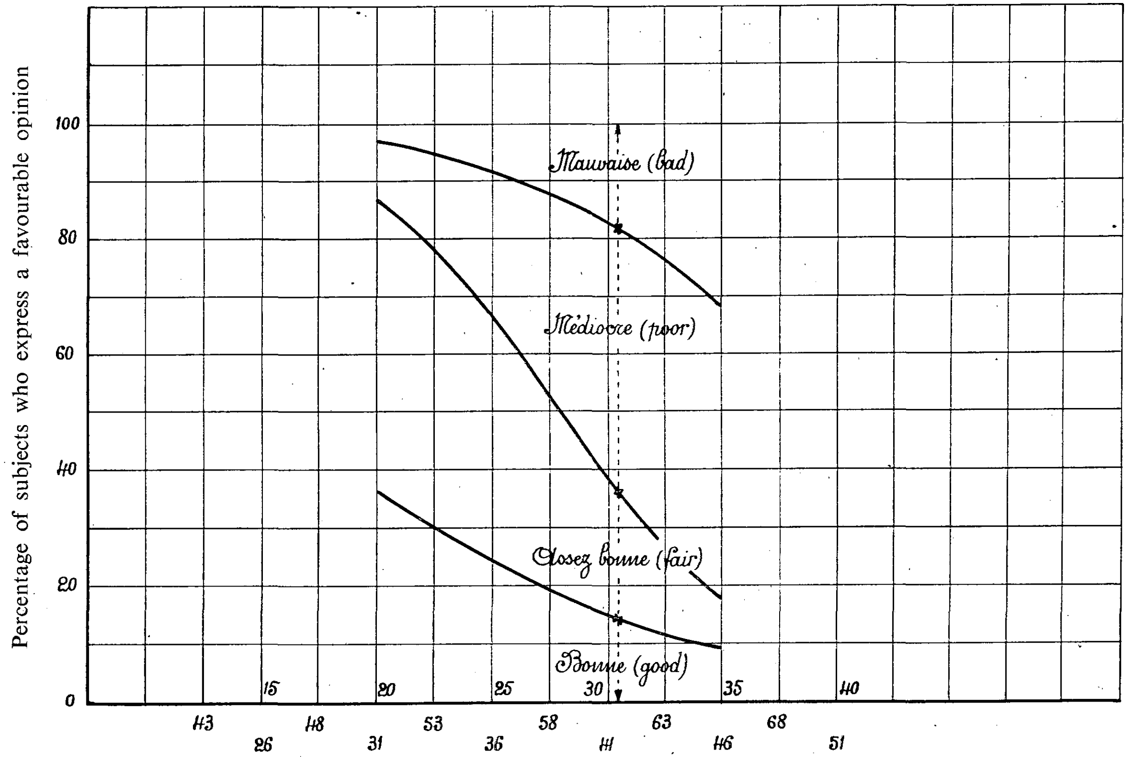
(*d*) World articulation (untrained Dollis Hill subjects, free vocal level).

Word articulation (trained crew, controlled vocal level).

← Junction attenuator, db
← A.E.N., db
← Reference equivalent, db

FIGURE 1. — *Various measures of transmission performance as functions of overall Transmission performance rating*

Circuit :—two British local telephone circuits in 60 db (Hoth) room noise; the circuits are connected together by an attenuation whose setting is given in the abscissa scale.



Each circuit was classified by the users after a Message Rate test.

GOOD No appreciable effort required.

FAIR Some effort required in conversing.

POOR Conversation possible but much difficulty on unfamiliar words.

BAD Usable only with extreme difficulty.

FIGURE 2. — *Subjective opinion percentage as a function of overall A.E.N.*

the percentage who expressed GOOD opinions as a function of attenuation setting; the middle curve shows the percentage who expressed GOOD or FAIR opinions; and the upper curve gives the percentage who expressed GOOD, FAIR or POOR opinions.

2.3 *Discussion of the results*

The experimental plotted on Figure 1 and 2 can be applied to the problem of definite limits of transmission. For this purpose it is convenient to consider also the scales of Reference Equivalent and A.E.N. which have been added.

The experimental results do not indicate any definite limit beyond which it can be stated that telephone conversation is impossible; indeed the most conspicuous feature of most of the data is the relatively gradual decrease in effectiveness. Nevertheless an attenuator setting in the range from 25 to 30 db does seem a reasonable choice for an ultimate limit of transmission performance. Thus a setting of 27 db corresponding to a Reference Equivalent of 38 and an A.E.N. value of 58 db would ensure that substantially all population A users were able to make contact with each other; such users would then be able to converse at a Message Rate Efficiency of about 70% i.e. they would require 40% longer to transmit a given set of messages than they would have required in direct air-air conversation. This efficiency may not seem very high but even a 10 db improvement increases the Message Rate Efficiency only to 84%.

Reference to Figure 2 (which applies to a sample from a population of users having slightly higher telephonic ability than the general telephone public) shows that a setting of 27 db would be regarded as BAD (though still usable) by only 12% and as many as 20% would regard it as GOOD. An improvement of 10 db would reduce the proportion of BAD opinions to about 2% but no less than about 55% would still consider the circuit as other than GOOD. The transition from POOR to FAIR is more critical so that a 27 db setting would result in 55% giving an opinion FAIR or GOOD while 10 bd improvement would increase this to about 90%.

The ultimate limit of transmission performance discussed above would apply to the admittedly rare connection which is at the planning limit and which is used in the presence of the rather high level of room noise of 60 db. The transmission performance of the most common type of international telephone connexion is likely to be about 18 db better than the limiting value. Taking into account the fact that the most probable level of room noise is at least 10 db lower than the value taken above is equivalent to further reducing the A.E.N. value by at least 10 db.

Assuming these figures, the effectiveness of the most probable (model) overall connexion when planned to an A.E.N. limit of 58 db would therefore be given by reading off the curves of Figures 1 and 2 at an A.E.N. value of about 58-18-10 = 30 db. This lies beyond the range of these results but clearly corresponds to an entirely satisfactory connexion.

3. Other applications of free conversation assessment methods

Section 2 above relates mainly to the establishment of a transmission limit. The same methods may, however, be applied to the problem of rating one telephone circuit in terms of another.

ANNEX 25

METHOD USED BY THE SWISS TELEPHONE ADMINISTRATION FOR THE DETERMINATION OF TRANSMISSION QUALITY BASED ON OBJECTIVE MEASUREMENTS

Summary. — The Research Laboratory of the Swiss Telephone Administration, in agreement with the C.C.I.F. Laboratory at Geneva, undertook to complete by some objective measurements the subjective articulation measurements which the latter had made on certain telephone apparatus. The object of these measurements was to obtain, by objective methods, the A.E.N. values which were found at the C.C.I.F. Laboratory by subjective methods.

After a description of the objective methods employed, the author compares the results thus acquired with those obtained during the 8th, 9th and 10th Series of tests at the C.C.I.F. Laboratory and finds a satisfactory agreement.

1. Principle of the calculation of articulation

H. Fletcher and R. Galt describe a method which allows of the calculation of the articulation of a transmission system from its physical characteristics (1). This method is applied to the articulation calculations presented below and a summary description is given of its essential features.

In the calculation of articulation if has been observed that it was desirable to establish an " articulation index ". This is based on the supposition that each frequency band of a speech sound makes a contribution to the articulation independently of the other bands. The sum of these different contributions gives the total articulation. In their article, Fletcher and Galt establish the relationship between the articulation index "A" and the sound articulation " s ". They deduce therefrom the following expression (see Figure 1) :

$$s = 1 - 10^{-\frac{A \cdot p}{0.55}}$$

where " p " is a " practice factor " which, for a normal manner of talking and listening, can generally be considered as equal to 1.

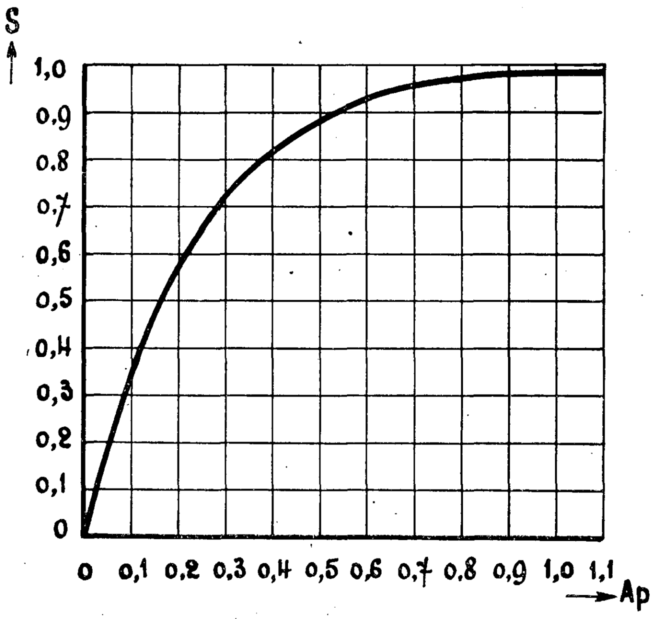


FIGURE 1. — *Relation between the sound articulation and the articulation index*

To find the important weighting factor indicating in what measure each frequency band contributes to the articulation, the articulation index for an ideal transmission system was examined for various cut-off frequencies. If A_f represents the greatest articulation index possible which occurs when the tests are made at the optimum level received, the derivative $\mathrm{d}A_f/\mathrm{d}f$ gives the required weighting factor D. We have, therefore, the relation :

$$A_f = \int_0^f D.\mathrm{d}f$$

The article in question indicates furthermore, that the articulation index of any transmission system is composed of four factors :

$$A = V.E.F.H.$$

The two factors V and E are functions of the received level above that of the threshold of hearing and take account of the change in A when supplementary attenuations or gains are inserted in the system. V takes account of the fact, that, when the level decreases, ever increasing numbers of vocal components fall below the threshold of hearing and can no longer make any contribution to the articulation. When room noise is present a supplementary masking effect is produced. If the energy spectrum of the room noise at the entry of the ear is known, it is possible, with the help of the " critical bandwidth ", to calculate the rise in the threshold of hearing.

If the received level exceeds a certain value the high acoustic pressure fatigues the ear. This fatigue is expressed by the factor E. The influence of these two effects has been examined in articulation tests and the values of V and E are shown in the tables as a function of the received level.

The factor F depends uniquely on the shape of the frequency characteristic. It is a maximum value and indicates the influence on the articulation index when the received level for this system has been so chosen that the articulation is greatest. This optimum received level is at 68 db. The value of F is 1 when the characteristic is flat and is included between 0 and 1 for any other shape of curve. This value is given by the following equation :

$$F = \int_0^\infty D.W.\mathrm{d}f$$

D representing the weighting factor. The factor W determines the reduction of dA in the interval df when the received level is less than optimum. It has been possible to obtain the function W in an ideal manner by means of articulation measurements carried out on a system for which the level of each frequency interval could be separately adjusted. The other frequency bands were only influenced when it again became necessary to bring the received level to the optimum value of 68 db for the total transmission.

Since, during the course of normal mouth-to-ear transmission, there is a modification of the acoustic pressure due to the dimensions of the listener's head, account must be taken of the " orthotelephonic transmission " factor during the tracing of the frequency characteristic. To calculate the articulation it is necessary to take account, at each frequency, of the difference in decibels between the system considered and the orthotelephonic reference system.

The fourth factor H on which depends the articulation index includes all the effects which are not included in the three other factors. By this is meant certain kinds of non-linear distortion, misalignment of frequencies in carrier systems, etc.

The object of the present work is to calculate a curve which gives the relation existing between the articulation of a system and the attenuation introduced in this system. This allows the calculation, on an objective basis, of the A.E.N. value introduced by the C.C.I.F. as a new quantity for the evaluation of the quality of a telephone system. This new quantity is based on the comparison of a commercial telephone system with a reference system and is consequently independent, to a large extent, of the characteristics of the testing crew. It is not absolutely necessary, for the calculation, to know the training factor and the hearing acuity. The absolute value of the articulation will therefore vary but the difference in the attenuation as defined will remain constant.

To calculate the A.E.N. value it is therefore necessary to know the following quantities :

1. Frequency characteristics of the reference system and of the telephone system.
2. Speech power at the microphone.
3. Interfering noise at the entry of the ear.

2. Measurement of the frequency characteristics of a carbon microphone

It is difficult to make an exact and reproducible measurement of the frequency characteristic of a carbon microphone, since the position of the carbon granules is not defined in an absolute manner and may vary to an appreciable extent according to the nature of the acoustical excitation. This is why the methods of measurement known at the present time often give very different results.

In principle, these objective measurements may be divided into two groups :

1. Measurement of the frequency characteristic using a continuous spectrum.
2. Measurement of the frequency characteristic using pure tones.

In the two cases, the microphone is excited acoustically, in the position indicated by the C.C.I.F. recommendations, by means of an artificial mouth.

2.2 *Measurement of the frequency characteristic using a continuous spectrum*

For the measurement of the frequency characteristic using a continuous spectrum, a noise generator is used as a source of energy. The acoustic pressure generated is thus composed of all the frequencies of the band concerned simultaneously. The spectral composition of the output voltage of the carbon microphone is determined with the aid of a noise analyser. When this noise spectrum is known it is easily possible to calculate the frequency characteristic of the carbon microphone.

Since the energy supplied by the voice extends simultaneously over a large number of frequency bands this form of excitation corresponds to normal working conditions.

Although this method appears to be clear, it presents, however, a fundamental difficulty. The relation between the contact resistance of the carbon granules and the acoustic pressure in not linear. Hence, when the microphone is excited by several frequencies, its distortion gives rise to new frequencies. If, in any one of the bands, the sum of the amplitudes of these supplementary frequencies is greater than the useful voltage of that band, the measurement of the frequency characteristic is rendered inaccurate.

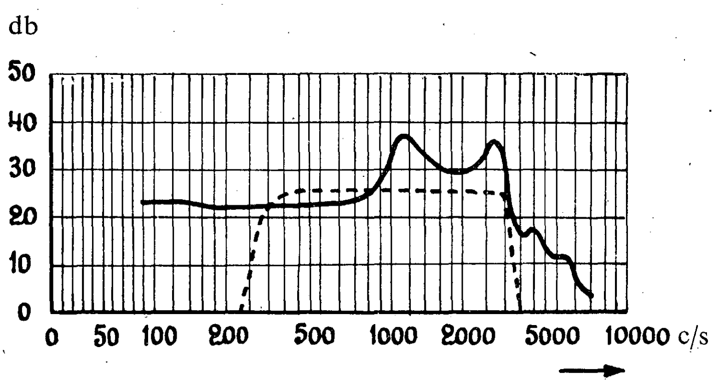


FIGURE 2. — *Response curve for microphone A (dotted curve indicates the energy spectrum of the sound)*

In order to determine for what proportion of cases this is true for commercial carbon microphones, the following experiment was made : by means of an artificial mouth two types of microphone, A and B, were artificially excited. The sound source used was a noise generator having a flat frequency characteristic. With the aid of filters its bandwidth was restricted to 280-3400 c/s. The effective acoustic pressure on the diaphragm of the microphone was 11.6 dynes/cm². The two microphones were measured while fitted in a telephone set.

Figures 2 and 3 show the frequency characteristics obtained.

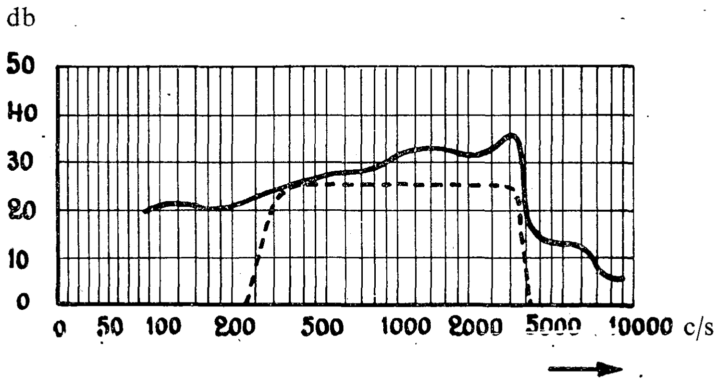


FIGURE 3. — *Response curve of microphone B (the dotted curve indicates the energy spectrum of the sound)*

Since no fall in sensitivity occurs below the pass band, it may be concluded that, due to the distortion in the carbon microphone, the sum of the modulation products in the range of the lower frequencies is greater than the useful voltage. This appears particularly clearly for microphone A where the energy per cycle remains constant below 500 c/s. At the high frequencies, the harmonics and the supplementary frequencies are also noticed, but they are less troublesome than at the low frequencies.

Since the intermodulation factor of a carbon microphone lies between 5 and 30% it cannot be expected to obtain with the noise analyser, as experience has confirmed, differences of level greater than 10 to 25 db in the pass band. If it was desired to measure greater fluctuations in the frequency characteristic of a carbon microphone with the aid of a continuous spectrum noise and an analyser, it would be necessary to modify the spectrum of the acoustic pressure. For those frequencies near which the sensitivity of the carbon microphones is small, it would be necessary to increase the acoustic pressure so as to reduce the effect of those frequencies in the sensitive range.

2.2 *Measurement of the frequency characteristic using pure tones*

For the measurement of the frequency characteristic of carbon microphones with pure tones use is made of a heterodyne oscillator whose frequency is varied by means of an automatic device and whose output voltage remains constant throughout the whole range of frequencies. The requirements imposed, by this method, on the frequency characteristic of the attificial mouth are not great. Since at any given instant there is only one sound, the pressure may be monitored by means of a standard microphone and maintained constant with the aid of an amplifier with automatic gain control.

According to the time necessary to cover the given frequency range a distinction may be made between a slow sweep or a rythmic variation of frequency.

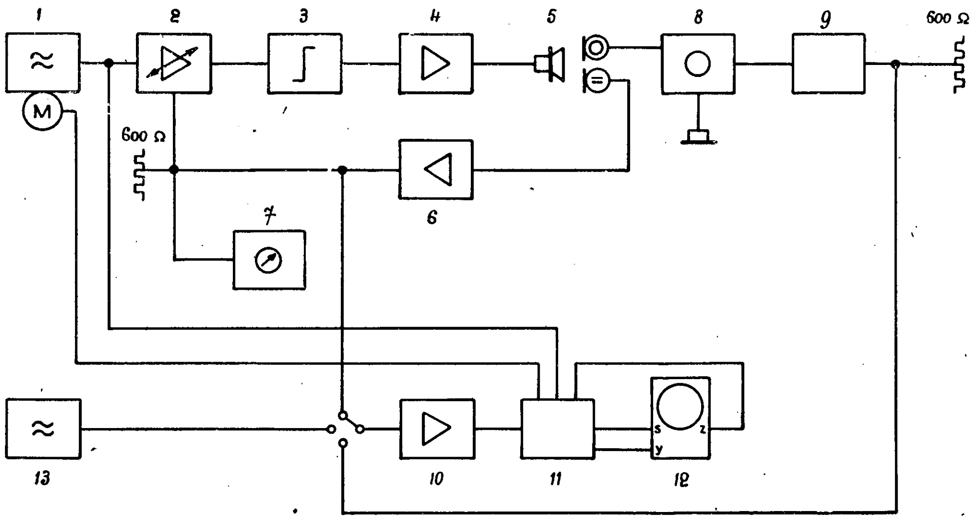
For the measurement of the frequency characteristic using a slow sweep the frequency generator constructed according to C.C.I.F. recommendations is used. The frequency range of 30 to 10,000 c/s is swept automatically in 121 seconds. Since the sweep is slow the instantaneous variations in voltage remain small and a mechanical instrument can be used for recording the characteristic. The objection to the slow sweep is that, during the acoustic excitation, the carbon granules of the microphone have time to rearrange themselves. The microphones become " packed " and entirely lose their sensitivity (see Figure 10).

For the measurement of the frequency characteristic using a rhythmic variation of frequency the oscillator should be set up for a rapid sweeping of the frequency band. The frequency with which the band is swept lies between 1 and 5 c/s. For this reason, the instantaneous variations of voltage are large. A level recorder can therefore no longer be used and it is necessary to record the frequency characteristic by means of a cathode ray oscilloscope. However, in order to measure large differences of level this is preceded by an amplifier whose output voltage is the logarithm of its input voltage.

A measurement device is described below which has proved very serviceable for the measurement of the frequency characteristics of carbon microphones. Figure 4 shows the overall block diagram of an arrangement used for measuring the sending loss of a subscriber's telephone set.

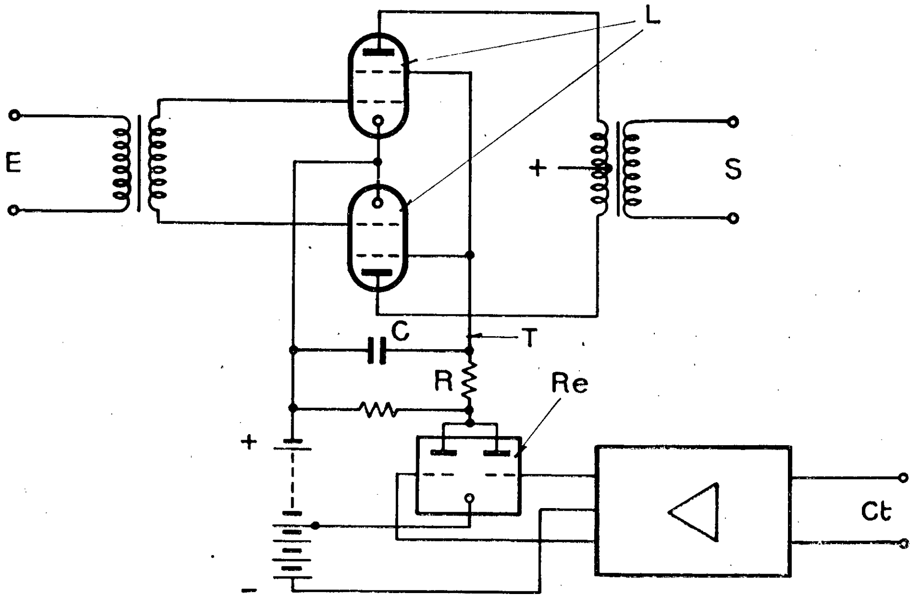
For the rhythmic generator a heterodyne oscillator having a frequency range of 50 to 10,000 c/s was used. The frequency scale is divided logarithmically and the frequency can be adjusted by means of a precision condenser that can be easily rotated. This condenser has a working angle of about 230° but may be turned through 360° and is directly coupled to the shaft of the driving motor.

As it is indispensable to have an acoustic pressure which is as constant as possible, an amplifier with automatic volume control is used which compensates the variations of the frequency characteristic of the artifical mouth. The choice of a time constant for this amplifier presents certain difficulties. On the one hand it is necessary to compensate for the rapid variations and, on the other hand, to provide for the passage of the low frequencies without distortion. An experiment showed that the time constant for one sweep per second ought to be about 0.05 seconds. If, in addition, the degree of regulation is fixed at 3.3% so that all the variations are reduced to one-thirtieth of their value, the lowest measuring frequency which remains stable and without distortion is about 200 c/s. It is to be noted that the rectifier providing the control voltage is of the push-pull type (frequency doubling) and that the time constants are the same in both directions of control. The schematic diagram of the automatic volume control amplifier is illustrated in Figure 5.



1 = Rhythmic oscillator
2 = AVC amplifier
3 = Equaliser
4 = Amplifier
5 = Artificial mouth
6 = Amplifier
7 = Volume meter
8 = Subscriber's telephone set
9 = Feeding bridge
10 = Logarithmic amplifier
11 = Impulse generator for synchronisation and frequency ordinates
12 = Cathode ray oscillograph
13 = Sine-wave generator
M = Driving motor

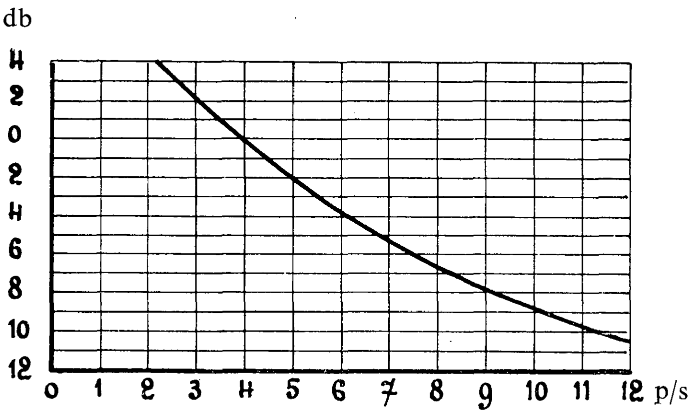
FIGURE 4. — *Arrangements for the objective measurement of sending sensitivity*



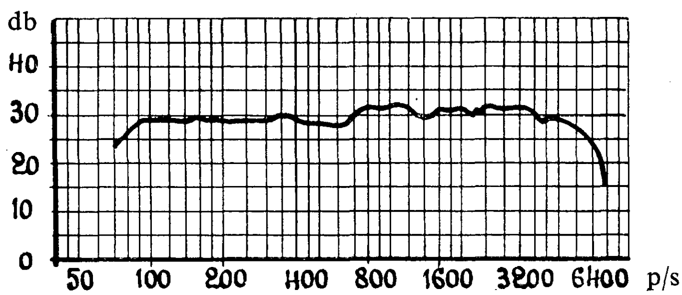
T = A.V.C. voltage
L = Control valves
Ct = Input for the controlling voltage
Re = Rectifier
E = Input
S = Output

FIGURE 5. — *Schematic diagram of the regulating amplifier*

The artificial mouth which generates the acoustic field has already been described in a previous publication (2). The principal details may be summarised as follows : the orifice of the mouth, as determined by empirical means, is a form of short mouthpiece having a rapidly increasing diameter and so preventing the formation of very pronounced beams at high frequencies : it offers no plane surfaces, which therefore prevents the formation of stationary waves between the microphone to be measured and the artificial mouth. Figure 6 gives the properties of the artificial mouth provided with an amplifier and an equalizer.



1. Decrease of the mean acoustic pressure along the axis



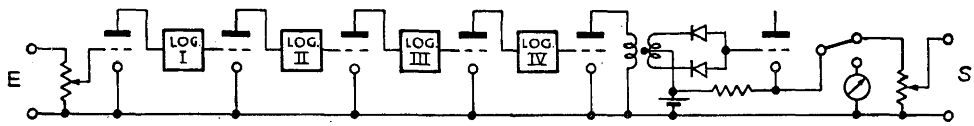
2. Response curve

FIGURE 6. — *Characteristics of the artificial mouth*

As is shown in Figure 4, the microphone is measured in a normal circuit with the subscriber's telephone set and the usual exchange feeding bridge. The terminating impedance is 600 ohms.

Unlike the usual logarithmic amplifiers which have a time constant, the amplifier used provides an instantaneous logarithmic amplification. At any instant the output voltage corresponds to the logarithm of the input voltage. It is thus possible to make an exact study of the large and rapid variations which occur during the measurement of microphones using rhythmical frequency variation.

In the amplifier in question the logarithmic function is obtained by approximation from linear sections. Between the various stages of amplification linear networks are inserted which are composed of pure resistances and crystal diodes with predetermined priming voltages. Figure 7 gives an overall diagram of the amplifier and Figure 8 shows the connection of the coupling units.



E = Input S = Output

FIGURE 7. — *Basic principles of the logarithmic amplifier*

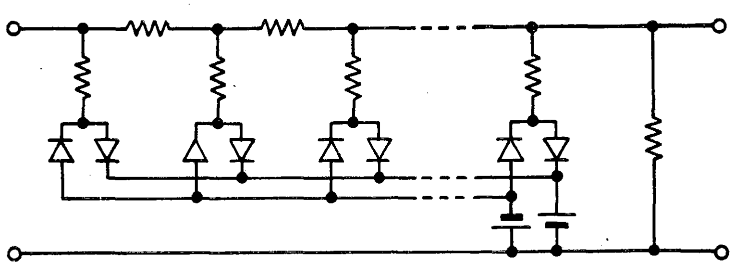


FIGURE 8. — *Coupling circuit (combination of linear elements)*

With the push-pull rectifier it is possible to apply both half waves to the same side of the neutral. If the envelopes of the two half waves coincide it can be said with certainty that even-order non-linear distortions are small. The distance between the two envelopes is thus a measure of these distortions and the measuring device in question allows not only the determination of the frequency characteristic of a carbon microphone but also the evaluation of its non-linear distortion. In order to render the curve of the frequency characteristic visible on the screen of the cathode ray oscilloscope, use may be made of a special setting of the amplifier which is provided for this purpose whereby the rectified voltage may be smoothed by means of a condenser.

The electrical characteristics of the logarithmic amplifier are as follows:

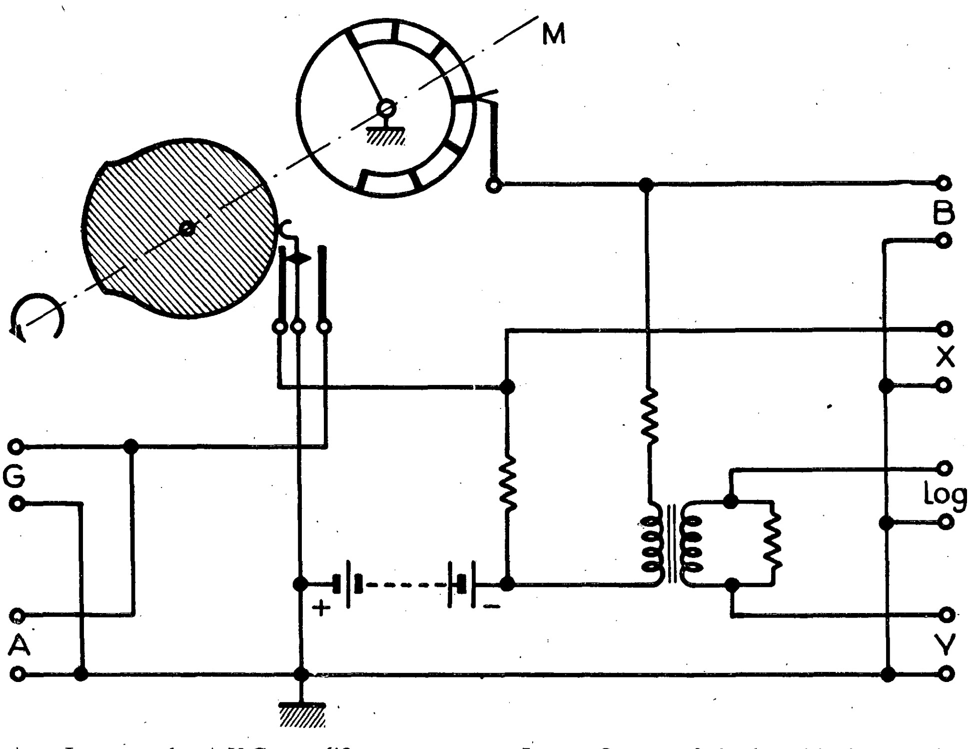
— Range of amplification without input potentiometer	50 db
— Accuracy of the logarithmic indications	±0.3 db
— Input potentiometer calibrated in steps of	5 db
— Input resistance	100,000 ohms
— Voltage with input potentiometer	0.7 mV to 70 V
— Rectified output voltage for the cathode ray oscilloscope	0 to 5.5 V
— Frequency range	30 to 50,000 c/s

The positions of the switch are as follows :

1. Direct indication on an instrument up to 50 db.
2. Connection of an external instrument consuming 5 mA (e.g. a recording device).
3. Direct output of the amplifier giving the instantaneous logarithm of the input voltage for the modulation of the cathode ray oscilloscope.
4. Rectified output from amplifier.
5. Rectified and filtered output for the modulation of the cathode ray oscillograph.

The calibrated potentiometer connected to the input of the amplifier allows the checking at any time of the accuracy of the logarithmic indications.

The indicating instrument used is the cathode-ray oscillograph with a direct-current amplifier. Figure 9 shows how the synchronisation of the time base and the frequency marks is arranged.



A = Input to the A.V.C. amplifier
B = Modulation of the spot intensity
G = Output of the rhythmic generator
Log = Output of the logarithmic amplifier
M = Shaft of the driving motor
X = Synchronisation of the time base
Y = Plates of the oscillograph

FIGURE 9. — *Generator of pulses for the synchronisation of the cathode-ray oscillograph and for the frequency ordinates*

The motor shaft of the rythmic generator also rotates two discs with cams and contact springs. The first disc operates a contact which makes connection to the output of the oscillator during its useful angle and short-circuits it during the remainder of that angle. In addition it supplies a synchronising pulse for the start of the time base of the cathode ray oscillograph. The second disc with the contact cams generates short pulses at certain adjustable angles of the shaft. By means of a transformer these pulses are superimposed on the useful voltage applied to the Y plates and give rise to visible frequency ordinates on the screen of the cathode ray oscillograph. If the cathode ray oscillograph is provided with the facility of modulating the spot intensity by means of an external voltage the frequency ordinates can be made even more prominent.

As has been shown by experience the best method is to sweep the frequency band from the high frequencies to the low frequencies. The carbon granules which are acoustically excited at the resonant frequencies of the microphone require some time to return to their rest position. It is for this reason that the method of measurement by rythmic variation of the frequency allows the exact recording of the steep flanks of the frequency characteristic only if the change is from the frequency at which the sensitivity is lowest to that at which the frequency is greatest. Since, for the majority of carbon microphones the upper cut-off frequency is vey marked, it is best to start the acoustic excitation at the high frequencies.

An accessory apparatus enables the photographing of the frequency characteristics shown on the screen. The calibration of the Y axis is accomplished with the aid of the potentiometer of the logarithmic amplifier. When taking the photograph of the frequency characteristic the 5 db scale markings are immediately superimposed. Even the smallest errors in the logarithmic indication are thus nullified and the frequency characteristic can be measured very exactly on an enlargement of the photograph.

Figure 10 shows the frequency characteristic of a microphone measured using a show frequency sweep. Figure 11 shows the frequency characteristic of the same microphone on the screen of the cathode ray oscilloscope. It can clearly be noticed that with the first method the microphone " packs " between 400 and 1400 c/s. With the other method the resonant frequency reactivates the carbon granules and the microphone recovers its normal sensitivity.

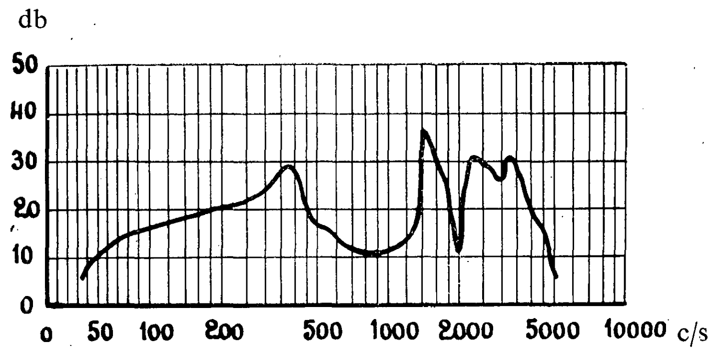


FIGURE 10. — *Response curve of a carbon microphone measured using a slow frequency sweep*

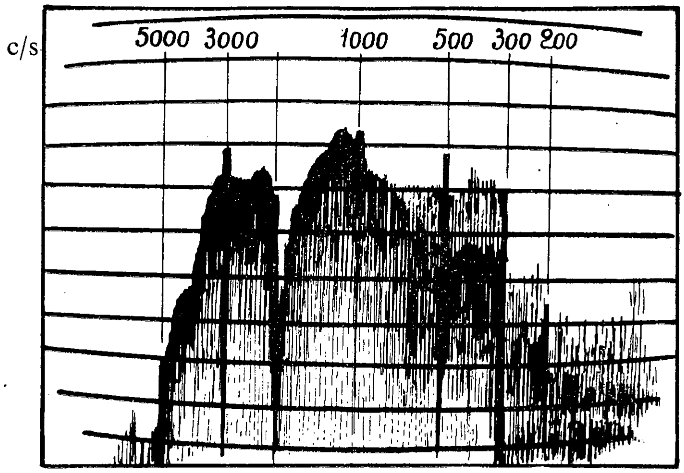


FIGURE 11. — *Response curve of the same microphone measured using a rhythmic oscillator*

3. The artificial ear used for the measurement of the telephone receivers

In the case of modern telephone receivers of high efficiency and whose mechanical impedance is relatively small it is noticed that the frequency characteristic is greatly dependent upon the acoustic loading. Receivers of this kind must therefore be measured by means of an artificial ear whose acoustic properties correspond to those of the human ear. Moreover, for the calculation of articulation it is indispensable to know the absolute acoustic pressure at the entry of the human ear.

H. Weber (3) has described an artificial ear whose acoustic properties have already proved themselves. In consequence, the basic principles of its construction were adopted for the development of the new artificial ear.

This new artificial ear is also composed of two cavities which are interconnected by a heavily damped passage. For frequencies above 1000 cps an acoustic impedance is thus obtained which is greater than that corresponding to the total volume of the two cavities. In addition the interconnecting passage contributes an acoustic resistance.

The first cavity between the condenser microphone (Western Electric type 640 AA) and the opening of the artificial ear has a volume of 1.5 cm^3. The volume of the circular cavity to which the first is coupled is 1.4 cm^3. The interconnecting passage is formed of four 45 degree segments of a ring having internal and external diameters of 19 and 30 mm and a thickness of 0.16 mm. The total volume at low frequencies is therefore 3.0 cm^3 and about 1.5 cm^3 for the high frequencies. To these volumes should be added, in each case, the volume included between the receivers ear-cap and the outer surface of the artificial ear.

The outer form of the artificial ear has been so chosen as to correspond, for the forms of receiver car-caps considered, as exactly as possible to the human ear. For this purpose account was taken of the results of the measurements made by K. Braun (4). As some ear-caps have a large diameter opening, the aperture of the new artificial ear has been given a conical form. The new form of construction has permitted the extension of the field of application and the enlargement of the frequency band. Figure 12 shows the frequency characteristic of a standard receiver measured with the new artificial ear.

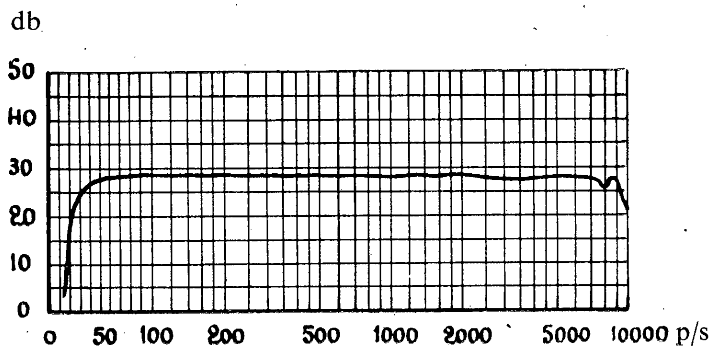
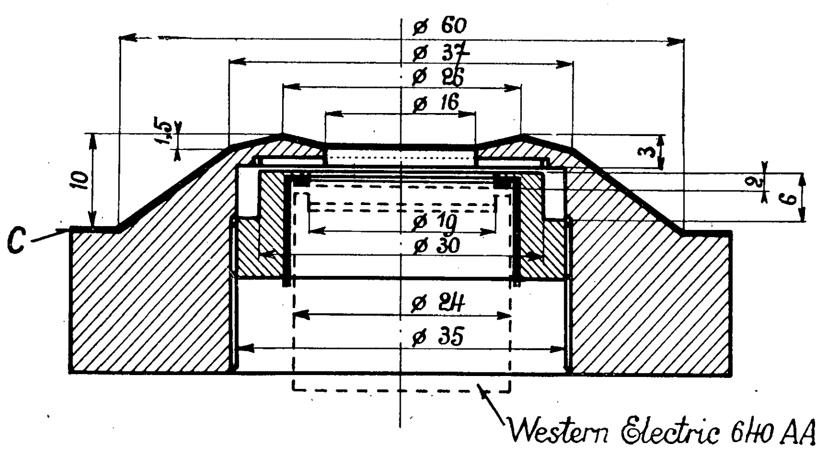


FIGURE 12. — *Response curve of a standard receiver measured with the new artificial ear*

In order to verify the exterior shape, measurements were made of the receiving systems of telephone sets from six different administrations and the equivalent attenuation was calculated as indicated by K. Braun (57). Table I allows the values thus obtained to be compared with those measured subjectively at the C.C.I.F. Laboratory.

The dispersion of these values is within the limits imposed by the precision of the measurements and calculations and the agreement can be considered as very satisfactory.

Figure 13 shows a section through the new artificial ear.



C = Rubber, thickness : 0.5

FIGURE 13. — *Section of the new artificial ear*

TABLE 1

System No.	Form of the receiver ear-cap	Calculated db	Measured in C.C.I.F. Laboratory db	Difference db
1	Thick, conical cavity	2.5	3.1	0.6
2	Large, flat	0.9	2.8	1.9
3	Medium, spherical cavity	6.1	4.9	−1.2
4	Medium, spherical cavity	8.1	7.6	−0.5
5	Thich, conical cavity	3.5	4.5	1.0
6	Medium, conical cavity	1.3	2.5	1.2

4. Determination of the interfering noise at the entry of the ear

The masking effect of a noise has the result of raising the threshold of hearing. Some speech conmponents disappear below the noise and no longer make any contribution to the articulation.

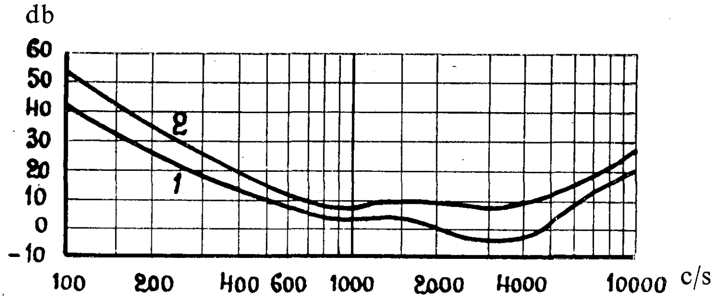
For the purposes of the calculation of articulation, the question must be posed whether, in monaural listening, as is the case with the telephone, only the noise reaching the listening ear influences the articulation, or whether the noise reaching the other ear has also an influence. To check this point the following experiment was carried out. The thresholds of hearing of twelve persons were measured using a telephone receiver. The room noise of 60 phones had a flat frequency characteristic. During the first test the free ear was exposed and during the second test it was covered. Table II shows, in decibels, with reference to 1V, the average voltage for the threshold as measured at the input of the telephone set.

TABLE 2

Frequency	2nd ear		Difference
	open	covered	
p/s	db	db	db
200	−58.7	−59.7	+1.0
400	−70.0	−69.7	−0.3
800	−77.2	−78.8	+1.5
1600	−77.5	−78.5	+1.0
3200	−77.5	−79.7	+2.2

The last column of Table II shows that the noise penetrating into the ear exposed to the room has so little effect that it can be neglected.

The room noise penetrates into the ear covered by the receiver ear-cap by two paths : through the leak between the ear-cap and the ear and by the microphone and the receiver i.e. the side-tone path. If equipment is available for the generation of a room noise and if the persons taking part in the experiment have normal thresholds, the masking effect can be directly determined by a measurement of the threshold of hearing. However, it is often preferable to separate the two paths and to have knowledge of their effects independently of one another. If a probe microphone is not available the acoustic attenuation of the leak between the ear-cap and the ear can be measured by determining the threshold of hearing for pure tones. To this end a loud-speaker provides sound pulses whose amplitudes can be adjusted by means of a precision attenuator. The thresholds determined by this method are very exact. The difference between the values obtained for rising and falling level is only about a decibel. At the input to the loud-speaker a measurement is made of the voltages for the thresholds when the ear is exposed and for when it is covered by the receiver. The ratio of these voltages diminished by the rise in the threshold with one ear covered gives the required attenuation directly. Figure 14 brings out the difference in the thresholds when listening under free field conditions and when listening with the ear covered by a receiver ear-cap (see H. Fletcher and R. Galt, *loc. cit.*).



Curve 1: for mon-aural listening under free field conditions.
Curve 2 : for mon-aural listening with the ear covered by a receiver.

FIGURE 14. — *Thresholds of hearing in decibels above* $10^{-16} W/cm^2$

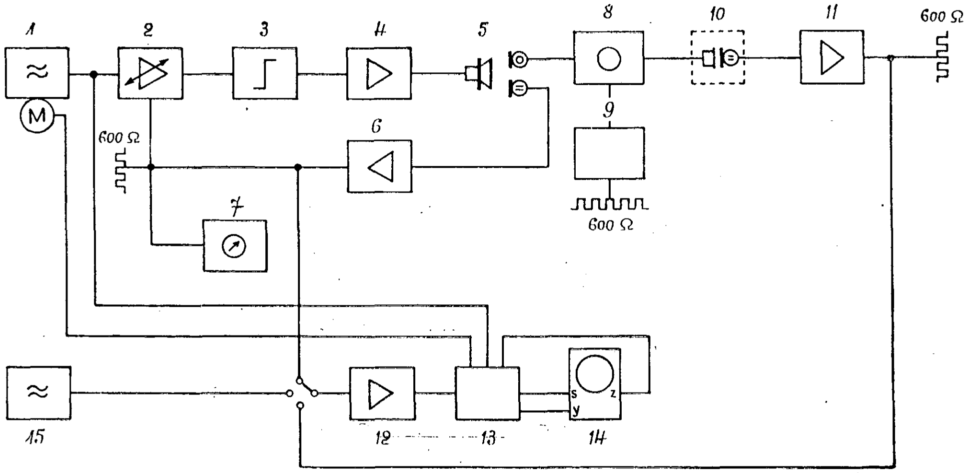
For the calculation of the noise penetrating into the ear through the leak between the ear and the ear-cap of the receiver account will, of course, be taken of the increase in the acoustic pressure by dimensions of the listener's head.

In order to calculate the noise which reaches the ear through the receiver a measurement is made of the attenuation of the side-tone path. This can be measured directly with the aid of the artificial mouth and the artificial ear as shown in Figure 15.

The various items of the apparatus have already been described in § 2. The only difference lies in the fact that the output voltage of the subscriber's set is not required but that the acoustic pressure under the receiver ear-cap is measured by means of the artificial ear. As the attenuation of the side-tone path depends to a large extent on the impedance of the termination to the telephone set the feeding bridge is closed with a resistance of 600 ohms.

Because the carbon microphone suffers from non-linearity, the acoustic pressure generated at the microphone must correspond to the room noise used during the articulation tests. If, in order to exclude interfering noises during the measurement of the frequency characteristic, it is desired to make the measurement using a higher acoustic level, account must be taken, in the calculation, of the change in sensitivity of the microphone.

The change in pressure due to the dimensions of the listener's head is no longer appreciable owing to the distance between the mouth and the microphone. According to our measurements the increase of pressure at a distance of 3 cms from the mouth is about $^1/_3$ or $^1/_2$ of the orthotelephonic increase. This increase is, to a large extent, compensated at high frequencies by the directional effects of the microphones.



1 = Rhythmic oscillator
2 = A.V.C. Amplifier
3 = Equalizer
4 = Amplifier
5 = Artificial mouth
6 = Amplifier
7 = Volume meter
8 = Subscriber's telephone set
9 = Feeding bridge
10 = Artificial ear
11 = Amplifier
12 = Logarithmic amplifier
13 = Impulse generator for synchronization and frequency ordinates
14 = Cathode-ray oscillograph
15 = Sine-wave generator
M = Driving motor

FIGURE 15. — *Arrangements for the measurement of the side-tone equivalent for a subscriber's telephone set*

5. Example of calculation and measurement

A subscriber's telephone system of the Swiss Telephone Administration has been measured under conditions which are identical with those under which the articulation tests were made at the C.C.I.F. Laboratory. The system is com-

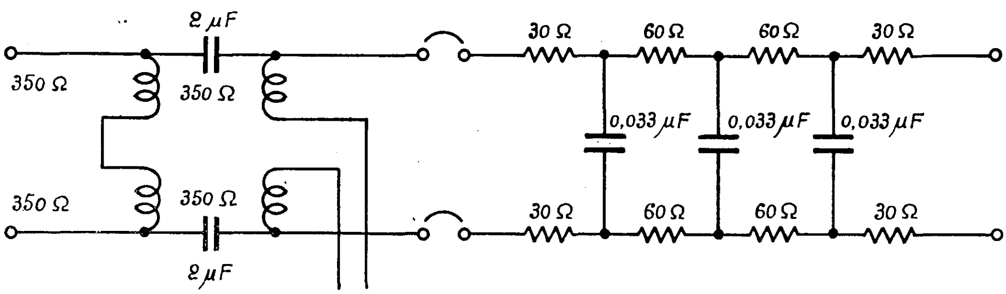


FIGURE 16. — *Circuit of the feeding bridge and the subscriber's line*

posed of three parts : a telephone set, model 1950, a subscriber's line of 3 km and a feeding bridge. Figures 16 and 17 give the circuit diagrams of the arrangements.

With the aid of the measuring device described in § 2, measurements were made on the sending systems of 5 subscriber's sets. The acoustic pressure at the mouthpiece of the microphone was 11.3 dynes/cm². Figure 18 shows, by way of example, the trace obtained on the screen of the cathode ray oscillograph of the reference system and of the subscriber's system. The average sensitivity of the reference system was 26.6 mV/dyne/cm²; from this the sensitivity of the commercial system can immediately be deduced.

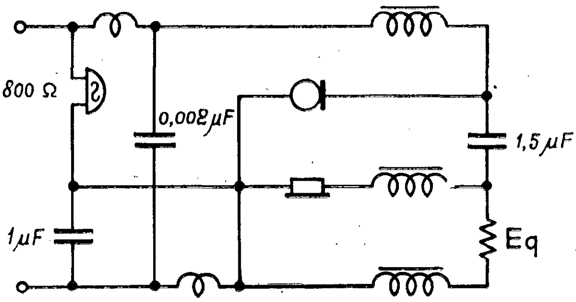


FIGURE 17. — *Circuit of the subscriber's telephone set*

The receiving systems of the five subscriber's telephone sets have been measured by the usual method using an oscillator with a slow sweep, the artificial ear and a level recorder.

As the sending system and the receiving system are measured separately any one system can be combined with another system for the calculation of articulation. The combinations with the reference system used in the C.C.I.F. Laboratory for the articulation measuments (A.R.A.E.N.) ares particularly interesting.

For the four following combinations the transmission qualities have been determined and the articulation then calculated :

	Sending	Receiving
1	R	R
2	CH	R
3	R	CH
4	CH	CH

R = Reference system (A.R.A.E.N.)
CH = Subscriber's telephone system of the Swiss Administration

The 300/3400 c/s band-pass filter of the reference system was inserted in each connection.

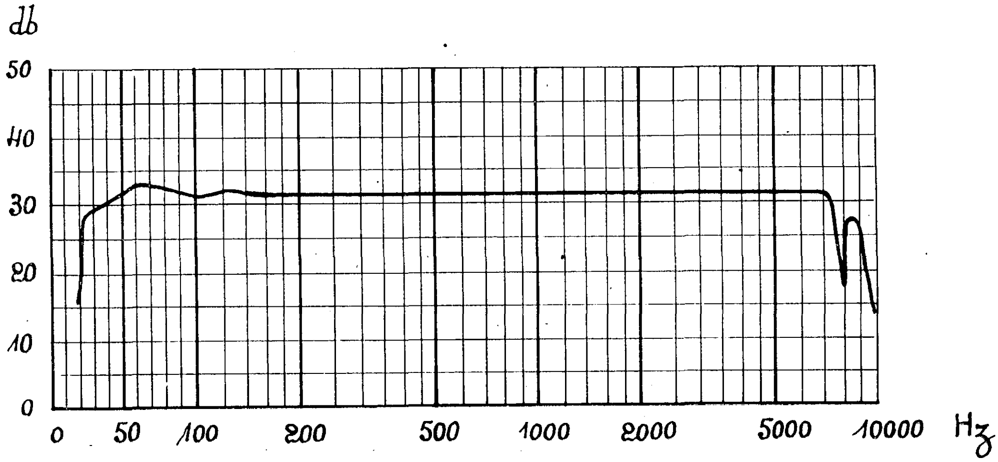


FIGURE 18 *a.* — *Response curve of the reference system*

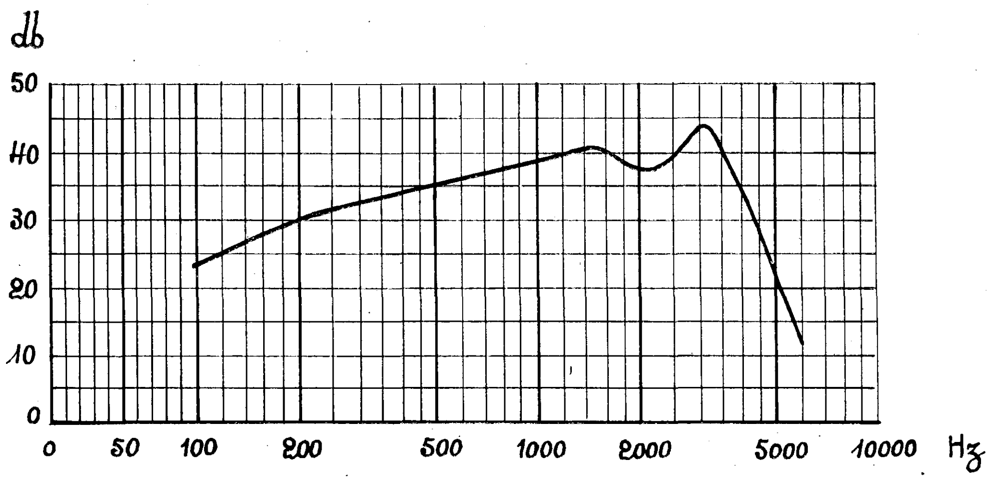
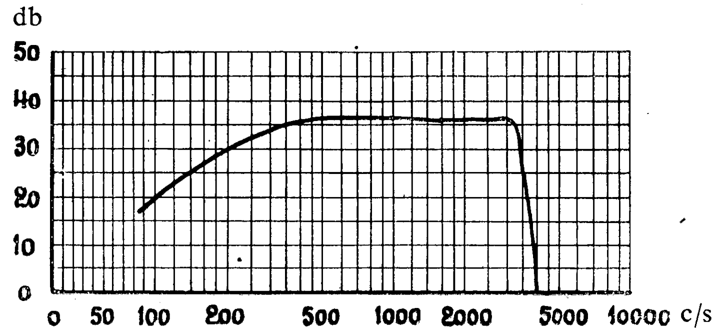


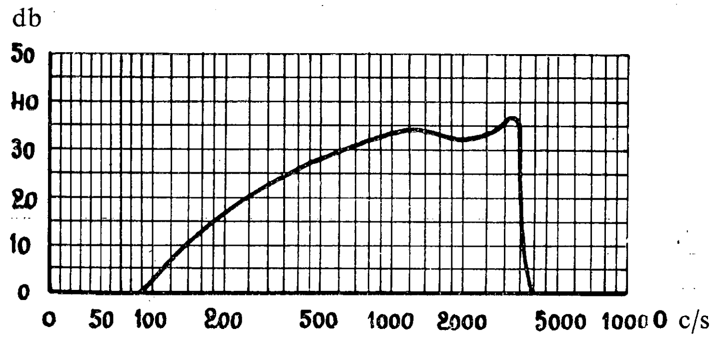
FIGURE 18 *b.* — *Response curve of the commercial system*

The frequency characteristics shown in the Figures 19 to 22 show the differences between these systems and a 1 metre air path with mon-aural listening and an orthotelephonic correction; the latter being taken as a basis for comparison. For the definition of the speaking level it is supposed that the sound pressure at a distance of 33.6 cms from the lips of the talker is 1 dyne/cm^2. It results from this, on the one hand, that for the distance standardized by the C.C.I.F. recommendations, the sound pressure at the mouthpiece of the Swiss handset, model 1946, is 11.3 dynes/cm^2 or 95 db above 10^{-16} W/cm^2. On the other hand we have, at a distance of 1 metre from the lips of the talker, an acoutic pressure of 64.5 db above 10^{-16} W/cm^2. To obtain the acoustic pressure in the ear for the purposes of articulation the last mentioned pressure must be increased by the values in decibels shown in Figures 19 to 22.

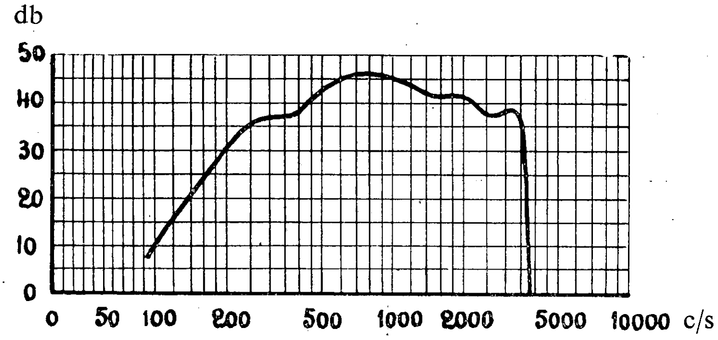
To determine the interfering noise in the listening ear a measurement has been made of the acoustic attenuation of the leak between the ear and the receiver



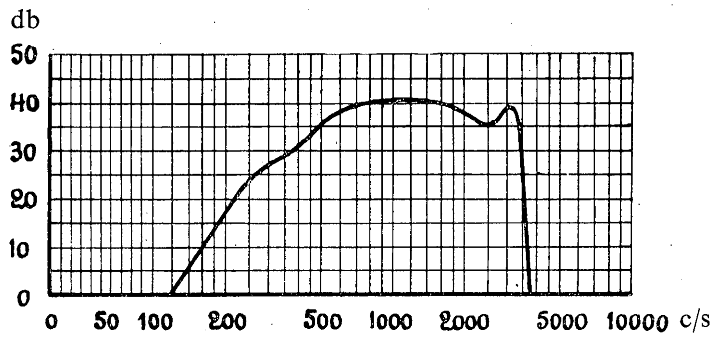
System 1 = R/R



System 2 = CH/R



System 3 = R/CH



System 4 = CH/CH

FIGURES 19 TO 22. — *Response curves of systems 1 to 4*

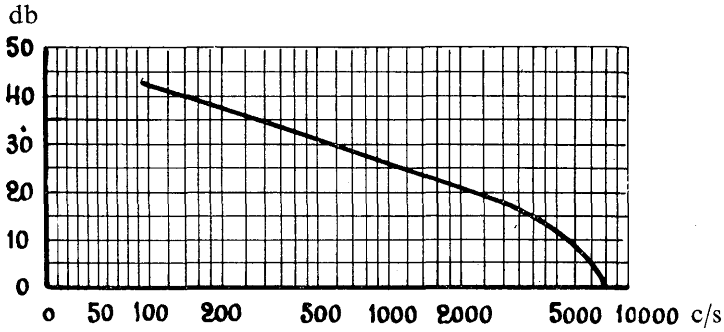


FIGURE 23. — *Energy spectrum of the room noise (according to Hoth) relative to* $10^{-16} W/cm^2$ *per cycle for a level of 60 phons*

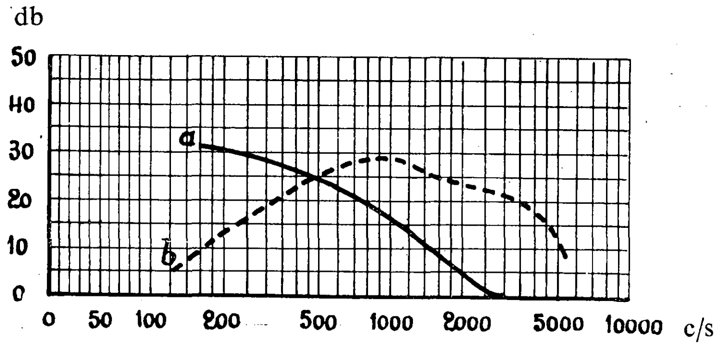


FIGURE 24. — *Interfering noise at the entry of the ear for the reference system*

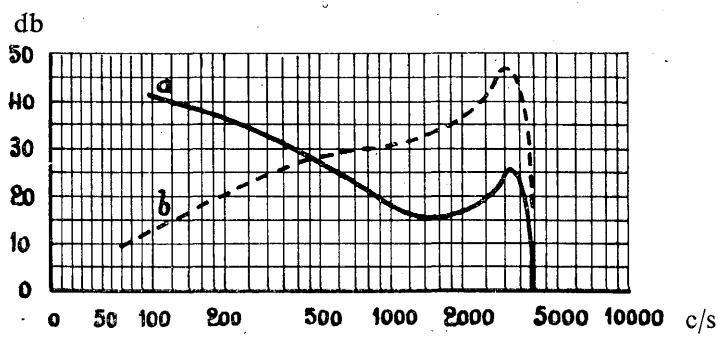


FIGURE 25. — *Interfering noise at the entry of the ear for the commercial system*

by means of a threshold test and by the measurement of the attenuation of the sidetone path, using the measuring arrangement shown in Figure 15.

The room noise generated in the listening room during the articulation measurements made at the C.C.I.F. Laboratory has an energy spectrum based on the observations made by Hoth. The level has been fixed at 60 db (see Figure 23).

The noise resulting from this is shown in Figure 24 for the reference system and in Figure 25 for the commercial system. The curve *(a)* shows the energy distribution per cycle at the entry of the ear and the curve *(b)* the increase in the threshold for speech due to the masking effect, account being taken of the critical bandwidth.

The non-linear distortion of the carbon microphone has been measured by the method using two sinusoidal signals as recommended by the C.C.I.F. The difference between the two frequencies remains constant and equal to 200 c/s. If β_o db represents mean output level due to the two fundamental signals and β_d db the output level of the difference tone, the influence on the articulation index, if $(\beta_o - \beta_d)$ is less than 25 db, will be given by the following equation :

$$H = l - 0.009\,[25 - (\beta_o - \beta_d)]$$

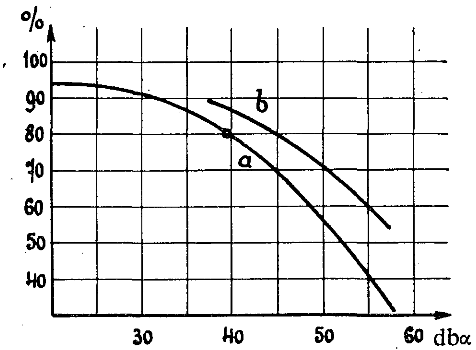
For the difference $(\beta_o - \beta_d)$, the average value has been taken of 8 different measurements uniformly distributed over the frequency band.

In Table III are grouped together the principal quantities which occur in the calculation of H. Fletcher and R. Galt and values are given for the four systems.

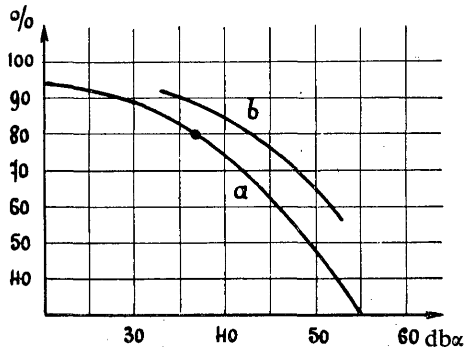
TABLE 3

		R/R	CH/R	R/CH	CH/CH
β_t	Speaking level in db above 10^{-16} W/cm^2	64.5	64.5	64.5	64.5
β_H	Hearings loss in db (Approximate estimated mean)	− 4	− 4	− 4	− 4
R	Frequency characteristic referred to the mean speaking level β_t . . .	Fig. 19	Fig. 20	Fig. 21	Fig. 22
B	Spectral distribution of the interfering noise	Fig. 24	Fig. 24	Fig. 25	Fig. 25
M	Masking effect	Fig. 24	Fig. 24	Fig. 25	Fig. 25
γ	Influence of the frequency characteristic on the mean hearing level . .	0	0	0	0
$(R-M)_1$	Mean value of $(R\text{-}M)$	+11.0	+8.0	+14.0	+9.5
$(R-M)_4$		+ 8.0	+5.0	+ 7.5	+3.5
α	Supplementary gain in db inserted in the system	α	α	α	α
$-\alpha_o$	Attenuation to be inserted in the system to bring the speaking level down to the threshold of audibility	−67.5	−64.5	−70.5	−66
F	Influence of the frequency characteristic on the hearing index	0.825	0.821	0.775	0.779
p	Training factor (for a trained crew)	1	1	1	1
$\beta_o-\beta_d$	Distortion attenuation in db . . .	∞	26	∞	26
H	Influence of the non-linear distortions on A	1	1	1	1

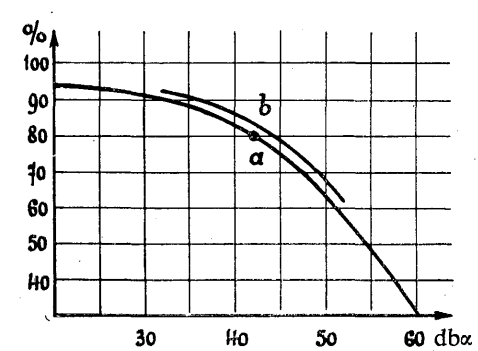
The masking effect of the operator's own voice is small owing to the steep slope of the high frequencies and can be neglected for the calculation of the A.E.N. since it exerts more or less the same influence on all the systems. In the calculation no correction has been made for the factor F. It has also been noticed that the influence of the non-linearity factor of the microphone on the factor *F* was negligible.



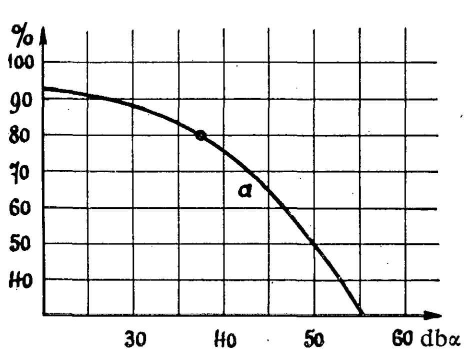
System 1 = R/R



System 2 = CH/R



System 3 = R/CH



System 4 = CH/CH

FIGURES 26 TO 29. — *Curves of the variation of sound articulation of the four different systems as a function of the line attenuation*

The Figures 26 to 29 show the sound articulation as a function of the supplementary attenuation inserted in the system. Curve *(a)* has been established by calculation and curve *(b)* represents the results of subjective measurements made at the C.C.I.F. Laboratory.

In Table IV have been recorded the values of α for a sound articulation of 80 %. Comparison with the reference system (A.R.A.E.N.) gives the required A.E.N. values.

TABLE 4

	Systems	α for 80% Calculated db	Measured at the C.C.I.F. Laboratory db	Difference db	A.E.N. Calculated db	Measured at the C.C.I.F. Laboratory db	Difference db
1	R/R	−40.0	−45.6	−5.6	0	0	
2	CH/R	−37.2	−43.2	−6.0	2.8	2.4	0.4
3	R/CH	−42.1	−44.6	−2.5	−2.1	1.0	3.1
4	CH/CH	−37.8	—	—	2.2	—	—
				−4.7 (mean)			

6. Comparison of the objective calculations and the subjective measurement of A.E.N.

As the A.E.N. has been introduced as a new criterion for the judgement of the quality of a telephone system the comparison between the calculated and the subjective determination is of great interest. To this end measurements have been made, according to the method described above, on the subscriber's telephone systems belonging to other administrations on which the C.C.I.F. Laboratory has already made articulation measurements during the 8th and 9th Series, of Tests, and the articulation has been calculated. Except for the speaking level which was 2 db lower, all the conditions were the same for the example described above. Table V gives a summary of the results. The different subscriber's systems are numbered from I to V.

The column of $\Delta\alpha$ shows that during the determination of the A.E.N.s at the C.C.I.F. Laboratory the attenuation was on the average 7.7 db greater. As the hearing loss and training factor of the testing crew are unknown, little can be said about this difference.

It is to be supposed, however, that the Esperanto consonants used for the tests were more readily understood than the English consonants on which the articulation calculations are based.

The column of A.E.N. allows the following mean values to be obtained :

Combination Sending	Combination Receiving	Average discrepancy
X	R	2.0
R	X	3.3
X	X	−0.2
R = Reference system X = Commercial system.		

TABLE 5

Sending system	Receiving system	F	H	α_o	α for 80% Calculated	α for 80% Measured at the C.C.I.F. Laboratory 8th Series	α for 80% Measured at the C.C.I.F. Laboratory 9th Series	$\Delta\alpha$	A.E.N. Calculated	A.E.N. Measured at the C.C.I.F. Laboratory	Δ A.E.N.
				db	db	db	db		db	db	db
R	R	0.825	1.0	−65.5	−38.0	−46.1	−48.0	− 9.1	0	0	0
I	R	0.800	0.865	−49.0	−17.0		−21.5	− 4.5	21.0	26.5	5.5
R	I	0.777	1.0	−69.9	−35.6		−41.7	− 6.1	2.4	6.3	3.9
I	I	0.779	0.865	−44.4	− 8.6	−61.7		− 8.1	29.4	29.4	0
II	R	0.814	0.946	−54.5	−26.0		−36.3	−10.3	12.0	11.7	−0.3
R	II	0.785	1.0	−63.6	−30.6		−35.2	− 4.6	7.4	12.8	5.4
II	II	0.767	0.946	−50.1	−16.2	−19.0		− 2.8	21.8	27.1	5.3
III	R	0.788	0.955	−54.3	−22.6		−32.6	−10.0	15.4	15.4	0
R	III	0.746	1.0	−62.8	−24.0		−32.1	− 8.1	14.0	15.9	1.9
III	III	0.734	0.955	−41.8	− 2.9	−16.6		−13.7	35.1	29.5	−5.6
IV	R	0.797	0.820	−45.5	−13.7		−21.5	− 7.8	24.3	26.5	2.2
R	IV	0.759	1.0	−62.3	−28.7		−36.3	− 7.6	9.3	11.7	2.4
IV	IV	0.746	0.820	−40.2	− 1.5	− 7.4		− 5.9	36.5	38.7	2.2
V	R	0.804	0.820	−56.3	−23.5		−31.1	− 7.6	14.5	16.9	2.4
R	V	0.752	1.0	−71.2	−32.8		−39.9	− 7.1	5.2	8.1	2.9
V	V	0.775	0.820	−52.8	−14.3	−24.8		−10.5	23.7	21.3	−2.4
								− 7,7			

As experience has shown, the method of measurement and calculation described enables the A.E.N. of an unknown system to be determined from its physical characteristics. The discrepancies with respect to the direct subjective measurement are due to the difficulty of understanding the actual circumstances under which the articulation measurements are made and of taking them into account in the calculations. Even if the absolute values are subject to certain discrepancies, the effect of the various transmission path elements can, however, be determined for an unknown system. This is of great importance in planning telephone networks.

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