THE INTERNATIONAL TELEGRAPH AND TELEPHONE CONSULTATIVE COMMITTEE

(C.C.I.T.T.)

FIFTH PLENARY ASSEMBLY

GENEVA, 4-15 DECEMBER 1972

GREEN BOOK

VOLUME V

Telephone transmission quality, local networks and telephone sets

Part 1 — Series P Recommendations

Part 2 — Supplements to Series P Recommendations

Part 3 — Questions (Study Group XII)

Published by

THE INTERNATIONAL TELECOMMUNICATION UNION

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CONTENTS OF THE C.C.I.T.T. BOOKS STILL APPLICABLE AFTER THE FIFTH PLENARY ASSEMBLY (1972)

GREEN BOOK

Each volume also contains, where appropriate:

— Definitions of specific terms used in the field of this volume.
— Supplements for information and documentary purposes.

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NOTICE TO VOLUME V OF THE GREEN BOOK

Volume V of the *Green Book* fully supersedes Volume V of the C.C.I.T.T. *White Book* (Mar del Plata, 1968).

It has been indicated (immediately after the titles of Recommendations or Supplements) whether the texts are new ones approved by the Plenary Assembly of Geneva, 1972 or are texts amended at the same period. Texts without any such an indication date from at least as far back as the Plenary Assembly of New Delhi, 1960, when Volume V was divided into numbered recommendations; certain of these texts may be even older.

Units

The units used in this Volume are in conformity with C.C.I.T.T. Recommendations B.3 and B.4 (Volume I of the *Green Book*).

The indication "amended Geneva, 1972" has not been affixed to those Recommendations in which the only amendment has been an editorial change concerning units.

The following abbreviations are used, particularly in diagrams and tables, and always have the following clearly defined meanings:

dBm	the absolute power level in decibels;
dBm0	the absolute power level in decibels referred to a point of zero relative level;
dBr	the relative power level in decibels;
dBm0p	the absolute psophometric power level in decibels referred to a point of zero relative level.

INTRODUCTORY NOTE

In order to simplify the wording of the Recommendations in this Volume, the expression "Administration" is used, for shortness, to indicate both a telecommunication administration and a recognized private operating agency.

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PART I

SERIES P RECOMMENDATIONS

Quality of telephone transmission; local networks and telephone installations

SECTION 1

General Recommendations on the transmission quality for an entire International Telephone Connection[1]

Recommendation G.111 (P.11) (Geneva, 1964; amended at Mar del Plata, 1968, and at Geneva, 1972)

REFERENCE EQUIVALENTS IN AN INTERNATIONAL CONNECTION

In the new transmission plan, the total nominal reference equivalent between two subscribers is not strictly limited; its maximum value results from all the various recommendations indicated below.

A. Nominal reference equivalents of the national systems

a) *Definition*

National sending and receiving reference equivalents should be those calculated at the virtual switching points of the international circuit; that is to say, at points a and b of Figure 1/G.111 (P.11) for a country of average size.

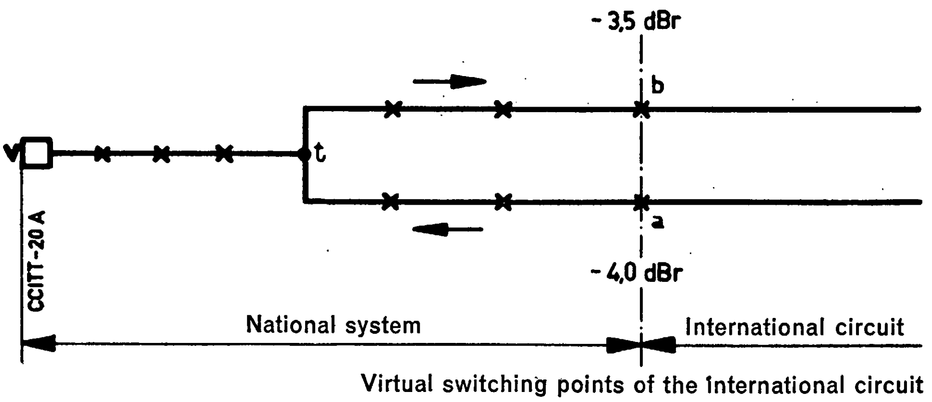


Figure 1/G.111 (P.11) – **Definition of the virtual switching points**

[1] The Recommendations appearing under this heading constitute sub-section 1.1 of Section 1, *Green Book*, Volume III, Part I and Section 1 of Volume V, Part I.

The virtual switching points of an international four-wire telephone circuit are fixed by convention at points of the circuit where the nominal relative levels at the reference frequency are:

—3.5 dBr, sending
—4.0 dBr, receiving

The nominal transmission loss of this circuit at the reference frequency between virtual switching points is therefore 0.5 dB.

Note. — The relative level at a given point of a four-wire circuit is determined by reference to the specifications of the transmission system on which the circuit is set up, the performance of the system (noise, crosstalk, limiting, linearity, etc) being evaluated at a point of zero relative level. For example, the nominal mean power of signals during the busy hour, at a point of zero relative level, is indicated in Section 1 of Recommendation G.223. For further details, see Recommendation G.141, A.

b) *Maximum values*

Provisionally, the national sending and receiving system used to set up 97% of actual incoming or outgoing calls in a country of average size (see Recommendation G.101, B, b, or Figure 1 of Recommendation G.121) should individually comply with both the following conditions:

- the nominal reference equivalent of the sending system between the subscriber and the first international circuit should not exceed 21 dB; and
- the nominal reference equivalent of the receiving system between the same two points should not exceed 12 dB.

(For further details, see Recommendation G.121 (P.21).)

B. Nominal overall loss of the international chain

The nominal loss between the virtual switching points of each international circuit should in principle be 0.5 dB at 800 Hz or 1000 Hz. However, some circuits can be operated with higher losses (see Recommendation G.131, B, a) and certain circuits may be operated at zero loss (see Note 3 of Recommendation G.141, A, a).

As far as transmission is concerned, there is no strict limit on the number of international circuits which may be interconnected in tandem, provided each of them has a nominal loss, between the virtual switching points, of 0.5 dB in the transit condition and provided there is four-wire interconnection. Naturally, the fewer the number of interconnected circuits the better the transmission performance is likely to be (see Recommendation G.101, C).

C. Nominal reference equivalent of a complete connection

a) *Nominal values for each direction of transmission*

The C.C.I.T.T. Laboratory has ascertained the loss to be inserted between a local sending and a local receiving system to obtain an overall reference-equivalent of 36 dB. In this test one, two or three A.R.A.E.N. 300-3400-Hz filters, identical with that used in the S.R.A.E.N., were inserted into the line connecting the two local commercial systems. (Recommendation P.44, *Green Book*, Volume V.)

The frequency-loss characteristic of each filter meets the requirements of Graph No. 2B of Recommendation G.232 the set of three filters conforms to Figure 1 in Recommendation G.132 showing the objective for a chain of 12 carrier circuits in tandem.

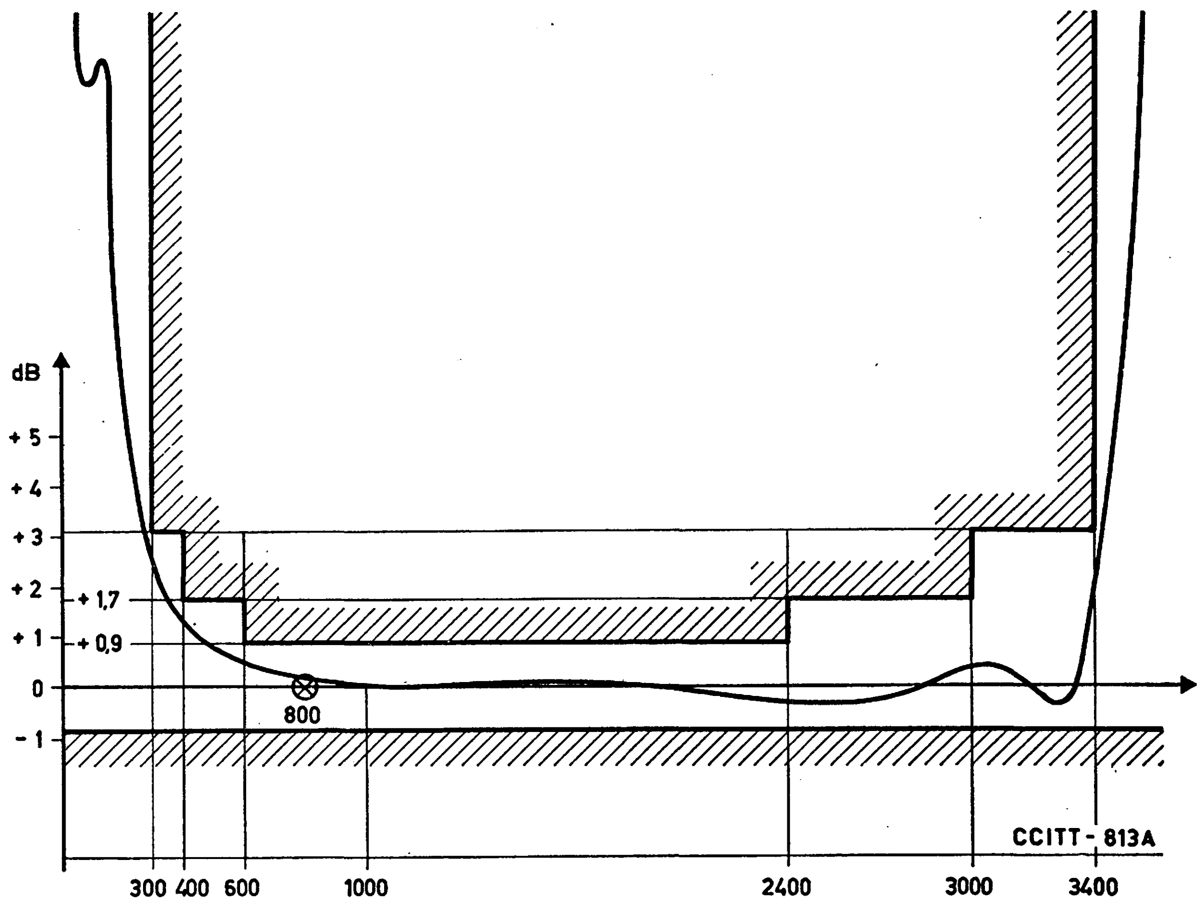


FIGURE 2/G.111 (P. 11) — **Characteristic of A.R.A.E.N. filter**

Figure 2/G.111 (P.11) shows the actual characteristic of one A.R.A.E.N. filter and reproduces Graph No. 2B of Recommendation G.232.

The sending and receiving reference equivalents of the local systems were also determined by the customary procedure.

In view of the results of these tests, it is recommended that Administrations which use modern telephone apparatus should assume, for network planning purposes, that the reference equivalent corresponding to a complete connection is satisfactorily represented (with an error of less than 1 dB) by the sum of the sending and receiving reference equivalents of the local systems, measured separately, and of the equivalent at 800 Hz (or at 1000 Hz) of the chain of long-distance circuits.

Note. — This recommendation makes allowance for the fact that the sending and receiving reference equivalents are determined for conventional conditions in which, for example, the level of the received speech sounds is not usually that to be expected in an international connection close to the acceptable limit. In planning, moreover, allowance cannot be made for all the factors which may vary from one connection to another, such as the exact reflection loss at certain interconnection points, the effects of attenuation distortion, the level of speech sounds transmitted and received, etc.

b) *Difference in transmission loss between the two directions of transmission in national systems*

It is recommended that the absolute value of the difference between loss $t-b$ and loss $a-t$ (see Recommendation G.122) should not exceed 4 dB so that in theory no greater nominal difference than 8 dB could be introduced in international connections.

The following points should be noted:

1. Bearing in mind that most Administrations allocate the losses of their national extension circuits in much the same sort of way (see Annex 1 to Recommendation G.121), connections set up in practice should not exhibit nominal differences much in excess of 3 dB.

2. As far as speech transmission is concerned, from the studies carried out by several Administrations in 1968–1972, it is clear that for connections with overall reference equivalents falling within the range found in practice no great disadvantage attaches to any reasonable difference in nominal overall reference equivalent between the two directions of transmission.

D. Variations in time and effect of circuit noise

The nominal reference equivalents given for national systems include the systematic differences between the sensitivities of the subscriber's set at the sending and receiving ends and their nominal values; however, they do not include the variations of loss with time in the various parts of the national system, nor random variations of the reference equivalents determined by subjective methods. Recommendation G.151, C sets forth the objectives recommended by the C.C.I.T.T. in connection with variations in transmission losses of international circuits and national extension circuits relative to the nominal values.

According to the results of measurements supplied by one Administration the reference equivalent of its transmitting system rises by an average of 0.6 dB per annum, a systematic increase due to ageing of the microphone [see also Note 3 of paragraph B, b of Recommendation G.121 (P.21)].

Annex A in the *Red Book*, Volume V *bis*, gives information on the statistical variations of reference equivalents.

Annex B in the *Red Book*, Volume V *bis*, mentions the effect on transmission performance of these variations in the equivalent and of the limits recommended for circuit noise.

E. Practical limits of the reference equivalent between two operators or one operator and one subscriber

The limits are being studied for the new transmission plan; the values hitherto recommended are given in the *Red Book*, Volume V, page 10, Note 1, and in applying them Note 2 of the same text should be borne in mind.

The values for the complete connections shown in the table in the *Red Book*, Volume V, page 9, and reproduced in the table of the appendix to Section 1 (Volume III) are not applicable to the transmission plan now recommended by the C.C.I.T.T.

Recommendation G.112 (P.12) (modified at Geneva, 1964, and at Mar del Plata, 1968)

ARTICULATION REFERENCE EQUIVALENT (A.E.N.)

The transmission quality of international telephone calls will always be satisfactory if the reference equivalent limits fixed in Recommendation G.111 (P.11) are respected together with the limits fixed in Volume III of the *Green Book* for noise, crosstalk, etc., and if, in addition, use is made of telephone sets of modern types which have satisfactory sensitivity/frequency characteristics and efficient anti-sidetone arrangements [see Recommendation G.121, E (P.21, E)].

Administrations wishing to make a thorough study of the transmission quality of their national sending and receiving systems could be guided by the A.E.N. method described below.

A. *Definition of the articulation reference equivalent (A.E.N.)*

Articulation reference equivalent (A.E.N.) (G.B.) [Equivalent articulation loss (Am.)—Affaiblissement équivalent pour la netteté (A.E.N.) (F)]

If articulation tests are made under specified conditions alternately on a telephone system to be tested and on the "reference system for the determination of A.E.N." (S.R.A.E.N.) with different values of line attenuation, up to the point where values of articulation on both systems are substantially reduced, then the results of these tests may be recorded in the form of curves showing the variation of sound articulation against attenuation. The value A_1 of the attenuation of the system under test, and the value A_2 of the attenuation of the S.R.A.E.N. at a fixed value 80% sound articulation can then be determined.

$(A_2–A_1)$ is by definition equal to the *articulation reference equivalent* (A.E.N.).

B. *Calculation of the nominal articulation reference equivalent of a national sending or receiving system* [1]

The nominal A.E.N. of a national sending or receiving system is the sum of the following quantities:

1. The nominal A.E.N. (average value in service) of the local system;
2. The nominal A.E.N. of the connection between the local exchange and the international exchange (average value in service).

The articulation reference equivalent, in service, of the connection between the local exchange and the international exchange is equal to the sum of the following numbers [2]:

— the equivalent of the trunk circuits between the last trunk exchange and the international exchange, measured at 800 Hz, increased by the transmission impairment due to bandwidth limitation [see Recommendation G.113 (P.13) below] when these circuits have an attenuation/frequency distortion greater than that which is allowed in the recommendations of the C.C.I.T.T.;

— the average articulation reference equivalent of the toll circuits given by the following expression:

$$i = K \times L$$

where

i = average A.E.N. in decibels;
L = length of the trunk-junction in kilometres;
K = coefficient which depends on the type of trunk-junction considered, in decibels per kilometre (see the Annex below),

— the mean A.E.N. of each intermediate exchange. The A.E.N. resulting from the insertion of a circuit element which, in accordance with the recommendations of the C.C.I.T.T., effectively transmits frequencies from 300 to 3400 Hz can be calculated by taking the arithmetic mean of the four values of insertion loss (or gain) of the element considered measured at 500, 1000, 2000 and 3000 Hz and expressed in decibels. Until there are more accurate values of this rating available, as will result from any measurements that Administrations may make in this respect, a provisional value of 1 dB for each exchange introduced into the connection will be used.

Note 1. — Circuit noise which is within the limits fixed by C.C.I.T.T. recommendations is not taken into account.

Note 2. — The "composite attenuation" of the lines connecting the international exchanges to the local exchanges should be such that the reference equivalent of the national sending system and the reference equivalent of the national receiving system remain within the limits considered compatible with good telephone transmission.

C. *Determination of A.E.N.*

The reference system for the determination of the A.E.N. (S.R.A.E.N.) and the method of determining the A.E.N. of commercial telephone systems at the C.C.I.T.T. Laboratory are described in Recommendations P.44 and P.45 (*Green Book*, Volume V).

[1] It is agreed for international purposes that the result obtained by this calculation B represents the magnitude of the articulation reference equivalent for a national transmitting or receiving system. This number is called the nominal articulation reference equivalent, to distinguish it from the articulation reference equivalent measured on the complete national sending or receiving system.

[2] Articulation tests have shown that the A.E.N. can be calculated approximately for such a link, in the manner shown above.

D. *Nominal A.E.N. values for the national sending system and the national receiving system*

By way of information, it is pointed out that Administrations using the A.E.N. method consider it very desirable that national sending and receiving systems used to set up 90% of actual outgoing or incoming calls should individually meet both of the following requirements:

— the nominal A.E.N. of the national sending system should not exceed 24 dB;
— the nominal A.E.N. of the national receiving system should not exceed 18 dB.

Note 1. — The values 24 dB and 18 dB given above for the national sending and receiving systems refer to the two-wire terminals of the international circuit, whereas the reference equivalents recommended in Recommendation G.111 (P.11) refer to the virtual switching points of the international circuit. These A.E.N. values do not include the probable variations, as a function of time, of the equivalents of the trunk circuits which form part of the national system.

Note 2. — These values apply to the A.E.N. values deduced from the values measured for a local system at the C.C.I.T.T. Laboratory, as described in Recommendation P.45 with, in particular, 60 dB room noise at the receiving end for commercial systems and an electrical background noise (having a psophometric e.m.f. of 2 millivolts) injected into the input of the receiving system of the S.R.A.E.N.

Note 3. — The A.E.N. method does not make allowance for the effect of sidetone on subscribers' speech power.

Administrations wishing to prepare transmission plans for their national network, on the basis of transmission performance rating, will find in Annex 2 to Volume IV of the *Green Book* of the C.C.I.F. (Geneva, 1954), information on the corrections to be made to the values of A.E.N. to allow for sidetone at the sending end.

ANNEX

[to Recommendation G.112 (P.12)]

Average A.E.N. of trunk-junctions

A trunk-junction may be considered as a quadripole inserted between the impedance of the first trunk circuit, seen through the switchboard (or switches), and the impedance of the local system (feeding bridge + subscriber's line + subscriber's apparatus).

For a given frequency, the loss introduced by such a circuit is represented by its composite attenuation[1] which is the sum of the image attenuation of the circuit itself and of the other terms representing all the effects due to reflections introduced by mismatch between the image impedance of the circuit and the impedances of the terminations defined above.

According to tests made by the United Kingdom Post Office, the A.E.N. due to the reflections can be represented by the arithmetic mean of the reflection losses measured at frequencies of 500, 1000, 2000 and 3000 Hz.

The transmission performance rating of an unloaded line is measured by its image attenuation at 1500 Hz and this is approximately equal to the arithmetic mean of the image attenuations at the four frequencies quoted above[2].

Therefore, the A.E.N. of the trunk-junction may be obtained directly, taking account not only of the effect due to the image attenuation but also of the effect of reflections, by taking the arithmetic mean of the composite attenuations measured at the four frequencies referred to above.

As the impedance of the local systems varies widely, it is not possible to define a single value for the average A.E.N. for a trunk-junction, but only an average value obtained by taking the arithmetic mean of several values of the A.E.N., measured under several terminal conditions (see C.C.I.F.—1952/1954—4th S.G.—Document No. 32, Annex).

For each type of trunk-junction (defined by the electrical characteristics of the circuit), the average A.E.N. is proportional to the length of the circuit, the ratio being *easily determined* when three or four values of the A.E.N. are known. It is given by the formula:

[1] In practice, instead of using the composite attenuation, insertion loss may be used.

[2] The attenuation of a non-loaded cable circuit is proportional to the square root of the frequency. The frequencies 500, 1000, 2000, 3000 Hz are in the ratio 1, 2, 4, 6 and their square roots in the ratio 1, 1.41, 2, 2.45 of which the arithmetic mean is 1.72, i.e. almost the square root of 3; therefore this mean corresponds to a frequency of $3 \times 500 = 1500$ Hz.

$$i = K \times L \tag{1}$$

where

i = average A.E.N. in decibels;

L = length of trunk-junction in kilometres;

K = coefficient, which depends on the type of trunk-junction considered, in decibels per kilometre.

To determine, once and for all, the different values of the coefficient K, the composite attenuation of three or four different lengths of each type of trunk-junction used in a particular network (if necessary using artificial lines) can be measured; for this purpose the technique described in Document 32 referred to above (see also Annex 2 to Question No. 10 in the *Yellow Book* of the C.C.I.F., Volume I *ter*, page 400), and one of the methods of measuring of the composite attenuation described in the *Blue Book*, Volume IV, Part III, Supplement No. 1 can be used.

From equation (1) the value of the average A.E.N. may be calculated for any length and any type of trunk-junction in the national network considered.

Recommendation G.113 (P.13) (amended at Geneva, 1964, and at Mar del Plata, 1968)

TRANSMISSION IMPAIRMENTS AND NOISE

A. Transmission impairment

a) *Due to bandwidth limitation (cut-off impairment) effectively transmitted by the trunk circuit*

Observations have been made in the United States of America of the repetitions during conversations and articulation measurements have been made in various national laboratories as well as in the C.C.I.T.T. Laboratory. The results obtained permit the mean curve given in Figure 1/G.113 (P.13) to be plotted showing the impairment due to cut-off frequency by a trunk circuit.

The equation to this curve is $y = 2(3.7 - f)^2$, where y is the transmission impairment (in decibels) due to the limitation of the frequency bandwith effectively transmitted, and f is the frequency (in kHz) for which the loss of the circuit exceeds its loss at 1000 Hz by 10 dB.

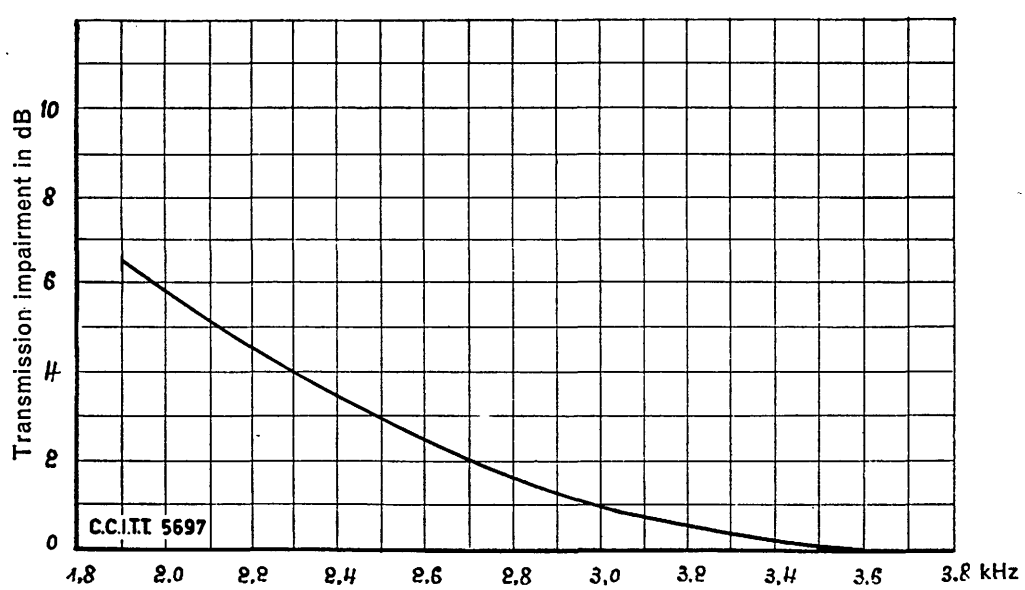


FIGURE 1/G.113 (P.13). — **Transmission impairment due to bandwidth limitation (cut-off impairment)**

Note. — The frequencies shown on the abscissa are the maximum frequencies effectively transmitted according to the definition adopted in the United States of America, i.e. those for which the attenuation is greater by 10 dB than the attenuation at 1000 Hz.

Note. — The cut-off impairment for a chain of national trunk circuits or for a connection between two international exchanges made up of several international circuits is not obtained by adding the individual impairments. It is necessary to consider the impairment for the circuit which transmits effectively the narrowest band of frequencies.

b) *Due to room noise*

The method of measuring A.E.N. takes account of 60 dB of room noise (Hoth spectrum)[1] at the receiving end; information regarding the method of evaluating the "impairment due to room noise" used in the United States of America is given in Annex 3, *Red Book*, Volume V, Part II.

Although the transmission impairment values mentioned in this annex are now out of date, they show the adverse effect on speech transmission in telephony of a high level of room noise.

B. Effect of circuit noise

The C.C.I.T.T. recommends that the mean value, expressed in decibels and taken over a large number of world-wide connections (each including six international circuits), of the distribution of one-minute

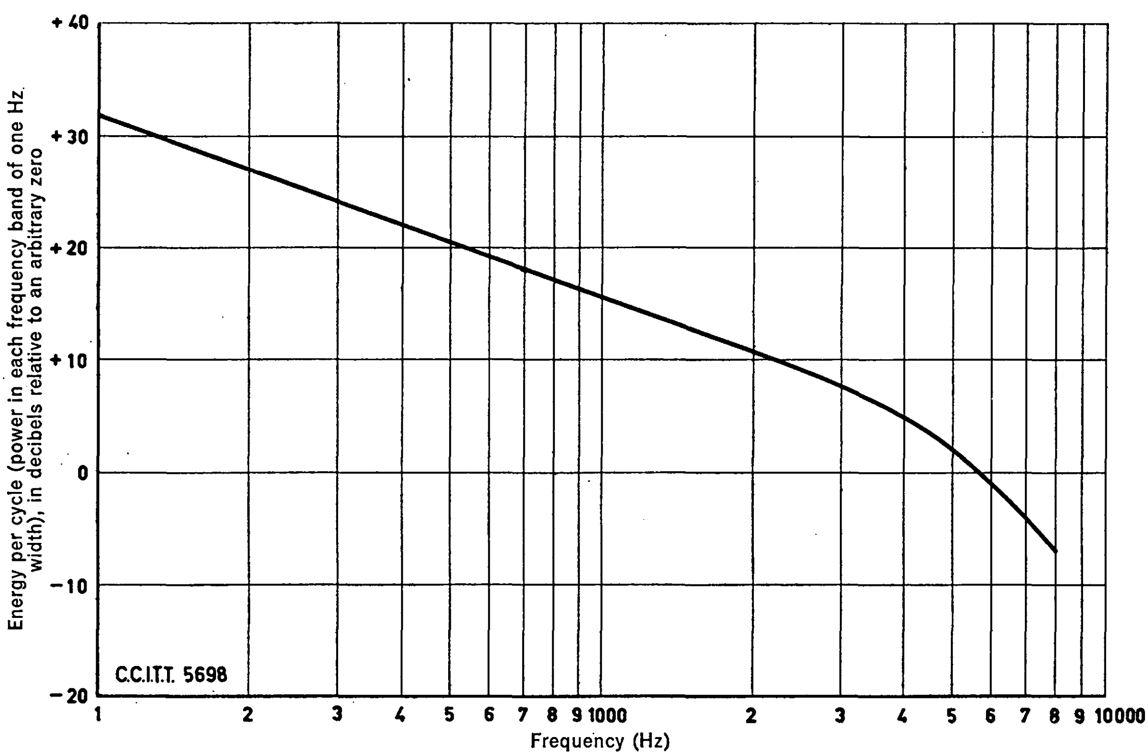


FIGURE 2/G.113 (P.13). — **Power density spectrum of the room noise produced in the listening cabinet of the C.C.I.T.T. Laboratory**

This curve conforms to the mean power density spectrum of noise observed in locations where telephone sets are situated, published by Hoth.

[1] The power density spectrum of the room noise used in A.E.N. measurements is given in Figure 2/G.113 (P.13). The following articles give information on room noise at locations where commercial telephone sets are located:

1. A Room Noise Survey of Business Subscribers' Telephone Locations. *B.P.O. Research Report*, No. 8990, 1935.
2. Room Noise at Telephone Locations. D. F. SEACORD, *Electrical Engineering*, Part 1, 58, 255, 1939.
3. Room Noise Spectra at Subscribers' Telephone Locations. D. F. HOTH, *Journal of the Acoustical Society of America*, 12, 499, 1941.

mean values of noise power of the connections, should not exceed −43 dBm0p referred to the input of the first circuit in the chain of international circuits.

Annexes B, C and D in the *Red Book*, Volume V *bis*, Part II, describe how the C.C.I.T.T. made allowance for the effect of noise on transmission performance in planning the international network. The procedure does not make explicit use of any transmission impairment due to circuit noise.

By way of information, the method used in the United States to fix objectives for circuit noise is described by D.A. Lewinski, in an article entitled: A New Objective for Message Circuit Noise (*Bell System Technical Journal*, Volume XLIII, pages 719–740, No. 2, March 1964).

Further information can be found in the C.C.I.T.T. handbook *Transmission Planning of Switched Telephone Networks*.

Note. — Annex 2 to the *Red Book*, Volume V, Part II, is out of date and should be deleted.

Recommendation G.114 (P.14) (Geneva, 1964; amended at Mar del Plata, 1968)

MEAN ONE-WAY PROPAGATION TIME

A. Limits for a connection

It is necessary in an international telephone connection to limit the propagation time between two subscribers. As the propagation time is increased, subscriber difficulties increase, and the rate of increase of difficulty rises. Relevant evidence is given in the References below, particularly with regard to paragraph b.

The C.C.I.T.T. therefore *recommends* the following limitations on mean one-way propagation times when echo sources exist and appropriate echo suppressors are used:

a) 0 to 150 ms, acceptable.

Note. — Old-type echo suppressors may be used; they should be modified for delays above 50 ms.

b) 150 to 400 ms, acceptable, provided that increasing care is exercised on connections as the mean one-way propagation time exceeds about 300 ms, and provided that echo suppressors designed for long delay circuits are used;

c) above 400 ms, unacceptable. Connections with these delays should not be used except under the most exceptional circumstances.

Until such time as additional, significant information permits Administrations to make a firmer determination of acceptable delay limits, they should take full account of the documents referred to in the References in selecting, from alternatives, plans involving delays in range b above.

References

C.C.I.T.T. *Red Book*, Volume V *bis*, Annex E (United States).

C.C.I.T.T. *Red Book*, Volume V *bis*, Annex F (United Kingdom).

C.C.I.T.T. *Red Book*, Volume V *bis*, Annex 4 to Question 6/XII (Italy).

C.C.I.T.T. *White Book*, Volume V, Supplements 1–6.

BARSTOW, J. M.: Results of user reaction tests on communication via Early Bird satellite; *Progress in Astronautic Aeronautics*, 19, 1966, Academic Press, New York and London.

HELDER, G. K.: Customer evaluation of telephone circuits with delay; *Bell System Technical Journal*, 45, September 1966, pp 1157–1191.

RICHARDS, D. L.: Transmission performance of telephone connexions having long propagation times; *Het P.T.T.-Bedrijf*, XV, No. 1/2, May 1967, pp 12–24.

KARLIN, J. E.: Measuring the acceptability of long delay transmission circuits used during the "Early Bird" transatlantic tests in 1965; *Het P.T.T.-Bedrijf*, May 1967, pp 25–31.

DE JONG, C.: Observations on telephone calls between the Netherlands and the U.S.A.; *Het P.T.T.-Bedrijf*, May 1967, pp 32–36.

HUTTER, J.: Customer response to telephone circuits routed via a synchronous-orbit satellite; *P.O.E.E.J.*, Volume 60, p. 181, October 1967.

B. VALUES FOR CIRCUITS

In the establishment of the general interconnection plan within these limits the one-way propagation time of both the national extension circuits and the international circuits must be taken into account.

a) *National extension circuits*

The main arteries of the national network should consist of high-velocity propagation lines. In these conditions, the propagation time between the international centre and the subscriber farthest away from it in the national network will probably not exceed

$$12 + (0.004 \times \text{distance in kilometres}) \text{ ms.}$$

Here the factor 0.004 is based on the assumption that national trunk circuits will be routed over high-velocity plant (250 km/ms). The 12-ms constant term makes allowance for terminal equipment and for the probable presence in the national network of a certain quantity of loaded cables (e.g. three pairs of channel translating equipments plus about 160 km of H 88/36 loaded cables). For an average-sized country the one-way propagation time will be less than 18 ms.

b) *International circuits*

International circuits will use high-velocity transmission systems; the one-way propagation times, or velocity, that should be assumed for planning purposes are:

1. *Terrestrial lines* (land lines and submarine cables)

160 km/ms

This propagation velocity includes an allowance for terminal and intermediate multiplex equipment likely to be associated with a transmission line.

2. *Satellite links*

The mean one-way propagation times between earth stations for two illustrative single-hop communication satellite systems are:

Satellite at 14 000 km altitude	110 ms
Satellite at 36 000 km altitude	260 ms

The one-way propagation times do not include any allowance for the distance from the earth stations to locations where the satellite circuits can either be extended on other international transmission systems or switched to other national or international circuits. These additional times should be taken into account for planning purposes. The practical distances between earth stations depend not only on the altitude of the satellites but also on the orbits and positions of the satellites relative to the earth stations. Exact account should be taken of these parameters in particular applications.

The magnitude of the mean one-way propagation time for circuits on high altitude communication satellite systems makes it desirable to impose some routing restrictions on their use. Details of these restrictions are given in Recommendation Q.13, Section 4.

Note. — The propagation time referred to above is the group delay as defined in the I.T.U. *List of Definitions of Essential Telecommunication Terms* (Definition No. 04–17); the numerical values are calculated at a frequency of about 800 Hz.

Recommendation P.15 (amended at Geneva, 1964)[1]

GROUP-DELAY DISTORTION

The permissible differences for a world-wide chain of 12 circuits, each on a single group connection, between the minimum group delay (throughout the transmitted frequency band) and the group delay at the lower and upper limits of this frequency band are indicated in the table below:

	Lower limit of frequency band	Upper limit of frequency band
	ms	ms
International chain	30	15
Each of the national four-wire extensions	15	7.5
On the whole four-wire chain	60	30

Typical group delays at various frequencies for a chain of 12 circuits in tandem are given in Recommendation G.232 (*Green Book*, Volume III).

Recommendation G.116 (P.16) (Geneva, 1972)

SUBJECTIVE EFFECTS OF DIRECT CROSSTALK

Thresholds of audibility and intelligibility

1. *Factors which affect the crosstalk thresholds*

The degree of audibility and intelligibility of a crosstalk signal depends on a large number of factors.

A simple and generally applicable method for estimating the required loss in the crosstalk path as a function of the factors affecting the audibility or the intelligibility of the speech crosstalk signal can be obtained if certain simplifications are made.

[1] Same as Recommendation G.133 (*Green Book*, Volume III).

The main factors influencing the intelligibility of the vocal crosstalk signal are listed below:

a) *Quality of transmission of telephone apparatus*

The sending and receiving reference equivalents are decisive factors. The same is true of the reference equivalent of sidetone when room noise is present. The use of modern telephone apparatus with smooth frequency curves is assumed.

b) *Circuit noise*

The circuit noise on the connection of the disturbed call must be taken into account. This is measured by a psophometer equipped with a weighting network for telephone circuits.

c) *Room noise*

Room noise affects the ear directly through ear-cap leakage between the ear and the receiver and indirectly by sidetone. Sidetone, which consists of the transmission of room noise to the receiver, also depends on operating conditions. Unlike circuit noise, the effect of room noise can be reduced to some extent by the user of the telephone. For this reason and to allow for unfavourable cases, the measurements were made with slight [40 dB(A)] room noise as well as with negligible room noise.

d) *Conversation on the disturbed connection*

While there is active speech on the disturbed connection practical levels of crosstalk are inaudible. However, before the conversation starts or during long pauses in the conversation it is possible for crosstalk to be heard and perhaps understood. The information given in this Recommendation assumes no conversation on the disturbed connection.

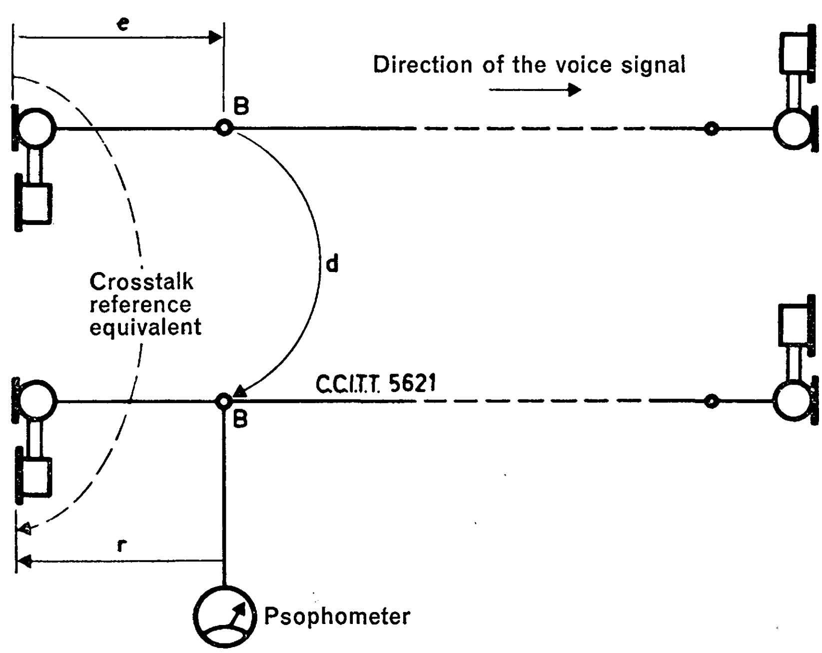


FIGURE 1/G.116 (P.16). — **Conventional subdivision of crosstalk reference equivalent**

e = Sending reference equivalent of the disturbing subscriber's set;
r = Receiving reference equivalent of the disturbed subscriber's set;
d = Crosstalk path attenuation, so that: crosstalk reference equivalent = e + d + r;
B = Terminals of subscribers' sets
Disturbing and disturbed subscriber's sets at the same end: near-end crosstalk
Disturbing and disturbed subscriber's sets at opposite ends: far-end crosstalk

e) *Microphone noise*

The noise produced by the carbon microphone of the disturbed telephone may slightly reduce the intelligibility of the vocal crosstalk signal owing to sidetone. Good quality modern microphones have been assumed in this Recommendation.

f) *Crosstalk coupling*

The intelligibility of a crosstalk signal also depends on the nature of the crosstalk coupling which is generally a function of frequency. The reference equivalent of the crosstalk transmission path can be conventionally divided into the sending reference equivalent of the subscriber's set causing the disturbance, the receiving reference, equivalent of the subscriber's set subject to disturbance, and the transmission loss of the crosstalk transmission path. Figure 1/G.116 (P. 16) illustrates this conventional subdivision.

In the absence of further information, the reference equivalent of the crosstalk coupling may be taken to be the attenuation measured or calculated at a frequency of 1100 Hz, as advocated in Recommendation G.134 (*Green Book*, Volume III) for telephone exchanges.

2. *Median listener thresholds of the audibility and intelligibility of vocal crosstalk*

The curves in Figure 2/G.116 (P.16) represent the nominal overall reference equivalents of the crosstalk transmission path corresponding to the thresholds of audibility and intelligibility as a function of the receiving reference equivalent; their parameter is the circuit noise; room noise is negligible or equal to 40 dB (having Hoth spectrum and measured with A weighting).

These curves represent median values for the various conditions such that in each case 50% of subscribers' opinions are respectively above and below the particular curve. The standard deviation for listeners has been observed to lie in the range 4 to 6 dB.

The results of the original experiments [which form the basis of the curves in Figure 2/G.116 (P.16)] were expressed in terms of speech level (e.g. in volume units) and on that basis showed a satisfactory degree of agreement among themselves.

The thresholds are based on the assumption that a subscriber set with a sending reference equivalent of 0 dB corresponds in practice to a speech level of -10 VU at the subscriber set terminals with a load of 600 ohms.

However, in order for the results to be directly useful for planning purposes for networks designed and characterized on the basis of reference equivalents, it is necessary to introduce a factor (c) which effectively establishes the relationship between speech level and sending reference equivalent.

The correction factor c has been defined in the following manner:

$$c = V_c - V_L \text{ dB}$$

where

V_c = speech level in decibels under normal conversational conditions at a particular point on the disturbing connection;

V_L = speech level in decibels at the same point in the disturbing connection under conditions corresponding to a speech level of -10 VU at the output of a subscriber set with a sending reference equivalent of 0 dB (i.e. it is assumed that the listening tests have been carried out at this speech level).

Thus the correction factor c is positive for conditions in which the speech level on the disturbing circuit is greater than that corresponding to -10 VU at the output of a subscriber set with 0 dB sending reference equivalent. This correction factor must be added to the value of nominal crosstalk reference equivalent given in Figure 2/G.116 (P.16).

In general the values of c will be a function of the overall reference equivalent and to some extent of the circuit noise and sidetone reference equivalent on the disturbing circuit. Typical values have been estimated from speech level measurements made by several Administrations and are given in Table 1 together with standard deviations.

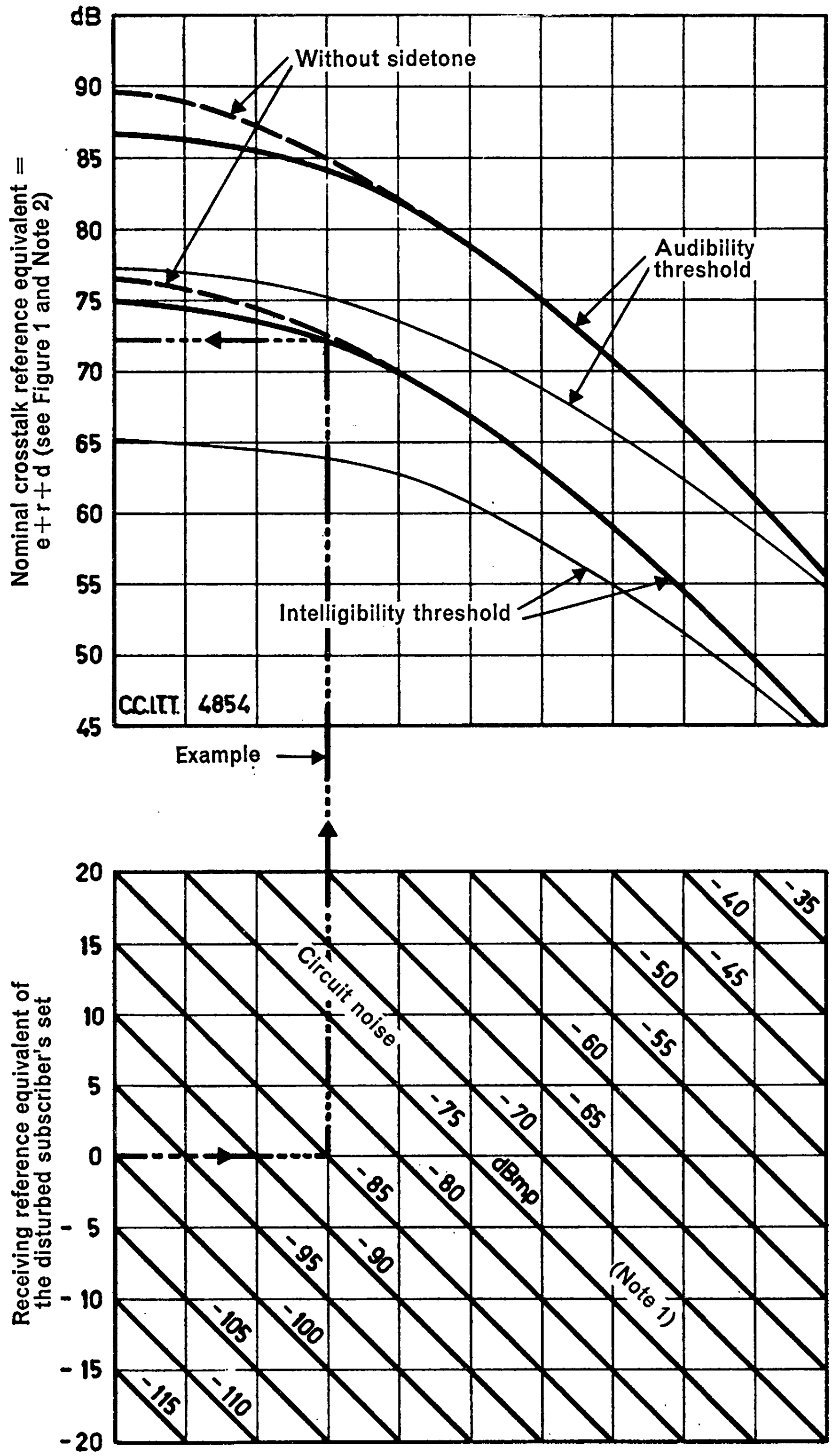


FIGURE 2/G.116 (P.16). – **Crosstalk reference equivalent as a function of the receiving reference equivalent and of circuit noise**

——— Negligible room noise
——— 40 dB (A) room noise

The sidetone reference equivalent of the disturbed subscriber's set appropriate to these curves is +13 dB.

Note 1. – The circuit noise is referred to the terminals of the subscriber's set having the reference equivalent indicated.

Note 2. – The value of factor *c* must be known before the curves can be used (see text).

TABLE 1

MEAN VALUES AND STANDARD DEVIATIONS OF THE FACTOR *c* FOR VARIOUS ADMINISTRATIONS

Administration	Nominal overall reference equivalent of the disturbing connection	Estimated mean value of factor *c* $\overline{C}$	Estimated standard deviation of the factor *c* σ_c
	dB	dB	dB
A. T. & T.	10 20 30	−2 +2 +5	4
Switzerland	35	+3.5	4
Sweden (Note 1)	10	−6	5
United Kingdom Post Office (Note 2)	10 20 30	+3 +4 +5	4.8

Note 1. — Preliminary results.

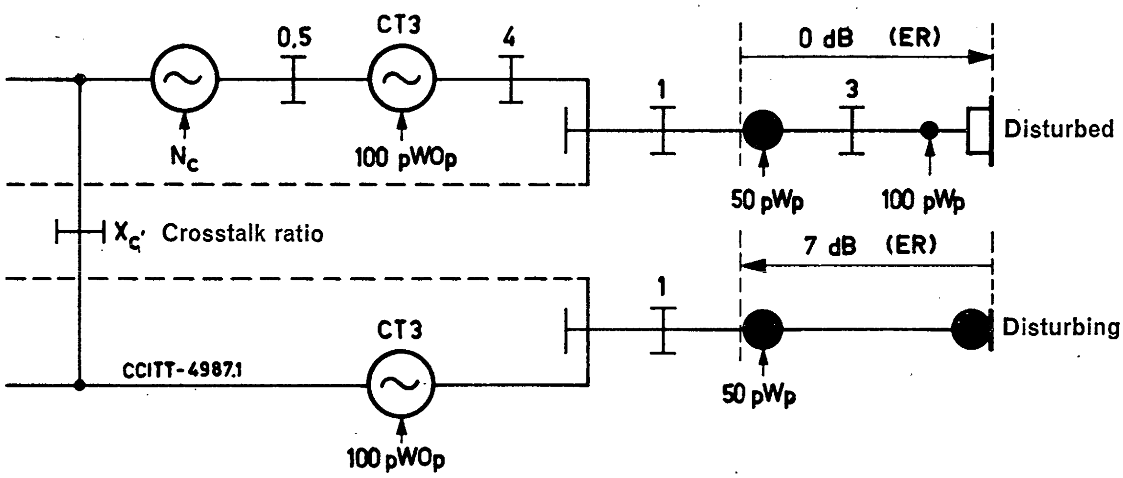
Note 2. — With a circuit noise level of −60 dBm on the disturbing connection.

ANNEX

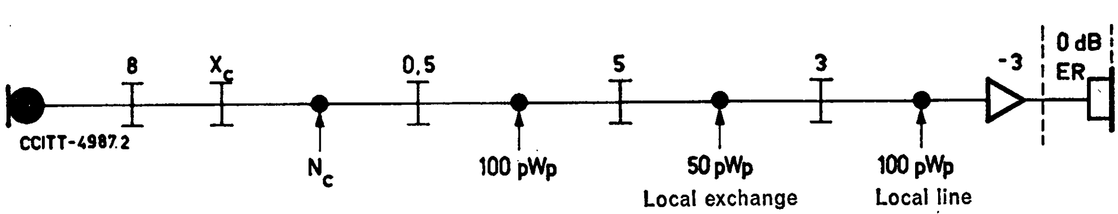
[to Recommendation G.116 (P.16)]

An example of the method of calculation

In order to demonstrate the method of using the information given in Recommendation G.116 (P.16) to calculate the probability of encountering (for example) intelligible crosstalk, a hypothetical reference connection is needed. An example of an important class of connection is given in Figure 3 of Recommendation G.103, *Green Book*, Volume III, and appropriate portions of two such connections with crosstalk between them introduced by, for example, the international circuit, are shown below:

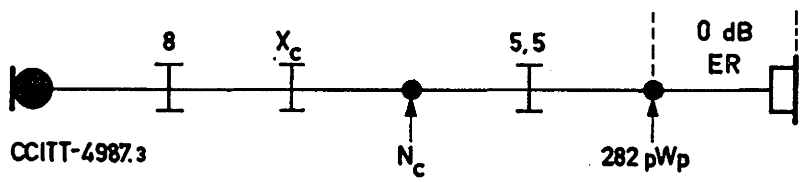


The crosstalk path of interest may be redrawn as shown:



(The 50 pWp and 100 pW0p sources prior to the X_c-pad are ignored, for after traversing the X_c-pad, the resultant noise power contributions will be negligibly small.)

The diagram may be further simplified by referring all the given noise powers to the input of a local system having a reference equivalent of 0 dB and summing (as far as possible) the various losses.



CCITT-4987.3

Considering for the sake of example two specific cases, namely $X_c = 58$ dB; $N_c = 500$ pW0p and $X_c = 62$ dB; $N_c = 200$ pW0p, the corresponding values of overall X and total N are:

Examples studied	Corresponding values X	N
58 dB; 500 pW0p	71.5	—63.7
62 dB; 200 pW0p	75.5	—64.7

associated with the following arrangement:

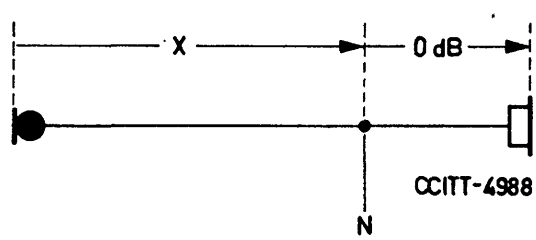


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Table 1 records the values of the median threshold of intelligible crosstalk between an active talker and a silent listener. The values have been taken from the curves given in Figure 2 of Recommendation G.116 (P.16).

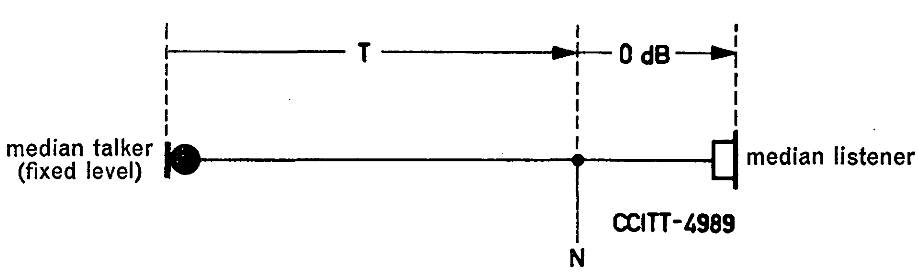

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TABLE 1

Median listener thresholds of intelligible crosstalk as a function of the noise power level at the input to a 0 dB reference equivalent receiving end, for a variety of listening conditions [based on Figure 2 of Recommendation G.116, (P.16)]

<table>
<tr><th rowspan="3">N dBm0p, noise power level at input to 0 dB RRE</th><th colspan="3">T dB, nominal overall reference equivalent of the crosstalk path</th></tr>
<tr><th colspan="2">Negligible room noise</th><th rowspan="2">+40 dB (A) room noise with sidetone</th></tr>
<tr><th>Without sidetone</th><th>With sidetone</th></tr>
<tr><td>−100</td><td>76.5</td><td>75.0</td><td>65.1</td></tr>
<tr><td>− 95</td><td>75.7</td><td>74.5</td><td>64.9</td></tr>
<tr><td>− 90</td><td>74.0</td><td>73.0</td><td>64.2</td></tr>
<tr><td>− 85</td><td>72.5</td><td>72.0</td><td>64.0</td></tr>
<tr><td>− 80</td><td colspan="2">70.0</td><td>62.5</td></tr>
<tr><td>− 75</td><td colspan="2">67.0</td><td>60.5</td></tr>
<tr><td>− 70</td><td colspan="2">63.0</td><td>58.0</td></tr>
<tr><td>− 65</td><td colspan="2">59.0</td><td>55.0</td></tr>
<tr><td>− 60</td><td colspan="2">54.5</td><td>51.5</td></tr>
<tr><td>− 55</td><td colspan="2">49.5</td><td>47.5</td></tr>
<tr><td>− 50</td><td colspan="2">44.0</td><td>43.0</td></tr>
</table>

Note. — The sidetone reference equivalent is +13dB. For intermediate values, use linear interpolation. The talker delivers −10 VU from a 0 dB reference equivalent end.

The distribution of pairs of connections between which there is intelligible crosstalk is in general a combination of three separate distributions:

1) The distribution of talker volumes on the disturbing connection;

2) The distribution of crosstalk reference equivalent introduced by cables and equipment;

3) The distribution of listener perception of the speech crosstalk levels received on the disturbed connection.

In order to take account of the distribution of real talker volumes a correction factor, c, is needed which, being characteristic of national networks, at this present time must be supplied by the user. As indicated in Table 1 of Recommendation G.116 (P.16) the value of $\overline{C}$ lies in the range −6 to +5 dB and σ_c in the range 4 to 5 dB.

For this example we will use $\overline{C} = 4$ dB, and $\sigma_c = 4$ dB.

We do not have a distribution of crosstalk reference equivalents to take account of at this time; just two specific values.

The standard deviation of the listeners' threshold about the median value is in the range 4 to 6 dB. We will take the value $\sigma_t = 5$ dB in this example.

If t is the threshold value for a particular listener, c the speech volume correction factor for a particular talker and x the actual value of the reference equivalent of the crosstalk path between them, then when x is equal to or less than $t + c$, intelligible overhearing occurs. Denoting the difference $x - (t + c)$ by z, intelligible overhearing for this particular pair arises when z is zero or less.

If x, t, and c are each normally distributed (or may fairly be assumed to be so) with mean values $\overline{X}$, $\overline{T}$, and $\overline{C}$ and standard deviations σ_x, σ_t, and σ_c, then z is also normally distributed with mean value $\overline{Z} = \overline{X} - (\overline{T} + \overline{C})$ and standard deviation $\sigma_z = \sqrt{\sigma_x^2 + \sigma_t^2 + \sigma_c^2}$.

The normal deviate at $z = 0$ is given by $\overline{Z}/\sigma_z$ and the probability of $z' \leqslant 0$ can be found from tables of the cumulated normal distribution (single upper tail).

Taking the particular case of 58 dB; 500 pW0p and considering +40 dB(A) room noise with +13 dB sidetone then $N = -63.7$ gives $T = 54.1$ (by interpolation of Table 2) so that $\overline{Z} = \overline{X} - (\overline{T} + \overline{C}) = 71.5 - (54.1 + 4.0) = 13.4$ and $\sigma_z = \sqrt{(\sigma_x^2 + \sigma_t^2 + \sigma_c^2)} = \sqrt{(0 + 25 + 16)} = \sqrt{41} = 6.4$. Hence $\overline{Z}/\sigma_z = 13.4/6.4 = 2.10$ corresponding to 1.8% risk of intelligible overhearing.

Table 2 displays the results for each combination used in this example.

TABLE 2

PROBABILITIES OF INTELLIGIBLE OVERHEARING BETWEEN ACTIVE TALKERS AND SILENT LISTENERS

$\sigma_x = 0; \sigma_t = 5; \sigma_c = 4; \overline{C} = 4$

Example studied	+40 dB (A) room noise; 13 dB sidetone	Negligible room noise; 13 dB sidetone (or no sidetone)[a]
58 dB; 500 pW0p	1.8%	6.7%
62 dB; 200 pW0p	0.5%	2.4%

[a] With the values used in the examples, the presence or absence of sidetone has no effect.

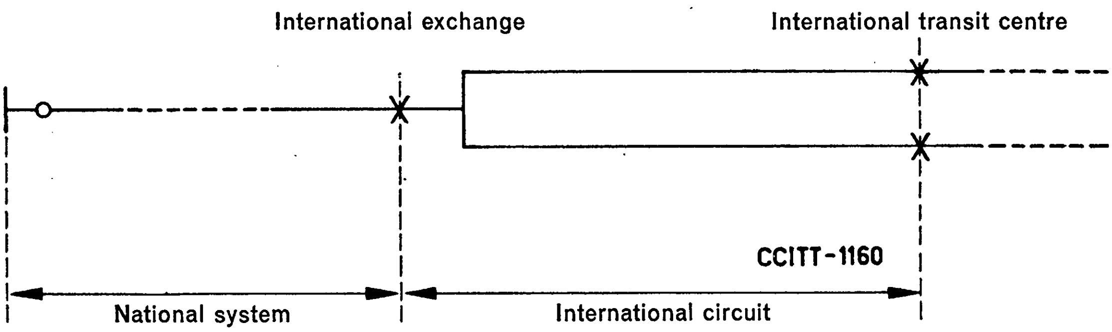
Another method of calculating the probability of intelligible crosstalk using Monte Carlo methods is described in the C.C.I.T.T. handbook *Transmission Planning of Switched Telephone Networks.*

SECTION 2

General characteristics of national systems forming part of international connections

The following sub-section groups together the recommendations which national systems must conform to if international communications are to be of reasonable quality.

The principles of these recommendations also apply in cases where an international circuit is two-wire switched at one end in an international centre. This case may arise while the C.C.I.T.T. transmission plan is being implemented. The figure below illustrates the arrangement.



Recommendation G.120 (P.20)[1]

TRANSMISSION CHARACTERISTICS OF NATIONAL NETWORKS

A. Application of C.C.I.T.T. recommendations on telephone performance to national networks

The different parts of a national network likely to be used for an international connection should meet the following general recommendations:

1. The national sending and receiving systems should satisfy the limits recommended in:

- Recommendation G.121 (P.21) as regards reference equivalent;
- Recommendation G.133 (P.15) as regards group-delay distortion;
- Recommendation G.122 as regards balance return loss and transmission loss;
- Recommendation G.123 for circuit noise.

Note. — Reference should also be made to Recommendations G.112 (P.12) and G.113 (P.13).

[1] Former Recommendation P.21 of Volumes V and V *bis* of the *Red Book* amended at Mar del Plata, 1968.

The Recommendations (series G) referred to in this text appear in Volume III of the *Green Book*; references are also given to those Recommendations which likewise appear in Series P in Volume V of the *Green Book*.

2. Long-distance trunk circuits forming part of the main arteries of the national network should be high-velocity propagation circuits which enable the limits fixed in Recommendation G.114 (P.14) to be respected. They should conform to Recommendations G.151 and G.152.

Loaded-cable circuits should conform to Recommendation G.124 and carrier circuits to Recommendation G.123.

3. National trunk circuits should have characteristics enabling them to conform to Recommendations G.131, G.132 and G.134 in the *Green Book*, Volume III, Section 1 as regards the other characteristics of the four-wire chain constituted by the international telephone circuits and the national trunk extension circuits.

4. International centres should satisfy Recommendation Q.45 in the *Green Book*, Volume VI.

National automatic four-wire centres should observe the noise limits specified in Recommendation G.123, C.

Manual telephone trunk exchanges should satisfy Recommendation P.22.

Information on the transmission performance of automatic local exchanges is given in the new C.C.I.T.T. handbook *Transmission Planning of Switched Telephone Networks*.

B. National transmission plan

Every Administration is free to choose whatever method it considers appropriate for specifying transmission performance and to adopt the appropriate limits to ensure satisfactory quality for national calls, it being understood that in addition the C.C.I.T.T. recommendation relating to reference equivalent [Recommendation G.121 (P.21)] must be satisfied for international calls.

Note. — To meet this twofold condition with respect to national and international calls, each Administration must draw up a national transmission plan, i.e. it must specify limits for each part of the national network. The new handbook *Transmission Planning of Switched Telephone Networks* contains descriptions of the transmission plans adopted by various countries and also some indications concerning the methods that can be used to establish such a plan.

Recommendation G.121 (P.21) (Geneva, 1964; amended at Mar del Plata, 1968, and at Geneva, 1972)

REFERENCE EQUIVALENTS OF NATIONAL SYSTEMS

A. Definition

By definition, the virtual switching points of the national system are the theoretical points at which the system is interconnected to the virtual switching points of the international telephone circuits—i.e. points a and b of Figure 1 of Recommendation G.111 (P.11) and the figure appearing in Recommendation G.122.

All reference equivalents in this recommendation are referred to the virtual switching points of an international circuit at the CT3, when the country is of average size.

B. Maximum nominal sending and receiving reference equivalents

a) *Nominal values for each direction of transmission*

Provisionally, national sending and receiving systems used to set up 97% of actual outgoing or incoming calls in an average-sized country (see Recommendation G.101, B, b) should individually meet both the following requirements:

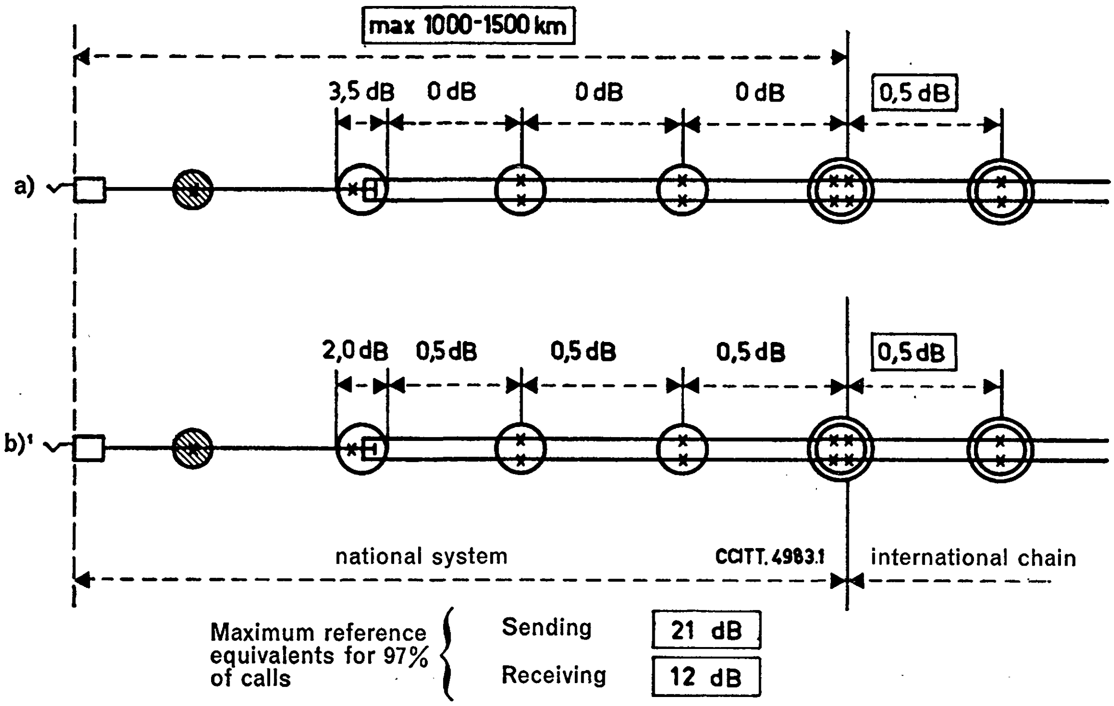


FIGURE 1/G.121 (P.21) — **Distribution of equivalents for an international call, in a country of average size**

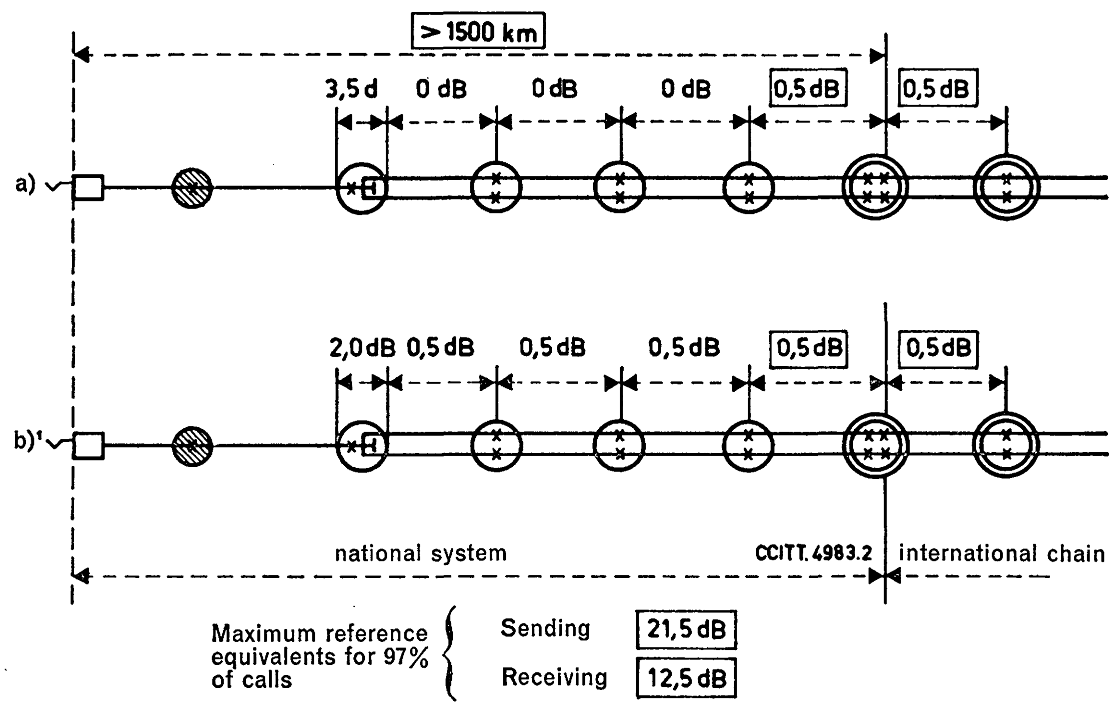


FIGURE 2/G.121 (P.21) — **Distribution of equivalents for an international call, in a large country**

(See on next page the legend for figures 1 and 2)

— the nominal reference equivalent of the sending system between a subscriber and the first international circuit should not exceed 21 dB;

[1] The division of nominal transmission losses is theoretical and can readily be achieved by means of pad-switching, for example.

— the nominal reference equivalent of the receiving system between the same two points should not exceed 12 dB.

In a large country, these limits shall be, respectively: 21.5 dB and 12.5 dB if a fourth national circuit is part of the four-wire chain, or 22 dB and 13 dB if five national circuits form part of the four-wire chain.

In Figures 1/G.121 (P.21) and 2/G.121 (P.21), the numbers in rectangles are figures recommended by the C.C.I.T.T. The others are given only as examples of possible arrangements, subject to Recommendation G.122.

Note 1. — It is possible that, in some existing networks, constructed in accordance with old C.C.I.F. recommendations (see the Appendix to Section 1 (Volume III)), the limits of 21 dB and 12 dB cannot be met immediately, but an attempt should be made to abide by them when telephone sets of a new type are introduced.

Note 2. — The 97% limit refers to the traffic-weighted distribution of calls (including the distribution of the length of subscribers' lines) and is provisional at this stage. It is desirable to use a higher percentage when planning new networks.

Note 3. — The nominal reference equivalents given for national systems are to be based upon the average values of the sensitivities of the transducers of a large number of working telephone sets. As far as carbon microphones are concerned, the average value of the sensitivity of the carbon microphones in the national network is effectively represented by the value of sensitivity corresponding to the long stable period of their lifetime.

As regards systematic differences between actual and nominal sensitivities, if the average sensitivity is obtained by measuring a sample of working sets, any systematic differences will automatically be taken into account. However, if this average value is calculated, then the systematic difference must be part of the calculation.

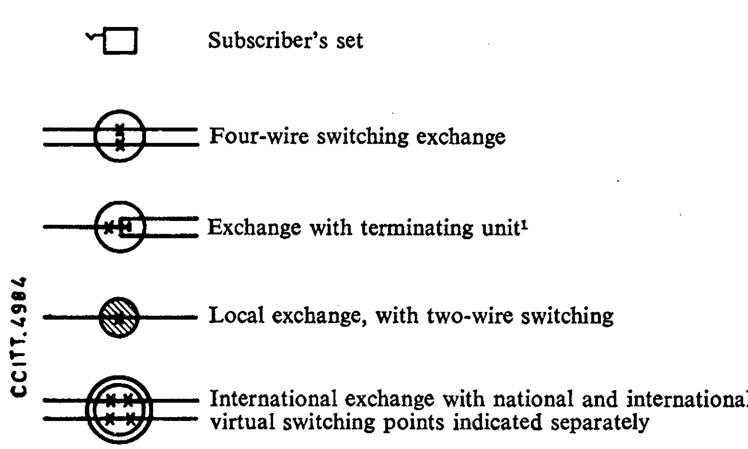
The average values of sensitivity do not include fortuitous variations introduced by subjective methods used when reference equivalents are assessed.

Note 4. — The variations with time of the items of equipment (circuits, etc.) connecting the local exchange with the international centre are not included in the estimated values of nominal reference equivalent.

b) *Difference in transmission loss between the two directions of transmission in national systems*

It is recommended that the absolute value of the difference between loss $t-b$ and loss $a-t$ (see Recommendation G.122) should not exceed 4 dB so that in theory no greater nominal difference than 8 dB could be introduced in international connections.

LEGEND FOR FIGURES 1 AND 2



[1] A switchable pad may also be used at that point to conpensate for losses on the two-wire side, provided that the limits given in Recommendation G.122, A for stability and attenuation are respected.

The following points should be noted:

1. Bearing in mind that most Administrations allocate the losses of their national extension circuits in much the same sort of way (see Annex 1 to this Recommendation), connections set up in practice should not exhibit nominal differences much in excess of 3 dB.

2. As far as speech transmission is concerned, from the studies carried out by several Administrations in 1968–1972, it is clear that for connections with overall reference equivalents falling within the range found in practice no great disadvantage attaches to any reasonable difference in nominal overall reference equivalent between the two directions of transmission.

Supplement No 7 to Volume V is a résumé of the test results from various Administrations concerning the subjective effects of such asymmetry.

3. When devising national transmission plans Administrations should take into account the needs of data transmission between modems complying with C.C.I.T.T. Recommendations (e.g. V.2, V.21, V.23). Annex 2 gives some information on this point.

C. Minimum reference equivalents

Administrations must take care not to overload the international transmission systems if they reduce the attenuation in their national trunk network. This aspect of the problem must be studied separately before any precise recommendation can be prepared.

Provisionally, a nominal minimum value of 6 dB sending reference equivalent referred to the send virtual switching point of the international circuit is recommended in order to control the peak value of the speech power applied to international transmission systems. It should be noted that the imposition of such a limit does not serve to control the long-term mean power offered to the system.

In some countries a very low sending reference equivalent may occur if unregulated telephone sets are used. Nor should the speech power applied to the international circuits by operators' sets be excessive.

D. Determination of the reference equivalents of a national system

Administrations can use various methods to see that the limits for reference equivalents are not exceeded. Thus, for example, simulating networks can be set up representing the main combinations of a subscriber commercial telephone set, subscriber lines, trunk-junctions, and local and trunk exchange equipments, each of these networks representing a complete national sending system or receiving system, which would be compared, in a voice-ear test, with the New Master System for the determination of reference equivalents (N.O.S.F.E.R.) or with a working standard system already compared with N.O.S.F.E.R or S.F.E.R.T.

Another way would be merely to measure the reference equivalent of the telephone apparatus under certain specific conditions. To this reference equivalent would be added the systematic difference between the actual sensitivity of the particular subscriber's telephone set and the nominal value of this sensitivity, the reference equivalent of the subscriber line, the image attenuation (calculated or measured at 800 Hz or at another suitable frequency) of the toll and trunk circuits connecting this set to the international centre, and the composite attenuation (measured or calculated at 800 Hz for a non-reactive resistance of 600 ohms) of the exchange equipments used in the connection between this set and the international centre (including the equipment of the exchange serving the subscriber and that of the international centre). Due account must be taken of the send relative levels at highest order two-wire switching exchanges in the national system when making these calculations.

In any event, however, these calculations ought to be checked by a voice-ear test on the artificial networks, representing the most typical complete national sending and receiving systems.

Administrations may need to calculate the reference equivalent of a subscriber line, as defined in Note 1, for local network transmission planning. The C.C.I.T.T. advises Administrations which do not possess many measurement results to apply the calculation methods described in Section 5.2 of the handbook *Transmission Planning of Switched Telephone Networks.* It is understood that Administrations which have the necessary means to assess the reference equivalent of the various types of lines used by them, with the telephone sets of the types used in their networks, may in all cases continue to apply any simple calculation methods which they may have already developed.

Note 1. — It is assumed that the reference equivalent has the same value q at the sending and receiving ends of a subscriber line, defined by: $q = Q - Q_0$

where Q is the overall reference equivalent of the line and of a subscriber set and Q_0 is the reference equivalent of the same set, without a line;
It is also assumed that the required precautions have been taken to assess separately the effect of the variations in the feed current.

Note 2. — Annex 2 to Question 3/XVI relates to the possible effect of the position of the zero relative level point in a national network on the actual values of the reference equivalents of the national send and receive systems.

Note 3. — The N.O.S.F.E.R. has replaced the Master Reference System (S.F.E.R.T.) used in the C.C.I.T.T. Laboratory before transfer to the new I.T.U. building. It, and other reference systems, are described in Recommendation P.42 *Green Book,* Volume V).

E. Sidetone reference equivalent

Every precaution must be taken to avoid further transmission impairment in communications which reach the reference equivalent and noise limits.

Tests have shown that in these unfavourable conditions the sidetone reference equivalent (for speech) should be at least 17 dB.

In fact, this value cannot be achieved without additional networks, which increase line costs and are only justified when the subscriber has to exchange calls frequently in very bad conditions. In most cases, values between 7 and 10.5 dB are to be expected.

Note 1. — Strong sidetone (corresponding to a low value for sidetone reference equivalent) impairs transmission in two ways. At the sending end, a subscriber who hears himself clearly is tempted to lower his voice; at the receiving end the room noise is transmitted as sidetone to the ear of the listener thus increasing the total noise received.

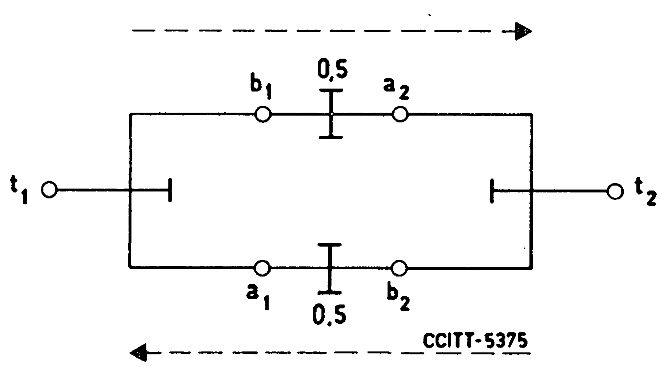
Note 2. — Even when the value of 17 dB is attained, room noise at the expected levels may still have an adverse effect. See Recommendation G.113 (P.13).

ANNEX 1

[to Recommendation G.121 (P.21)]

Evaluation of the nominal differences of loss between the two directions of transmission

a) Consider an international connection between primary centres in two countries, established over one international circuit as shown below:



The nominal overall losses in each of the two directions of transmission are:

$1 \rightarrow 2 = t_1b_1 + 0.5 + a_2t_2$ dB

and $2 \rightarrow 1 = t_2b_2 + 0.5 + a_1t_1$ dB

Where *at* and *tb* are defined as in Recommendation G.122, so that the difference between the two directions is

$$(t_1b_1 - a_1t_1) - (t_2b_2 - a_2t_2)$$
$$= (tb - at)_1 - (tb - at)_2$$
$$= d_1 - d_2$$

in which d signifies $tb - at$.

b) The value in decibels of losses *at* and *tb* for each of several countries is given in the following table together with the corresponding values of *d*, their difference. It will be seen that a maximum nominal difference of 3 dB between the two directions of transmission can arise on connections between any of the countries with a $d = 0$ dB (e.g. Netherlands) and any of the countries with $d = 3$ dB, e.g. North America. It will also be noted that most nominal differences are $d = 0$ dB, so that the nominal differences on connections between the countries concerned is also 0 dB.

c) The nominal differences of loss between the two directions of transmission on international connections between local exchanges and also between subscribers' premises (i.e. telephone instrument disconnected) may also be calculated from the table, but the results will be true only if national two-wire switched trunk-junctions, etc., are nominally symmetrical. This is usually the case.

d) The last column in the table indicates the sum of *tb* and *at*. This value represents that component of the loss *a—t—b* that is attributable to the national transmission plan and if, for example, the loss of the path *a—t—b* from the point of view of stability (or echo) is required, the value in the last column must be augmented by the stability (or echo) balance return loss at *t*.

TABLE

	at	*tb*	$d = tb - at$	$s = tb + at$
*Australia	–0.5	0.5	1.0	0.0
Belgium	3.5	3.5	0.0	7.0
*Denmark	1.9	1.9	0.0	3.8
F. R. G.	3.5	3.5	0.0	7.0
*France	2.2	2.2	0.0	4.4
Hong Kong	1.5	3.0	1.5	4.5
Japan	4.0	4.0	0.0	8.0
*Netherlands	3.5	3.5	0.0	7.0
*New Zealand	–1.5	1.5	3.0	0.0
*North America	–0.5	2.5	3.0	2.0
*Norway	0.5	3.5	3.0	4.0
*Sweden	3.5	3.5	0.0	7.0
Switzerland	3.5	3.5	0.0	7.0
United Kingdom (old)	3.5	3.5	0.0	7.0
United Kingdom (new)	0.5	3.5	3.0	4.0
U.S.S.R.	0.0	0.0	0.0	0.0

Note. — For countries marked * a range of values is appropriate and in each case the nominal minimum values of *at* and *tb* are given. In each case the nominal difference is maintained for all values within the range. For such countries, the indicated sum is the nominal minimum value. North America signifies A. T. & T. and the Canadian Telecommunications Carriers Association. Values shown in decibels.

ANNEX 2

[to Recommendation G.121 (P.21)]

(Contribution of the Netherlands Administration)

The influence of the telephone transmission plan on data transmission

The application of "differential gain" will often result in a higher circuit loss in one direction of transmission and a lower loss in the other one, because the sum of both will be held constant for stability reasons. This means that in international connections with an unfavourable combination of "differential gains" at both ends, one direction of transmission indeed can have an extra loss of 4 dB.

A rough calculation, based on the existing Recommendations and taking into account the following aspects:

— maximum circuit losses in national networks, estimated from national transmission plans (see the handbook *Transmission Planning of Switched Telephone Networks*),
— a reasonable number of international circuits,
— variation of transmission loss of international circuits and national extension circuits (Recommendation G.151),
— the sending and receiving levels for data equipment and the attenuation range indicated for the design (Recommendations V.2, V.21 and V.23),

shows that, in some cases, the maximum loss which can be expected on international connections is such that data transmission may encounter problems.

The introduction of differential gain will influence this situation in an unfavourable way.

Recommendation P.22

MANUAL TRUNK EXCHANGES

A. Operators' positions

The C.C.I.T.T.,

considering,

that it is necessary to reduce as much as possible the disturbance due to room noise as well as to the bridging losses due to operators' sets,

unanimously recommends

1. That the operators' sets used for international telephony should be provided with an arrangement allowing the microphone to be disconnected, this device being preferably a changeover key;

2. That the operator's set while being used on an international telephone call should not cause, in the silent listening position (microphone out of circuit), a bridging loss greater than 0.43 dB at any frequency between 300 and 3400 Hz. To reduce this insertion loss sufficiently (while assuring the operator satisfactory reception), a suitable impedance can be introduced, in the silent listening position, in series with the operator's receiver; alternatively the connection between the operator's receiver and the telephone circuit can be made by means of a transformer of sufficiently high transformation ratio.

Note 1. — It is necessary to ensure that the speech signals of the operators do not overload the amplifiers or modulators of carrier systems. The operators' sets and associated equipment should be so designed that, under service conditions, the operators do not produce a speech volume greater than that of a subscriber situated very close to the trunk exchange considered. When Administrations put any new type of operator's set into service they must check that this is still so.

Note 2. — The limits for reference equivalent on an international telephone call between two operators or between an operator and a subscriber are being studied. The values previously recommended will be found in the Appendix to Section 1 of Volume III of the *Green Book*.

B. Supervisors' desks

The C.C.I.T.T.

recommends unanimously

1. that the equipment of the supervisor's desk should allow the supervisor who is using the desk:

a) to listen on the circuits,

b) to listen on the operators' sets,

c) to listen on the order wires,

d) to be connected with the section supervisors;

2. that the desk should be provided with a clock;

3. that the equipment of the desk and the circuit of the operators' sets should be such that no indication of any nature can reveal to an operator that she is being observed from the supervisor's desk;

4. that where the trunk operator calls a subscriber or an exchange by automatic routing, the supervisor's desk equipment should permit verification of the correctness of the dialled impulses.

The C.C.I.T.T.,

considering too,

that observation on a given circuit by the supervisor's desk is in general of a prolonged character and that supervisors' desks at international terminal exchanges exercise this supervision simultaneously; that, consequently, it is appropriate, from the point of view of insertion loss caused by observation, to be more severe in the case of observation on the part of the supervisor's desk than in the case of supervision by an operator,

unanimously recommends

1. That the additional bridging loss caused by observation on the part of the supervisor's desk of a circuit or of an operator's set should in no case exceed the value of 0.26 dB at any frequency effectively transmitted by the trunk circuits (any frequency between 300 and 3400 Hz);

2. That it is, furthermore, desirable to reduce to as small a value as possible the bridging loss caused by observation, for example by using, if need be, an amplifier.

C. Arrangements for conference calls

The arrangements for conference calls should satisfy the following provisional recommendations:

a) *Setting-up and supervision of conference calls*

Supervision and determination of chargeable time of a conference call should always be the responsibility of a special trunk operator attached to the exchange, of those at which the conference call equipments are installed which, by agreement between the Administrations concerned, is the master exchange.

On being requested to do so by this special trunk operator, the trunk operators at the exchanges concerned should be able to swiftly insert the conference call equipments either automatically or manually (if manual, this plays no part in the operating procedure).

This special trunk operator has, on her position, the necessary means of calling individually the various trunk exchanges concerned, of receiving the clearing signals, of reconnecting the subscribers of the local network to the circuits concerned in the normal manner, and of supervising the conference call.

b) *Connecting equipment for interconnecting several long-distance international telephone circuits and several local circuits*

The connecting equipment for conference calls should permit interconnection of two-wire or four-wire circuits without any change in setting up the circuits; the connecting equipment should equally permit two-wire or four-wire subscriber lines to be connected to the international circuits.

The loss at the frequency of 800 Hz of two international circuits interconnected by means of the connecting equipment should not exceed 11.3 dB.

The reference equivalent of a conference call between any two subscribers should not exceed the value prescribed for a normal call (see Recommendation P.11).

The additional attenuation distortion introduced by the connecting equipment in the various paths should be as little as possible.

The connecting equipment should not noticeably reduce the stability of the interconnected circuits.

Where special microphones or loudspeakers are used in the subscribers' sets, separate lines should preferably be used for sending and receiving and precautions should be taken against the effect of acoustic coupling between microphones and loudspeakers.

The power output of the microphones and special amplifiers in the subscribers' stations should not exceed that given by the normal microphones of subscribers' sets in order to avoid overloading the repeaters in circuit.

At any receiving position the power from any of the various sending positions should be roughly equal.

SECTION 3

Subscribers' lines and sets

Recommendation P.31

CONDITIONS WHICH SHOULD BE SATISFIED BY SUBSCRIBERS' STATIONS USED WITH INTERNATIONAL CIRCUITS LEASED FOR PRIVATE PURPOSES

The C.C.I.T.T. is at present studying the conditions imposed generally on the sensitivity of local sending and receiving telephone circuits[1]. Until the results of this study become available, Administrations should refer to the recommendation below, which lays down the conditions which should be satisfied by subscribers' stations used with international circuits leased for private purposes.

The C.C.I.T.T.,

considering

that the sets connected to a leased international telephone communication channel should in no case be made generally available for public use and that the leased circuit should in no way be given over to a third party,

unanimously recommends

that it is desirable for leased circuits to terminate, at the subscriber's premises, in installations of which the equipment is forbidden to be used on these circuits except under the conditions set out in the rental agreement;

considering, too,

that connections set up over leased circuits should satisfy the same electrical conditions as commercial connections between subscribers,

unanimously recommends

1. that it is desirable for Administrations to forbid, wherever possible, the use of microphones giving greater power output than that given by normal microphones and also the use of special receivers;

2. that it is desirable for Administrations to reserve themselves the right to verify by means of volume meters, that the volume transmitted over a leased telephone circuit does not reach an excessive level;

3. that, where Administrations authorize the use of receiving amplifiers, it is desirable that the gain given by this apparatus should be limited so that it is not possible for the user to overhear, by means of crosstalk, conversations on neighbouring circuits;

4. that it would be desirable for the above recommendations to be applied to all telephone sets used on international connections as well as those used on leased international telephone circuits.

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[1] See Question 10/XII

Recommendation P.32

DEVICES FOR RECORDING MESSAGES OR TELEPHONE CONVERSATIONS

The C.C.I.T.T.,

considering

that only Administrations are in a position to decide whether to allow in their respective networks devices for recording messages or telephone conversations;

that, where certain Administrations have decided to permit these, they would be interested to know the essential technical clauses to be imposed upon such recording equipment,

unanimously recommends

that the essential technical characteristics that can be recommended for these devices for recording messages or telephone conversations are as follows:

The devices for recording messages or telephone conversations have three applications:

a) such a device can serve as an auxiliary in a telephone installation to record the conversation exchanged by the calling subscriber with his correspondent;

b) such a device can also, in the absence of the called subscriber, record the message from the caller after indicating by means of a suitable phrase that the called subscriber is absent but that the recording of the conversation is going to take place;

c) such a device can be used on supervisors' desks in local or trunk telephone exchanges.

In order that such apparatus shall have no harmful effect on the plant and shall not adversely affect the transmission quality, it is desirable that it should comply with a certain number of conditions which are enumerated below; the conditions which are mentioned are not general but apply to each particular method of use.

1. Input impedance. — The input impedance of the recording device, connected in parallel with a connection on which a conversation is taking place, should be high enough at all frequencies above 300 Hz to ensure that the bridging loss does not exceed 0.5 dB for any amplitude of speech signal likely to occur during a conversation.

Whenever the recording device is, in the absence of the subscriber, substituted for the set, it should present an input impedance close to that of the subscriber's set for which it is substituted.

2. The recording device should be well balanced to earth so that its connection to the line shall not produce or aggravate any noise disturbance on the telephone circuit; furthermore the power supplies to the recorder should not produce any disturbance on the telephone circuit.

3. There should be sufficient margin between the background noise of this recording device and its overload point so that the weakest speech sound to be recorded should be at least 20 dB above the background noise, while at the same time the highest level of speech should not overload the device. Alternatively the recording device may contain a volume compressor which, on the one hand amplifies the very weak speech sounds so that they reach a level of 20 dB above the background noise of the recording device but which, on the other hand, attenuates the very loud speech sounds so that they do not cause overloading during recording.

4. The recording device should reproduce a conversation recorded on a circuit of total reference equivalent, subscriber to subscriber, corresponding to an attenuation between subscribers' sets of 29 dB with sufficient clarity considering the quality of telephone systems and with a subjective acoustic intensity comparable to that given by a telephone receiver connected to the same circuit.

5. In order to preserve the secrecy of telephone conversations, a conversation recorded with the maximum possible gain should be quite unintelligible if the speech volume is lower than 55 dB at least below reference volume.

6. If the recording device contains, after the amplifier, a listening arrangement to monitor the recording of the conversation when the subscriber is present, it should, so as to avoid acoustic couplings in this listening arrangement, employ only a headband receiver, this being connected by means of a fixed pad so as to provide a subjective acoustic intensity at the most equal to that given by the receiver of the subscriber's telephone equipment connected to the line.

7. Where the recording device is such that, when the called subscriber is absent, it connects itself automatically in place of the subscriber's set, it is necessary for the device to send out a reply signal on being called and then to give a spoken announcement (film or disc for example) to make it known to the calling subscriber that his correspondent is absent but that a recorder is ready to take a message. This announcement should be sent out at a volume not exceeding values normally encountered in telephone conversations.

8. In order to be able easily to disconnect the recording device when it is out of order and so avoid any possible disturbance to the conversation, it would be useful to provide a key to break both wires of the connecting circuit; on the other hand, so as to limit any danger due to an insulation breakdown between the power supply circuits and the connecting wires, it is desirable to insert protectors in accordance with the normal practice in the countries concerned. Finally, to avoid giving rise to a calling signal at the exchange when the device is connected by means of the isolating key, it is necessary to insert in each leg of the circuit either a capacitor of appropriate maximum capacitance and designed so as to avoid distortion of automatic dialling impulses, or any other device which fulfils this purpose.

9. The general arrangement of recording devices should conform to the general installation conditions in force.

Recommendation P.33 (Mar del Plata, 1968; amended at Geneva, 1972)

SUBSCRIBER TELEPHONE SETS CONTAINING EITHER LOUDSPEAKING RECEIVERS OR MICROPHONES ASSOCIATED WITH AMPLIFIERS

Since an increasing number of loudspeaker sets is being used in the telephone network; and

in view of the complex nature of the effect of factors introduced by these equipments on telephone transmission performance, and

in order to help Administrations to determine the conditions in which the use of such equipment may be authorized in telephone networks,

the C.C.I.T.T. makes the following provisional recommendation:

In order to avoid overload of carrier systems, the mean long term power of speech currents should not exceed the mean absolute power level assumed for system design. In Recommendation G.223 the value adopted for this mean power level is −15 dBm0 (mean power = 31.6 microwatts). Furthermore, in order to avoid excessive crosstalk from high level speech currents and/or inadequate received volume from low level speech currents, care should be taken to ensure that the variation of speech currents is not substantially greater than that from modern telephone instruments.

Administrations should take the necessary precautions so that the person listening may be able to break the sending circuit if oscillations occur or devise suitable methods so that a device controlled by the voice may prevent oscillations.

SECTION 4

Transmission standards

Recommendation P.41

DESCRIPTION OF THE A.R.A.E.N.

A set of equipment which is kept in the C.C.I.T.T. Laboratory is known, for historical reasons, as the A.R.A.E.N. (Reference apparatus for the determination of transmission performance ratings). Actually, A.R.A.E.N. is used in the constitution:

— of N.O.S.F.E.R., for the determination of reference equivalents (see Recommendation P.42),

— or of S.R.A.E.N., for the determination of the A.E.N. (see Recommendation P.44).

The A.R.A.E.N. comprises three main parts:

1. The transmission path proper, subdivisible into sending end, junction, and receiving end;
2. A centralized apparatus for the supply of room noise and intercommunication facilities;
3. A calibration equipment arranged to facilitate the proper maintenance of the reference system.

The transmission path incorporates a moving coil microphone, send and receive amplifiers, junction attenuators and four moving coil receivers. There is a junction filter having a transmission characteristic similar to that of an average carrier channel (4-kHz carrier spacing). This filter can be inserted either in the transmission path of the A.R.A.E.N. or in the test telephone circuit. The complete transmission path, when the filter is switched out of circuit, is designed to reproduce the transmission characteristics of a free field air path, one metre long, the air path being assumed to be used with monaural listening. Normal settings of the send and receive amplifiers are such that these characteristics are reproduced with 30 dB non-reactive attenuation in the junction.

Room noise is produced, as a continuous-spectrum sound, by amplifying the random fluctuations of the anode current of a gas-filled triode. The spectrum is adjusted to the average observed at telephone locations.

Calibrated probe-tube microphones are provided as secondary standards and are for use with:

a) an artificial ear for observing the performance of the moving coil receivers, and

b) a closed coupler for observing the performance of the microphones.

Rayleigh discs and a standing-wave tube are provided as a primary standard and used to calibrate the probe-tube microphones. An oscillator, milliammeters and ancillary equipment complete the electro-acoustic testing gear.

Supplement No. 9 to *White Book*, Volume V, describes the method for the absolute calibration of the A.R.A.E.N. in the C.C.I.T.T. Laboratory. The main purpose of the calibrations effected in the Laboratory is to verify the stability of moving coil microphones and that of the receivers under specified conditions of measurement.

This system is completely defined in documents held by the C.C.I.T.T. Secretariat and the C.C.I.T.T. Laboratory; furthermore the mimeographed document entitled "Draft summary of instructions for the use and maintenance of the C.C.I.F. Laboratory" gives a shortened description of the equipment and its method of use.

A. Transmission path

This transmission path consists essentially of the items whose characteristics are given in Table 1 and which are interconnected according to the arrangement in Figure 1/P.41 by means of the junction switching panel.

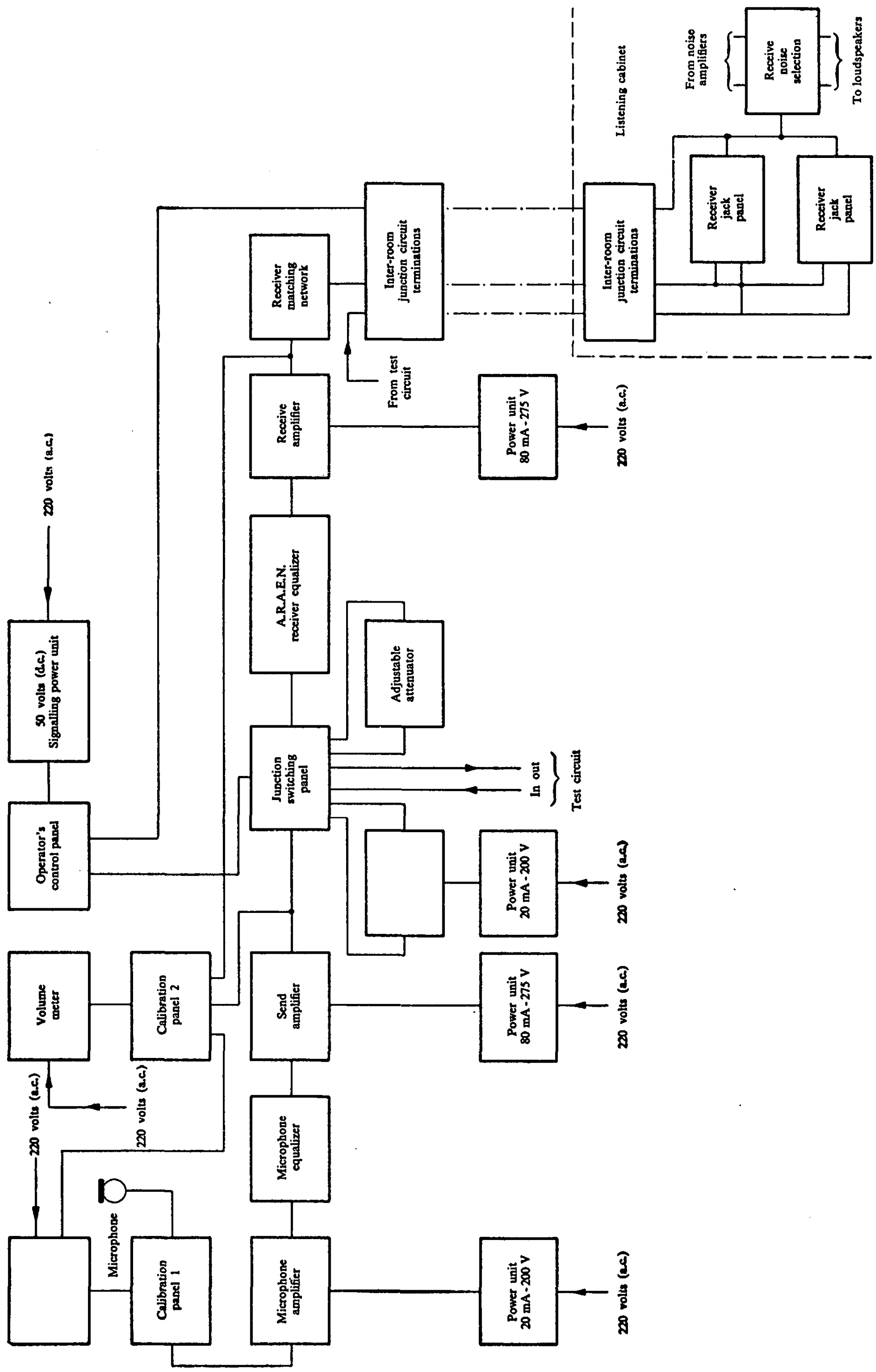


FIGURE 1/P.41. — **Schematic diagram of the reference equipment for the determination of A.E.N.**

TABLE 1

Item	Performance characteristics
Microphone Standard Telephone and Cables type 4021 E	Attenuation distortion +2.5 dB; 80-6000 Hz (equalized to still closer limits by separate equalizer circuit)
Microphone amplifier	Input impedance: high as compared with 20-ohm microphone Output impedance: 600 ±50 ohms over range 80-600 Hz A fixed value of gain is provided Gain without feedback: 68 dB Gain with feedback: 47 ±0.2 dB over range 80-600 Hz Maximum noise level at output (input closed with 20 ohms): −82 dB rel. to 1 volt across 600 ohms
Send (or receive) amplifier	Input and output impedances: 600 ±50 ohms Gain without feedback: 100 dB Maximum gain with feedback: 64 dB Attenuation distortion: ±0.3 dB over range 50-6000 Hz Range of gain control: 48 dB (in 0.2 dB steps)
Telephone receiver Standard Telephone and Cables type 4026 A	Attenuation distortion (on real ear) ±5 dB over range 80-6000 Hz (before equalization)

B. Equipment for supply of room noise and intercommunication circuit

This equipment, of which Figure 2/P.41 shows connections in schematic form, comprises:

1. A source of noise (gas-filled triode);
2. Power amplifiers for feeding loudspeakers;
3. A sound-level meter, which can be switched to the various listening points; and
4. A loudspeaking telephone equipment to facilitate intercommunication between members of the testing crew.

C. Calibration equipment

The general arrangement of the electro-acoustic gear is shown in Figure 3/P.41. The method of using this equipment at the C.C.I.T.T. Laboratory is described in the Supplement No. 9 to the *White Book*, Volume V.

The Rayleigh disc is suspended in the centre of the standing-wave tube and optical means are provided at the operator's desk for observing its angular deflection (from which sound pressures at the end of the tube can be calculated). The probe of the microphone under test is inserted in a hole in a plate closing one end of the standing-wave tube; the other end is closed by a moving-coil receiver fed from an oscillator at the operator's right hand. The output of the probe-tube microphone is read on a meter mounted in front of the operator.

Calibration of the probe-tube microphone is effected by adjusting the frequency of the oscillator to produce a stationary wave in the tube and give simultaneous maxima of the deflection of the Rayleigh disc and of the output of the microphone. At any one setting of the length of the standing-wave tube, frequencies for calibration can be used which are those of the fundamental mode of resonance in the tube (about 100 Hz) and any odd harmonic thereof. To obtain calibration points at other frequencies it is necessary to alter the length of the tube; means are provided for doing so, but it will not be necessary to use this facility for routine checks of the sensitivity of the probe-tube microphones.

The rack on the left of the operator's desk contains equipment for checking the sensitivities of the microphones and receivers of the A.R.A.E.N. against a calibrated probe-tube microphone. The main items of equipment for this work are:

Probe-tube microphone.—For calibration of the A.R.A.E.N., two microphones and one amplifier and equalizer are provided; the equalized frequency characteristic of the probe-tube microphone and amplifier is substantially flat from about 80 to 6000 Hz.

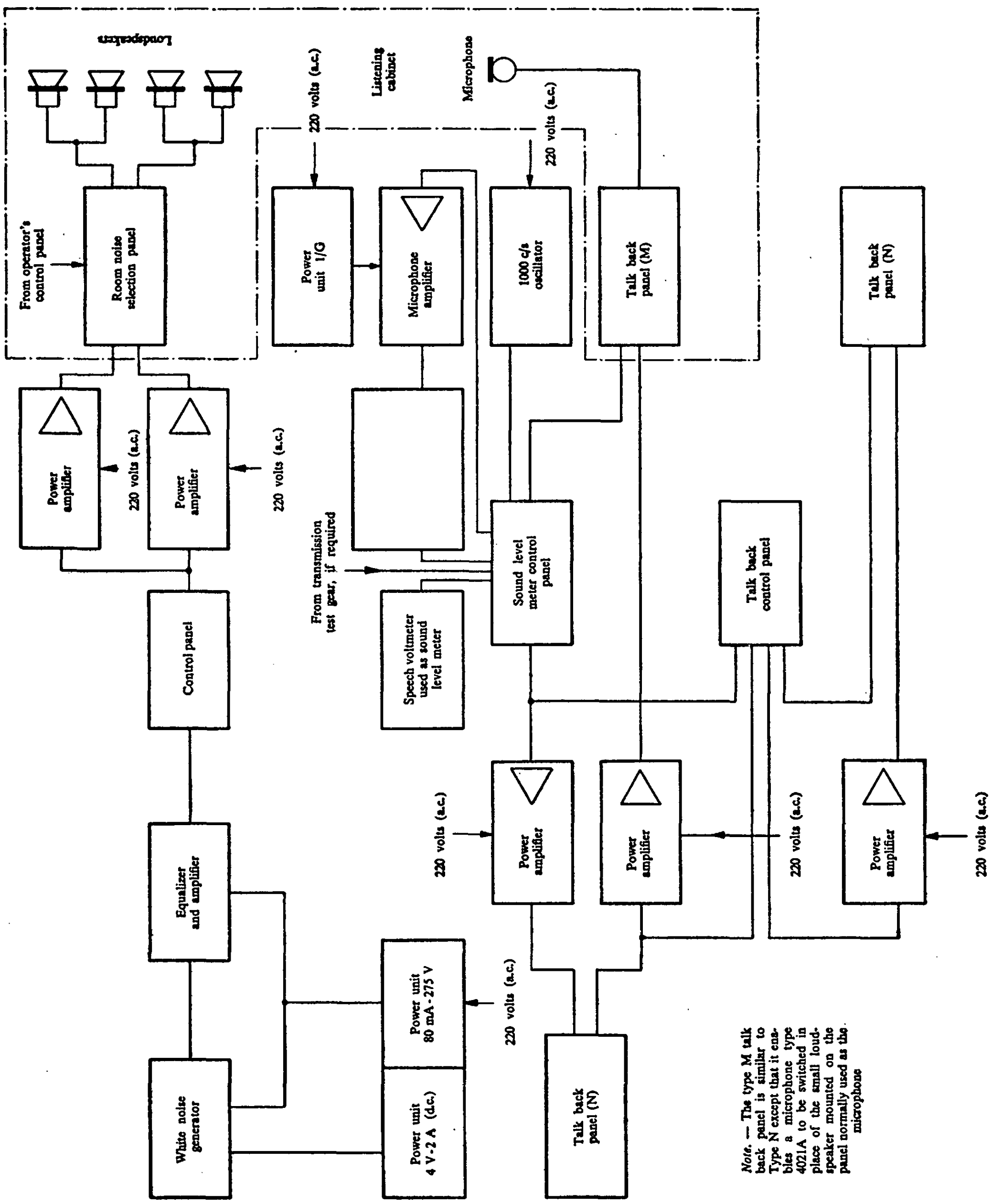


FIGURE 2/P.41. — Noise generation, level measurement and talk-back equipment

Artificial ear.—A device for presenting to a telephone receiver an acoustical load equivalent to that of a human ear, and permitting the measurement of sound pressure at a specified point therein by means of a probe-tube microphone.

Closed coupler.—A small cylindrical chamber closed at one end by a moving-coil receiver (the source of sound) and at the other end by the microphone under test, with means for admitting the tip of a probe-tube microphone for measuring the acoustic pressure. A microphone calibration at constant pressure under specified conditions of test can thus be obtained which is sufficient for detecting any change of sensitivity of the microphone.

A high-grade moving coil milliammeter and a thermocouple milliammeter are associated with the equipment as primary and secondary standards (respectively) for electrical measurements, and arrangements are provided for switching the different items of electrical equipment to facilitate routine calibrations.

Note. — It is sometimes convenient when using a reference telephone system for articulation testing to make a recording of the operator's speech to assist training in correct pronunciation. A recording equipment suitable for use in conjunction with the microphone and receivers of the A.R.A.E.N. exists and has been sent to C.C.I.T.T. Laboratory. This equipment should not be regarded as forming a specific part of the reference system.

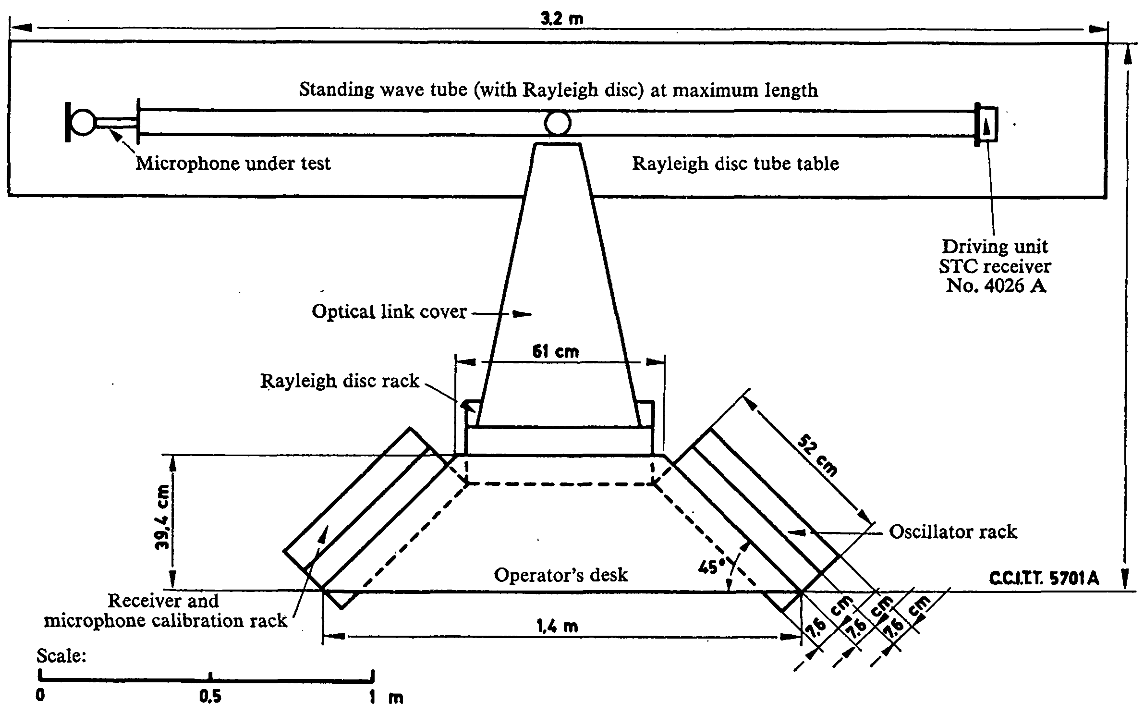


FIGURE 3/P.41. — **Plan of microphone and receiver calibration equipment A.R.A.E.N.**

D. THEORETICAL EFFICIENCY OF THE COMPLETE A.R.A.E.N.

The construction of the A.R.A.E.N. is such that, in the standardized position of the microphone (defined below), the whole system included between the talker's mouth and the listener's ear represents from the acoustical standpoint the equivalent of a one-metre air path. Thus the A.R.A.E.N. represents that portion included between a point situated at 33.5 cm from the talker's lips (the position of the centre of the microphone)[1] and the head of the listener, the latter being situated at a point one metre from the talker's lips and facing the talker.

[1] The rim of the baffle plate of the microphone is situated at about 30.5 cm from the talker's lips.

Neglecting the effect upon the sound field caused by the obstruction effect of the listener's head, the difference in acoustic pressure between these two points is theoretically:

$$20 \log_{10} \frac{100}{33.5} = 9.5 \text{ dB}$$

Taking into account the obstruction effect caused by the listener's head according to the curve *b* of Figure 4/P.41, the values of Table 2 are obtained:

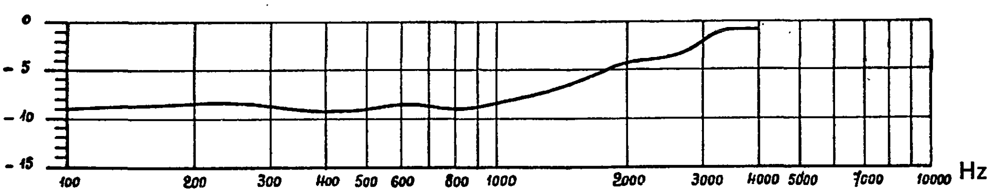
TABLE 2

Frequency	Pressure increase due to obstruction effect	Theoretical loss
100 Hz	0 dB	9.5 dB
300 Hz	0 dB	9.5 dB
1000 Hz	1 dB	8.5 dB
2000 Hz	4.6 dB	4.9 dB

Sensitivity of the A.R.A.E.N. sending end. — The sensitivity of the sending end of the A.R.A.E.N. has been fixed at a value permitting the control of the speaking level by means of a specified speech voltmeter (see Recommendation P.52) connected to the output of the sending system.

The speech voltage applied to the input of the junction and read on this speech voltmeter is one volt when the operator speaks at the "A.R.A.E.N. reference vocal level" (see Recommendation P.45). Under these conditions the acoustic pressure applied to the diaphragm of the microphone is 0.1 Pa.

Sensitivity in decibels relative to Pa/Pa

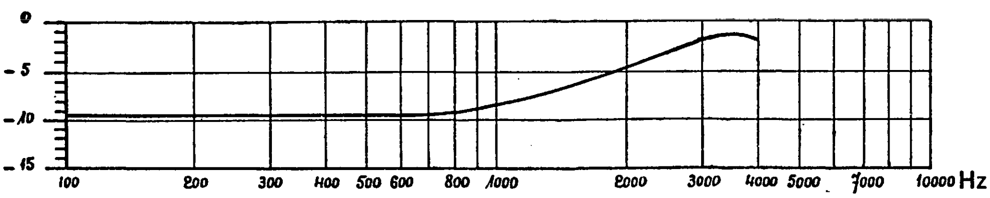


Frequency

(*a*) Overall working characteristic of the A.R.A.E.N. taken with microphone No. 1284 (type 4021E) and a typical receiver (type 4026A) the band-pass filter being out of circuit.[1]

Adjustment $\left\{ \begin{array}{l} \text{Send amplifier "normal"} \\ \text{Junction: 30 dB} \\ \text{Receive amplifier "normal" +1 dB} \end{array} \right.$

Sensitivity in decibels relative to Pa/Pa



Frequency

(*b*) Characteristic of transmission in free air over a distance of 1 metre — "conversation distance", account taken of the distortion of the acoustic field caused by the presence of the listener's head (theoretical definition of the frequency characteristic of the A.R.A.E.N. set up in accordance with the adjustments shown above).

FIGURE 4/P.41. — A.R.A.E.N.

[1] The effect of the filter is to cause a sharp cut-off below 300 and above 3400 Hz; between these frequencies the loss introduced is less than ± 0.5 dB.

Sensitivity of the A.R.A.E.N. receiving end. — The sensitivity of the receiving end has been determined conventionally such that the condition indicated above (for the "air to air" efficiency of the A.R.A.E.N.) is complied with for a junction attenuation equal to 30 dB.

Table 3 gives the values of acoustic pressure (in decibels relative to 0.1 Pa) produced by a receiver when a level of −30 dB relative to 1 volt is applied to the input of the receiving system—i.e. when an acoustic pressure of 0.1 Pa is applied to the microphone.

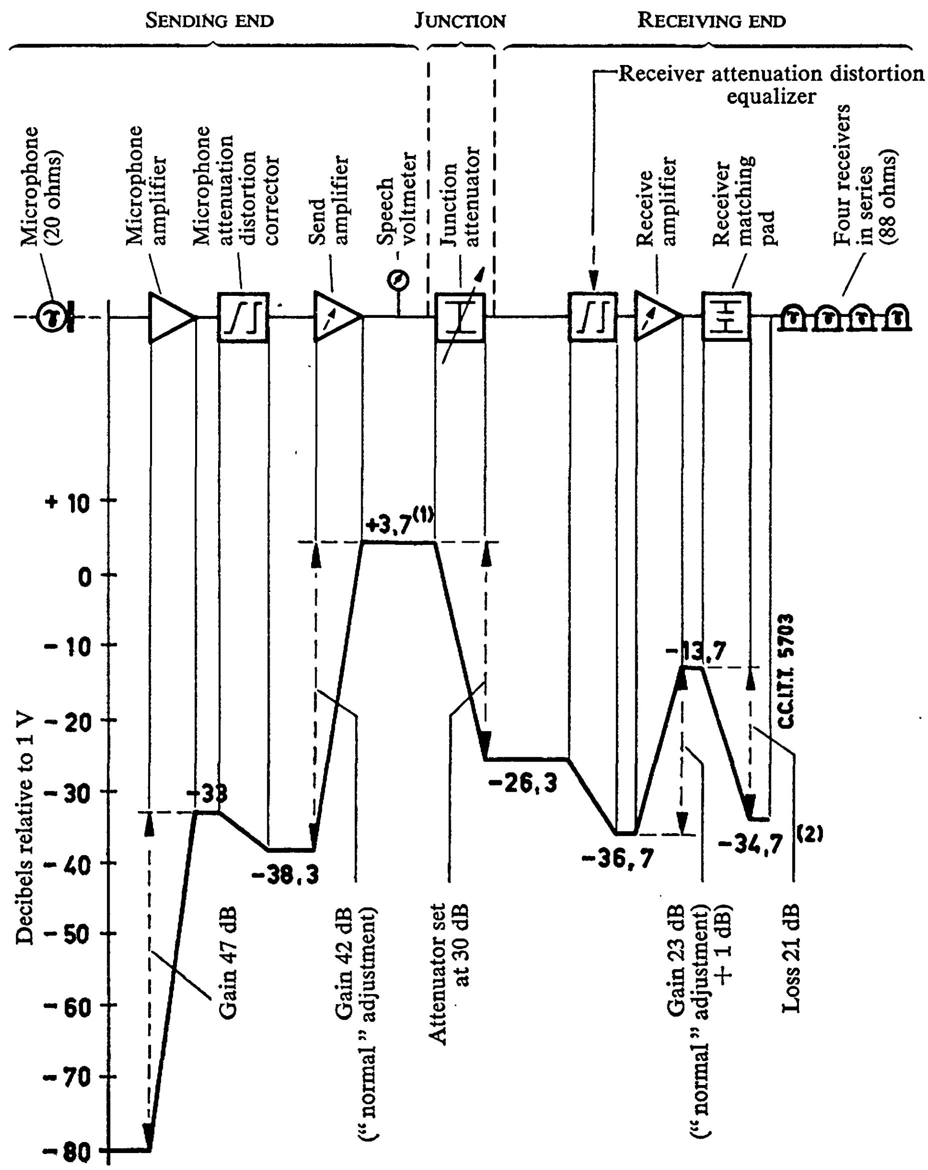


FIGURE 5/P.41. — Diagram showing the levels at various points in the A.R.A.E.N. when a pure tone of 1000 Hz at a level of −80 dB relative to 1 V is applied to the microphone sockets, in the following conditions of adjustment

Send amplifier: "normal"
Receive amplifier: "normal" +1 dB
Junction attenuator: 30 dB

[1] The speech volume is 0 dB (relative to 1 V) a this point when the microphone is connected and the talker speaks at the reference vocal level for the A.R.A.E.N.

[2] With tolerance of ±1.0 dB without the band-pass filter in circuit.

TABLE 3

Frequency	Voltage at the input of the receiving system (output of the junction)	Total loss of the electrical part of the receiving system	Voltage applied to one receiver	Average receiver efficiency	Acoustic pressure produced by one receiver
Hz	dB relative to 1 volt	dB	dB relative to 1 volt	dB relative to 1 dyne/cm²/volt	dB relative to 1 dyne/cm²
100	−30	25.8	−55.8	46.0	−9.8
300	−30	25.2	−55.2	46.1	−9.1
1000	−30	19.5	−49.5	41.2	−8.3
2000	−30	15.4	−45.4	41.4	−4.0

Table 4 below shows the comparison of the theoretical and actual values of the overall attenuation of the A.R.A.E.N.

TABLE 4

Frequency	Overall attenuation of the A.R.A.E.N.		
	Theoretical value	Actual value	Actual value corrected to take account of the position of the probe in the artificial ear [a]
Hz	dB	dB	dB
100	9.5	9.8	9.8
300	9.5	9.1	9.1
1000	8.5	8.3	8.3
2000	4.9	4.0	4.3

[a] This correction is necessary because the value of pressure taking account of the presence (in the acoustic field) of the listener's head is referred to the external opening of the ear canal, whilst in the artificial ear the probe of the microphone is placed at the lower part of the artificial ear cavity; the region corresponding to the external opening of the real ear canal is close to the upper part of the artificial ear cavity. This correction becomes very important at high frequencies. The differences between the measured values (corrected in this way) and the theoretical values are due to small variations in the frequency characteristics of the receivers.

In practice, for the adjustment of the gain of the sending and receiving amplifiers, account must be taken of the differences in the frequency characteristics of the individual microphones and receivers. The C.C.I.T.T. Laboratory is in possession of the necessary documentation for the calculation of these corrections from the small changes in sensitivities of the microphones and receivers as obtained during calibration measurements. Figure 5/P.41 gives a diagram showing the levels at various points in the A.R.A.E.N. when normally adjusted.

Recommendation P.42 (amended at Mar del Plata, 1968)

SYSTEMS FOR THE DETERMINATION OF REFERENCE EQUIVALENTS

Three systems are in existence for the determination of reference equivalents. These three systems should comply with the conditions shown below and are designated as follows:

1. The new fundamental system for the determination of reference equivalents (N.O.S.F.E.R.);
2. Primary systems for the determination of reference equivalents;
3. Working standard systems.

The new fundamental system for the determination of reference equivalents (N.O.S.F.E.R.) is the system used in the C.C.I.T.T. Laboratory. Formerly, reference equivalents were determined by comparison with the European master reference system for telephone transmission (S.F.E.R.T.), defined on pages 27 to 43 of Volume IV of the C.C.I.F. *Green Book.*[1]

Values of reference equivalents determined by comparison, directly or indirectly, with the S.F.E.R.T. remain valid.

In the past, other telephone transmission reference systems were also used; these are described on pages 27 to 43 of Volume IV of the C.C.I.F. *Green Book.*

A. The new fundamental system for the determination of reference equivalents (N.O.S.F.E.R.)

This system consists of the A.R.A.E.N. (described in Recommendation P.41) with the following modifications:

1. *Sending end*

The talking distance (measured between the plane of the guard-ring nearest to the talker's lips) and the centre of the protective cover of the microphone is 14 cm.

An equalizer defined by Figures 1/P.42 and 2/P.42 and Tables 1 and 2 is inserted at the output of the sending amplifier.

The A.R.A.E.N. volume measuring set having the characteristics given in Supplement No. 10, *White Book,* Volume V, is bridged across the output terminals of the N.O.S.F.E.R. sending system.

2. *Receiving end*

An equalizer as defined by Figures 3/P.42 and 4/P.42 and Tables 3 and 4 is inserted at the input of the receiving amplifier in place of the A.R.A.E.N. receiver equalizer[2] (see Figure 1 in Recommendation P.41).

TABLE 1

INSERTION LOSS OF THE N.O.S.F.E.R. SENDING END EQUALIZER
(measured in the C.C.I.T.T. Laboratory between two non-reactive resistances of 600 ohms)

Hz	dB	Hz	dB	Hz	dB
100	12.6	1200	8.1	4000	7.6
200	12.3	1300	7.9	4500	9.6
300	12.2	1400	7.8	5000	12.2
350	11.8	1500	7.5	5500	15.5
400	11.5	1800	7.0	6000	19.0
450	11.1	2000	6.8	6500	21.8
500	11.0	2200	6.7	7000	23.7
550	10.7	2500	6.5	7500	24.0
600	10.5	2700	6.4	8000	23.8
700	10.3	3000	6.2	8500	24.6
800	9.6	3200	6.3	9000	25.8
900	9.1	3400	6.3	9500	27.5
1000	8.7	3600	6.6	10000	28.9
1100	8.3	3800	7.0		

[1] Since the C.C.I.T.T. Laboratory has high quality transmission apparatus (A.R.A.E.N.), it appeared reasonable to keep only one reference system at the C.C.I.T.T. Laboratory which, after appropriate modification, could replace the S.F.E.R.T.; tests have shown this to be possible.

The S.F.E.R.T. is an old system, using parts which are difficult to replace; furthermore, its physical characteristics have been defined arbitrarily. It would therefore be difficult to reconstruct in case of partial or total destruction.

[2] In the present constitution of the A.R.A.E.N., this network fulfils two functions:

a) it corrects the distortion of the A.R.A.E.N. receivers, and

b) it provides the transmission characteristics of a metre-long free air path with allowance made for the distortion of the acoustic field by the presence of the listener's head.

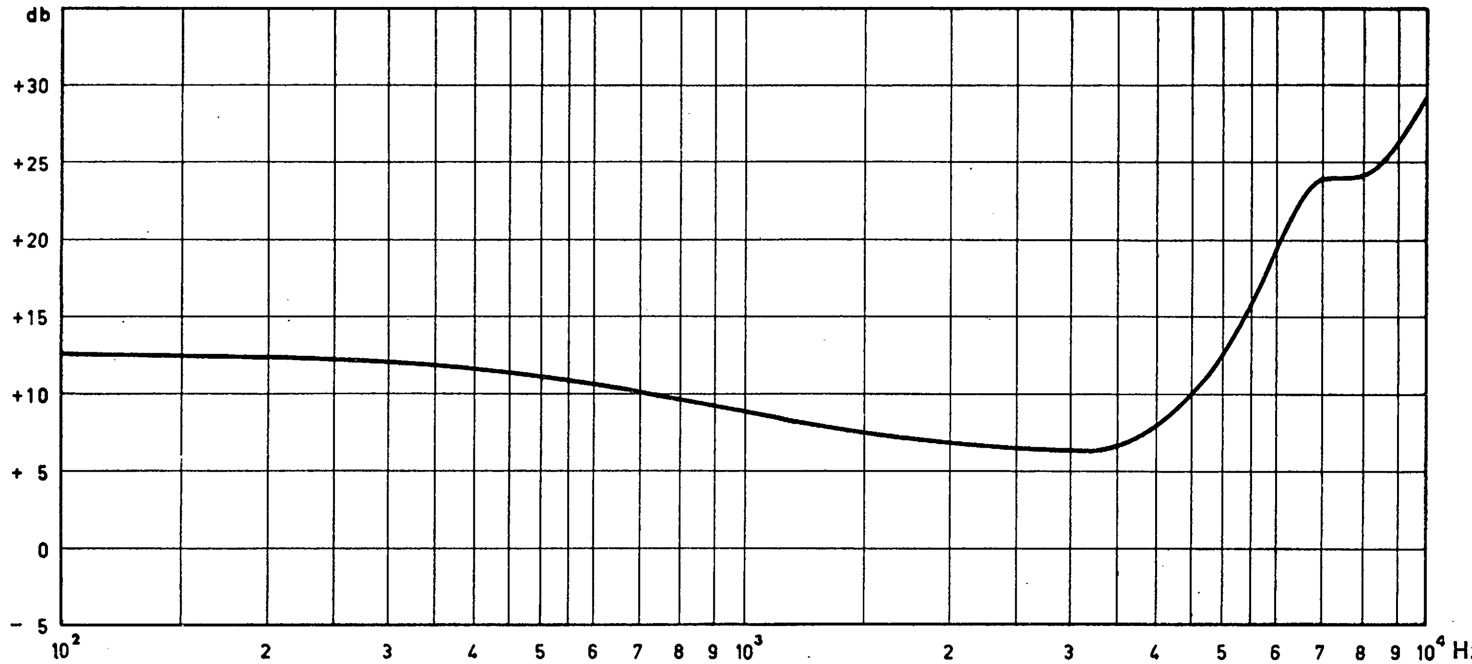


FIGURE 1/P.42 — **Insertion loss characteristic of the N.O.S.F.E.R. sending end equalizer (measured between 600-ohm terminations)**

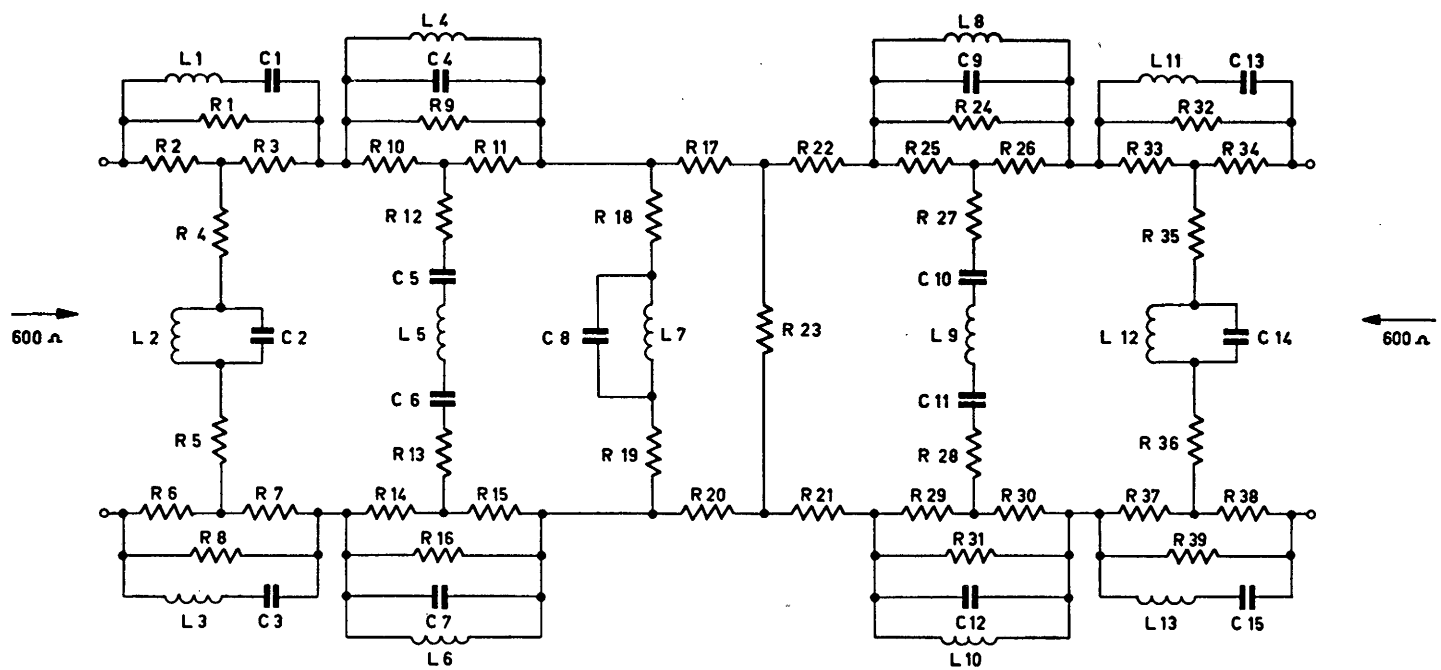


FIGURE 2/P.42 — **Circuit of the N.O.S.F.E.R. sending end equalizer**

TABLE 2

VALUES OF THE COMPONENTS USED IN THE N.O.S.F.E.R. SENDING END EQUALIZER (FIGURE 2/P.42)

R		L					C	
(non-inductive)								
	ohm		mH	d.c. resistance in ohms	Q (at f_r)	f_r Hz		μF
R_1 R_8	372	L_1 L_3	2.265	0.61	106	3 900	C_1 C_3	0.736
R_2 R_3	300	L_2	132.7	32.81	94.5	3 900	C_2	0.0126
R_4 R_5	241.5	L_4 L_6	9.09	2.37	209	10 000	C_4 C_7	0.0217
R_6 R_7	300	L_5	5.01	1.31	205	10 000	C_5 C_6	0.101
R_9 R_{16}	3477	L_7	4.04	1.02	203	10 000	C_8	0.1475
R_{10} R_{11}	300	L_8 L_{10}	4.33	1.10	157	6 700	C_9 C_{12}	0.1298
R_{12} R_{13}	25.88	L_9	23.4	5.54	159	6 700	C_{10} C_{11}	0.0483
R_{14} R_{15}	300	L_{11} L_{13}	5.25	1.34	92.5	3 850	C_{13} C_{15}	0.318
R_{17} R_{20}	13.81	L_{12}	55.8	13.94	88.5	3 850	C_{14}	0.029
R_{18} R_{19}	579							
R_{21} R_{22}	13.81							
R_{23}	6505							
R_{24} R_{31}	765							
R_{25} R_{26}	300							
R_{27} R_{28}	113							
R_{29} R_{30}	300							
R_{32} R_{39}	125							
R_{33} R_{34}	300							
R_{35} R_{36}	722							
R_{37} R_{38}	300							
Tolerances	± 0.5%		± 0.5%					± 0.5%

TABLE 3

INSERTION LOSS OF THE N.O.S.F.E.R. RECEIVING END EQUALIZER
(measured in the C.C.I.T.T. Laboratory between two non-reactive resistances of 600 ohms)

Hz	dB	Hz	dB	Hz	dB
100	28.7	1200	18.3	4000	27.0
200	27.3	1300	18.0	4500	23.3
300	25.8	1400	17.9	5000	20.2
350	24.7	1500	17.8	5500	17.6
400	23.8	1800	17.8	6000	16.4
450	22.2	2000	18.0	6500	18.0
500	21.4	2200	18.6	7000	19.7
550	21.1	2500	19.8	7500	21.3
600	21.2	2700	21.0	8000	22.2
700	20.9	3000	23.3	8500	23.1
800	20.2	3200	25.3	9000	23.8
900	19.7	3400	27.0	9500	24.4
1000	19.0	3600	28.3	10000	24.7
1100	18.7	3800	28.2		

B. NORMAL ADJUSTMENT OF THE N.O.S.F.E.R.

The A.R.A.E.N. having been adjusted to take into account the characteristics of the microphone used, the equalizers described in section A above are inserted and the talking distance is set to 14 cm. The gain of the receiving amplifier is increased by 14 dB with respect to its normal value for the A.R.A.E.N. (normal + 1 dB); the gain of the sending amplifier is not to be changed.

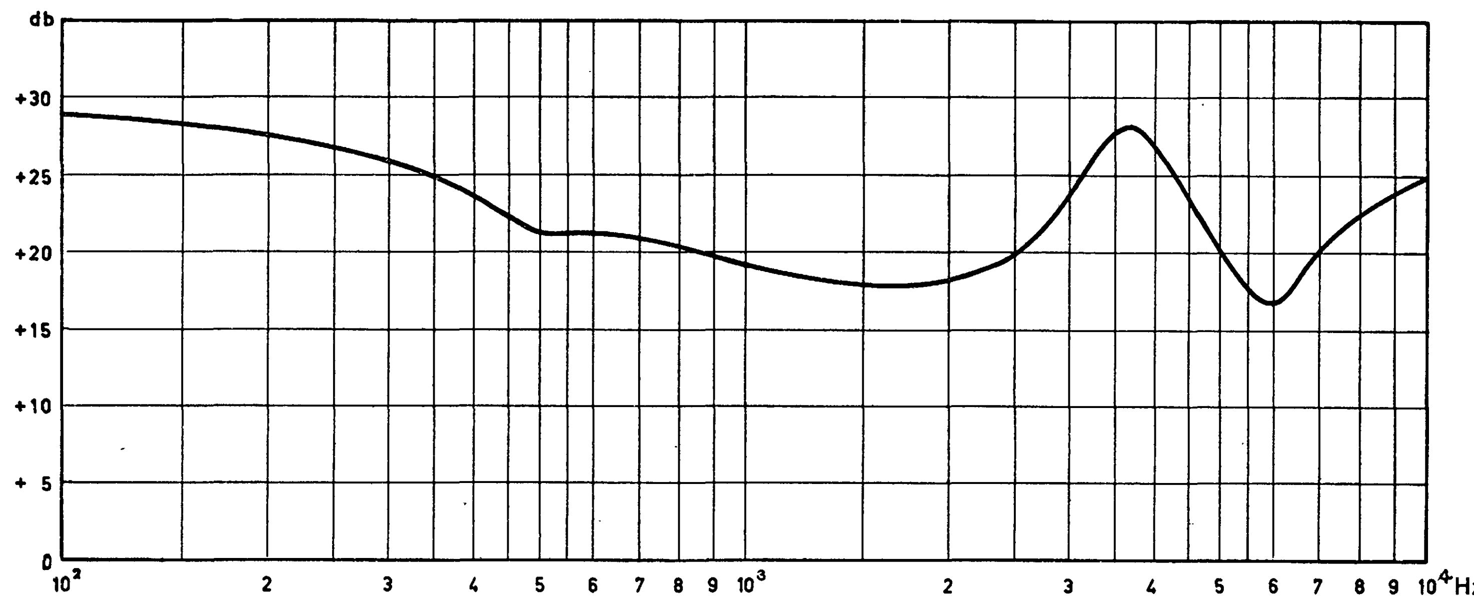


FIGURE 3/P.42 — **Insertion loss characteristic of the N.O.S.F.E.R. receiving end equalizer (measured between 600-ohm terminations)**

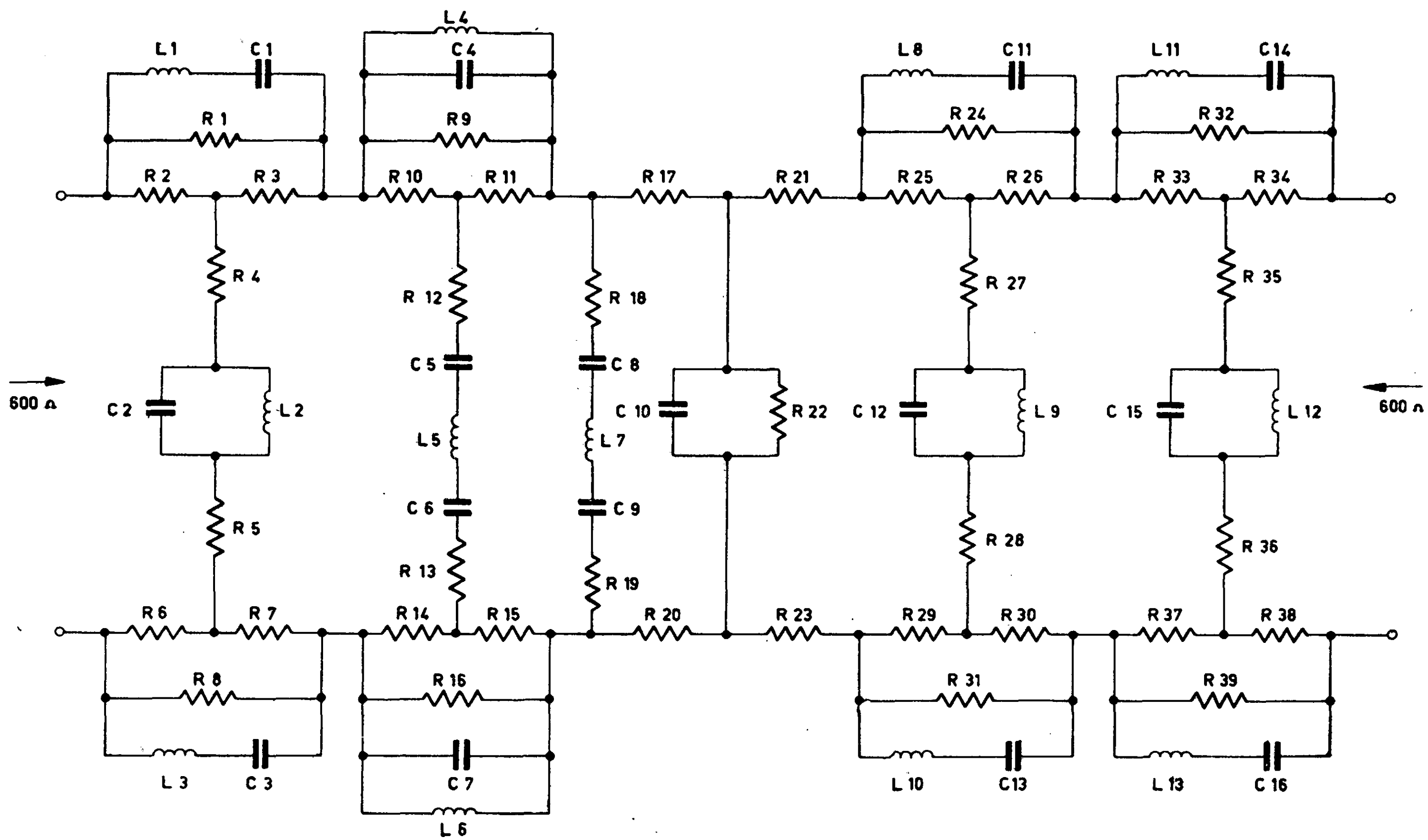


FIGURE 4/P.42 — **Circuit of the N.O.S.F.E.R. receiving end equalizer**

TABLE 4

VALUES OF THE COMPONENTS USED IN THE N.O.S.F.E.R. RECEIVING END EQUALIZER (FIGURE 4/P.42)

R		L					C	
(non inductive)				d.c. resistance in ohms	Q	r		
	ohm		mH		(at f_r)	Hz		μF
R_1 R_8	1071	L_1 L_3	10.64	2.63	50.5	2 000	C_1 C_3	0.5956
R_2 R_3	300	L_2	107.2	29.61	43.6	2 000	C_2	0.05906
R_4 R_5	84	L_4 L_6	7.975	1.90	90	3 700	C_4 C_7	0.2318
R_6 R_7	300	L_5	41.74	11.42	81.6	3 700	C_5 C_6	0.08862
R_9 R_{16}	764.5	L_7	658	167.2	71	3 300	C_8 C_9	0.00709
R_{10} R_{11}	300	L_8 L_{10}	18.27	5.16	122	5 900	C_{10}	0.022
R_{12} R_{13}	117.8	L_9	7.171	1.78	136	5 900	C_{11} C_{13}	0.03984
R_{14} R_{15}	300	L_{11} L_{13}	91.2	23.6	12.1	500	C_{12}	0.1015
R_{17} R_{20}	108.8	L_{12}	200	54.34	11.4	500	C_{14} C_{16}	1.111
R_{18} R_{19}	1650						C_{15}	0.5068
R_{21} R_{23}	108.8							
R_{22}	718.4							
R_{24} R_{31}	411.4							
R_{25} R_{26}	300							
R_{27} R_{28}	218.8							
R_{29} R_{30}	300							
R_{32} R_{39}	100.2							
R_{33} R_{34}	300							
R_{35} R_{36}	898							
R_{37} R_{38}	300							
Tolerances	± 0.5%		± 0.5%					± 0.5%

1. *Sensitivity of the N.O.S.F.E.R. transmitting system*

As indicated above, the adjustment of the sending amplifier gain is not changed when passing from the A.R.A.E.N. sending system to the N.O.S.F.E.R. sending system.

The nominal gain of the microphone pre-amplifier (47 dB), plus that of the sending amplifier (42 dB) independent of the frequency, is equal to 89 decibels.

The sending amplifier gain may be altered slightly to allow for the particular microphone being used.

The amplifier gain is adjusted according to the result of the following operations, described in the table below:

a) Take the arithmetic mean of the three values of the microphone sensitivity (expressed in decibels with respect to 1 volt/dyne/cm^2) measured in a free acoustic field at the frequencies of 100, 300 and 900 Hz; substract 6.1 dB which represents the mean attenuation at these three frequencies for the microphone equalizer.

Hz	Nominal gain of the whole (microphone pre-amplifier plus sending amplifier) (dB) 1	Sensitivity of the microphone (No. 1292) in a free field (dB with respect to 1 volt dyne/cm²) 2	Microphone equalizer attenuation (dB) 3	(2–3) 4	Correction to be made to sending amplifier adjustment (with microphone No. 1292) 5
100	89.0	−85.2	4.5	−89.7	−(−89.2)−89
300	89.0	−81.1	8.0	−89.1	
900	89.0	−83.0	5.8	−88.8	
Average	89.0	−83.1	6.1	−89.2	+0.2

b) Change the sign of the result obtained by a (to obtain the value to which the sending amplifier gain should be adjusted) and substract 89 dB (normal adjustment); in this way the correction to be made to the sending amplifier adjustment is determined.

These corrections were determined by the United Kingdom Post Office. In the particular case of microphone No. 1292, the correction is +0.2 dB. The two sending amplifier gain adjustment controls are therefore set at "normal" and "+0.2".

The Laboratory periodically calibrates microphones on a special closed coupler associated with the Laboratory's calibrating equipment. By these measurements, the stability of the microphones can be checked and their variation (if any) in time determined. If a variation of more than 1 dB is noted, the microphone is rejected. If a variation in the mean sensitivity of less than 1 dB is noted, the transmitter amplifier gain has to be altered.

Table 5 gives the characteristic value defining variations, as a function of frequency, of the sensitivity of the N.O.S.F.E.R. sending system calculated at each frequency on the basis of the mean value (for a certain number of microphones) for the free field sensitivity.

TABLE 5

CHARACTERISTIC VALUES DEFINING THE VARIATION, AS A FUNCTION OF THE FREQUENCY, OF THE SENSITIVITY OF THE N.O.S.F.E.R. TRANSMITTING SYSTEM, CALCULATED FROM THE MEAN SENSITIVITY VALUES OF A CERTAIN NUMBER OF MICROPHONES MEASURED IN A FREE FIELD

Hz	Gain of the electrical part of the sending system (the sending amplifier being adjusted to normal +0.4) 1	Mean sensitivity of a certain number of microphones measured in free acoustic field [a] (dB relative to 1 volt/dyne/cm²) 2	Sensitivity of the sending system in free acoustic field (dB volt/1 dyne/cm²) (1 + 2) 3
80	+73.2	−86.8	−13.6
100	+72.9	−85.6	−12.7
120	+72.4	−84.6	−12.2
200	+70.8	−82.4	−11.6
300	+69.5	−81.6	−12.1
400	+69.6	−81.7	−12.1
500	+70.4	−81.7	−11.3
600	+71.5	−81.5	−10.0
700	+72.6	−82.0	−9.4
800	+73.7	−82.3	−8.6
900	+74.5	−82.7	−8.2
1000	+75.4	−83.4	−8.0
1500	+77.9	−85.8	−7.9
2000	+79.2	−86.6	−7.4
2500	+79.9	−87.4	−7.5
3000	+80.2	−86.5	−6.3
3500	+80.2	−86.0	−5.8
4000	+79.1	−85.9	−6.8
4500	+77.2	−85.6	−8.4
5000	+74.5	−85.4	−10.9
5500	+71.4	−85.9	−14.5
6000	+67.5	−85.6	−18.1
6500	+65.0	−84.3	−19.3
7000	+62.9	−84.7	−21.8

[a] Values extracted from Research Report No. 13200 of the United Kingdom Post Office (April 1950).

Table 6 gives the characteristic values defining the variation, as a function of the frequency, of the N.O.S.F.E.R. sending system sensitivity determined on the basis of the sensitivity of microphone No. 1292, measured in a free field (figures supplied by the United Kingdom Post Office) and also on a closed coupler. Transmitter amplifier gain is adjusted to the value corresponding to this microphone ("normal" +0.2).

TABLE 6

CHARACTERISTIC VALUES DEFINING THE VARIATION, AS A FUNCTION OF THE FREQUENCY, OF THE SENSITIVITY OF THE N.O.S.F.E.R. SENDING SYSTEM, CALCULATED FROM THE SENSITIVITY VALUES OF A GIVEN MICROPHONE (NO. 1292)

Hz	Gain of the electrical part of the sending system (the sending amplifier is set at normal +0.2) 1	Sensitivity of microphone No. 1292 measured in free speech field (dB relative to 1 volt/dyne/cm²) 2	Sensitivity of the sending system in the free speech field for the associated microphone No. 1292 (1+2) 3	Sensitivity of microphone No. 1292 measured on the closed coupler (dB relative to 1 volt/dyne/cm²) 4	Sensitivity of the sending system with microphone No. 1292 measured on the closed coupler (1+4) 5
80	+73.0	−86.8	−13.8	−89.9	−16.9
100	+72.7	−85.2	−12.5	−87.7	−15.0
120	+72.2	−83.9	−11.7	−86.2	−14.0
200	+70.6	−81.6	−11.0	−83.3	−12.7
300	+69.3	−81.1	−11.8	−82.6	−13.3
400	+69.4	−81.5	−12.1	−82.6	−13.2
500	+70.2	−81.1	−10.9	−82.6	−12.4
600	+71.3	−81.0	−9.7	−82.6	−11.3
700	+72.4	−81.7	−9.3	−82.7	−10.3
800	+73.5	−82.6	−9.1	−82.8	−9.3
900	+74.3	−83.0	−8.7	−83.0	−8.7
1000	+75.2	−83.2	−8.0	−83.2	−8.0
1500	+77.7	−85.6	−7.9	−84.6	−6.9
2000	+79.0	−86.7	−7.7	−85.8	−6.8
2500	+79.7	−87.8	−8.1	−86.2	−6.5
3000	+80.0	−86.6	−6.6	−85.9	−5.9
3500	+80.0	−85.3	−5.3	−85·3	−5.3
4000	+78.9	−85.0	−6.1	−85.0	−6.1
4500	+77.0	−84.9	−7.9	−84.6	−7.6
5000	+74.3	−84.7	−10.4	−84.1	−9.8
5500	+71.2	−86.0	−14.8	−83.0	−11.8
6000	+67.3	−84.8	−17.5	−79.2	−11.9
6500	+64.8	−83.2	−18.4	−76.6	−11.8
7000	+62.7	−84.7	−22.0		

Table 7 gives, for information, the sensitivity of the sending system determined from measurements made in the anechoic chamber, and with the Swiss Administration's artificial mouth, the microphone being placed at 14 cm from the mouth with its protective grille placed horizontally. The acoustic pressure was measured before the microphone was put into position.

The artificial mouth is described in Annex 10, Part II of Volume V of the *Red Book.*

These measurements were made in the anechoic chamber of the Swiss Administration in Bern (July 1958).

Figure 5/P.42 gives the sensitivity/frequency characteristics of the N.O.S.F.E.R. sending end, calculated from the sensitivity values of the microphone as measured under various calibration conditions.

2. *Sensitivity of the N.O.S.F.E.R. receiving system*

The two receiving amplifier gain controls are set to the positions "+14 dB" and "+1 dB". The nominal receiving amplifier gain of the N.O.S.F.E.R. receiving system is adjusted to the fixed value of 37 dB.

Table 8 (column 5) gives the characteristic values of the sensitivity of the N.O.S.F.E.R. receiving system. The sensitivity values of the receiver taken into account in the calculation are extracted from Research Report No. 13200 (April 1950) of the United Kingdom Post Office.

These values correspond to the average sensitivity, less 1 dB, of a number of receivers. The average nominal sensitivity of a receiver, at frequencies of 100, 300, 1000 and 2000 Hz, is fixed at +43.7 dB in relation to 1 dyne/cm² per volt.

In practice the four receivers used have sensitivity/frequency characteristics which differ from the average characteristic defined above. Generally the sensitivity of a receiver is above the average value; moreover, a correction of 1 dB has been introduced above, so that the variations in the individual receivers in relation to the average value can be compensated by means of attenuators.

TABLE 7

Hz	Gain of the electric part of the sending system (dB) 1	Sensitivity of microphone No. 1292 measured in a free acoustic field (dB relative to 1 volt/dyne/cm²) 2	Sensitivity of the sending system in free acoustic field (1+2) (dB relative to 1 volt/dyne/cm²) 3
100	+72.7	−85.6	−12.9
200	+70.6	−82.9	−12.3
300	+69.3	−82.4	−13.1
400	+69.4	−82.9	−13.5
500	+70.2	−83.6	−13.4
600	+71.3	−83.7	−12.4
700	+72.4	−83.6	−11.2
800	+73.5	−83.6	−10.1
900	+74.3	−84.4	−10.1
1000	+75.2	−84.8	−9.6
1100	+75.9	−85.2	−9.3
1200	+76.5	−85.7	−9.2
1300	+77.0	−85.7	−8.7
1400	+77.3	−86.2	−8.9
1500	+77.7	−86.3	−8.6
1800	+78.7	−87.3	−8.6
2000	+79.0	−87.3	−8.3
2200	+79.2	−87.6	−8.4
2500	+79.7	−87.0	−7.3
2700	+79.8	−87.4	−7.6
3000	+80.0	−86.4	−6.4
3300	+80.0	−86.6	−6.6
3500	+80.0	−89.6	−9.6
4000	+78.9	−84.9	−6.0
4500	+77.0	−84.8	−7.8
5000	+74.3	−87.1	−12.8
5500	+71.2	−87.2	−16.0
6000	+67.3	−84.0	−16.7
6500	+64.8	—	—
7000	+62.7	−82.7	−20.0
8000	+62.4	−87.0	−24.6
10000	+56.9	−92.6	−35.7

When the characteristic of a receiver lies within the limits fixed, a special attenuator, variable by steps of 0.25 dB, is adapted to the receiver, so that the average value of its efficiency at frequencies of 100, 300, 1000 and 2000 Hz is equal to +43.7 dB ±0.4 dB in relation to 1 dyne/cm² per volt.

Table 9 gives the characteristic values defining, for each of the four listening channels, the sensitivity of the N.O.S.F.E.R. receiving system, with the particular set of four receivers used.

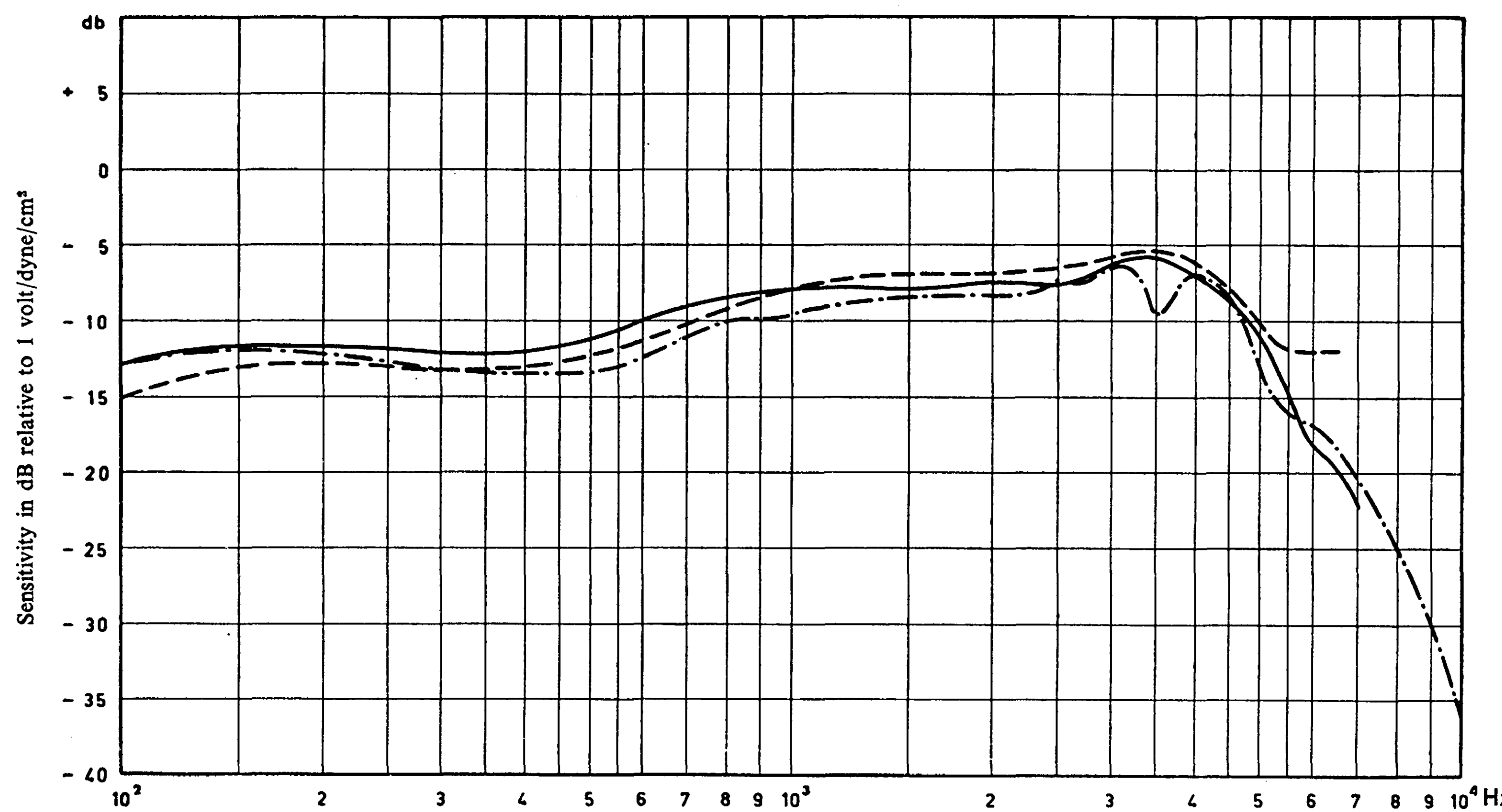
Figure 6/P.42 gives the sensitivity/frequency characteristics of the N.O.S.F.E.R. receiver system.

3. *Level diagram of N.O.S.F.E.R.*

Figure 7/P.42 gives the theoretical level diagram of N.O.S.F.E.R.

C. Normal speech power for voice-ear measurements

The volume measuring set of A.R.A.E.N. is connected at the output of the sending system of N.O.S.F.E.R. The sensitivity controls of the volume measuring set must be adjusted to −10 dB and the operator speaks at the microphone of the sending system of N.O.S.F.E.R. using a speech power such that the needle of the indicating instrument reaches the mark. This speech power is the "normal speech power for voice-ear measurements". The volume (of speech sounds) corresponding to this normal speech power is the "normal volume for voice-ear measurements".



——————— Values calculated from the mean sensitivities of a certain number of microphones, measured in a free acoustic field.

— — — — Values calculated from the sensitivity of microphone No. 1292, measured on a closed coupler.

—·—·— Values calculated from the sensitivity of microphone No. 1292, measured in the anechoic chamber (Bern).

FIGURE 5/P.42 — **Response curve of the N.O.S.F.E.R. sending end**

TABLE 8

Hz	Gain of the electric part of the sending system (terminated with 88 ohms) 1	Correction of 12 dB at the output of the impedance adapter of the receivers (four receivers in series) 2	Average sensitivity −1 dB, of a receiver (dB relative to 1 dyne/cm²/volt) 3	Nominal sensitivity of the receiving system (dB relative to 1 dyne/cm²/volt) 4
80	−12.5	−12.0	+45.4	+20.9
100	−12.2		+46.0	+21.8
120	−12.0		+46.3	+22.3
200	−10.8		+46.6	+23.8
300	−8.8		+46.1	+25.3
400	−6.9		+45.3	+26.4
700	−4.0		+43.1	+27.1
1000	−2.8		+41.2	+26.4
1500	−1.2		+40.0	+26.8
2000	−1.1		+41.4	+28.3
2500	−3.0		+43.3	+28.3
3000	−6.7		+45.9	+27.2
3500	−11.2		+47.8	+24.6
4000	−10.7		+47.9	+25.2
4500	−7.0		+47.0	+28.0
5000	−3.7		+45.5	+29.8
5500	−2.0		+46.3	+32.3
6000	−0.3		+48.2	+35.9
6500	−1.9		+52.0	+38.1
7000	−3.8	−12.0	+55.2	+39.4
Average sensitivities at frequencies of 100, 300, 1000 and 2000 Hz			+43.7	+25.4

Note 1. — In the same conditions, a vu measuring set (see Supplement No. 11, *White Book*, Volume V), connected at the output of the sending system of N.O.S.F.E.R. would give a reading of −9.4 vu.

Note 2. — The normal volume for voice-ear measurements was formerly defined by means of the Volume Indicator (see Annex 18, Part II of Volume V, *Red Book*), which, connected at the output of the sending system of S.F.E.R.T., should give a reading of −15 dB.

Note 3. — The relationships between the readings of the A.R.A.E.N. volume measuring set, the Volume Indicator and a vu measuring set, resulting from Notes 1 and 2, are valid only for the determination of reference equivalent relationships between the indications of the various types of volume measuring sets, during a telephone conversation, are given in Supplement No. 14, *White Book*, Volume V.

D. Primary systems for the determination of reference equivalents

"Primary system for the determination of reference equivalents" is the name given to:

a) a system consisting of a replica of N.O.S.F.E.R.,

b) a system conforming to the description given in section 3.1.1.II of the C.C.I.F. *Green Book*, Volume IV, pages 27 to 34.

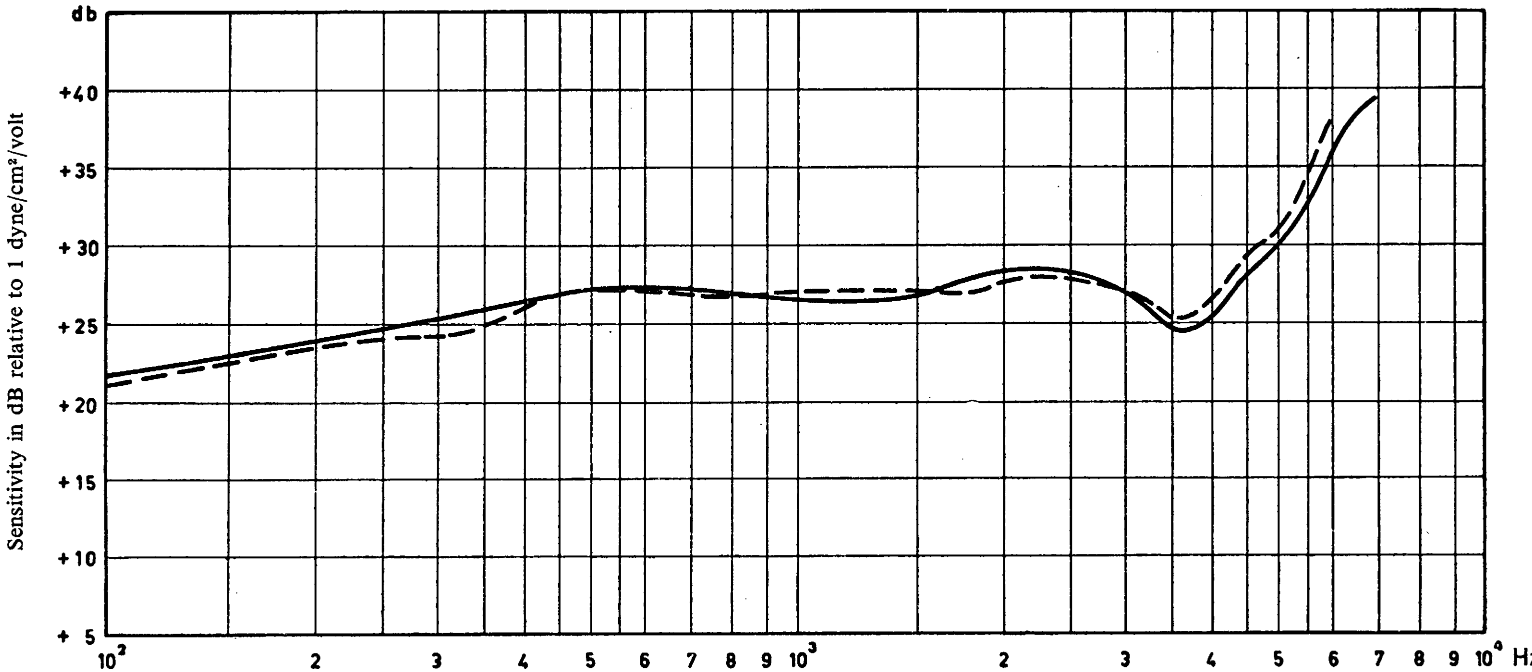
It is assumed:

1) that such a system is defined by a detailed description including the relevant method of objective calibration of the physical parameters of the system;

2) that such a system has been compared directly or indirectly with N.O.S.F.E.R.

TABLE 9

Hz	Gain of the electric part of the receiving system (terminated by the 4 receivers in series)	Gain of the electric part of the receiving system relative to each listening channel (4 receivers in series) Receivers Nos.				Sensitivity of the receivers (dB relative to 1 dyne/cm²/volt) Receivers Nos.				Sensitivity of the receiving system with each of the 4 receivers (dB relative to 1 dyne/cm²/volt) Receivers Nos.				Mean sensitivity of the receiving system with the 4 receivers (dB relative to 1 dyne/cm²/volt)
		936	946	1039	1140	936	946	1039	1140	936	946	1039	1140	
100	−12.3	−24.3	−24.4	−25.0	−24.8	**+45.0**	**+45.5**	**+45.5**	**+47.5**	**+20.7**	**+21.1**	**+20.5**	**+22.7**	**+21.2**
200	−11.0	−22.7	−22.8	−23.2	−23.1	+46.1	+46.9	+46.6	+46.4	+23.4	+24.1	+23.4	+23.3	+23.5
300	−9.3	−20.9	−21.0	−21.7	−21.6	**+45.5**	**+46.0**	**+45.1**	**+45.6**	**+24.6**	**+25.0**	**+23.4**	**+24.0**	**+24.2**
400	−7.3	−18.9	−19.0	−19.8	−19.7	+45.2	+45.4	+45.2	+45.1	+26.3	+26.4	+25.4	+25.4	+25.9
500	−5.0	−16.6	−16.7	−17.3	−17.2	+44.5	+44.5	+44.5	+43.5	+27.9	+27.8	+27.2	+26.3	+27.3
600	−4.8	−16.4	−16.6	−17.1	−17.0	+43.9	+44.0	+43.8	+43.5	+27.5	+27.4	+26.7	+26.5	+27.0
700	−4.4	−16.0	−16.2	−16.8	−16.7	+43.5	+43.5	+43.0	+43.0	+27.5	+27.3	+26.2	+26.3	+26.8
800	−3.8	−15.4	−15.6	−16.2	−16.1	+42.7	+42.7	+42.4	+42.0	+27.3	+27.1	+26.2	+25.9	+26.6
900	−3.2	−14.8	−15.0	−15.7	−15.5	+42.4	+42.2	+42.0	+41.5	+27.6	+27.2	+26.3	+26.0	+26.8
1000	−2.7	−14.3	−14.4	−15.0	−14.9	**+42.0**	**+41.8**	**+41.5**	**+41.0**	**+27.7**	**+27.4**	**+26.5**	**+26.1**	**+26.9**
1100	−2.3	−13.9	−14.0	−14.7	−14.6	+41.5	+41.5	+41.0	+40.7	+27.6	+27.5	+26.3	+26.1	+26.9
1200	−1.8	−13.5	−13.6	−14.3	−14.2	+41.0	+41.0	+40.7	+40.5	+27.5	+27.4	+26.4	+26.3	+26.9
1300	−1.6	−13.2	−13.3	−14.0	−13.9	+41.0	+41.0	+40.6	+40.1	+27.8	+27.7	+26.6	+26.2	+27.1
1500	−1.1	−12.8	−12.9	−13.6	−13.4	+40.7	+40.8	+40.5	+39.4	+27.9	+27.9	+26.9	+26.0	+27.2
1800	−1.0	−12.7	−12.8	−13.4	−13.3	+40.7	+40.5	+39.7	+39.0	+28.0	+27.7	+26.3	+25.7	+26.9
2000	−1.2	−12.7	−12.9	−13.6	−13.5	**+41.4**	**+41.2**	**+40.5**	**+39.9**	**+28.7**	**+28.3**	**+26.9**	**+26.4**	**+27.6**
2100														
2400														
2500	− 2.8	−14.8	−14.5	−15.1	−15.0	+42.9	+43.1	+42.2	+41.8	+28.1	+28.6	+27.1	+26.8	+27.6
2700	− 4.0	−15.6	−15.8	−16.4	−16.3	+44.0	+44.5	+43.5	+43.0	+28.4	+28.7	+27.1	+26.7	+27.7
3000	− 6.5	−18.0	−18.2	−18.9	−18.9	+45.8	+46.2	+45.2	+44.5	+27.8	+28.0	+26.3	+25.6	+26.9
3300	− 9.2	−20.7	−20.9	−21.6	−21.6	+47.5	+48.0	+47.0	+46.5	+26.8	+27.1	+25.4	+24.9	+26.0
3600	−11.0	−22.6	−22.8	−23.5	−23.4	+48.5	+49.0	+48.0	+48.0	+25.9	+26.2	+24.5	+24.6	+25.3
4000	−10.0	−21.7	−21.9	−22.6	−22.3	+48.8	+48.7	+48.2	+48.7	+27.1	+28.8	+25.6	+26.4	+26.5
4500	−6.8	−18.4	−18.6	−19.2	−18.9	+48.0	+48.1	+47.5	+48.7	+29.6	+29.5	+28.3	+29.8	+29.3
5000	−3.8	−15.4	−15.6	−16.2	−16.0	+46.2	+46.3	+45.8	+47.8	+30.8	+30.7	+29.6	+31.8	+30.7
6000	−0.2	−11.8	−12.0	−12.6	−12.4	+49.9	+49.9	+48.8	+52.4	+38.1	+37.9	+36.2	+40.0	+38.0
7000	−3.4	−15.0	−15.1	−15.8	−15.7									
	Mean sensitivities at frequencies of 100, 300 1000 and 2000 Hz . . . . Supplementary attenuator					+43.5 b=1.5dB	+43.6 b=1.5dB	+43.2 b=2.0dB	+43.5 b=2.0dB	+25.4	+25.4	+24.3	+24.8	+25.0



——————— Values calculated from the mean sensitivity of a certain number of receivers.

— — — — — Values relative to the mean of the four receivers used in the C.C.I.T.T. Laboratory.

FIGURE 6/P.42 — **Frequency response curve of the N.O.S.F.E.R. receiving system (values calculated from the calibration of the receivers on the A.R.A.E.N. artificial ear)**

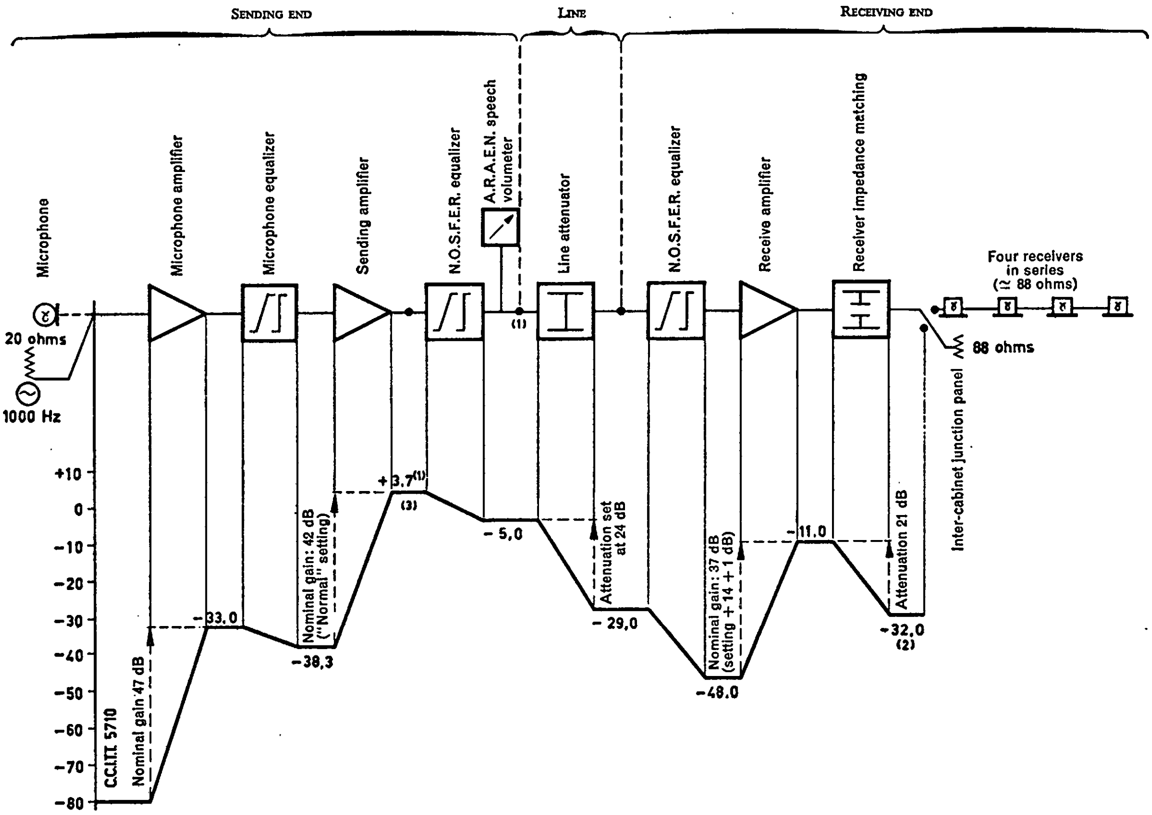


FIGURE 7/P.42 — **Level diagram for the N.O.S.F.E.R. for a 1000-Hz tone at a level of −80 dB relative to 1 volt supplied by a generator of 20-ohm internal impedance and applied to the jack of the microphone, the following settings being used:**

send amplifier "normal"
receive amplifier: "14 dB + 1 dB"
line attenuator: 24 dB

[1] The volume measured at this point with the A.R.A.E.N. volume meter indicator is −10 dB (relative to 1 volt) when the operator speaks with the normal speech power for measurements.

[2] With a tolerance of ±0.3 dB (value determined from maintenance measurements taken over a period of six months).

[3] This value, and the levels measured at the following different points of the transmission chain, depend on the microphone used. (See B.1 above and Supplement No. 9, *White Book*, Volume V, gain adjustment of the sending amplifier.)

The indirect verification of a primary system can be carried out by determining the reference equivalents of some stable sending or receiving systems against the given primary system and against the N.O.S.F.E.R.

E. WORKING STANDARD SYSTEMS

It is admitted for the purposes of the application of C.C.I.T.T. recommendations that the reference equivalent of a commercial system may be determined by taking the sum of the relative equivalent of this commercial system obtained by comparison with a working standard and the reference equivalent of the working standard system (see Recommendation P.72).

By way of information the descriptions of the working standard systems are reproduced in Annexes 1 and 2 to this Recommendation.

Before being officially put in service, any working standard that has not already been compared with the S.F.E.R.T. should be compared with the N.O.S.F.E.R. or to a primary system for determining reference equivalents.

This comparison is intended to define the transmission qualities of a component of the working standard as compared with the corresponding component of N.O.S.F.E.R. or a primary system for the determination of reference equivalents. It indicates in decibels the amount by which the respective sending or receiving system of the working standard is worse or better relative to the sending or receiving system of N.O.S.F.E.R. (or a primary system for the determination of reference equivalents).

The measuring method used in the C.C.I.T.T. Laboratory is the so-called "two-operator, hidden-loss method" (see Recommendation P.72).

The tests are carried out by telephonometric comparison (voice and ear tests), substituting the component to be compared (sending or receiving system) for the corresponding component of N.O.S.F.E.R. An artificial line of adjustable loss, in series with the more efficient system, enables the efficiencies of the two systems to be made equal.

The circuit diagrams showing the general method of calibrating the sending and receiving systems of the working standard with the S.F.E.R.T., are shown in Figures 8/P.42 and 9/P.42 respectively.

The method of comparison employed in the C.C.I.T.T. Laboratory is based on tests (elementary balances, see later) by only two operators (one operator speaking and one listening) and the use of three distortionless attenuators with characteristic impedances of 600 ohms at zero angle.

The first attenuator *A* is set to a value of 24 dB at the C.C.I.T.T. Laboratory (in any case the value of this attenuator should be greater than 15 dB) in order:

1) to adjust the current in the receiving systems to a value such that the best conditions for comparative listening tests are obtained, and

2) to prevent electrical interaction effects between the sending and receiving systems.

The second attenuator *S* introduces the hidden-loss; its value is not known to the listening operator and may vary from 0 to 10 dB by 1-dB steps.

The third attenuator *E*, called a "balancing attenuator", is adjusted by the listening operator and is to enable equality of loudness to be obtained.

A combination of three keys (see Figures 8/P.42 and 9/P.42), which can be operated simultaneously, provides the switching necessary for telephonometric comparisons.

A volume indicator (the A.R.A.E.N.) enables the speaking operator to maintain the normal volume for telephonometric tests as defined above under C. The reference equivalents of the transmitting and receiving systems of the working standard considered are obtained from the average of a certain number of telephonometric tests called "individual balances".

To make an individual balance, the following procedure is adopted:

a) *Tests on a sending system* (Figure 8/P.42)

Each individual balance is carried out between two operators. The talker repeats a predetermined sentence[1] in front of each microphone alternately; the hidden loss is set at a particular value.

[1] In the C.C.I.T.T. Laboratory the conventional sentence is as follows: Paris, Bordeaux, Le Mans, Saint-Leu, Léon, Loudun.

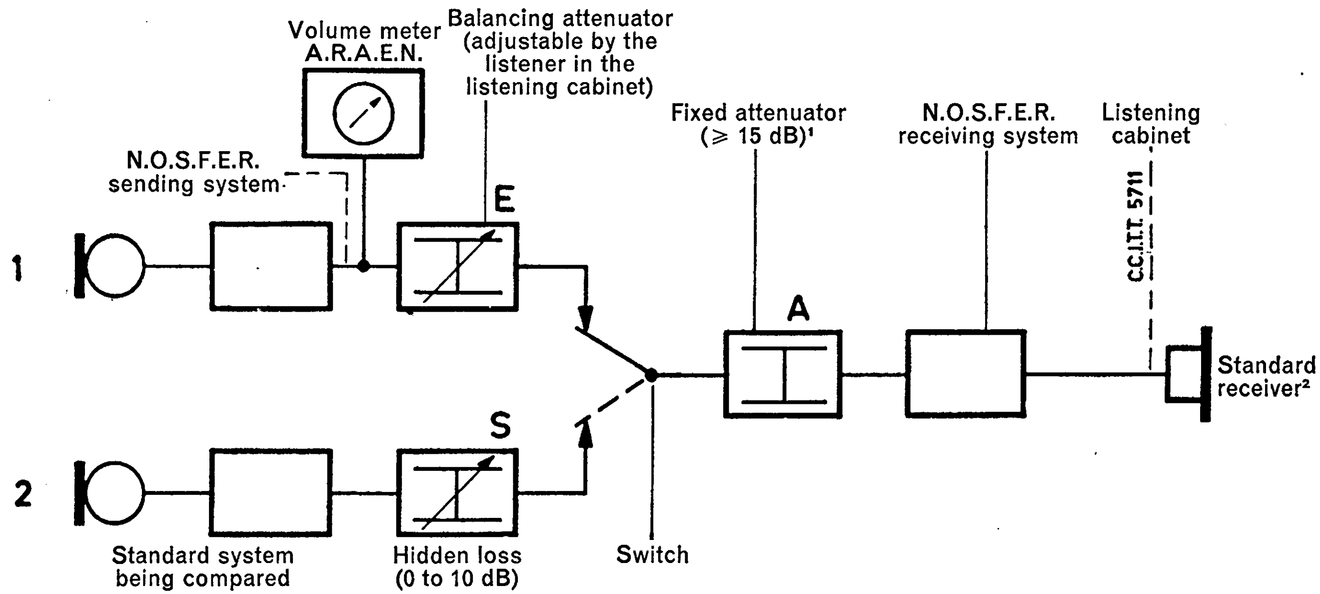


FIGURE 8/P.42 — **Comparison of a standard sending system with a N.O.S.F.E.R. sending system (method termed "two-operator-hidden-loss method")**

[1] In the C.C.I.T.T. Laboratory, this value is fixed at 24 dB.

[2] In N.O.S.F.E.R. there are four receivers in series. During an elementary balance, the three other receivers available are laid face downwards, although they remain connected.

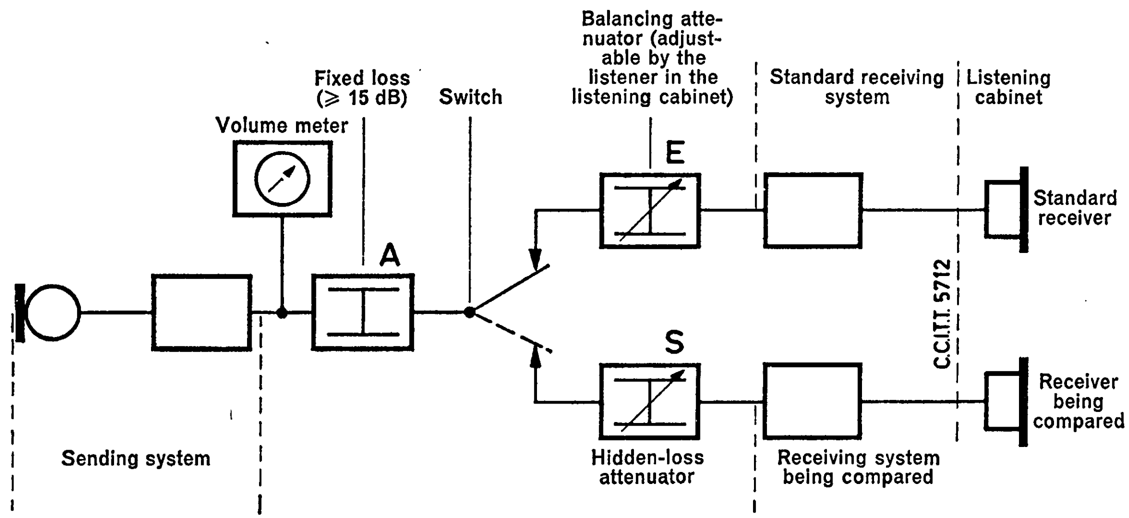


FIGURE 9/P.42 — **Comparison of a standard receiving system with a N.O.S.F.E.R. receiving system (method termed "two-operator hidden-loss method")**

1, 2 } See footnotes to Figure 8/P.42.

In Figure 8/P.42 it will be noted that the total attenuation inserted between the sending system to be measured and the N.O.S.F.E.R. receiving system varies between 24 dB and 34 dB (according to the attenuation of the hidden loss). A variant of this method is now used in the C.C.I.T.T. Laboratory so as to leave a greater margin of variation in the balancing attenuator, which appeared necessary with apparatus having a reference equivalent in the neighbourhood of that of the standard system: $S + A$ of the attenuations of the lines S and A varies between 24 dB and 34 dB; thus, the hidden loss can vary from 0 to 34 dB.

The operator (P) endeavours to speak in a normal tone at a normal conversational speed and to preserve the normal volume for telephonometric tests. At the same time he operates the keys in such a manner that the appropriate connections are made according to the microphone employed. The listening operator (E) adjusts the balancing attenuator, of which he has control, to obtain equality of sound intensity for the two positions of the keys.

b) *Tests on a receiving system* (Figure 9/P.42)

Each individual balance is made by two operators. The speaking operator (P) repeats, in a normal tone and at a normal conversational rate and maintaining the normal volume for telephonometric tests, the conventional sentence into the microphone of the N.O.S.F.E.R. sending system. He operates the keys putting the N.O.S.F.E.R. receiving system and the working standard receiving system successively into circuit with the N.O.S.F.E.R. sending system. The operator (E) listens with the two receivers (N.O.S.F.E.R. receiver and the receiver of the working standard under test) successively. He also adjusts the balancing attenuator so as to obtain equality of sound intensity for each of the two receivers. The Laboratory uses the same technique in this test as under a above, for the adjustment of the attenuators S and A.

c) *Recording of results and statistical analysis of tests*

Each replication of a telephonometric test consists of a certain number of balances. The number of individual balances which makes up a replication is at least six; it is normally twelve at the C.C.I.T.T. Laboratory with a normal crew of six operators who work in sets of three at a time; the number of balances can be increased whenever considered necessary.

In each replication, the results are entered in appropriate forms, on which the hidden loss values and balance attenuations are shown respectively for each elementary balance. The value of the reference equivalent for a replication is the arithmetical mean of the values obtained for all the elementary balances of the replication concerned. When a single replication does not suffice to determine the reference equivalent, two replications are carried out in periods with a spacing of one week between the two. The test results are then submitted to statistical analysis. The test results and the statistical analysis are sent to Administrations in the form of a technical report by the C.C.I.T.T. Laboratory which also gives the confidence limits as defined in Annex 3 below.

Note. — By way of information, Annex 7 (Part II of Volume V of the *Red Book*) describes another method for the analysis of loudness efficacy balances.

d) *Measurement of microphone resistance*

When the sending system to be tested includes a carbon microphone (S.E.T.A.B. or S.E.T.A.C. system) the measurement of the microphone resistance is made during the speech test by the voltmeter-ammeter method. The voltmeter and ammeter used are of a damped type.

Several observations are made while somebody speaks into the microphone to be measured, and the mean resistance is that obtained during these observations.

e) *Periodical calibration of working standard systems*

Working standard systems must be periodically compared against the international telephonometric standard consisting of N.O.S.F.E.R. or a primary system for the determination of reference equivalents. Recommendations for forwarding such apparatus are contained in Recommendation P.43.

ANNEX 1

(to Recommendation P.42)

Rules concerning the composition of working standards with subscriber's equipment (S.E.T.A.B.)

Working standards with subscriber's equipment consist of a sending system, an attenuator and a receiving system. The sending and receiving systems consist respectively of subscribers' sets of a commercial type associated with a subscriber's line and a feeding bridge. The feeding current should be low enough to avoid any risk of damage to or instability of the microphone.

The attenuator connected between the sending and receiving systems should have a minimum loss of 15 dB and an impedance of 600 ohms.

The system should be complete with a volume meter to enable the vocal power used during telephonometric tests to be maintained.

It is, of course, essential for the microphones and receivers to satisfy certain conditions to enable them to be considered as standards. Administrations which have not already done so may therefore send to the C.C.I.T.T. Laboratory six handsets which appear to have been stable during preliminary tests extending over a period of six months.

The C.C.I.T.T. Laboratory will first carry out measurements of sensitivity/frequency characteristics to assess the quality of the apparatus; then it will conduct at intervals of two months five measurements of sending and receiving reference equivalent in order to check the stability of the apparatus.

After these preliminary measurements the C.C.I.T.T. Laboratory will choose, from the six items of the same type which have been sent there, three items which will serve as sending standards and three items which will serve as receiving standards. It will proceed to the calibration of the standard apparatus thus selected under the following conditions:

Determination of the sending and receiving reference equivalents. For each measurement at least 12 individual balances will be made in order to obtain reliable values of reference equivalents.

ANNEX 2

(to Recommendation P.42)

Description of a working standard having an electro-dynamic microphone and receiver (S.E.T.E.D.)

The S.E.T.E.D. working standard was originally designed for use as a reference system for loudness rating and for articulation rating (A.E.N.). A detailed description of this working standard is given in Annex 2, Recommendation P.42 (*White Book*, Volume V). Basically it consists of a calibrated speech path, having a frequency characteristic similar to that which would be given by a 1-metre air path including the obstacle effect of the human head, but band-limited to 300 to 3400 Hz. It uses a moving coil microphone of a special type designed for close talking which is substantially protected against the effects of breath moisture. To standardize the lip position a guard ring is fitted to the microphone at a distance of 25 mm.

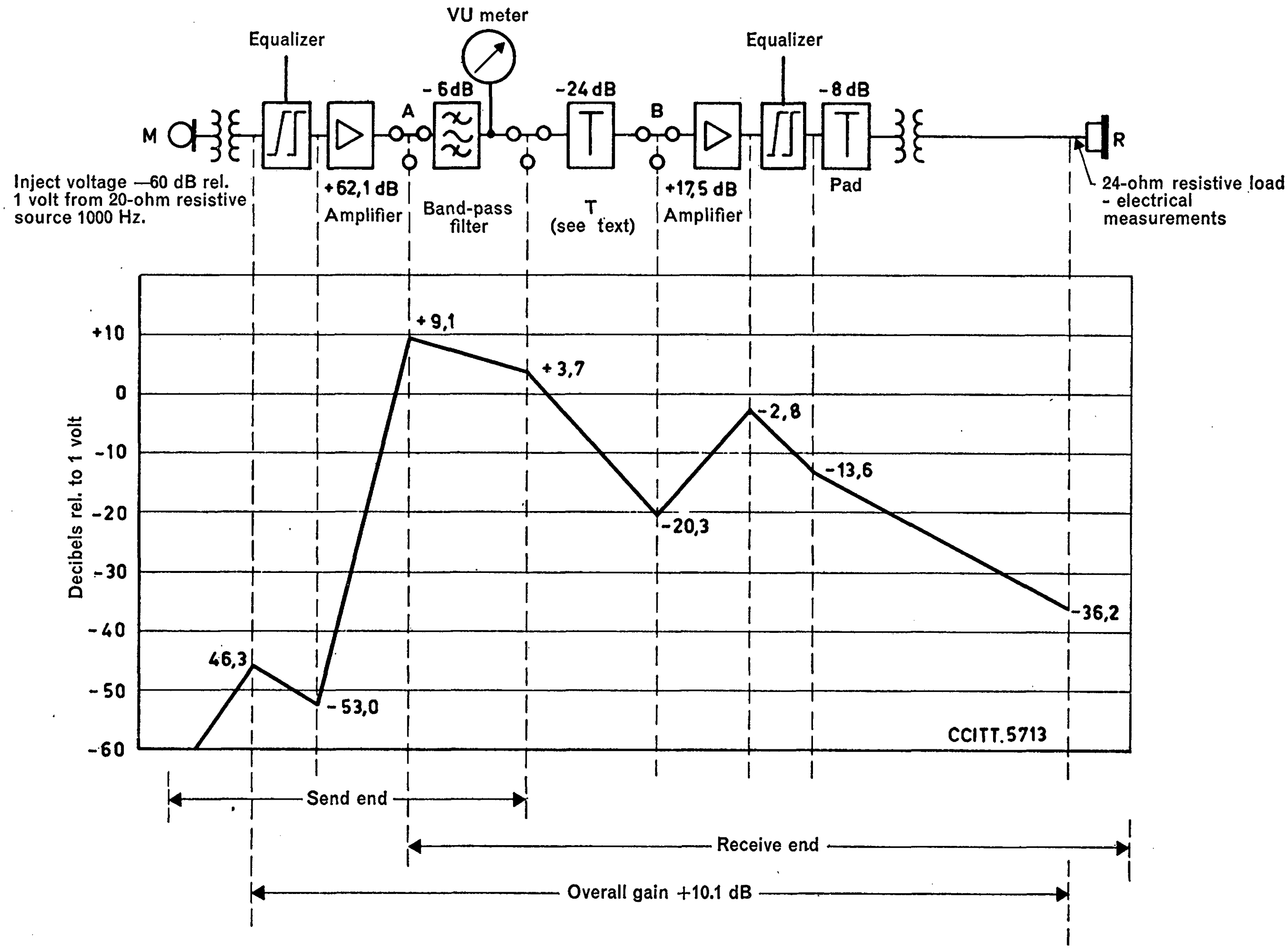
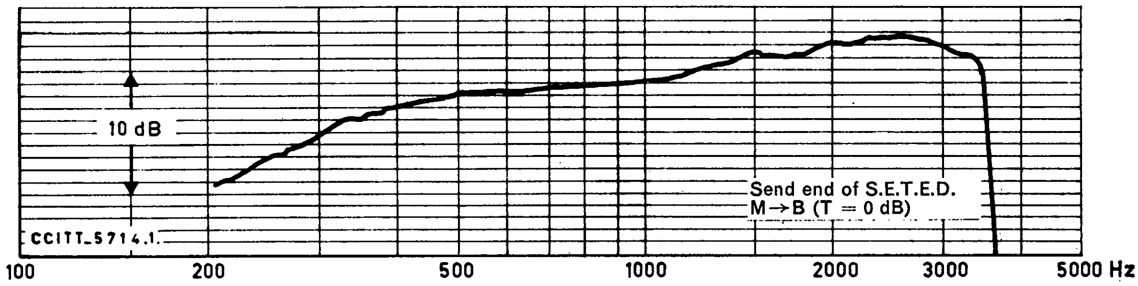
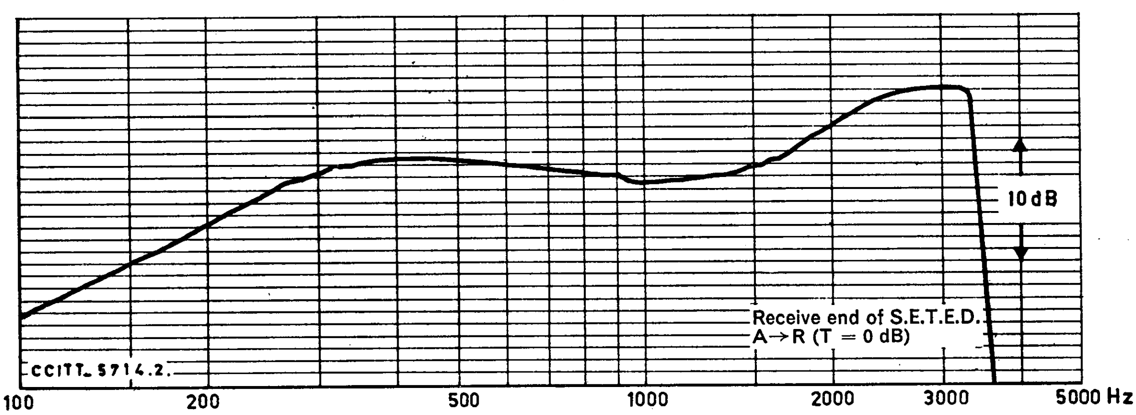


FIGURE 10/P.42. — **S.E.T.E.D. distribution of gains and levels**



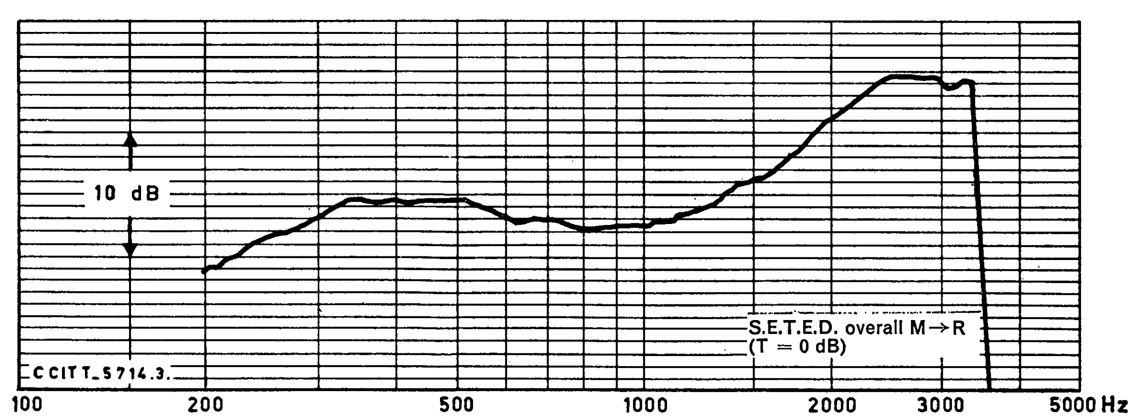
Sensitivity at 1000 Hz = −6.0 dB rel. 1 Volt/N/m² referred to 1 Newton/m² free field sound pressure 25 mm on axis in front of artificial mouth (Bruel & Kjaer 4219). At 1000 Hz reference sound pressure level caused reading of −2.8 dB on S.E.T.E.D. vu meter.

FIGURE 11/P.42



Sensitivity at 1000 Hz = +7.7 dB rel. 1N/m²/volt with receiver loaded in I.E.C. artificial ear (no leak). At 1000 Hz, 1 volt input at A caused a reading of −3.6 dB on S.E.T.E.D. vu meter.

FIGURE 12/P.42



Mouth to ear gain at 1000 Hz = 8.5 dB referred to free field sound pressure 25 mm on axis in front of artificial mouth, and with receiver loaded in I.E.C. artificial ear (no leak).

FIGURE 13/P.42

S.E.T.E.D. is provided with means for absolute calibration of its microphones and receivers and for this purpose makes use of a calibrated quartz crystal microphone. In recent years it has become possible to confirm calibrations with modern capacitor microphones.

By means of the input/output facilities provided, appropriate circuits can be set up for determining the relative equivalent or the A.E.N. value of a commercial telephone system for sending, receiving, sidetone, or for a complete connection. Arrangements at the input to the receive amplifier allow noise to be injected or sidetone to be provided.

A vu meter connected to the output of the sending part of S.E.T.E.D. allows the talking level to be monitored and controlled.

The distribution of gains and losses throughout the S.E.T.E.D. working standard are given in Figure 10/P.42, and the model at the C.C.I.T.T. Laboratory is within 0.1 dB of these design figures. Sensitivity frequency characteristics have been determined for the send and receive ends and the overall connection of S.E.T.E.D. and are shown in Figures 11/P.42, 12/P.42 and 13/P.42 respectively. For these measurements the sensitivities quoted under each curve refer to the condition with the band-pass filter and its in-band loss of 6 dB included, and the trunk attenuator (T) set to zero.

Reference equivalents determined for the S.E.T.E.D. by direct comparison against the N.O.S.F.E.R. in the C.C.I.T.T. Laboratory in 1973 were as follows referring to Figure 1:

Sending reference equivalent	—	M	→	B	=	+7.8 dB, quieter than N.O.S.F.E.R.
Receiving reference equivalent	—	A	→	R	=	+4.5 dB, quieter than N.O.S.F.E.R.
Overall reference equivalent	—	M	→	R	=	+0.8 dB, quieter than N.O.S.F.E.R.

The determinations were carried out with loudness balancing at constant listening level employing a trunk attenuation (T) of 24 dB in S.E.T.E.D. and varying the loss in the appropriate N.O.S.F.E.R. path to obtain a loudness balance.

Reference equivalents given above for the S.E.T.E.D. are for the condition excluding the 24 dB trunk attenuation T but including the band-pass filter with its 6 dB in-band loss. Figures given in previous editions of C.C.I.T.T. Volume V have omitted the 6 dB loss of the attenuator path associated with the band-pass filter. However, since the S.E.T.E.D. is always used with its band-pass filter complete with pad it seems right to include it here. In making comparisons with previous test results, due consideration should, of course, be given to the probable variations due to test-team changes.

ANNEX 3

(to Recommendation P.42)

Confidence limits

Supposing that a suitable form of statistical analysis has furnished an estimate s_0^2 of the true error variance and an estimate s_D^2 of the true listener (or s_C^2 talker) variance, each with its own number of degrees of freedom depending on the number of operators (n) in the team and the number of times (r) the test was replicated. An estimate of the true value of the reference equivalent is furnished by the mean value $\bar{x}$ of all the results. The word "true" is to be taken to mean those values to which the estimated values would tend if the tests were repeated indefinitely with an indefinite number of operators.

On the basis of these estimates it is possible to state, with a long-term probability P of being correct, that the true value of reference equivalent X lies somewhere between two limits x_1 and x_2, $x_1 \leqslant X \leqslant x_2$. The numerical values of x_1 and x_2 can be determined, with some degree of approximation in certain cases, from s_1, s_2 (taking account of their numbers of degrees of freedom) and $\bar{x}$: the distribution of the relation being given by Student's t function. The values x_1 and x_2 are known as the confidence limits of the mean and where, as in this instance, they are symmetrically disposed about it are represented by $\pm Lp\%$.

If the confidence limits involve only the error, they are referred to as internal limits and apply in the case of repeated determinations, *under the same test conditions*, with the same crew in the C.C.I.T.T. Laboratory. (In this case only one estimated variance is involved and the errors due to approximations are negligible.)

If the limits are based on the listener (or talker) variance as well as the error variance they apply to repeated determinations, *under the same test conditions*, in the C.C.I.T.T. Laboratory but with an indefinite variety of teams, each of n operators, drawn from the same population as the sample of operators used in the test analyzed.

(In this case both estimated variances are involved but the value of t to be used depends only upon the factor (D) as it has the *smallest* number of degrees of freedom: the degree of approximation is therefore greater.)

Note. — The method to be used for the analysis of normally arranged volume tests is given in Supplement No. 15, *White Book*, Volume V.

Recommendation P.43

INSTRUCTIONS FOR FORWARDING STANDARD SYSTEMS AND COMMERCIAL TELEPHONE APPARATUS TO THE C.C.I.T.T. LABORATORY TO HAVE THE REFERENCE EQUIVALENTS OF THESE SYSTEMS DETERMINED

Administrations are requested to follow the instructions given below when they forward reference systems or commercial telephone systems to the C.C.I.T.T. Laboratory for determinations of reference equivalents.

A. Primary systems for the determination of reference equivalents

If an Administration wishes to have the reference equivalent of its primary system for the determination of reference equivalents determined, assuming that the system concerned can be transported without risk of deterioration, it must supply the C.C.I.T.T. Laboratory with the necessary documentation and, if necessary, instructions for checking the various parts of the system (amplifier, attenuator line, etc.).

If the volume meter associated with the system does not possess the basic characteristics of the volume meter of A.R.A.E.N. (Supplement No. 10, *White Book*, Volume V), the volume meter must be sent to the C.C.I.T.T. Laboratory at the same time as the system itself, and the method for reading it should be indicated.

B. Working standard systems

1. *Working standard systems using microphones other than carbon microphones*

If a working standard system is designed for the use of one or more stable receivers and chiefly of one stable microphone, it is not necessary to calibrate such systems periodically by comparison with N.O.S.F.E.R. (or a primary system for the determination of reference equivalents).

Administrations wishing to have their systems calibrated (or re-calibrated) by the C.C.I.T.T. Laboratory should follow the instructions given under A above.

2. *Working standard systems using carbon microphones*

2.1 *Working standard systems using subscribers' sets (S.E.T.A.B.).* — When a S.E.T.A.B. system is set up, the Administration should first make preliminary checks to see whether the microphones and receivers are stable, whether they are subject to frying, and whether the transmission quality is acceptable. These tests should be spread over a fairly long period (six months).

After these preliminary tests, the Administration should forward six systems composed of the same type of apparatus (each system should bear a suitable distinctive mark) namely:

— six subscribers' handsets equipped with a microphone and receiver (each bearing a number);
— six feeding bridges (with an indication of their characteristics);
— where necessary, six artificial subscribers' lines if the system concerned comprises such lines;
— a guard-ring for the reference equivalents;
— a guard-ring for the A.E.N.s, should the Administration want the C.C.I.T.T. Laboratory to carry out articulation tests using the method specified for A.E.N.;
— the associated volume meter.

The method of reading the volume meter should be indicated. During the measurements, the C.C.I.T.T. Laboratory will thus be able to calibrate the volume meter using speech and determine the adjustment corresponding to the normal speech power for telephonometric measurements.

The Administration will thus have six systems that may be used, as required, e.g.:

— three systems as a sending standard,
— three systems as a receiving standard, when reference equivalents are measured, or
— one system as a sending standard,
— four systems as a receiving standard, in the case of A.E.N. measurements.

In the case of periodic re-calibrations by means of reference equivalent determinations where the object is mainly to verify the stability of the microphones and receivers the Administration need not send all the above-mentioned apparatus. In this case the essential items are—

— three subscribers' sets,
— six microphones and six receivers,
— one subscriber's artificial line,
— one feeding bridge,
— one guard-ring for the reference equivalents.

2.2 *Working standard systems using a Solid Back carbon microphone and a Bell receiver (S.E.T.A.C.)* — The C.C.I.T.T. does not recommend the use of such systems as working standard systems; however, Administrations which still use them and which wish to have their microphones and receivers re-calibrated should send only the microphones and receivers to the C.C.I.T.T. Laboratory, as the latter already has some S.E.T.A.C. systems (see Volume IV of the C.C.I.F. *Yellow Book*, Paris, July 1949).

General comment on sections A and B

The object of the general recommendations given above is to guide Administrations. When an Administration wishes to have a system for the determination of reference equivalents calibrated (or re-calibrated) it should get into touch with the C.C.I.T.T. Laboratory before sending the apparatus, so that the technical and experimental conditions of tests may be fixed in advance.

C. Commercial telephone systems

Determinations of reference equivalents are not, strictly speaking, calibration measurements; their aim is to determine reference equivalents by direct comparison with the new master systems for the determination of reference equivalents (N.O.S.F.E.R.). This being so, it is desirable for the technical conditions to be defined by agreement between the Administration and the C.C.I.T.T. Laboratory.

The cost of determining reference equivalents in the C.C.I.T.T. Laboratory is generally assessed on the basis of the number of hours of work by the Laboratory team. The relevant information is given in Recommendation P.47.

Recommendation P.44

DESCRIPTION AND ADJUSTMENT OF THE REFERENCE SYSTEM FOR THE DETERMINATION OF A.E.N. (S.R.A.E.N.)

The reference system for the determination of A.E.N. (S.R.A.E.N.) is a system consisting of the following elements:

- reference equipment for the determination of A.E.N. (A.R.A.E.N.),
- a band-pass filter cutting off at 300 and 3400 Hz,
- a device allowing electrical background noise (Hoth spectrum) to be injected at the input of the receiving system (point M in Figure 1/P.44) at a psophometric e.m.f. of 2 mV.

The schematic diagram of the S.R.A.E.N. is given in Figure 1/P.44.

a) *A.R.A.E.N.*

The A.R.A.E.N. is described in detail in Recommendation P.41; this also contains a definition of the normal adjustment of the A.R.A.E.N.

b) *300* to *3400-Hz band-pass filter*

The band-pass filter has cut-off frequencies of 300 and 3400 Hz; it simulates the transmission characteristics of a typical carrier system telephone channel. The insertion loss is within the limits ± 0.5 dB in the band 300 to 3400 Hz (see Figure 2/P.44). For frequencies above 3400 Hz the insertion loss increases to reach at least 30 dB at 4000 Hz and remains above this value for all frequencies above 4000 Hz.

c) *Electrical background noise*

At the input of the A.R.A.E.N. receiving system an electrical background noise is injected; this noise has the Hoth spectrum and has a psophometric e.m.f. of 2 mV as measured with the psophometer specified by the C.C.I.T.T. for commercial telephone circuits (see Recommendation P.53 A). Figure 3/P.44 gives the mean power density spectrum observed at subscribers' telephone stations (Hoth spectrum) (curve *a*) together with typical graphs *b* and *c* obtained at the C.C.I.T.T. Laboratory with two sets of half-octave filters.

Note. — Administrations may consider the use of other working standards for the determination of A.E.N. values, these systems being capable of being calibrated by comparison with the S.R.A.E.N.

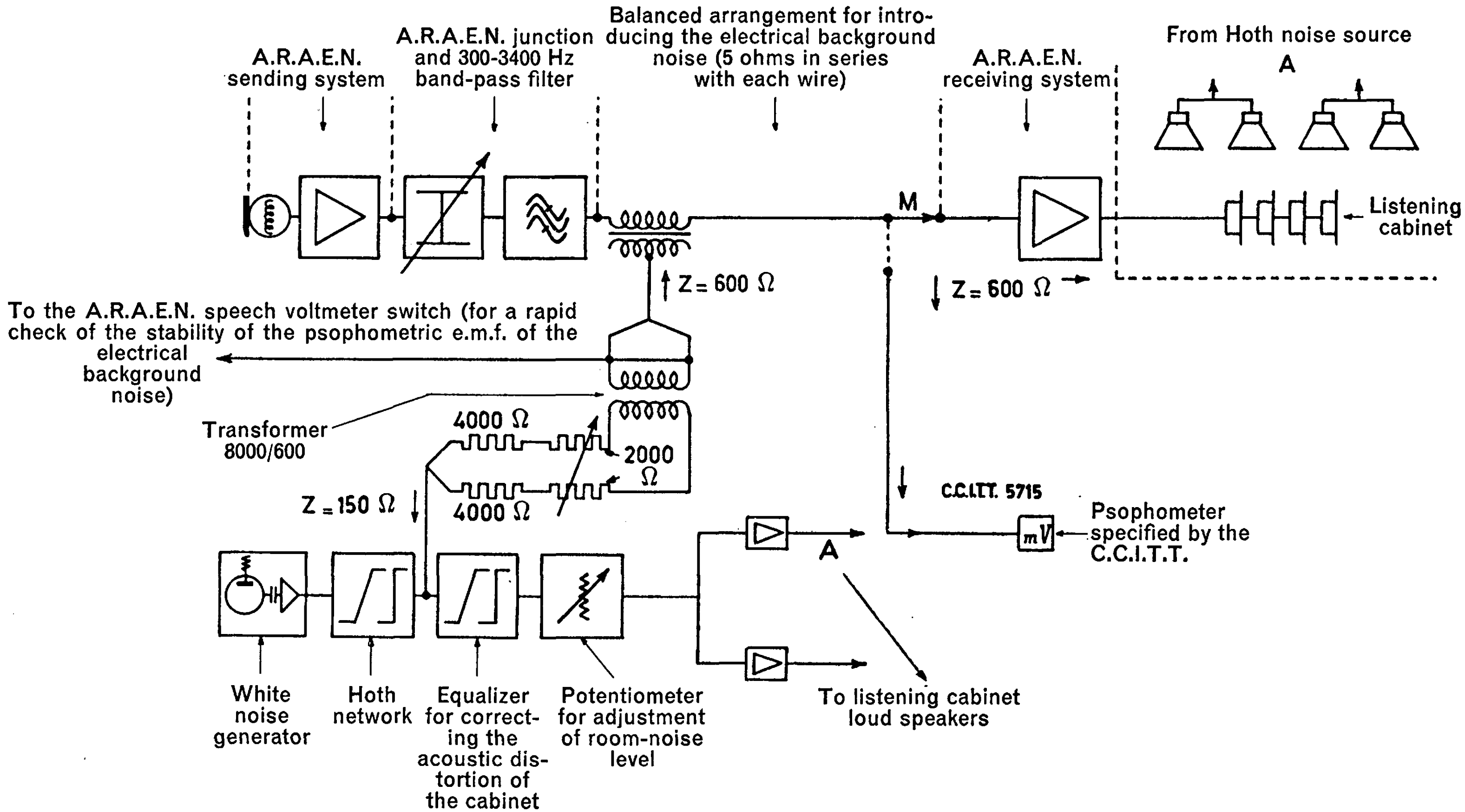


FIGURE 1/P.44. — **Schematic diagram of the S.R.A.E.N. including the arrangement for injecting electrical background noise into the A.R.A.E.N., and the measurement of the psophometric voltage of this noise**

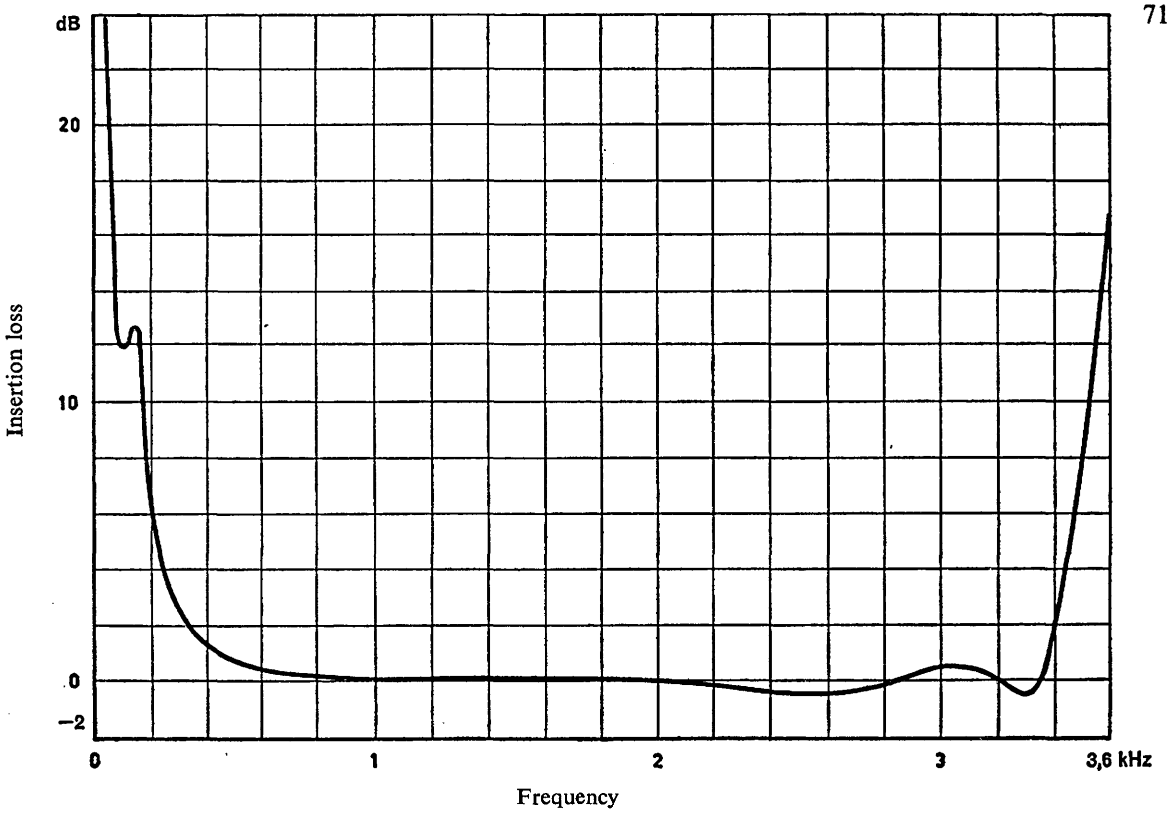
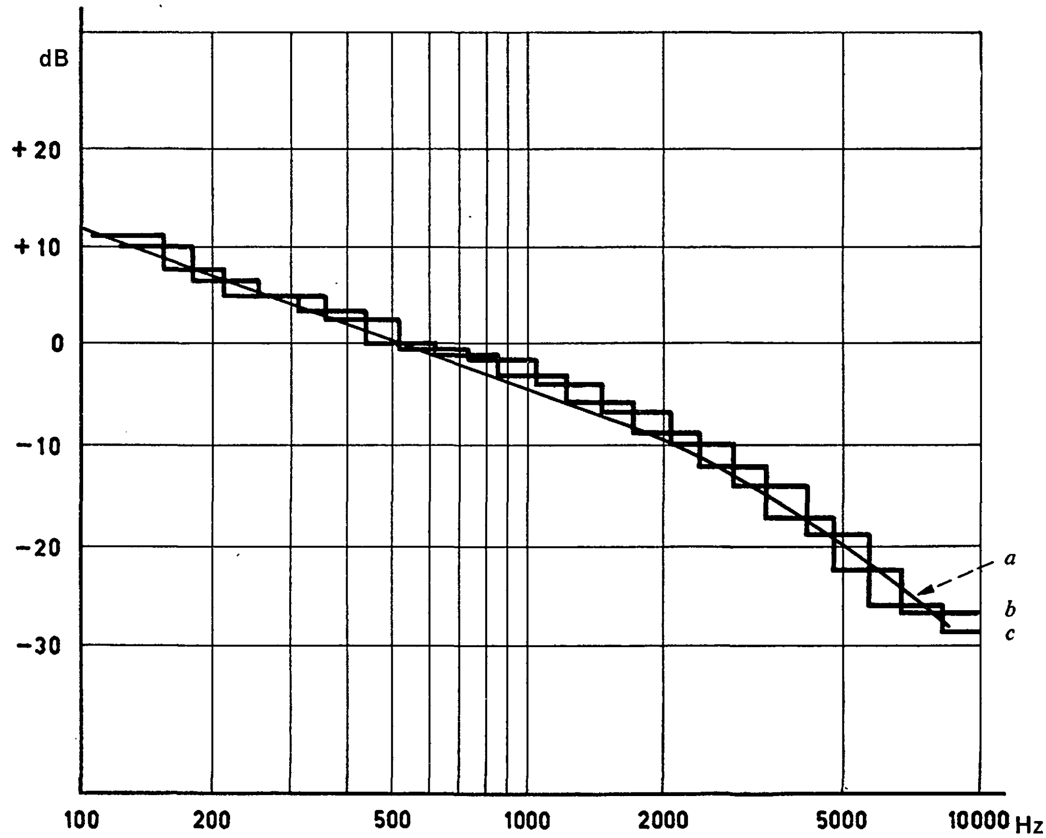


FIGURE 2/P.44. — **Insertion loss (between 600-ohm terminations) of the 300 to 3400-Hz band-pass filter**



a ——— Mean power density spectrum of noises observed at subscribers' telephone stations (published by Hoth).

b ━━━ } Typical power density graph of the electrical background noise injected at the input of the A.R.A.E.N.
c ━━━ } receiving end (obtained at the C.C.I.T.T. Laboratory with two sets of half-octave filters).

FIGURE 3/P.44. — **Power density spectrum of the electrical background noise injected at the input of the A.R.A.E.N. receiving end**

Recommendation P.45 (amended at Mar del Plata, 1968)

MEASUREMENT OF THE A.E.N. VALUE OF A COMMERCIAL TELEPHONE SYSTEM (SENDING AND RECEIVING) BY COMPARISON WITH THE S.R.A.E.N.

This measurement method is described for information in the former text of Recommendation P.45 (*Red Book*, Volume V, pages 69–114). It mentions, *inter alia*, the following conditions of measurement, which differ from the conditions for determining reference equivalents.

a) *Talking distance*

The talking distance used for measurement of a sending A.E.N. value is determined by the mean values of the following parameters (defined in Recommendation P.72):

$$\alpha = 22° \qquad \beta = 12°54' \qquad \delta = 13.6 \text{ cm}$$

The Administration concerned must then supply at the same time as the five subscribers' telephone sets a total of two guard-rings for this "speaking distance" as well as two guard-rings for the measurement of the reference equivalents; the values of the parameters defining this latter "speaking distance" are indicated in the Annex to Recommendation P.72 and are reproduced below:

$$\alpha = 15°30' \qquad \beta = 18° \qquad \delta = 14 \text{ cm}$$

b) *Acoustical speech power to be used during the tests*

The speech power used will be the reference vocal level for A.R.A.E.N. — The reference vocal level for A.R.A.E.N. is that speech power which produces, at a point 33.5 cm directly in front of the lips of the talker, an acoustical speech pressure for each of the three syllables "CAN-CON-BY" of the carrier phrase (used in articulation tests), a deflection of the needle of the indicating instrument of the specified speech voltmeter (see Supplement No. 10, *White Book*, Volume V) connected to a specified microphone and amplifier system equal to that obtained when an acoustic pressure of 1 dyne per cm^2 at 1000 Hz is continuously applied at this same point.

c) *Mounting of the telephone handsets*

With the above values of α, β and δ, it is possible to determine the position of a guard-ring which fixes the position of the talker's mouth relative to the handset. The plane of this ring will be perpendicular to the plane of symmetry of the handset and the centre of the guard-ring will be situated in that plane of symmetry.

Its position is defined by the following geometrical construction carried out in the plane of symmetry of the handset. An origin is taken at the centre of the receiver ear-cap. From this origin a straight line is drawn forming an angle α with the plane of the surface of the ear-cap and in the plane of symmetry of the handset and having a length β. The point thus determined is the centre of the guard-ring and should coincide with the centre point of the lips.

The intersection of the plane of this ring with the plane of symmetry of the handset will be a straight line perpendicular to the direction of speaking as just defined, i.e. that the perpendicular to this straight line will form an angle β with the intersection of the plane of the receiver with the plane of symmetry of the handset.

The position of the guard-ring is thus determined and fixed with respect to the handset.

All that remains is to fix the position in space of the guard-ring during the articulation tests. It is assumed that the operator will talk in such a manner that the plane of symmetry of his face will be vertical. The centre of the guard-ring will be in this plane and the plane of the guard-ring will be perpendicular to it.

Apart from this it has been decided (as a convention) that the plane of the guard-ring will be vertical.

The Administration concerned is requested to supply a setting gauge for each type of handset such that when fixed on the receiver ear-cap the plane of symmetry of the gauge being coincident with that of the handset, the indications marked on the gauge determine the correct position of the guard-ring relative to the handset as has been defined above. In addition this gauge must be fitted with a spirit level placed so that the plane of the guard-ring is vertical when the air bubble is within the central outlined area. By way of example Figure 1/P.45 shows a gauge used at the C.C.I.T.T. Laboratory for one particular type of handset.

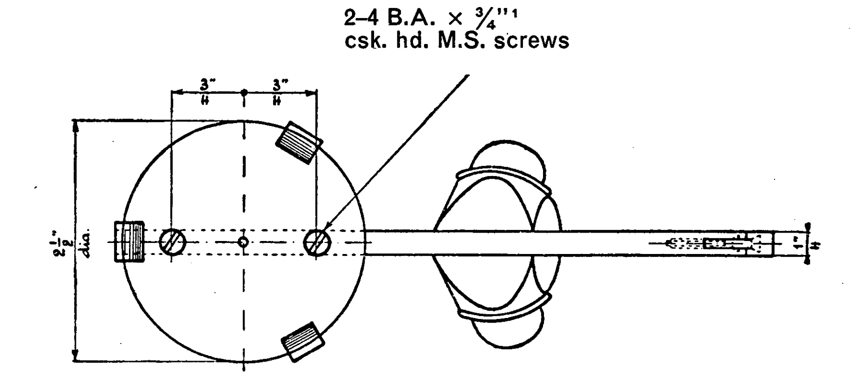
Note. — The position of the guard-ring with respect to the handset is determined uniquely for A.E.N. measurements by the conditions defined above. Provisionally, for each type of handset, it would be desirable to define a gauge which will determine the position of the whole (handset and guard-ring) such that the two following conditions will be satisfied simultaneously:

1) the plane of the guard-ring is vertical;

2) the position with respect to the vertical of the plane of the diaphragm of the microphone capsule is as nearly as possible the same as it would occupy during normal conversation.

d) *Preliminary treatment of the microphone before each talk*

Before each talk and after the handset has been fixed in its support in the appropriate manner, the



Dimensions in inches (1 inch = 25.4 mm) Material: duralumin Finish: clean

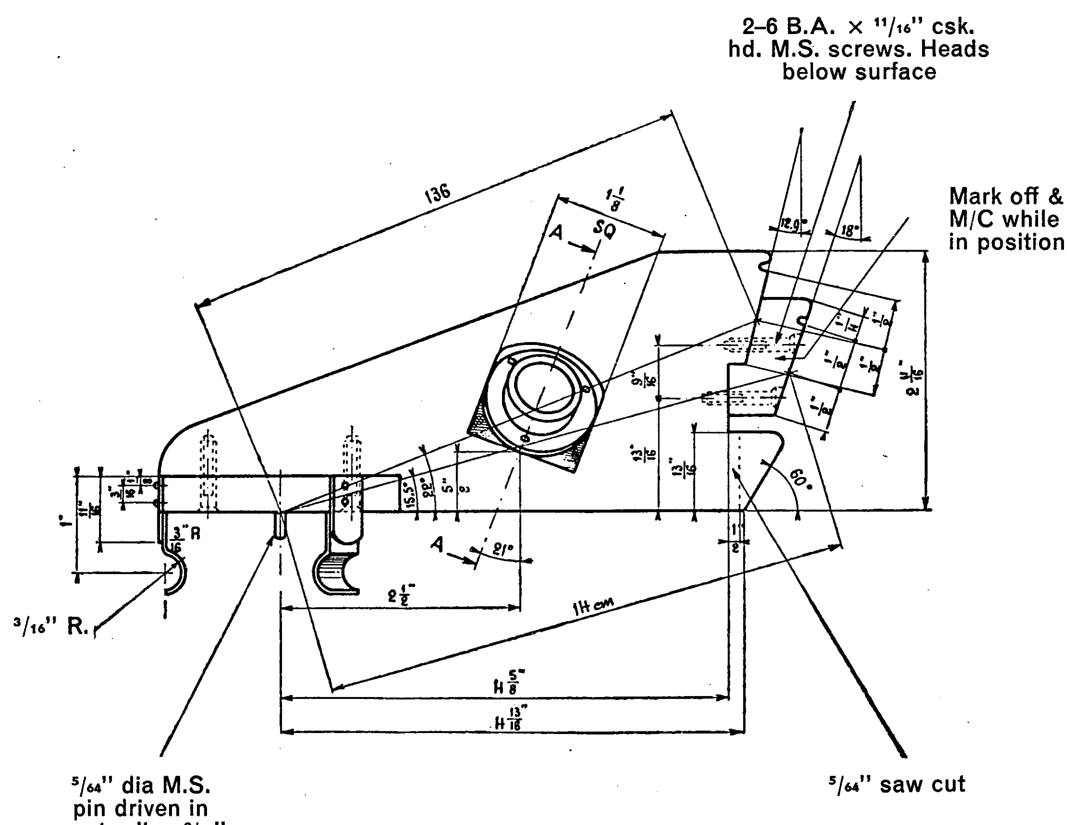


FIGURE 1/P.45. – **Type of gauge used for setting the handsets in articulation tests at the C.C.I.T.T. Laboratory**

Note. – As the numbers refer to the dimensions of a device, the Secretariat considered it better to retain the original units.

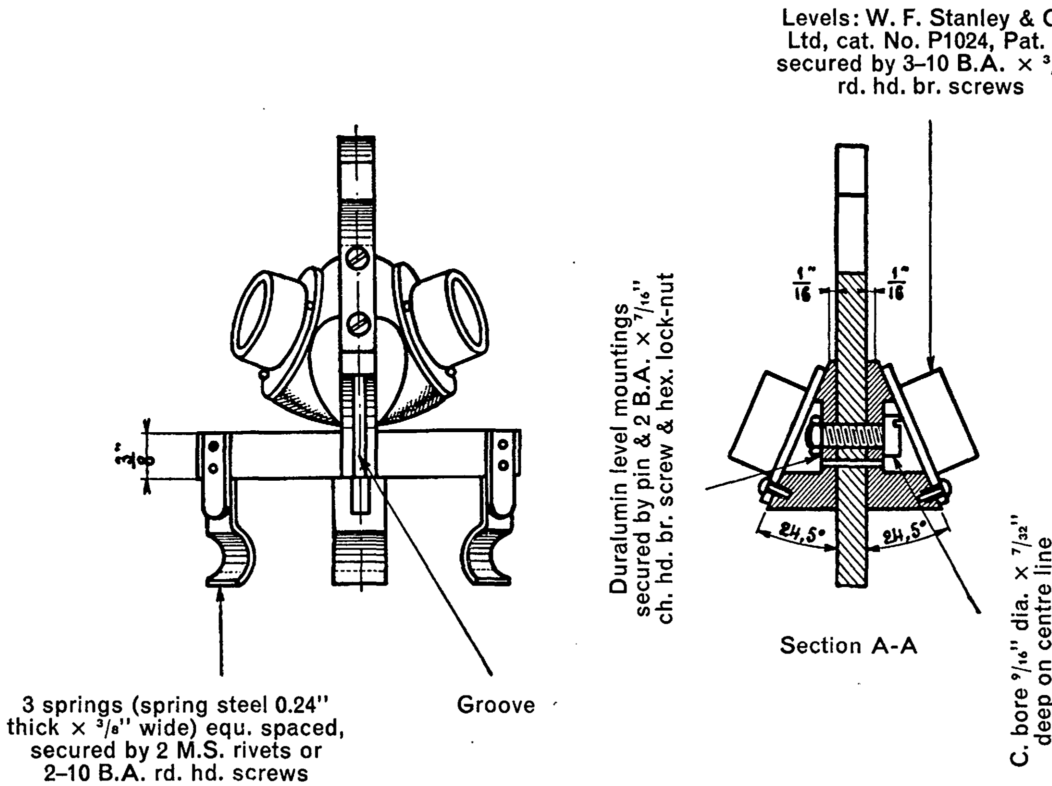


FIGURE 1/P.45 (contd.)

feeding current is applied and the microphone is rotated gently, once forward and once back, about ¾ of a circle and is then fixed in position while avoiding any mechanical shock.

(Paragraphs e and f of the former Recommendation have not been reproduced)

g) *Noise at the receiving end*

For *sending* A.E.N. measurements on a commercial telephone circuit electrical background noise is injected at the input of the A.R.A.E.N. receiving end having a psophometric e.m.f. of 2 mV measured with the commercial telephone circuit psophometer specified by the C.C.I.T.T. (see Recommendation P.53 A). Figure 1 of Recommendation P.44 gives a schematic diagram of the circuit for introducing electrical background noise at the input of the A.R.A.E.N. receiving end and Figure 3 of Recommendation P.44 gives the power density spectrum of this noise.

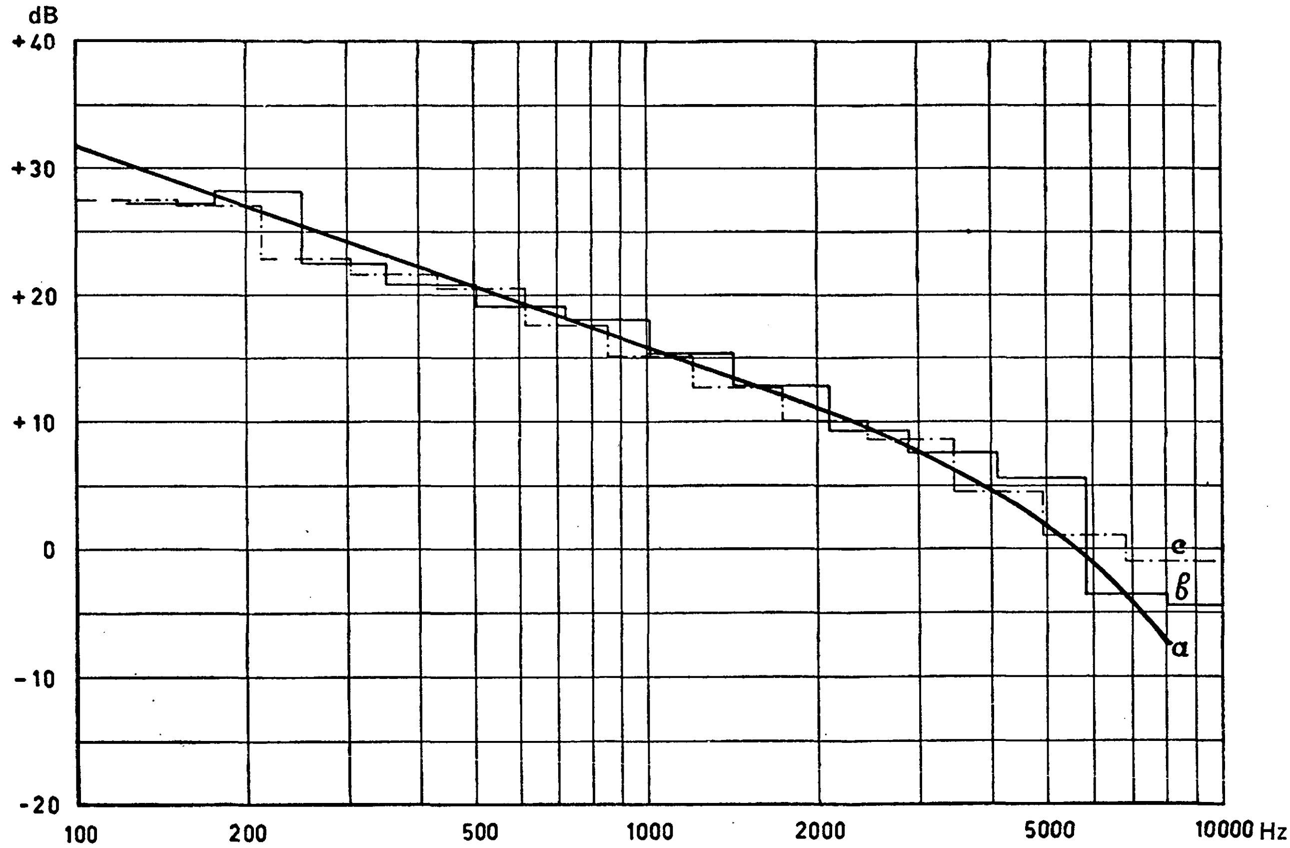
For measurements using a *commercial receiving circuit*, a room noise is used at the receiving end only. This room noise should have a power density spectrum corresponding to that published by Hoth; this is reproduced in Figure 2/P.45 which also shows the spectral distribution of a typical room noise measured in the listening cabinet of the C.C.I.T.T. Laboratory; graphs *b* and *c* represent respectively the results of measurements on this noise made with two sets of half-octave filters.

The acoustic intensity will be 60 dB above a reference point defined by 2×10^{-4} dyne/cm^2 at 1000 Hz in a free progressive wave; this acoustic intensity will be measured with the American sound level meter equipped with weighting network A (Standard Z 24.3.1944 of the American Standards Association, reproduced in Annex 24, Part II of Volume V, *Red Book*).

Note. — Before the XVIIth Plenary Assembly of the C.C.I.F. (Geneva, October 1954), the C.C.I.F. Laboratory determined A.E.N. values in all cases (sending and receiving) with room noise at the receiving end; the present method introduces, with respect to the values previously measured, a difference of −2 dB in the receiving transmission performance rating of a commercial telephone circuit.

h) *Junction*

The junction used throughout the tests comprises a band-pass filter 300–3400 Hz and a variable distortion-less attenuator (the junction of the A.R.A.E.N.). This junction has always the same composition whatever the system, S.R.A.E.N. or commercial, under test.



a ——— Mean power density spectrum of noise observed at telephone subscribers' stations (published by Hoth).

b ——— } Typical graphs obtained at the C.C.I.T.T. Laboratory with two sets of filters.
c —·—·—·— }

FIGURE 2/P.45. — **Power density spectrum of room noise produced in the listening room of the C.C.I.T.T. Laboratory**

Recommendation P.47[1]

CHARGES FOR THE DETERMINATION OF REFERENCE EQUIVALENTS AND A.E.N.s (SENDING AND RECEIVING) OF WORKING STANDARD SYSTEMS AND COMMERCIAL TELEPHONE CIRCUITS IN THE C.C.I.T.T. LABORATORY

These costs are assessed on the basis of the number of hours of work carried out in the C.C.I.T.T. Laboratory; cost per hour of work of the C.C.I.T.T. Laboratory team (of five technical operators) is assessed periodically in Swiss francs (general running costs of the C.C.I.T.T. other than heating and lighting are excluded).

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[1] The IVth Plenary Assembly of the C.C.I.T.T. (Mar del Plata, 1968) discontinued the former recommendations which appeared in Volume V of the *Red Book*:

P.46 Instructions for the forwarding to the C.C.I.T.T. Laboratory of commercial telephone systems with a view to A.E.N. measurements.

P.48 Instructions on how to forward apparatus submitted for reference equivalent or A.E.N. measurements.

1. The number of hours of work for the measurement of reference equivalents depends on the type of apparatus measured and on the purpose of the measurements, i.e. on whether they are for calibrating or re-calibrating equipment.

a) *Calibration of systems using carbon microphones (S.E.T.A.B. or S.E.T.A.C.)*

a.1 Calibration test (sending): 5 hours.
a.2 Calibration test (receiving): 5 hours.

b) *Re-calibration of systems using carbon microphones (S.E.T.A.B. or S.E.T.A.C.)*

b.1 Re-calibration test (sending): 3 hours.
b.2 Re-calibration test (receiving): 3 hours.

c) As regards the calibration or re-calibration of systems other than those mentioned above, e.g. for the measurement of reference equivalents of commercial telephone systems (sending, receiving and side-tone), the Laboratory assesses the actual time spent in carrying out the measurements, in agreement with the Administration or operating agency concerned.

2. The number of hours of work corresponding to measurements of the A.E.N. of a commercial telephone system are as follows:

a) Measurement of A.E.N. (sending): 28 hours;
b) Measurement of A.E.N. (receiving): 28 hours;
c) Measurement of A.E.N. for a complete telephone system: 35 hours.

SECTION 5

OBJECTIVE MEASURING APPARATUS

Recommendation P.51 (amended at Mar del Plata, 1968, and at Geneva, 1972)

ARTIFICIAL VOICES, ARTIFICIAL MOUTHS, ARTIFICIAL EARS

A. General

The C.C.I.T.T.,

considering

that it is highly desirable to design an apparatus for telephonometric measurements such that in future all these measurements may be made with it, without using the human mouth and ear,

that the standardization of the artificial voices, mouths and ears used in the construction of such apparatus is a subject for general study by the C.C.I.T.T.,

that the standardization of an accurate artificial mouth can only be obtained after conclusion of the studies undertaken by various Administrations, comparison of their results and study of the models to check their characteristics;

that in the meantime it would be useful to issue a provisional Recommendation regarding a "sound source" designed in accordance with the sensitivity-frequency characteristics;

provisionally recommends

the use of the artificial ear and the sound source described in Sections B and C below.

Note 1. — The above is still on the understanding that it is considered essential that all telephonometric measurements at the C.C.I.T.T. Laboratory should continue to be made with the human mouth and ear.

Note 2. — Administrations may, if they wish, use in the future, devices which they have been able to construct for large-scale testing of telephone apparatus supplied by manufacturers, provided that the results obtained with these devices are in satisfactory agreement with results obtained by real voice-ear methods.

Note 3. — The Plenary Assembly at Copenhagen in 1936 considered that it would be of interest to deal separately with the design, on the one hand, of an artificial speech source and, on the other, of apparatus for producing a defined acoustic field according to certain specified conditions which will reproduce artificially a human mouth. The term "artificial voice" may be used for the former and "artificial mouth" for the latter.

B. Artificial ear provisionally recommended by the C.C.I.T.T.

1. *Introduction*

For many years the C.C.I.F. studied the possibility of standardizing an artificial ear internationally so that voice-ear measurements could be carried out without using the human ear. Pending such standardization, the 1954 Plenary Assembly recommended that Administrations and the C.C.I.F. Laboratory use a "provisional reference artificial ear" consisting of a simple coupler for the comparison of objective measurements of telephone receivers made in various laboratories. Afterwards, this device was more accurately called the "C.C.I.T.T. reference coupler"[1].

[1] The most recent description of this coupler is to be found in former Recommendation P.51 (*Red Book*, Volume V *bis*, pp. 29-33), with which Annex 17 in Volume V of the *Red Book* is associated.

The International Electrotechnical Commission (I.E.C.), on the other hand, set up a working group in 1960 to draw up certain specifications and recommendations relating to the design of artificial ears, "Objective apparatus replacing the human ear for calibrating different types of earphone".

During the meeting at Liège in 1960, the Working Group proposed the definition of five types of artificial ear:

1 — Simple conventional type
2 — Simple type used for telephonometric applications
3 — Wideband type for audiometric measurements
4 — Special type for calibrating insert earphones
5 — A type which faithfully reproduces the characteristics of the average human ear, for use in laboratory.

Artificial ear type 1 (or reference coupler) is the subject of I.E.C. publication 303; this coupler is different from the "C.C.I.T.T. reference coupler".

The I.E.C. Working Group then concentrated on a study of specifications relating to an ear of type 3. Agreement was reached on the acoustic impedance of the average human ear, after which the Working Group defined an electrical network equivalent to the average human ear and prepared specifications for constructing the type 3 artificial ear. The IVth Plenary Assembly of the C.C.I.T.T. (Mar del Plata, 1968) decided to recommend provisionally that this ear be used for telephonometric measurements, in cases where acoustic leaks do not have to be introduced; the pertinent passages of the I.E.C. publication 318, with some minor amendments, are reproduced below.

The study of type 2 artificial ear and the study of acoustic leaks have therefore been deleted from the programme of work of the I.E.C. and are carried on by the C.C.I.T.T.

2. *Scope, purpose and definition*

2.1 *Scope and purpose*

The present recommendation relates to the specification of an artificial ear which covers the frequency band 20 to 10 000 Hz and is intended for calibrating supra-aural earphones applied to the ear without acoustical leakage.

2.2 *Definition*

The artificial ear is a device at the entry of which the acoustic impedance is the same as the acoustic impedance of the average external human ear, as given in Annex 1. The artificial ear comprises an acoustic network and a measurement microphone which permit calibration of earphones used in audiometry and telephonometry.

3. *Description of the artificial ear for audiometric measurements*

3.1 *Basic design*

The artificial ear is composed of three cavities coupled acoustically. The dimensions of the primary conical cavity and the volumes of the coupled cavities are defined in Figure 1/P.51. The lumped parameter values of the coupling elements shall be adjusted as follows:

$$\begin{aligned}
\mathrm{L}_2 &= 5 \times 10^2\ \mathrm{Ns^2\,m^{-5}}\\
L_3 &= 1 \times 10^4\ \mathrm{Ns^2\,m^{-5}}\\
R_2 &= 6.5 \times 10^6\ \mathrm{Ns\,m^{-5}}\\
R_3 &= 2 \times 10^7\ \mathrm{Ns\,m^{-5}}
\end{aligned}$$

These values relate to normal atmospheric conditions.

Note. — Volume V_1 includes the equivalent volume of the microphone capsule; a corresponding correction for the presence of a protective grid also being taken into account.

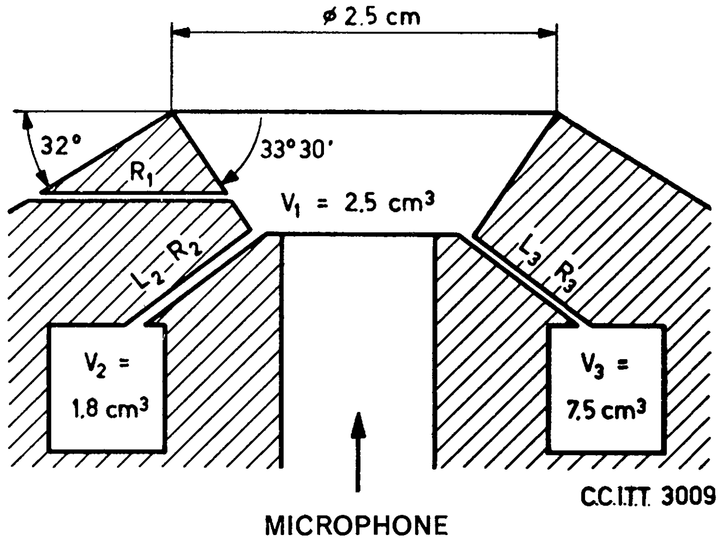


FIGURE 1/P.51

(Tolerances: see sub-clause 3.2)

3.2 *Tolerances*

The linear dimension specified shall be met within a tolerance of ± 0.02 cm, the magnitude of coupled volumes within $\pm 1\%$ and the magnitude of the coupling elements within $\pm 5\%$. The angular dimension 33°30′ shall have a tolerance of $\pm 00°30'$.

Note. — No tolerance has been specified by the C.C.I.T.T. for the angle 32° because it was agreed that when telephone receivers are measured it may be necessary to deviate considerably from this value to ensure that the earphone is properly applied to the artificial ear. In this context Administrations may refer to Annex 1 to Question 12/XII.

3.3 *Pressure equalizing leak*

A leak provided to equalize the pressure shall have an acoustic resistance R_1 greater than 5×10^8 Ns m^{-5} and less than 10^9 Ns m^{-5}. This leakage can be coupled to any one of the three volumes.

3.4 *Microphone*

A microphone forms the base of cavity V_1. The acoustical impedance of the microphone shall be high, the equivalent volume being less than 0.02 cm^3 over the specified range of frequencies. The overall pressure sensitivity of the microphone and associated measuring system over the specified frequency range shall be known with an accuracy of ± 0.2 dB. The microphone shall be coupled to the volume V_1 without leakage.

3.5 *Material*

The artificial ear shall be constructed of a hard, stable, non-magnetic material such as brass.

3.6 *Example of design*

A specific example of the artificial ear is shown in Appendix 2.

4. *Method of use*

See also Annex 1 to Question 12/XII.

The earphone to be calibrated shall be applied to the artificial ear without acoustic leakage with a force of between 4 and 5 N, not including the weight of the earphone itself.

Note that the earphone should not rest on the sloping side of the artificial ear, but only on the upper edge (or rim).

If the earcap of the earphone to be calibrated is made of a very hard material, a wax or grease film of minimal thickness shall be used between earcap and artificial ear in order to eliminate leakage.

5. *Calibration*

For an artificial ear complying with the above requirements, the calibration depends on a knowledge of the overall pressure sensitivity of the microphone and associated measuring system.

It is recommended that manufacturers of artificial ears conforming to this specification describe method(s) for determining the overall stability in an instruction manual.

6. *Use of A.R.A.E.N. earphones with the I.E.C./C.C.I.T.T. artificial ear*

The measurement results contained in Contribution COM XII-No. 125 (period 1968–1972) of the United Kingdom Post Office which coincide, moreover, with those given in Technical Report No. 355 of the C.C.I.T.T. Laboratory demonstrate that the sensitivity of A.R.A.E.N. receiver No. 4026A with rubber earpad may be measured on either the I.E.C./C.C.I.T.T. or A.R.A.E.N. artificial ear to yield substantially the same result providing the receiver is seated in each case on a flat plate flush with the rim of the artificial ear (see Figure 2/P.51).

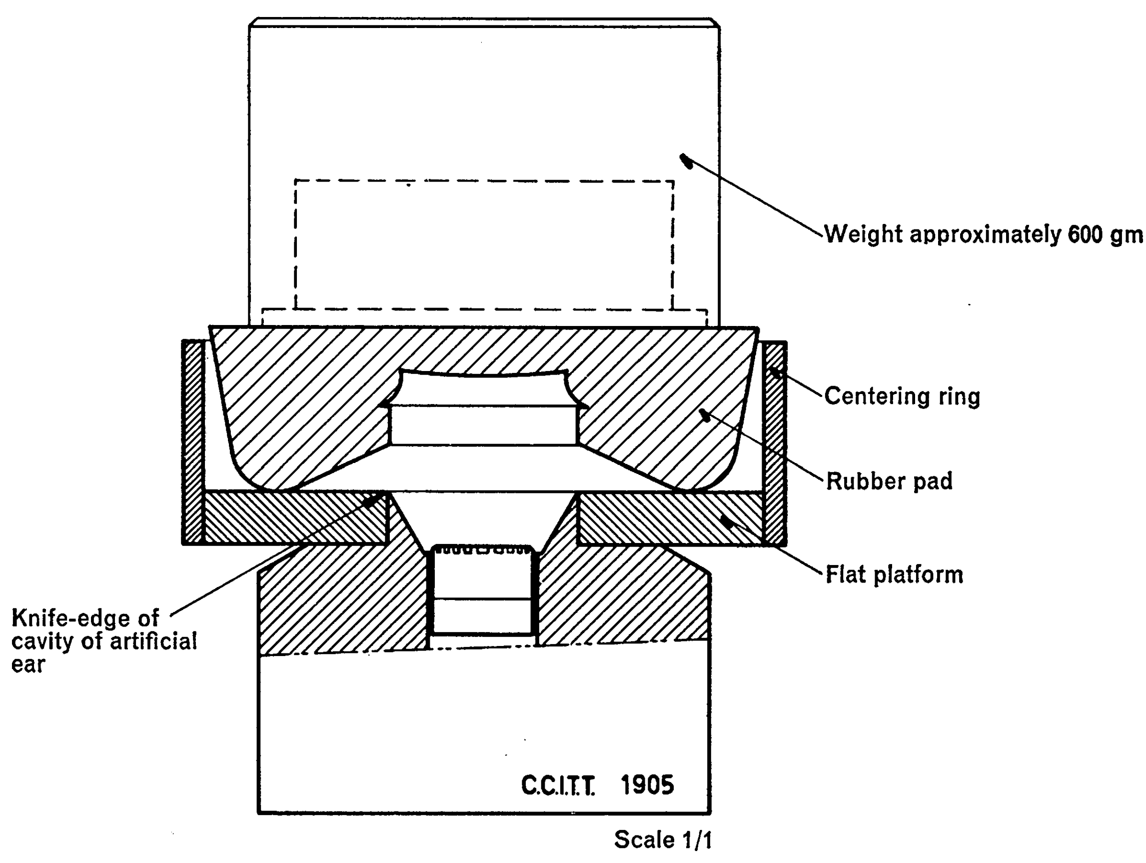


FIGURE 2/P.51. — **A.R.A.E.N. receiver applied to a platform mounted flush with the top of the input cavity of the I.E.C./C.C.I.T.T. artificial ear**

Furthermore, it is known that the setting-up of the A.R.A.E.N. receiving end was originally based on the agreement between real ear calibrations of the receiver No. 4026A with rubber earpad and the calibration method used above with the A.R.A.E.N. artificial ear.

The C.C.I.T.T. therefore recommends that, for future objective measurements of the *A.R.A.E.N. receive end* of the type used for exploring the correlation between subjectively measured loudness ratings and calculated ratings based on objective measurements, the I.E.C./C.C.I.T.T. artificial ear be used with a flat plate as described. The receiver should be seated on the artificial ear with a mass of 600 grams (excluding the receiver mass).

Note 1. — This recommendation is made solely in connection with the calibration of receiver No. 4026A with rubber earpad. It is assumed that receivers of conventionally shaped telephone handsets will be seated directly on the artificial ear as prescribed in Section 4 of I.E.C. Publication 318 and C.C.I.T.T. Recommendation P.51.

Note 2. — This recommendation applies not only to the A.R.A.E.N. receiving end but also to that of N.O.S.F.E.R., for tests of the type described above. It implies no change in the absolute calibration of A.R.A.E.N. described in Recommendation P.41 and Supplement No. 9 to Volume V of the *White Book*.

C. Sound source provisionally recommended by the C.C.I.T.T.

1. *Introduction*

Before recommending a particular type of artificial mouth as suitable for objective telephonometric measurements, it is proposed that, as a first stage, experience should be gained in the use of one form of sound source to determine the shape of the sensitivity-frequency characteristic of a commercial sending system to be obtained whatever type of microphone inset is used.

Such a sound source is required to permit useful comparisons to be made between the results obtained in various laboratories. This advantage already exists for comparison of the sensitivity-frequency curves of earphones since the adoption by the C.C.I.T.T. at Mar del Plata of the I.E.C./C.C.I.T.T. artificial ear.

It would be desirable to supplement existing documentation on the human mouth.

Note. — It is not proposed that the choice of sound source should prejudice definition of a more precise artificial mouth that can be used universally for measuring objective ratings.

2. *Acoustic characteristics of the sound source*

2.1 The sound source must permit calibration of microphone at short distances.

2.2 At the measuring distances normally used, the acoustic properties should be close to those of the average human mouth; in particular, the law of decrease in sound pressure on the axis should be close to that of the average human mouth from a distance of about 10 mm onwards from a plane called the lip plane of the source.

The following table shows the sound pressures measured by some Administrations at points along the axis and expressed in relation to the sound pressure at 25 mm from the lip plane. The sound pressures should be measured with a very small (say, about 6 mm diameter) microphone or a probe microphone.

Distance from lip plane (mm)	Relative sound pressure level (dB relative to the sound pressure 25 mm from the lip plane)		
	U. K. Post Office	Chile Telephone Co.	L. M. Ericsson
10	+4.8	+5.5	+4.6
20	+1.5	+1.5	+1.3
25	0	0	0
40	−3.3	−3.3	−3.4
60	−6.5	(See Note)	

Note. — Beyond 40 mm, the sound pressure can be assumed to be inversely proportional to the distance from an equivalent point source lying 6 mm behind the lip plane.

2.3 The directivity, in a region of space around the axis, should be close to that of the average human mouth.

2.4 For the measurements obtained with different specimens of the source to be comparable, it is necessary to define a reference point on the main axis at which the characteristics of the source will be checked and which will serve as reference in inter-laboratory tests. It is suggested that a point on the axis 25 mm from the lip plane would be suitable.

2.5 As a preliminary indication, the sound source should be able to deliver to the above reference point acoustic pressure levels of not less than 90 dB (relative to $2 . 10^{-5}$ Pa (pascal)) in a frequency range comprising at least the 200–4000 Hz band. (Sound pressure levels up to 100 dB over the frequency range 100–8000 Hz would be desirable.)

2.6 The source should be stable and reproducible.

3. *Choice of a model*

The results of measurements made with the modified B & K 4216 source and the United Kingdom Post Office artificial mouth have shown good agreement between the two models.

These results are not very different from the values measured for the human mouth so far as the distribution of acoustic pressure in a free field along the axis is concerned.

These two models also meet the other specifications of paragraph 2 above.

Note 1. — The modification made to the B & K 4216 sound source consists largely in bringing the lip ring nearer to the regulation microphone. The distance between the ring and the plane of the microphone orifice on the modified source is 9.7 mm (see C.C.I.T.T. Laboratory Technical Report No. 397).

Note 2. — The artificial mouth of the United Kingdom Post Office is not commercially available and the B & K 4216 model is not now manufactured. However, the B & K 4219 model is believed to have performances equivalent to those of the 4216 model, as modified by the laboratory of C.C.I.T.T.

List of references

Brüel, P. V., Frederiksen, E. and Rasmussen, G.: Artificial ears for the calibration of earphones of the external type; *B. & K. Tech. Rev.* No. 4 (1961) and No. 1 (1962).

Delany, M. E.: The acoustical impedance of human ears; *J. Sound Vib. 1* (1964), 455.

DELANY, M. E., WHITTLE, L. S., COOK, J. P. and SCOTT, V.: Performance studies on a new artificial ear; *Acustica 18* (1967), 231.

ITHELL, A. H.: A determination of the acoustical input impedance characteristics of human ears; *Acustica 13* (1963), 311.

ITHELL, A. H., JOHNSON, E. G. T. and YATES, R. F.: The acoustical impedance of human ears and a new artificial ear; *Acustica 15* (1965), 109.

ANNEX 1

(to Recommendation P.51)

Lumped-parameter electrical network analogue of the average human ear

In the analogue network shown in Figure 1 one electrical ohm corresponds to 10^5Ns m^{-5}.

Three independent determinations of the acoustical impedance of the average human ear under no-leak conditions were available (see references) covering various earcap contours used on audiometric earphones. In each case an analogue network of the type shown in Figure 3/P.51 was devised with values of the elements adjusted to produce optimum fit to the experimental impedance data. The values of the lumped-parameters shown in Figure 3/P.51 are average values corresponding to a plane earcap.

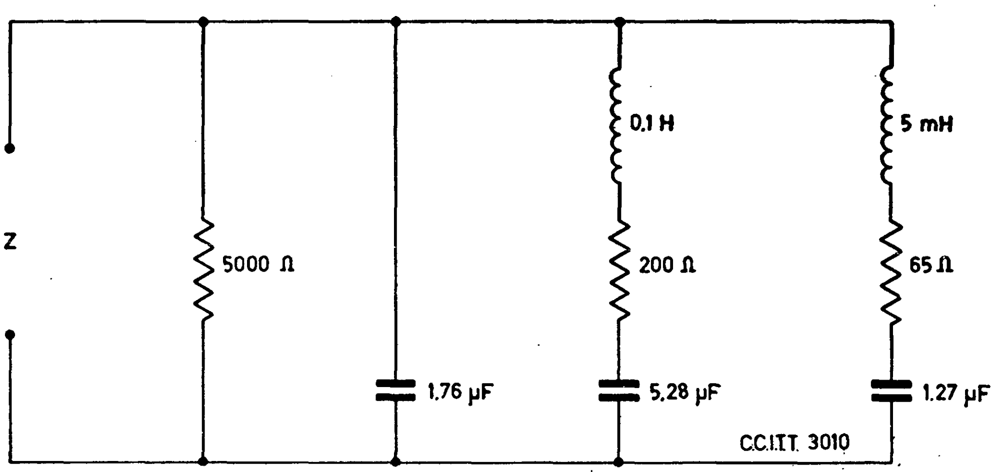


FIGURE 3/P.51. — **Lumped-parameter electrical network analogue of the average human ear. The real and imaginary components of the impedance Z are shown as functions of frequency in Figures 4/P.51 and 5/P.51 following**

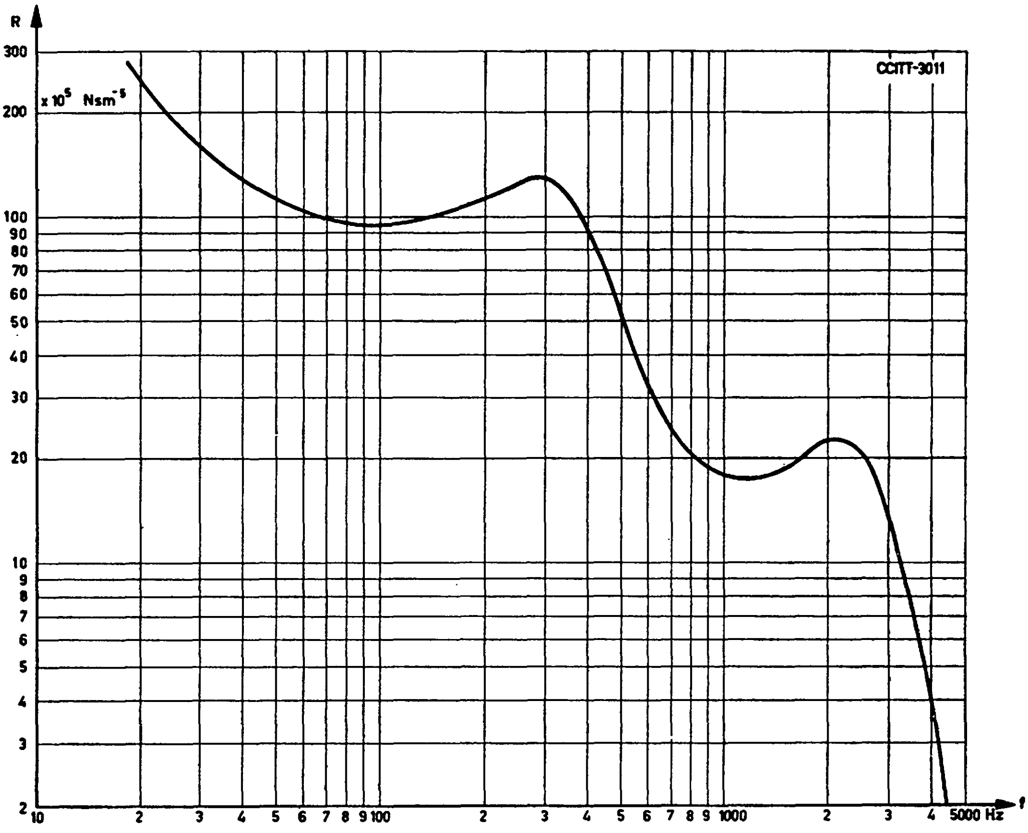


FIGURE 4/P.51. – **Real component of the impedance of the electrical analogue network**

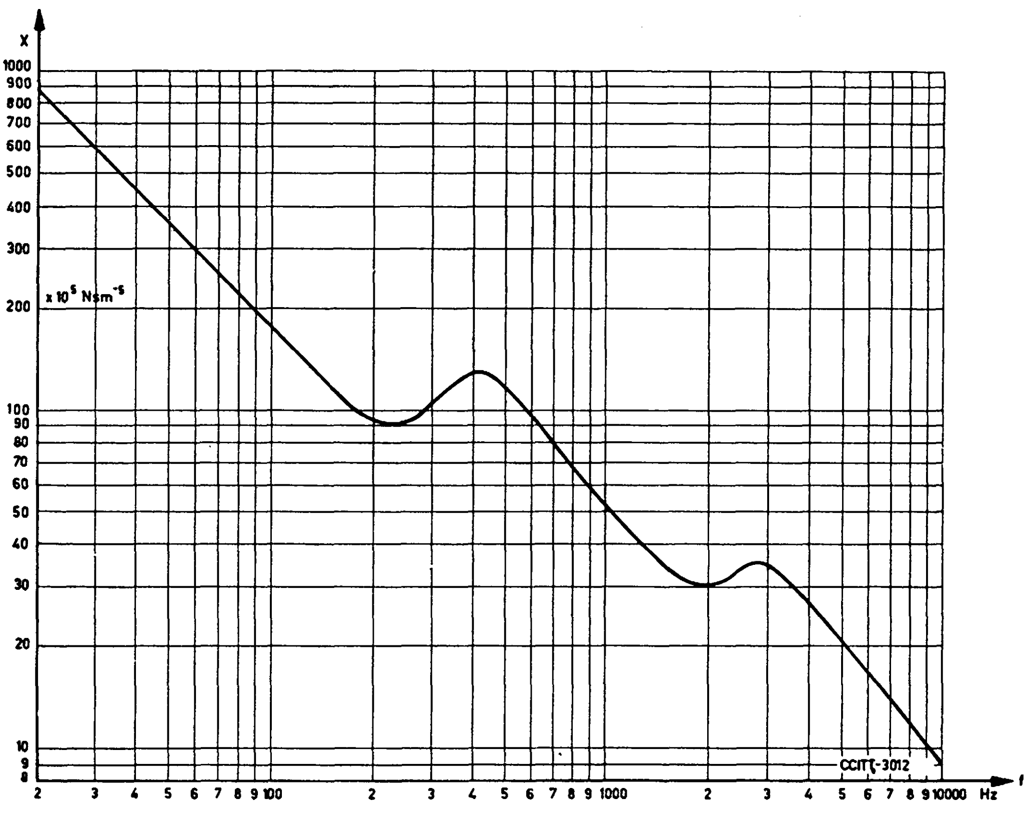
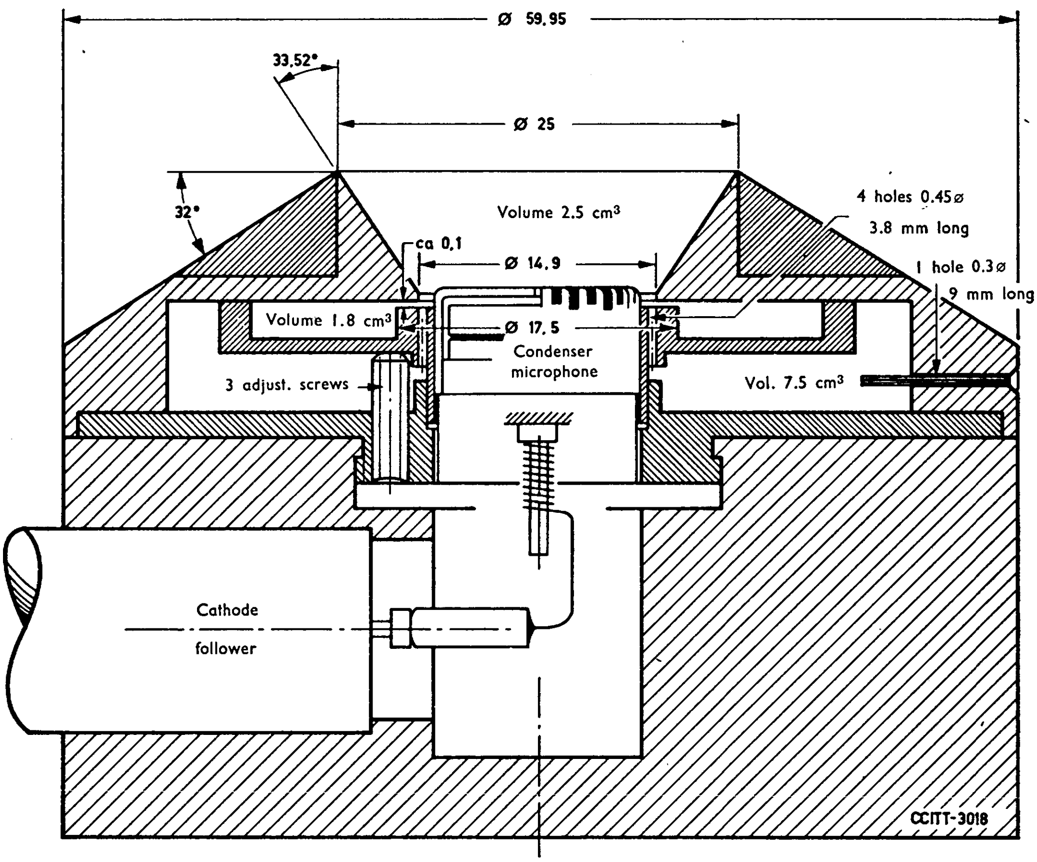


FIGURE 5/P.51. – **Imaginary component of the impedance of the electrical analogue network**

ANNEX 2

(to Recommendation P.51)

Example of one specific design of the artificial ear



All linear dimensions in mm.

Note. — The three adjusting screws are set so the corresponding flow resistance is 6.5×10^6 Ns m^{-5}.

Recommendation P.52

VOLUME METERS

The C.C.I.T.T. considers that, in order to ensure continuity with previous practice, it is not desirable to modify the specification of the volume meter of the A.R.A.E.N. employed at the C.C.I.T.T. Laboratory.

The table below gives the principal characteristics of various measuring devices used for monitoring the volume or peak values during telephone conversations or sound-programme transmissions.

Note. — Descriptions of the following devices are contained in the Supplements to *White Book*, Volume V:
1. A.R.A.E.N. volume meter or speech voltmeter: Supplement No. 10.
2. Volume meter standardized in the United States of America, termed the "vu meter": Supplement No. 11.
4. Peak indicator used by the British Broadcasting Corporation: Supplement No. 12.
5. Maximum amplitude indicator Types U21 and U71 used in the Federal Republic of Germany: Supplement No. 13.

The Volume Indicator — S.F.E.R.T. Volume Indicator (3) which used to be used in the C.C.I.T.T. Laboratory is described in Annex 18 (2nd Part of Volume V of the *Red Book*).

Principal characteristics of the various instruments used for monitoring the volume or peaks during telephone conversations or programme sound-transmissions

Type of instrument	Rectifier characteristic (Note 4)	Time to reach 99% of final reading (milliseconds)	Integration time (milliseconds) (Note 5)	Time to return to zero (value and definition)
(1) "Speech voltmeter" United Kingdom Post Office Type 3 (S.V.3) identical to the speech power meter of the A.R.A.E.N.	2	230	100 (approx.)	equal to the integration time
(2) vu meter (United States of America) (Note 1)	1.0 to 1.4	300	165 (approx.)	equal to the integration time
(3) Speech power meter of the "S.F.E.R.T. volume indicator"		around 400 to 650	200	equal to the integration time
(4) Peak indicator for programme transmissions used by the British Broadcasting Corporation (B.B.C. Peak Programme Meter) (Note 2)	1		10 (Note 6)	3 seconds for the pointer to fall 26 dB
(5) Maximum amplitude indicator used by the Federal German Republic (type U 21)	1	around 80	5 (approx.)	1 or 2 seconds from 100% to 10% of the reading in the steady state
(6) OIRT — Programme level meter:		for both types:		for both types:
type A sound meter		less than 300 ms for meters with pointer indication and less than 150 ms for meters with light indication	10 ± 5	1.5 to 2 seconds from the 0 dB point which is at 30% of the length of the operational section of the scale
type B sound meter			60 ± 10	

Notes to the table

Note 1. — In France a meter similar to the one defined in line (2) of the table has been standardized.

Note 2. — In the Netherlands a meter (type N.R.U.-ON301) similar to the one defined in line (4) of the table has been standardized.

Note 3. — In Italy a sound-programme meter with the following characteristics is in use:

Rectifier characteristic: 1 (see note 4)

Time to reach 99% of final reading: approx. 20 ms

Integration time: approx. 1.5 ms

Time to return to zero: approx. 1.5 s from 100% to 10% of the reading in the steady state.

Note 4. — The number given in the column is the index n in the formula $[V_{(\mathrm{output})} = V_{(\mathrm{input})}{}^{n}]$ applicable for each half-cycle.

Note 5. — The "integration time" was defined by the C.C.I.F. as the "minimum period during which a sinusoidal voltage should be applied to the instrument for the pointer to reach to within 0.2 neper or nearly 2 dB of the deflection which would be obtained if the voltage were applied indefinitely". A logarithmic ratio of 2 dB corresponds to a percentage of 79.5% and a ratio of 0.2 neper to a percentage of 82%.

Note 6. — The figure of 4 milliseconds that appeared in previous editions was actually the time taken to reach 80% of the final reading with a d.c. step applied to the rectifying/integrating circuit. In a new and somewhat different design of this programme meter using transistors, the performance on programme remains substantially the same as that of earlier versions and so does the response to an arbitrary, quasi-d.c. test signal, but the integration time, as here defined, is about 20% greater at the higher meter readings.

Comparative tests with different types of volume meters

A note which appears on pages 270 to 293 of Volume IV of the *White Book* of the C.C.I.F. (Budapest, 1934) gives some information on the results of preliminary tests conducted at the S.F.E.R.T. Laboratory to compare the Volume Indicator with different impulse indicators.

The results of comparative tests made in 1952 by the United Kingdom Post Office appear in Supplement No. 14 to the *White Book*, Volume V.

Recommendation P.53

PSOPHOMETERS (APPARATUS FOR THE OBJECTIVE MEASUREMENT OF CIRCUIT NOISE)

A. Psophometer for commercial telephone circuits

The C.C.I.T.T.,

considering

that, since the psophometer for commercial telephone circuits was specified (*Directives concerning the protection of communication lines against the interfering effects of electric power lines*, Rome Edition, 1937, revised at Oslo, 1938), considerable progress has been made in the construction of the subscriber's telephone apparatus, especially so far as the smoothness of the sensitivity-frequency characteristic is concerned,

that the "Joint Subcommittee on development and research of the Edison Electric Institute and the Bell Telephone System" (*Engineering Report*, No. 45) has carried out numerous tests to determine the curve to be prescribed for the psophometer filter network in order to take account of the improved characteristics of the subscriber's telephone equipment,

that numerous tests and measurements made in the course of the last few years show that the electro-acoustic characteristics of the subscriber's telephone equipment used in Europe are very similar to those of American equipment and that, consequently, it is unnecessary to repeat in Europe similar tests to those described by the Joint Subcommittee, *unanimously recommends*

that the weights attributed to different frequencies in the weighting network of the psophometer used for measurements at the terminals of a commercial trunk telephone circuit should be those in Table 1 (see also the curve given in Figure 1/P.53); only the values in bold type in the table should be considered as specifying the psophometer filter network and should be taken into consideration for check tests of the apparatus; the other values, obtained by interpolation, are given to facilitate any calculations.

By convention, the numerical values are determined by attributing the value 1000 to the frequency 800 Hz. The logarithmic weighting values are obtained by attributing the value corresponding to 0 dB to the frequency 800 Hz.

Permissible tolerances

The following permissible tolerances are:

50 to 300 Hz: ± 2 dB
300 to 800 Hz: ± 1 dB
800 Hz: 0 dB
800 to 3000 Hz: ± 1 dB
3000 to 3500 Hz: ± 2 dB
3500 to 5000 Hz: ± 3 dB

Note 2. — During the XVIth Plenary Assembly (Florence, 1951), the C.C.I.F. considered that it would be extremely undesirable to make any modifications in the weighting table or to the specification of the psophometer for as long a period as possible, for example for ten years.

Measurements at the terminals of a subscriber's telephone receiver

The psophometer, which was standardized by the XVIth Plenary Assembly of the C.C.I.F. for relatively stable circuit noise measurements, consists, for use at the end of an international telephone circuit (see above), of a filter network which takes account of the characteristics of a fairly modern type of telephone set used in the United States of America together with the mean characteristics of the national telephone network of that country. According to American practice, if it is desired to use this psophometer at the terminals of the telephone receiver, it is adapted for this purpose by removing that part of the filter network which takes account of the characteristics of the commercial telephone circuits. It seems unnecessary to have recourse to such a modification in Europe since the characteristics of telephone sets used in Europe cover a wide range. Choice of a single characteristic for the filter network which would result from a modification of this kind would probably be as arbitrary as would be the use, without modification, for measurements at the terminals of the telephone receiver, of the psophometer with filter network specified by the XVIth Plenary Assembly of the C.C.I.F. for measurements at the terminals of a commercial trunk telephone circuit (see above).

When only comparative measurements are needed, the psophometer specified by the XVIth Plenary Assembly of the C.C.I.F. can very well be used, without modification, as a voltmeter of which the characteristics have been arbitrarily fixed, to make measurements at the terminals of the subscriber's telephone receiver.

For studies of a fundamental nature, Administrations may very well wish to use filter networks specially chosen to be appropriate for the studies concerned.

Correspondence with the readings of American psophometers

Information now used by the American Telephone and Telegraph Company in assessing noise impairment is given in an article by D. A. Lewinski in the *Bell System Technical Journal*, March 1964 [1]. In this article, noise is expressed in terms of readings with C-message weighting on the 3A noise meter now used in the United States. Because the weighting differs from that associated with the older 2B noise meter and the C.C.I.T.T. 1951 psopho-

TABLE 1

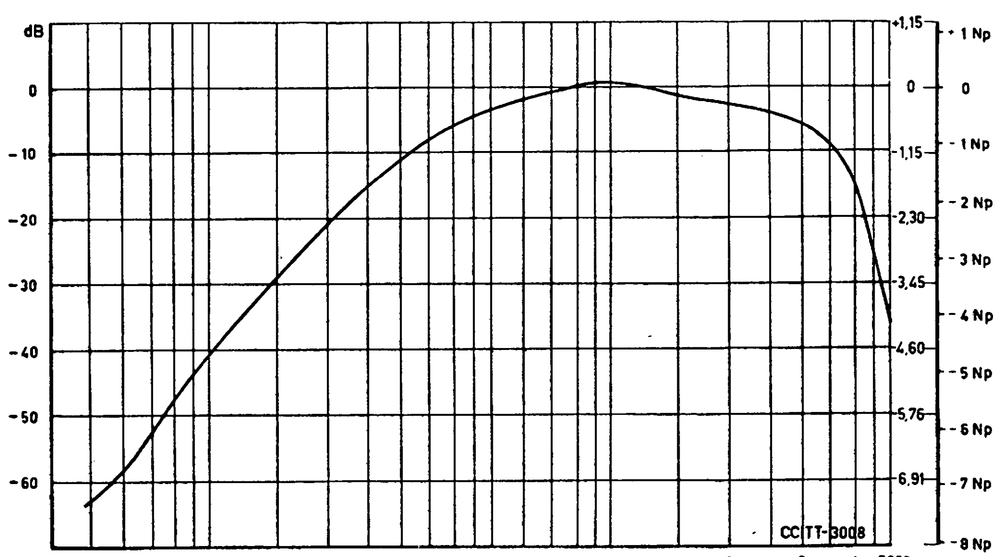
TABLE OF COMMERCIAL TELEPHONE CIRCUIT PSOPHOMETER WEIGHTING COEFFICIENTS

Frequency Hz	Weight		
	Numerical value	Numerical value squared	Value in decibels
16.66	0.056	0.003136	−85.0
50	0.71	0.5041	−63.0
100	8.91	79.3881	−41.0
150	35.5	1 260.25	−29.0
200	89.1	7 938.81	−21.0
250	178	31 684	−15.0
300	295	87 025	−10.6
350	376	141 376	− 8.5
400	484	234 256	− 6.3
450	582	338 724	− 4.7
500	661	436 921	− 3.6
550	733	537 289	− 2.7
600	794	630 436	− 2.0
650	851	724 201	− 1.4
700	902	813 604	− 0.9
750	955	912 025	− 0.4
800	1 000	1 000 000	0.0
850	1 035	1 071 225	+ 0.3
900	1 072	1 149 184	+ 0.6
950	1 109	1 229 881	+ 0.9
1 000	1 122	1 258 884	+ 1.0
1 050	1 109	1 229 881	+ 0.9
1 100	1 072	1 149 184	+ 0.6
1 150	1 035	1 071 225	+ 0.3
1 200	1 000	1 000 000	0.0
1 250	977	954 529	− 0.20
1 300	955	912 025	− 0.40
1 350	928	861 184	− 0.65
1 400	905	819 025	− 0.87
1 450	881	776 161	− 1.10
1 500	861	741 321	− 1.30
1 550	842	708 964	− 1.49
1 600	824	678 976	− 1.68
1 650	807	651 249	− 1.86
1 700	791	625 681	− 2.04
1 750	775	600 625	− 2.22
1 800	760	577 600	− 2.39
1 850	745	555 025	− 2.56
1 900	732	535 824	− 2.71
1 950	720	518 400	− 2.86
2 000	708	501 264	− 3.00
2 050	698	487 204	− 3.12
2 100	689	474 721	− 3.24
2 150	679	461 041	− 3.36
2 200	670	448 900	− 3.48
2 250	661	436 921	− 3.60
2 300	652	425 104	− 3.72
2 350	643	413 449	− 3.84
2 400	634	401 956	− 3.96
2 450	626	390 625	− 4.08
2 500	617	380 689	− 4.20
2 550	607	368 449	− 4.33
2 600	598	357 604	− 4.46
2 650	590	348 100	− 4.59
2 700	580	336 400	− 4.73
2 750	571	326 041	− 4.87
2 800	562	315 844	− 5.01
2 850	553	305 809	− 5.15
2 900	543	294 849	− 5.30
2 950	534	285 156	− 5.45
3 000	525	275 625	− 5.60

TABLE 1 (contd.)

TABLE OF COMMERCIAL TELEPHONE CIRCUIT PSOPHOMETER WEIGHTING COEFFICIENTS

Frequency Hz	Weight		
	Numerical value	Numerical value squared	Value in decibels
3 100	501	251 001	− 6.00
3 200	473	223 729	− 6.50
3 300	444	197 136	− 7.05
3 400	412	169 744	− 7.70
3 500	376	141 376	− 8.5
3 600	335	112 225	− 9.5
3 700	292	85 264	−10.7
3 800	251	63 001	−12.0
3 900	214	45 796	−13.4
4 000	178	31 684	−15.0
4 100	144.5	20 880.25	−16.8
4 200	116.0	13 456	−18.7
4 300	92.3	8 519.29	−20.7
4 400	72.4	5 241.76	−22.8
4 500	56.2	3 158.44	−25.0
4 600	43.7	1 909.69	−27.2
4 700	33.9	1 149.21	−29.4
4 800	26.3	691.69	−31.6
4 900	20.4	416.16	−33.8
5 000	15.9	252.81	−36.0
>5 000	<15.9	<252.81	<−36.0
Note. — If, for the planning of certain telephone transmission systems, calculations are made on a basis of the psophometric weighting values and if it appears useful to adopt, for frequencies above 5000 Hz, more precise values than those given in the above table, the following values may be used:			
5 000 to 6 000	<15.9	<252.81	<−36.0
>6 000	< 7.1	< 50.41	<−43.0



Frequency in Hz

FIGURE 1/P.53. — **Characteristic curve of the psophometer filter network used for measurements at the terminals of a commercial trunk telephone circuit** meter, the relationship among measurements with these instruments is influenced by the spectrum of the noise measured. If one milliwatt of white noise in the band 300-3400 Hz is applied to each, the following readings are obtained:

3A noise meter (C-message weighting)	88 dBrn
2B noise meter (F1A weighting)	81.5 dBa
C.C.I.T.T. psophometer (1951 weighting)	−2.5 dBm

Recognizing that the relationship will change for other noise spectra, the following rounded conversion factors are proposed for practical comparison purposes:

C.C.I.T.T. 1951 weighting		3A noise meter C-message weighting		2B noise meter F1A weighting
0 dBm	=	90 dBrn	=	84 dBa
−90 dBm	=	0 dBrn	=	−6 dBa
−84 dBm	=	6 dBrn	=	0 dBa

These conversion factors include the effect of the difference between the reference frequencies used (800 Hz in the C.C.I.T.T. psophometer, 1000 Hz in the American noise meters).

Detailed information concerning the noise meters used in the United States is referred to in references [2] and [3] below.

References

[1] Lewinski, D. A.: A New Objective for Message Circuit Noise. *Bell System Technical Journal*, 43, March 1964, page 719.

[2] Cochran, W. T. and Lewinski, D. A.: A New Measuring Set for Message Circuit Noise. *Bell System Technical Journal*, 39, July 1960, page 911.

[3] Aikens, A. J. and Lewinski, D. A.: Evaluation of Message Circuit Noise. *Bell System Technical Journal*, 39, July 1960, page 879.

Measurement of impulsive noise

(See Recommendation P.55)

Essential clauses of a model specification for the provision of a psophometer for commercial telephone circuits

The C.C.I.T.T.,

considering, on the one hand,

that the design of a psophometer for commercial telephone circuits which will permit measurements to be made at frequencies lower than 40 Hz and particularly 16 2/3 Hz would present construction difficulties and would result in a heavy and cumbersome instrument,

that the need to use the instrument at these frequencies would arise infrequently,

that, when these frequencies are encountered, it seems possible that the instrument could be used as it stands with the addition of a suitable correcting network;

considering, on the other hand,

that the provisional essential clauses of a model specification for a psophometer for commercial telephone circuits appears to be in insufficient detail so far as measurement of voltages of these types is concerned,

that it would seem proper to provide a check test to this effect,

that, nevertheless, by reason of the variety of designs of psophometers, it would seem impossible to recommend uniform testing clauses but it would seem useful to draw attention to this point, *unanimously recommends*

that it is advisable that psophometers for commercial telephone circuits should conform to the following conditions:

1. *Graduation.* — The psophometer should be so graduated that, for each sensitivity provided, it gives by direct reading (or after multiplication by a factor defined by the sensitivity setting) the exact value of the voltage when a voltage at 800 Hz is applied to the input of the psophometer.

2. *Sensitivity.* — The psophometer should enable a clear reading to be obtained when a voltage at 800 Hz of at least 0.05 millivolt is applied to the input. It should also permit a direct reading of voltages at least up to 100 millivolts without the use of external potentiometer devices.

3. *Measurements.* — For every measuring range and under every condition of use of the instrument, for each sensitivity and for each frequency applied separately, the readings should be equal to the product of the applied voltage and the weighting coefficient for that frequency, divided by one thousand.

When the applied voltage consists of a number of different frequency components, the reading on the indicating instrument should be equal to the square root of the sum of the squares of the readings corresponding to the individual components applied separately.

To check that this condition is satisfied, it is possible to use, for example, the following procedure. Two sinusoidal voltages are applied successively at different frequencies which are not harmonically related and which give the same deflection on the needle of the indicating instrument; the resultant of these two voltages is then applied by means of an arrangement which allows them to be attenuated equally and adjusted so as to restore the deflection previously obtained. The loss introduced should be equal to 3 dB with a tolerance of ± 0.5 dB.

The test should be made using different pairs of frequencies, some close together and others well apart. It should be repeated at different deflections of the needle of the psophometer.

4. *Linearity.* — When the periodic voltage waveform applied is peaky so that the peak value is much greater than the effective value, the corresponding weighted voltage measurement should be as much as possible free from any error caused by overloading the amplifier or other parts. It is possible to check whether this source of error has been eliminated by one of the following methods given as examples.

First method. — A voltage is applied to the psophometer at a frequency of the order of 2000 Hz in 5-millisecond pulses separated from each other by 20-millisecond intervals of silence. When the applied voltage is decreased from a value corresponding to the highest which can be measured by the apparatus, the readings should be proportional to the applied voltage with a tolerance of $\pm 5\%$ (or ± 0.5 decibel).

Second method. — The United Kingdom Post Office has adopted the following rule:

The psophometer contains a d.c. indicating instrument preceded by a square-law rectifier. The instrument is so graduated that Condition 1 is satisfied.

For a sinusoidal voltage of given frequency and for a fixed adjustment of the gain controls, the operating current of the indicating instruments should be proportional to the square of the voltage applied to the psophometer for values of this voltage between 0.4 to 2.5 times that required to produce a full-scale deflection with a tolerance of $\pm 10\%$ corresponding to an error in reading of about $\pm 5\%$ (or ± 0.5 decibel).

The following method of check is adopted: between the rectifier and the indicating instrument is inserted a network such that a known fraction of the rectified current passes through this instrument whilst the impedance presented to the rectifier remains the same as is presented to it by the indicating instrument when this is directly connected. By these means the deflection can be brought back to a value lower or equal to the maximum of the scale graduations and thus check that the condition is satisfied.

Third method. — Another convenient recognized test in the case of a psophometer containing a d.c. indicating instrument preceded by a square-law rectifier consists of carrying out the test described in 3, but applying a voltage having two sinusoidal components with values equal to 0.4; 1; 1.5; 2 and 2.5 times that corresponding to the full deflection of the indicating instrument. The deflection is reduced to a value equal to or less than the full scale by using a reducing network such as was involved in the description of the second method.

5. *Dynamic characteristic.* — The dynamic characteristic of the psophometer should be such that a noise of duration of the order of 0.15 to 0.25 second produces the same deflection as a continuous noise, whilst noises of shorter duration produce proportionately smaller deflections. This period is that which seems necessary for the noise to be entirely heard.

6. *Input impedance.* — The input impedance of the psophometer should be as large as possible over the whole frequency band 15 to 5000 Hz. In particular it should be at least 6000 ohms from 40 to 5000 Hz.

The impedance between the two terminals connected together and the case of the psophometer should be as high as possible at all frequencies from 15 to 5000 Hz. In particular it should be greater than 200 000 ohms at 800 Hz.

7. *Balance.* — The balance of the psophometer with respect to the case should be such that the application between the mid-point of a 600-ohm resistor connected to the input terminals and the case (Figure 2/P.53) of a voltage of 200 volts at 50 Hz, or 30 volts at 300 Hz or 10 volts at 800 Hz does not give a reading greater than 0.1 millivolt.

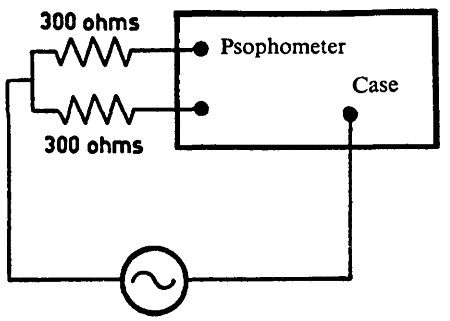


FIGURE 2/P.53

8. *Freedom from the effects of external fields.* — The apparatus should be free from the effects of external magnetic and electric fields even when used in the vicinity of power plant. In this respect it is necessary to note that external fields can affect the apparatus either in front of or after the range control (potentiometer) and accordingly the effects of these fields may or may not depend upon the setting of the range control.

The psophometer together with the boxes containing the power supplies should be screened; the various external connections should be made with twisted and screened conductors. It is desirable to provide terminals so that all parts of the apparatus and their boxes can be earthed while the psophometer is in use.

Note. — As an example the United Kingdom Post Office carried out the following tests:

a) A magnetic field of 0.01 oersted (alternating field at 300 Hz) is produced by means of a square coil of dimensions as follows:

— length of side, 102 cm

— cross section not greater than 6.4 cm^2 consisting of n turns and carrying a current I amperes such that $nI = 0.84$. The psophometer under test is placed at the centre of this coil and its sensitivity is adjusted to that value for which an applied voltage of 0.2 millivolt would give the greatest deflection on the measuring instrument. Under these conditions the magnetic field should not produce a deflection greater than 0.04 millivolt.

b) The magnetic field is then made 0.05 oersted corresponding to $nI = 4.2$. Under these conditions for any sensitivity of the psophometer other than that mentioned under a the needle of the measuring instrument should not reach full scale.

9. *Adjustment.* — When the amplifier is not sufficiently stable an appropriate adjustment should be provided so as to maintain the amplifier gain at the desired value with an error less than $\pm 5\%$.

10. *Construction.* — No inconvenience should be experienced in practice due to the effect of mechanical vibration.

The characteristics of the psophometer should be as stable as possible under practical conditions of use—i.e. in spite of transport, temperature variation, etc.

The apparatus should be transportable and its weight reduced as much as the above conditions permit.

B. Psophometer used on circuits for sound-programme transmission

The general form as well as the principal characteristics given above for the psophometer used on commercial telephone circuits are applicable also for the psophometer used on sound-programme circuits except for the reference frequency and the data on the filter network which in this case should have a characteristic curve as in Table 2 below.

TABLE 2

SPECIFICATION OF THE CHARACTERISTIC CURVE FOR THE FILTER NETWORK OF THE PSOPHOMETER USED ON A SOUND-PROGRAMME CIRCUIT

(See curve in Figure 3/P.53)

Frequency Hz	Weighting relative to 1000 Hz	
	Nominal	Tolerance
	dB	dB
20 and below	⩽ −40	—
50	−34.3	±1.5
60	−32.2	”
100	−26.1	”
200	−17.3	”
400	− 8.8	”
800	− 1.9	”
1 000	0	0.0
2 000	+ 5.3	±1.5
4 000	+ 8.2	”
5 000	+ 8.4	”
6 000	+ 8.2	”
7 000	+ 7.3	”
8 000	+ 5.1	”
9 000	− 0.3	±3.0
10 000	− 9.7	”
13 000	⩽ −30	—
20 000 and above	⩽ −35	—

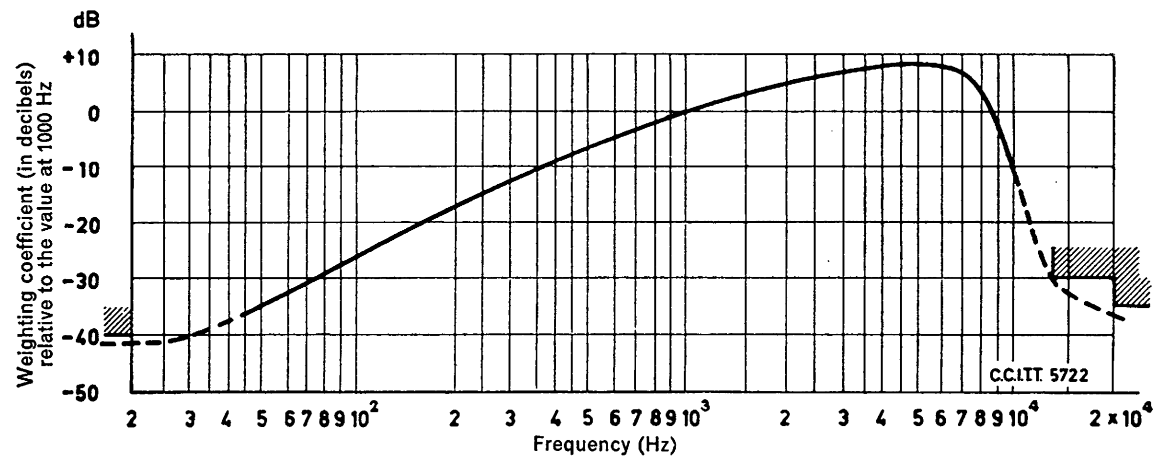


FIGURE 3/P.53. — **Characteristic curve of the filter network for the psophometer used for measurements on sound-programme circuits**

Recommendation P.54 (amended at Mar del Plata, 1968, and Geneva, 1972)

SOUND LEVEL METERS
(APPARATUS FOR THE OBJECTIVE MEASUREMENT OF ROOM NOISE)

The C.C.I.T.T. recommends the adoption of the sound level meter specified in I.E.C. Publication 179 in conjunction, for most uses, with the octave, half, and third octave filters in accordance with I.E.C. Publication 225.

Recommendation P.55 (Mar del Plata, 1968)[1]

APPARATUS FOR THE MEASUREMENT OF IMPULSIVE NOISE

Experiments have shown that clicks or other impulsive noises which occur in telephone calls come from a number of sources, such as faulty construction of the switching equipment, defective earthing at exchanges and electromagnetic couplings in exchanges or on the line.

There is no practical way of assessing the disturbing effect of isolated pulses on telephone calls. A rapid succession of clicks is annoying chiefly at the start of a call. It is probable that these series of clicks affect data transmission more than they do the telephone call and that connections capable of transmitting data, according to the noise standards now under study, will also be satisfactory for speech transmission.

In view of these considerations, the C.C.I.T.T. recommends that Administrations use the impulse noise counter defined in Recommendation H.13 (Volume III of the *Green Book*) for measuring the occurrence of series of pulses on circuits for both speech and data transmission.

Note. — At the national level, Administrations might continue to study whether the use of this impulse noise counter is sufficient to ensure that the conditions necessary to ensure good quality in telephone connections are met. In those studies, Administrations may use whatever measuring apparatus they consider most suitable—for example a psophometer with an increased overload factor—but the C.C.I.T.T. does not envisage recommending the use of such an instrument.

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[1] Former Recommendation P.55 (*Red Book*, Volume V, page 134) has been deleted.

SECTION 6

Objective electro-acoustical measurements

Recommendation P.61

MEASUREMENT OF THE ABSOLUTE SENSITIVITY OF A SENDING OR RECEIVING SYSTEM

For such measurement, in general one of the following methods can be used:

a) *Thermophone method*

The principle and description of this method appear in the following articles:

H. D. Arnold & I. B. Crandall: *Physical Review*, Vol. 10 (1917), p. 22.
E. C. Wente: *Physical Review*, Vol. 19 (1922), p. 333.
S. Ballantine: *Journal of the Acoustical Society of America*, Vol. 3 (1932), p. 319.

Note. — This method was used in the C.C.I.T.T. Laboratory for the absolute calibration of the S.F.E.R.T. This method is described in Annex 5 of the Book of Annexes to Volume IV of the *Green Book* of the C.C.I.F., (Geneva, 1954).

b) *Rayleigh disc method*

The principle and description of this method appear in the following articles:

W. König: *Annalen der Physik*, Vol. 43 (1891), p. 43.
E. J. Barnes & W. West: *Journal of the Institution of Electrical Engineers*, Vol. 65 (1927), p. 871.
W. West: *Acoustical Engineering* (Pitman edition, London), Chapter XI (1932).
R. A. Scott: *Proceedings of the Royal Society A*, Vol. 183 (1945), p. 296.
W. West: *Proceedings of the Physical Society B*, Vol. 62 (1949), p. 437.

Application of this method at the C.C.I.T.T. Laboratory for the absolute calibration of the A.R.A.E.N. is described in Supplement No. 9 to the *White Book*, Volume V.

c) *Compensation method and electrostatic actuator method*

The principle and description of these methods appear in the following articles:

E. Gerlach: *Wiss. Veröff. Siemens-Konzern*, Vol. 3 (1923), p. 139.
M. Grützmacher & E. Meyer: *Elektrische Nachrichten Technik*, Vol. 4 (1927), p. 203.
S. Ballantine: *Journal of the Acoustical Society of America*, Vol. 3 (1932), p. 219.

d) *Reciprocity method for the calibration of condenser microphones*

The principle and description of this method appear in an article by R. K. Cook published in the *Journal of Research of the National Bureau of Standards* (Washington), Vol. 25, p. 489 (November 1940). Some complementary details are given in Volume V of the *Red Book.*

The physical basis of this method is given in the following books and articles:

Rayleigh: *The Theory of Sound.* Macmillan & Co., London (1896).
W. Voigt: *Lehrbuch der Kristallphysik.* B. G. Teubner, Leipzig (1910).
D. A. Keys: *Philosophical Magazine*, Vol. 46 (1923), p. 999.
S. Ballantine: *Proceedings of the Institute of Radio Engineers*, Vol. 17 (1929), p. 929.
L. J. Sivian: *Bell System Technical Journal*, Vol. 10 (1931), p. 96.

S. Ballantine: *Journal of the Acoustical Society of America*, Vol. 3 (1932), p. 319.
H. Osterberg & J. W. Cookson: *Review of Scientific Instruments*, Vol. 6 (1935), p. 347.
W. R. MacLean: *Journal of the Acoustical Society of America*, Vol. 12 (1940), p. 140.

Recommendation P.62

MEASUREMENTS ON SUBSCRIBERS' TELEPHONE EQUIPMENT

A. Measurement of the attenuation distortion of a telephone set

The curve of the variation of the absolute sensitivity of an item of telephone equipment (sending or receiving system) as a function of frequency does not supply complete information on the manner in which this equipment reproduces the human voice or music, although such a curve may often be called the frequency characteristic.

However, the curve of variation of the absolute sensitivity of telephone equipment as a function of frequency gives useful indications from the point of view of the transmission of speech. On the other hand, for the transmission of music, in the absence of a precise criterion of the quality of transmission (corresponding to articulation, or repetition rate, in commercial telephony) such curves should be sufficient to enable the quality of the terminal equipment used (microphone or loudspeakers) to be appreciated.

For tracing sensitivity-frequency characteristics several modern commercial instruments are available which fall into two categories:

1) Recording devices which trace the frequency characteristics of telephone equipment automatically.

2) Devices which employ a cathode-ray tube and which allow rapid determination of the frequency characteristics of equipment.

Information on the measurement methods used by various Administrations for the maintenance of telephone apparatus and factory acceptance testing is given in Recommendations P.81 and P.82 below.

B. Measurement of the non-linear distortion of a telephone set and of microphone noise

Whilst the non-linear distortion of telephone receivers is in general negligible, microphones (and particularly carbon microphones of the type generally used in commercial telephone equipment) show considerable non-linearity: the relationship between the variation of microphone resistance and the acoustic pressure on the diaphragm is not linear. This non-linearity becomes more important as the variation of resistance in relation to the total resistance of the microphone increases, i.e. when the microphone is more sensitive. Furthermore, there are two supplementary effects:

1. The microphone is insensitive to acoustic pressure lower than a certain value (threshold of excitation).

2. As a consequence of the mechanical inertia of the carbon granules (delay in establishing electrical contact between the granules), the various states of agitation of the carbon under the influence of acoustic waves are not the same for all frequencies (for example, slow beats between two sounds are in general enhanced in reproduction by a carbon microphone).

Microphone noise is directly related to non-linearity. When non-linear distortion is measured, harmonic distortion as well as the variation of sensitivity with amplitude can be measured. As an example of such measurements reference can be made to a contribution of the Federal Republic of Germany described in Annex 26, Part II of Volume V of the *Red Book*.

C. Objective measurement of the reference equivalent (sending and receiving) and of the sidetone reference equivalent

1. As far as the objective measurement of reference equivalent (sending and receiving) of subscribers' telephone equipment is concerned, attention may be drawn to the equipment, described in Annexes 27 to 29, Part II of Volume V of the *Red Book* and in Annex 9 (Part II of Volume V *bis* of the *Red Book*), used by the Administrations of France, the Federal Republic of Germany, Switzerland and Czechoslovakia.

2. As far as the objective measurement of the sidetone reference equivalent of subscribers' telephone equipment is concerned, no objective method is recommended, this whole question being studied by the C.C.I.T.T. (see Recommendation P.63).

Recommendation P.63

METHODS FOR EVALUATING TRANSMISSION QUALITY ON THE BASIS OF OBJECTIVE MEASUREMENTS

These measuring methods are being studied by the C.C.I.T.T.

Methods which have been used by the Swiss Administration and the U.S.S.R. Administration are described in Annexes 30 and 31, Part II of Volume V of the *Red Book*.

A new method for evaluating relative transmission performance ratings of complete connections by means of objective measurements, has been studied by the American Telephone and Telegraph Company. The basic measuring equipment is described in Annex 1 to Question 15/XII studied in 1968–1972 (*White Book*, Volume V):[1]

[1] See also Sullivan, J. L. "A laboratory system for measuring loudness loss of telephone connections": B.S.T.J., 50, No. 8, October 1971, pp. 2663–2739.

SECTION 7

Subjective voice-ear measurements

Recommendation P.71 (amended at Mar del Plata, 1968)

MEASUREMENT OF SPEECH VOLUME

Each volume meter should be used in accordance with the relevant specifications (see Recommendation P.52). When the normal speech power for voice-ear measurements is to be used, the information provided in Recommendation P.42, C, should be borne in mind.

Recommendation P.72

MEASUREMENT OF REFERENCE EQUIVALENTS AND RELATIVE EQUIVALENTS

A. Measurement of true reference equivalents

This measurement consists of a comparison by voice and ear with the new master system for the determination of reference equivalents (N.O.S.F.E.R.); such a measurement is called a telephonometric measurement.

This comparison may be direct, and in that case gives the reference equivalent of the complete system, or of the sending system, or of the receiving system considered. But generally, only working standards are compared directly with the N.O.S.F.E.R., before they are put in service, and then from time to time afterwards for checking (see Recommendation P.42, E). Consequently the reference equivalent of a system or part of a system is usually determined indirectly—that is to say, the reference equivalent of the system (or part of the system) is determined by means of an auxiliary system (working standard system) whose own reference equivalent has been previously determined by direct comparison with the master reference system.

B. Measurement of relative equivalents[1]

The working standard systems used at present being either of the carbon microphone type (S.E.T.A.B.) or of the electrodynamic microphone and receiver type (S.E.T.E.D.), the special precautions to be taken when making a telephonometric measurement are given below, especially in the measurement of the relative equivalent of a handset type equipment. Two methods of measurement are given as examples:

a) Use of a working standard system of the S.E.T.A.B. type

The telephonometric measurement to be made for determining the relative equivalent of a system or part of a system by comparison with a working standard having a carbon microphone (S.E.T.A.B.) can be made in one of the two following methods:

[1] This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the C.C.I.T.T. Laboratory by using reference systems are described in Section 4 of this book.

*a.*1 *Method termed "two-operator with hidden-loss method"*

The method is based on the simultaneous use of two adjustable attenuators; one of these (balancing attenuator) serves the purpose of equalizing the sound intensities at the receiving end; the second attenuator (hidden-loss attenuator) can be adjusted arbitrarily, before the test and unknown to the listening operator, in order to modify the apparent sensitivity of one of the instruments compared.

The results must be expressed as: x dB "better" (M) or "worse" (P) than the N.O.S.F.E.R. taking account of the reference equivalent of the S.E.T.A.B.

The particulars given below refer to setting-up details, and are given only as examples.

*a.*1.1 *Comparison of a sending system with a standard sending system*

The schematic diagram together with the necessary switching arrangements for this comparison are shown in Figure 1/P.72.

To carry out an elementary balance a first operator A adjusts the hidden-loss attenuator to a certain value; he then talks alternately into microphones 1 and 2 repeating successively into each, one of the following conventional phrases, chosen so as to contain each of the principal vowel sounds:

Berlin, Hamburg, München, Koblenz, Leipzig, Dortmund (used in Germany).
One, two, three, four, five (used in Great Britain).
Joe took father's shoe bench out
She was waiting at my lawn } (used in the United States of America).
Paris, Bordeaux, Le Mans, Saint-Leu, Léon, Loudon (used in France and in the C.C.I.T.T. Laboratory).

He maintains, when talking, the normal volume for telephonometric measurements defined above in "Transmission standards", Recommendation P.42, section C, and places his lips so that they are approximately tangential to the plane of the circle which bounds the guard-ring.[1] At the same time he operates the switch in appropriate manner for controlling the switching system.

A second operator B receives, in a single receiver, the signals from the two microphones compared. He compares them by ear and adjusts the balancing attenuator so as to obtain the same sound intensity.

To enable the listening operator to follow the respective positions of the key, it is advisable to use a lamp the lighting circuit of which is controlled synchronously by the key. When glowing, it indicates that the balancing attenuation is inserted in the listening circuit. When the balance is thus obtained, the test is completed, and it is sufficient to record the readings of the two attenuators, and to interpret them according to the example given below.

*a.*1.2 *Comparison of a receiving system with a standard receiving system*

The schematic diagram together with the switching arrangements necessary for this comparison are shown in Figure 2/P.72.

To make an elementary balance a first operator A adjusts the hidden-loss attenuator to a certain value, then talks into the standard microphone (always the same one) repeating the same conventional phrase at regular intervals and with normal volume for telephonometric measurements (see above). He operates the key synchronously in order to obtain the appropriate circuit connections.

A second operator B holds the two receivers in one hand, and places them alternately to his ear (in the position giving the best reception) in step with the switching of the key. He then adjusts the balancing attenuator so as to obtain equality of sound from the two receivers. If the operator B cannot obtain equality of sound, i.e. when the system compared is more sensitive than the standard system, he asks operator A

[1] The position of the guard-ring is defined in section C of the present Recommendation.

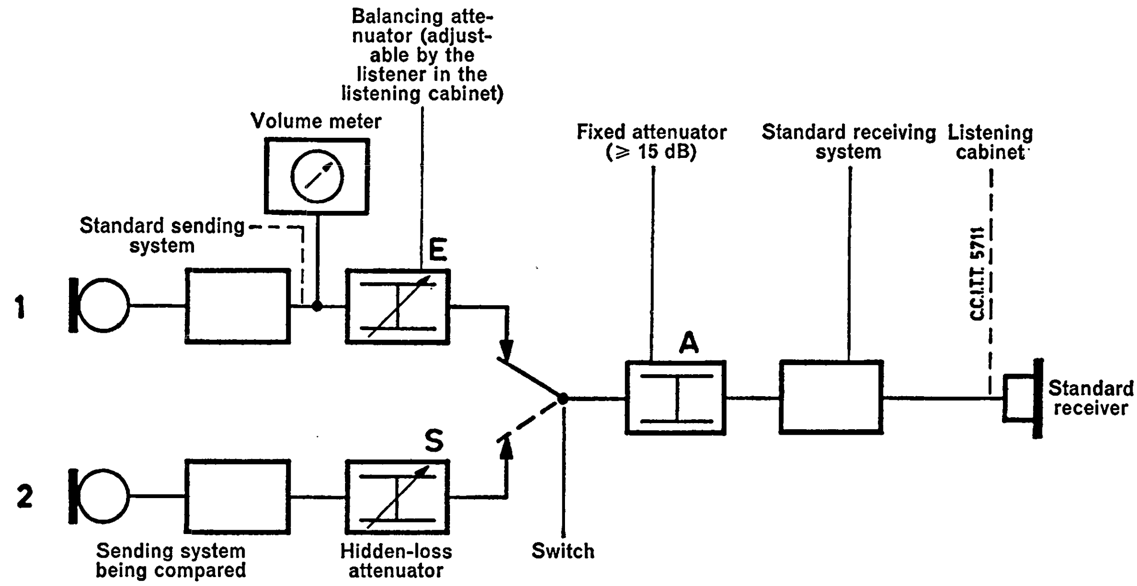


FIGURE 1/P.72. — **Comparison of a given sending system with a standard sending system (method termed "Two-operator, hidden-loss method")**

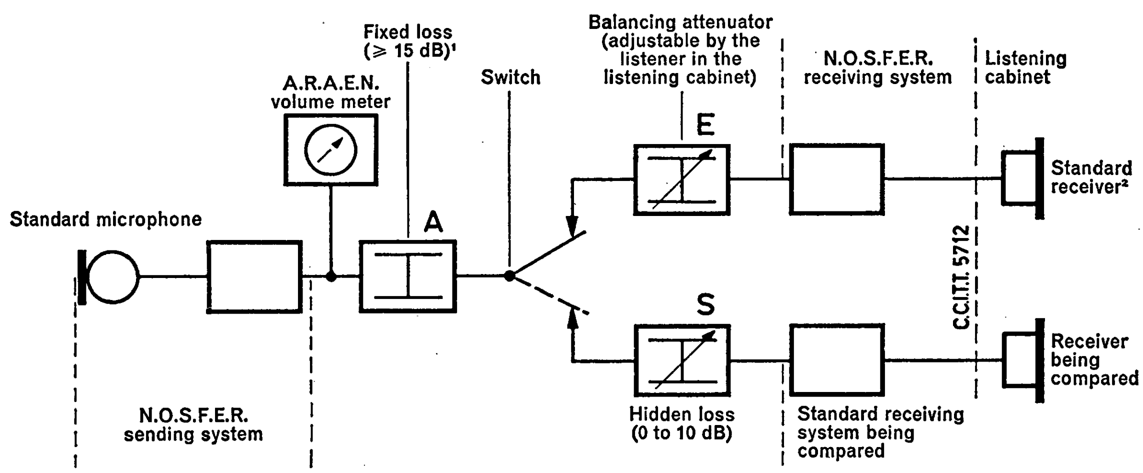


FIGURE 2/P.72. — **Comparison of a given receiving system with a standard receiving system (method termed "Two-operator hidden-loss method")**

(by means of some type of signalling system, as, for instance, a suitable audible signal) to change the respective settings of the hidden-loss and balancing attenuators.

A lamp, the circuit of which is controlled synchronously by the key, indicates to operator B that the balancing line is inserted in the listening circuit; it thus gives him information regarding the position of the switch at any instant.

The reference equivalent (or relative equivalent) cannot be obtained by only one test. It is obtained from the mean of a sufficiently large number of elementary balances made according to the method described above. The minimum number of tests is six, and twelve should normally be made. When three operators are available, they can be grouped in six different ways, and it will then be necessary to make only one test, or preferably two, for each possible combination of operators.

It is recommended that the test results be recorded on special forms; entries being made of the values of the hidden-loss and balancing attenuation used during each elementary test, together with the mean values which indicate the final results of the telephonometric measurements. The table below gives an example of the recording of a telephonometric measurement conducted at the Laboratory with a crew of five.

System (type of telephone system tested) *Date:*..................................

Operators			
1	....	4	....
2	....	5	....
3	....		

Reference equivalent (or relative equivalent) for sending (or receiving)

Measuring conditions (details of feeding bridge, with or without subscriber's line, voltage of feeding supply and value of microphone current)

Test No. ....

Listeners

Talker	1			2			3			4			5			Total	Talker mean
	s	eq	r	s	eq	r	s	eq	r	s	eq	r	s	eq	r		
1				8	12	+4	9	6	−3	5	7	+2	5	7	+2	+5	+1.2
2	10	11	+1				6	10	+4	10	8	−2	7	11	+4	+7	+1.7
3	4	9	+5	4	9	+5				6	6	0	2	4	+2	+12	+3.0
4	8	16	+8	9	15	+6	9	7	−2				10	12	+2	+14	+3.5
5	6	13	+7	3	7	+4	9	7	−2	9	11	+2				+11	+2.7
Total			+21			+19			−3			+2			+10	49	
Listener mean	+5.2			+4.7			−0.7			+0.5			+2.5				

Reference equivalent +2.45 dB (or 2.45 dB worse)
Standard deviation of the mean: ....

Symbols
- s denotes the hidden loss
- eq denotes the setting of the balance attenuator
- r denotes the result of the comparison (eq-s)

When it is desired to determine the reference equivalent of a sending (or receiving) system by means of a comparison measurement with a sending (or receiving) working standard system (whose reference equivalent has been determined at the C.C.I.T.T. Laboratory), it is necessary to take account of the value of reference equivalent of this sending (or receiving) standard system. The reference equivalent of a sending (or receiving) system is then determined from the test results in the following manner, e.g.:

Uncorrected mean result	−5.0 (5 dB better)
Reference equivalent of the working standard system .	+1.3 (1.3 dB worse)
Reference equivalent of the system under test	(—5.0) + (+1.3) = —3.7 dB or (3.7 dB better)

a.2 Method termed "Three-operator without hidden-loss method"

This method requires positions for three operators:

a) Sending position;

b) Receiving position (where the telephonometric comparisons are made);

c) Balancing position.

The sending and receiving positions are identical with those already described, the only difference between the two methods being in the number and positions of the attenuators. The comparison method employing three operators requires, in effect, only one adjustable attenuator in addition to the fixed attenuator. This is adjusted by operator C, who occupies the balancing position and receives signals from operator B at the receiving end. The hidden-loss attenuator is replaced by direct metallic connections.

The method of operations is as follows:

a.2.1 Comparison of a sending system with a standard sending system (Figure 3/P.72)

Operator C adjusts the balancing attenuator to a preliminary value a_1, he then signals by lamp, by buzzer, or orally to operator A that he may begin talking. The latter repeats into the two microphones alternately the conventional phrase adopted once for all, maintaining the normal volume for telephonometric measurement defined above in Recommendation P.42, section C. Operator B receives, in a standard receiver, the signals produced successively by the two microphones. A luminous indicator, controlled by the general switching system, indicates to him the microphone being spoken into at any instant (No. 1 or No. 2). If the sound intensity corresponding to microphone 2 is less than the sound intensity corresponding to microphone 1 (standard), B presses the signalling button marked P (worse). A luminous signal (lighting of a lamp on the cap of which is marked the letter P), together with, if necessary, a buzzer signal, indicates to operator C the first decision. A signal of the same type is also used to inform operator A that he may stop talking. Operator C records immediately the test result in a table in the form a_1 P.

The number a_1 can be entered in either of two columns. In the first, it indicates that the attenuation was introduced into the circuit at the same time as the standard, with the effect of attenuating the standard; inserted in the second column, it indicates that the attenuation was introduced into the circuit at the same time as the test apparatus, with the effect of increasing the attenuation of the latter.

In the opposite case, if the sound intensity corresponding to microphone 2 is greater than the sound intensity corresponding to microphone 1 (standard), operator B presses the signalling button marked M (better). A luminous signal (lighting of a lamp on the cap of which is marked the letter M), accompanied by a buzzer signal if necessary, then appears in front of operator C. If the test result corresponds to an exact balance, operator B presses a third button controlling the circuit of a third lamp, which is used for signalling exact balance.

The balancing operator C then sets the balancing attenuator at a second value a_2. He then signals to operator A that he may resume talking. The result of this measurement will be a second decision, for instance M, signifying that the microphone compared appears to be better than the standard, when the

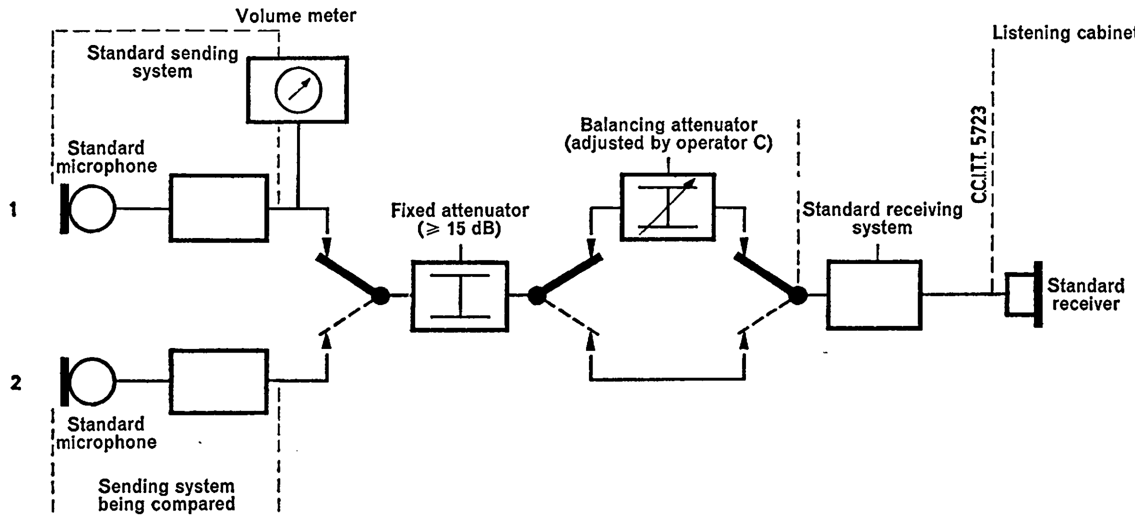


FIGURE 3/P.72. — **Comparison of a sending system with a standard sending system (three-operator without hidden-loss method)**

Test of sending system ....

Standard sending system used for comparison No. ....

A-B (Talker) (Listener)			B-C			C-A		
Attenuation			Attenuation			Attenuation		
Standard side	Instrument side		Standard side	Instrument side		Standard side	Instrument side	
6		M	1		P	1		M
0		P	5		M		3	P
3		M	3		M		1	P
1		P	1		P	1		M
2		M	2		M		0	P
						1		M
Mean 1.5 P			Mean 1.5 P			Mean 0.5 P		
B-A			C-B			A-C		
Attenuation			Attenuation			Attenuation		
Standard side	Instrument side		Standard side	Instrument side		Standard side	Instrument side	
0		P		2	P	4		M
2		M	3		M		2	P
1		P	0		M	2		M
2		M		1	P	0		M
				0	M		1	P
Mean 1.5 P			Mean 0.5 M			Mean 0.5 M		

Uncorrected mean result . . . . 0.7 P

Reference equivalent of standard . . . . . 5.0 P

Reference equivalent of the instrument tested 5.7 P or + 5.7

latter is in series with an attenuation of a_2 dB; operator C records the corresponding information in the form a_2 M.

He then adjusts the attenuation, at his discretion, to new values in order to diminish the interval between the two values for which the balancing result changes its sign. When successive intervals (forming a convergent series) have determined, if not the number corresponding to an exact equality of the sound impressions, at least two values a and a' differing at the most by 1 or 2 decibels, and for which one of the two instruments appears better or worse than the other, the test is considered as finished. Operator C at the control position signals the end of the test to the other two operators A and B and a new balance can then begin.

A single determination of equality cannot be considered sufficient to denote balance, and must be confirmed by at least two decisions (M and P) enclosing it.

In order to facilitate scrutiny of the results, it is convenient to arrange the individual test results in such a way that they show clearly the position of the balance attenuator on the one hand (standard or test side) and on the other hand the corresponding decision given by the listener.

The table above is an example of such an arrangement. The uncorrected result of the balance is either the number corresponding to the exact balance of the telephonometric estimations (when the exact balance has been obtainable, and confirmed, by enclosing values), or the mean of the two most adjacent numbers, one with the letter M (better) and the other with P (worse). The mean is then recorded, followed by the letter P or M according to whether the larger of the two numbers on either side of it is placed in the column marked "standard" or "instrument".

The uncorrected test result for a series of six balances is the mean of the results of the six elementary balances. The net result of the telephonometric measurement or series of six balances is equal to the uncorrected result corrected for the reference equivalent of the standard. The final result, instead of being followed by the letter M or P, can be prefixed by the sign — or +.

α.2.2 Comparison of a receiving system with a standard receiving system

The operating method is similar to that for comparing two sending systems; the only difference is, naturally, in the switching arrangement, which changes the receiving system instead of the sending system. For the general arrangement of the results the same instructions should be followed.

β) *Use of the S.E.T.E.D. type working standard*

The S.E.T.E.D. can be used for measuring the reference equivalent of any sending (or receiving) system, particularly of systems normally employed in telephone service.

The method of comparison employed can be either of the two methods previously described.

Note. — In the past, the C.C.I.T.T. recommended use of working standards either with a carbon microphone (S.E.T.A.C.) or with an electromagnetic microphone (S.E.T.E.M.). Administrations which still use these working standards will find information concerning them in Volume IV of the *Yellow Book* (Paris, 1949), pp. 254 to 266.

C. Precautions to be taken during telephonometric measurements

Volume to be maintained. — The speech volume produced during telephonometric measurements is of great importance in the conduct of such measurements as it influences the absolute and relative sensitivities of the equipment (especially in the case of carbon microphones). This volume must correspond to the normal power for telephonometric measurements employed in the C.C.I.T.T. Laboratory and determined as shown above (see Recommendation P.42, C).

It is necessary to adjust this volume by means of a volume indicator whose needle is in view of the talker and which is connected at the input of the fixed junction attenuator (which has an input impedance of 600 ohms). This volume indicator must have been compared with the S.F.E.R.T. Volume Indicator, at the same time as its associated working standard (or with another volume indicator of the same type having itself already been compared with the S.F.E.R.T. Volume Indicator).

Packing effect. — To prevent packing of carbon microphones under test, it is recommended that the microphone case be tapped lightly before each test.

Contact resistance. — In order to reduce to a minimum the effect of contact resistances, it is recommended that good quality spring blades be used, exerting sufficient contact pressure.

The contact points must be made of a suitable metal, for example, silver and gold, or platinum, several springs being in parallel to provide a single connection when the contact points are made of silver and gold.

It is, moreover, necessary to check frequently the electrical contacts of the plugs and of the switching system, by measuring the transmission equivalent of the electrical part of the system at a given frequency, for instance, 1000 Hz and with a very small current.

Position of the lips with respect to the microphone. — Not only is it necessary to use the normal volume for telephonometric measurements but it is also essential that the position of the lips with respect to the microphone should be rigorously defined. In the case of a fixed microphone the operator when speaking must place his lips so that they are approximately tangential to the plane of the external opening of the microphone, and maintain this position throughout the test. To this end, a device termed a guard-ring consisting of a circular ring of 2.5 cm diameter may be fitted to the microphone mouthpiece by means of a light attachment, and fixed so that the plane of the microphone opening is tangential to the plane of the lips when the operator applies his lips to the ring while talking. In any case, the front of the microphone must be inclined backwards, making an angle of 20° with the vertical.

In the case of a handset telephone, a guard-ring conforming to the details below must always be used.

In the first place, from measurements made on the heads of a large number of individuals, the characteristic head dimensions of an average subscriber have been determined together with the position in which he holds the handset to his ear during a telephone conversation. Such measurements have been made in various countries by means of an instrument referred to as a "Device for measuring the dimensions of the head".

This device is shown in Figure 4/P. 72. It consists of a telephone receiver to which is applied a complex voice frequency tone and to which is fixed a system of graduated scales. The device is held in the plane passing through the centres of the ears and of the mouth, the individual placing the receiver to his ear as he would normally do. The distance d_1 between the centre of the ear and the line of the lips and the distance d_2 of the displacement of the centre of the mouth are read on the scales. By means of the abac (Figure 5/P. 72) the following data are deduced:

1. The distance δ between the centre of the ear and the centre of the mouth;
2. The angle α between the plane of the earpiece of the telephone receiver and the straight line from the centre of this earpiece to the centre of the mouth.

The distance l between the mid-points of two telephone receiver ear-caps placed one against each ear is also measured (distance between the centres of the ears). The angle β is computed; the intersection of the plane of the telephone ear-cap placed against the ear and the plane through the centres of the ears and the centre of mouth defines one straight line; β is the angle between this line and the "direction of speech". The "direction of speech" is the straight line formed by the intersection of the median plane of the head with a plane through the centres of the ears and the centre of the mouth.

The value of β is obtained from the formula:

$$\beta = \arcsin \frac{l}{2\,\delta} - \alpha$$

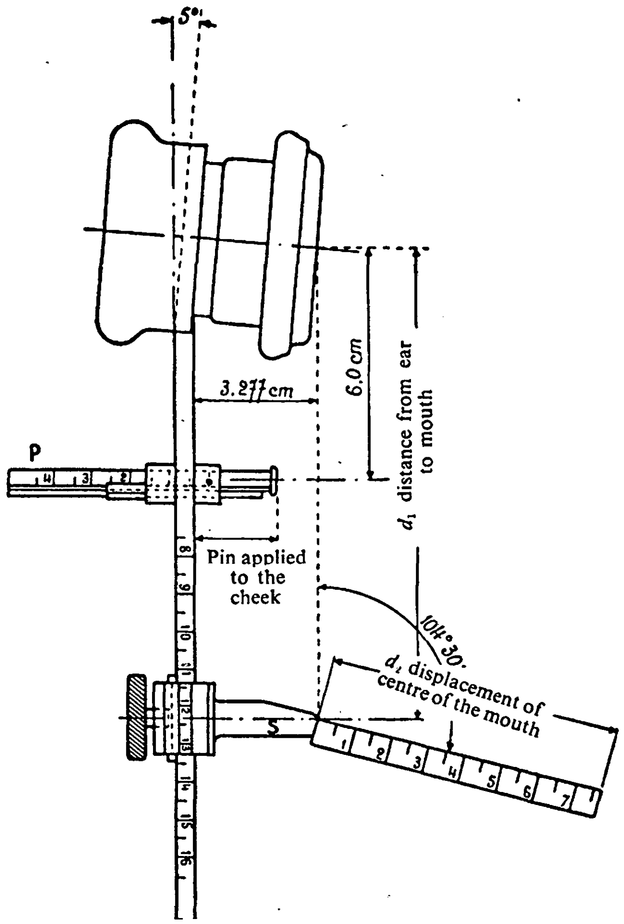


FIGURE 4/P.72. — **Device for measuring the dimensions of the head**

The C.C.I.T.T. recommends the following values for α, β and δ in the case of reference equivalent measurements:

$$\alpha = 15°30'$$
$$\beta = 18°$$
$$\delta = 14 \text{ cm}$$

These figures are the most probable values observed in the United States. Although other measurements of the dimensions of heads of subscribers have given slightly different values, it is desirable to keep the above values for the sake of world-wide standardization and also because, on the basis of these values, much information concerning the reference equivalents of commercial telephone instruments has already been determined.

Using the above values of α, β and δ, it is possible to determine the position of a guard-ring to fix the position of the mouth of the operator who is talking into a handset. The plane of the guard-ring will be at right angles to the plane of symmetry of the instrument and its centre will be located in that plane.

Its position will be defined by the following geometrical construction in the plane of symmetry of the handset. The mid-point of the earcap of the receiver is taken as the origin. From this origin a straight line is drawn making an angle α with the intersection of the plane of the earpiece of the receiver and the plane of symmetry of the handset and a distance δ is marked off along this line. The point thus determined is the centre of the guard-ring, which should coincide with the mid-point of the lips.

The intersection of the plane of this ring with the plane of symmetry will be a straight line, perpendicular to the direction of speech above defined, i.e. the perpendicular to this straight line will make an angle β with intersection of the plane of the receiver.

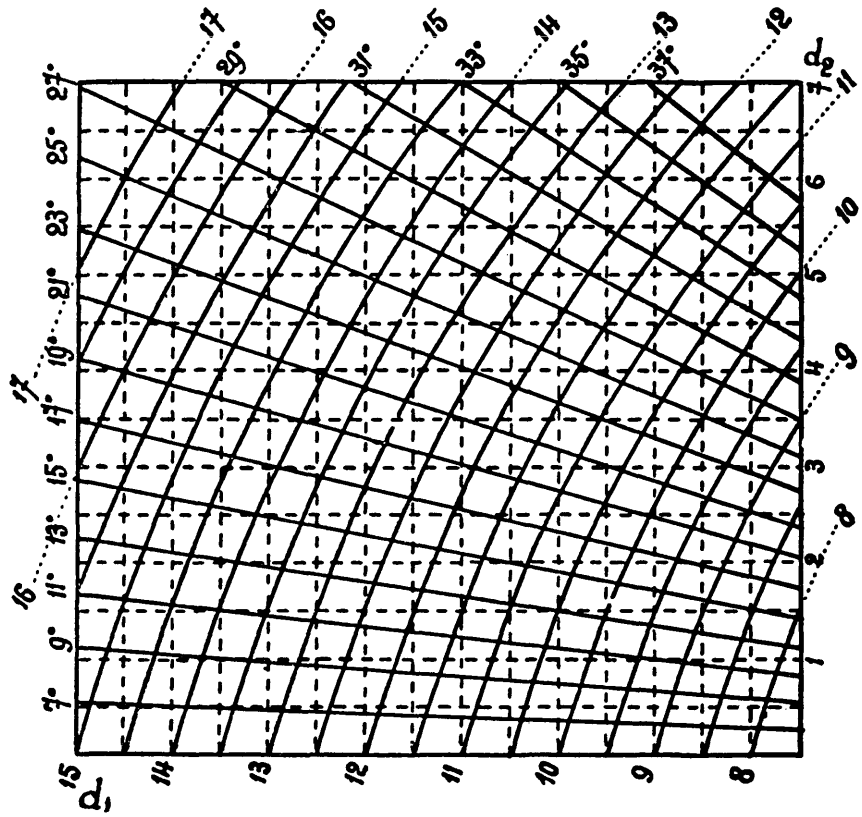


FIGURE 5/P.72. — **Abac used with the device for measuring the dimensions of the head**

d_1 Distance between the centre of the ear and the line of the lips (cm)
d_2 Displacement of the centre of the mouth (cm)
15-15, 14-14, etc. Distance δ in cm
7°, 9°, etc. Angle α in degrees

The position of the guard-ring is thus completely determined and fixed with respect to the instrument.

It then remains to determine the position of the guard-ring in space during telephonometric measurements. It is assumed that the operator talks in such a manner that the median plane of his face is vertical. The centre of the ring will be in that plane and the plane of the ring will be perpendicular to it.

It remains to determine the inclination of the ring with respect to the horizontal plane. This is taken at 45°, which corresponds to a normal posture during conversation, the head being inclined forward slightly.

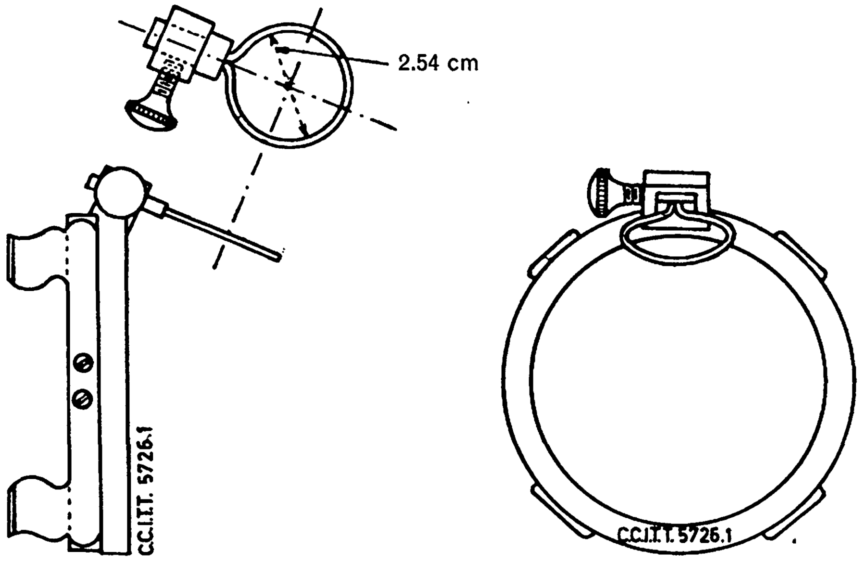
It should be noted that the position of the guard-ring, thus defined, has been fixed without reference to the inclination of the diaphragm of the microphone and does not necessarily correspond to the best operating conditions of the latter.

If, when the handset is in the position described above, the receiver is near the operator's ear, care must be taken to ensure that the volume remains constant. In fact, with the volume meter connected to the standard, when the operator speaks into the handset he is inclined to vary his speech intensity on account of sound heard in the receiver by sidetone. This inconvenience is most likely to occur in instruments without an anti-sidetone circuit.

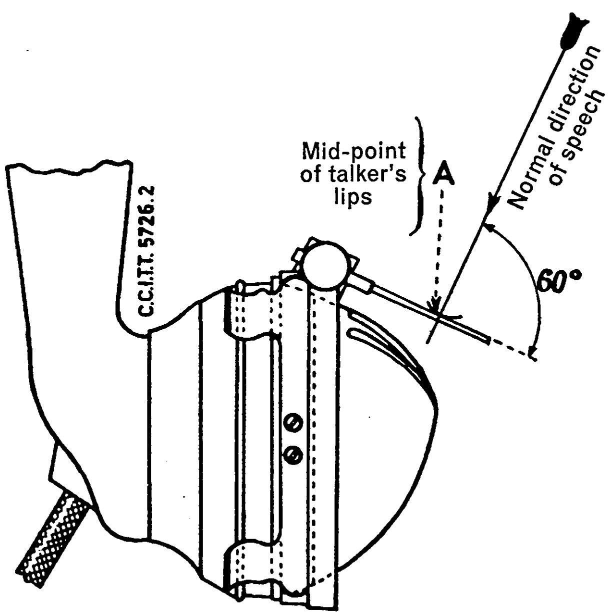
In order to avoid this trouble the receiver of the handset should be disconnected and is not to be applied to the operator's ear; in addition, in the test arrangement a similar receiver should be inserted in place of the disconnected receiver which should be placed face downwards on the table so as to present an impedance similar to that of the receiver held to the ear.

It is essential that the guard-ring and its mounting should be of light construction in order not to cause any disturbance in the acoustic field in front of the microphone. It is equally important that the strain on the microphone case should not affect the mechanical and electrical properties of the microphone.

A device similar to that shown in Figure 6/P.72 and Figure 7/P.72 is recommended.

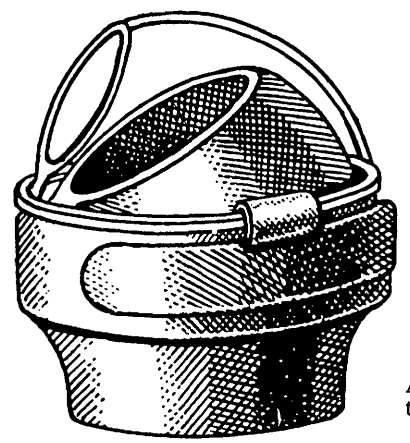


1. Example of guard-ring for tests of handsets

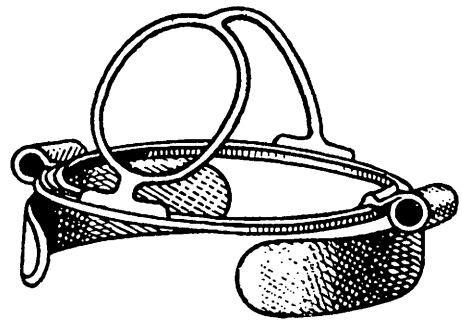


2. Attachment of guard-ring to a handset

FIGURE 6/P.72



Attachment of guard-ring
to a handset



Perspective sketch
of the guard-ring

FIGURE 7/P.72. — **Guard-ring used by the American Telephone and Telegraph Company for tests of handsets**

ANNEX

(to Recommendation P.72)

Remark on measurements of reference equivalent

It is necessary to draw a very clear distinction between, on the one hand, measurements required in the design and development of commercial telephone equipment to satisfy service conditions as well as possible and, on the other hand, the exchange between Administrations of numerical data which enable different types of equipment to be compared, so far as from the standpoint of reference equivalent considered as one of the factors which affect transmission quality.

In the first case it is necessary to measure the sending and receiving sensitivities of the equipment over a wide range of variation of either the position of the subscriber's mouth with respect to the microphone or of the volume used or even of the feeding current value.

In the second case it is sufficient to give for each item a value of sending and receiving reference equivalent corresponding to a conventional position of the mouth with respect to the microphone and at a conventional volume measured with a specified volume meter.

The C.C.I.T.T. considers only the second case and for this reason it is not absolutely essential that the conventional position adopted for the mouth should correspond exactly with the mean position of the subscriber's mouth nor that the normal volume for telephonometric tests should coincide exactly with the mean value of volumes found in service.

On the other hand, it is a great advantage if this conventional mouth position and this normal volume for telephonometric tests is used universally when it is simply a matter of communicating from one country to another general information on reference equivalents.

It follows from this that the values of sending and receiving reference equivalents corresponding to this conventional mouth position and normal volume for telephonometric tests are not necessarily the same as those that would be obtained for the same items when in actual service.

From these considerations the above conventions can be admitted so far as the mouth position and the normal volume for telephonometric tests are concerned, although the results of measurements of the head dimensions in Europe have given appreciably different mean values from those which appear above, particularly for the angles α and β. These values do, however, fall within the range of variation in service of the measured values. (Actually, the statistical mean values found in Europe as a result of several determinations conducted in various countries and which have been adopted for A.E.N. determinations in the C.C.I.T.T. Laboratory are:

$$\alpha = 22^\circ \qquad \beta = 12^\circ 54' \qquad \delta = 13.6 \text{ cm}$$

while the values retained for reference equivalent measurements are:

$$\alpha = 15^\circ 30' \qquad \beta = 18^\circ \qquad \delta = 14 \text{ cm.})$$

Recommendation P.73

MEASUREMENT OF THE SIDETONE REFERENCE EQUIVALENT[1]

It is necessary to consider two kinds of sidetone: speech sidetone and room-noise sidetone.

The determination of speech sidetone reference equivalent must be made with speech or equivalent arrangements: the speech power to be used for these tests is the normal speech power for telephonometric measurements.

The determination of room-noise sidetone reference equivalent must be made with reference to subjective acoustic intensity for room noise.

Whenever a result of a sidetone reference equivalent measurement is quoted for a telephone set it is necessary also to state the value of the impedance to which it was connected during the measurement, the value of the feeding current and the sending and receiving reference equivalents of the telephone set.

[1] This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the C.C.I.T.T. Laboratory by using reference systems are described in Section 4 of this volume.

a) If it is a question of speech sidetone, a telephonometric measurement is made of the sidetone reference equivalent (voice and ear measurements), while speaking in a silence cabinet into the microphone of the set concerned, with the mouth at the normal speaking distance (see above) from the diaphragm of the microphone; the receiver of the set situated some distance away in another silence cabinet where the sound level heard in this receiver is compared with that in the receiver of the Master Reference System (or with that in the receiver of a working standard whose reference equivalent is known).

Equality of sounds heard is obtained by adjusting the balancing attenuator. A hidden-loss attenuator situated close to the talking position enables the apparent sensitivity value of the complete N.O.S.F.E.R. to be varied at will before the measurement and by an amount unknown to the listener. The value of sidetone reference equivalent of the telephone system is equal to the sum $S + Q$ of the values of the hidden-loss and balancing attenuators.

b) For measurement of the room-noise sidetone reference equivalent of a telephone set by aural comparison between the Master Reference System (or a calibrated working standard) and the sidetone path from microphone to receiver of the telephone set considered, one should, strictly, employ "normal room noise" produced by a loudspeaker situated at a specified distance from the microphone.

The noise source could consist, for example, of a gramophone pick-up reproducing from a disc on which typical room noises had been recorded. The C.C.I.T.T., having adopted a reference room noise for A.E.N. determinations (see Recommendation P.45), advises the use of such a noise.

The measurement technique used in the C.C.I.T.T. Laboratory is given in Figure 1, where the real voice is replaced by a noise source giving the reference room noise at the positions of the two microphones (1 and 2).

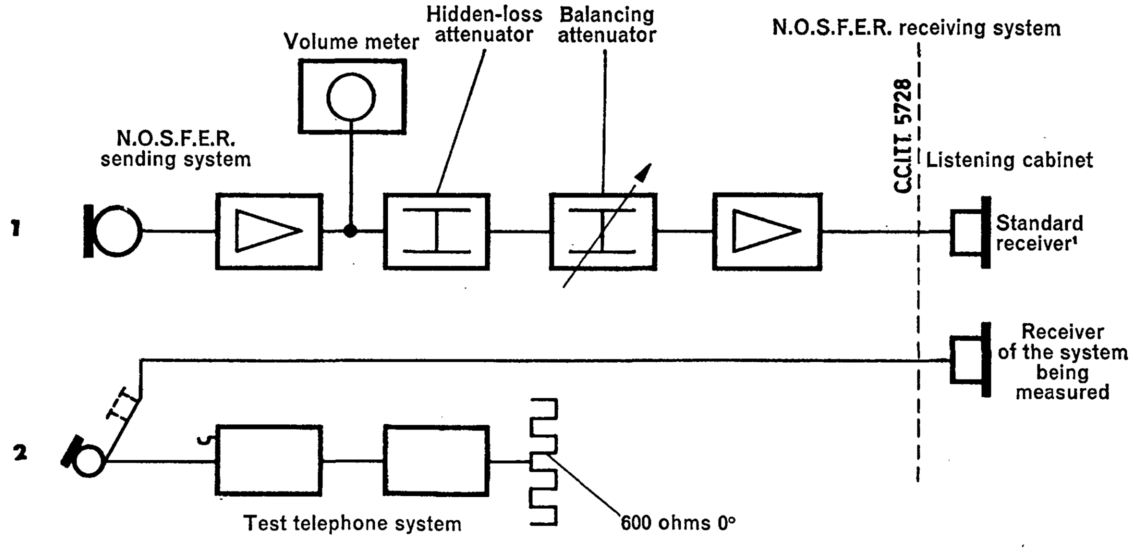


FIGURE 1/P.73. — **Measurement of the sidetone reference equivalent of a commercial telephone system**

[1] In N.O.S.F.E.R. there are four receivers in series. During an elementary balance, the other three receivers available are laid face downwards but remain connected.

The value of the reference equivalent of sidetone for room noise is equal to $S + Q - 17$ dB, where S is the hidden-loss, Q is the loss of the balancing attenuator, the correction of 17 dB takes account of the fact that, under these conditions of measurements, the N.O.S.F.E.R. is more efficient than the S.F.E.R.T. and it is with respect to the latter that sidetone reference equivalents have been defined.

Note. — The C.C.I.T.T. is at present studying the test conditions together with a measuring technique for determining the speech and room-noise sidetone reference equivalents.

Recommendation P.74

METHODS FOR SUBJECTIVE DETERMINATION OF TRANSMISSION QUALITY[1]

A. Repetition observation tests

One of the criteria used for assessing the quality of transmission in service is based on the observation of repetitions in the course of telephone conversations conducted under commercial service conditions.

No direct measurement of effective transmission losses exists having international acceptance.

So far as trunk telephone circuits are concerned, attention is confined to the individual measurement of various transmission impairments due respectively to circuit noises, distortions, etc., without even being certain that close agreement to the effective transmission can always be obtained by calculation, for example by adding the reference equivalent of the circuit (which is approximately equal to the loss at 800 or 1000 Hz) to the transmission impairments due to the circuit distortions (attenuation distortion, phase distortion, non-linear distortion) and the transmission impairments due to various noises (induced noise, repeater noise, crosstalk noise, etc.), these transmission impairments being defined as has been shown in the first part of this book and measured as indicated below.

To measure the transmission impairment due, for example, to a certain circuit noise present on a trunk telephone circuit by means of repetition rate, the following method is employed:

During a sufficiently long period (for example 50 000 to 100 000 seconds), the repetitions are noted of one or other of the correspondents conversing on the test circuit of constant reference equivalent q on which are introduced successively various levels of an artificial noise of the same characteristics as the circuit noise considered, but of adjustable level; the curve is drawn of repetition rate (number of repetitions per 100 seconds) as a function of the level of the artificial circuit noise.

On the other hand, the reference equivalent of the test circuit (which remains noise-free) is increased from the value q and the curve is drawn of the repetition rate as a function of the increase Δq in reference equivalent of the test circuit.

By comparing these curves, it is possible to determine the increase in reference equivalent of the test circuit which produces the same increase in repetition rate as the circuit noise of the specified level and characteristics which are considered: this increase in reference equivalent Δq is equal to the transmission impairment due to this circuit noise, expressed in decibels.

[1] This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the C.C.I.T.T. Laboratory by using reference systems are described in Section 4 of this volume.

The test circuit used for measurements by this method should reproduce the average conditions of a typical commercial trunk telephone call. Each Administration should set it up using the sending and receiving systems of its own working standard (with typical room noise) and connecting these sending and receiving systems together by means of a circuit (or better still an artificial line) of adjustable attenuation and similar in all respects to the trunk circuit considered (particularly from the point of view of the various distortions), except that this circuit (or artificial line) used for the measurements is noise-free (no circuit noise).

The method of repetition observations indicated above can be adapted for measurement of the transmission impairment due to a certain distortion (for example: limitation of the band of frequencies effectively transmitted, or attenuation distortion) of the trunk circuit considered, on condition that, instead of the circuit noise, the quantity is taken which characterizes the magnitude of the distortion considered (in the above example, the bandwidth of frequencies effectively transmitted by the trunk circuit).

In the case of such a measurement, the test circuit used comprises (in addition to the sending and receiving systems of the working standard together with the typical room noise) an artificial line with an artificial circuit noise similar to that of the trunk circuit, and of which all the other characteristics are equally similar to those of the trunk circuit, except that this artificial line does not present the distortion of the type considered.

B. Immediate appreciation tests

The method of immediate appreciation tests is described in Annex 32, Part II, of Volume V of the *Red Book.*

C. Other methods

For example, Annex 1.2 to Section 1 of the C.C.I.T.T. Handbook on *Transmission Planning of Switched Telephone Networks* describes a method used by the United Kingdom Post Office for subjectively determining telephone transmission quality.

Recommendation P.75 (Geneva, 1972)

STANDARD CONDITIONING METHOD FOR HANDSETS WITH CARBON MICROPHONES

1. The C.C.I.T.T. considers that, since the characteristics of carbon microphones are strongly dependent on conditioning techniques, it is necessary to follow a consistent procedure prior to measuring sensitivity/frequency characteristics in order to obtain reproducible results. The following steps are specified for the *standard conditioning method*:

a) Place the handset in a holding fixture with the handset clamped in a position corresponding to that in which the microphone is going to be measured.

b) Connect the microphone or telephone set terminals as required to the d.c. feed circuit and appropriate terminating load.

c) Turn the feed current on. After 5 seconds, condition the microphone by rotating it through an arc slowly and smoothly. The microphone face should reach a vertical plane during the initial part of the rotation. In this vertical plane, a reference vector should be visualized which passes through the centre of the microphone and points straight up. Rotation should then be continued until this reference vector points straight down (that is, a movement of the reference vector through 180°). The direction of rotation should then be reversed and the microphone returned to its starting position.

Without interrupting the d.c. current or jarring the microphone, repeat this process two more times. The speed of rotation is not critical, but should be slow enough to ensure that the effect of centrifugal force on the carbon granules is negligible. Finally, return the handset to the measuring position.

Note. — Depending on the axis of rotation in c), e.g. about a diameter or about the axis, the carbon granules can flow smoothly in the granule chamber in a number of ways. Any of these is allowable.

2. For any type of microphone which does not give repeatable results by the standard method, the following alternative method may be used. In this case, the artificial mouth is fed from alternate sources of measuring tone or noise of Hoth spectrum [1].

Following "standard conditioning" during the frequency characteristic measurement the normal tone stimulus is interrupted at approximately 1.5-second intervals by a short burst of noise at a sound level of the order of 98 dB (linear weighting of sound level meter). This level is referred to the same point as the tone used for measurement.

Note. — The timing of the noise bursts quoted here is based on the assumption that the total measurement time is about 60 seconds. Precise timing of the two signals is not thought to be essential.

3. For measurements other than sensitive/frequency characteristics, e.g. subjective and objective loudness ratings, it may not be possible to use the above methods. It is, however, desirable to simulate the movements of the standard method so far as possible, even for situations where the handset is held in the hand for subsequent measurements.

Reference

[1] C.C.I.T.T. *Reed Book*, Volume V, p. 76.

SECTION 8

Measurements for maintenance of subscribers' telephone equipment and for factory acceptance testing

Recommendation P.81 (modified in Geneva, 1964, and at Mar del Plata, 1968)

Maintenance of subscribers' equipment

To ensure good transmission on international connections, the C.C.I.T.T. recommends periodic testing of each subscriber's equipment.

Different procedures exist for making check tests from the exchange of subscribers' stations under working conditions by means of subjective or objective measurements.

The most important of these are the following:

A. *Subjective measurements.* — a) Quick conversation test; b) Complete telephonometric test.

B. *Objective measurements.* — It is possible to envisage maintenance also based on the procedures used for factory acceptance testing. (This form of maintenance does not involve the exchange.)

A. Subjective measurements

a) *Quick conversation test*

This method is used mainly in the United States (by the American Telephone and Telegraph Company) and in Switzerland. Furthermore, in the United Kingdom Post Office, the transmission quality of telephone equipment in public call-boxes and subscribers' stations having extension sets is assessed by means of a conversation with the exchange test desk clerk[1].

b) *Complete telephonometric test*

This method seems to be no longer used.

B. Objective measurements

a) *Measurements made from the test desk*

In Switzerland the checking of transmission quality is made subjectively by an exchange of conversation with the test desk which deals with fault control for each terminal trunk exchange. From this test desk, measurements and checks of subscribers' lines can also be made from the point of view of insulation, loop resistance, transmission of dialling impulses, etc.

b) *Electrical measurements of a general nature*

To check the transmission quality of subscribers' telephone equipment in service, the Dutch Administration uses the same measuring equipment and methods as for factory testing; nevertheless it must be

[1] The United Kingdom Post Office considers that the cost of applying preventive maintenance to telephone sets other than public call-boxes and private branch exchanges would not be justified.

understood that the permissible limits are somewhat larger. These measuring methods are described in Recommendation P.82.

c) *Use of special measuring equipment for checking telephone equipment*

Australia. The Australian Administration has developed a subscribers' instrument tester for use in the maintenance of subscribers' telephone instruments. The tester which is small and portable (measuring approximately 15 × 8 × 8 cm) is inserted between the subscribers' transmission line and the telephone instrument. This insertion is facilitated by the plug and socket connections of modern telephones. The tester enables the following parameters to be checked:

1) line current in mA (d.c.);
2) transmitting volume efficiency;
3) receiving volume efficiency;
4) subscribers' transmission line insertion loss (600 ohms termination).

The tester is excited by a warble tone oscillator, and a considerable reduction in size and weight of the portable tester has been achieved by locating this oscillator within the local exchange. The oscillator which is a solid-state device, generates a sinewave swept linearly with respect to time over the range 300—3000—300 Hz, 25 times per second. In the tester a moving coil microphone unit with suitable frequency characteristics is made to serve as either an artificial voice or an artificial ear, as required. A closed coupler is used for each measurement and the telephone microphone is subjected to a sound pressure of approximately 95 dB rel. 20 microbar during the transmitting efficiency test.

A number of types of subscribers' instruments are in use in Australia and have a variety of mouthpieces and receiver insets. Each of these will give slightly different performance figures, and to avoid the need for a multipoint switch to adjust the instrument calibration for each type, a common calibration adjustment is used and the figure of merit for the measurement is read off a meter scale calibrated in decibels and is compared against the appropriate limit value listed on a card attached to the tester. On the same meter a 0–100 scale is used to indicate the line current directly in milliamperes. The decibel scale is used also in measuring the subscriber's line loss.

The tester is calibrated *in situ* so that its readings approximate to those which the telephone would give under limiting line conditions which are the conditions under which new telephones are acceptance tested. Consideration was originally given to measuring the volume efficiency of the telephone plus its local line, on the basis that instruments which had fallen below their initial performance might still give adequate performance on shorter lines. This possibility was rejected because only the volume efficiency is measured and if this is very low other impairments may be present (e.g. the receiver diaphragm may be poling), and because telephone instruments possessing faults of a marginal nature may have unsatisfactory service life.

A prototype instrument received a very favourable reception in a field trial. Both the maintenance technicians and the subscribers expressed satisfaction that faults could now be more readily diagnosed and demonstrably remedied. Further field testing on a wider scale with about 20 testers distributed over a number of local exchange areas is being arranged.

United States of America. The American Telephone and Telegraph Company assesses subscriber sets with an electroacoustic rating system (E.A.R.S.)[1]. This system is used in the laboratory to determine re-

[1] See Annex 1 to Question 15/XII in Part II of this volume of the *White Book*, Volume V and the following article: SULLIVAN, J. L.: A Laboratory System for Measuring Loudness Loss of Telephone Connections; B.S.T.J. 50, No. 8, October 1971, pp. 2663–2739.

lative ratings of telephone set designs and local transmission plans which correlate with subjective loudness ratings. There are no present plans for using this system to evaluate subscribers' sets on an in-service basis for maintenance purposes.

Federal Republic of Germany. The Administration of the Federal Republic of Germany uses the following methods.

The testing of telephone equipment to check the transmission quality of subscribers' telephone equipment in service is applied mainly to the measurement of microphone and receiver capsules, because their transmission quality depends very much on the material used and the quality of manufacture. Specifications have been fixed for microphone and receiver capsules against which they are checked by means of the equipment for objective measurement of reference equivalents described in Annex 28, Part II of Volume V of the *Red Book*.

The equipment for the objective measurement of reference equivalents enables the reference equivalents of microphone and receiver capsules to be measured. For microphone capsules, non-linear distortion and microphone noise are measured at the same time as reference equivalent by means of the modulation products. Furthermore, it is possible to check the sensitivity-frequency characteristic by means of a visual display.

The microphone capsules are divided according to their sensitivity into groups in steps of 3.5 dB and the receiver capsules in steps of 2.5 dB. These groups correspond for microphone capsules to values of sending reference equivalent 8 to 4.5 dB, 4.5 to 1 dB and 1 to −2.5 dB and, for receiver capsules, to values of receiving reference equivalent 0 to −2.5 dB, −2.5 to −5 dB and −5 to −8 dB. This allocation into groups is then used to associate the capsules with corresponding groups of subscribers' lines (loop resistance 0 to 250 ohms, 250 to 500 ohms and 500 to 750 ohms (see Annex 2.3 of the C.C.I.T.T. Handbook: *Transmission Planning of Switched Telephone Networks*, under "Federal Republic of Germany").

For this allocation the capsules are stamped with the figures I, II, or III. Thus it is possible not only to compensate for too high values of reference equivalent of subscribers' lines but also, on replacement of capsules when the telephone set is repaired, to make sure that the capsules have not been changed after being put into service. For this reason the lineman who is dealing with the location of faults must always have with him some capsules of the various groups; capsules which are removed from subscribers' sets are checked at the headquarters stores depot by the equipment for the objective measurement of reference equivalents so as to determine whether they are still serviceable.

The measurement and grouping of microphone and receiver capsules with the aid of the equipment for the objective measurement of reference equivalents were introduced several years ago in the Federal German Posts and Telecommunications Administration. Each headquarters has at its telecommunications stores depot one such measuring equipment operated by non-specialist female staff. The measuring precision is so high that when the same capsule is measured with a different measuring equipment the differences are less than 1 dB. The grouping of capsules and their correct allocation to the telephone sets can, so far as present experience has shown, be done without difficulty. They are considered by the telephone service staff, particularly the officers on fault location duties, as a great step forward because they are able to ensure that, by means of this grouping, the variations of receiving loudness can be compensated for different lengths of subscriber's line. A large percentage of capsules in service (about one-third of the microphone capsules and one-sixth of the receiver capsules) had to be replaced, which resulted in a great improvement in transmission quality. It was noted that most of the microphone capsules in service did not correspond to the present conditions. This also applies for receiver capsules, but to a lesser extent.

Recommendation P.82 (modified in Geneva, 1964)

FACTORY ACCEPTANCE TESTING OF SUBSCRIBERS' EQUIPMENT

The methods used in various countries are described below for information.

Denmark

In addition to inspection and mechanical examination, the equipment is given the following transmission test:

The handset is placed in a support containing a sound source (artificial mouth) and a microphone (artificial ear).

With an 800-ohm generator connected to the terminals of the equipment, the acoustic pressure produced by the telephone receiver is measured and this appears on a cathode-ray oscillograph as a function of frequency over the frequency band 300–3400 Hz. In this way a simultaneous check is provided of the receiver capsule and the electrical receiving circuit.

A feeding bridge and a line impedance of 800 ohms are connected to the terminals of the equipment and the voltage at these terminals is measured while a constant acoustic pressure of 20 dynes per square centimetre, provided by the sound source, is applied to the microphone. The voltage obtained appears on a cathode-ray oscillograph as a function of frequency over the frequency band 300–3400 Hz. In this way a simultaneous check is provided of the microphone capsule and the electrical sending circuit.

The oscillograph is provided with a transparent scale on which are drawn the limit curves for sending and receiving, i.e. the mean curves ± 2 dB.

United States of America

In addition to measurements on the various component parts of the telephone equipment, the principal measurements made upon subscribers' telephone equipment in the factory by the American Telephone and Telegraph Company are the following:

1. Once the assembly of the handset is complete:

a) both the shape and the level of the "sensitivity-frequency" characteristics of the microphone and receiver are determined by means of a cathode-ray oscillograph on the screen of which curves corresponding to the tolerance limits are drawn;

b) the d.c. resistance of the carbon microphone is measured for which upper and lower limits have been established;

c) the impedance of the varistor used to protect the receiver is measured at 60 Hz and compared with an established upper limit.

2. When the telephone set is completely assembled:

a) the ringing is tested, a given input voltage being applied;

b) to check the circuit continuity rather than to detect faulty components, a howling sound is applied to the microphone by an acoustical path in order to excite it and the following measurements are made:

1) the output voltage across an artificial line representing the subscribers' line,

2) the acoustic pressure produced by the receiver and transmitted by the side-tone path;

c) a test of the isolation-to-ground of the telephone circuit is made using a break-down voltage of 500 volts d.c.

The tests and measurements described above are made on all sets and not on a sampling basis.

France

The French P.T.T. Administration has studied and prepared a set of apparatus for:

- the checking and maintenance of telephone sets on the subscribers' premises;
- bulk factory acceptance tests of consignments of subscribers' sets, conforming to an accepted type, submitted by manufacturers for the approval of the Administration;
- maintenance in regional exchanges.

Descriptions of these various types of apparatus are given in paragraphs II, III and IV of Annex 27, Part II of Volume V of the *Red Book*.

Netherlands

The Dutch Administration has put measuring equipment at the suppliers' disposal by means of which they are required to examine the sensitivity of each microphone and receiver capsule delivered to this Administration.

In addition it is necessary to measure the resistance of each microphone capsule when white noise of spectrum restricted to the band 300–3400 Hz is applied to the microphone in an acoustic chamber. The microphone is connected to an electrical circuit which, for both a.c. and d.c., is equivalent to the average conditions obtained when the microphone is connected in the telephone network. The d.c. resistance is also measured in this condition at the current which would apply in practice. The noise voltage produced by the microphone is measured by means of a d.c. voltmeter connected in a Graetz circuit. The voltmeter indicates approximately the r.m.s. value.

For measuring the telephone receiver, the reciprocity principle is used by applying the white noise to the receiver, acoustically, and measuring the voltage across the receiver.

In this case too, the receiver is connected in a circuit which has the same nominal impedance as that of normal telephone equipment.

The levels measured in this way yield a statistical distribution and the Administration requires that no microphone or receiver capsule may be accepted which departs more than ± 3 dB from the mean. The absolute level of the mean is also fixed by the Administration.

So far as the "sensitivity-frequency" characteristic is concerned the manufacturers are required to guarantee, for each capsule, that this complies with the tolerances specified in the Administration's standard. Experience has shown that the Dutch Administration can confine itself to checking from time to time by sampling whether the relevant clauses concerning the sensitivity-frequency characteristic are being observed. In general, the Administration uses the same measuring equipment for checking as is used in the factory. The measuring equipment used by the manufacturer for final checking in the factory must have been approved by the Administration. Furthermore, the Administration has reserved itself the right to make measurements on the microphones and receivers in the factory.

The transmission characteristics of each induction coil must be guaranteed by the manufacturer. He can conduct his checking during manufacture in a manner approved by the Dutch Administration.

Federal Republic of Germany

For tests made from the transmission point of view, the Federal German Posts and Telecommunications Administration uses, for the acceptance of subscribers' telephone equipment by its telecommunications stores depots, the equipment for objectively measuring reference equivalent described in Annex 28, Part II of Volume V of the *Red Book*. It has been possible to prove that, in the case of good manufacture, there are scarcely any faults in assembling telephone equipment. It is therefore sufficient to make random tests at the time of acceptance. Nevertheless, on delivery all microphone and receiver capsules are again measured and grouped as described in Rec. P.81. Furthermore, all reconditioned telephone equipment must be tested, but this is an easy matter because only a small number of items is generally involved.

When testing telephone equipment the mean sending and receiving loss is measured between the frequency limits of 200 and 4000 Hz. The resistances of line, receiver and microphone are each replaced by a 600-ohm resistor.

United Kingdom Post Office

General. — The processes of manufacture and the measurements made by the manufacturer are liable for inspection at any time by the Inspection Branch. Acceptance measurements are made on every piece of equipment manufactured or on samples chosen at random at the discretion of the Inspection Branch. The nature of the acceptance measurements is determined by agreement between the purchasing authority and the manufacturer before the contract is placed.

Electro-acoustical measurements on telephone microphones and receivers. — Each manufacturer is required by the Post Office to equip himself with measuring equipment of an approved design. This employs specified bandwidths of continuous-spectrum noise. To ensure that each manufacturer used the same testing signal, these bands of noise have been recorded by the Post Office in the form of optical soundtracks on a glass disc; each manufacturer is supplied with discs which are positive prints from the master negative.

For testing microphones, the appropriate noise signal is fed into an artificial mouth (see Annex 11 in Part II of Volume V of the *Red Book*), the output level of which is adjusted to a specified value with the aid of a probe microphone. The carbon microphone under test is given a conditioning treatment, placed in a standardized position in front of the artificial mouth, and the output voltage across a standard circuit is observed by means of a voltmeter. Special steps are taken to check the pressure calibrations of all the probe microphones in use at the various factories.

The voltmeter indicates true r.m.s. values, is scaled in decibels relative to 1V, has an integration time of 1.4 seconds and its reading, when a sinusoid is applied, is independent of frequency over the range 300–3400 Hz.

The noise signals used are as follows. First, the over-all sensitivity of the microphone is measured by applying wide-band noise (300–3400 Hz); for the microphone Transmitter Inset No. 16, the permitted tolerance is ± 2 dB on the specified value. Secondly, a measurement is made with each of three narrow bands, to check that the frequency response is within tolerance.

For testing telephone receivers, the noise source feeds the receiver being measured which is placed on an artificial ear (see Annex 11 in Part II of Volume V of the *Red Book*); the output voltage of the artificial ear is measured across a standard circuit by means of a voltmeter. For the type of subscriber's telephone receiver normally manufactured, acceptance measurements are specified with three narrow bands of noise.

Complete telephone sets are not measured for performance. As all the component parts have been separately tested before assembly, only a simple check to ensure that the telephone does work is considered necessary.

Switzerland

Subscribers' equipment and spares purchased by the Swiss Telephone Administration are acceptance-tested. This work is entrusted to the stores testing section of the Research and Testing Division. Bulk tests are made with appropriate measuring apparatus. Some components are furnished to the manufacturers of telephone apparatus after being tested by the PTT, for example, the capacitance and insulation of condensers, the handset and various cords. The return speed and impulse ratio of the dial contacts and the short-circuit contacts are tested in a few seconds with SC12/SC14 (Sodeco) equipment.

The subscribers' equipment (without handset) is acceptance-tested with TLPG3 (Zellweger) measuring apparatus in a minute or so. This test covers insulation, loudness of the bell, impedance for the circuit conditions applying when calls are being received, composite attenuations for sending, receiving and sidetone, at 400 and 1600 Hz if necessary for two different feeding currents; the check may also be extended to cover high-frequency interference suppression, the busy impulse for party lines and any auxiliary circuits in the subscriber's equipment.

Microphones and earphones are checked rapidly with the KP51/MPG12 (Autophon/Zellweger) measuring apparatus described in Annex 29, Part II of Volume V of the *Red Book*. Checks are made of the reference equivalent and the frequency curve as well as the resistance and noise level of the microphones and the centring of the transmission system in respect of the earphones.

Manufacturers of telephone equipment and components use similar measuring apparatus.

Apparatus returned as faulty by the operating services is tested in the same way and by the same testing section as that delivered by the suppliers.

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PART II

SUPPLEMENTS TO RECOMMENDATION SERIES P

Foreword

Supplements Nos. 1 to 6, and 8 to 15 are conserved in Volume V of the *White Book* (Geneva, 1968) and have not been reproduced in this Volume, which contains only Supplement No. 7.

Former Supplement No. 7 to Volume V of the *White Book* has been transferred to the C.C.I.T.T. Handbook: *Transmission Planning of Switched Telephone Networks.*

Supplement No. 7 (Geneva, 1972; referred to in Recommendation P.21 (G.121))

INFLUENCE OF SPEECH PATH UNBALANCE IN TERMS OF A REFERENCE EQUIVALENT ON THE QUALITY OF SPEECH TRANSMISSION

If Recommendation P.11 is satisfied, the transmission loss, circuit noise and distortion in each direction of transmission will not be more unfavourable than certain limits which will ensure satisfactory conversational performance even when the limits are reached in both directions simultaneously. Asymmetry will be present if the nominal overall reference equivalent in one direction is at the limit but that in the other falls short of the limit. Under such conditions one participant in a conversation may be able to hear the other very well but have some difficulty in making his speech understood due to the greater loss in the direction from him to his respondent than in the other direction. Clearly this participant will rate the connection less favourably than if the other participant can hear him as well as he can hear the distant participant. It might also be supposed that the person who has to listen to speech over the worse direction path would rate this less highly because the other person is not experiencing the same difficulty as he is; however, this is not so.

The results of the tests made by some Administrations are given below in this supplement:

A. The upper part of Figure 1/Suppl. No. 7 reproduces some results of Laboratory conversation tests conducted by the L.M. Ericsson Company, Stockholm, Sweden [1]. The transmission loss of the connection was made different in the two directions and the results can be compared with those for symmetrical connections of various losses. Curve (a) shows the conversational mean opinion score [1] for symmetrical conditions as a function of the nominal overall reference equivalent. Curves (b) and (c) show the results when the loss from A to B was kept constant at 42 dB nominal overall reference equivalent and that from B to A varied over the range 0 to 42 dB; curve (b) shows the results for the participants at A and curve (c) shows those for B. A's opinion [curve (b)] is slightly reduced by the larger loss in the direction A to B but B's opinion [curve (c)] was hardly affected by the smaller loss in the direction B to A. The speech levels measured at the outputs of the respective telephone sets are shown in curves (d), (e) and (f); it will be noticed that the speech level of B is raised in the asymmetrical cases as well as that of A, although A can hear B quite well without B

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[1] The scores in this figure are based on opinions using the 5-category scale: "Excellent-Good-Fair-Poor-Bad". Scores of 4-3-2-1-0 were allocated and the arithmetic mean was taken for each test condition.

talking any louder. This effect in which the subjects try to share their difficulties has also been observed in other tests [2].

B. The lower part of Figure 1/Suppl. No. 7 shows similar results from tests conducted in the same way but in the laboratories of the Swedish Board of Telecommunications [3]. Curve (g) shows the results for symmetrical connections; the asymmetry in these tests was, however, arranged so that the mean loss of the two directions remained constant at about 22 dB. Amounts of asymmetry (differences between the losses in the two directions) up to ± 44 dB were introduced. The results for the asymmetrical cases are plotted against the loss (nominal overall reference equivalent) in the direction B to A; a value of 22 dB on the abscissa scale therefore corresponds to zero asymmetry, 12 dB represents 20 dB asymmetry, etc., as indicated in the figure. It will be seen that for asymmetry within the range ± 20 dB, A's opinion is affected only by the loss from B to A and B's opinion only by that from A to B as shown in curves (h) and (i). Even when the asymmetry is very large, the opinion of the less fortunate participant (who suffers by receiving the more attenuated speech) is not worse than it would have been had the connection been symmetrical and of loss equal to the worse direction of the asymmetrical case.

C. Tests have also been carried out by the U.S.S.R. Administration on a high-quality experimental transmission path (see reference [4]) and the results were assessed by the operators in the form of opinion scores according to the method described in Volume V of the C.C.I.T.T. *Red Book*, pages 594 to 598.

The 24 operators (10 men and 14 women) who took part in the test, were not experts in the assessment of telephone transmission quality; 6 sets of testing were carried out and 720 marks were obtained. The different values of unbalance, as 0, 4.3, 8.7, 13 and 17.4 dB, have been introduced by changing the path reference equivalent in one of the transmission directions and keeping constant that of the other direction.

The tests were carried out in the two trials. In the first trial the value of 36.5 dB was adopted for the path with a constant reference equivalent value.

The reference equivalent in the second path was changed in the following sequence: 36.5, 32, 27.8, 23.5 and 19 dB. In the second trial the constant reference equivalent was of 23.5 dB value. The changing of the reference equivalent was carried out in the following sequence: 23.5, 19, 14.8, 10 and 6 dB. The assessment of transmission quality in each trial was performed under the three following conditions:

1. The paths under test were to be assessed by operators in increasing order of an unbalance value, as 0; 4.3; 8.7; 13 and 17.4 dB.

2. The paths under test were to be assessed by operators in decreasing order of an unbalance value, as 17.4; 13; 8.7; 4.3 and 0 dB.

3. The paths under test were to be assessed by operators in a random sequence of unbalance values.

The relationship between the operators' assessment and unbalance value for every test condition is illustrated in Figure 2/Suppl. No. 7.

Consideration of test results illustrated in Figure 2/Suppl. No. 7 enables us to draw the following conclusions:

1. An unbalance increase from 0 to 17.4 dB does not, in practice, lead (= 0.2 points) to the change of transmission quality assessment for the paths with a constant reference equivalent value in the first trial, as well as in the second one.

2. The improvement of assessment marks for the paths with the adjustable reference equivalent is caused by the increase of a receive loudness and does not depend on unbalance either.

3. The results obtained under the different conditions of testing (assessment of the paths in decreasing or increasing order of an unbalance value and in a random sequence of these values) were in sufficiently close agreement with each other.

D. Finally, in the document indicated in reference [5], the American Telephone and Telegraph Company reports the results of subjective tests on actual telephone calls within the Holmdel location of Bell Telephone Laboratories.

The influence of unbalance was evaluated for a total of 47 test conditions. Four values of noise were included in the range from −73 to −50 dBmp at the input to a telephone set with an estimated receive reference equivalent of −5 dB. In all conditions the noise was equal in the two directions of transmission. For each value of noise, overall reference equivalents were varied from 10 to 30 dB. Differences in the overall reference equivalent for the two directions were varied from 0 to 20 dB.

The test conditions were inserted on a portion of the normal calls made by employees of Bell Telephone Laboratories who had agreed to participate as subjects in the tests. All calls were to other employees in the Laboratories.

The rating results indicated that:

1. Quality ratings made by the subjects depended primarily on the circuit noise and overall reference equivalent in the direction of transmission toward the subject.
2. Variations in the overall reference equivalent in the direction of transmission away from the subject had a much smaller effect on ratings. In general, 20 dB changes in the overall reference equivalent in the direction of transmission away from the subject resulted in changes in mean opinion score which were approximately equivalent to those caused by 5 dB changes in overall reference equivalent in the direction of transmission toward the subject.
3. Although the mean opinion scores decreased with increased circuit noise, the effects noted in points 1 and 2 above were essentially independent of the noise.

Asymmetry can arise from other factors than line transmission loss. For example, the telephone sets in different countries have somewhat different relationships between their sending and receiving sensitivities (or reference equivalents) and so asymmetry of as much as 8 to 10 dB can exist when particular types are interconnected over an international circuit. Naturally, within each country, the connections will remain symmetrical notwithstanding differences in the quantity SRE-RRE. Certain other circumstances can increase asymmetry of loss by a few decibels and additional asymmetry can be caused by circuit noise or room noise being different for the two participants. Even when the connection is physically symmetrical and the room noise is the same at both ends, talkers may emit their speech at levels differing by as much as 15 dB; participants will experience the effects of this in ordinary, non-telephone conversation and accept the situation and this probably accounts for their willingness to accept quite large physical asymmetries in a telephone connection.

From the foregoing results it can be concluded that such asymmetries as are likely to arise when Recommendation P.11 is satisfied are not important from the point of view of subscribers using such connections. It is consequently unnecessary to make any further recommendation on the matter.

References

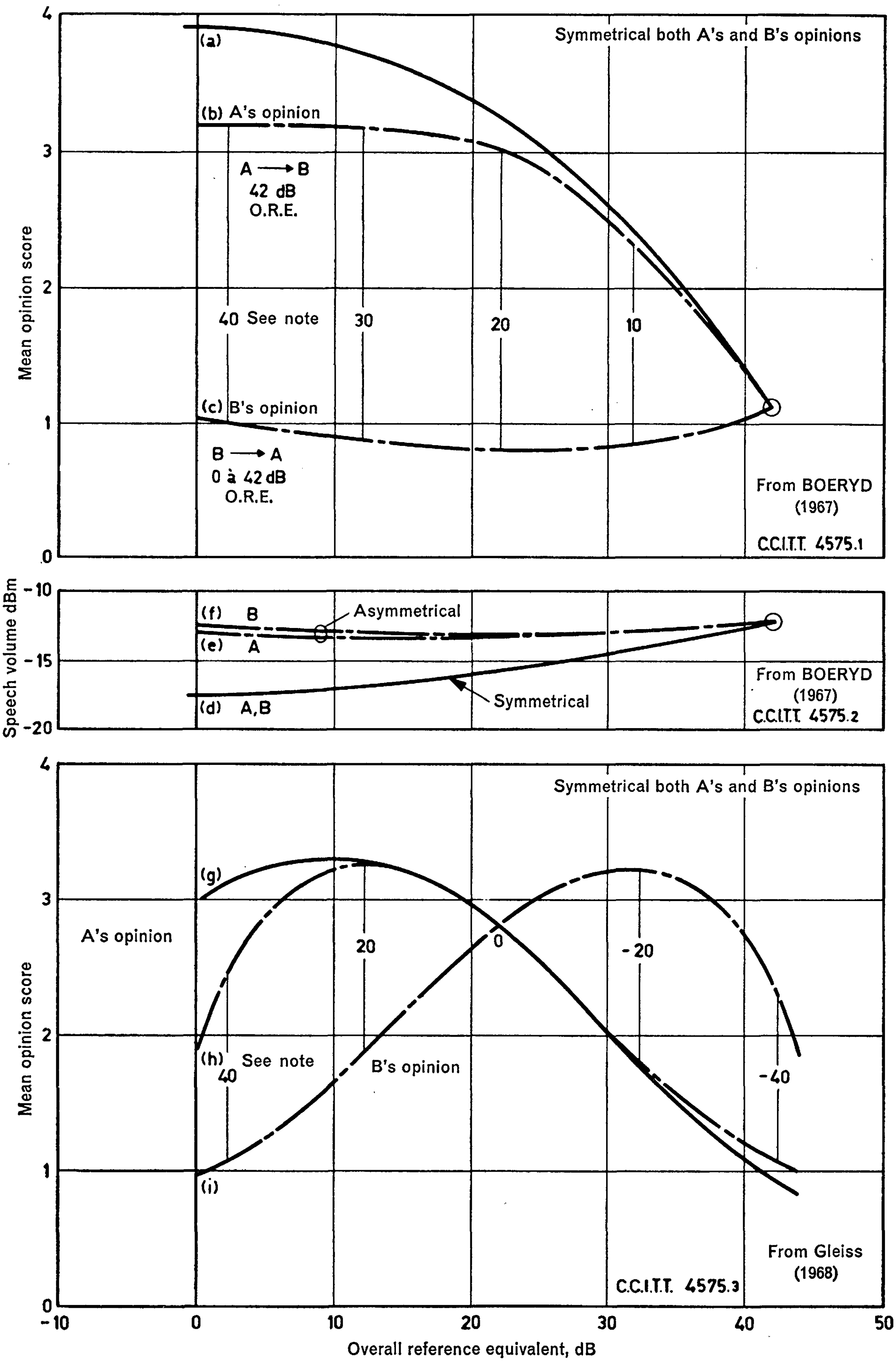
[1] Boeryd, A.: Subscriber reaction due to unbalanced transmission levels: *Het-Bedrijf*, 1967, *15*, pp. 39–43.

[2] Richards, D. L.: Some aspects of the behaviour of telephone users as affected by the physical properties of the circuit; Communication Theory, 1953, Butterworth's Scientific Publications, pp. 442–449.

[3] Gleiss, N.: Subjective measurements in telephony: *Tele*, English Edition 1968, *20*, pp. 29–44.

[4] Influence of speech path unbalance in terms of a reference equivalent on a quality of speech transmission. Contribution of the U.S.S.R. Administration, COM XII-No. 85 (period 1968/1972).

[5] Effect of asymmetry between the two directions of transmission. Contribution of the American Telephone and Telegraph Co., COM XII-No. 107 (period 1968/1972).



Note – Numbers show amount of asymmetry

FIGURE 1/Suppl. No. 7. – **Effect of asymmetrical transmission loss on conversational opinion scores**

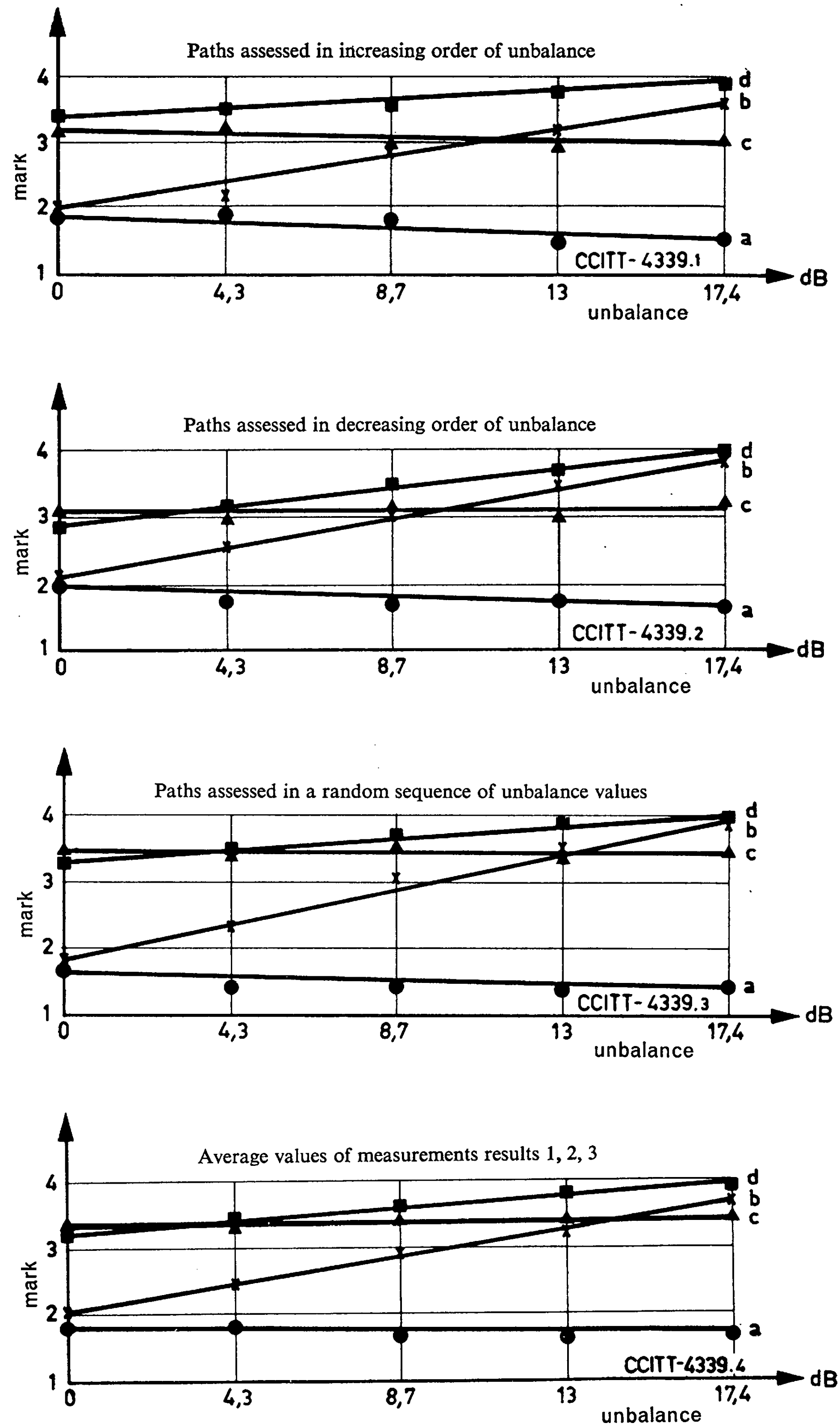


FIGURE 2/Suppl. No. 7. — **Operator mark vs. unbalance**

a = path with constant attenuation 36.5 dB
b = path with adjustable attenuation 36.5; 32; 27.8; 23.5; 19 dB
c = path with constant attenuation 24.4 dB
d = path with adjustable attenuation 24.4; 19.9; 15.7; 10.9; 6.9 dB

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PART III

QUESTIONS ON TRANSMISSION PERFORMANCE AND LOCAL NETWORKS TO BE EXAMINED BY STUDY GROUP XII IN THE 1972/1976 STUDY PERIOD

Important note

Notes to the effect that a particular question is of interest to various Study Groups when no joint study group has been set up to study it are intended primarily for the information of the members of the Study Group dealing with the question, to enable them to arrange for the necessary coordination within their national administrations, in accordance with the decision taken by the Plenary Assembly.

Question 1/XII — Reference equivalents of national systems in the international transmission plan

(Continuation of Question 1/XII studied in 1968–1972)

a) *Sending reference equivalent*

What are the nominal maximum, traffic-weighted median, and nominal minimum values to be recommended for the distribution of the sending reference equivalent of the national system calculated to the international send virtual switching point?

b) *Receiving reference equivalent*

What are the nominal maximum and traffic-weighted median values to be recommended for the distribution of the receiving reference equivalent of the national system calculated from the international receive virtual switching point?

c) *Overall reference equivalent*

What, according to listener preferences, should the short- and long-term planning objectives be for the median value of the distribution of nominal overall reference equivalents and how should it be apportioned between national sending and receiving systems?

Note 1. — Annex 1 is an extract from the reply to Question 1/XII studied during the 1964–1968 period.

Note 2. — Administrations should submit information concerning national sending and receiving reference equivalents. Annexes 2 and 3 are contributions from the Australian Post Office.

Note 3. — Annex 4 gives instructions concerning how contributions to Part c) of the question should be made, and Annex 5 contains proposals concerning opinion scales for listening-only tests.

ANNEX 1

(to Question 1/XII)

Extract from the Report (approved by Study Group XII in October 1967) by Mr. D. L. Richards' Working Party[1]

The percentage of international calls for which it seems possible to satisfy the limits of 21 dB and 12 dB for the national reference equivalents is at least 95%, and for some Administrations the percentage reaches 97.

[1] The C.C.I.T.T. Secretariat has deleted those portions of the report which dealt with parts of the question which have now been abandoned.

The Fifth Plenary Assembly amended section B.a in Recommendation P.21 accordingly.

This part of the question requires further study to clarify the section as a whole and for the adoption of final recommendations. The following considerations should be borne in mind:

The Working Party considered that a percentage value associated with the limits could not easily be used for actual planning by administrations which do not use laws of statistical distribution for this purpose.[1] In practice, the percentage must be determined later from a survey of the actual network after the operation of any given planning rules. Annex 2 (especially Figure 3) gives the results of such a survey of the Australian network. Some discussion took place on the possibility of re-wording Recommendations P.11 and P.21 to explain more clearly what the percentages were supposed to refer to; no agreement could be reached on such re-wording, but it was the general opinion that, in principle, planning ought to aim at complying with the stated limits for all international calls and that the few per cent of exceptions was to be treated as a reserve.

ANNEX 2

(to Question 1/XII)

National sending and receiving reference equivalents of international calls outgoing from Australia

(Contribution by the Australian Administration)

A one-day sample of 1 233 outgoing international calls from Australia (see appendix) was analysed to determine the originating subscriber and the transmission connection by which each subscriber was switched to the international exchange at Sydney. Similar information on incoming calls is more difficult to obtain and is not yet available.

The transmission loss of each connection from the terminal exchange (end office) to the international exchange in Sydney was obtained with the aid of cable plans, estimated switching losses, maintenance records and transmission measurements.

The reference attenuation of this connection in each direction was then calculated with respect to the virtual switching points — 3.5 dBr sending and — 4 dBr receiving at the international exchange. These distributions were called D'_s (sending) and D'_r (receiving) and are shown in Figure 1.

The reference equivalents of each subscriber's line and telephone were not determined. However, information was available from a previous sample survey of Australian subscribers' local ends which yielded representative distributions D''_s (sending) and D''_r (receiving) of subscribers' local ends. These distributions are shown in Figure 2.

The statistical sums of D'_s and D''_s and of D'_r and D''_r are shown in Figure 3 and represent the present distributions of sending and receiving reference equivalents respectively for outgoing international calls from the Australian national network, referred to the virtual switching points at the Sydney international exchange.

Key points on the distribution are as follows:

	Sending	*Receiving*
Median	13.8 dB	2 dB
95% less than	20.2 dB	8.4 dB
97% less than	21.2 dB	9 dB

The Australian network is therefore within the 95% limits (21.3 dB sending, 12.7 dB receiving) of Recommendation P.11/G.121 (1964 Plenary Assembly) for a large country with four four-wire circuits as part of its national chain.

Furthermore, these limits wouid also be respected by more than 97% of connections, thus complying with the provisional recommendation in Recommendation P.11 (Volume V of the *Green Book*).

[1] The A. T. & T. Co. was one of the Administrations which stated that much of their system planning was done in terms of distributions rather than specific limits. The following papers published in *Bell System Technical Journal*, Volume XLIII, No. 2, March 1965, give examples of this approach:

I. NÄSELL : The 1962 survey of noise and loss on toll connections, pp. 697–718.

D. A. LEWINSKI : A new objective for message circuit noise, pp. 719–740.

Comments

The distributions of reference equivalents for subscribers' local ends assume the use of telephones with microphones and receivers having an efficiency equal to the lowest permitted for new telephones; also, no corrections have been made for the effect of teed cable pairs.

The average efficiencies of new telephone instruments are about 2 dB higher on sending and receiving than the minimum permitted values, and only about one-quarter of sample subscribers' microphones have been found to be less efficient than the minimum permitted for new instruments, over an age range of more than 20 years.

Nothing is so far known about the deterioration of receiver efficiency with time.

Teed cable pairs have an adverse transmission effect on less than 4% of subscribers' lines and more than 60% have no teed cable.

If these effects are taken account of in the distribution of Figure 2, there is no significant change to the previous conclusions.

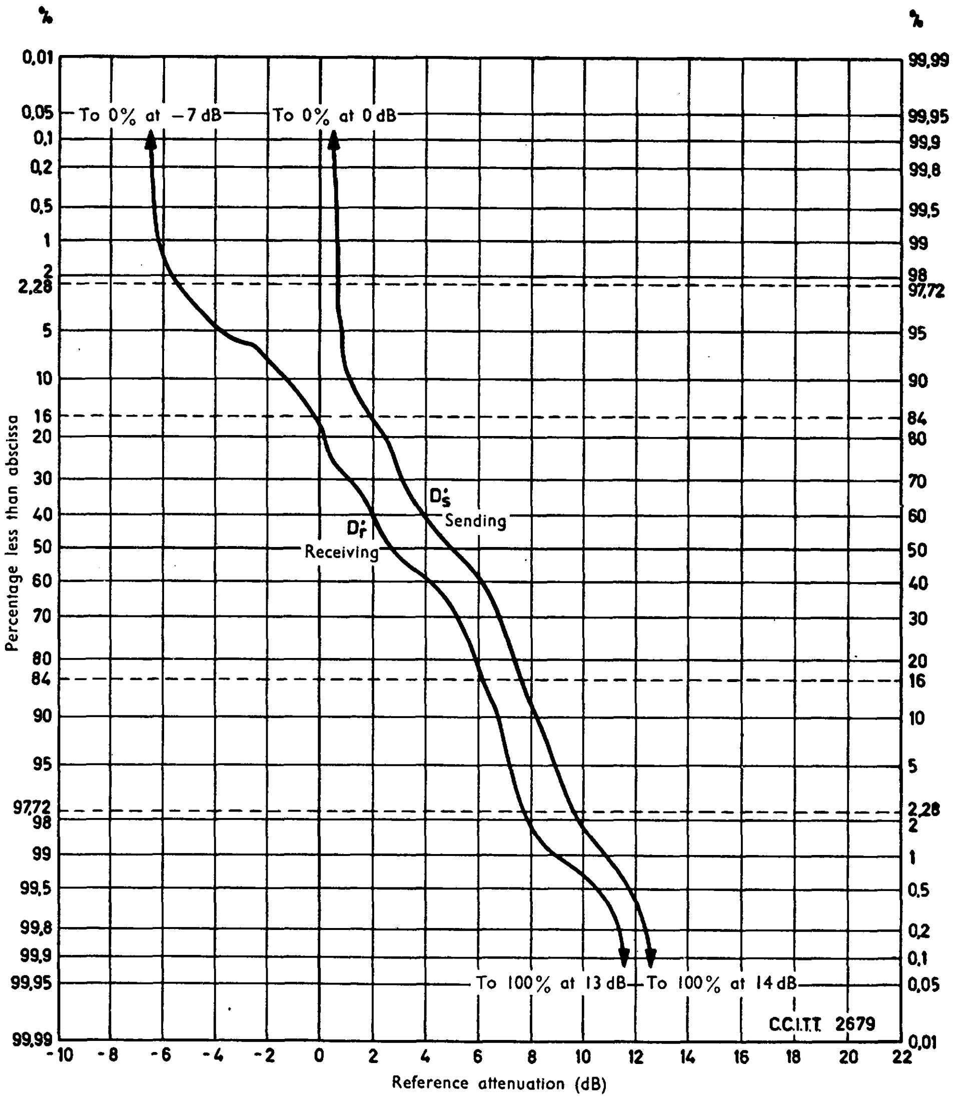


FIGURE 1. — **Reference attenuation of inter-exchange network between terminal exchange and international virtual switching points**

(International outgoing calls—10 May 1967)

APPENDIX (to Annex 2)

Make-up of sample of 1233 outgoing international calls from Australia

All calls were routed through the main trunk switching centre of the State of origin to the international exchange in Sydney. The distances from the State main trunk switching centres to the international exchange are included as an indication of the lengths of connections in the Australian national network.

State	Calls from capital city area		Calls from country areas		Total calls		Distance from State main trunk switching centre to international exchange
	Number	%	Number	%	Number	%	
New South Wales	570	89.5	67	10.5	637	51.7	Nominal distance 0
Victoria	316	88.7	40	11.3	356	28.9	950 km
Queensland	75	63.5	43	36.5	118	9.55	1045 km
South Australia	43	79.6	11	20.4	54	4.35	1750 km
Western Australia	44	89.8	5	10.2	49	3.96	4080 km
Tasmania	15	79	4	21	19	1.54	1590 km
Totals	1063	86.1	170	13.9	1233	—	

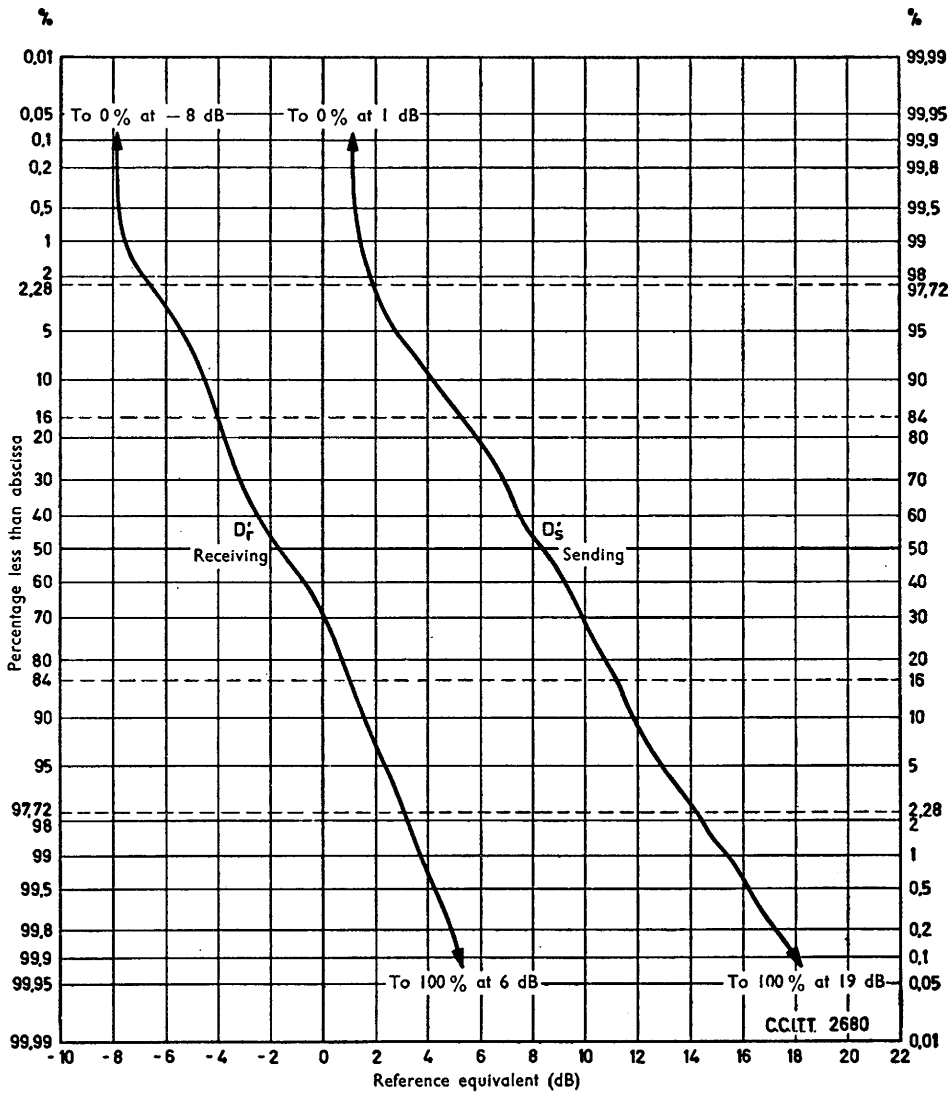
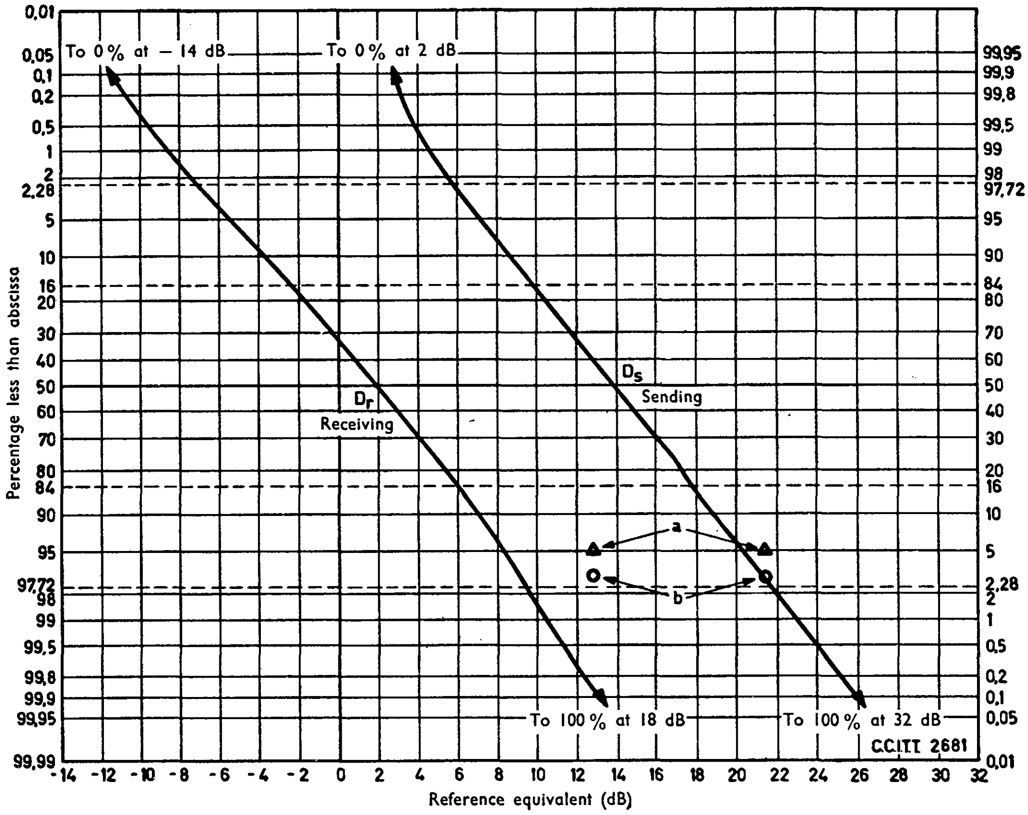


FIGURE 2. — **National subscribers' sending and receiving reference equivalents**
(Survey of subscribers' network, 1965)



a = 95% points of former Recommendation G.121 (for a large country). b = 97% points.

FIGURE 3. — **Nominal sending and receiving reference equivalents of the Australian national system**
(International calls outgoing from Australia on 10 May 1967: number in sample = 1233)

ANNEX 3

(to Question 1/XII)

Maximum, minimum and preferred reference equivalents

(Note by the Australian Administration)

In the experience of several Administrations, certain connections involving short-lines and P.A.B.X.s and local calls are too loud unless special precautions are taken to reduce the volume, that is, to increase the overall reference equivalent. Several Administrations fit resistors to build out short-lines on certain types of telephones; other telephones are automatically regulated to control the minimum reference equivalent.

New equipment now appearing will enable losses in the network to be reduced, in many cases without significant expense. The following are examples of such equipment: subscribers' carrier lines, p.c.m. junctions, amplified receiver and microphone insets, integrated switching and transmission. With the spread of such equipment, and the possibility of adaptive echo suppressors becoming available, the "too loud" problem hitherto encountered on short subscribers' lines only could be encountered even on inter-exchange calls. As a consequence, short- and long-term planning objectives are needed for preferred overall reference equivalents as well as the minimum and maximum limits.

ANNEX 4

(to Question 1/XII, Part C)

Short-term and long-term planning objectives for reference equivalents

It is difficult to usefully compare measurement results from various Administrations, because of the different conditions used in conducting the tests. Study Group XII agreed upon the following outline guide for the conduct of tests to determine the preferred nominal overall reference equivalent:

Essential features of tests to determine the preferred range and preferred value of nominal overall reference equivalent (or nominal overall loudness rating)

Conversation tests

As suggested for the study of Question 4/XII:

— Range: —10 to +20 dB nominal overall reference equivalent (or loudness rating).
— Room noise: to include the value +50 dB Hoth spectrum. A filter weighting.
— Circuit noise: to include —60 dBmp referred to a 0 dB receive reference equivalent end.
— Sidetone reference equivalent: to be stated.
— Speech volume referred to the output of a 0 dB send reference equivalent and to be measured and stated.
— Opinions recorded on the scales given in Question 4/XII.

Listening only tests

— Range: 0 to —50 dBV referred to a 0 dB receive reference equivalent end.
— Room noise, Circuit noise, Sidetone reference equivalent } As for conversation tests
— Opinion recorded on
  a) 5-category listening effort scale
  b) 5-category loudness preference scale

Annex 5 contains proposals for the exact wording of the listening effort and loudness preference scales.

It should be noted that results obtained by listening-only tests can be compared with those obtained by conversation tests by taking into account the speech levels applicable under the two different conditions.

ANNEX 5

(to Question 1/XII, Part C)

The following opinion scales have been found useful by the United Kingdom Post Office for listening tests in which the talking conditions are held constant or recorded speech is used.

Opinion scale for listening effort

A Complete relaxation possible; no effort required.
B Attention necessary; no appreciable effort required.
C Moderate effort required.
D Considerable effort required.
E No meaning understood with any feasible effort.

Opinion scale for loudness preference

A Much louder than preferred.
B Louder than preferred.
C Preferred.
D Quieter than preferred.
E Much quieter than preferred.

With either scale the categories can be scored 4, 3, 2, 1 and 0 respectively for A to E; mean scores can then be calculated and plotted as functions of transmission loss or listening level. Such results have been found to produce parabolic graphs for the listening effort scale and straight lines for the loudness preference scale. The preferred value of transmission loss or listening level may be taken as the maximum of the parabola obtained from the listening effort scale; this value is found to correspond to a mean score of 2.0 to 2.5 from the loudness preference scale, depending somewhat upon the noise present.

Question 2/XII — Assessment of service transmission quality

(Continuation of Question 2/XII studied in 1967–1968)

Considering

that with the advent of world-wide automatic and semi-automatic networks operating personnel will be less able to detect the onset of unsatisfactory service conditions,

that connections in such world-wide networks will be more complex and include more elements as potential sources of transmission difficulty, and

that customers will expect higher quality of service as their use of the world-wide networks increases,

the following questions shall be studied:

a) What methods are suitable for the evaluation of service from the standpoint of speech transmission quality?

b) Is it desirable to standardize methods to be used as part of overall appraisals of world-wide subscriber-to-subscriber connections?

Note 1. — General approaches which might be considered for this purpose include service observations by third party observers and subscriber interrogation by interview or questionnaire. Annex 1 to Question 2/XII, *White Book*, Volume V, describes this kind of method and Annex 2 to Question 2/XII, *White Book*, Volume V, gives an example of test programmes. See also the Annex to this question.

Note 2. — It has been noted by certain Administrations that some measures of service quality vary during the course of a conversation. For example, it has been observed that repetition events are more frequent at the beginning of a conversation (see Annexes 3 and 4 to Question 2/XII, *White Book*, Volume V). This factor should be considered in studying methods of evaluating transmission quality.

Note 3. — Attention is drawn to the importance of using suitable methods for preparing test programmes and for the analysis of results.

Note 4. — This question is related to Question 7/XII part (a) which asks what criterion should be adopted to determine quality of performance of different transmission systems, preliminary to developing objective measurement methods.

Note 5. — Study Groups II and XIII study questions relating to service quality by means of service observations and questionnaires which include elements on speech transmission quality which are also relevant to the study of Question 2/XII. Coordination is particularly desirable between these Study Groups on transmission data arising from the use of Recommendations Q.61, 62 and 63 and study of Questions 9/XIII and 12/XIII in Study Group XIII and from the use of Recommendation E. 425 and study of Question 8/II in Study Group II.

ANNEX

(to Question 2/XII)

Extract from the questionnaires annexed to Recommendation E. 425

Reproduced below are the questions relating to transmission quality which appear, with the same wording, in the questionnaires annexed to Recommendation E.425:

Points 9.0 to 10.8 in Appendix I: for National Users.
Points 12.0 to 13.8 in Appendix II: for Foreign Visitors.

The extract below follows the numbering in Appendix I.

9.0 *Which of these four words comes closest to describing the quality of the connection during conversation?*

9.1 — *Excellent*	☐		
9.2 — *Good*	☐		
9.3 — *Fair*	☐		
9.4 — *Poor*	☐		
10.0 *Did you or the person you were talking to have any difficulty in talking or hearing over that connection?*		Yes ☐	No ☐

(If answer is "Yes" probe for nature of difficulty, but without suggesting possible types of difficulty, and copy down answers verbatim: e.g. *Could you describe the difficulty a little more?*)

(at end of interview, categorize the answers in terms of the items below):

10.1 — Low volume	☐
10.2 — Noise or hum	☐
10.3 — Distortion	☐
10.4 — Variations in level, cutting on and off	☐
10.5 — Crosstalk	☐
10.6 — Echo	☐
10.7 — Complete cut off	☐
10.8 — Other (specify)	☐

Note. — Item 4 of the associated "Notes on Intended Uses of Questionnaires" reads as follows: "4) It is intended that *only the underlined questions in italic type* should be asked by the interviewer. The sub-sections which are not underlined are to be used by the interviewer to categorize the answers."

Question 3/XII — Reference equivalents of operators' headsets

(New Question, replaces point f of Question 1/XII studied in 1968-1972)

What methods of measurement should be recommended for the determination of reference equivalents and corresponding sensitivity/frequency characteristics of operators' headsets for sending and receiving?

Note 1. — Due consideration should also be given to the distribution of the speech power which operators' sets may produce.

Note 2. — Maximum and minimum values of reference equivalent of operators' telephone circuits cannot be recommended until a method of determining these quantities has been agreed. The Annex to this question indicates the criteria guiding the choice made by the United Kingdom Post Office concerning the sensitivity of operators' sets.

Note 3. — Question 12/XII is concerned with artificial mouths and ears suitable for measuring operators' headsets (see Note 3 is Question 12/XII).

ANNEX

(to Question 3/XII)

Criteria used by the United Kingdom Post Office for choosing the sensitivities of operators' sets

(Contribution by the United Kingdom Post Office)

The sensitivities, sending and receiving, of operators' sets have been chosen according to the following criteria:

Sending

The speech volume sent into the trunk line, with the local subscriber disconnected, when the operator is talking to the remote operator, should be about the same as that sent into the trunk line by actual subscribers when they are conversing with the remote subscribers. Under these conditions the speech volume referred to the two-wire input (point of zero relative level) is —14.4 dBm0 (standard deviation 3.4 dB) for operators and from —19.3 to —13.9 dBm0 for subscribers, depending upon the location of their local exchange relative to that of the trunk switching centre at which the measurements are being made. (Standard deviation for subscribers is of the order of 5.7 dB.)

Values of reference equivalent cannot be expressed for operators' sets in the sending direction because no directives exist for choosing the distance to be used in measurements of reference equivalent between lip position and mouthpiece opening. Suitable instructions for this are required and these should ensure that the same distance is used as that adopted by operators in actual service. Corresponding prescriptions are necessary regarding the talking level to be used.

It is important to note that operators' sets in general have a wide degree of adjustment of talking distance which will be used to advantage by the operator to improve sending sensitivity under difficult talking conditions; the adjustment could, of course, also be misused to affect transmission adversely. The specification of standardized conditions of talking distance and talking level for realistic measurements of reference equivalent must take these facts into account. It is therefore necessary to determine the median value of these quantities in actual service, for example by measurements of speech volume and other observations under actual conditions of use.

Indirect estimates of "reference equivalent" can be arrived at as follows. Under conversation conditions in the laboratory but in an environment representative of ordinary telephone use, a subscriber's local telephone circuit having a send reference equivalent of 7 dB yields a speech volume of about —15 dBm. It might therefore be expected that a speech level of —14.4 dBm0 would be produced by a set having a send reference equivalent of 6.4 dB. This value would refer to the two-wire input to the trunk circuit and would correspond to 9.9 dB when referred to the virtual switching point. Allowance must also be made before comparing this figure with that provisionally proposed in part d of Annex 1 to Question 1/XII, *White Book*, Volume V (namely 15.6 dB) for connection of the local subscriber; this will probably impose a bridging loss of about 3 dB giving an estimate for operators' sets of the United Kingdom Post Office of 12.9 dB which is comfortably within the value proposed.

Receiving

A subscriber's set with a negligibly short line has a receiving reference equivalent of about —6 dB (Telephone 706 of the United Kingdom). Such a set will reproduce received speech that is never uncomfortably loud when used by a subscriber; the same loudness rating associated with an operator's set will, however, be uncomfortably loud when the received speech volume is high and a reduction in sensitivity of about 3 dB must be made. The receive reference equivalent for United Kingdom operators' sets referred to the virtual switching point (3 dB adjustment assuming a trunk circuit having 7 dB loss between two-wire terminations) and including allowance for bridging loss by the local subscriber's line (say 3 dB) will be: $-6 + 3 + 3 + 3 = 3$ dB. Again this figure is within the provisionally proposed value of 7 dB.

The United Kingdom Post Office therefore has no objection to the provisionally proposed values provided that the directions, which are now lacking, for measurement of sending reference equivalents are determined in a realistic manner. These must take account of the manner in which operators actually use their sets.

Question 4/XII — Effect of circuit noise on transmission performance

(Continuation of Question 4/XII, studied in 1968-1972)

What is the family of curves which gives the noise transmission impairment as a function of the indications given by a psophometer standardized by the C.C.I.T.T. and for different values of nominal reference equivalent likely to be encountered in international connections?

Note 1. — Annex 1 provides an outline of suggestions and desired information which will facilitate comparison of results from various Administrations.

Note 2. — Annex 2 outlines a method for conducting conversational tests as used by the United Kingdom Post Office.

Note 3. — Annex 3 provides results submitted by A.T. & T. along the lines outlined in Annex 1.

ANNEX 1

(to Question 4/XII)

Suggestions for the study of Question 4/XII

The following outline of suggestions and desired information is provided to facilitate comparison of results submitted by various Administrations on the effect of circuit noise on transmission performance.

1. *Types of tests*

Tests should simulate as realistically as possible actual conversations of customers. Conversational type tests are therefore preferred. However, it is recognized that this type of testing is more difficult and it may be useful to use the results of listening tests to supplement the results obtained for conditions included in the conversational tests. It is suggested that the results from listening tests be validated by conversational tests and that the results be submitted in the form of conversational test results. Annex 2 below describes a method for conducting conversational tests which is used by the United Kingdom Post Office.

2. *Subjective evaluation of performance for the various test conditions*

Two types of subjective evaluation are preferred and Administrations are requested to use and submit results on both for the same test conditions whenever possible. The first involves the use of the five point opinion scale (excellent, good, fair, poor and bad) as described in the *Red Book*, Volume V, Annex 3, page 598. The second involves asking whether the subjects experienced difficulty in talking or hearing. A suggested wording is contained in paragraph 6 of Annex 2 below:

"Did you or the person who spoke to you have any difficulty in talking or hearing over the connection?"

Where both types of evaluation are used the order of presentation is left to the discretion of Administrations but should be specified.

3. *Test conditions*

Results are desired which cover the range of values of circuit noise and reference equivalents which occur on international connections. It is suggested that these be expressed in terms of the overall reference equivalent in decibels and circuit noise as measured at the junction side of the receiving office but referred to a 0 dB receiving reference equivalent and expressed in dBmp. Useful ranges would include overall reference equivalents from 0 to 40 dB and noise from —80 to —50 dBmp.

4. *Types of noise*

It is recognized that there are various types of noise (white, thermal, hum, impulses, tones, etc.) which are of importance and that their effect on transmission performance may differ. Results pertaining to these various types are of interest to Study Group XII. However, it is suggested that results be obtained initially using band-limited (approximately 300 to 3400 Hz) white thermal noise such as could be achieved with a filter characteristic similar to that shown in Figure 2 of Recommendation P.44, *Green Book*, Volume V. This would permit comparisons of the results from different Administrations. Subsequently it should be possible to study other types of noise by means of tests which compare these types of noise with band-limited white noise (pair comparison tests, for example). Study Group XII is especially interested in the results of tests with sinewaves. Administrations which may already have test results on other types of noise are invited to submit their results, including a description of the particular type of noise (amplitude and spectral density distributions, for example).

5. *Presentation of results*

Contributions should provide a complete description of the test method, test circuit, telephone instruments and environmental conditions. Desirable information includes:

— description of test arrangements,
— subjects (number, age, sex, training, etc.),
— instructions to subjects,
— room noise (spectrum and level),

— telephone instruments (type, frequency response, sending and receiving reference equivalents, sidetone frequency response and reference equivalents, type of transmitter, etc.),
— connecting circuitry (frequency response, battery supply arrangements).

It is suggested that the raw test results be presented in tabular form and the following entries are desirable:

Test condition (noise level and reference equivalent),
Number of subjects,
Percentage 'excellent',
Percentage 'good',
Percentage 'fair',
Percentage 'poor',
Percentage 'bad',
Percentage 'excellent' and 'good',
Percentage 'excellent', 'good' and 'fair',
Percentage 'poor' and 'bad',
Mean opinion score,
Percentage difficulty.

It is also suggested that the smoothed test results for these quantities be presented as curves; for example, percentage of difficulties plotted as a function of overall reference equivalent. Separate curves could be presented on the same figure for the different values of noise included in the tests. Examples of such curves are given in the *Red Book*, Volume V, page 591.

Note. — The mean opinion score should be calculated by assigning numerical values as follows:

Excellent	—4
Good	—3
Fair	—2
Poor	—1
Bad	—0

6. *Additional information*

As an aid in the comparison of test results from various Administrations, information on the speech volume of the subjects as a function of overall reference equivalent and circuit noise for the different test conditions are also requested. Although these may be measured at some other point in the test connection, it would be desirable for the results to be presented in terms of speech volumes at the output of a subscriber set and line having a sending reference equivalent of 0 dB.

ANNEX 2

(to Question 4/XII)

Method for conducting conversation tests

(Contribution of the United Kingdom Post Office)

1. *Choice of subjects*

Subjects taking part in the conversation tests were chosen at random (actually in alphabetical order of surname initial) from the Research Station personnel, with the provisos that:

a) they were not directly involved in work connected with assessment of the performance of telephone circuits, and

b) they had not participated in any objective test whatever for at least the previous six months. The numbers of male and female subjects were generally found to be roughly equal. Subjects were arbitrarily paired in the experimental design prior to the test and remained thus paired for its duration, but steps were taken, in the pairing process, to avoid gross inequalities in rank.

2. *Environment*

Subjects were seated in separate sound-proof cabinets near the point from which the experiment was controlled. Room noise (50 dB Hoth spectrum[1]) was introduced for all circuit conditions in each test.

3. *Method of establishing the connection*

The telephone sets used by subjects were, in appearance and feel, identical to the standard United Kingdom Post Office. Telephone No. 706. The means of establishing telephone contact between subjects was made as realistic as possible. The calling subject, on lifting the handset, obtained dialling tone; the dialling of a pre-determined number of digit trains established the connection, and suitable fixed delays were provided between completion of dialling and commencement of ringing tone, and between the commencement of ringing tone and the application of ringing current to the called subject's bell.

4. *Conversation task*

The means adopted to ensure that the conversations were purposeful, and to ensure that subjects fully exploited the transmission capabilities of the test circuit were as follows:

Each subject was provided, for each conversation, with a set of six postcard-size reproductions of paintings in the Tate Gallery. The pictures embraced diverse schools of art (i.e. impressionist, abstract as well as pictorial views, etc.) and the sets used by each pair were identical. Cards were identified by a code, comprising pairs of letters on the rear only (all other descriptive matter being obscured): identical pictures in each set bore the same codes.

The subjects were asked to imagine that a choice was to be made from the set for hanging in the Post Office Research Station Refreshment Club and immediately before each conversation, each subject was given a minute or so to arrange the cards, independently, in his own order of preference, and to note the identifying letters shown on the backs of the cards. The probability of the orders of preference of the two subjects being identical was extremely remote, and on this basis, when connection had been established over the test circuit, they were to negotiate so as to arrive at an order of the six pictures which satisfied both. The final or compromise order was also noted.

The time occupied for each conversation was about five or six minutes.

5. *Instruction of subjects*

In order to standardize the instructions given to subjects, and to ensure that each subject was fully apprised of what was required of him, the following procedure was adopted:

a) After selection of prospective subjects as detailed in Section 1, initial approach was made by telephone to ascertain their willingness to take part in the tests. A confirmatory note was sent to those agreeing, with a brief description of the conversational task.

b) Prior to the first condition of the first entry, the subjects were invited to listen to a recording in the presence of the controlling officer, which gave more detailed instructions on the conversational task. Any queries raised by the subjects were dealt with at this stage. A text of the recording is given in Appendix 1.

6. *Procedure and subjects' assessment of the connection*

After listening to the recording, the subjects were taken to their respective cabinets and were handed their picture card sets and an opinion form (see Appendix 2). After a short period during which they arranged and noted their orders of preference of the pictures, a lamp indication was given by the controlling officer to the subject nominated to be the caller, who then set up the connection by dialling, and the conversation proceeded normally. At the conclusion of the conversation, the subjects noted the agreed picture set order and replaced the handsets. They were then called, separately, by the controlling officer over the test connection and reminded to complete section 5 of the opinion form. They were then asked if they had any difficulty (the actual question being :"Did you, or the person who spoke to you, have any difficulty in talking or hearing over that connection?"). If the answer was "Yes" the controlling officer asked the subject to explain the nature of the difficulty which he (the controlling officer) noted.

[1] For this measurement, Dawe Instruments Sound Level Meter Type 1400F, weighting characteristic "A" meter "slow" was used.

7. Experimental design

Each experiment is usually based on a 12×12 graeco-latin square, with twelve circuit conditions, twelve pairs of subjects and twelve sets of picture cards. As stated in Section 1, subjects were paired in the experimental design prior to each experiment and remained thus paired for its duration. Also, subjects occupy respectively the same cabinets throughout each test. For subject pairs 1 to 6, subject A of the pair initiates the call to subject B for odd-numbered conditions: for subject pairs 7 to 12, subject A initiates the call for even-numbered conditions.

Although the experimental design is based on twelve conditions, subjects sometimes take part in thirteen conversations. The first conversation of the first entry is regarded as a "conditioning" conversation, and the circuit parameters are set at arbitrary fixed values. The results from the "conditioning" conversation are not taken into account in the analysis of the experiment.

APPENDIX 1

(to Annex 2 of Question 4/XII)

Text of recording replayed to subjects immediately prior to participation in a conversation test

Tests over simulated telephone connections

A series of tests is being carried out to study the performance of telephone connections, for which your assistance is requested. The tests involve conversing with another person, who has also agreed to take part, over a telephone connection which includes certain test equipment. You will be given a task which, it is hoped, will result in vigorous conversation devoted to argument and negotiation. The arrangement will be as follows. In the cabinet from which you make each test call, you will be provided with a set of six cards, showing reproductions of paintings. You will be asked to imagine that a choice is to be made from these for hanging, say, in the Refreshment Club at Martlesham, and during a period of about two minutes before each test call you should arrange all six cards in your own order of preference, and write the six identifying numbers, in this order, on the form provided. Your partner will also have an identical set of pictures, and his order will almost certainly differ somewhat from yours. The purpose of the conversation is to *negotiate* with your partner to arrive at an agreed compromise order of the pictures that satisfies both of you. It is important that the agreed list should again include all six pictures; you should then enter this list on the form. To make the situation realistic, one person will be nominated as the caller and will dial the number given on the form. At the conclusion of the conversation you will replace the handset and enter your opinion of the connection on the form. You will then be called individually by the operator to answer a few additional questions about the connection.

Subsequent conversations will be similar, but using different sets of pictures.

The complete test will require three visits, during each of which you will make three calls.

APPENDIX 2

(to Annex 2 of Question 4/XII)

Opinion form

1. Before you commence your call, would you please insert below your order of preference for the picture cards. For identification, please use the serial numbers on the backs.

	1st	2nd	3rd	4th	5th	6th
Your order of preference						

2. The green light indicates that it is your turn to originate the call.

If the green light is on, please call your companion as soon as you are ready by dialling . . .

If the green light is off, it is your turn to be called.

3. Please insert below the order of preference arrived at after discussion with your companion.

	1st	2nd	3rd	4th	5th	6th
Agreed order of preference						

4. When you have completed the conversation, please replace the handset.

5. Please mark by a cross, your opinion of the telephone call you have just had.

N.B. Please do not discuss your opinion with your companion.

Excellent	Good	Fair	Poor	Bad

6. You will then be called by the operator who will ask you to complete the following:

Did you, or the person who spoke to you, have any difficulty in talking or hearing over the connection?

YES	
NO	

If YES, please explain the nature of your difficulty to the operator.

ANNEX 3

(to Question 4/XII)

Effect of noise and reference equivalent on transmission performance

(Contribution by the American Telephone and Telegraph Company)

Introduction

This contribution describes an experiment which was performed to study the subjective effects of noise and reference equivalent on actual telephone calls made within the Murray Hill location of Bell Telephone Laboratories. The effects of noise and reference equivalent on talker volume as well as on transmission quality ratings are presented.

Method—Experimental conditions

Four levels of noise and six levels of overall reference equivalent were used in this study. Each level of noise was combined with each value of reference equivalent, thus making a total of 24 experimental conditions. The noise and reference equivalent were the same for the two directions of transmission during an experimental call.

White noise, band-limited to 300–3300 Hz, was used. The average basic circuit noise was —90 dBmp at the line terminals of the telephone sets which was measured by replacing the transmitters and receivers by 90 and 150 ohm resistors respectively. With the transmitters and receivers in place, the average basic circuit noise was approximately —73 dBmp. Three other values of noise (—66 dBmp, —58 dBmp and —50 dBmp) were also tested by adding noise to the circuit. (In the remainder of this contribution the noise values will be expressed as noise at the input of a set with 0 dB receiving reference equivalent by adding 3 dB to each of the values given above.)

The six values of reference equivalent were obtained by means of adjustable attenuators inserted between the subscriber lines. The minimum overall reference equivalent was estimated as approximately 6 dB. For the other conditions attenuation was added in 5 dB steps to provide reference equivalents of 11, 16, 21, 26, and 31 dB.

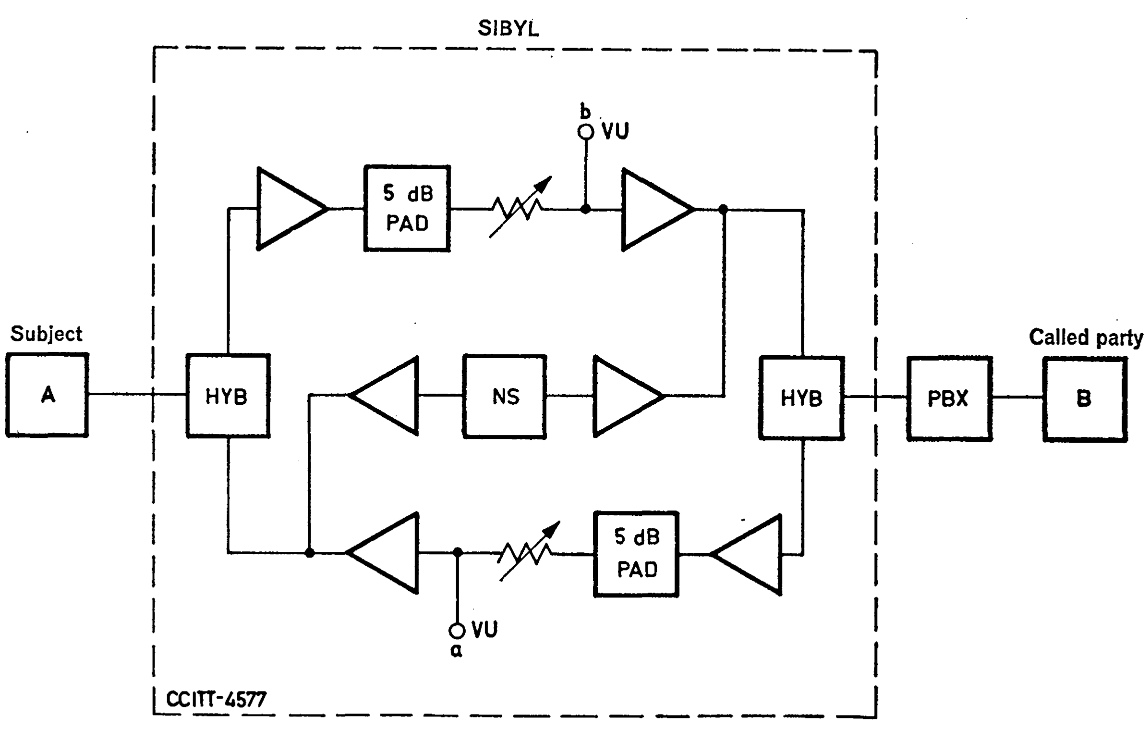
The experimental system

Figure 1 presents a block diagram of the experimental system used in this study. Each experimental call originated at the subject's telephone set denoted by A in the diagram and terminated, via the system called SIBYL, at the called party's set denoted by B. SIBYL automatically introduced a predetermined condition of noise and reference equivalent into the circuit during each experimental call.

SIBYL was able to process only one call at a time so that when it was busy all calls made by the subjects went through their normal circuits, i.e. circuits without any added noise or attenuation. Experimental calls thus comprised only a portion of normal business calls placed by the subjects. All subjects used 500-type telephone sets which were provided with a current of 50 mA. The loops comprised 5 kft (1525 m) of 26-gauge (0.405 mm) non-loaded cable. The receiving reference equivalent of set and subscriber line was estimated at 0 dB and the transmitting reference equivalent as 7 dB. A Northern Electric vu-meter was used to make electrical measurements of the subjects' talking volume. At no time did anyone listen to any of the conversations.

Sidetone loss

It was observed before the experiment that due to impedance changes, the introduction of any loss in the SIBYL circuit would significantly change the normal sidetone path loss function of 500-type handsets, especially around 1.5 kHz. This was corrected by introducing into the system a network which effectively controlled the im-



NS = Noise generator

FIGURE 1. — **A block diagram of the experimental system**

pedance of the subject's loop as seen from his telephone set. Sidetone reference equivalent is estimated as approximately 17 dB.

Room noise

All test calls were made in the normal office environment at the Murray Hill Laboratories. Room noise was approximately 45 dB (A).

Subjects

A random sample of 60 male and female subjects was chosen from the members of Bell Telephone Laboratories at Murray Hill, New Jersey. All subjects agreed to take part in the experiment, prior to its start.

Procedure

There were two main stages in the experiment; Stage A—the pre-experimental stage, and Stage B—the experimental stage. In Stage A no added noise or attenuation was used. The normal talking volume of the subjects was measured using a volume indicator (vu-meter) for a period of 12 weeks.

At the beginning of Stage B all subjects were notified that the experiment was about to begin. During this period which lasted for 13 weeks, SIBYL introduced into the subjects' telephone lines the 24 experimental conditions in random order, according to a preset schedule.

An experimental call occurred when one of the 60 subjects made a telephone call to any person within the Murray Hill Laboratories, and the call was routed through SIBYL. The talking levels of both the calling and called parties were independently measured by two trained vu-meter readers who were calibrated against each other during training sessions.

At the end of each experimental call, the subject received a telephone ring at which time he was required by previous instructions to give a rating of the overall transmission quality according to the following scheme:

Dial 9 for *excellent*; Dial 8 for *good*;
Dial 7 for *fair*; Dial 6 for *poor*.

If at any time *during* the experimental call the circuit was so unsatisfactory that the subject did not want to continue to talk over the circuit, he could get his normal circuit back by simply dialing the digit 6, without hanging up. These calls were classified as *unsatisfactory* in the final data analysis.

Results

Talker volume

Table 1 below presents the normal talking levels of the subjects for the two stages of the experiment. The average vu and the standard deviations are shown.

TABLE 1. — TALKING LEVELS (GROUP AVERAGE) OF SUBJECTS FOR TWO STAGES OF THE EXPERIMENT

	Stage A	*Stage* B
Average	−16.2	−18.9
Standard deviation	4.2	4.1

The numbers shown correspond to talker volume at the line terminals of the subjects' telephone sets. Talker volumes at the output of a set with 0 dB transmitting reference equivalent would be approximately 3 dB higher.

For Stage B the data shown in Table 1 are for the "reference" condition, that is, for "no added noise or attenuation" condition. The subjects, however, lowered their normal talking level in Stage B by about 3 dB although there was no change in the basic noise-loss conditions between Stages A and B for these data. We assume here that since the subjects had to raise their normal talking level to compensate for some of the high noise-loss conditions in Stage B, they slightly lowered their normal talking levels to accommodate for the wide range of experimental conditions in the experiment.

The effects of overall reference equivalent and circuit noise on talker volume in Stage B can be seen in Figure 2. The subjects' talking volume as a function of reference equivalent is shown for the two extreme noise conditions, i.e. —87 and —47 dBmp. Two general observations can be made from the data presented in Figure 2.

1. Talker volumes vs reference equivalents are approximately linear. For 25 dB change in total reference equivalent talker volume increased by about 6.5 dB.
2. Circuit noise affected talker volume less than reference equivalent. Over the whole range of circuit noise between —87 and —47 dBmp (between —70 and —47 dBmp with transmitters and receivers in place) considered here, talker volume increased by only about 3 dB.

Talking volume data of the called persons were also analyzed. The results were very similar to those obtained with subjects' talking volume.

Rating data

Table 2 presents the rating data in the following manner. For various levels of noise and overall reference equivalent percentage ratings are shown for the following five rating categories: excellent, good, fair, poor and unsatisfactory. The number of calls rated for each condition are also shown. Mean opinion score has been calculated by assigning numerical values of 4, 3, 2, 1 and 0 to the five rating categories.

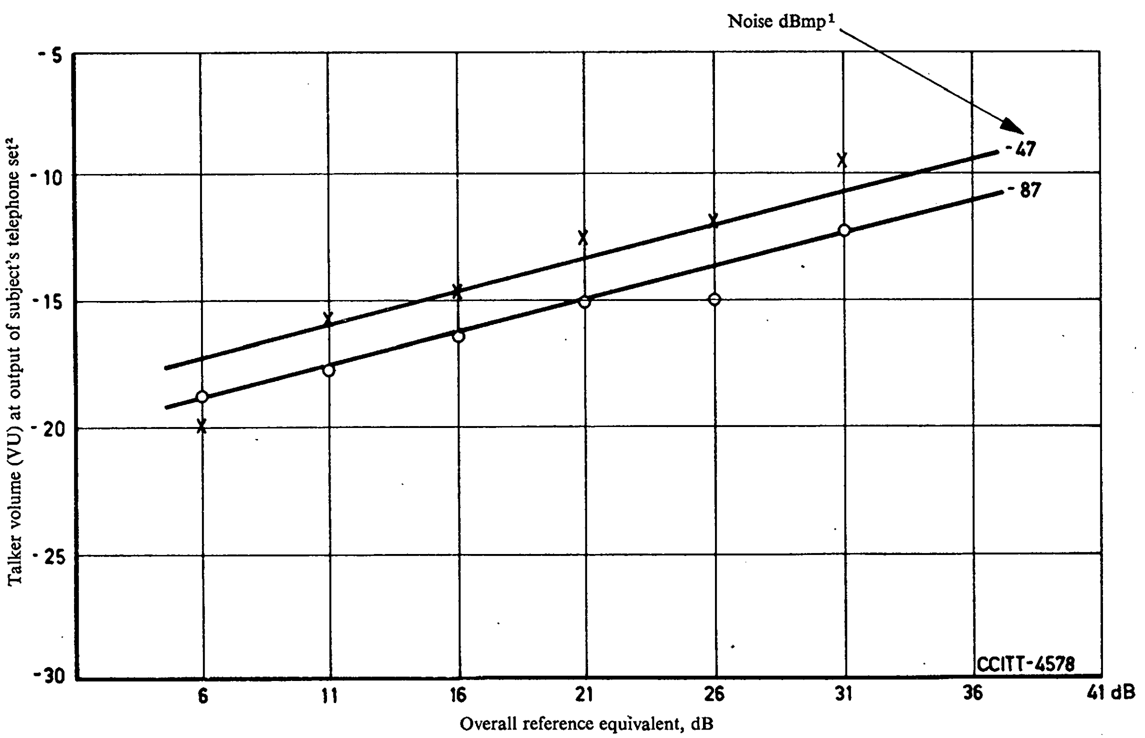


FIGURE 2. — Average talker volume as a function of overall reference equivalent and circuit noise

[1] At input to set with 0 dB receiving reference equivalent.
[2] Talker volume at output of set with 0 dB transmitting reference equivalent would be approximately 3 dB higher than values shown.

TABLE 2

RATING RESULTS

Test condition			Subjective opinion percent					Mean opinion score
Noise dBmp	Overall reference equivalent	Number of calls	E	G	F	P	U	
−87	6	35	60.0	11.4	14.3	14.3	0.0	3.17
−87	11	61	63.9	26.2	3.3	6.6	0.0	3.48
−87	16	33	66.7	21.2	12.1	0.0	0.0	3.55
−87	21	29	41.3	27.6	24.1	3.5	3.5	3.00
−87	26	35	34.3	14.3	31.4	8.6	11.4	2.51
−87	31	12	8.3	25.0	33.3	16.7	16.7	1.92
−63	6	12	50.0	33.3	16.7	0.0	0.0	3.33
−63	11	36	66.7	30.5	0.0	2.8	0.0	3.61
−63	16	33	54.5	30.3	9.1	6.1	0.0	3.33
−63	21	43	25.6	30.2	18.6	16.3	9.3	2.47
−63	26	32	25.0	15.6	37.5	12.5	9.4	2.34
−63	31	12	8.3	8.3	33.3	16.8	33.3	1.42
−55	6	24	54.2	20.8	16.7	8.3	0.0	3.21
−55	11	34	64.7	23.5	5.9	2.9	2.9	3.44
−55	16	27	33.3	29.6	22.2	14.8	0.0	2.81
−55	21	23	21.7	8.7	34.9	21.7	13.0	2.04
−55	26	29	13.8	3.5	17.2	34.5	31.0	1.34
−55	31	23	0.0	13.0	26.1	34.8	26.1	1.26
−47	6	20	15.0	30.0	25.0	20.0	10.0	2.20
−47	11	34	17.6	26.5	11.8	20.6	23.5	1.94
−47	16	28	17.9	25.0	28.5	14.3	14.3	2.18
−47	21	35	8.6	5.7	20.0	40.0	25.7	1.31
−47	26	24	0.0	4.2	25.0	50.0	20.8	1.13
−47	31	11	0.0	0.0	0.0	36.4	63.6	0.36

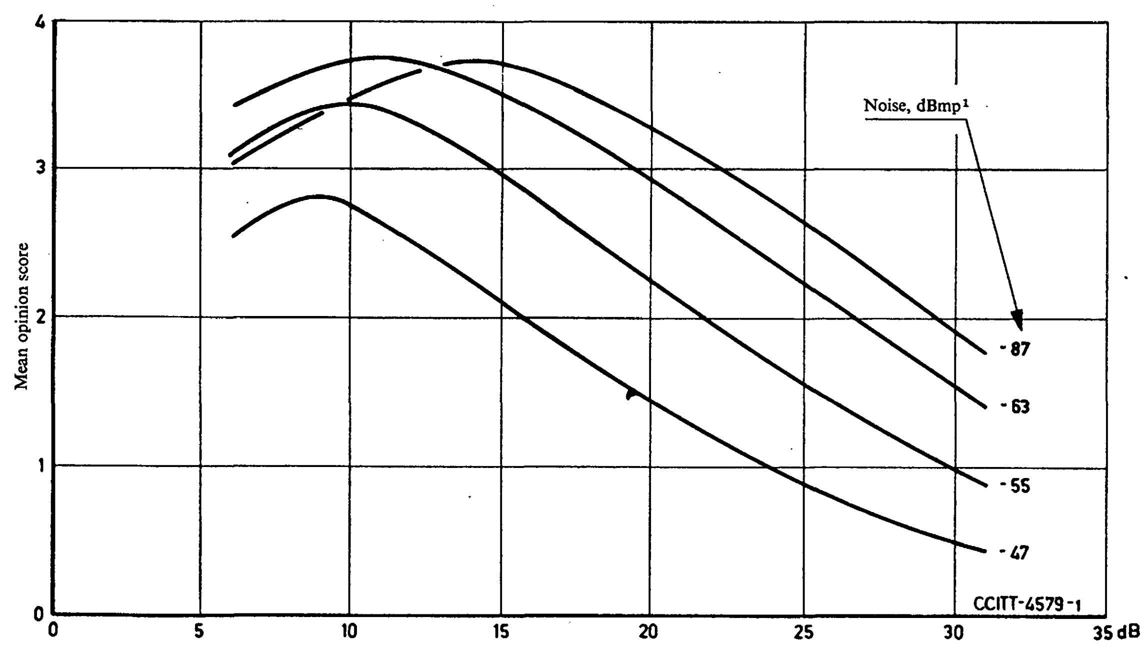


FIGURE 3. — **Mean opinion score**

[1] At input to set with 0 dB receiving reference equivalent.

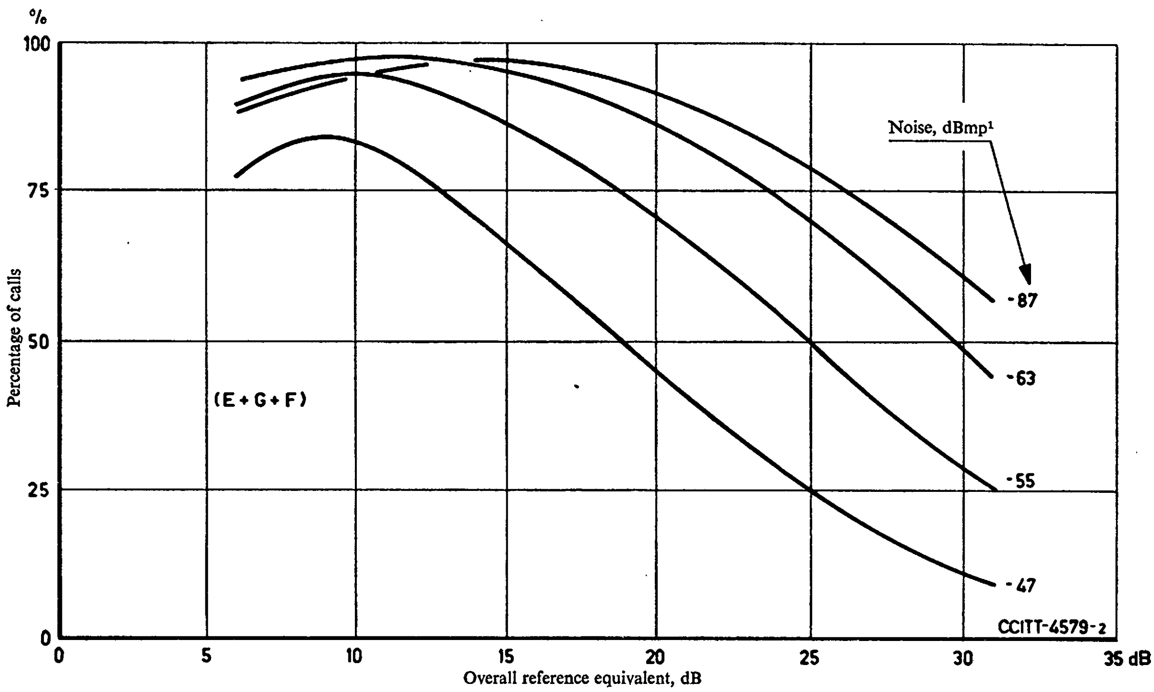


FIGURE 4. — **Percentage of calls rated E + G + F**

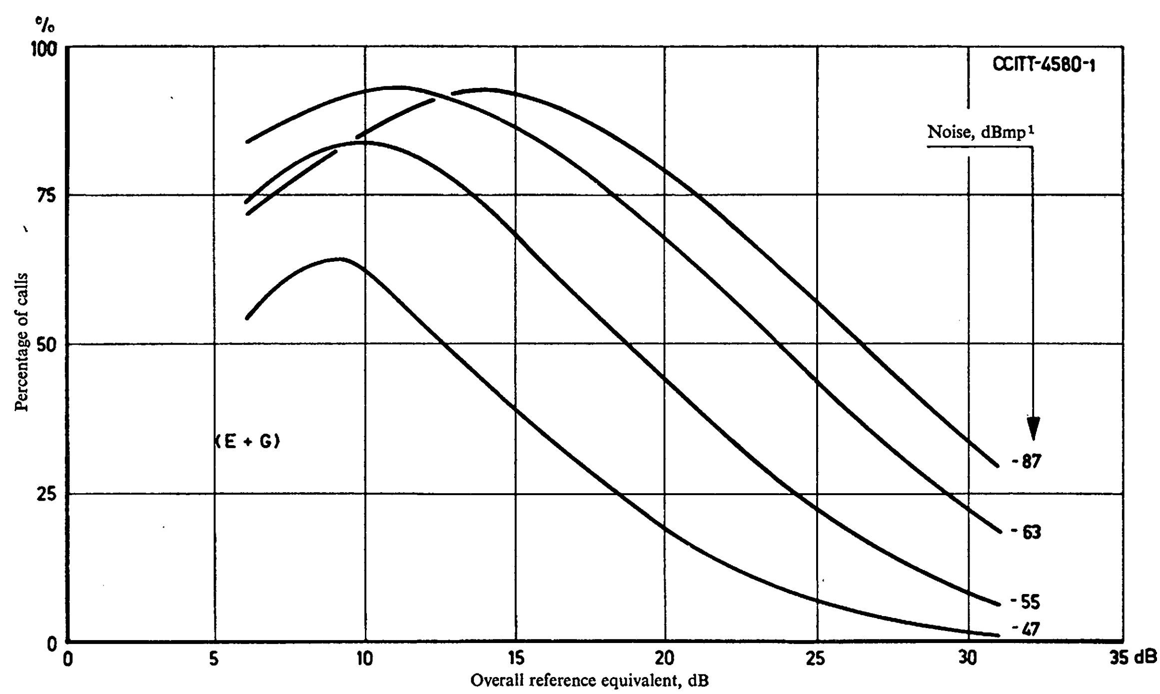


FIGURE 5. — **Percentage of calls rated E + G**

[1] At input to set with 0 dB receiving reference equivalent.

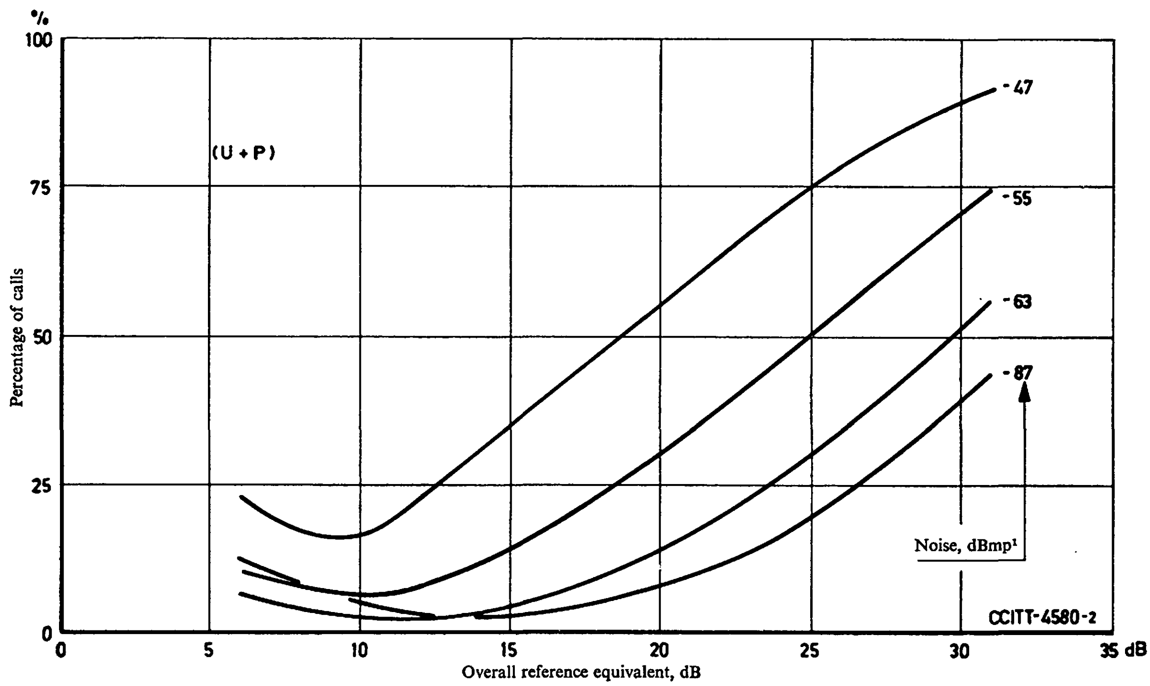


FIGURE 6. — **Percentage of calls rated U + P**

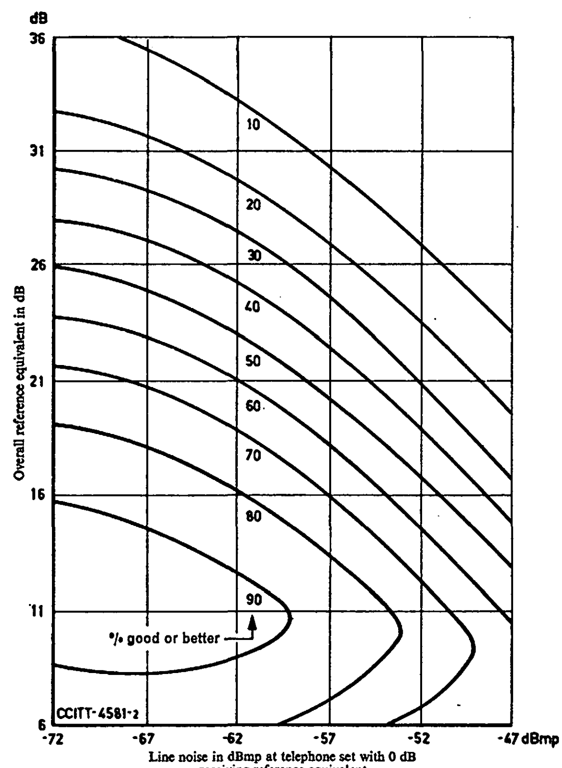


FIGURE 7. — **Good or better contours**

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¹ At input to set with 0 dB receiving equivalent.

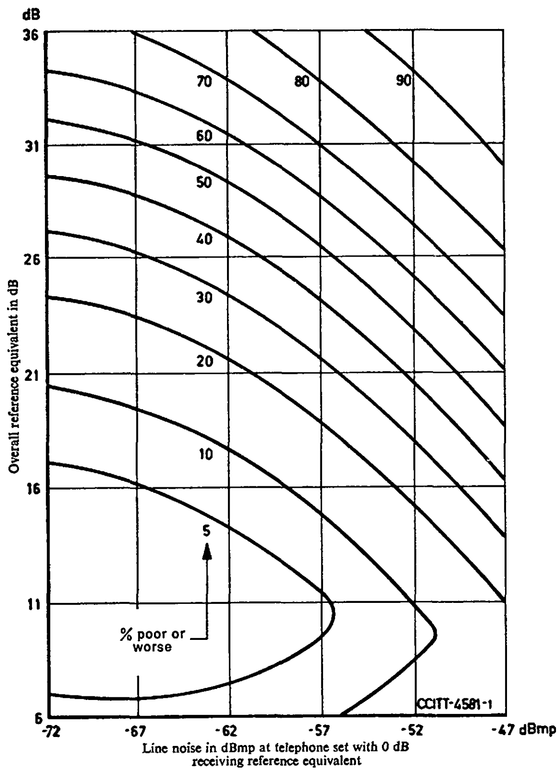


FIGURE 8. — **Poor or worse contours**

Curves have been fitted to the data and are shown in Figures 3 to 6 for mean opinion score; percent fair, good and excellent; percent good and excellent; and percent poor and unsatisfactory as a function of the overall reference equivalent with noise as a parameter.

Figures 7 and 8 present the same results as contours of constant percent good or better and percent poor or worse as a function of overall reference equivalent and noise at the input to a telephone set with 0 dB receiving reference equivalent.

Question 5/XII (8/XVI) — Hourly noise clause

(New question, related to Question 4/C; the results of this study are to be sent to Joint Special Study Group C)

a) Is the mean value of the noise power over any hour suitable for assessing the performance of a telephone connection?

b) If not, which is the most appropriate method for assessing this performance during hours of high noise incidence?

In particular, should a recommendation be made to fix a value for the percentage X as defined in the Note below?

Note. — Joint Special Study Group C draws the attention of Study Groups XII and XVI to C.C.I.R. Report 442. This report is concerned with clause 1.1 of C.C.I.R. Recommendation 393-1 (which corresponds to clause 1.1 of C.C.I.T.T. Recommendation G.222).

Recommendation G.222 has a footnote attached to clause 1.1 and the report is an attempt to solve the problem mentioned in the footnote.

The report proposes tentatively that clause 1.1 of C.C.I.R. Recommendation 393-1 should be modified to read:

"1.1 7500 pW psophometrically weighted hourly mean noise power for more than X% of hours in any month."

The C.C.I.R. is continuing the study of this clause (Resolution 55).[1]

Study Groups XII and XVI may consider the possibility of suggesting a desirable value or range of values for the figure X, the percentage of hours during any month for which the hourly mean noise in radio-relay links could be allowed to exceed 7500 pW (corresponding to 10 000 pW in Recommendation G.222) without undesirable effects in the telephone network.

Question 6/XII — Subscribers' tolerance of echo and propagation time

(Continuation of Question 6/XII, studied in 1968–1972)

a) In the light of the results of opinion tests, what is the mean curve indicating subscribers' tolerance of echo when modern telephone sets are used, and what is the standard deviation of this tolerance's variation, for individual subscribers, in relation to this mean curve?

Note. — Annex 4 contains contributions from various Administrations to this part of the question.

b) Can the method of calculating (and measuring) loss in an echo path, provisionally recommended in Recommendation G.122 B, b) (*White Book*, Volume III) be endorsed as providing an indication of loss which sufficiently corresponds to the subjective effect produced by speech echo in that path?

c) Does Recommendation G.161 (*Green Book*, Volume III) sufficiently control the quality of telephone transmission on connections fitted with echo-suppressors complying with the Recommendation?

d) What is the effect on transmission performance of the following factors in telephone connections having mean one-way propagation times of 150 ms and upwards?

1. The presence of several interconnected circuits, each having a separate pair of echo suppressors. Information is particularly needed for the case of three or more such circuits, with attention to the specific cases defined by Study Group XVI in Annex 1.
2. Interaction between end-delay and return loss. This should be established by co-operation between Study Groups XII and XV for a range of values including 36-ms round trip end-delay.
3. The effects of asymmetry at the echo suppressors of speech levels in the two directions of transmission.
4. Higher return losses that might be achieved by special techniques.
5. Echo control devices of different types at the two ends of the international circuit. Both types should, of course, comply with the specification prepared by Study Group XV.

Note. — Annex 2 below shows the progress made on this question in the 1964–1968 and 1968–1972 study periods. Annex 3 contains some explanations concerning the path a-t-b.

[1] See C.C.I.R. Progress Report of the Interim Working Party 9/1, pages XVI to XXI of the Conclusions of the Interim Meeting of Study Group 9 (Fixed service using radio-relay systems), Geneva, 5–21 July 1972, Part I.

e) What characteristics of echo-cancelling devices need to be specified to ensure satisfactory telephone transmission quality on connections to which they are fitted at one or both ends?[1]

1. for connections with MOPTs less than 400 ms.
2. for connections with MOPTs greater than 400 ms.

Note. — Annex 5 contains contributions from two Administrations to this part of the question.

ANNEX 1

(to Question 6/XII)

Studies to be pursued within the framework of point d) 1 of Question 6/XII

Study Group XVI has defined, in Figures 1, 2a and 2b, a number of circuit and echo suppressor layouts that might be encountered in actual service and that might create difficulties.

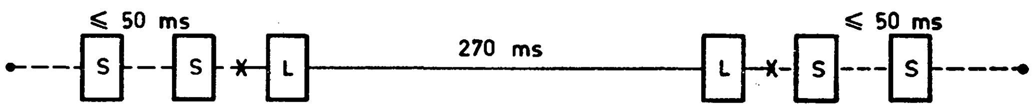


Figure 1

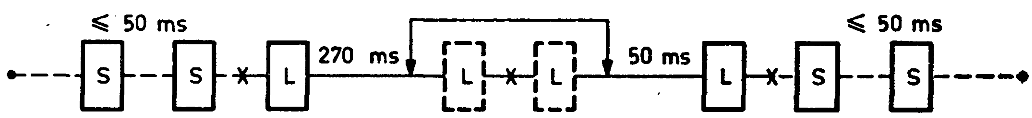


Figure 2a

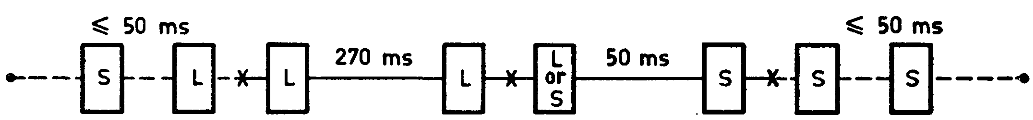
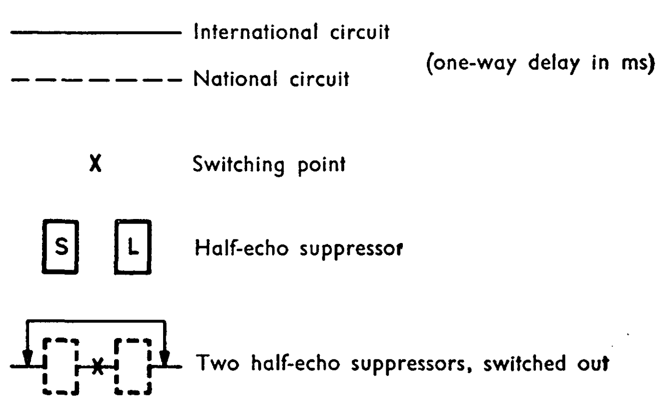


Figure 2b

C.C.I.T.T. 2371

Legend



[1] These techniques are described in the following articles:
Becker, F. K. and Rudin, H. R.: B.S.T.J., Brief — Application of automatic transversal filters to the problem of echo suppression; B.S.T.J., December 1966, Volume XLV, No. 10, pp 1847–1850.
Sondhi, M. M.: An adaptive echo canceller; B.S.T.J., March 1967, Volume XLVI, No. 3, pp. 497–511.
Thies, A. W. and Zmood, R. B.: New ways of echo suppression; *Australian Telecommunication Research*, November 1967, Volume 1, Nos. 1 and 2, pp. 14–19.

In these figures:

L is a half-echo suppressor designed to work with long propagation times such as are likely to be found on a high-altitude satellite circuit—for example, the American type 3A (*Blue Book*, Volume III, Recommendation G.161),

S is a half-echo suppressor conforming to the C.C.I.T.T. specification in force in 1966, or a modern half-echo suppressor for moderate propagation times—for example, a simplified version of type 3A.

Note 1. — In a national network two half-echo suppressors may be replaced by one full echo suppressor.

Note 2. — It should be assumed that on each circuit the go-to-return crosstalk is in agreement with Recommendation G.151 D.

In further contributions to the study of point d.1 of Question 6/XII which Administrations will be submitting, they are asked to study the quality of connections with these layouts and in particular the performance of echo suppressors of the various types used in them.

ANNEX 2

(to Question 6/XII)

Extract from the reply of Study Group XII made at the end of the 1968-1972 period

Part a) — *Mean curve for subscribers' tolerance of echo*

Information given in Annex 4 below could not be compared thoroughly with the echo tolerance curves of Figure 2 in Recommendation G.131.B (Volume III, *White Book*) in the time available, in view of the different forms of presentation used. However, it is considered that the usefulness of the latter figure would be improved if the ordinate were in terms of mean reference equivalent of talker echo path (as used in Annex 4) instead of mean talker echo attenuation which excludes the effect of the sending and receiving reference equivalents of the talker's local end.

Further examination of existing data (including former Annex 2, Question 6/XII, *Red Book*, Volume V*bis*) is required to make this adjustment and to take account of any other changes which might be desirable following proper examination of Annex 4 below.

Care should be taken to avoid confusion between the reference equivalent of the talker echo path and the transmission loss of the path a-t-b from the point of view of echo as described in Recommendation G.122 (Figure 1 and section B—Volume III, *White Book*) the latter being only part of the talker echo path[1].

Part b) — *Method of calculating echo balance return loss*

No contributions have been received to justify a departure from the method provisionally recommended in Recommendation G.122 B, b), and that method should continue to be used until further evidence becomes available.

The calculating and measuring methods will, of course, apply to echo paths other than balance return loss, which is the subject of this part of the question[2].

Part c) — *Maximum propagation time for echo suppressors in accordance with Recommendation G.161*

No evidence has been presented to warrant a modification to Recommendation G.114 (P.14) at this stage. It can be concluded from contributions (COM XII-No-11, period 1968–1972) from the United Kingdom Post Office[3] and the A.T. & T. (Annex 5 below) that with high values of echo return loss (ERL), the presence or absence of echo

[1] See Annex 4 to this question.
[2] See Annex 3 to this question for the necessary explanations.
[3] Material found in Richards, D. L., *Telecommunication by Speech*, Butterworths, 1973.

suppressors has no noticeable effect on the results, irrespective of overall reference equivalent (ORE) or of propagation time up to mean one-way propagation time (MOPT) = 600 ms; this implies that mutilation caused by echo suppressor action during double-talks is unimportant (for high values of ERL). This confirms earlier work by the United Kingdom Post Office. This finding indicates that echo suppressors operate less and less satisfactorily as the ERL decreases, and raises the question as to whether it might be possible to use echo cancellers in conjunction with echo suppressors to accomplish significantly better performance than that possible with the "conventional" echo suppressor (Recommendation G.161, *Green Book*, Volume III).

Current developments in echo control devices may lead the way to an extension of acceptable propagation time for telephone connections, and a relaxation of the restrictions of Recommendation G.114 (P.14) A, c). A necessary condition for such a relaxation will be the assurance of a high reference equivalent in the talker echo path. Designers of new echo control devices should note that there will be a tendency for overall reference equivalents in international connections to decrease over the next few years, and that connections having a reference equivalent from subscriber to subscriber of less than 10 dB may be encountered.

Subjective comparisons of the performance of six pairs of echo suppressors generally conforming objectively with the requirements of *Green Book* Recommendation G.161 have been conducted using circuits simulating single hop satellite connections (see Supplement No. 3 in Volume III). The degree of conformance to Recommendation G.161 was determined by conducting objective measurements in accordance with the procedures set forth in the Annex to the Recommendation. The results indicate that the difference in performance, as determined by subjects conversing over the simulated connections, were not statistically significant. This test also included an echo canceller equipped circuit as a test condition and this circuit performed significantly better than any of the echo suppressor equipped circuits. Analysis of the conversations was performed by the use of a 7 × 3 Youden square design wherein seven test conditions are ranked, three at a time, in a 1, 2, 3 order. Two repetitions of the Youden square were used in this experiment. It is possible that tests involving a larger number of exposures, i.e. more conversations by test subjects, may produce statistically significant differences. Administrations are invited to conduct similar observations and to make results and opinions available to Study Group XII.

Study Group XII notes that an increase in the break-in hangover time has been shown to be beneficial when used on satellite circuits. Before the Study Group can approve the general adoption of the modification, it requires an assurance that the transmission quality of echo-suppressed connections having shorter propagation delays will not be degraded by the increased hangover time. It awaits further evidence on this aspect.

Part d) — *Transmission performance on long-delay connections*

1. *Echo-suppressed circuits in tandem*

No contributions have been received. The attention of Administrations, etc., is drawn to the need to study the performance of three or more echo-suppressed circuits in tandem (see Annex 1 to this question).

2. *End-delay and return loss*

Study Group XII endorses the conclusion of the Laboratory Working Party that Administrations should conduct the necessary tests, required by Study Group XVI, which are described in the Annex to the preliminary reply of Study Group XII to Question 6 given at the Melbourne (1970) meeting (COM XII-No. 49/COM XVI-No. 26, pp. 27–29). No contributions have been received.

3. *Asymmetry of speech levels*

No contributions have been received.

4. *Higher return losses by special techniques*

Annex 5 below and reference [1] describe experiments which suggest that for high values of reference equivalent in the talker echo path, a mean one-way propagation time of up to 600 ms should provide commercially acceptable quality. It should be noted that the experiments were on circuits having overall reference equivalents higher than might be expected on many international connections in the future.

5. *Echo suppressors of different types at opposite ends*

No contributions have been received.

An additional point

With reference to the Document reproduced in Annex 5 below, Study Group XII expressed interest in the test results of the Rio de Janeiro-Tokyo double-hop circuit but felt that there was insufficient information regarding the circumstances under which the test was conducted to make an adequate evaluation of the significance of the results.

Supplements Nos. 1–6 (*White Book*, Volume V) to Recommendations P.14 (G.114) relate to this question.

REFERENCE

[1] WILLIAMS, G. and MOYE, L. S.: Subjective evaluation of unsuppressed echo in simulated long-delay telephone communications, *Proc. I.E.E.*, Vol. 118, No. 3/4, March/April 1971, p. 401.

ANNEX 3

(to Question 6/XII)

Explanation of terms associated with the path a-t-b

(Contribution of the United Kingdom Post Office)

a) *Return loss*

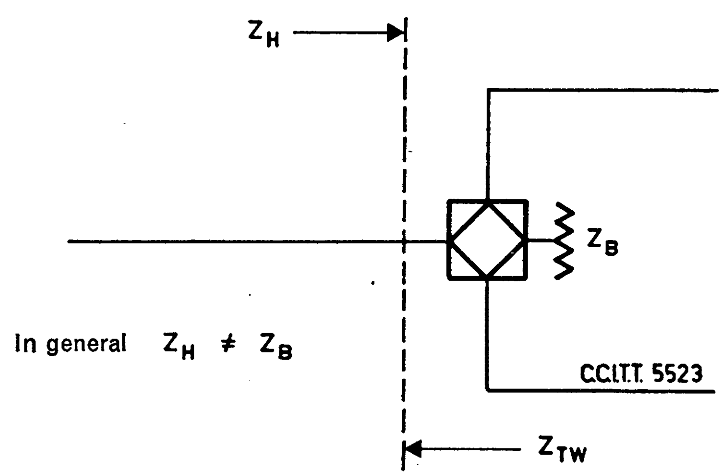
This is a quantity associated with the degree of match between two impedances and is given by the expression:

$$\text{Return loss of } Z_1 \text{ versus } Z_2 = 20 \log_{10} \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| \text{ dB}$$

The use of the expression "return loss" should be confined to two-wire paths supporting signals in the two directions simultaneously.

b) *Balance return loss*

A clear definition is given in the preamble of Recommendation G.122. Figure 1 illustrates the definition.



$$\text{Balance return loss} = 20 \log_{10} \left| \frac{Z_B + Z_{TW}}{Z_B - Z_{TW}} \right| \text{ dB}$$

FIGURE 1

The two-wire portion must be in the condition appropriate to the study, e.g.: if speech echo is being studied the telephone-set must be in the speaking condition.

In the particular case (which occurs very often) in which the impedances presented by each of the paths in the four-wire portion is also Z_B (e.g. 600 ohms) then the terminating-set presents an impedance at the two-wire point which is substantially equal to Z_B. Figure 2 illustrates this case.

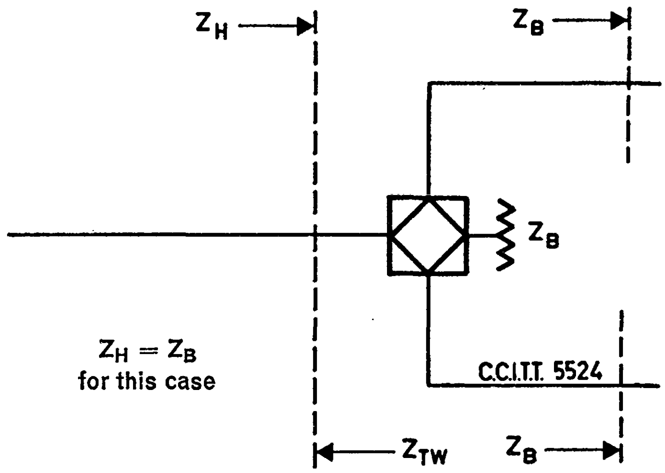


FIGURE 2

The term "balance return loss" (*not* return loss) should always be used for the contribution to the loss of the path a-t-b attributable to the degree of match between Z_B and Z_{TW}.

c) *Transmission loss of the path a-t-b*

There is room for confusion here because the concept can be applied to arrangements in which there is no physical point "t" at all, e.g. as in some laboratory simulations of long connections in which echo is introduced by a controlled unidirectional-path bridging the two four-wire paths. The point "t" is necessary in the recommendation because practical public switched telephone networks are being dealt with.

Thus in general two cases arise.

Case 1: There does exist a point "t" (Figure 3)

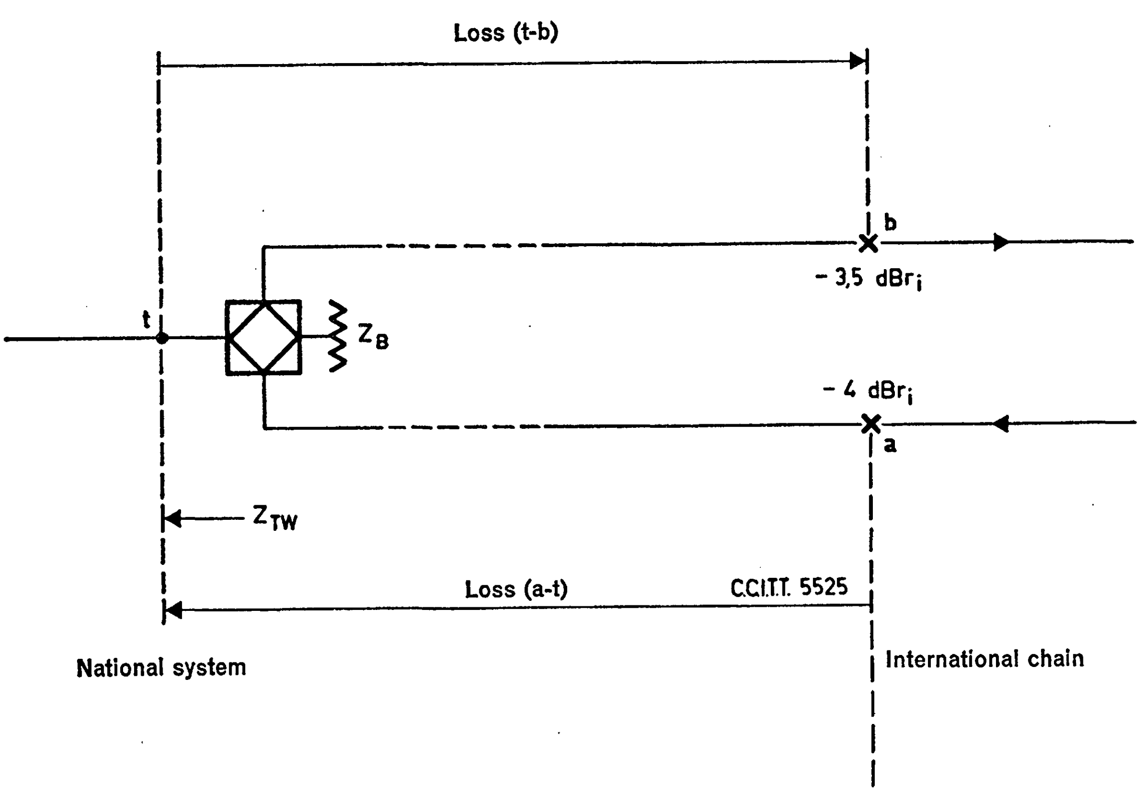


FIGURE 3

The transmission loss of the path a—t—b may be calculated from

$$\text{loss (a—t)} + 20 \log_{10} \left| \frac{Z_B + Z_{TW}}{Z_B - Z_{TW}} \right| + \text{loss (t—b)}$$

The diagram is drawn in terms of the virtual switching points of the international circuit with their associated relative levels. The subscript i in the abbreviation dBr_i signifies that these relative levels are with respect to a 0 dBr point of the international circuit.

It is clear that any other convenient pair of relative levels (differing by 0.5 dB in the correct sense) can be used in practice, e.g.: the actual switching levels used in an international centre.

Case 2: There does not exist any "t" (Figure 4)

This relates particularly to laboratory testing arrangements.

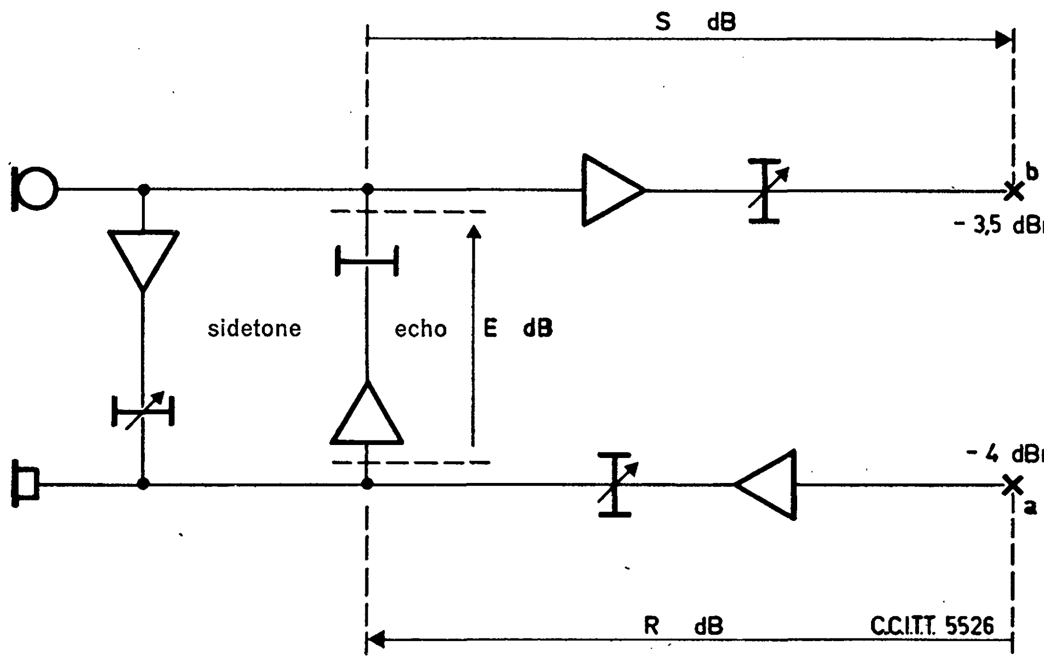


FIGURE 4

In this case the loss of the path a—'t'—b may be calculated from: R + E + S dB (assuming acoustic feedback at the four-wire telephone to be negligible).

In both cases the loss of 'the path a—t—b' can in principle be directly measured by the principles described in the Annex 1 to Recommendation G.122, i.e. by injecting a signal at a and measuring the result at b, so that one may properly say for all cases

$$\left\{ \begin{array}{l} \text{transmission loss} \\ \text{of the path a—t—b} \end{array} \right\} \equiv \left\{ \begin{array}{l} \text{transmission loss} \\ \text{between a and b} \end{array} \right\}$$

or, more shortly

$$\text{loss (a—t—b)} \equiv \text{loss (a—b)}$$

d) *Stability and echo losses*

So far the quantities dealt with are functions of frequency and yield a graph of attenuation/frequency distortion. When it is required to characterise such a graph with a single number additional qualifying phrases are used, e.g. 'transmission loss of the path a—t—b from the point of view of stability (or of echo)'. One may use the shorter expressions: stability loss (a—b) and echo loss (a—b).

Recommendation G.122 gives the definitions of these single-number descriptions thus: the transmission loss from the point of view of stability is the least measured or calculated value in the band 0–4 kHz (Recommendation G.122. A).

Similarly, the transmission loss from the point of view of echo (Recommendation G.122.B) is equal to one quarter of the sum of the power ratios of the following frequencies weighted as indicated:

Hz	Weight
500	½
1000	1
1500	1
2000	1
2500	½

In both cases, in principle, the attenuation distortion of the whole path from a to b should be calculated (or measured), before the stability loss (a—b) or the echo loss (a—b) are calculated in accordance with their respective definitions. In the particular case of echo, however, if (as is often the case) the mid-band distortion over the transmission paths a—t and t—b is relatively small then the averaging process required for the echo calculation need only be carried out on the balance return loss component of the total loss which is then added to the transmission loss (at 800 Hz or 1000 Hz) of the a—b and t—b paths. A balance return loss distortion curve so averaged may be referred to as the 'echo balance return loss in accordance with the definition of Recommendation G.122' or, more shortly, the 'echo balance return loss'.

e) *Reference equivalent of the echo path*

Recommendation G.131 is concerned with complete talker echo paths and it is convenient to characterise this path in terms of reference equivalent. Part b) of Annex 2 above provides a provisional rule as far as the echo balance return loss is concerned, and by convention we may regard the echo balance return loss as the contribution it makes to the overall reference equivalent of the mouth-ear echo path. Naturally, as indicated in Recommendation G.122.B, the echo loss (a—b), when this is already known, may be used instead of the sum of three quantities: — the loss (a—t), the echo balance return loss at t, and the loss (t—b).

Hence the nominal overall reference equivalent of the echo path or, more shortly, the 'echo reference equivalent' may be calculated as illustrated in Figure 5.

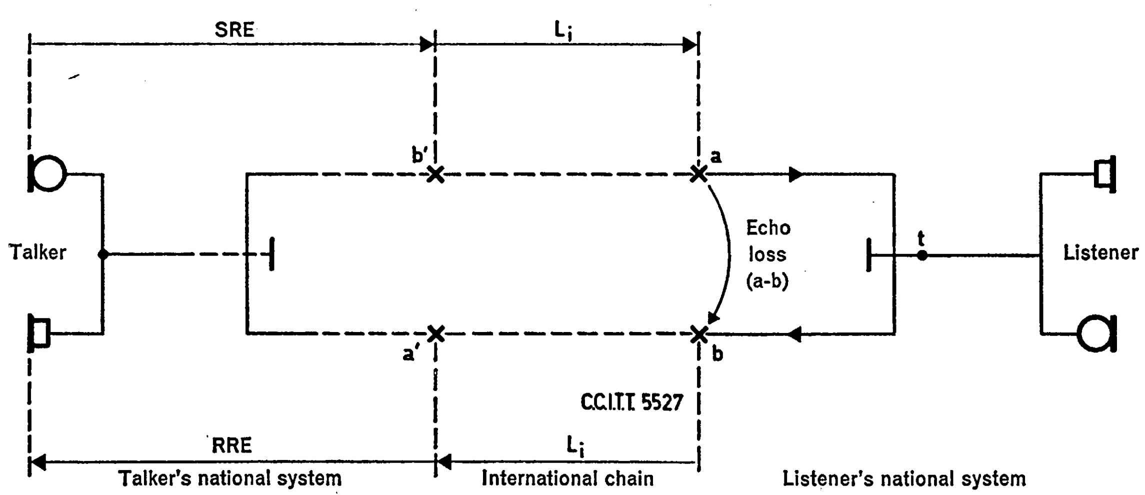


FIGURE 5

Echo reference equivalent =

SRE + RRE of the talker's national system

+

twice the loss of the international chain (i.e.: $2L_i$ at 800 Hz or 1000 Hz)

+

the echo loss (a—b) of the listener's national system (i.e. averaged according to Recommendation G.122).

f) *Résumé of useful terms*

Return loss. — Relates to a two-wire bi-directional circuit; classical definition.

Balance return loss. — Proportion of loss in the a—t—b path attributable to the degree of match between the two-wire impedance and the balance impedance at the terminating unit. Applicable only if there is a point "t".

Transmission loss of the path a—t—b. — Can be regarded as the loss (a—b), whether there exists a physical point "t" or not.

Stability loss (a—b). — The least value of the loss (a—b) in the band 0–4 kHz.

Echo loss (a—b). — The loss (a—b) averaged according to the definition in Recommendation G.122.B.

Echo balance return loss. — A balance return loss averaged according to Recommendation G.122.B.

Echo reference equivalent. — The sum of the sending reference equivalent and receiving reference equivalent of the talker's national system, twice the loss of the international chain, and the echo loss (a—b) of the listener's national system.

ANNEX 4

(to Question 6/XII, Part a)

I

AMERICAN TELEPHONE AND TELEGRAPH CO.

A. — Observer reaction to talker echo

1. *Introduction*

This contribution provides a summary of two laboratory experiments to determine observer assessment of talker echo over a range of delay conditions typical of terrestrial facilities. Experiment I was designed to collect data on talker echo threshold in the presence of various amounts of analogue circuit noise. Experiment II was designed to collect rating data on talker echo in the presence of various amounts of analogue circuit noise. Independent variables for both experiments were echo path delay, echo path loss, and circuit noise level.

2. *Method*

2.1 *Test methods*

In Experiment I the subject adjusted the echo path loss to obtain his threshold of detectability. By means of an attenuator, he was allowed to alternately increase and decrease the echo path loss until he was satisfied that his threshold was determined. (Threshold of detectability was defined for the subject as that attenuator setting at which the talker echo was just perceptible.)

In Experiment II the subject was presented with a number of different experimental conditions and rated each condition on a 5-point rating scale: E = Excellent, G = Good, F = Fair, P = Poor, and U = Unsatisfactory. Each condition comprised a preselected combination of echo path delay, echo path loss and circuit noise level.

The experimenter initiated each test condition by reciting a sentence (from the Harvard list of phonetically balanced sentences) [1] to the subject. The subject repeated the sentences to determine threshold settings for Experiment I or ratings for Experiment II.

2.2 *Test system*

A diagram of the laboratory test system is shown in Figure 1. The system provided a sidetone path, an echo path, and a transmission path from the experimenter to the subject. Transmission from the subject to the experimenter was, unknown to the subject, accomplished by means of an intercom system not shown in Figure 1, thus enabling the experimenter to monitor the test sentences and questions from the subject.

The sidetone path amplitude response was uniform (± 1 dB) with frequency over the band 300–3000 Hz, and incorporated sharp band limiting with 3 dB down points at 200 Hz and 3300 Hz. The echo path was similarly bandlimited. The passive delay equipment used in the tests was such that the echo path amplitude response within the band 300–3000 Hz varied depending on the delay setting: for low delays, the response was uniform within about ± 1 dB while at the highest delay tested the variation was within ± 3 dB. Envelope delay distortion over the band 500–2500 Hz ranged from ± 0.5 milliseconds at the lowest delay tested to ± 5 milliseconds at the highest delay.

The experimenter's talking circuit incorporated an average customer loop and a 500-type telephone set, the present Bell System standard.

2.3 *Experimental conditions*

In Experiment I three circuit noise levels were used, corresponding to −72 dBmp, −62 dBmp, and −52 dBmp at the input to a telephone set with an estimated receive reference equivalent of 0 dB. Echo path delays of 1.5, 5, 10, 20, 26, 36, 46, 56, 72 and 90 milliseconds were tested at each noise level.

There were thus a total of 30 experimental conditions. Each condition was presented twice, resulting in a total of 60 determinations by each subject, requiring six test sessions. For each test session, the circuit noise level was constant and all delays were tested in random order.

In Experiment II the same three circuit noise levels were used. Echo path losses are given in Table 1 for a circuit noise level of −62 dBmp. Echo path losses shown for echo path delays of 1.5, 20, 56 and 90 milliseconds were also tested at noise levels of −72 dBmp and −52 dBmp. A condition without echo was also included at each circuit noise level.

TABLE 1

ECHO PATH LOSSES (dB REFERENCE EQUIVALENT) FOR CIRCUIT NOISE LEVEL = −62 (dBmp)

Round trip delay —RTD (milliseconds)						
1.5	*10*	*20*	*36*	*56*	*72*	*90*
2	4	6	13	16	21	23
7	9	11	18	21	26	28
12	14	16	23	26	31	33
17	19	21	28	31	36	38
22	24	26	33	36	41	43
27	29	31	38	41	46	47

Thus, there were a total of 93 different conditions for Experiment II. These conditions were arranged in random order, then presented in two test sessions.

2.4 *Fixed conditions*

The room noise for both experiments was 35 dB (A) at the subject's location.

The sidetone path loss of the test system was estimated to be about 14 dB reference equivalent.

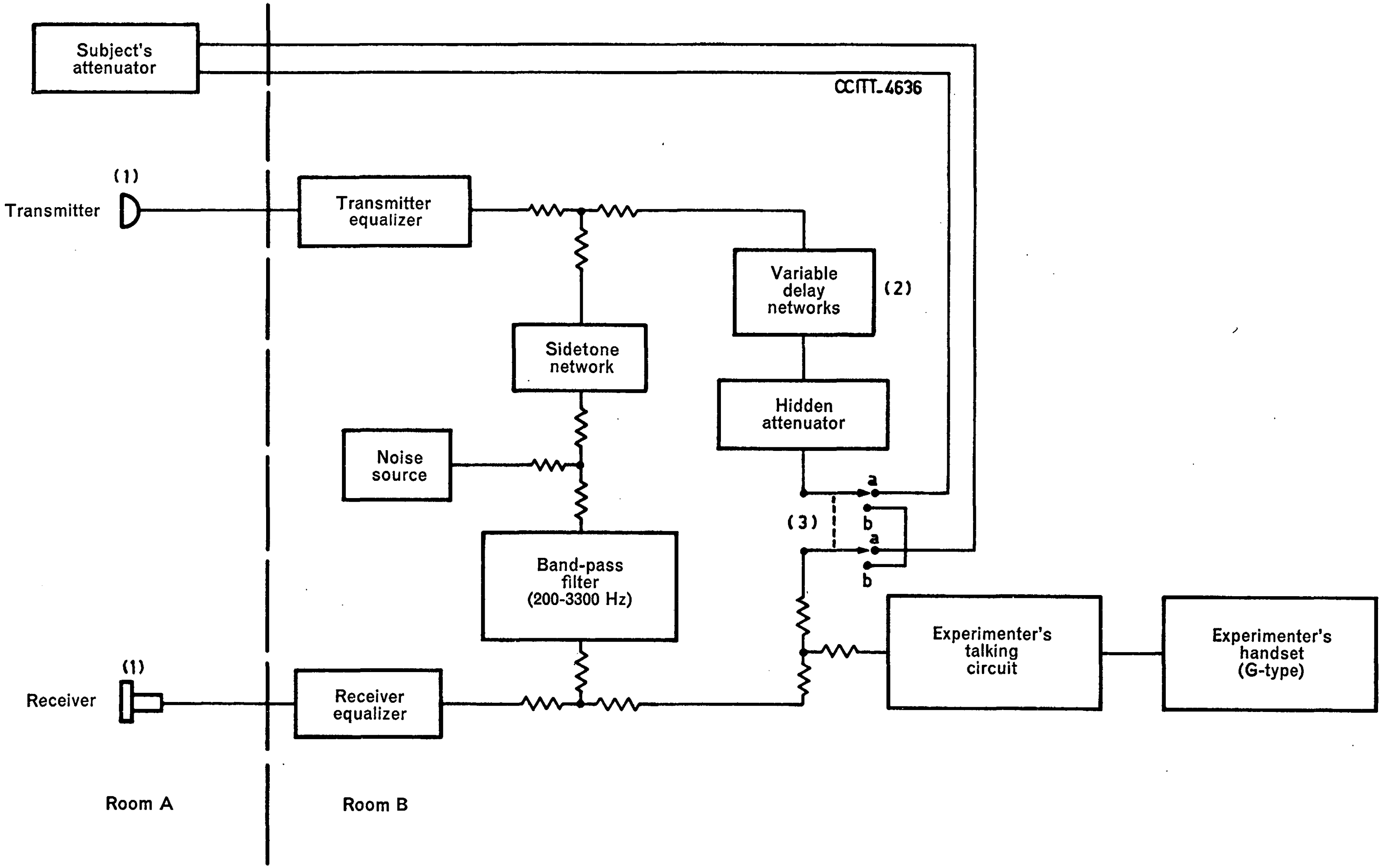
The experimenter-to-subject transmission path had an estimated rating of 15 dB reference equivalent. The experimenter maintained a constant speech level at the subject's telephone set corresponding to —26 vu into a telephone set with a receive reference equivalent of 0 dB.

2.5 *Subjects*

Fifteen subjects, seven male and eight female, participated in Experiment I. Their ages were in the range from 21 years to 41 years. Nine of these subjects also participated in Experiment II.

Thirty subjects, fifteen male and fifteen female, participated in Experiment II. Their ages were in the range from 20 years to 56 years.

Subjects were selected on the basis of having hearing acuity (for their age group) in the normal range over the band 500–2000 Hz.



(1) Linear transducers installed in the G-type handset used by the subject.
(2) Includes a bandpass filter, 200-3300 Hz.
(3) Switch in position "a" for Experiment I, position "b" for Experiment II.

FIGURE 1. — **Block diagram of the talker echo test system**

3. *Results and discussion*

Thirty data points comprising echo path losses for detection of talker echo were obtained for each of the combinations of echo path delay and circuit noise level. Means and standard deviations for each combination are given in Table 2.

Table 2 supports the following observations:

1. Threshold echo path loss increased by about 32 dB for an increase in echo path delay from 1.5 milliseconds to 90 milliseconds;
2. Threshold echo path loss increased by about 0.7 dB for each 10 dB increase in noise level;
3. Standard deviations which range from 3.0 dB to 5.2 dB did not depend on noise level or echo path delay in a systematic manner.

The data was further analyzed assuming that the mean echo path loss was independent of noise level and the standard deviation was independent of noise level and echo path delay. The model resulting from this analysis had a mean echo path loss described by the exponential function

$$[14.72 + 40.31\,(1—e^{-0.016\,D})]\ \mathrm{dB}$$

(where D = echo path delay in milliseconds) and a standard deviation of 3.94 dB.

Results of the rating tests are given in Table 3 for a circuit noise level of —72 dBmp; these results most nearly reflect effects of talker echo alone. Results obtained for circuit noise levels of —62 dBmp and —52 dBmp were very similar in shape. However, the average ratings were lower as a result of the higher circuit noise levels.

The results for all three noise conditions were used to develop smooth functions for representing all of the data. Curves of mean opinion score resulting from the smoothing operations are shown on Figure 2 together with mean opinion scores computed directly from the data of Table 3.

Per cent good or better and per cent poor or worse contours developed from the rating data are plotted on Figures 3A and 3B respectively. Also included in these figures is the mean threshold computed from the exponential function given earlier.

The mean threshold curve and 50% contours of good or better and poor or worse are plotted on Figure 4 together with a curve plotted from mean echo tolerance values adopted by the C.C.I.T.T.[1]

4. *Results from other tests*

Experiment I was conducted using a fixed sidetone path loss estimated to be about 14 dB reference equivalent. Results of another experiment, not reported herein, indicates that on the average the threshold echo path loss increases by about 0.5 dB for each 1 dB increase in sidetone path loss over the range 9–19 dB sidetone path reference equivalent.

Experiment II to determine ratings was conducted under laboratory conditions. Another experiment, also not reported herein, has recently been completed under conditions more typical of those experienced by customers. This experiment utilized a facility called SIBYL which was used to introduce preselected echo conditions on actual telephone calls within the Holmdel location of Bell Telephone Laboratories. Mean opinion scores obtained from this study were in close agreement with Experiment II results.

References

[1] Egan, J. P.: Articulation Testing Methods, *Laryngoscope*, Vol. 58, No. 9, September 1948, pp. 955–991.

[2] Supplement No. 2: Talker Echo on International Connections, *C.C.I.T.T. Green Book*, Volume III.

[3] Notes on Distance Dialing: Copyright, 1961, by American Telephone and Telegraph Company.

[1] The mean C.C.I.T.T. echo tolerance values are tabulated in Reference [2] for RTDs from 20 milliseconds to 100 milliseconds; a value for 0 milliseconds was obtained from Reference [3]. The curve of mean echo tolerance plotted on Figure 4 was obtained by adding to these values a constant which accounts for the transmit and receive efficiency (in reference equivalent terms) of an average Bell System customer loop.

TABLE 2

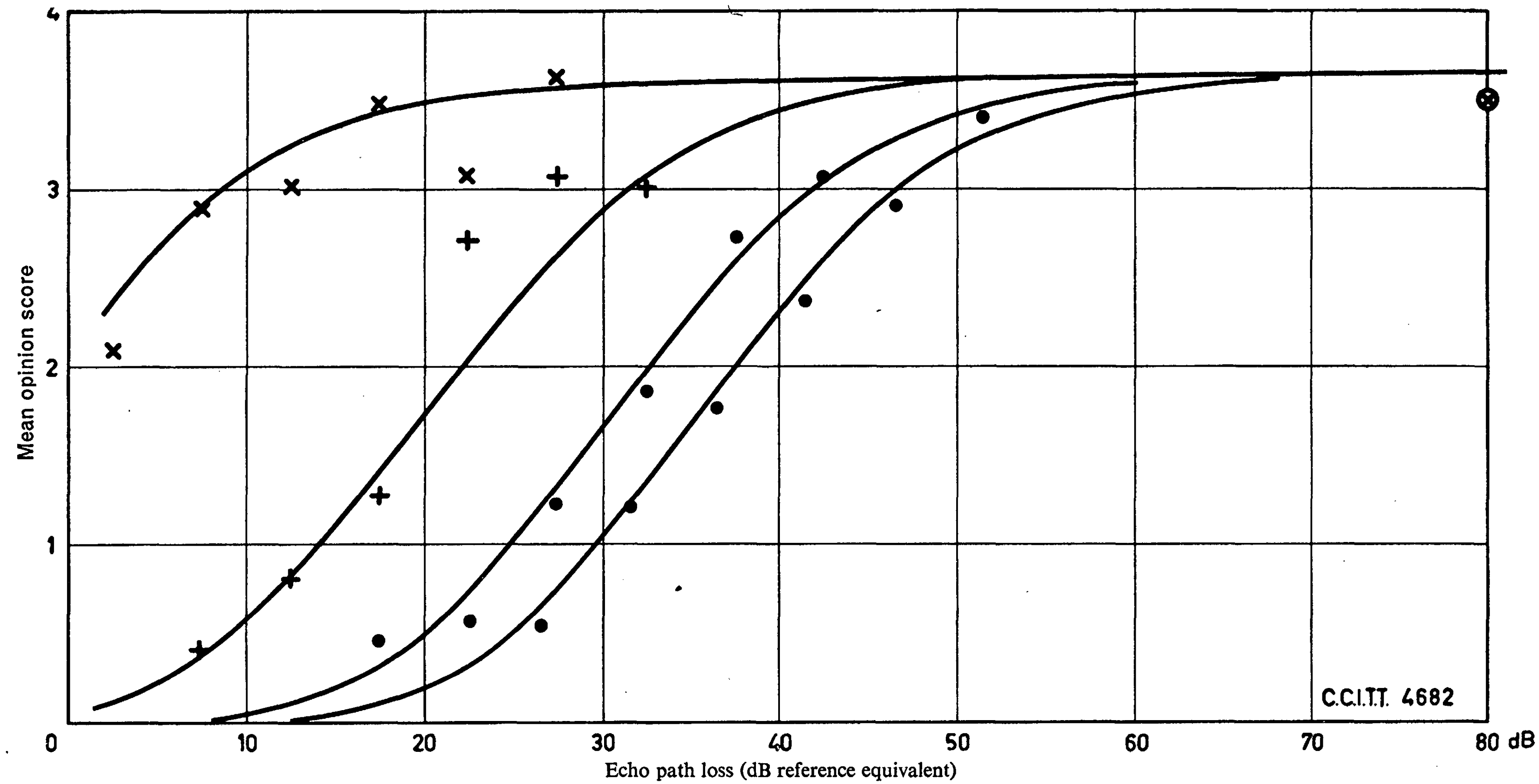
RESULTS OF EXPERIMENT I

Talker echo detectability thresholds

Statistic	Noise (dBmp)	Round-trip delay — RTD (milliseconds) 1.5	5	10	20	26	36	46	56	72	90
Mean [a]	—72	15.0	20.9	21.7	25.8	30.8	33.2	36.7	38.6	44.0	47.9
	—62	14.6	19.8	21.3	25.6	29.9	32.9	35.7	37.7	43.2	46.4
	—52	14.0	18.5	21.0	25.2	29.7	31.6	35.2	36.3	42.2	44.1
Standard [b] Deviation	—72	4.1	4.7	3.6	3.3	3.7	3.9	3.0	4.2	3.1	4.2
	—62	3.2	3.7	3.6	3.1	3.2	4.4	3.5	4.3	3.5	4.6
	—52	4.0	3.6	3.9	3.1	4.0	3.8	3.6	4.0	5.2	4.8

[a] Entries are echo path losses in dB reference equivalent.

[b] Standard deviation $= \sqrt{\dfrac{\sum^{n} \left| X_i - \overline{X} \right|^2}{n}}$



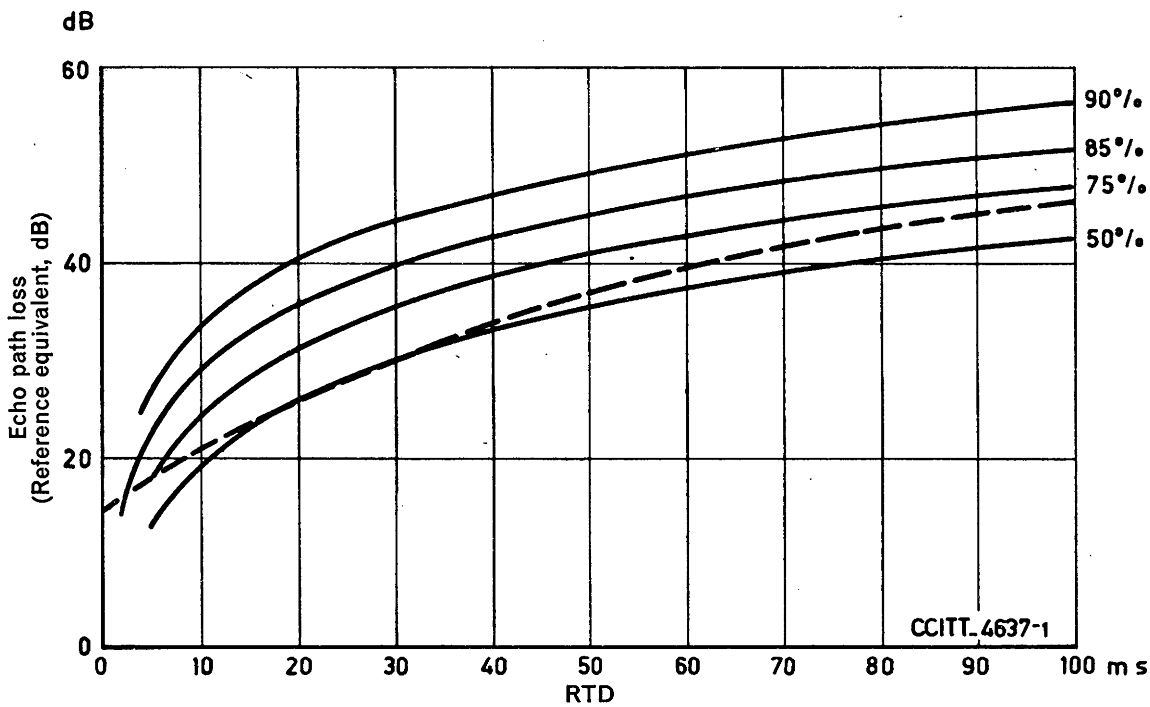
Curves determined from smoothing process applied to all of rating data.
Points computed from raw data of table.
RTD = 0 milliseconds, echo path loss = 100 dB
RTD = 1 millisecond
RTD = 20 milliseconds
RTD = 56 milliseconds
RTD = 90 milliseconds

FIGURE 2. – **Mean opinion scores for CNL = −72 dBmp**

TABLE 3

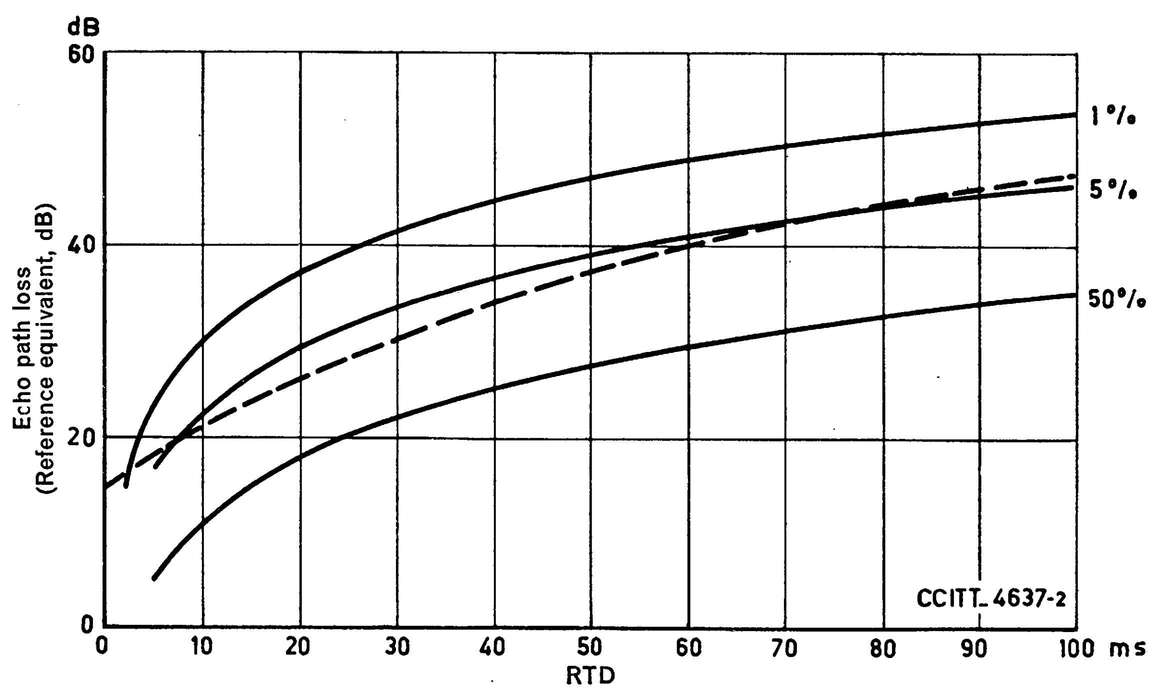
EXPERIMENT II RESULTS FOR CNE = −72 dBmp

Delay (milliseconds)	Echo path loss (dB)	Rating (%)				
		Excellent	Good	Fair	Poor	Unsatisfactory
0	100	63.3	30.0	3.3	0.0	3.3
1.5	2.4	3.3	33.3	36.7	23.3	3.3
	7.4	26.7	43.3	23.3	6.7	0.0
	12.4	33.3	46.7	10.0	10.0	0.0
	17.4	60.0	26.7	13.3	0.0	0.0
	22.4	33.3	46.7	13.3	6.7	0.0
	27.4	63.3	36.7	0.0	0.0	0.0
20	7.5	0.0	3.3	3.3	23.3	70.0
	12.5	3.3	0.0	13.3	40.0	43.3
	17.5	0.0	10.0	36.7	23.3	30.0
	22.5	33.3	30.0	16.7	13.3	6.7
	27.5	36.7	36.7	23.3	3.3	0.0
	32.5	26.7	56.7	6.7	10.0	0.0
56	17.6	0.0	0.0	6.7	33.3	60.0
	22.6	0.0	0.0	10.0	36.7	53.3
	27.6	3.3	6.7	26.7	36.7	26.7
	32.6	3.3	20.0	40.0	33.3	3.3
	37.6	20.0	43.3	26.7	10.0	0.0
	42.6	33.3	46.7	16.7	0.0	3.3
90	26.5	0.0	0.0	6.7	40.0	53.3
	31.5	3.3	10.0	20.0	36.7	30.0
	36.5	3.3	13.3	43.3	36.7	3.3
	41.5	23.3	26.7	16.7	30.0	3.3
	46.5	30.0	40.0	23.3	3.3	3.3
	51.5	50.0	40.0	10.0	0.0	0.0



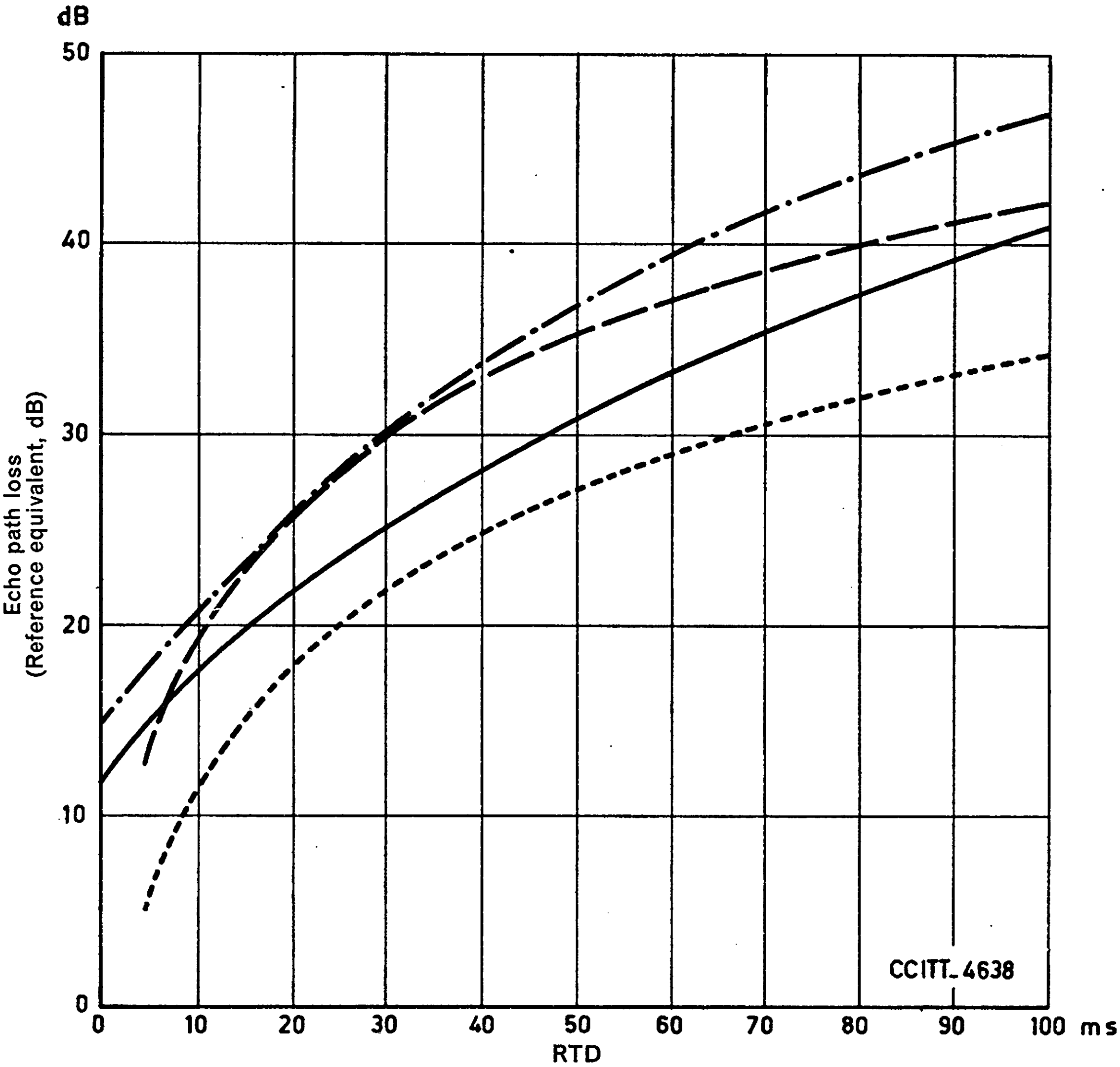
——— = Per cent good or better contours
– – – – = Mean threshold curve

FIGURE 3A. — **Per cent good or better contours and mean threshold curve (CNL = −72 dBmp)**



——— = Per cent poor or worse contours
– – – – = Mean threshold curve

FIGURE 3B. — **Per cent poor or worse contours and mean threshold curve (CNL = −72 dBmp)**



— = Mean C.C.I.T.T. echo tolerance curve
— — — = 50% good or better contour for CNL = −72 dBmp
- - - - = 50% poor or worse contour for CNL = −72 dBmp
—.—.—.— = Mean threshold curve

FIGURE 4. — **Mean C.C.I.T.T. echo tolerance curve and results of present study**

B. — Proposed revision to Recommendation G.131

Limitation of echoes

The main circuits of a modern telephone network providing international communications are high-velocity carrier circuits on symmetric or coaxial pairs or radio-relay systems, and echo suppressors are not normally used except on connections involving very long international circuits. There is often no general need for echo suppressors in national networks but they may be required for the inland service in large countries. Echo suppressors may also be needed on loaded-cable circuits (low-velocity circuits) used for international calls.

Echoes may be controlled in one of two ways; either the overall loss of the four-wire chain of circuits may be adjusted so that echo currents are sufficiently attenuated (which tacitly assumes a particular value for the echo balance return loss) or an echo suppressor can be fitted.

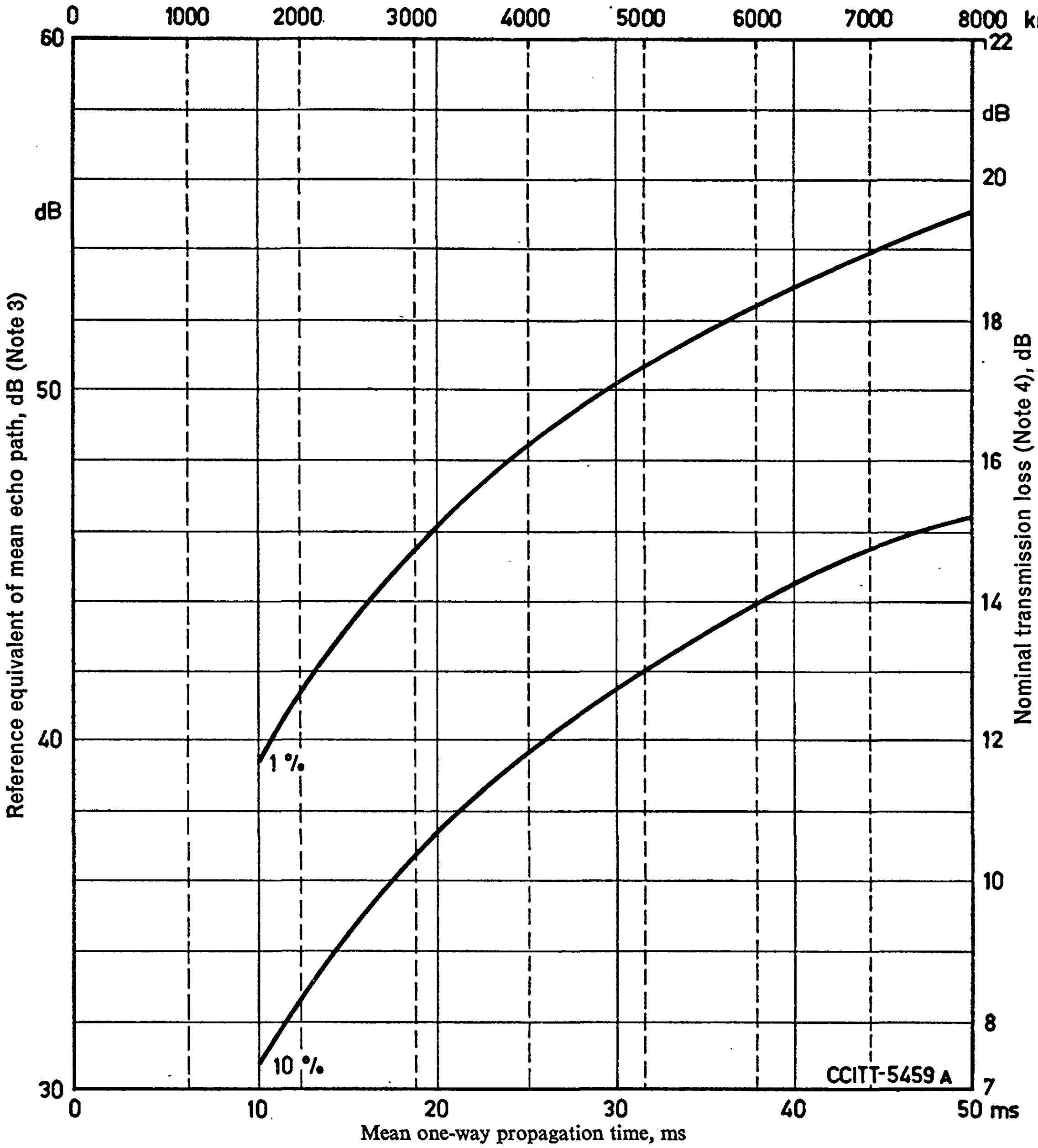


FIGURE 5. — **Echo tolerance curves**

Notes to the Figure

Note 1. — The percentages refer to the probability of encountering objectionable echo and are based on a rating of poor or unsatisfactory on a five-point scale which includes categories of excellent, good, fair, poor, and unsatisfactory.

Note 2. — The distance scales assume a velocity of propagation of 160 km/ms.

Note 3. — The reference equivalent of the mean echo path is here defined as the sum of the mean values of the transmission loss in the two directions of transmission between the two-wire ends of the subscriber lines in the terminal local exchanges, the mean value of the echo return loss at the listener's end, together with the sending and receiving reference equivalents for the telephone set and subscriber line at the talker's end.

Note 4. — The nominal transmission loss is here defined as the nominal transmission loss between two-wire ends of the subscriber lines in the terminal local exchanges assuming (a) that there is no difference between the nominal value and the mean value; (b) that the nominal transmission loss is the same in both directions of transmission; and (c) the mean value of the echo return loss at the listener's end is 11 dB and that the sum of the sending and receiving reference equivalents for the telephone set and subscriber line at the talker's end is 5 dB.

Note 5. — In constructing the tolerance curves, nine four-wire circuits were assumed for the four-wire chain.

Transmission loss adjustment

The curves of Figure 2 of Recommendation G.131 (Figure 5 hereafter) indicate the minimum value of the reference equivalent of the mean echo path that must be introduced if no echo suppressor is to be fitted. The required reference equivalent is shown as a function of the mean one-way propagation time. A distance scale has been added which assumes a velocity of propagation of 160 km/ms; if low velocity plant is used in any part of the connection then the propagation-time scale should be used. Supplement No. 2 to this Volume explains how these curves have been derived.

The curves of Figure 2 of G.131 used with the scale at right also show the nominal transmission loss between the two-wire ends of the subscriber's lines in the local exchanges that must be introduced if no echo suppressor is to be fitted. This scale has been derived assuming that the mean value of echo return loss at the listener's end is 11 dB and that the sum of the sending and receiving reference equivalents for the telephone set and subscriber line at the talker's end is 5 dB.

The curves are applicable to a chain of circuits which are connected together four-wire, but they may also be used for circuits connected together two-wire if precautions have been taken to ensure good return losses at these points, for example, a mean value of 27 dB with a standard deviation of 3 dB.

C. — Proposed revision of Supplement No. 2 to Volume III

(Geneva, 1964, amended at Mar del Plata, 1968; referred to in Recommendation G.131.B)

Talker echo on international connections

The curves of Figure 5 may be used to determine whether a given international connection requires an echo suppressor. Alternatively they may be used to find what value of nominal overall transmission loss shall be adopted for the four-wire chain of a complete connection so that an echo suppressor is not needed.

Before the curves can be used it must be decided what proportion of calls are to be allowed to exhibit an objectionable echo and Recommendation G.131.B gives guidance on this matter.

The co-ordinates of the graph are two of the parameters of a telephone connection that govern echo—the reference equivalent of the echo path and total length of the four-wire chain. By making certain assumptions (discussed below) these two parameters become the principal ones.

Each curve divides the co-ordinate plane into two portions and the position, relative to the curve, of the point describing the connection indicates whether an echo suppressor is needed, bearing in mind the percentage of calls permitted to exhibit an objectionable echo.

Factors governing echo

The principal factors which must be considered in order to decide whether an echo suppressor is needed on a particular connection are:

a) the number of echo paths;

b) the time taken by the echo currents to traverse these paths;

c) the reference equivalent of the echo paths including the subscriber lines;

d) the tolerance to echo exhibited by subscribers.

These factors are discussed in turn in the following paragraphs.

When circuits are switched together four-wire there is only one echo path, assuming negligible go-to-return crosstalk. This is also substantially true if the circuits are switched together two-wire and reasonable values of echo return losses are achieved at the connection points because the principal echo currents are those due to the relatively poor echo return losses at the ends of the two extreme four-wire circuits, where the connection is reduced to two-wire.

The time taken to traverse the echo path is virtually dependent solely on the length of the four-wire connection because the main circuits of modern national and international networks are high-velocity circuits.

The reference equivalent of the talker echo path for a symmetrical connection is approximately given by the sum of twice the transmission loss of the complete connection between the two-wire points in the terminal local exchanges, the echo return loss presented at the two-wire point in the far-end terminal local exchange, and the reference equivalent of the subscriber's lines connected to these two-wire points. In general, values of reference equivalents corresponding to low loss subscriber lines should be used. The echo experienced by subscribers on lines with more loss will be further attenuated. This is therefore a safe assumption.

The echo return loss is assumed to have a mean value of not less than 11 dB with a standard deviation of 3 dB expressed as a weighted mean power ratio over the band 500–2500 Hz. The mean value of the transmission loss is assumed to be uniform over this band and the standard deviation of transmission loss for each four-wire circuit is assumed to be 1 dB for each direction of transmission. The correlation between the variations of loss of the two directions of transmission is assumed to be unity.

The data on tolerance to echo exhibited by subscribers given in the table below are furnished by the American Telephone and Telegraph Co. and are based on a series of studies completed in 1971. These tests provided information on the reference equivalent of the echo path for just detectable echo as a function of echo path delay. In addition, ratings of quality on a five-point scale (excellent, good, fair, poor, unsatisfactory) were also obtained. The table indicates the mean echo path loss for the threshold of detectability and for ratings of good or better and fair or better. These mean values are the reference equivalent of the echo path for 50% detectability, 50% good or better and 50% fair or better. The standard deviation is also given.

Results of echo tolerance tests

One-way propagation time — ms	Reference equivalent of echo path —dB					
	Threshold		Good or better		Fair or better	
	Mean dB	Std. dev. dB	Mean dB	Std. dev. dB	Mean dB	Std. dev. dB
10	26	≈4	25	≈7	18	≈6
20	35	≈4	32	≈7	25	≈6
30	40	≈4	36	≈7	29	≈6
40	45	≈4	39	≈7	32	≈6
50	50	≈4	41	≈7	34	≈6

Construction of Figure 5

The mean margin against poor or unsatisfactory echo performance is given by:

$$M = 2T + B - E + SRE + RRE$$

where T = mean overall loss between the two-wire points in the terminal local exchanges. The loss is assumed to be the same in both directions of transmission;

B = mean echo return loss at the listener end;

E = mean value of reference equivalent of the echo path required for opinion rating of fair or better (Note 1);

SRE = sending reference equivalent to two-wire point in the originating local exchange for short subscriber lines;

RRE = receiving reference equivalent to two-wire point in the originating local exchange for short subscriber lines.

The standard deviation of the margin is given by:

$$m^2 = n(t^2_1 + 2rt_1t_2 + t^2_2) + b^2 + e^2$$

where m = standard deviation of the margin;

t_1, t_2 = standard deviation of the transmission loss in the two directions of transmission of one four-wire circuit, national or international;

Note 1. — This corresponds to the value of reference equivalent of the echo path at which 50% of the opinion ratings are fair or better.

b = standard deviation of echo return loss;

e = standard deviation of the distribution of reference equivalents echo path required for opinion ratings of fair or better;

r = correlation factor between t_1 and t_2;

n = the number of four-wire circuits in the four-wire chain.

Inserting $t_1 = t_2 = 1$ dB; $r = 1$; $b = 3$ dB; $e = 6$ dB gives $m^2 = 4n + 45$.

In Recommendation G.131.B, c), Rules A and E refer to 10% and 1% probabilities of encountering poor or unsatisfactory echo and for these cases 9 four-wire circuits will be assumed (3 national + 3 international + 3 national). For both the 1% and 10% curves therefore $m = 9.0$ dB.

For 10% probability, the margin may fall to 1.28 times the standard deviation. The corresponding factor for the 1% curve is 2.33. Hence the corresponding values of M are:

$M = 1.28 \times 9.0 = 11.5$ for 10% probability
$M = 2.33 \times 9.0 = 21$ for 1% probability.

Putting these values into $M = 2T + B - E + SRE + RRE$ gives the following values for the mean talker echo attenuation, $2T + B + SRE + RRE$.

$2T + B + SRE + RRE = 11.5 + E$ for 10% probability
$2T + B + SRE + RRE = 21 + E$ for 1% probability.

The values in the table below have been calculated (to the nearest whole decibel) using these equations. The figures in the length of connection column have been calculated assuming a velocity of propagation of 160 km/ms.

Mean one-way propagation time	Length of connection	Reference equivalent of mean echo path $2T + B + SRE + RRE$	
		10% Poor and unsatisfactory	1% Poor and unsatisfactory
ms	km	dB	dB
10	1600	30	39
20	3200	37	46
30	4800	41	50
40	6400	44	53
50	8000	46	55

Figure 5 has been constructed from these values. The scale of ordinates on the right-hand side of the figure has been determined on the assumptions that:

a) there is no difference between the nominal value and the mean value of the loss of the connection between the two-wire points in the terminal local exchanges;
b) the nominal loss is the same in both directions of transmission;
c) the mean value of the echo return loss at the listener end is 11 dB.

II

Echo-path attenuation and delay time

JAPANESE ADMINISTRATION

1. *Introduction*

This report concerns studies of the relationship between the delay-time and the necessary echo-path attenuation for Japanese-language communications. The results show practical echo-path attenuation on the basis of the speaker's echo. They also show the echo-path attenuation for which the echo is hardly perceptible.

2. *Tests made*

Figure 1 gives a block-schematic diagram of the test circuit. A magnetic tape-recorder (MTR) was used as a delay-device, and a variable attenuator (ATT) was used to adjust the echo-path attenuation. The telephone set used is a No. 4A automatic telephone set, a standard model in Japan. The subjects were 14 men with normal hearing. The place in which the test was made was an office room.

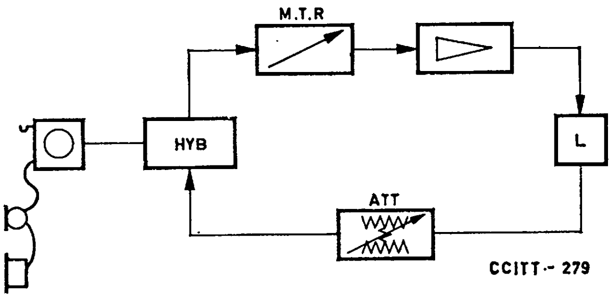
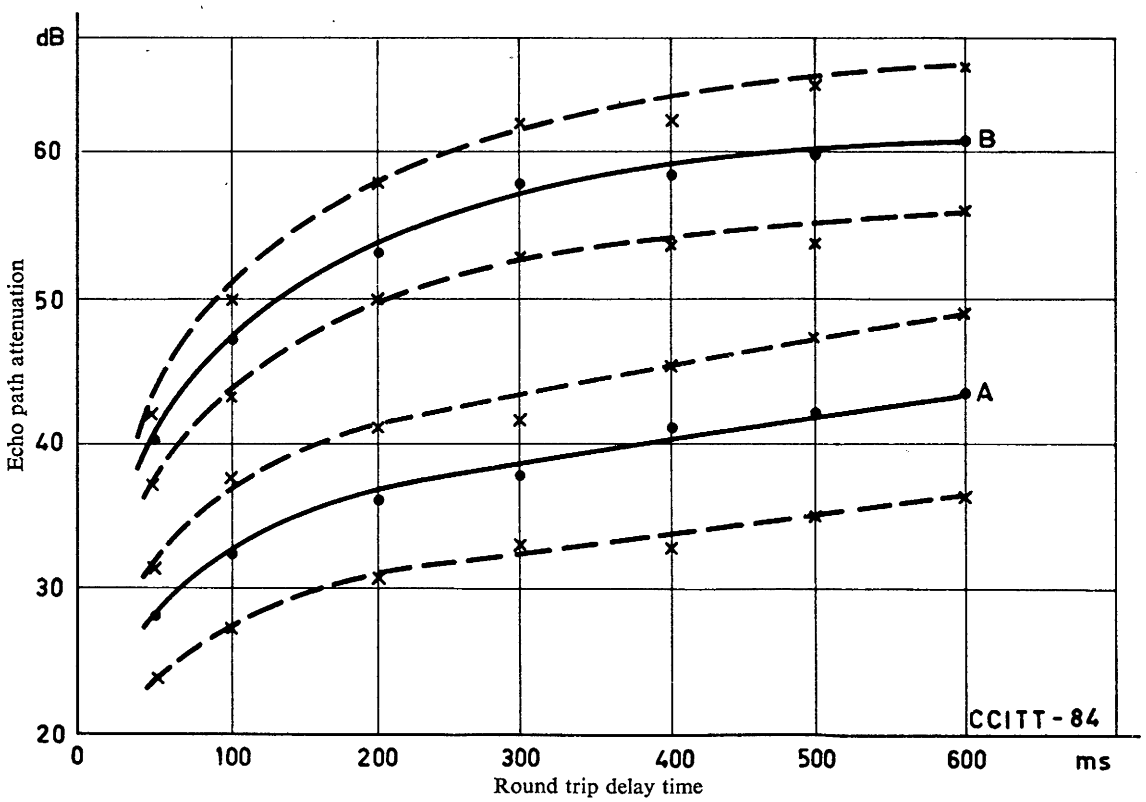


FIGURE 1. — **Block-schematic diagram of the test circuit**

Several values of delay-time between 50 ms and 600 ms were selected. For each delay-time, a subject was required to adjust the attenuator according to his own auditory sensation, thereby making it possible to find the echo-path attenuations for which the echoes are imperceptible or perceptible but not objectionable to him. Incidentally, the subjects repeated the same experiment with four telephone sets to minimize variations caused by the sets. The telephone transmitter varies in characteristics when it is moved. An appropriate current was therefore set flowing through the transmitter to eliminate this effect.

3. *Test results*

The results are shown in Figure 2.



Curve A: Perceptible but not objectionable

Curve B: Hardly perceptible

FIGURE 2. — **Echo-path attenuation for delay time (Japan)**

Curve A shows delay time and average echo-path attenuation for which an echo just becomes intolerable to the subjects. The dotted lines above and below the curve denote the standard deviation of variations depending on the subjects.

Curve B shows, in a similar manner, the average echo-path attenuation for which an echo is just perceptible for each delay-time.

The echo-path attenuations increase rapidly up to 100 ms, but thereafter increase but slowly and the variations in the measured values are spread with an almost normal distribution, the standard deviation of which is approximately 5 dB.

By using the observed average echo-path attenuation and the standard deviation shown by curve A, the necessary echo-path attenuation can be obtained for any desired grade of service from the point of view of echo. Subtraction of the two-wire reflection loss at the distant end from this value makes it possible to find the necessary attenuation to be given by the echo suppressor.

4. *Conclusion*

When Japanese-language communications are carried out under the aforementioned operating conditions, echo-path attenuation of about 37 dB is required, to fulfil the condition that 86% of the speakers will perceive the echo but do not consider it to be objectionable, for a round-trip delay of about 100 ms in the earth satellite communication of low altitude for example. With stationary satellite communication in which the round-trip delay is about 600 ms, the value is about 48 dB.

III

United Kingdom Post Office

The results of tests carried out by the United Kingdom Post Office in 1956–58 have been published (reference [1]). These were not then expressed in terms of reference equivalents but the information has now been converted to this form (reference [2]) and Figures 1, 2 and 3 give extracts from this publication giving the more important conclusions. These results are based on the use of United Kingdom Post Office 300-type telephone sets (extensively in service at the time of the tests), having median-length (1.6 km) local lines, and a corresponding sidetone reference equivalent of + 10.6 dB. Figure 1 shows a pair of sigmoid tolerance distribution curves that apply for a round-trip delay of 50 ms, giving the percentage of naive subjects who found echo "detectable" [curve (a)] and objectionable [curve (b)] the results being expressed in terms of the nominal overall reference equivalent of the echo path (*SRE* + *RRE* + losses of intervening lines). The shape of these sigmoids and their relative displacements do not change

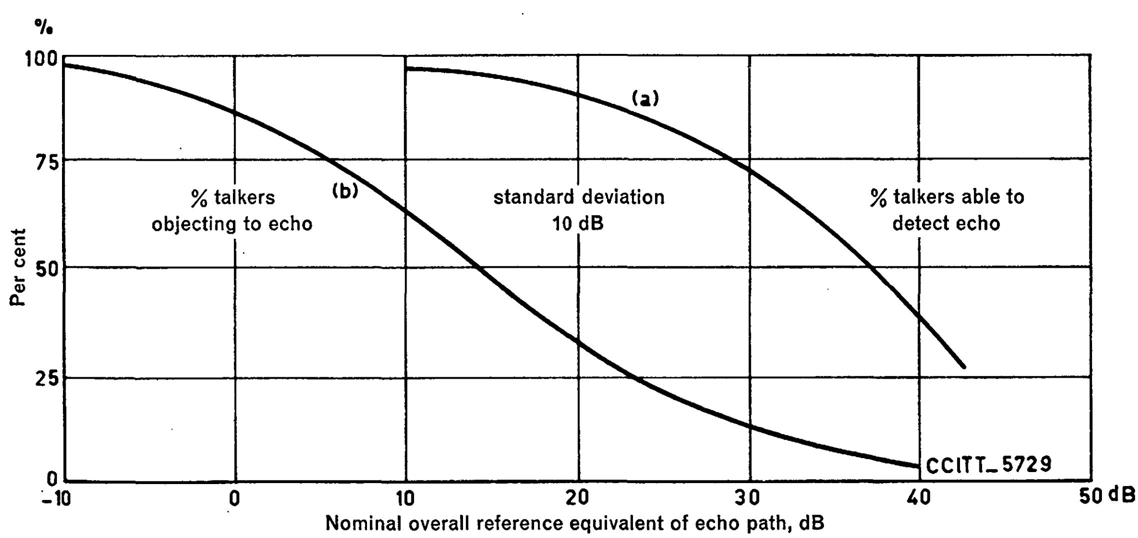


Figure 1. — **Sigmoids showing detectable and objectionable thresholds of echo for a round-trip delay of 50 ms**

Note 1. — See note 1 of Figure 2.
Note 2. — See note of Figure 3.

appreciably for round-trip delays in the range 15 to at least 60 ms but their joint location along the scale of nominal overall reference equivalent of the echo path is a function of round-trip delay. The information given in this figure enables the fundamental information to be presented in the convenient form shown in curves (c) to (g) of Figure 2. (When comparing Figures 1 and 2, attention is drawn to Note 2 of Figure 2.)

The relationships shown in Figure 2 apply, as stated above, for a sidetone reference equivalent of + 10.6 dB. The threshold of detectability and objectionableness of echo are, however, functions also of the value of reference equivalent of the sidetone path of the talker's telephone set; the effect on the mean intercategory thresholds is shown in Figure 3. Masking of echo by sidetone has been found to be substantially independent of echo delay over the range investigated (0–60 ms), hence the correction factors shown in Figure 3 can be added directly to the values of echo path nominal overall reference equivalent given in Figure 2. For example, 5% of listeners object to echo when the delay is 20 ms and the nominal overall reference equivalent of the echo path is 19 dB.

If the sidetone reference equivalent is increased to + 15 dB, the nominal overall reference equivalent of the echo paths must, using curve (h) of Figure 3, accordingly be increased to approximately 20.5 dB, other factors remaining constant.

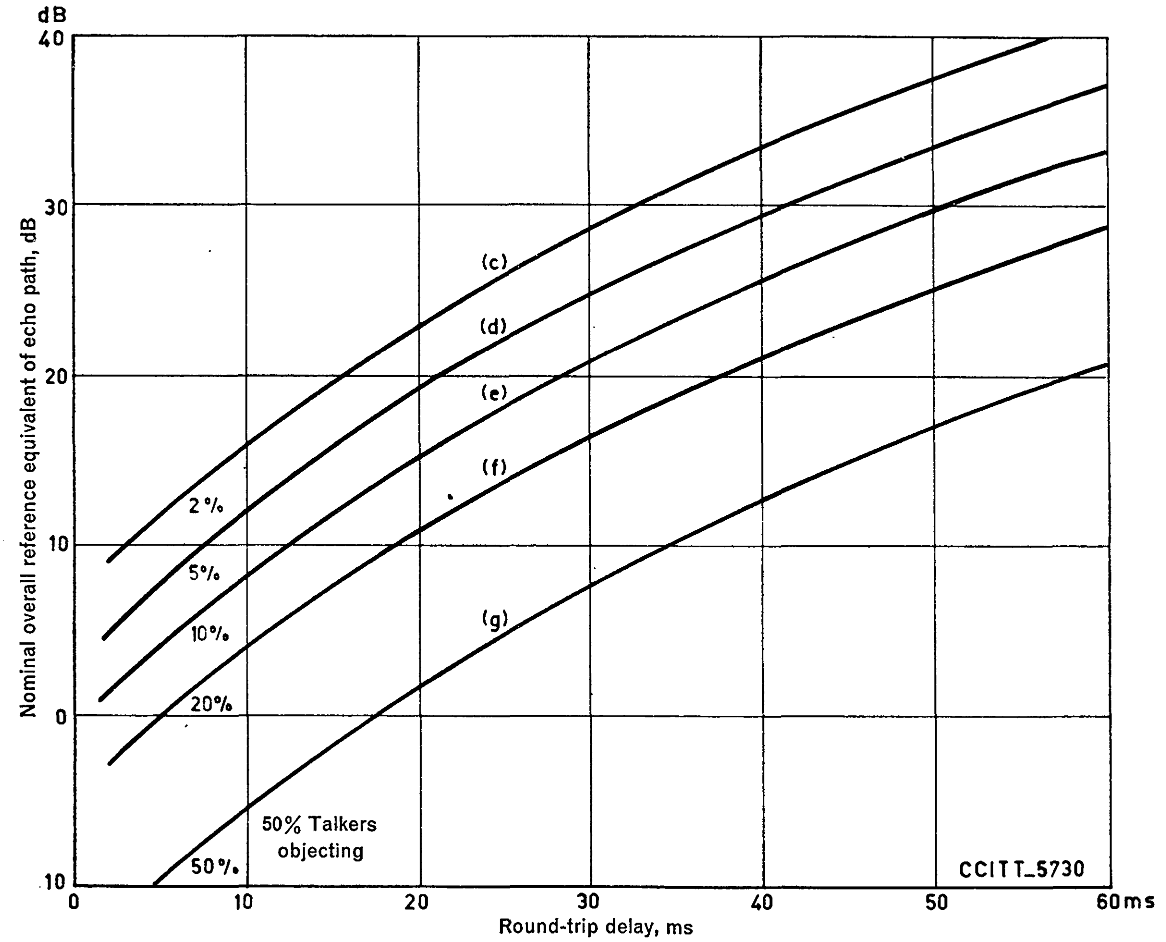


FIGURE 2. — **Relationships between nominal overall reference equivalent of echo path and round-trip delay for various percentages of subjects who would find the echo objectionable**

Note 1. — The above relationships apply for a sidetone reference equivalent of +10.6 dB.

Note 2. — It will be found that the values from curve (b) of Figure 1 are not exactly consistent with those from curves (c) to (g) of Figure 2 above; this is because different subjects were used and because the latter curves include information from several other experiments.

The tests also showed that the median value of loss at which naive subjects will object to echo is 26 dB less than the median value at which an experienced testing crew can just detect the presence of echo. In the latter case the standard deviation is very much less than the 10 dB which applies for untrained subjects.

Acknowledgement: Figures 1, 2 and 3 have been prepared from Figures 4.26 and 3.26 of reference [2].

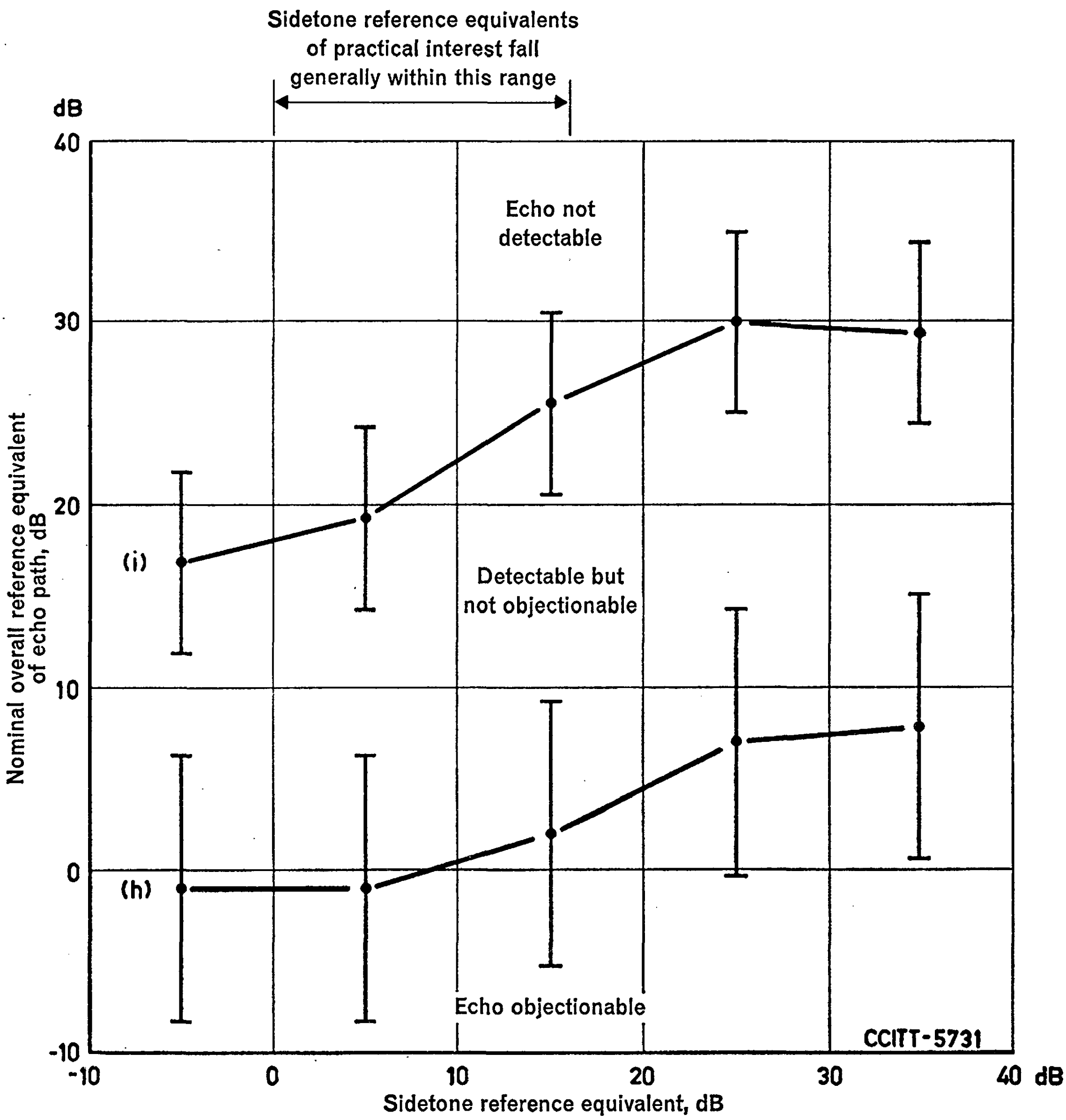


Figure 3. — **Effect of sidetone reference equivalent on mean intercategory thresholds of echo for round-trip delay of 25 ms**

Note. — Median thresholds are very close to mean thresholds. Standard deviation of distributions of subjects' tolerances is approximately 9 dB.

References

[1] Richards, D. L., and Buck, G. A.: Telephone Echo Tests, *Proceedings of the Institute of Electrical Engineers*, 1960, 107, Part B, pp. 553-556.

[2] Richards, D. L.: Telecommunication by speech, 1973, Butterworths, London (Chapters 3.5.2 and 4.3.2).

ANNEX 5

(to Question 6/XII, Part e)

I

User reaction to simulated overseas telephone circuits with long propagation times

CONTRIBUTION BY COMSAT

1. *Introduction*

Results from studies on the subjective evaluation of simulated overseas telephone communications, under a variety of conditions of echo return loss and delay are here presented.[1] The purpose of the studies was to evaluate the potential advantages of echo return loss enhancement by echo cancellation techniques and to provide guidance with regard to the amount of echo return loss necessary to achieve satisfactory service in the international telecommunications satellite environments. No echo suppressors were used in the test circuits.

Separate studies were performed at the laboratories of Standard Telephone and Cables (S.T.L.) [1] and [2] and at the K.D.D. Research Laboratories [3]. Two languages were used, English [1] and [2] and Japanese [3].

2. *Circuit configurations*

The general circuit configuration used in both laboratories is shown in Figure 1. Four-wire handsets were used, and the echo path was a circuit in parallel with each handset consisting of an attenuation pad.

The circuit parameters, used by each laboratory, are listed below. Note that the parameter designated as echo reference equivalent (ERE) is the sum of the overall reference equivalent (ORE) and the echo return loss (ERL); it represents the loss of the path from the mouth to the ear of the talker via the telephone echo path and is used to identify the various echo conditions in the figures that follow.

	S.T.L. [1] [2]	*K.D.D.* [3]
Overall reference equivalent (ORE)	16.5 dB	25 dB average
Echo return loss (ERL) (five values)	15, 25, 31, 37, 50 dB	21, 31, 37, 43, 56 dB
Echo reference equivalent (ERE)	31.5, 41.5, 47.5, 53.5, 66.5 dB	46, 56, 62, 68, 81 dB
Noise levels	—46 dBm0p	—51 to —35 dBm0p (uniformly assigned)
Mean one-way propagation time (MOPT)	50, 300, 450, 600 ms	50, 300, 450, 600 ms

Upon placing the calls, test subjects had simulated operator assistance, mixed with a background of sounds which typically accompany such calls.

3. *Testing procedure*

Test subjects were exposed to 40 test conditions, or a fraction thereof, at random. The exact procedure depended on the chosen statistical design administered by the two laboratories. By providing operator assistance, and introducing the usual sounds during the time of setting up the call, test subjects were placed in a realistic overseas calling environment. Eighty people (involved in 800 exposures) participated in the S.T.L. experiments, and sixty (involved in 9600 exposures) in the K.D.D. experiments.

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[1] The views and conclusions on these results are COMSAT evaluations and do not necessarily reflect those of the International Telecommunications Satellite Consortium (INTELSAT).

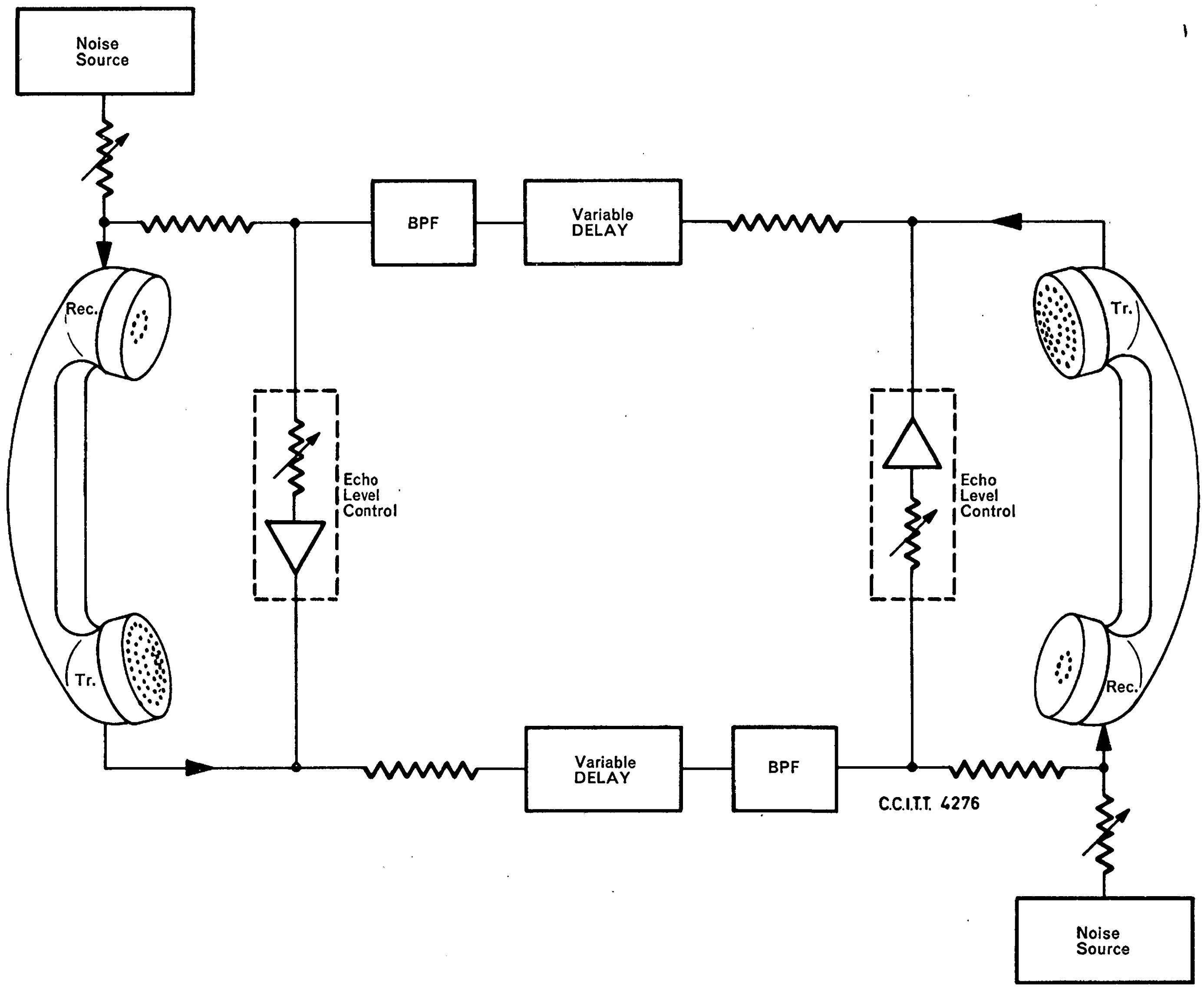


FIGURE 1. — **General test circuit for echo and delay experiments**

In both studies, the subjects were required to perform a task in order to simulate a two-way conversation. Upon completion of a call, a call-back interview was conducted during which the following questions were asked:

Question	*Quality rating*	*Corresponding numerical rating*
(1) How would you rate the circuit?	Excellent	4
	Good	3
	Fair	2
	Poor	1
	Unacceptable (or bad)	0
(2) Did you have any difficulty conversing?	Yes	
	No	

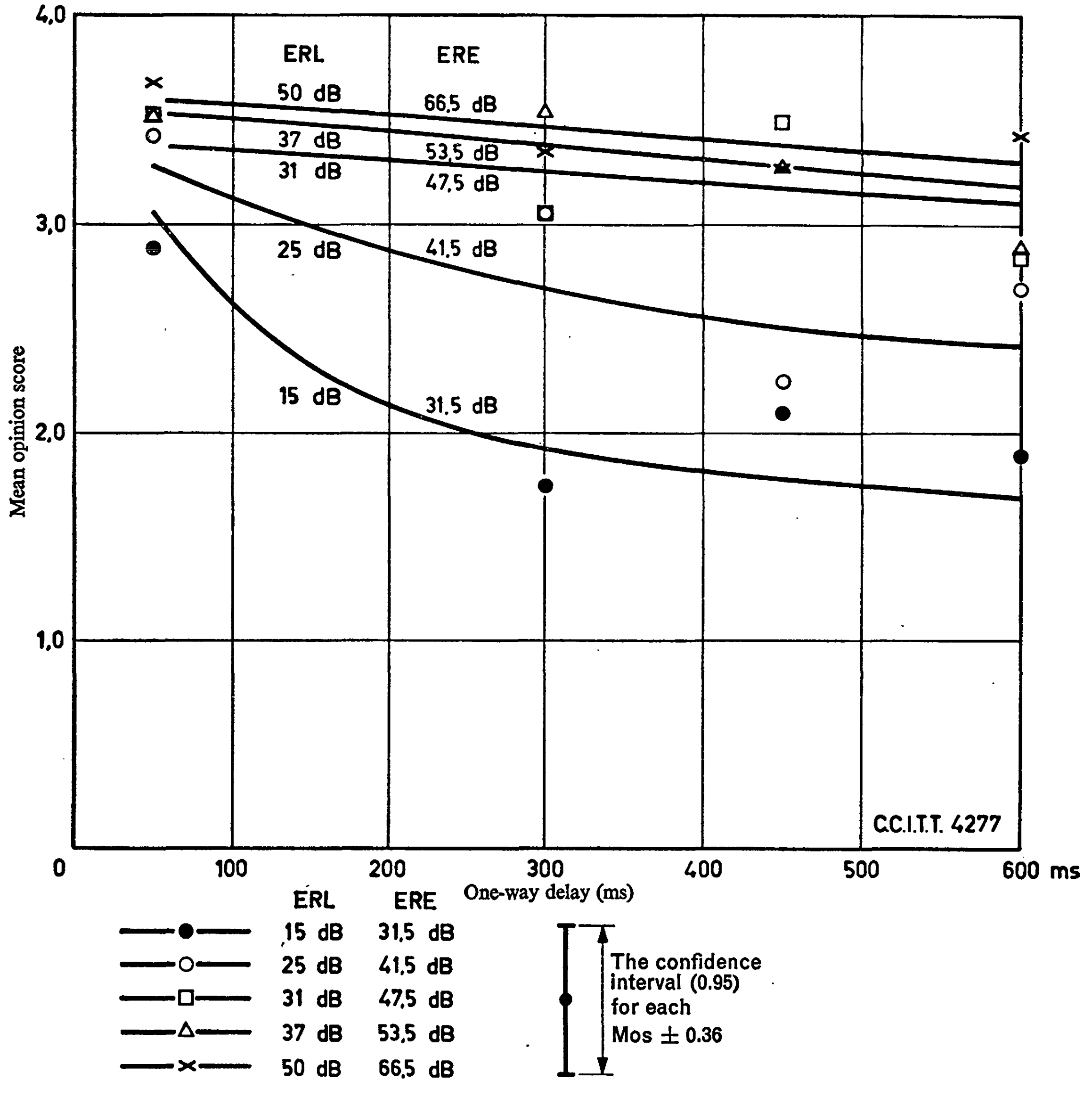


FIGURE 2. — S.T.L. studies

Mean opinion score (MOS) as a function of delay, with echo return loss (ERL) or echo reference equivalent (ERE) as a parameter.

The reverse order of above questions (1) and (2) was also asked an equal number of times. From Question (1), a mean opinion score (MOS) was calculated for each test condition. From Question (2) the percentage difficulty scores were determined.

4. *Test results*

The results obtained for mean opinion score (MOS) and percentage difficulty as a function of mean one-way propagation time, with echo return loss or echo reference equivalent as parameters are shown in Figures 2 and 4 for the S.T.L. studies and Figures 3 and 5 for the K.D.D. study. Another point of view of the same data is obtained by plotting MOS and percentage difficulty as a function of echo return loss or echo reference equivalent with mean one-way propagation time as a parameter. This is done in Figure 6 for the S.T.L. studies and Figure 7 for the K.D.D. study.

The data shown in Figures 2 and 3 show that for the low values of echo return loss and echo reference equivalent, the mean opinion score exhibits a rapid initial decrease with increasing propagation time, with the rate of change decreasing for propagation delays in excess of 300 milliseconds. As echo return loss and echo reference equivalent are increased, the initial rapid decrease vanishes and the curves show a slow relatively uniform drop in mean opinion score as propagation time increases. The differences in mean opinion score between the 300 and 600 ms mean one-way propagation times (MOPTs) are not significant at the 95% level.

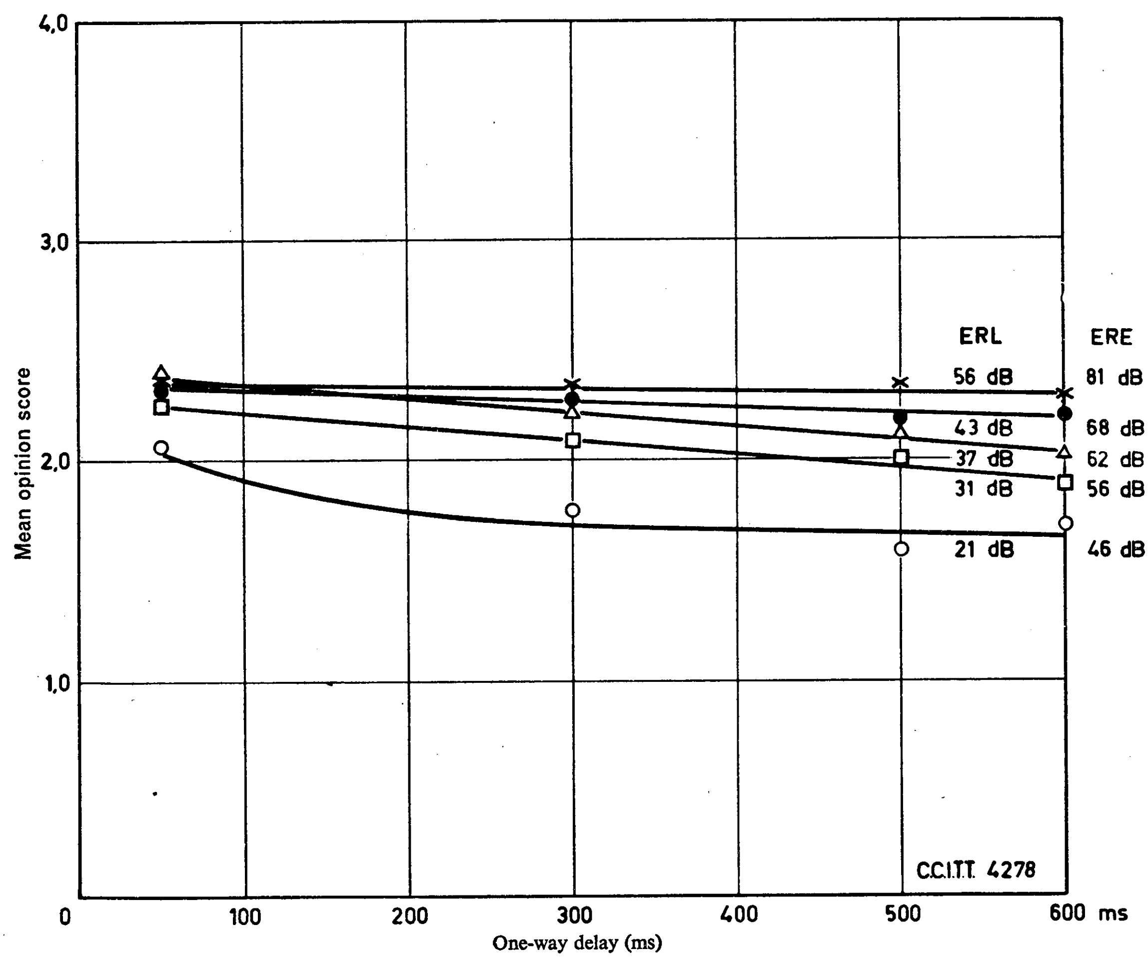


FIGURE 3. — **K.D.D. study**

Mean opinion score (MOS) as a function of delay, with echo return loss (ERL) or echo reference equivalent (ERE) as a parameter.

Referring to the per cent difficulty plots shown in Figures 4 and 5, it is seen that difficulty shows a rapid initial increase with increasing mean one-way propagation time, with the rate of increase diminishing as propagation time exceeds 300 milliseconds. As in the case of the mean opinion scores, the difference in per cent difficulty for the 300 and 600 ms cases is not significant. The S.T.L. data, Figure 4, indicates that for an echo return loss of 31 dB, corresponding to an echo reference equivalent of 47.5 dB, the percentage difficulty for 300 ms mean one-way propagation time is less than 10% and rises to 12% for 600 ms. The K.D.D. data indicates that for an echo return loss of 37 dB, corresponding to an echo reference equivalent of approximately 62 dB, the percentage difficulty is 10% at 300 ms and 13% at 600 ms.

Referring to Figures 6 and 7, where the data from both studies is plotted as a function of echo return loss and echo reference equivalent, it is seen that values of percentage difficulty and mean opinion score show a high initial rate of change as echo return loss and echo reference equivalent increase and approach relatively stable values at higher values of these variables. The S.T.L. study shows that increases beyond 35 dB ERL, corresponding to 51.5 dB ERE, produce little additional decrease in percentage difficulty or increase in mean opinion score for the 300 and 600 ms mean one-way propagation times. Similar results are indicated in the K.D.D. data relative to ERL. It

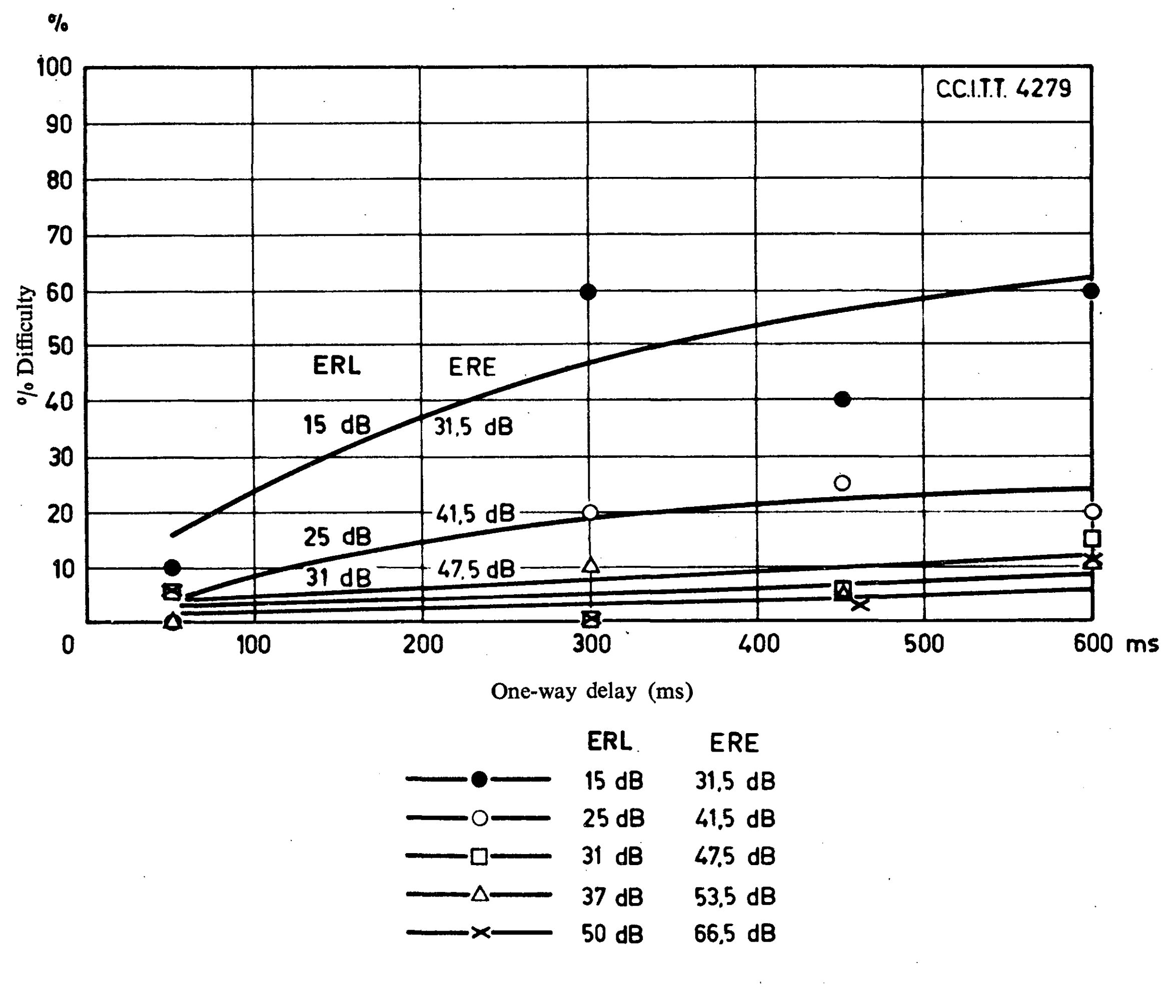


FIGURE 4. — S.T.L. studies

Percentage difficulty as a function of delay, with echo return loss (ERL) or echo reference equivalent (ERE) as a parameter.

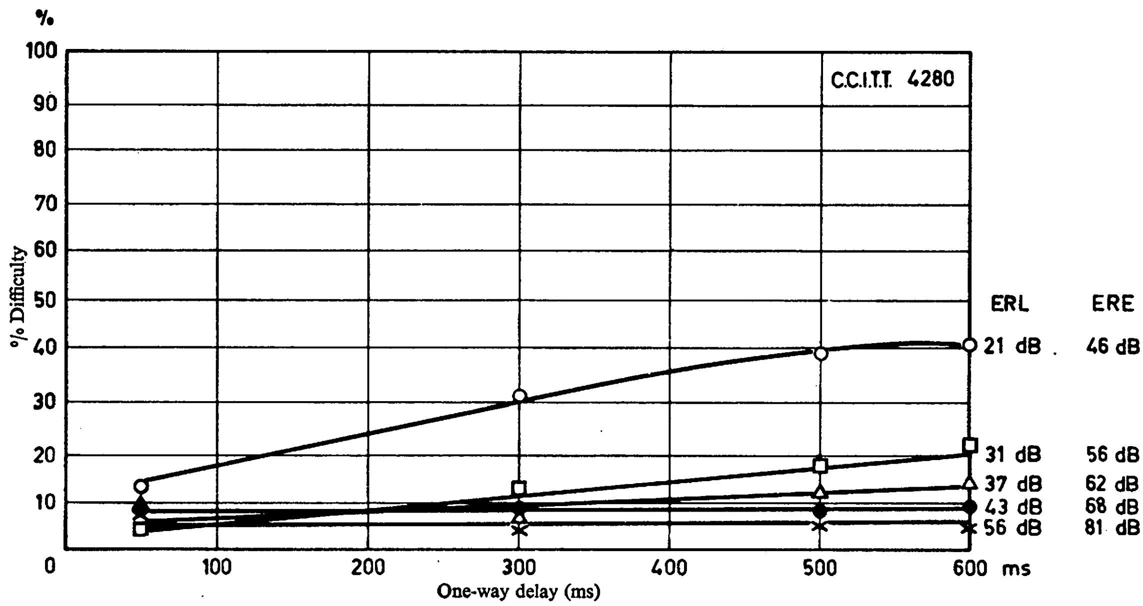


FIGURE 5. — **K.D.D. study**

Percentage difficulty as a function of delay, with echo return loss (ERL) or echo reference equivalent (ERE) as a parameter

should be noted that in the case of Figures 6 and 7 dependence on both echo return loss and echo reference equivalent are illustrated. Comparison of the data appears to indicate that echo return loss was the predominating controlling factor.

The fact that the MOS values obtained by S.T.L. are higher than those obtained by K.D.D. can be ascribed to the difference on OREs used in the two test circuits. S.T.L. used an ORE of 16.5 dB whereas that used by K.D.D. averaged 25 dB. Previous experiments [4], which assess the influence of OREs on the MOS, show that the MOS decreases with the increasing ORE and, hence, support the data obtained from the S.T.L. and K.D.D. studies.

5. *Conclusions*

The S.T.L. data indicates that for an echo return loss of 31 dB and greater, corresponding to an echo reference equivalent of 47.5 dB, the limiting values of mean opinion score and percentage difficulty are for all practical purposes attained. From the K.D.D. data a similar situation was attained with an echo return loss of 30 dB (corresponding to an echo reference equivalent of 55 dB) for 300 ms MOPT and with an echo return loss of 37 dB (corresponding to an echo reference equivalent of 62 dB) for 600 ms MOPT.

The above data gives information on the level of echo cancellation needed to attain echo-free four-wire performance of long propagation delay circuits without switching-type echo suppressors. If it is assumed that an average circuit in the international system has an echo return loss of 15 dB, then an echo canceller providing an enhancement in echo return loss of an additional 20 to 25 dB should provide echo-free performance on the average.

For the levels of echo return loss stated above, the laboratory experimental results reported here indicate that circuits with MOPT values up to 600 ms should approach commercially acceptable quality.

References

[1] Final report - COMSAT Contract CSC-SA-145: The subjective evaluation study of simulated long-delay telephone communications (COMSAT, contracting on behalf of INTELSAT).

[2] Williams, G. and Moye, L. S.: Subjective evaluation of unsuppressed echo in simulated long-delay telephone communications. *Proc. I.E.E.*, Volume 118, No. 3/4, March/April 1971, pp. 401–408.

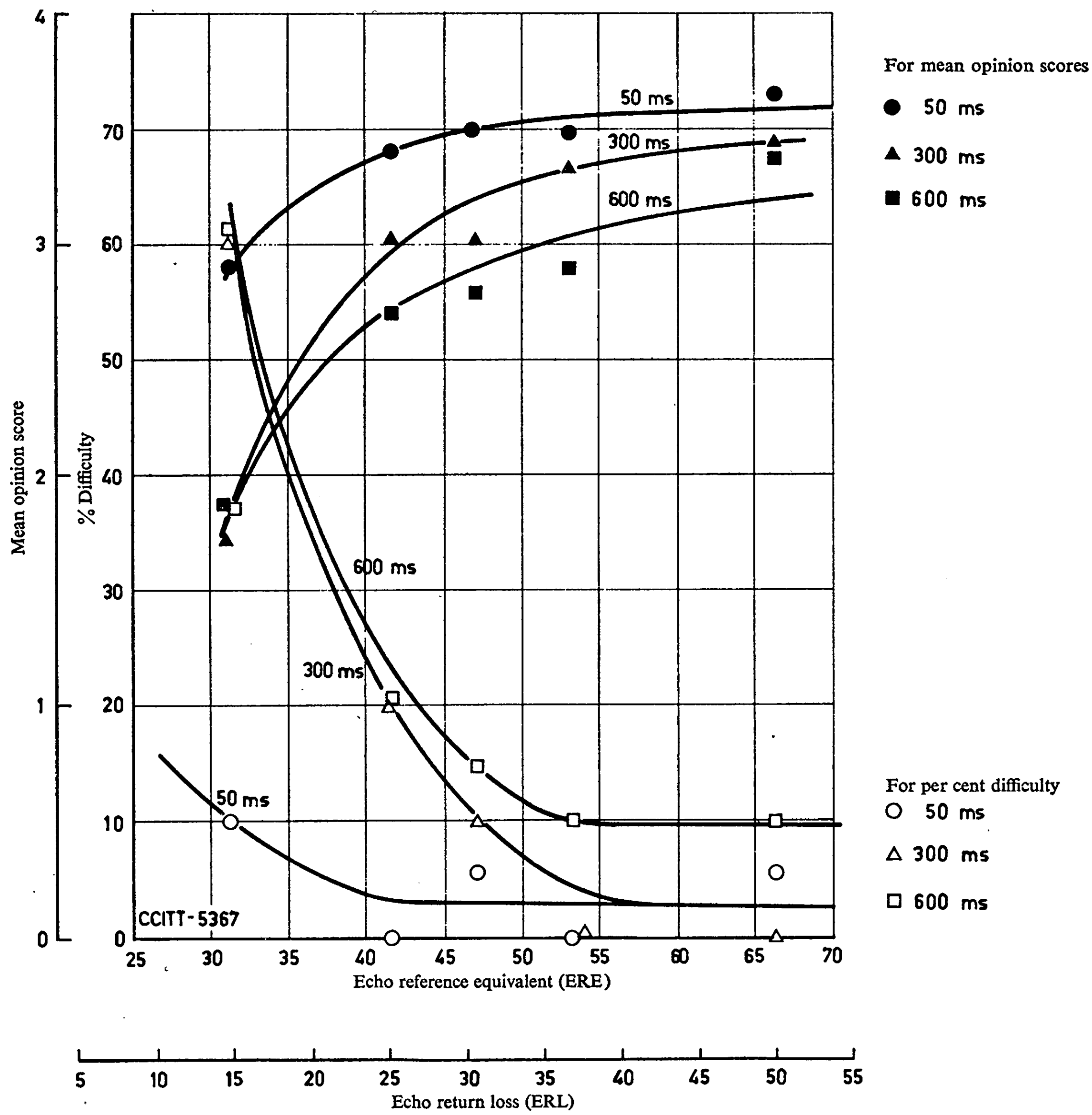


Figure 6. — S.T.L. studies

Percentage difficulty as a function of ERE or ERL with MOPT as a parameter.

[3] Final report - COMSAT Contract CSC-SA-149: The subjective evaluation study of simulated long-delay telephone communications (COMSAT, contracting on behalf of INTELSAT).

[4] Introduction of four-wire switching system in toll centres in Japan—*N.T.T. Report*, presented at the International Seminar on National Telephone Transmission Planning, Melbourne, Australia, March 1970.

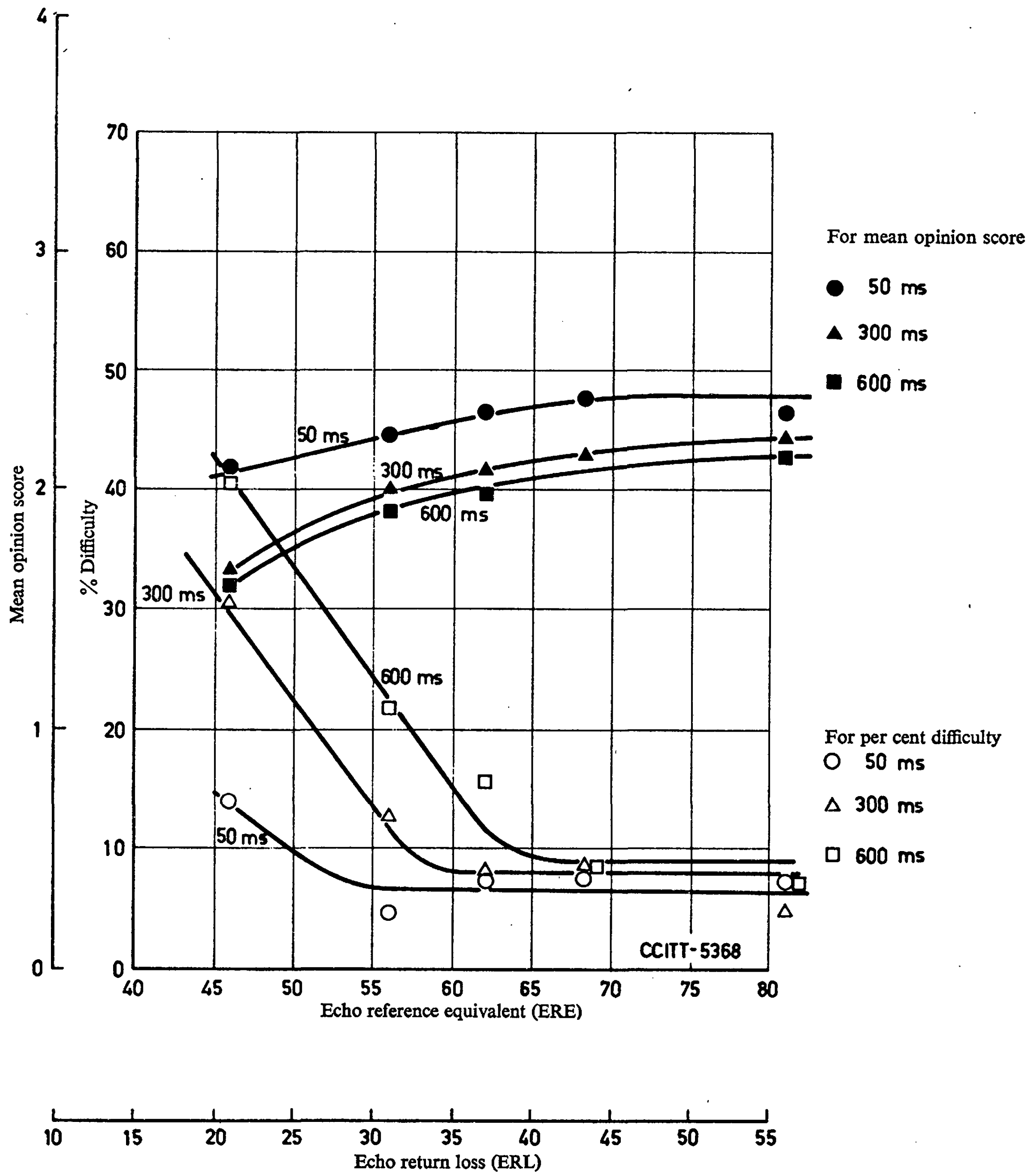


FIGURE 7. — K.D.D. study

Percentage difficulty as a function of ERE or ERL with MOPT as a parameter.

II

Telephone circuits with long propagation time

CONTRIBUTION BY COMSAT

1. *Introduction*

The influence of propagation delay on telephone subscriber reaction to long-distance telephone communications via a communications satellite has been a subject of considerable discussion and investigation since the advent of satellite communications.

Much of the data on the subject collected during the 1964–1968 C.C.I.T.T. Study Period appears in a bibliography to C.C.I.T.T. *White Book*, Recommendation G.114 (P. 14). During that period, a significant amount of data was available for actual and simulated single-hop satellite communications circuits based on call-back interview and service observation methods. The actual single-hop satellite circuits possessed a MOPT[1] of approximately 300 ms, including terrestrial extensions [1]. By extending single-hop satellite circuits with a cascade of terrestrial links, circuits with greater than single-hop propagation delay were contrived [2]. In this manner, MOPTs up to 420 ms were obtained. Customer call-back interview tests were performed on these circuits with results which indicated rapidly increasing percentage difficulty in talking or hearing (percentage difficulty is determined by the number of "yes" responses to the question, "Did you have difficulty?") for propagation delays greater than that of the single-hop satellite. For example, one circuit, comprising a single-hop satellite across the Atlantic extended across Canada and the COMPAC submarine cable through the Pacific Ocean to New Zealand gave a MOPT of 420 ms. For this circuit, the percentage difficulty was found to be approximately 65%.

As a consequence of this data, Part A (c) of C.C.I.T.T. *White Book* Recommendation G.114 (P. 14) designates circuits with mean one-way propagation times greater than 400 ms as unacceptable and recommends their use only under the most exceptional circumstances. All of the data used in arriving at the wording at present appearing in the various C.C.I.T.T. *White Book* Recommendations which touch upon the use of circuits with propagation delays greater than 400 ms MOPT was based on either greatly extended terrestrial circuits or single-hop satellite circuits extended by the use of long terrestrial circuits. It is not possible to determine what portion of the difficulty scores so obtained may have been due to the limitations and faults attending the operation of the tandem arrangement of the many terrestrial links to form a long MOPT circuit. No data was available at that time on double-hop satellite circuits which would avoid the use of many terrestrial links in tandem.

As a result of deliberation on collected data in the bibliography now appearing in C.C.I.T.T. Recommendation G.114 (P. 14) concerning Part A (b) of the Recommendation, the last paragraph of Part A reads: "Until such time as additional, significant information permits Administrations to make a firmer determination of acceptable delay limits, they should take full account of the documents referred to in the bibliography in selecting, from alternatives, plans involving delay, in range b) above Recommendation G.114." The statement quoted above refers to MOPTs covered in sub-paragraph (b) of the second paragraph of Part A, pertaining to the range from 150 to 400 ms. On the other hand, no reference in this last paragraph of Part A was considered necessary for Part A (c) referring to MOPTs greater than 400 ms. These MOPTs were considered to be unacceptable, and connections with these delays were to be used under the most exceptional circumstances.

2. *Post-1968 data on long propagation delay circuits*

With respect to circuits having delays from 150 to 400 ms MOPT and circuits with MOPTs in excess of 400 ms and up to 600 ms MOPT, much additional significant information has been obtained regarding the reaction of users of telephone systems to long propagation time. This new significant information is summarized as follows:

A. *S.T.L. and K.D.D. studies:*

Two studies sponsored by INTELSAT were conducted independently by Standard Telecommunications Laboratories (S.T.L.) in the United Kingdom [3], [4] and Kokusai Denshin Denwa Company, Ltd. (K.D.D.),

[1] MOPT denotes "mean one-way propagation time".

in Japan [5]. These studies were devoted to assessment of user reaction to simulated international telephone circuits with MOPT delays of 50, 300, 450 and 600 ms. No echo suppressors were used, and precisely adjusted values of echo and circuit noise typical of international satellite circuits were admitted. The assessment of user reaction was accomplished by the call-back method. The percentage difficulty obtained by affirmative responses to the question, "Did you have difficulty in talking or hearing?" for various combinations of MOPT, echo reference equivalents (ERE) and echo return loss (ERL) are given in Table 1. Only data pertaining to single-hop (300 ms) and double-hop (600 ms) are given.

From the data given in Table 1, the following points regarding long propagation time circuits can be made:

a) For single-hop circuits (300 ms MOPT) without echo suppressors, the percentage difficulty decreases to less than 15% for ERLs of 31 dB and greater. Refer to tests numbers 1, 2, 3, 7, 8 and 9. Assuming an average international circuit ERL of 15 dB, this performance can be realized with an echo canceller possessing an ERL enhancement capability of 16 dB or more;

b) For double-hop circuits (600 ms MOPT) without echo suppressors, the S.T.L. study showed that the percentage difficulty dropped to 10% for an ERL of 37 dB (refer to tests numbers 4, 5 and 6), while the K.D.D. study showed a drop to 13% for the same ERL. For an average international circuit ERL of 15 dB, this performance can be realized with an echo canceller having an echo return loss enhancement capability of 22 dB or more;

c) The difference in performance of single-hop and double-hop circuits, gauged in terms of user percentage difficulty appears to be small when appropriate echo control is realized (compare tests 3, 6, 9 and 12).

The data presented above is discussed in greater detail in Part I of this Annex above. It shows that acceptable levels of user percentage difficulty scores can be expected on circuits with double-hop satellite delay provided adequate echo control is incorporated.

TABLE 1

S.T.L. and K.D.D. data pertaining to circuits with MOPTs of 300 and 600 ms without echo suppressors and with varying amounts of echo reference equivalent (ERE) and echo return loss (ERL). The ERE is the sum of the ERL of the a–t–b path and the overall reference equivalent (ORE). For the S.T.L. experiments, the ORE was 16.5 dB; for the K.D.D., the ORE was 25 dB.

Test No.	MOPT ms	Source	ERE dB	ERL dB	Difficulty %
1	300	STL	41.5	25	20
2	300	STL	47.5	31	0
3	300	STL	53.5	37	10
4	600	STL	41.5	25	20
5	600	STL	47.5	31	15
6	600	STL	53.5	37	10
7	300	KDD	46	21	31
8	300	KDD	56	31	12
9	300	KDD	62	37	8
10	600	KDD	46	21	40
11	600	KDD	56	31	22
12	600	KDD	62	37	13

B. *A.T. & T. subjective tests of long delay circuits:*

In Part III of this Annex A.T. & T. presents data taken on laboratory circuits with MOPTs of 300 and 600 ms (as well as 32.5 ms which will not be discussed here).

The data pertinent to the performance of the simulated single- and double-hop circuits, viz. that for MOPTs of 300 and 600 ms, is summarized in Table 2.

For A. T. & T. experiments, the overall reference equivalent (ORE) was 20 dB. A. T. & T. also reported that it can be assumed that the average international circuit has an ERL of 15 dB.

From the A. T. & T. information, the following points can be made regarding user reaction to circuits with single-hop and double-hop propagation delays:

a) On single-hop circuits with echo suppressors and an average ERL of 15 dB (corresponding to an ERE of 35 dB), the percentage difficulty is about 18% (refer to tests numbers 4 and 5);

b) On single-hop circuits with echo suppressors and an average ERL of 25 dB (ERE of 45 dB), the percentage difficulty drops to approximately 8% (refer to test No. 7). An echo canceller capable of achieving an ERL enhancement of 10 dB can accomplish this;

c) On a double-hop circuit with echo suppressors and an average ERL of 25 dB (ERE of 45 dB), the percentage difficulty drops to approximately 13% (refer to test No. 16). An echo canceller capable of achieving an ERL enhancement of 10 dB used in conjunction with the echo suppressor can achieve this;

d) On a single-hop circuit with no echo suppressors and a 35 dB ERL (ERE of 55 dB), the percentage difficulty drops to approximately 12% (refer to test No. 10). This operation can be achieved with an echo canceller of 20 dB ERL enhancement capability;

e) On a double-hop circuit with no echo suppressors and an average ERL of 35 dB (ERE of 55 dB), the percentage difficulty drops to 18% (refer to test No. 19). This can be accomplished with an echo canceller capable of an ERL enhancement of 20 dB.

In summary, the A. T. & T. data shows that echo cancellation capable of ERL enhancement of 10 dB used in conjunction with an echo suppressor of conventional design should provide improved performance of single-hop and certainly acceptable performance of double-hop satellite circuits. An echo canceller capable of at least 20 dB of ERL enhancement should achieve acceptable performance on single- and double-hop circuits without echo suppressors.

C. *United Kingdom Post Office data:*

The United Kingdom Post Office reported to C.C.I.T.T. Study Group XII [6] data taken on simulated long propagation time circuits equipped with conventional echo suppressors. These results show that on circuits with OREs lying in the range from 16 to 24 dB, an increase in ERL significantly decreases the percentage difficulty indicated by users. For example, for a MOPT of 300 ms, the percentage difficulty decreased from 39% at an ERL[1] of 17 dB to 11% at an ERL of 23 dB, and for a MOPT of 600 ms for the same values of ERL, the percentage difficulty decreased from 44% to 22%. It was noted that increasing the MOPT from 300 ms to 600 ms did not seem to degrade performance as much as previous customer interview results suggested [2].

The above data indicates that acceptably low user percentage difficulty can be realized with echo suppressor-equipped double-hop satellite circuits when the ERL is 23 dB or greater.

D. *Echo canceller data:*

Echo cancellers capable of providing in excess of 25 dB enhancement in ERL have been developed both at COMSAT Laboratories [7] and by the Nippon Electric Company (N.E.C.) under contract with INTELSAT [8]. A field trial involving four echo cancellers is currently being initiated in the INTELSAT System. With these echo cancellers, circuits having an average ERL of 15 dB (which is about average for international circuits) can have their ERL increased to 40 dB.

Based on the data obtained from the A. T. & T., S.T.L. and K.D.D. studies, this level of echo cancellation is sufficient to provide effective echo-free performance on non-echo suppressor-equipped circuits and, hence, such echo cancellers can be expected to provide acceptable performance on double-hop circuits. It is important to note further that both A. T. & T. and the United Kingdom data indicates that on circuits equipped with echo suppressors,

[1] The echo return losses (ERL) quoted from these United Kingdom Post Office results can be converted to echo reference equivalent (in the sense of Annex 3 to this Question) by adding 20 dB.

considerably less ERL enhancement is needed. For example, on the average circuit with 15 dB of ERL, to achieve the 25 dB ERL indicated in the United Kingdom and A. T. & T. studies as yielding acceptable performance of a double-hop circuit, the echo canceller need provide only 10 dB of ERL enhancement. The COMSAT and N.E.C. echo cancellers are achieving considerably in excess of this amount. Cancellers of similar design and consequently lower cost which take advantage of knowledge gained in the studies reported herein are now being investigated by COMSAT. These cancellers will provide acceptable performance on double-hop circuits.

E. *Transmission quality assessments of Brazil–Japan double- and single-hop satellite circuits:*

A double-hop satellite circuit has been in operation between Rio de Janeiro, Brazil and Tokyo, Japan, since December 1971. It is routed via the Atlantic and Indian Ocean INTELSAT IV satellites. Simultaneously, between the same two locations, there has been operating another circuit comprising a single-hop via an Atlantic Ocean INTELSAT IV satellite in tandem with a chain of terrestrial microwave links across Eurasia. This circuit situation has provided a unique opportunity to obtain information comparing the transmission quality of very long-distance telephone circuits, of which one (1 hop + terrestrial) meets, and the other (2 hops) does not meet the 400 ms MOPT upper limit of C.C.I.T.T. Recommendation G.114 (P 14).

Information was obtained from EMBRATEL (Brazil) regarding customer (circuit user) reaction on both of these circuits with the specific intent of establishing if one is significantly different from the other. These data are summarized below.

EMBRATEL performed customer call-back tests asking the following questions: "Did you have difficulty?"; "What was the nature of the difficulty?"; and "How would you rate the call: excellent, good, fair, or poor?". The latter question is used to arrive at a mean opinion score by associating with each opinion level a numerical range as follows: 4-excellent, 3-good, 2-fair, 1-poor. The results are presented in Table 3.

Note that, while the double-hop circuit received a higher mean opinion score than the single-hop plus terrestrial circuit, the percentage difficulty was higher. This anomalous result is probably due to the fact that when the customer was asked to rate the circuit as either excellent, good, fair, or poor, he tended to rate it in terms of voice quality. Whereas when he was asked, "Did you have difficulty?", he tended to accentuate instances of more problems involving imperfect echo control. In reference to this observation, it should be pointed out that in the single-hop plus terrestrial configuration, one of the satellite hops was replaced by a tandem of three terrestrial links. This greater number of links may have resulted in more exposure to level fluctuations, in noise and frequency distortions. The difference noted in mean opinion score is statistically significant at the 1.5 per cent level, i.e. the probability of obtaining this result if both circuits were equal in mean opinion score is 0.015 for the sample sizes used. The difference noted in percentage difficulty is significant at a 15 per cent level and, therefore, the difference in this case is not considered significant.

From the above data, it appears that during the test period all components of both circuits operated satisfactorily. Conventional echo suppressors designed for long delay circuits were used, and overall acceptable performance was provided. Additionally, the results indicate that the double-hop circuit was not inferior to the single-hop plus terrestrial circuit.

3. *Conclusions*

A. Long propagation time of a circuit, *per se*, is not the cause for difficulty in talking or hearing, i.e. MOPTs of up to 600 ms;

B. The combination of echo, echo control and long propagation time is the cause for difficulty in talking or hearing;

C. When all components of a double-hop and of a single-hop satellite circuit in tandem with a long terrestrial circuit operate satisfactorily, the quality of transmission has been acceptable to the circuit users in Brazil, Japan, United States and United Kingdom.

TABLE 2

A.T. & T. data on subjective reaction to simulated long-delay circuits with and without echo suppressors. The ERE, ERL and ORE relationships are the same as in Table 1, but the ORE was 20 dB.

Test No.	MOPT ms	Echo suppressor	ERE dB	ERL dB	Difficulty %
4	300	IN	35	15	18
5	300	IN	35	15	17
6	300	OUT	35	15	70
7	300	IN	45	25	8
8	300	OUT	45	25	43
10	300	OUT	55	35	12
16.	600	IN	45	25	13
19	600	OUT	55	35	18

TABLE 3

		Single-hop plus terrestrial circuit	Double-hop circuit
Number of call-backs		88	70
Excellent		20	27
Good		56	39
Fair		10	2
Poor		2	2
Mean opinion score		3.07	3.30
Difficulty		9	11
% Difficulty		10	16
Difficulty categories	Noise	6	1
	Low volume	1	6
	Echo	2	1
	Cutting	1	5
	Crosstalk	0	0

REFERENCES

[1] BARSTOW, J. M.: Results of user reaction tests on communication via early bird satellite; Progress in astronautic aeronautics, 1966, Academic Press, New York and London.

[2] RICHARDS, D. L.: Transmission performance of telephone connections having long propagation times, *Het PTT-Bedrijf*, XV, No. 1/2, May 1967, pp. 12–24.

[3] WILLIAMS, G. and MOYE, L. S.: Subjective evaluation of unsuppressed echo in simulated long-delay telephone communications, *I.E.E. proceedings*, Volume 118, No. 3/4, March/April 1971, pp. 401–404.

[4] STANDARD TELECOMMUNICATIONS LABORATORIES, Final Report, COMSAT Contract CSC-SA-145: The subjective evaluation study of simulated long-delay telephone communication, February 1970 (COMSAT, contracting on behalf of INTELSAT).

[5] KOKUSAI DENSHIN DENWA COMPANY, LTD., Final Report, COMSAT Contract CSC-SA-149: The subjective evaluation study of simulated long-delay telephone communications, December 1969. (COMSAT, contracting on behalf of INTELSAT.)

[6] COM XII-No. 11: United Kingdom Post Office, Contribution to the study of Question 6/XII. This material with certain corrections will be found in RICHARDS, D. L.: *Telecommunication by speech* (Butterworths, London, 1973).

[7] Campanella, S. J., Suyderhoud, H. G., and Onufry, M.: Analysis of an adaptive impulse response echo canceller, *COMSAT Technical Review*, Volume 2, No. 1, Spring, 1972.

[8] Nippon Electric Company, Final Report, COMSAT Contract CSC-IS-351: Echo cancellers, September 1972. (COMSAT, contracting on behalf of INTELSAT.)

III

Subjective tests of long delay circuits with and without echo suppressors and having various return losses

Contribution of American Telephone and Telegraph Co.

Introduction

Q.6/XII relates to the subjective tolerance of echo and propagation time; Q.10/XV includes consideration of new means for controlling echo. In 1970 and 1971 subjective tests were performed relating to these questions. Laboratory subjects conversed over circuits in which the amount of echo and propagation delay were varied and on which echo suppressors were or were not used. The tolerance to echo on long delay circuits was thus determined as was the subjective effect of voice-operated echo suppressors on circuits having high echo path loss.

Experiment

The primary purposes of the experiment were to determine:

1. Echo tolerance on long delay circuits;
2. The effects of increased echo loss on the quality of long delay circuits having echo suppressors.

Secondary purposes included determination of:

1. The relative quality of simulated terrestrial, one-hop and two-hop satellite circuits;
2. The effects of noise on long delay circuits.

Two experiments were conducted, each involving 50 pairs of subjects, employees of Bell Laboratories, who were friends. They were seated in separate test rooms and were instructed to converse with each other over the test telephones about anything they wished except the experiment. Ten to thirteen experimental conditions were randomly presented, each lasting four minutes. At the end of each condition, both subjects independently rated the condition and indicated difficulty in talking or hearing by checking appropriate boxes on a form similar to the following:

bad			poor			fair			good			excellent		

Yes	No

Difficulty

Experimental circuits

The experimental equipment configuration is shown in Figure 1. Attenuation was incorporated into the four-wire telephone sets to achieve a Send Reference Equivalent (SRE) of 10 dB and a Receive Reference Equivalent

(RRE) of 1 dB. Toll circuit loss, assumed to be 9 dB and arbitrarily split as shown, results in an Overall Reference Equivalent (ORE) of 20 dB which is considered average for a long-distance connection. Sidetone loss was set to 14 dB reference equivalent. The delay, provided by a record-reproduce device, could be set to 0.65 ms, 600 ms or 1200 ms delay. The return loss, consisting of flat attenuation, was varied from 5 to 55 dB where 15 dB is considered to be average for a long-distance connection. As an example, when the return loss was 15 dB, the reference equivalent of the echo path (ERE) was $10 + 4.5 + 15 + 4.5 + 1 = 35$ dB. The standard Western Electric 3A echo suppressors could be switched in or out of the circuit.[1]

White noise was added to provide noise at the receiver equivalent to —60 dBmp at the line terminal of a standard 2-wire 500-type telephone set. This noise is equivalent to about —49 dBm0p. Room noise measured 38 dB(A) at the subject's location. This is considered typical of a quiet room.

Test conditions and results

The experimental conditions and test results are listed in Table 1. In each case the circuit conditions were the same for both subjects. Presented are the mean opinion scores (MOS) and standard deviations (SD) computed from the raw data, the per cent of the subjects having difficulty, and the rating breakdown. In computing the MOS and SD, the 15-point scale inherent in the response form was collapsed so that the three categories of Bad were combined and given an opinion score = 0, Poor = 1, Fair = 2, Good = 3, and Excellent = 4. Shown in Figure 2 are smooth MOS curves derived from the raw data.[2] "Eyeball" smoothed difficulty percentages are shown in Figure 3.

In test number 2, the last 32 subjects were presented with two additional conditions. These conditions A and B were the same as conditions 4 and 15 except that the added circuit noise was removed resulting in essentially a noise-free circuit. The results for these 32 subjects only are listed on Table 1 as test conditions A and B having no noise, and C and D having normal noise. (C and D are subsets of 4 and 15.)

Statistical analysis of results

Significance tests were performed using statistics computed from the raw data using the 15-point scale. Confidence statements based on these tests follow.

1. Replication

Conditions 4 and 5 constitute a replication using the different subjects of experiments 1 and 2. Neither the difference in ratings nor the difference in per cent difficulty was significant.

2. Comparison of average[3] terrestrial circuit (Condition 3) and best condition tested (Condition 1):

— No significant difference in rating;

— No significant difference in per cent difficulty.

3. Comparison of average terrestrial circuit (Condition 3) and average single-hop satellite circuit (Condition 5):

— Difference in ratings significant at 99% level;

— Difference in per cent difficulty significant at 99% level.

4. Comparison of worst echo loss terrestrial circuit (Condition 2) and average satellite circuit (Condition 5):

— No significant difference in rating;

— No significant difference in per cent difficulty.

[1] These WE 3A echo suppressors used speech compressors to provide the receive loss during double talking, and are in accordance with Recommendation G.161.

[2] See Appendix A for smoothing procedure.

[3] Average here implies average ORE and average return loss or ERE.

TABLE 1

Experimental conditions and results

Test condition number	Experiment number	RT prop. time msec	Echo ref. eq. dB	Echo supp.	Computed from raw data MOS	SD	Conv. diff. %	Excellent %	Good %	Fair %	Poor %	Unsatis-factory %
1	1	0	35	OUT	2.59	0.78	2	9	49	34	8	0
2	1	65	25	IN	2.00	0.92	18	6	27	38	16	13
3	1	65	35	IN	2.50	0.88	4	12	39	37	11	1
4	2	600	35	IN	2.14	0.86	18	6	28	45	16	5
5	1	600	35	IN	2.02	0.98	17	4	29	41	17	9
6	1	600	35	OUT	0.28	0.65	70	1	1	2	17	79
7	1	600	45	IN	2.29	0.85	8	4	29	41	17	9
8	1	600	45	OUT	1.19	1.03	43	2	11	18	40	29
9	1	600	55	IN	2.33	0.89	10	6	42	36	13	3
10	1	600	55	OUT	2.24	0.86	12	4	37	43	12	4
11	1	600	65	IN	2.30	0.84	7	5	39	42	14	0
12	1	600	65	OUT	2.42	0.88	11	7	49	28	15	1
13	1	600	75	IN	2.43	0.79	7	7	44	36	13	0
14	1	600	75	OUT	2.46	0.78	9	6	44	42	6	2
15	2	1200	35	IN	1.84	0.98	35	2	25	38	25	10
16	2	1200	45	IN	2.38	0.90	13	7	43	34	13	3
17	2	1200	45	OUT	0.95	0.92	53	1	5	19	38	37
18	2	1200	55	IN	2.37	0.91	15	10	34	42	11	3
19	2	1200	55	OUT	2.10	0.97	18	6	29	40	19	6
20	2	1200	65	IN	2.17	0.85	18	5	27	52	12	4
21	2	1200	65	OUT	2.37	0.81	18	4	45	36	14	1
22	2	1200	75	IN	2.33	0.93	18	5	45	33	12	5
23	2	1200	75	OUT	2.36	0.87	14	7	39	39	13	2
A	2	600	35	IN	1.81	1.46	59	6	28	25	22	19
B	2	1200	35	IN	1.78	1.70	66	6	31	22	16	25
C	2	600	35	IN	2.00	1.22	34	16	16	31	28	9
D	2	1200	35	IN	1.81	1.32	41	19	16	34	28	3

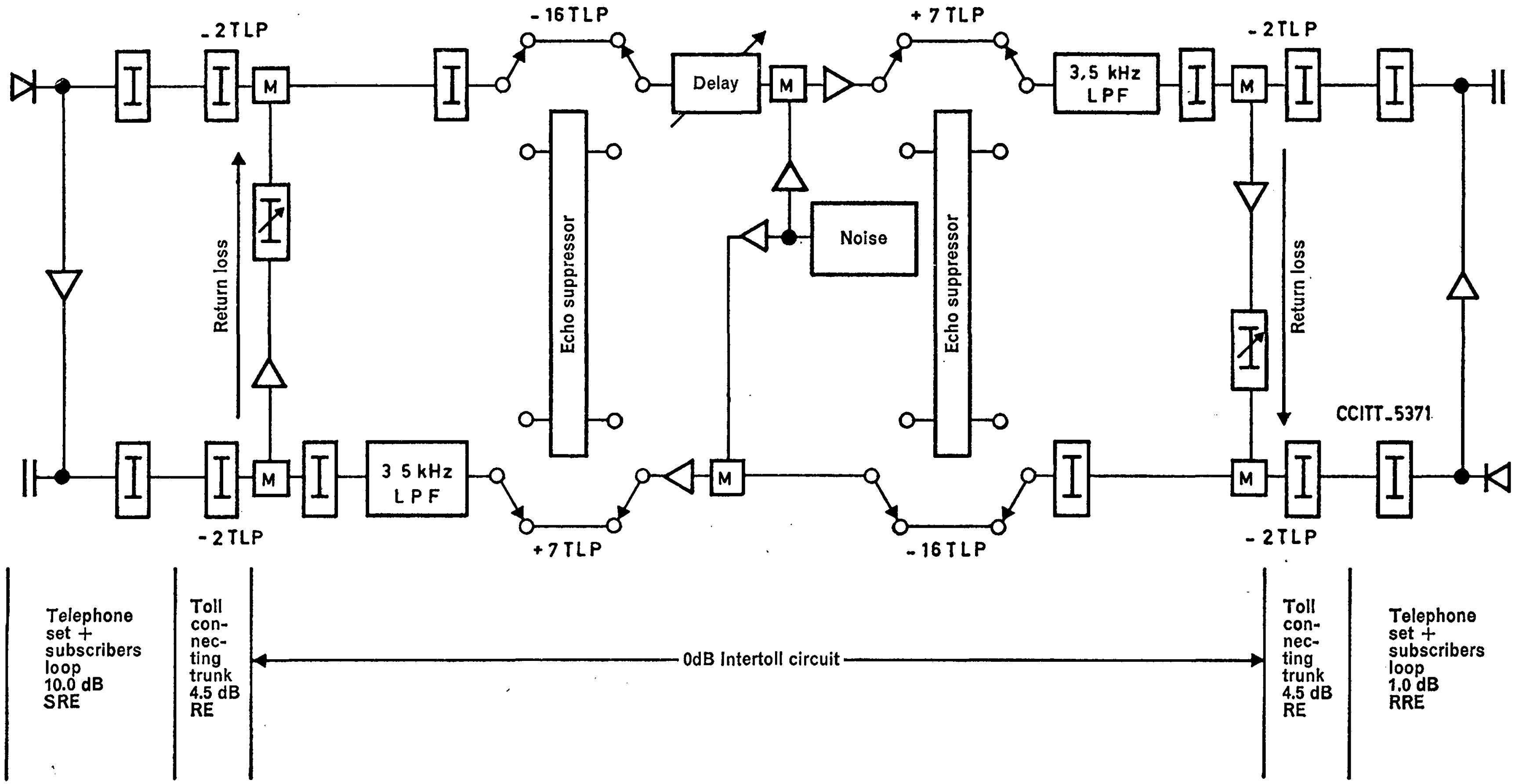


FIGURE 1. — Experimental equipment

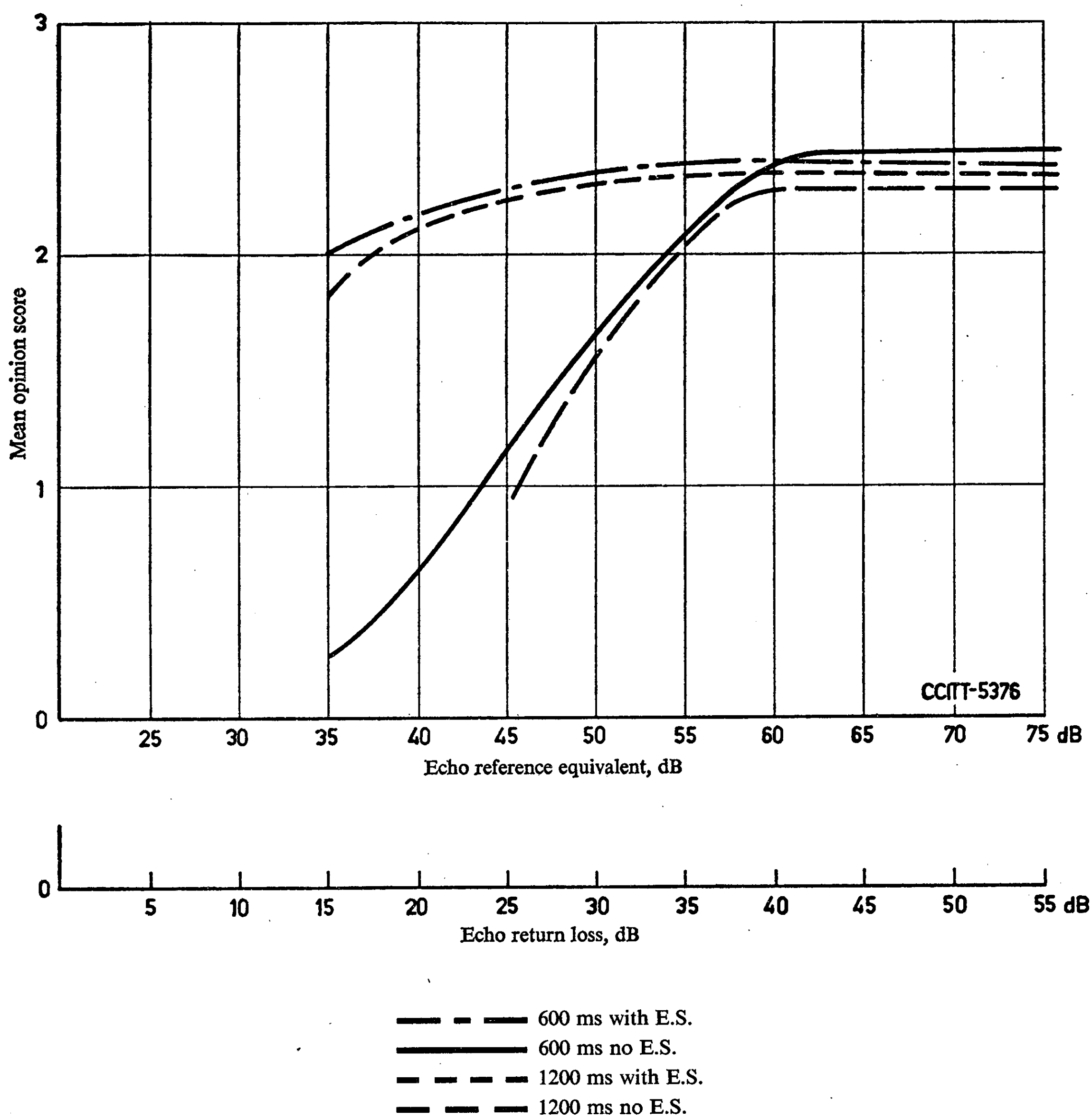


FIGURE 2

5. Comparison of average single-hop satellite circuit (mean of Conditions 4 and 5) and average double-hop satellite circuit (Condition 15):

— Difference in ratings significant at 99% level;

— Difference in per cent difficulty significant at 99% level.

6. Effects of ERE on circuit quality:

For both 600 and 1200 ms RT propagation times, no improvement in circuit quality is obtained by increasing the echo reference equivalent above about 60 dB when no echo suppressors are used or about 45 dB when echo suppressors are used. The circuit quality with and without echo suppressors for high ERE is about the same.

7. Comparison of long-delay circuits with echo suppressors having average echo reference equivalents (35 dB) with those having high echo reference equivalents (≥45–50 dB). The following applies to both 600 and 1200 ms RT propagation time:

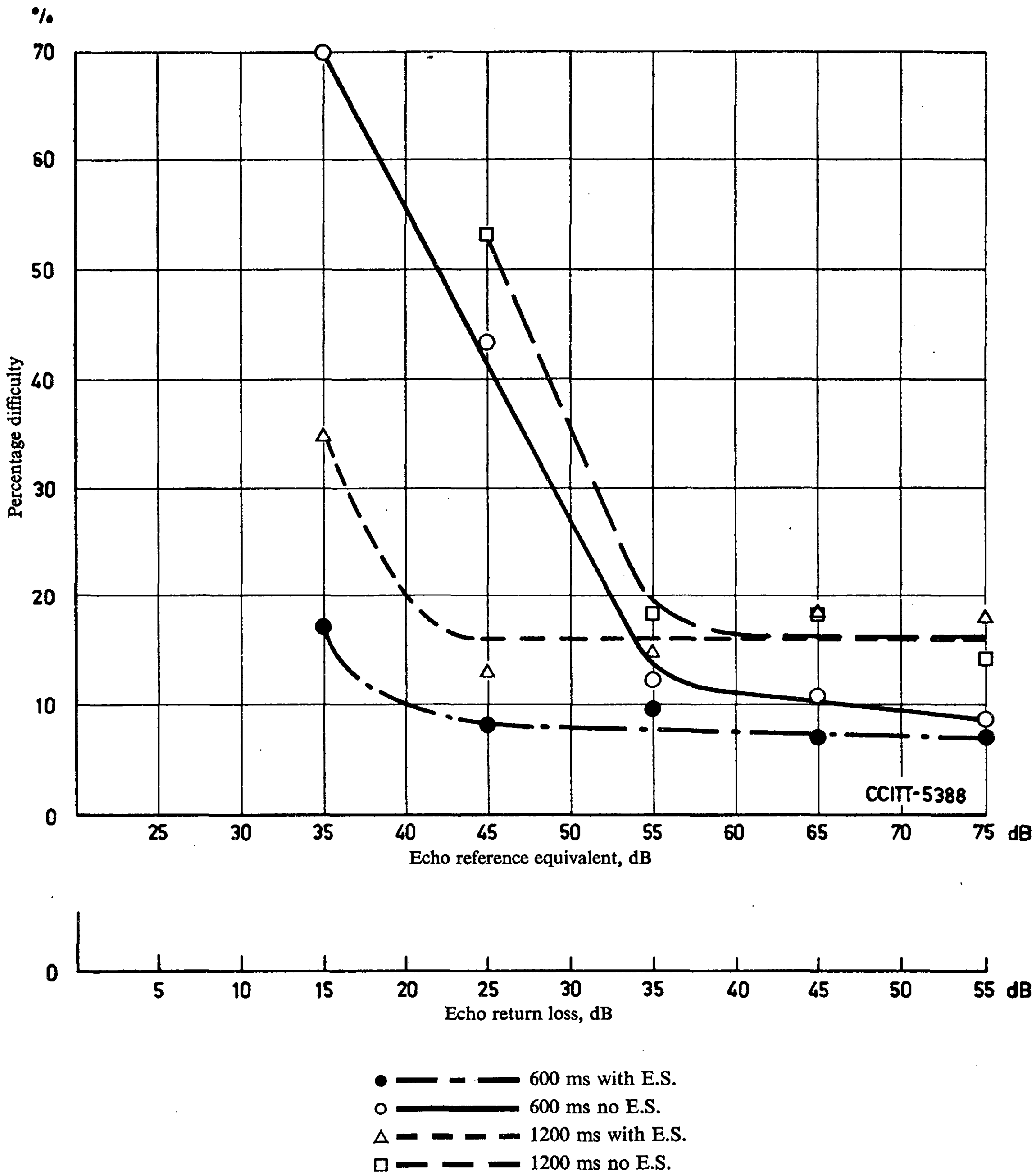


FIGURE 3

— Difference in ratings significant at 90% level;
— Difference in per cent difficulty significant at 90% level.

8. Comparison of simulated one-hop and two-hop satellite circuits having high ERE:

— Difference in ratings not significant;
— Difference in per cent difficulty significant at 95% level.

9. Comparison of average long-delay circuits having no circuit noise (Conditions A and B) with those having normal circuit noise (Conditions C and D) for 32 subjects only:

— No significant difference in rating;
— Difference in per cent difficulty is significant at the 95% level.

Discussion

As would be expected, when no echo suppressors are used there is a high value of echo reference equivalent beyond which no improvement in circuit quality is provided by increasing the ERE. This value, for RT propagation times up to 1200 ms, is about 25 dB or so greater than that provided today on the average call. For EREs below this, there is a rapid decrease in circuit quality.

We also observe that the presence of echo suppressors seems not to be noticed by the subjects when the ERE is greater than about 45 dB (about 10 dB more than that provided on the average call), i.e. the quality of these circuits is nearly the same as that of 4-wire circuits. From the point of view of echo cancellers, this implies that long-delay circuits having echo suppressors and incorporating echo cancellers which provide a modest amount of cancellation, 10 dB for the average return loss case with more for poorer return losses, should provide speech quality similar to that of 4-wire circuits.

The difference in the MOS of simulated terrestrial and satellite circuits agree with similar differences found in real user tests in 1964 and 1965.[1] However, the ratings of the simulated circuits in these laboratory tests are about one point lower than the ratings of similar circuits in real user tests. The reasons for this difference are not known.

Late in test number 2 it was postulated that the circuit noise, by itself, was sufficient to cause the laboratory subjects to give the circuits lower ratings. To investigate this, the last 16 pairs of subjects were presented with two very low noise conditions. The ratings of the low noise conditions were consistently lower than the normal noise conditions although the differences are not significant. The differences in difficulty percentages are significant. Thus, the low ratings in the main test were not due to the added noise; indeed, the noise may have raised the ratings by masking the echo. This finding may have some bearing on the quality of future low noise p.c.m. satellite circuits.

APPENDIX A

Derivation of smooth mean opinion score curves

1. The raw data are identified as belonging to one of four classes of condition, e.g. 600 ms with E.S., 600 ms no E.S., 1200 ms with E.S., and 1200 ms no E.S.

2. For each condition in a class, e.g. 600 ms with E.S. at 35 dB ERE, the raw data are put in the form of a vote histogram and the MOS and SD are determined (presented in Table 1).

3. A normal density function with mean, μ, and standard deviation, σ, is fitted to the raw data histogram for each condition so that this function produces the same MOS and SD as the raw data.

4. A weighted average of the σs in a class is found yielding a compromise σ for the class.

5. The mean of the normal density function for each condition in the class is then adjusted until a new normal density function having the compromise σ yields the same MOS as the raw data.

6. The resultant means in a class are then least squares fitted to an arbitrary smooth function of the echo return loss. The smooth function has the form

$$\mu = C_1 — C_2 \, (60\text{-ERL* } C_3)$$

where * denotes the power sum and the values of the constants evolve out of an iterative process of minimizing the least square error.

7. The MOS for each condition is then recomputed using the resultant smoothed mean and compromise σ. The smoothed MOSs are shown in Figure 2.

This method of smoothing was also used in analyzing the results given in Annex 4 above.

[1] C.C.I.T.T. Volume V*bis* *(Red Book)*, Annex E.

REFERENCES

[1] Supplement No. 1 in Volume V of the *White Book* (Tel. Assoc. of Canada).
[2] Supplement No. 2 in Volume V of the *White Book* (K.D.D., Japan).
[3] HUTTER, J.: Customer response to telephone circuits routed via a synchronous-orbit satellite; *P.O.E.E.J.*, Volume 60, p. 181, October 1967.
or RICHARDS, D. L.: Transmission performance of telephone circuits having long propagation times; *Het PTT-Bedrijf*, Volume XV, p. 12, May 1967.
[4] HELDER, G. K.: Customer evaluation of telephone circuits with delay; *B.S.T.J.* 45, September 1966, pp. 1157–1191.
or KARLIN, J. E.: Measuring the acceptability of long delay transmission used during the "Early Bird" transatlantic tests in 1965; *Het PTT-Bedrijf* (English edition), May 1967.
[5] COM XII–No. 113 (COM XV–No. 149), Annex 3 (United Kingdom).
[6] COM XII–No. 52 (COM XV–No. 93), Annex 6 (A. T. & T. Co.).

Question 7/XII — Determination of transmission quality by objective measurements

(Continuation of Question 7/XII, studied in 1968–1972)
(Documentary question)

a) What criterion should be adopted to determine if different transmission systems have equal performance?

b) What practical method, based on objective measurements, should be specified for the determination of the transmission performance of local telephone systems?

Note 1. — The following Annex 1 gives a classification of methods relevant to this question.

Note 2. — Attention is drawn to the importance of using suitable methods for preparing test programmes and for analyzing the results.

Note 3. — The problems connected with methods of assessing ratings based on loudness are the subject of Question 15/XII. Question 7/XII should therefore be confined to the study of methods other than those using loudness ratings. Attention is drawn also to Question 2/XII.

ANNEX 1

(to Question 7/XII)

Classification of assessment methods

Part a

The first essential of a method for measuring the quality of a telephone circuit is that it should give results which correspond to the experience of any user employing the telephone for the purposes of everyday life. Furthermore, the method should be both simple and practical.

The subjective testing methods at present recommended or under study by the C.C.I.T.T. are based on the following criteria:

a.1 *Loudness comparison for speech*

The only method of this type recommended by the C.C.I.T.T. is that of reference equivalents, which does not perhaps fully satisfy the first condition mentioned above. The possibility of improving this method will be investigated under Question 15/XII.

a.2 *The A.E.N. method*

This is no longer the basis of international recommendations because it is not sensitive enough for modern sets. However, articulation tests are still carried out for the study of certain particular cases (see Question 18/XII).

a.3 *Opinion tests*

Although this method does not form the subject of a C.C.I.T.T. Recommendation, it has already been used to study the effect of circuit noise and compandors (see Annexes B and D in Volume V*bis* of the *Red Book*). Study Group XII has advised its use in studies under way and some Administrations have used it (Questions 6a, 13, 18/XII).

a.4 Methods of assessing service quality contemplated in Question 2/XII, for example, observations made by third parties by means of interrogations, taking as criteria repeats, difficulties reported, etc. Some of these methods have already been used for the study of Question 6/XII (see documentation referred to in Recommendation P.14) and also in [20], [21], and [22]). Methods of this type are also described in the former Annexes to Question 2/XII (*White Book*, Volume V).

Part b

An attempt has been made to work out objective measurement methods that yield the same results as some of the methods mentioned under *a*; this involves developing objective measuring methods and presupposes i) a theory to relate the measurement results to the subjective factor taken as criterion, and ii) practical procedures for the correct measurement of the physical magnitudes considered. A general solution is not to be expected on this last point until a reply is given to Questions 12 and 8/XII.

There are also general methods for making objective measurements of transmission quality that are not connected with any of the existing subjective testing methods.

The status of work on objective methods for measuring transmission quality is outlined below. The order of the paragraphs of part *a* has been kept as far as possible.

b.1 *Methods based on loudness*

b.1.1 The theory of "objective reference equivalents" has been known for a long time ([1], [2], [3]). An attempt is being made to establish, on the international level, a relationship between the results measured objectively and subjectively on various types of sets (Question 15/XII).

b.1.2 Another possibility is to try to develop, on the basis of loudness, an absolute method comparable to that used in the calculation of line attenuation. Such methods are set forth in [4] and Annex 6 to Question 15/XII (*White Book*, Volume V), which could serve as a basis for developing similar methods for application to b.2.

b.2 *Methods based on articulation or information flow*

b.2.1 There are several theories, which are more or less equivalent, for calculating the articulation [5] to [8]. The constants of the theoretical formulae have been deduced from the articulation tests, which leads in practice to "tailored" methods, each of which is applicable to one type of telephone set. Articulation measurement methods deduced from these theories are described in [10] and [11].

b.2.2 Instead of trying to calculate the articulation for logatoms or sounds, one can also try to evaluate the information transmitted in word components. An objective measurement method based on this principle is described in Chapter 3.8 of [26].

b.2.3 The method described in [12] can be considered an absolute method since it calculates the information transmitted from objective measurement data. Although this theory applies to a one-way transmission channel, it yields results that correlate with those obtained with opinion tests for a type of set for which measurements performed in relation with a type b.2.1 theory are available. It will be impossible to get away from this limitation before a reply has been given to Questions 12 and 8/XII.

The United Kingdom Post Office has developed a method for the objective measurement of the lines and the electrical assembly of a set which, for a given type of microphone and receiver, yield results that correlate with AEN measurements and opinion test data [13] and [27]. It can be shown that the weighting of the frequencies used in this method is similar to that calculated in [12].

b.3 It appears to be very difficult to devise an objective measurement method based on the principle used in opinion tests, which rely on two-way conversations and take account of the mutual reactions of the two individuals conducting the conversation.

b.4 The considerations mentioned in b.3 also apply to the objective testing methods mentioned in a.4.

ANNEX 2

(to Question 7/XII)

Reply to the question given by Study Group XII at the end of the 1968–1972 period

Question 7/XII is an extremely general one and has been under study by Study Group XII for a very long time. In the course of this study many valuable contributions have been made and two specific problems have been formulated which are now subjects of separate study under Questions 2/XII (General assessment criteria) and 15/XII (Problems specifically related to loudness as a criterion).

Undoubtedly there are many more important aspects of the broad problems contained within Question 7/XII which it would be valuable to continue to study. To assist in clarifying the future course of such work, a distinction under the following two headings is suggested:

1. Feasibility of measuring instrumentation related to the *general* assessment criteria that might be formulated under Question 2/XII.
2. Measuring instrumentation and supporting theory related to *specific* criteria such as articulation, listening opinion tests, determination of: speech losses of sidetone, echo, crosstalk and other specific paths of speech communication systems. (Note that the criterion of received loudness of speech is being treated separately under Question 15/XII.)

It will be extremely difficult to achieve success under heading 1 (as mentioned in b.3 and b.4 of Annex 1) but the matter is extremely important in the long term and the difficulties ought to be thoroughly appraised. Annex 3 falls under this heading and contains some suggestions for future work. There have been no other contributions relevant to this heading. It is suggested that the study ought to be treated as one that will eventually lead to some international agreement although it will take a long time.

It is recommended that study of the question should be continued but that it should be treated, for the present, as documentary. Further contributions are invited, with special attention being paid to material already published and accessible. Some references have been assembled in Annex 4.

Perusal of such information could lead to the definition of practical calculation methods which might then be studied after Question 15/XII has been disposed of.

ANNEX 3

(to Question 7/XII)

Feasibility of automatic detection of loss of communication efficiency of speech links (Shortened version of paper presented to Fourth Symposium on Human Factors in Telephony, 1968, to be published)

(Contribution of the United Kingdom Post Office)

Measurements to determine the communication efficiency of a speech link or telephone connection [19][1] may be considered to fall into four basic categories, namely:

a) Examination of the speech link itself. This involves measurement of the physical magnitudes of the individual causes of loss of efficiency, such as loss, noise, distortion, etc.;

b) Measurement of the information capacity of the speech link in terms of its potential ability to transmit specified test signals chosen to represent particular features considered important in the transmission and reception of speech;

c) Examination of conversations between subjects who are actually using the speech link;

d) Examination of the subjects who are conversing over the speech link. This may take place either during the conversations or immediately afterwards.

Because of the complexity of the problems, it is intended here to confine discussion to phenomena upon which instrumentation might be based for connection to actual telephone networks so as to indicate any need for remedial action to restore the intended standard of transmission quality of connections being used. It is therefore sufficient to consider those features that indicate serious loss of efficiency and to initiate remedial action when the proportion of telephone calls so affected exceeds some predetermined critical value. For other purposes, it may be desirable to study methods that could indicate transmission efficiency over a much wider range.

[1] The references are to the bibliography in Annex 4.

Although methods based on category c) present formidable practical problems, they seem to offer the only real hope of devising a method of providing surveillance over complete telephone connections by use of observation locations relatively remote from the extremities of the connection. There is hope, also, that a single observation location would suffice.

Features of conversation structure worthy of examination

The features of the conversations that are examined must be those affected quantitatively in a characteristic manner closely associated with loss of efficiency of the connection as a whole or the incidence of unsatisfactory conversations experienced by the participants. The features examined must also, in principle, be affected to the appropriate extent by all possible causes of loss of efficiency. In practice, however, it is unlikely that these requirements will be completely satisfied by examination of any single feature; several features will need to be examined and a suitable combination made of the resulting changes in each.

Physical quantities that could be observed from a near-central location in telephone connections can be distinguished in type according to the following classification:

1. amplitude statistics;
2. time statistics;
3. frequency spectrum statistics.

1. *Amplitude statistics*

The simplest example of this class is speech volume. When this is measured at a point in a telephone connection definitely related in respect of loss[1] to the terminals of the talker's microphone, the value will increase as the opinions formed of the connection by the participants become more unfavourable. Figures 1a) and 1b) show talker volumes as functions of the percentage of unfavourable opinions for two different causes of loss of efficiency. The results in Figure 1a) related to a high-quality speech link degraded by random interruption of the two speech paths at a mean rate of about 500 per second, the extent of the degradation being governed by varying the speech-time fraction. In this case raising their speech volumes confers no advantage on the participants and seems to be a purely unconscious response. In Figure 1b) the speech link was of ordinary telephone sets, in rather loud room noise with varied loss between them; in this example the increase in speech volume does confer some advantage (although the amount goes very little towards offsetting the transmission loss) and the rate of change is greater.

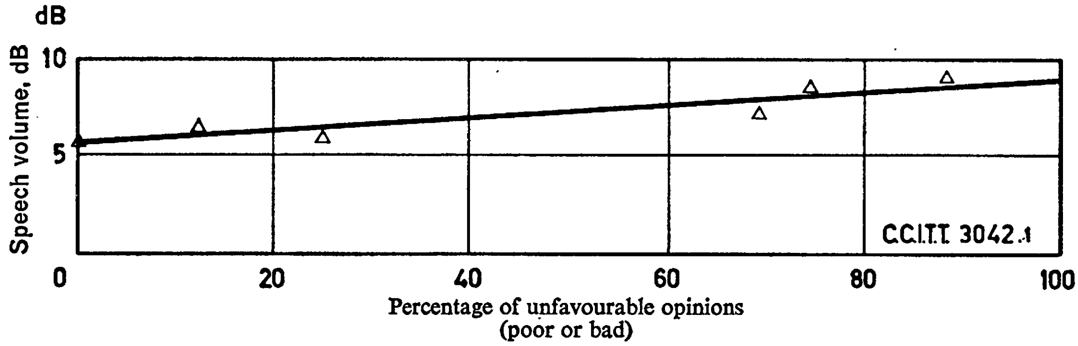
It is also interesting to note that the topic of conversation has quite a marked effect upon the speech volume.

2. *Time statistics*

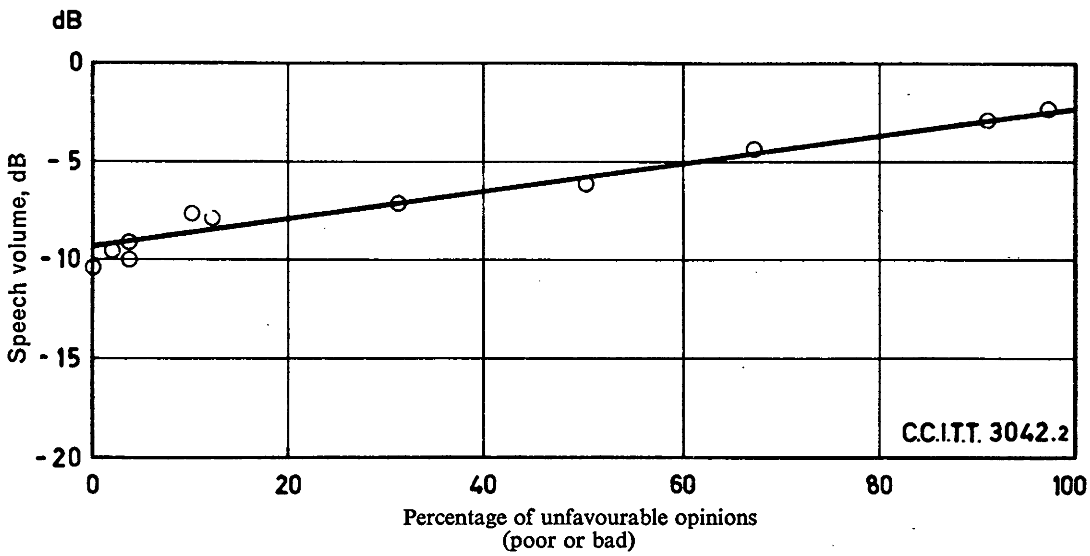
These include a range of time intervals, for example, the total duration of the conversation, incidence and duration of pauses, rate of emission of syllables, etc.; consideration must be given to both directions of the conversation to include the relative timing of events in the two directions. Figure 2 shows the time required to complete a defined task requiring conversation over a speech link as a function of the percentage of unfavourable opinions expressed afterwards by the participants. It is believed that this relationship is largely independent of the type of degradation, but it will be noticed that the time starts to increase only after the percentage of unfavourable opinions has become rather large; only in a few exceptional cases would a public telephone connection become so degraded as to cause this percentage to rise as high as 50% and then only when factors outside the control of the telephone Administration (such as ambient noise) were particularly adverse as well as those (like loss) that the Administration could control.[2] When a connection is very much degraded, the additional time required is partly due to a lower rate of talking but also to more circumlocutory phrasing and to the need for more requests for material to be repeated. The effects have not been studied separately.

[1] It may be difficult to choose an observation location that is both nearly central and at which the losses from the talking locations are known.

[2] A pair of telephone sets having exceptionally poor sidetone characteristics situated in loud (60 dB) room noise and connected through lines with very high losses to give an overall reference equivalent of 40 dB would correspond to a standard yielding about 50% of unfavourable opinions.



a) **Connections between high-quality transducers but with various degrees of non-linear distortion in the transmission paths**



b) **Commercial quality telephone connections with different losses in the transmission paths**

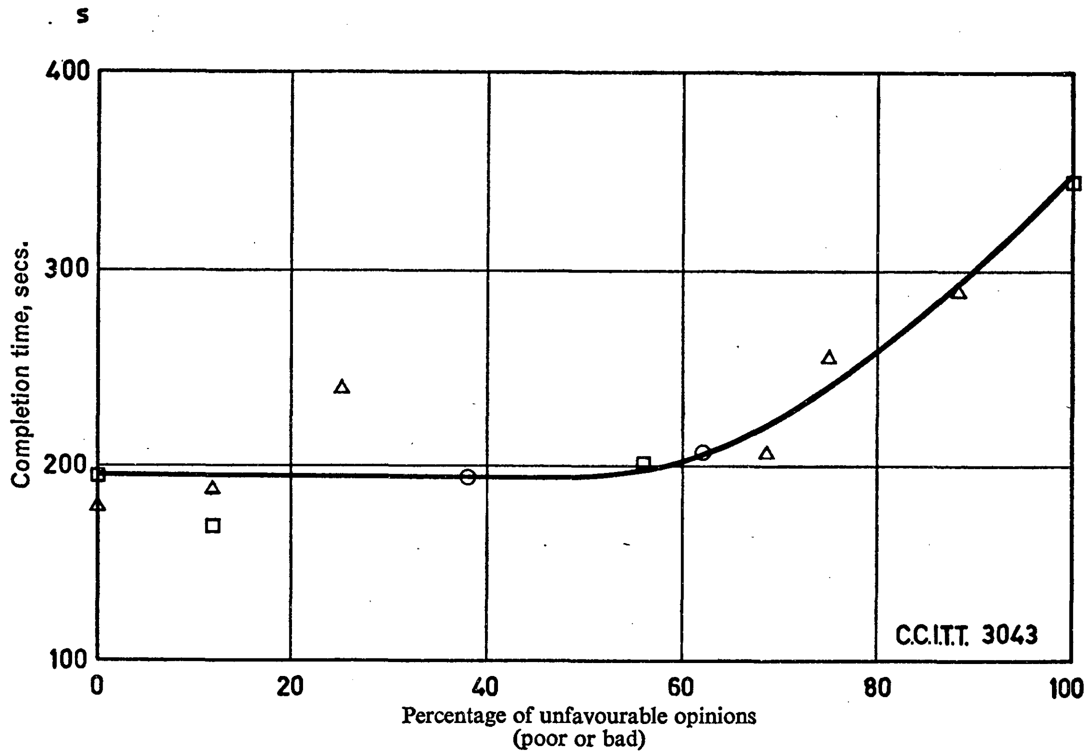
FIGURE 1. — **Speech volume as a function of the percentage of unfavourable opinions**

Care will be needed in making use of time statistics of a crude form derived under actual service conditions because experience of unfavourable connections can cause customers to terminate their business sooner than otherwise (perhaps to make a new connection) and so give rise to an effect apparently opposite to that explained above. The most promising type of time statistics for the present purpose are those such as have already been found useful in studying the effects of long propagation time; it is possible to detect the occurrence of confusion in conversation by observing the alternations of speech utterances and noting when these become disrupted in a characteristic manner. Figure 3 shows some results from this study [21]. It has also been shown that there are characteristic relationships between the incidence of short pauses in the speech of one person when he is exerting intellectual effort while speaking [25]. Remembered material, which does not require such intellectual effort, is spoken more fluently than a spontaneous commentary on a similar subject.

3. *Frequency spectrum statistics*

This heading is intended to cover a wide range of statistics, from simple ones involving the mean frequency spectrum of the speech signals on the line to quite sophisticated analysis of the speech by methods such as are used in vocoder transmission systems.

It is known that the mean frequency spectrum of speech spoken loudly differs from that of speech spoken normally by the same talker; whispering also changes the shape of the mean spectrum. Speaking loudly mainly involves increasing the power levels of the vowels, the consonants remaining almost unchanged in level [14], [15]. The mean frequency spectrum of speech differs also with the identity of the talker, in particular between men and



△ Non-linear distortion
□ Circuit noise with "constant volume" speech
o Commercial telephone connection with varied loss

FIGURE 2. — **Time to complete a task involving conversation as a function of the percentage of unfavourable opinions**

women; the transmission characteristics of the connection between the talker and the observation point may also change the spectrum considerably. Examination in any simple manner of the mean frequency spectrum is not, therefore, likely to yield results that show useful correlations with the opinions of the users of the connection.

A more profitable line of study under this heading would seem to be to select some definite characteristic of speech such as the frequency of the fundamental pitch of the larynx and to seek among the patterns of these some changes that are characteristic indicators that difficulty is being experienced. No more specific guidance can be given at present.

Proposals for preliminary evaluation of prospects

It is clear that a great deal of work will be necessary before it can be established whether a method could be developed on which complete reliance could be placed and which would correctly distinguish "unsatisfactory" connections (or conversations) from satisfactory ones. Although the method is eventually required for use in the field, a start must be made under the controlled conditions of the laboratory where "favourable" and "unfavourable" can be distinguished by accepted means (such as by interrogating the participants). The initial work would therefore consist of assembling a collection of recorded conversations that had taken place under known conditions and examining the correlations between suitably chosen statistics (for example, those indicated above) and the opinions formed by the participants themselves. The recordings must be made in such a way that the required observations can subsequently be performed on them; a multi-channel (probably 4-channel) magnetic tape recorder will facilitate the association of pertinent indications and other information with the recorded conversation.

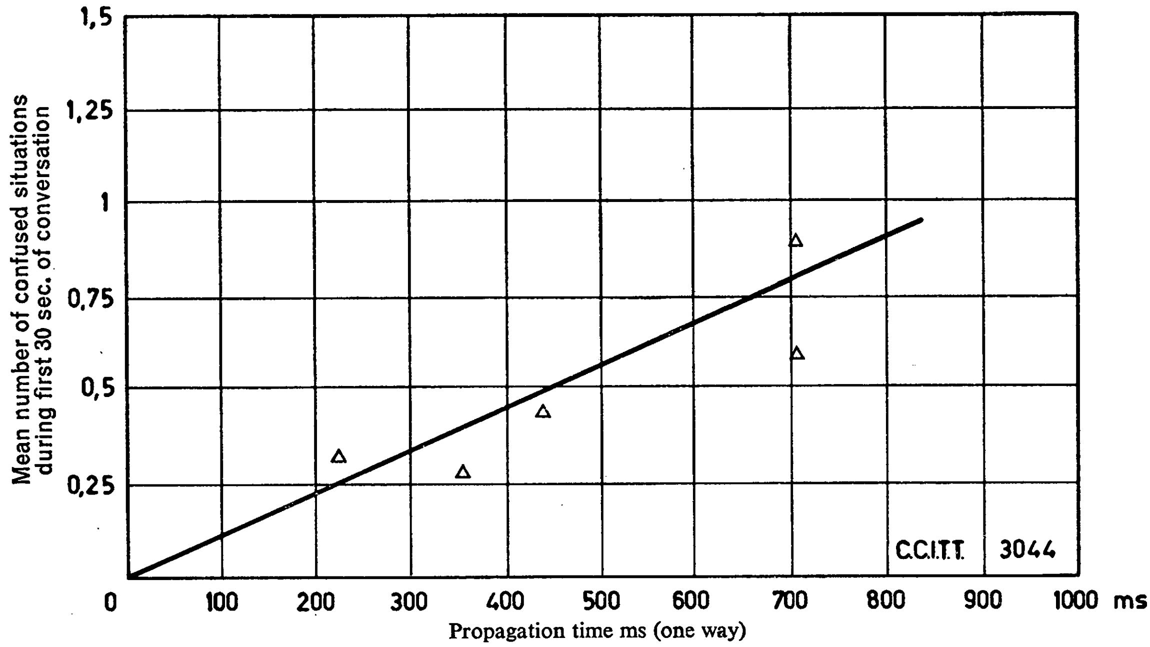


FIGURE 3. — **Incidence of confused situations in conversation over connections having long propagation times**

Note. — The connections used were laboratory speech links free from echo and without echo suppressors.

ANNEX 4
(to Question 7/XII)

BIBLIOGRAPHY

[1] BRAUN, K.: Die Bezugsdämpfung und ihre Berechnung aus der Restdämpfungskurve (Frequenzkurve) eines Übertragungssystems; *T.F.T.*, Volume 28, pp. 311–318, August 1939.

[2] BRAUN, K.: Theoretische und experimentelle Untersuchung der Bezugsdämpfung und der Lautstärke; *T.F.T.*, Volume 29, pp. 31–37, No. 2, 1940.

[3] BLYE, P. W., COOLIDGE, O. H., & HUNTLEY, H. R.: A revised telephone transmission rating plan; *B.S.T.J.*, Volume 34, pages 453–472, May 1955 (reproduced in the *Red Book*, Volume I, pages 636–651, and Volume V, pages 607–624).

[4] BRAUN, K.: Image attenuation of microphone and receiver insets; *N.T.Z.*, 1960, No. 8, pages 365–370 (translated in the *Red Book*, Volume V*bis*, pages 255–265).

[5] FRENCH, N. R. and STEINBERG, J. C.: Factors governing the intelligibility of speech sounds; *J.A.S.A.*, Volume 19, page 89, January 1947.

[6] RICHARDS, D. L. and ARCHBOLD, R. B.: A development of the Collard principle of articulation calculation; *P.I.E.E.*, Volume 103, part B, September 1956 (*Red Book*, Volume I, pages 681–707, Geneva, 1956).

[7] Contribution by the Italian Administration to the study of objective methods for measuring reference equivalent and articulation reference equivalent; *Red Book*, Volume I, pages 660–680, Geneva, 1956.

[8] FLETCHER, H. and GALT, R. H.: The perception of speech and its relation to telephony; *J.A.S.A.*, Volume 22, March 1950, page 89 (reproduced in [9], Chapter 15–17).

[9] FLETCHER, H.: *Speech and hearing in communication*, D. Van Nostrand, New York, 1953.

[10] Tonality method studied by the U.S.S.R. Administration to determine articulation; *Red Book*, Volume V, pages 516–524.

[11] Method used by the Swiss Telephone Administration for the determination of transmission quality based on objective measurements; *Red Book*, Volume V, pages 496–516.

[12] LALOU, J. Calculation of telephone transmission performance by information theory; *Red Book*, Volume V*bis*, pages 168–199.
[13] Methods used by the British Telephone Administration for rating telephone speech links; *Red Book*, Volume V, pages 167–173.
[14] SILVIAN, L. J.: Speech power and its measurement, *B.S.T.J.*, 1929, *8*, p. 646–661.
[15] LOYE, D. P. and MORGAN, K. F.: Sound picture recording and reproducing characteristics, *J. Soc. Motion Picture Engineers*, 1939, *32*, pp. 631–647.
[16] RICHARDS, D. L.: Some aspects of the behaviour of telephone users as affected by the physical properties of the circuit, Communication Theory, 1953, *Butterworths Scientific Publications*, pp. 442–449.
[17] ZAITSEV, T. E.: Correlation method for determining the fidelity and intelligibility of speech transmitted over telecommunication channels, *Elektrosvyaz*, 1958 (10) pp. 38–46.
[18] LICKLIDER, J. C. R., BISBERG, A. and SCHWARZLANDER, H.: An electronic device to measure the intelligibility of speech. *Proc. Nat. Electronics Conf.*, 1959, *15*, pp. 329–334.
[19] RICHARDS, D. L. and SWAFFIELD, J.: Assessment of speech communication links, *P.I.E.E.*, 1959, *106B*, pp. 77–89.
[20] RICHARDS, D. L. and BUCK, G. A.: Telephone echo tests, *P.I.E.E.*, 1960, *107B*, pp. 553–556.
[21] RICHARDS, D. L.: Conversation performance of speech links subject to long propagation times, International Conference on Satellite Communication (*Inst. Elec. Engrs.*, London, 1962), pp. 247–251.
[22] RICHARDS, D. L.: Transmission performance of telephone connexions having long propagation times, *Het PTT-Bedrijf*, 1967, *15*, pp. 12–24.
[23] BOERYD, A.: Subscriber reaction due to unbalanced transmission levels, *ibid*, pp. 39–43.
[24] RICHARDS, D. L.: Distortion of speech by quantizing, *Electronics Letters*, 1967, *3*, pp. 230–231.
[25] GOLDMAN-EISLER, F.: Sequential temporal patterns and cognitive processes in speech, *Language and Speech*, 1967, *10*, pp. 122–132.
[26] RICHARDS, D. L.: Telecommunications by speech, Butterworths, London, 1973.
[27] UNITED KINGDOM POST OFFICE: COM XII—No. 112 (1968–1972 period) concerned with estimating mean opinion from objective transmission measurements of mouth-ear paths.

B.S.T.J. = Bell System Technical Journal
J.A.S.A. = Journal of the Acoustical Society of America
N.T.Z. = Nachrichtentechnische Zeitschrift
P.I.E.E. = Proceedings of the Institution of Electrical Engineers
T.F.T. = Telegraphen-, Fernsprech-, Funk- und Fernseh-Technik

Note. — Although references [14] to [19] and [23] to [25] are not cited in the Annexes, they have been retained because of their general interest.

Question 8/XII — Measuring the efficiency of a carbon microphone

(Continuation of Question 8/XII, studied in 1968–1972)

The shape of the sensitivity/frequency characteristic of a sending system depends very much on the artificial mouth used and also upon the method of measurement employed.

Similarly, the shape of the sensitivity/frequency characteristic of a receiving system depends upon the artificial ear used and also upon the method of measurement employed.

What methods of measurement should be recommended for tracing these curves and what accuracy should be recommended for these measurements?

Note. — In studying this question, account should be taken of Recommendation P.75.

The attention of Administrations is also drawn to the following documentation (period 1964–1968):

1. COM XII-No. 28 (Australia, Sweden): with the following corrections to the English text: on pages 2 and 3, replace kc/s by Hz throughout and indicate the sound pressures: 0.1; 1; 3; etc. COM XII—No. 76 (Helsinki Telephone Co.)

2. For the period 1968–1972, COM XII-No. 20 (S.I.P. and S.T.E.T., Italy).

3. I.E.E.E. Standard 269–1971: "I.E.E.E. standard method for measuring transmission performance of telephone sets", I.E.E.E., Stockholm.
4. Annexes 27 to 31 (*Red Book*, Volume V, Part II).
5. Annex to Question 8/XII (*White Book*, Volume V).
6. Annexes 1 and 2 to Question 12/XII.

Question 9/XII — Sidetone

(New question)

Considering that:

1. Reference equivalents determined in accordance with Recommendation P.73 and used hitherto to rate sidetone levels of subscribers' telephone sets are rather unsatisfactory because the frequencies most contributing to loudness (the low frequencies) are those which are most subject to masking by the talker's own speech via bone conduction and other paths;
2. The new loudness rating proposals, if applied to sidetone paths may also be subject to the same objections, but in addition could introduce other problems associated with the threshold of detection of sidetone while speech is being uttered, and the response of the IRS used to determine the loudness rating;
3. Recommendation P.73 takes no account of the mechanical and acoustical transmission of sound along the handset to the receiver.

The following questions should be studied:

a) Is it appropriate to rate the equivalent loss of the sidetone path of a subscriber's set in terms of:

1. Some form of loudness rating? (see Question 15/XII);
2. Preferred and limiting frequency characteristics? or
3. Some other means?

Whichever method is used what numerical values should be recommended to replace those contained in Recommendation G.121 (P.21)?

b) In view of the effects of listener sidetone in which local room noise heard in the telephone receiver via the sidetone path may mask speech signals received from the line, and which may demand a different set of rating criteria and calculation procedures from those applying when a subscriber speaks, what recommendations, additional to those in reply to Part a) of this question, should be followed to ensure that listener sidetone does not adversely affect listener opinion?

Note. — In seeking an answer to this question it will be desirable to determine:

1. The frequency characteristic of the human sidetone referred to the mouth reference point, and measured in the ear-canal at the ear over which a telephone receiver is placed.
2. The masking threshold levels for different frequencies (or narrow frequency bands), determined for an ear over which a receiver is placed, when the subject is speaking.
3. The effects of level and frequency spectrum relating to objectional sidetone.
4. The conditions under which room noise received via the sidetone path has an adverse effect on the listener's opinion.

Question 10/XII — Increase in the sensitivity of local systems

(Continuation of Question 10/XII, studied in 1968–1972)

(Documentary question)

Considering that modern developments have enabled considerable improvements to be made in the sensitivity of telephone sets and that even further increases in sensitivity can readily be achieved, it is desirable to examine the consequences of such increases in sensitivity and the manner in which they may be turned to advantage.

In extracting advantages from these possibilities care is necessary to ensure that certain disadvantages do not follow (such as excessive levels on circuits, excessive loudness of received speech, excessive sidetone, excessive echo, excessive crosstalk, etc.).

Clearly, compromises are necessary to balance the advantages and disadvantages.

What transmission characteristics are desirable for subscribers' sets so that the best compromise will be obtained under conditions likely to apply in future telephone networks?

Note 1. — The fundamental characteristics to be considered in the specification of such subscribers' sets include the following, which must be expressed as functions of line current:

a) sensitivity/frequency characteristic, sending;
b) sensitivity/frequency characteristic, receiving;
c) line terminal impedance;
d) sidetone balance impedance.

The other transmission characteristics of importance can be derived from these.

Note 2. — The above-mentioned characteristics relate to conventional two-wire handset telephone sets. Additional characteristics would need to be considered for other types of subscribers' telephone equipment, e.g. loudspeaker telephones (Question 17/XII).

Note 3. — In the Annex there is reported the effect on speech volume due to microphones with an increased sensitivity.

ANNEX

(to Question 10/XII)

Consequences of increased efficiency of microphone capsules

(Contribution by the Administration of the Federal Republic of Germany)

To examine the consequences of increased efficiency of microphone capsules, it is recommended that in telephone sets on which tests are to be carried out microphone capsules with varying degrees of sensitivity should be used without letting the subscribers know that capsules have been changed. The volume of speech sounds produced by subscribers when they speak may be determined by volumeter observations. The mean statistical value obtained by these measurements enables the relation between volume and efficiency of the microphone capsule (sending reference equivalent) to be established.

The volume of speech currents transmitted by the telephone set equipped with test capsules must be measured at the ends of the line to the telephone set. For the purpose of changing microphone capsules without the subscriber's knowledge, extension telephone sets connected to a large office of the Administration were used. The users of these sets were not employed in the technical telephone services.

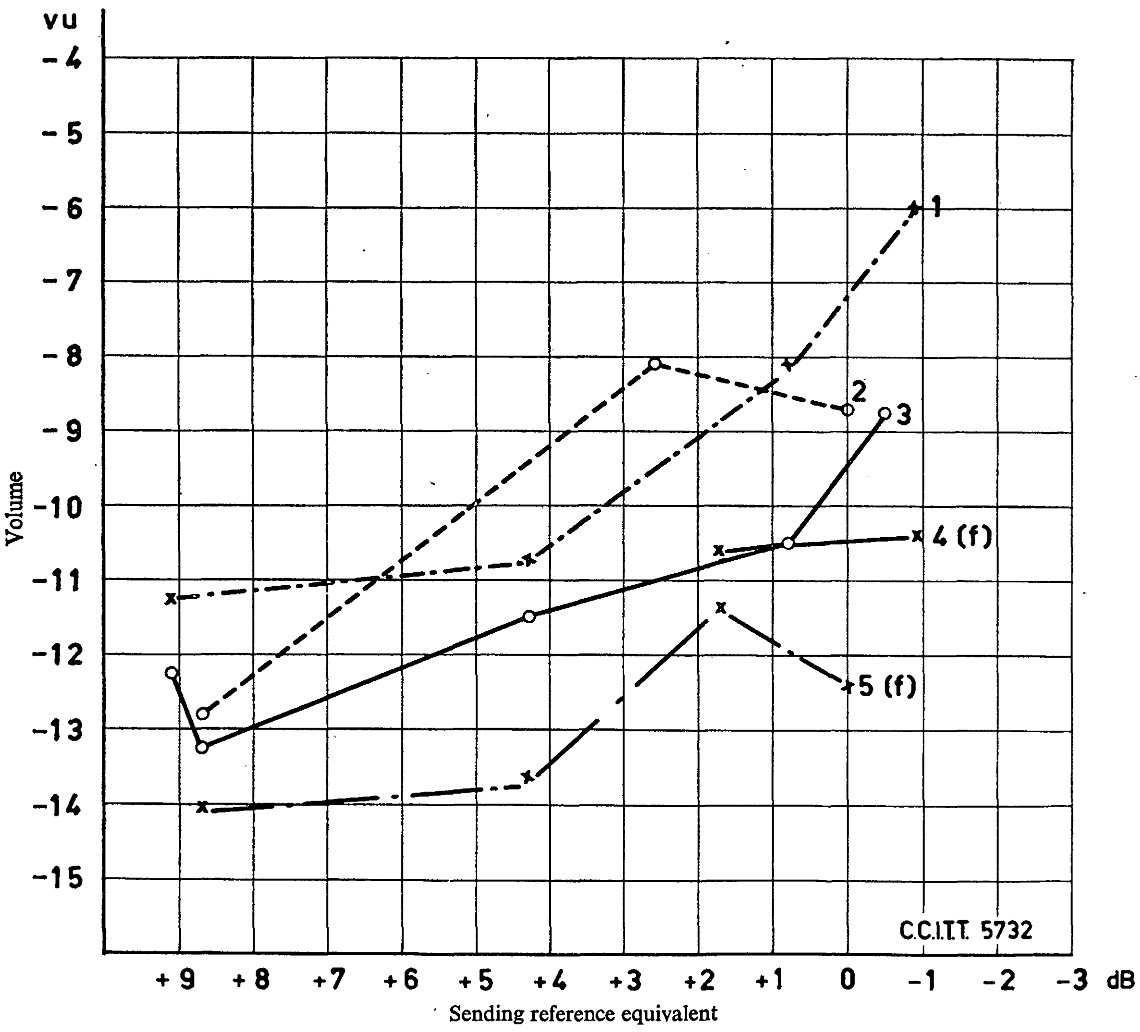
These telephone sets were equipped with microphone capsules having reference equivalents between —1 dB and +9 dB. The users of the sets in question usually made calls only within the building, or trunk calls. There was a percentage of about 40 to 60% trunk calls (an average of about 55%). The percentage of local calls was very small, i.e. less than 5%. The reference equivalent for trunk calls measured with microphone capsules having a high reference equivalent (worse) was less than 30 to 35 dB. The difference in volume between trunk calls and calls exchanged within the building was very small (< 1 dB). This result is explained by the fact that trunk calls were also exchanged with a fairly high sound intensity.

The figure below shows the statistical values of 50% of volume measurement results as a function of reference equivalent for sending for various telephone sets. The reference equivalent of microphone capsules used in telephone set equipment is shown by a dot or a cross. Users of telephone sets 1, 2 and 3 were men, while those using sets 4(f) and 5(f) were women.

The shape of the curves is somewhat irregular, some probably being caused by manufacturing differences and others by differences between the old- and new-type microphone capsules; the observations were also carried out with some old-type capsules which were already in service in the telephone sets. In spite of this, an increase in volume as a function of increase in efficiency (decrease in sending reference equivalent) is quite apparent. Given a constant speech power and equal transmission characteristics of capsules, excluding differences in sensitivity, a 45° straight line relation between the sending volume and reference equivalent could be expected. In general, the measurements showed a smaller increase. This increase could be accounted for by the fact that, with more efficient capsules, the user speaks less loudly as a result of the increase in sidetone or as a result of some other reaction on receiving speech from the distant subscriber, for in the calls under observation the received speech intensity was, in general, still good.

If, for all male subjects who spoke we derive a mean volume for the sending reference equivalent of 0 dB, we find — 8.3 dB, which corresponds well enough to the normal S.F.E.R.T. volume (about — 9 dB). The mean volume for female subjects who spoke is, for a sending reference equivalent of 0 dB, about 3 dB less than for male subjects. Almost exactly the same volume as the normal S.F.E.R.T. volume was obtained for all the persons who spoke, taken together.

The measurements show quite clearly that, as a result of an increased efficiency of about 10 dB (i.e. as a result of the reduction in the output reference equivalent) for relatively good calls, no inadmissible increase in volume is to be expected which might cause distortion in carrier systems or crosstalk in adjacent lines. With regard to room noise, no reduction in performance should be expected, since the room noise produced by telephone calls in rooms containing several telephone sets is reduced by the fact that the subscriber speaks less loudly when efficient microphone capsules are used.



Relation between sending volume and reference equivalent for different telephone sets

Question 11/XII — Limits of intelligible crosstalk

(Continuation of Question 11/XII, studied in 1964–1972)

PART A

How should the information given in Recommendation P.16 (G.116) be further refined?

Note. — Contributions to the study of this question should be drafted with due regard to the following conditions:

- — room noise: 40 dB, 50 dB (both with Hoth spectrum and weighting A) and negligible;
- — listening subscriber's set with a range of sidetone losses;
- — research to be so conducted as to give information capable of being directly compared with that given in Figure 2 of Recommendation P.16;
- — the standard deviation to be given for the listener threshold;
- — the values characterizing correction factor c also to be indicated with the standard deviation.

The law of the various distributions should also be studied.

It would also be desirable for the contributions to contain information on the test conditions, e.g.:

- — the arrangement and method adopted;
- — the participants (number, age, sex, education, etc.);
- — the instructions given to participants;
- — the telephone sets used (type, frequency response, sending and receiving reference equivalents and sidetone and also the frequency response of the crosstalk path).

PART B

What should be the planning rules governing the use of the noise/crosstalk threshold curves given in Recommendation P.16 (G.116)? The following points should be particularly studied:

a) Which threshold criterion should be used, audibility or intelligibility?
b) What levels of room and line noise should be assumed?
c) What transmission plan arrangements should be used (e.g. distribution of reference equivalents, junction and trunk-circuit losses and crosstalk attenuations)?
d) What are the probability limits appropriate to various classes of connection?
e) What allowances should be made for the joint probability of calls being made simultaneously and the incidence of mutual silences between the two correspondents on the disturbed connection?

ANNEX 1

(to Question 11/XII, Part B)

Notes concerning intelligible crosstalk

(Contribution of the United Kingdom Post Office)

1. Overhearing another conversation tends to undermine a subscriber's confidence in the privacy of his own conversation. This important subjective reaction to overhearing is difficult to quantify for it is clear that the opinions of test-subjects in laboratory experiments are not able to yield a precise or trustworthy measure of the effect. This aspect can be expected to become increasingly important in the future.

Clearly, a public switched telephone network cannot be expected to guarantee perfectly secure speech, but nevertheless we should plan to achieve reasonable freedom from overhearing in fault-free conditions. As a target for study let us aim at 99 % freedom from the risk of intelligible overhearing for each class of call, based on appropriate curves of Figure 2 of Recommendation P.16. It must be remembered that these curves were derived from laboratory experiments in which test-subjects were asked to listen attentively to faint speech and indicate if they could either hear it or hear and understand it. The curves are thus applicable only to an active disturber and a silent listener, and are thus likely to be inappropriate in respect of the ability of persons to understand crosstalk they might chance to encounter when they themselves are engaged in a telephone conversation in which their periods of mutual silence are the only ones in which the crosstalk has a chance of being heard. In view of the lack of knowledge concerning this matter we cannot now take it into account. It is common experience that relatively long pauses are not uncommon; for example the correspondent is looking for a document or has gone in search of someone. However, it might form the subject of an investigation.

2. There are at least two classes of potential disturber which should be distinguished in crosstalk studies:

— the disturber connected to the same distribution point as the disturbed subscriber;

— the disturber not so connected.

a) As regards the first case it would be desirable to ensure very little risk of overhearing between subscribers served by the same distribution point because of the social implications; they are likely to know each other. Furthermore, they are always in the same fixed condition relative to each other.

It follows that although a 99% target might ensure an adequate freedom for the Administration from general complaints of intelligible overhearing, the unfortunate subscribers who constitute the band of listeners in the uncatered-for 1% are permanently in situations in which all their calls are, in principle, subject to encountering intelligible overhearing. They will derive scant confort from knowing that the Administration is content that they are only part of a minority of 1%. However, the chance of intelligible overhearing between any pair within the 1% is governed by the probability of their making simultaneous calls (ignoring the probability of mutual silences in the disturbed connection coinciding with activity by the disturber), quite apart from the masking effect of the listener's room-noise.

b) When subscribers are served by different distribution points the social repercussions are less important and the 99% target is quite reasonable. This also applies to the risk of intelligible overhearing between subscribers making junction or long-distance calls.

For the purposes of establishing an equipment limit based on such cases, the routing of the disturbing and disturbed local lines may be assumed to be uncorrelated. A consequence of this is that subscribers served by the same distribution point and simultaneously engaged in junction calls will have a greater probability of overhearing than if they were making local calls, although there is now likely to be significant line noise which will reduce the probability of intelligible overhearing.

3. Multiple entries into a telephone connection of intelligible crosstalk signals all derived from one source is so unlikely an event that it should be ignored for the purposes of deriving design limits for equipments, etc. Hence, the crosstalk mechanism of interest is assumed to be the dominant one when deriving limits, and all other sources are deemed to be negligible.

Similarly, we ought not to rely on there being several different sources (which gives rise to unintelligible babble), when deriving design limits for an item of equipment. This is because , when considering the distribution of crosstalk attenuation introduced by equipment, etc., one should really consider the distribution of worst values. Why this should be so arises as follows:

Consider the near end crosstalk paths in a fictitious 10-pair junction cable. For any particular pair, say pair 1, there are 9 other pairs capable of crosstalking into it so there will (in general) be 9 values of near-end crosstalk attenuation associated with pair 1. One of these values will (in general) be the worst (i.e.: lowest) of the 9. During busy periods the junction route is, by design, offered more traffic than it can handle so that, in particular, this worst crosstalk path (for pair No. 1) is always activated. That is, the subscribers using pair 1 are always subject to this worst value of crosstalk. They may not be troubled by it, of course. But the fact that the other eight sources are relatively harmless is immaterial—it is sufficient that there is one source that is potentially harmful. Extending this idea to all ten pairs we see we need the values of the ten worst crosstalk paths and we are not to dilute the distribution with the other 80 better crosstalk values. This is clearly a counsel of perfection in some cases but is an important matter in others, e.g.: PCM terminal equipment in which it is often the case that intelligible crosstalk is introduced essentially only between channels adjacent in the multiframe and not between every possible pair of channels.

4. The hypothetical reference connections used in crosstalk studies are, naturally, similar to those used for studies of other transmission impairments (e.g.: loss and noise). However, crosstalk reference connections must be representative of very much simpler connections than is the case for the other impairments. There is little point in studying the direct crosstalk between a pair of 12-circuit connections of near maximum length and noise, in order to arrive at, for example, a limit for channel equipment crosstalk. This is because the majority use of the channel equipment bought and installed to the specification is in much simpler, quieter, and more numerous connections.

5. Noise on the disturbed connection masks low-level crosstalk signals and due account has been taken of it, but it must not be overlooked that limits devised now must, if possible, take the future into account. It is a wise principle that the successful performance of equipment in one part of the network should not be dependent upon adventitious imperfections of other parts of the network, particularly if such imperfections are likely to be eliminated or reduced in the future, e.g.: by new designs of local exchange or by extensive use of long-distance digital transmission systems.

The effect of room-noise (unlike circuit noise) can be reduced by a determined listener and we should at least study the consequences of deriving limits based on the no-room-noise condition particularly for subscribers connected to the same distribution point. For the other classes of connection a room-noise of +40 dB Hoth (which is fairly quiet) would not be unreasonable to assume.

Question 12/XII — Artificial voices, mouths and ears

(Continuation of Question 12/XII studied in 1968–1972)

a) How should Recommendation P.51 (Sections A and B) concerning the artificial ear provisionally advocated by the C.C.I.T.T. be amplified?

b) What general characteristics should be fixed for artificial voices and mouths?

c) Pending a reply to part b) above, how should Section C of Recommendation P.51 be amplified?

Note 1. — Details on the use of the artificial ear and information for the study of part a) of this Question are given in Annex 1. Contribution COM XII-No. 53 (1964–1968) gives the results of tests on the effect of acoustic leaks.

Note 2. — A considerable amount of documentation concerning artificial mouths and ears is to be found in Annexes 8 to 16 in Volume V of the *Red Book* (pages 241–415), in Annex G in Volume V*bis* of the *Red Book* (pages 119–131) and in Annexes 1 to 5 to former Question 12/XII (*Red Book*, Volume V*bis*, pages 202–244).

Note 3. — In studying this Question, attention should be paid not only to artificial mouths and ears used for measuring subscriber sets but also those used for measuring operators' microtelephone sets (which may have internal headphones). See Question 3/XII.

ANNEX 1

(to Question 12/XII)

Use of the artificial ear recommended by the C.C.I.T.T.

Paragraph B.4—"Method of use"—of Recommendation P.51 (Volume V of the *Green Book*) advocates a method of using the artificial ear whereby the earcap measured is applied tightly to the artificial ear without any acoustic leak.

On the basis of the test results submitted to it, Study Group XII considers that this is a limit condition which is approached under practical conditions when the overall reference equivalent of the call is near to the limit recommended by the C.C.I.T.T. or when room noise is relatively loud; in these limit conditions, the listener tends to place the receiver against his ear as tightly as possible. It is therefore desirable to recommend that this condition of use, which excludes acoustic leak, be adopted as a reference condition.

In other conditions, however, e.g. in assessing the transmission performance of calls which do not reach the lower limits of acceptable quality, it may be advisable to introduce a specific degree of acoustic leak so that the results would correspond to those obtained in subjective tests appropriate to such conditions. Further studies are required in this field.

With regard to paragraph B.3.1.—"Basic design"—of Recommendation P.51, it is proposed provisionally that the measurements be carried out with the commercial telephone receiver resting on the knife edge of the input cavity of the artificial ear. The C.C.I.T.T. Laboratory will make the necessary arrangements so that any type of commercial receiver can be applied to the artificial ear in this way. Every Administration can determine the angle (32° according to Recommendation P.51) which ensures that the above-mentioned condition is met when the receiver it normally uses is placed upon the artificial ear.

To ensure a good fit of the receivers on the artificial ear the sealing may be ensured by use of a small rubber ring, or of plasticine, or any other means which does not alter the volume of the coupling. The same methods may be used when measuring receivers with asymmetric earcaps.

ANNEX 2

(to Question 12/XII)

The state of the study of the question at the end of the 1968–1972 period

Artificial voices

Study Group XII notes that very little data are available on artificial voices; Administrations are therefore requested to undertake research on this point and to send in their results.

Artificial mouths

Study Group XII has continued its research with a view to defining a genuine artificial mouth suitable for the determination of objective ratings. Pending this, however, it considers it necessary to recommend provisionally a sound source (see Recommendation P.51).

The activities of Study Group XII can be summed up as follows:

There exist requirements for better artificial mouths than are at present generally available which are suitable for emitting complex sounds more representative of real speech than sinusoids. The exact characteristics required have not yet been formulated, partly because further information is needed on the characteristics of real mouths. Attention is drawn to the following information and to the suggestions made therein for further work on the subject:

1. Annex 15, Volume V of the *Red Book*, pages 374–399.
2. Leman, H. S., 4th International Acoustics Congress, August 1962, Document No. G. 36.
3. Contribution COM XII-No. 19 (1964–1968) by L. M. Ericsson.
4. Annex 2 to Question 12/XII in Volume V of the *White Book*.

In the period 1968–1972, the available documentation was augmented by numerous contributions from Administrations and the C.C.I.T.T. Laboratory. They were Contributions Nos. 50, 51, 52, 64, 68, 73, 84, 113, 118, 134, 136, 162, and Temporary Document No. 2 of the Laboratory Working Party, July 1972.

Law of decrease in acoustic pressure around the human mouth in a free field.

A substantial quantity of information already exists on this point (Annex 15, Volume V of the *Red Book*, and Contribution COM XII-No. 84 "The acoustical properties of human mouths" (period 1968–1972) by the United Kingdom Post Office).

Acoustic impedance of the human mouth

It would be desirable to have the results of measurements of this characteristic of the human mouth. Contribution COM XII-No. 162 (period 1968–1972) from I.T.T. shows that the human mouth is a high impedance source at a distance of more than 5 mm from the lips.

Obstacle effect

It was noted that the data at present available are insufficient and Administrations are strongly urged to furnish new contributions on this point. The study of the obstacle effect created by a disc 6.3 cm in diameter placed perpendicularly on the axis of the mouth is an initial necessary approach; but it is recognized that it would be desirable also to have results for the obstacle effect of different positions (inclined, at points around the axis, etc.) which would better simulate the position of a telephone handset during an actual telephone call.

The obstacle effect of the human mouth should be studied in detail (frequency analysis).

Importance of puffs

Although some Administrations consider that the puffs of air have an important influence on the sensitivity of carbon microphones it can hardly be asserted that an artificial mouth should also simulate the puffs of air produced by the human mouth so long as the properties of the average human mouth are not better known. Some Administrations have simulated the effect of these puffs on the functioning of carbon microphones by means of noise impulses.

Admissible tolerances

Some Administrations consider that it would be useless to specify strict tolerances for artificial mouths to be used for determining the characteristics of carbon microphones.

It was agreed, however, that, in order to determine the sensitivity/frequency characteristics of stable microphones and for the calculation of ratings based on loudness from these characteristics, narrow tolerances are desirable.

The magnitude adopted was ± 0.5 dB for deviations from the mean characteristics: an artificial mouth should follow a decay law of the pressure in a free field (mean value in the wanted frequency range, from 200 to 5000 Hz) which does not differ by more than ± 0.5 dB from the overall distribution for human speech; similarly, the average obstacle effect for all frequencies should not differ by more than ± 0.5 dB from the obstacle effect of the human mouth measured for speech (in the wanted frequency range, from 200 to 5000 Hz).

With regard to discrete frequencies, wider tolerances of approximately ± 1 dB are acceptable.

Question 13/XII — Non-linear distortion of telephone apparatus

(Continuation of Question 13/XII, studied in 1968/1972)

(Documentary question)

Collection of information:

a) on the effects which the non-linear distortion of a subscriber's telephone apparatus has on the quality of telephone transmission;

b) on methods of measuring the non-linear distortion of a subscriber's telephone apparatus; and

c) on the effects of carbon microphone noise in a subscriber's telephone apparatus on the quality of telephone transmission.

Note. — The documentation collected up to the present time is contained in Annex 26 (Part II, *Red Book*, Volume V). The following Annex contains a more recent contribution to the study of this question. For additional information, Administrations may refer to Contribution COM XII-No. 38 (period 1957–1960) which is a contribution by the Japanese Administration concerning the factors liable to affect the natural sound of the voice in a transmission system.

ANNEX

(to Question 13/XII)

Reply of Study Group XII at the end of the 1968-1972 study period

1. *Definition of non-linear distortion*

According to the I.T.U. "List of Definitions of Essential Telecommunications Terms" the term "Non-linearity distortion" is defined as: "Distortion which occurs due to the transmission properties of a system being dependent upon the instantaneous magnitude of the transmitted signal" (06.42). It is usually measured by means of a sinusoidal signal and expressed as the "Coefficient of harmonic distortion" (06.46), defined as the ratio between the r.m.s. value of all the harmonics and the r.m.s. of the total wave.

A difficulty in using sine voltages for the measurement of distortion in speech transmission systems comes from the fact that the crest factor of a sinusoidal signal is only $\sqrt{2}$, whereas for a speech signal the crest factor may be as high as 7. In a system where the distortion coefficient depends on the amplitude of the input signal the measurement by a sinusoidal voltage of a given r.m.s. value will not supply enough information about the performance of the system when a speech voltage of the same r.m.s. is applied. A way around this difficulty is to state the distortion of a system for a sinusoidal voltage having the same peak value as the speech voltage to be applied. This is suitable for laboratory tests and theoretical considerations and has often been used in literature. However, in practical applications the assessment of the peak value of a speech signal is a difficult problem. Because of the simpler measurement it may be a permissible compromise to compare the peak value of the test signal to the long-term r.m.s. value of the speech signal.

2. *Earlier investigations*

A direct evaluation by opinion tests of the degradation caused by non-linear distortion has been contributed to Study Group XII by the Italian Administration (see Annex to Question 13/XII, *White Book*, Volume V). The contribution gives an idea of the combined effect of distortion and attenuation. However, like in most earlier investigations, the effect of second and third order harmonics has not been separated. Instead, the distortion products are combined in an arbitrary ratio which may not be typical for a telephone apparatus.

The influence of frequency-dependent distortion has been investigated by Braunmühl and Weber (Akust. Zeitschrift 1937). They found that much higher distortion can be permitted at low frequencies than at medium and high frequencies. Another interesting finding was that for frequency-independent distortion in the range 50–10000 Hz a slight degradation for speech transmission was audible for 12% second order or 8% third order distortion.

Massa (Proc. IRE 1933) investigated the influence of bandwidth on the smallest perceptible distortion for speech transmission. He found that in a band 80–8000 Hz, 5% second or 5% third order distortion was perceptible in comparison with undistorted transmission. In the band 80–5000 Hz the threshold values were 12% and <10% respectively. However, Massa could not isolate one order of harmonics, so "second order distortion" here means "dominated by second harmonics".

An actual weighting of harmonics of different order according to their subjective effect was discussed by D. Shorter (Electronic Eng., 1950). He cites an old American recommendation (Radio Manufacturers' Association, 1937), where the nth harmonic is multiplied by n/2 (to keep the figure for the 2nd harmonic unchanged). Shorter himself suggests a more drastic weighting, in which the amplitude of the nth harmonic is multiplied by $n^2/4$. However, there were very few experimental data available to validate this proposal.

3. *Recent investigations*

H. Sjögren has studied the effect of second and third order distortion on speech intelligibility [1]. He found that third order distortion reduced the articulation score (typically from 100% to 80% at preferred listening level), while second order distortion at low listening level gave an appreciable increase in articulation. The phoneme confusion occurring without distortion between fricatives, particularly f, and clusiles vanishes when second order

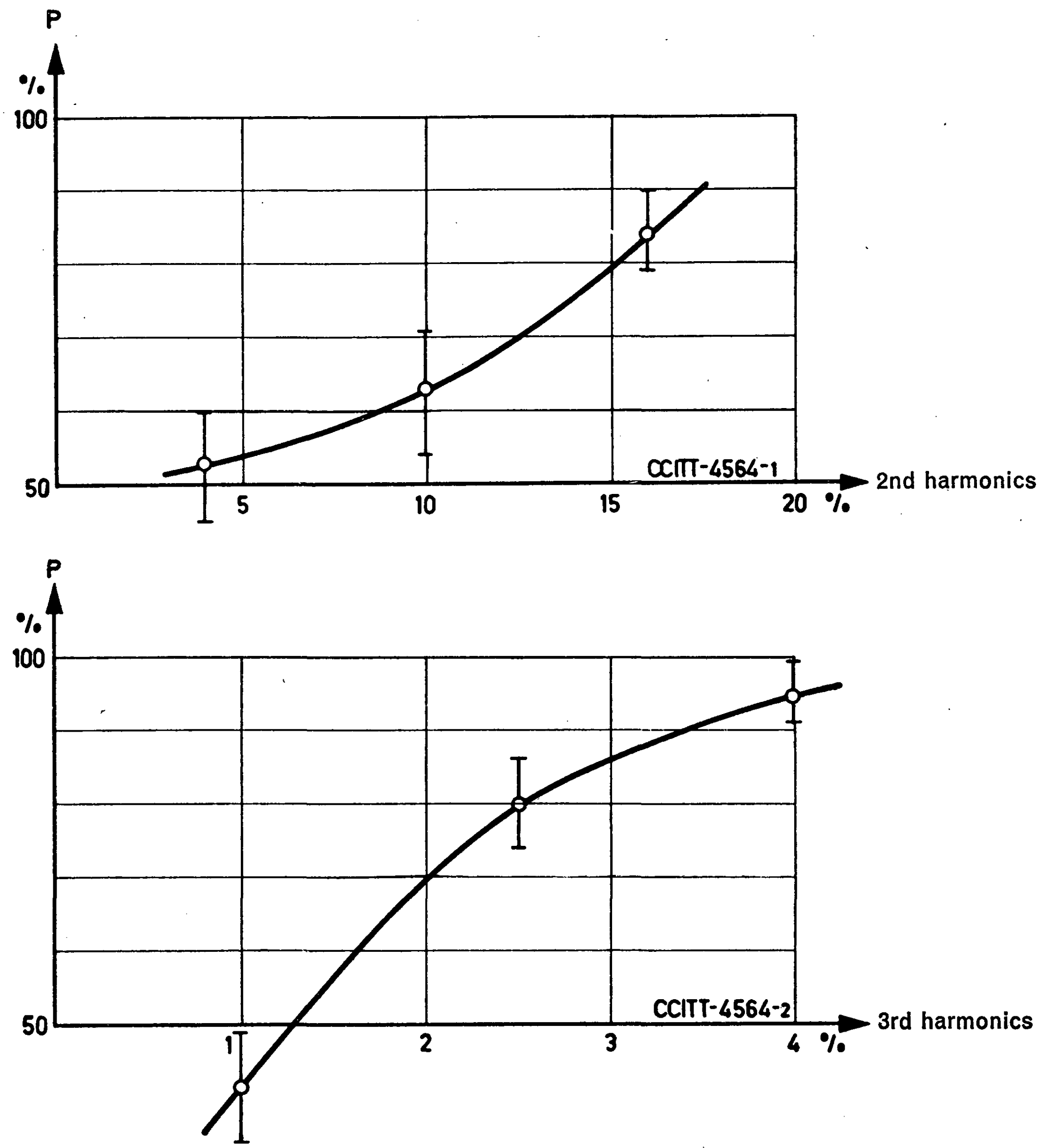


FIGURE 1. — **Percentage of correct responses for paired comparisons of distorted and undistorted speech. 95% confidence limits indicated**

distortion is introduced. Such distortion causes intermodulation products at low frequencies, which may give additional information to the listener.

Another important investigation is described in a contribution to I.E.C. 29 B/WG 9 from Czechoslovakia (February 1971; summary presented by J. Merhaut at the 5th I.C.A. 1965). [2] It covers the psycho-acoustic weight of non-linear distortion, though for music only. The degradation was rated by comparison to a variable bandwidth limitation. The effect of pure second and third order distortion was examined separately, and their relative weight could be determined by means of bandwidth as the comparison variable. It appears that 6% third order distortion is about equally severe as 14% second order. Threshold values are not directly obtained by the method, but third order distortion below 4% and second order below 12% does not seem to produce noticeable degradation.

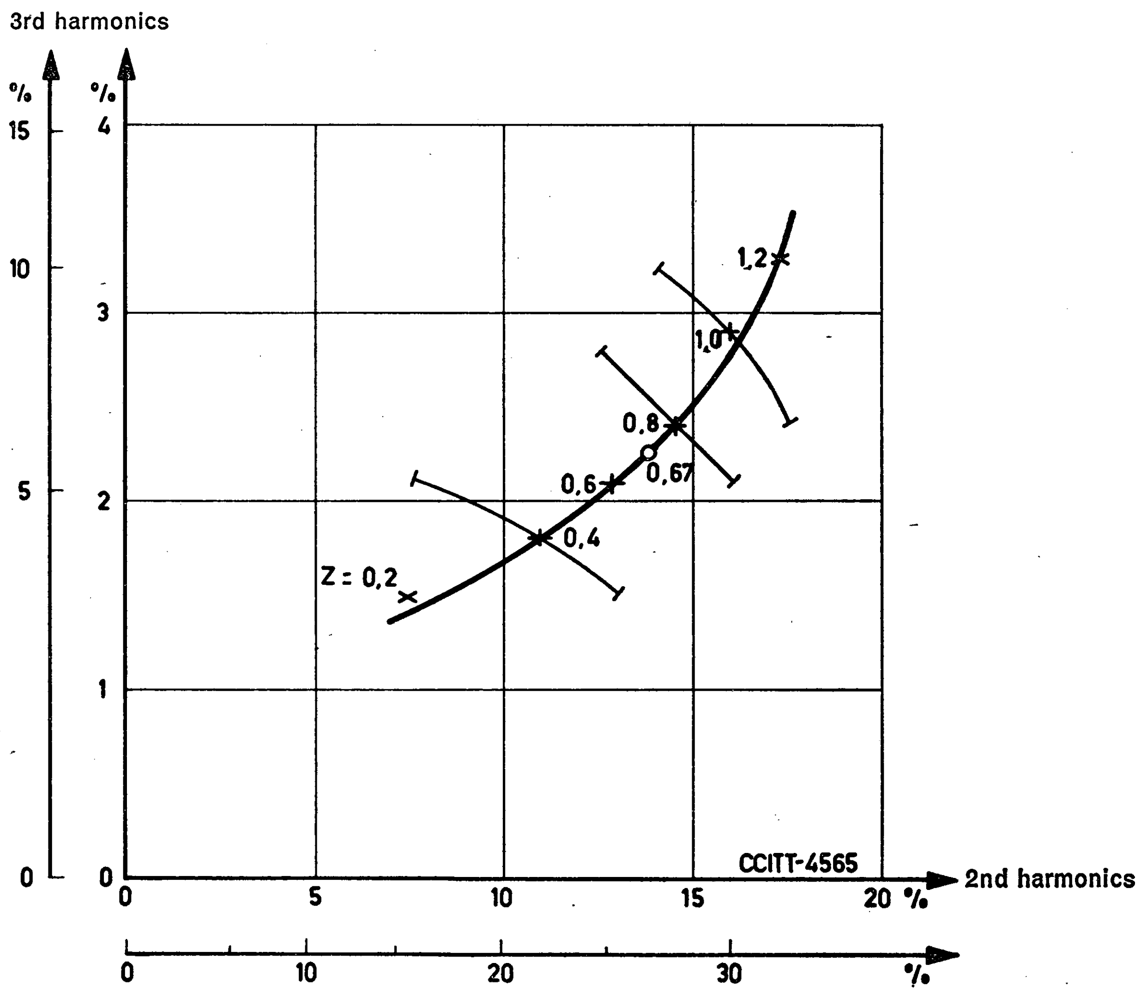
4. *Tests done by the Swedish Telecommunications Administration*

Earlier investigators of the effect of non-linear distortion on transmission quality have had difficulties in producing second and third order harmonics separately and in comparing their relative effect. In this investigation, distortion was introduced by piecewise linear networks, which made it possible to keep unwanted distortion products at a very low level. The amplitude characteristic of the network for second harmonic distortion was $V = v - bv^2$, and for third harmonic distortion $V = v - cv^3$.

In the test, random pairs of distorted and undistorted recordings of sentences were presented to listeners, who were asked to decide which of the samples in each pair was the distorted one. The speech material contained two male and two female voices and was band-limited to the telephone frequency range before presentation over a telephone receiver at about optimum level.

Three values of the content of second or third order harmonics were used after some preliminary tests. The distortion was defined as the distortion coefficient of a sine tone having the same peak value as the r.m.s. value of the speech signal. According to this definition, the distortion values presented corresponded to 4, 10 and 16% second harmonics and to 1, 2.5 and 4% third harmonics.

The test results are shown in Figure 1 as the proportion of correct answers as a function of the distortion coefficient. A relative frequency of 75% is normally used as a criterion for the detection threshold. By interpolation, threshold values for detectable distortion of 14% for second harmonics and 2.2% for third harmonics are obtained. If the distortion instead is defined for a sine tone having the same r.m.s. value as the r.m.s. of the speech signals, the thresholds will be 20% and 5.4% respectively.



The inner scales refer to a sinusoidal voltage whose peak value equals the r.m.s. value of the speech signal.

The outer scales refer to a sinusoidal voltage whose peak value equals the peak value of the speech signal

FIGURE 2. — **Relation between 2nd and 3rd-order harmonic distortion at same sensation. Threshold at z = 0.67. 95% confidence limits indicated**

According to Thurstone's "Law of Comparative Judgment" it may be assumed that the subjective sensation produced by the distortion in a pair comparison test can be scaled by means of a normal distribution transform, using the normal deviate z_i corresponding to the response proportion P_i. It is then possible to compare the effect of the two types of distortion by combining distortion values having the same z-values. This is done in Figure 2. The threshold is defined by $z = 0.67$.

The results show that the listeners are much more sensitive to third than to second harmonic distortion. However, the actual threshold values are probably much dependent on the sound level presented.

5. *New measurement methods*

An ideal method for measuring distortion should yield values that are directly related to the degradation assessed by subjective methods. In particular, different distortion spectra producing the same subjective effect in a speech transmission system should be rated equal by the objective measurement method.

Such a rating may be achieved by a suitable weighting of the components in the distortion spectrum measured for a sine tone. A more straight-forward method is described in a contribution from the United Kingdom Post Office (Annex COM XII—No. 117), using a 450–550 Hz noise band as a measuring signal.[1] This method was originally designed for tests on p.c.m. systems and although suitable for such studies, has not yet been validated by subjective/objective comparisons for non-linearity distortions of other types, e.g. those of ordinary telephone sets. Its value for the present purpose can only be decided upon when the appropriate tests have been carried out.

6. *Conclusions*

The information available at present contributes to the answering of Question 13/XII, point 1, on the effects of non-linear distortion on the quality of telephone transmission. In this respect it makes no difference in which part of the subscriber's telephone apparatus, or in any other part of a telephone connection, the distortion is introduced.

For the time being, it is necessary to analyze non-linear distortion products and to state the content of at least second and third harmonics separately in order to evaluate the degradation of speech quality. For instance, a distortion coefficient of 5%, which is sometimes used as a limit in telephony, may imply an obvious degradation if mainly third harmonics are present but would not be perceptible if the distortion is dominated by second harmonics.

Probably it will soon be possible to agree on a weighting procedure for the influence of harmonics. However, there is still the problem of definition and measurement to solve. As it is, the investigations reported here are not readily compared because of different or lacking definitions of the distortion measurement in relation to the level of the speech signal. Therefore, future work on Question 13/XII should be concentrated on point 2, concerning the methods of measuring the non-linear distortion of a subscriber's telephone apparatus.

7. *Bibliography*

[1] Sjögren, H.: Proc. 2nd Danavox Symposium Odense, Denmark, 1970.
[2] Merhaut, J.: Proc. 5th Internat. Congr. Acoust., Liége, 1965.

Question 14/XII — Effect of attenuation distortion

(New question)

Practical telephone networks can introduce attenuation/frequency distortion into the connection between two local telephone systems. Do the values of attenuation/frequency distortion likely to be encoun-

[1] This method is described in Annex 1 to Question 21/XII *White Book*, Volume V.

tered in practice introduce any significant impairment? If so, how should the impairment associated with a given attenuation/frequency distortion be characterized for planning purposes?

Note. — The hypothetical reference connections to be found in Recommendation G.103 may be useful in the study of this question.

Question 15/XII — Measurement of ratings based on loudness

(Continuation of Question 15/XII, studied in 1968/1972)

Considering,

1. that rather elaborate facilities, including the use of a highly-trained team of operators, are required to determine reference equivalents according to Recommendation P. 72;

2. that many Administrations use other methods (subjective tests, calculations and objective measurements) to determine loudness ratings and that it is very difficult to establish a relationship between the different results;

3. that Study Group XII has been studying the problems and difficulties of determining reference equivalents and loudness ratings for a very long time and that various suggestions have been put forward to improve methods;

4. that it is desirable that the method ultimately chosen as the result of the study of this question should yield measurements as precise as possible and readily reproduced in different laboratories. To this end, attention should be given to defining practical procedures that will ensure the greatest degree of precision. In the case of subjective measurements, for example, the best procedures for ensuring the stability of the testing crew should be studied. Similarly, in any method, the stability of commercial telephone sets, especially those with carbon microphones, will affect the precision obtained and attention should be given to the method of preconditioning the handsets before use (see Recommendation P. 75) and to the fact that a sufficiently large number of sets should be tested so as to ensure reasonably narrow confidence limits. For the same reason, attention is drawn to the importance of using appropriate methods for planning the experiments and for analyzing the results obtained;

5. that the Annex below assembles a comprehensive set of proposals embodying the results of previous studies:

a) Can these proposals and their consequences be accepted with a view to formulating a new recommendation to replace Recommendation P. 72 at a later stage?

b) If not, what other proposals can be made taking the following aspects into account?

1. it is desirable to check whether the methods suggested are accurate enough when applied;
2. the ultimate aim is to evolve an objective method which would give the best possible agreement with a certain number of results of subjective tests.

Note 1. — The Annex first defines certain basic principles involving the use of an Intermediate Reference System; it then indicates how these principles could be implemented in alternative forms as follows:

a) subjective tests;
b) based on defined methods of measuring sensitivity/frequency characteristics;
c) measurements involving use of special measuring instruments.

Note 2. — The study should be undertaken as follows:

1. Measurement of as many different local telephone systems as possible, according to the proposals in the Annex; these measurements should be conducted in both the C.C.I.T.T. Laboratory and other laboratories;
2. Detailed examination of the proposals and suggested improvements;

3. Proposal of methods which would allow for the modification of existing recommendations (e.g. P.11) to take account of new methods of determining loudness ratings in place of the reference equivalent method and the existence of national networks which meet the terms of the present recommendations.

Note 3. — Some of the work undertaken in connection with Questions 1/XII (on operators' sets), 8/XII and 12/XII also affects Question 15/XII.

Note 4. — Sidetone loudness ratings will continue to be studied under this question. Other aspects of sidetone are covered by Question 9/XII.

ANNEX

(to Question 15/XII)

Contribution proposed for study by Mr. D. L. Richards, special rapporteur; not approved by Study Group XII

PREFACE

This Annex consists of five parts as follows:

PART 1. — *Fundamental principles* — Description of the principles common to all loudness rating methods and specific proposals for a new, unified set of definitions.

PART 2. — *Subjective testing methods* — Description of a new subjective loudness rating method which would be consistent with the proposals in Part 1.

PART 3. — *Determination of sensitivity/frequency characteristics* — Description of the requirements for measuring sensitivity so that the results may be applied to the calculation of loudness ratings.

PART 4. — *Calculation of loudness ratings* — Description of calculation methods based on the information and proposals in Parts 1 and 3.

PART 5. — *Objective instrumentation* — Description of the principles underlying existing types of objective loudness rating measurement equipment and indication of changes needed to make them comply with the new proposals in Part 1.

Note. — For ease of cross reference each section, table, figure and equation number is prefixed by the number of the Part in which it appears, e.g.: Table 1.2 will be found in Part 1, etc.

The purpose of the proposals is to change the methods at present used for determining reference equivalents so that the following advantages would be obtained:

a) a variety of means could be employed, such as subjective testing, calculation and objective instrumental measurement without obtaining a variety of incompatible results;

b) to define the loudness ratings according to more useful principles than at present so that the results give a better indication of the relative merit of different local telephone circuits;

c) the principles would be suitable for extending the basis of rating in terms of loudness to include additional criteria, such as opinion scores and percentage of customers experiencing difficulty.

The problems encountered in using the present Recommendations P.42 and P.72 have been discussed in great detail by Study Group XII and its predecessors, as can be found in the documentation associated with study of the following questions since 1952:

Questions 7 and 16, C.C.I.F. Study Group IV, 1952/54;
Questions 7 and 14, C.C.I.F. Study Group IV, 1955/56;
Questions 7 and 14, C.C.I.T.T. Study Group XII, 1957/60;
Questions 7 and 14, C.C.I.T.T. Study Group XII, 1961/64;
Question 15 (new wording), C.C.I.T.T. Study Group XII, 1965/68.

A list of contributions to the study of the present question made available during the present study period (1968–1972) is given in Appendix 1 to this Preface.

Attention is also drawn to contributions made during previous study periods and listed as Annexes 1–6 to Question 15/XII in *White Book*, Volume V.

The general problem of loudness rating has been studied extensively also outside the C.C.I.T.T. and a selection of the published literature on this matter is given in Appendix 2 to this Preface; items 12 and 13 of that list contain reviews of the matter.

The proposals contained in Parts 1 to 5 are consistent so that advantage a) can be obtained. They also embody changes from present (reference equivalent) practice which have been advocated during study group meetings and in the associated documents and provide, it is believed, a firm base for extending the use of calculation and theoretical methods for assessing customer satisfaction.

Although quite definite and specific proposals are made, they are intended to form the basis for further discussion; if necessary, changes can be made without departing from the unity of the proposals as a whole. Alternative proposals are possible that may also give the desired advantages; it is hoped that the most acceptable outcome will be reached by discussion based on an examination of the present proposals offered by the rapporteur.

Most of the proposals made in this document have already appeared in the contributions listed in Appendix 1 to this Preface; many have also been examined by actual tests in the C.C.I.T.T. Laboratory.

Appendix 1 to the Preface

List of contributions to Question 15/XII made available during the study period 1968–1972

1. *Contribution COM XII — No. 1 — C.C.I.T.T. Laboratory (Technical Report No. 368):* "Report on results of subjective measurements (reference equivalent) and objective measurements (with Siemens-type O.D.B.M.) carried out by the C.C.I.T.T. Laboratory on various stable telephone systems", February 1969.
2. *Temporary Document No. 1* (Laboratory Working Party, Geneva, 21–25 April 1969): "Relationship between a reference equivalent (sending and receiving) and the corresponding objective rating measured with Siemens O.D.B.M. (and Tesla TMS2) equipment", April 1969. See Item 3.
3. *Temporary Document No. 2* (Laboratory Working Party, Geneva, 21–25 April 1969): "Relationship between a sidetone reference equivalent for speech (by comparison with N.O.S.F.E.R.) and the corresponding objective rating measured with the O.B.D.M. (Siemens)", April 1969. This document, together with Item 2, form C.C.I.T.T. Laboratory Technical Report No. 371; this report has not been reproduced as a separate document.
4. *Contribution COM XII — No. 2 — C.C.I.T.T. Laboratory:* "Complement and errata to Contribution No. 1", May 1969.
5. *Contribution COM XII — No. 3 — Laboratory Working Party:* "Report of the meeting at Geneva (21–25 April 1969)", June 1969.
6. *Contribution COM XII — No. 5 — Compania Telefonica Nacional de Espana:* "Reference equivalents of telephone sets. Deriving subjective values (N.O.S.F.E.R.) from objective statistical measurements (O.R.E.M.-A)", September 1969.
7. *Contribution COM XII — No. 10 — United Kingdom Post Office:* "Measurement of loudness ratings", November 1969.
8. *Contribution COM XII — No. 18 — C.C.I.T.T. Laboratory (Technical Report No. 381):* "Report on the results of tests carried out in connection with study of Question 15/XII", November 1969.
9. *Contribution COM XII — No. 23 (Part) — International Telephone and Telegraph Company:* "Measurement of ratings based on loudness", December 1969.
10. *Contribution COM XII — No. 33 — Philips' Telecommunicatie Industrie:* "The adjustment of 'The normal speech power for voice-ear measurements' by means of the vu-meter", December 1969.
11. *Contribution COM XII — No. 35 — Compania Telefonica Nacional de Espana:* "Corrigendum to Contribution COM XII — No. 5", December 1969.

12. *Contribution COM XII — No. 36 — C.C.I.T.T. Laboratory (Technical Report No. 390):* "Evaluation of objective ratings relative to the I.R.S. — Complement to Contribution COM XII — No. 18", December 1969.
13. *Contribution COM XII — No. 50 — Study Group XII:* "Report on the meeting in Melbourne (16–26 February 1970)", pp. 39–43 (Preliminary reply to Question 15/XII), May 1970.
14. *Contribution COM XII — No. 64 — Laboratory Working Party:* "Report on the meeting held in Geneva (17–19 November 1970)", January 1971.
15. *Contribution COM XII — No. 76 — C.C.I.T.T. Laboratory (Technical Report No. 414):* "Report on the results of tests carried out in connection with Question 15/XII", May 1971.
16. *Contribution COM XII — No. 86 — U.S.S.R. Telecommunication Administration:* "Loudness weighting factor of a standard Russian sentence for reference equivalent measurement", June 1971.
17. *Contribution COM XII — No. 88 — C.C.I.T.T. Laboratory (Technical Report No. 415):* "Report on the results of tests carried out with commercial telephone systems in connection with the study of Question 15/XII", July 1971.
18. *Contribution COM XII — No. 98 — United Kingdom Post Office:* "Calculation of loudness ratings", September 1971.
19. *Contribution COM XII — No. 120 — International Telephone and Telegraph Company:* "Subjective loudness loss of high-pass and low-pass filtered speech", October 1971.
20. *Contribution COM XII — No. 121 — C.C.I.T.T. Laboratory:* "Differences between objective and subjective ratings", October 1971.
21. *Contribution COM XII — No. 122 — Nippon Telegraph and Telephone Public Corporation:* "Estimated power index m using the various experimental and theoretical results", October 1971.
22. *Contribution COM XII — No. 123 — C.C.I.T.T. Laboratory (Technical Report No. 420):* "Relation between the reference equivalent and the O.B.D.M. (or O.R.E.M.) objective rating for a number of high-pass and low-pass filters", October 1971.
23. *Contribution COM XII — No. 124 — C.C.I.T.T. Laboratory (Technical Report No. 421):* "Comparison of the reference equivalents and various objective ratings of telephone systems (and the I.R.S.) measured in connection with Question 15/XII", October 1971.
24. *Contribution COM XII — No. 125 — United Kingdom Post Office:* "Method of calibration of A.R.A.E.N. Receiver No. 4026A using the I.E.C. artificial ear", October 1971.
25. *Contribution COM XII — No. 129 — Laboratory Working Party:* "Report on the meeting held in Geneva (20–24 September 1971)", November 1971.
26. *Contribution COM XII — No. 131 — Special Rapporteur (Mr. D. L. Richards) on Question 15/XII:* "1st Report by Special Rapporteur on Question 15/XII", October 1971 (Reproduction of Temporary Document No. 12 of Melbourne, February 1970, with amendments by Study Group XII).
27. *Contribution COM XII — No. 132 — Special Rapporteur (Mr. D. L. Richards) on Question 15/XII:* "Supplement to Report presented in February 1970 (Melbourne)", October 1971.
28. *Contribution COM XII — No. 133 — International Telephone and Telegraph Company:* "Contribution to the study of Question 15/XII — Measurement of ratings based on loudness", October 1971.
29. *Contribution COM XII — No. 149 —* "Report on the results of tests carried out with commercial telephone systems in connection with the study of Question 15/XII (Technical Report No. 427 of the C.C.I.T.T. Laboratory)", June 1972.
30. *Contribution COM XII — No. 150 —* "Report on the results of tests carried out with commercial telephone systems in connection with the study of Question 15/XII (Technical Report No. 428 of the C.C.I.T.T. Laboratory)", June 1972.
31. *Contribution COM XII — No. 162 — International Telephone and Telegraph Corporation:* "Contribution to the study of Questions 12 and 15 of C.C.I.T.T. Study Group XII", September 1972.
32. *Contribution COM XII — No. 164 — Laboratory Working Party:* "Report of the meeting in Geneva, July 1972", September 1972.
33. *Contribution COM XII — No. 167 — C.C.I.T.T. Laboratory:* "Report on the results of measurements carried out as part of the programme of work of the C.C.I.T.T. Laboratory for the period 1 August–1 November 1972 (Technical Report No. 432 of the C.C.I.T.T. Laboratory)", September 1972.
34. *Temporary Document No. 5 — C.C.I.T.T. Laboratory:* "Presentation of the results contained in Contributions COM XII — Nos. 88, 149, 150 and 167", November 1972. (The missing values of calculated ratings will be supplied by the United Kingdom Post Office.)

INDEX TO THE ABOVE-LISTED CONTRIBUTIONS

The 34 contributions to the study of Question 15/XII made available during the study period 1968–1972 have been carefully studied by the special rapporteur and the information and proposals made therein have as far as possible been incorporated in the five parts of this Annex. To assist reference and further study the following table has been prepared to show their relevance to these five parts.

Contribution Item No. as listed	Parts to which each contribution is relevant				
	1	2	3	4	5
1		X			X
2		X			X
3		X			X
4		X			X
5		X			X
6					X
7	X	X	X	X	
8		X	X		X
9		X			X
10		X			
11					X
12		X	X		X
13			X		
14		X	X		X
15		X	X		X
16				X	
17		X	X		X
18	X	X	X	X	
19				X	
20		X			X
21				X	
22				X	
23		X			X
24		X			
25		X	X		X
26	X	X	X	X	X
27	X	X	X	X	X
28					X
29–34	X	X	X	X	X

APPENDIX 2 TO THE PREFACE

References to Published Material

[1] Braun, K.: "A new reference equivalent meter with objective excitation and indicating device", *Telegr. u. Fernspr. Techn.*, 1940, *29*, pp. 223–227.

[2] Braun, K. and Koshel, H.: "A direct-reading reference equivalent meter and its effect on the improvement of telephone communications", *Fernmeldetechn. Z.*, 1952, *10*, pp. 447–455.

[3] French P.T.T. Administration: "Apparatus used by the French Administration for the objective measurement of reference equivalents", *C.C.I.T.T. Red Book*, 1962, Volume V, Annex 27, pp. 469–479.

[4] Telephone Administration of the Federal Republic of Germany: "Apparatus used by the Telephone Administration of the Federal Republic of Germany for the objective measurement of reference equivalents", *C.C.I.T.T. Red Book*, 1962, *5*, Annex 28, pp. 479–487.

[5] Swiss P.T.T. Administration: "Method and apparatus used by the Swiss Administration for the objective measurement of reference equivalents", *C.C.I.T.T. Red Book*, 1962, *5*, Annex 29, pp. 487–496.

[6] Czechoslovak Telephone Administration: "Portable telephonometer used by the Czechoslovak Administration for the objective measurement of reference equivalents", *C.C.I.T.T. Red Book*, 1965, *5bis*, Annex H, pp. 132–134.

[7] Turko, V.: "Equipments for objective measurement of the reference equivalent and their comparison with the N.O.S.F.E.R. system", *Tesla Electronics*, 1968, pp. 15–18.

[8] C.C.I.T.T. Recommendation P.45: "Measurement of the A.E.N. value of a commercial telephone system (sending and receiving) by comparison with the S.R.A.E.N.", *C.C.I.T.T. White Book*, 1968, *5*, pp. 1–5.

[9] C.C.I.T.T. Recommendation P.72: "Measurement of reference equivalents and relative equivalents", *C.C.I.T.T. White Book*, 1968, *5*, pp. 1–17.

[10] Cragg, W. D.: "Loudness rating of telephone subscribers' sets by subjective and objective methods", *Elec. Comm.*, 1968, *43*, pp. 39–43 and 228–232.

[11] Barnes, G. J.: "Loudness rating of telephone subscribers' sets by subjective and objective methods", *Elec. Comm.*, 1969, *44*, pp. 299–307.

[12] Richards, D. L.: "Loudness ratings of telephone speech paths", *Proc. I.E.E.*, 1971, *118*, pp. 423–436.

[13] Sullivan, J. L.: "A laboratory system for measuring loudness loss of telephone connexions", *BellSyst. Tech. J.*, 1971, *50*, pp. 2663–2739.

[14] Richards, D. L.: "New definitions for loudness ratings", *Proc. I.E.E.*, 1972, *119*, pp. 1429–1441.

Part 1. — **Fundamental principles**

1.1 *Introduction*

Transmission planning for international telephony has until now largely been based on loudness ratings expressed in terms of "reference equivalents".

The "reference equivalent" method is defined in C.C.I.T.T. Recommendations P.42 and P.72 and the fundamental principles are briefly explained in Annex 1.1 of the C.C.I.T.T. Handbook: *Transmission Planning of Switched Telephone Networks*. The present proposals for a new method for determining loudness ratings of local telephone circuits are in accordance with similar fundamental principles to those of the "reference equivalent" method but embody modifications which render the new method much more flexible so that the same values of "loudness ratings" should be obtained whether the determination is by subjective tests, by calculation based on sensitivity/frequency characteristics or by objective instrumentation. The fundamental principles of the new method are described below and these differ from those applicable to "reference equivalents" by the least possible extent to achieve the desirable flexibility.

1.2 *General considerations*

A speech path is, broadly, a transmission path that exists between a talker's mouth and the ear of a listener or, in the case of sidetone, between the mouth and ear of a talker. In typical face to face conversation, the speech is transmitted by means of the air path connecting the mouth and ear. Depending on environmental conditions, transmission may be, a) more or less direct, as in the case of two persons conversing in an open, unobstructed location, such as a golf course; b) largely indirect, as in the case of two persons conversing in a small, hard surfaced, room where a large proportion of the energy reaching the ear may be due to reflections from the walls, ceiling and floor; or c) something between the two extremes of a) and b).

In the case of telephony, the air path is replaced by a system comprising, a) an air path from the mouth to the telephone microphone; b) an air path between the telephone earphone and the ear and, c) a telephone connection consisting of the microphone, earphone and interconnecting circuitry together with a similar system for the reverse direction of transmission. The two situations—face to face and using the telephone—differ appreciably in detail but, for speech transmission purposes, they are alike so far as their function is to provide a means of both-way speech communication.

Telephone engineering is concerned with providing telephone connections which, while not identical to the face to face situation, are comparable in effectiveness for providing a means of exchanging information by speech; such telephone connections should also optimize customer satisfaction within technical and economic constraints.

Various tools are used by transmission engineers in planning, design and assessment of the performance of telephone networks. Loudness ratings, based on the criterion of loudness of speech emitted by the talker and perceived by the listener provides and continues to provide one of the most important of these tools; such ratings provide a very important measure of the transmission loss, from mouth to ear, of a speech path.

The loudness rating (which has the dimensions and sign of "loss") is, in principle, defined by the amount of loss inserted in a reference system to secure equality of perceived loudness to that obtained over the speech path being measured. Practical telephone connections are composed of several parts connected together. To enable the transmission engineer to deal with these parts in different combinations, loudness ratings must be defined in a suitable manner so that "sending" and "receiving" ratings can be used.

1.3 *Definitions and symbols*

Definitions and symbols used in the subsequent description of fundamental principles are listed below.

1.3.1 *Concerning talking*

These definitions and symbols characterize the situation where a subject is talking and they include his physical relationship to the telephone or reference connection.

P_M Point defining the *m*outh *r*eference *p*oint (M.R.P.); P_M is at a defined location relative to the talker's lips.

p_M Sound pressure at P_M.[1]

B'_S Spectrum density (long-term mean pressure)[2] of speech referred to a M.R.P. (p_M) in dB relative to 20 $\mu N/m^2$ in a bandwidth of 1 Hz.

VL Vocal level, i.e. speech sound pressure (long-term R.M.S.) level of talker at the M.R.P.; usually referred to a reference vocal level as datum.

SP Speaking position, i.e. the relative location of the microphone of the telephone or reference system and the lips of the talker.

1.3.2 *Concerning listening*

These definitions and symbols characterize the situation where a subject is listening and they include his physical relationship to the telephone or reference connection:

P_E Point defining the *e*ar *r*eference *p*oint (E.R.P.).

p_E Sound pressure at P_E.

β_0 Hearing threshold for pure tones referred to an ERP in dB relative to 20 $\mu N/m^2$.

K A number, related to Fletcher's critical frequency bands, required to convert hearing threshold for pure tones to that for continuous-spectrum sounds like speech.

β_0–K Hearing threshold for continuous-spectrum sounds referred to an ERP in dB relative to 20 $\mu N/m^2$ in a bandwidth of 1 Hz.

HL Hearing loss, usually referred to "normal" hearing threshold.

LC Listening conditions; the manner in which the earphone and its coupling to the ear is related to the E.R.P.

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[1] The reference level or datum must be specified, e.g. 1 N/m^2, 20 $\mu N/m^2$, etc.

[2] In practice, measurements are made in terms of sound pressure, and that convention is retained for convenience of explanation. It is worth noting that sound pressure relative to 20 $\mu N/m^2$ in a bandwidth of 1 Hz is approximately equal to sound intensity relative to 1 pW/m^2 per Hz.

1.3.3 *Concerning telephone or reference connections*

These definitions and symbols serve to characterize the telephone or reference connections in objective terms:

L_{ME} Air-to-air transmission loss, in dB, from a M.R.P. to an E.R.P.

J_S, J_R Electrical interfaces at the output of a sending local telephone circuit and the input to a receiving local telephone circuit.

LTC Local telephone circuit.

S_{MJ} Sending sensitivity of a local telephone circuit from the M.R.P. (P_M) to the electrical output (J_S).

S_{JE} Receiving sensitivity of a local telephone circuit from the electrical input (J_R) to the E.R.P. (P_E).

x Transmission loss between local telephone circuits, i.e. between J_S and J_R.

S_{RMJ}, S_{RJE}, L_{RME}, etc., are values of S_{MJ}, S_{JE}, L_{ME}, etc., applicable to a reference speech path, e.g. N.O.S.F.E.R.

S_{UMJ}, S_{UJE}, L_{UME}, etc., are values of S_{MJ}, S_{JE}, L_{ME}, etc., applicable to an unknown speech path, e.g. a telephone connection.

x_{UR}, x_{RU} – values of x applicable to combinations of "unknown" sending to reference receiving and reference sending to "unknown" receiving speech paths.

S_M Sensitivity of a telephone microphone referred to a M.R.P. (P_M).

S_E Sensitivity of a telephone receiver referred to an ERP (P_E).

L_S Electrical transmission loss from the terminals of a microphone to the line terminals of a telephone set.

L_R Electrical transmission loss from the line terminals of a telephone set to the terminals of a receiver.

$L(SL+FB)$ Transmission loss of the subscriber's line and feeding bridge.

1.3.4 *Concerning loudness, its relationship to sensation level and loudness ratings*

These definitions and symbols relate to factors concerning loudness and loudness ratings of telephone speech paths:

Z Sensation level, in dB, of the received speech signal at a given frequency; describes the portion of the received speech signal which is above threshold and is, therefore, effective in producing the sensation of loudness.

Z_{RO} Value of Z when $L_{ME} = 0$ dB.

$Q(Z)$ Function of Z related to loudness; transforms sensation level expressed in terms of Z, to loudness numerics.

m A parameter which can be used to define $Q(Z)$; represents the slope of $10 \log_{10} Q(Z)$ as function of Z.

S A monotonic function of frequency such that equal increments of S are of equal importance to loudness, provided the associated values of Z are the same.

S' The derivative of S with respect to frequency; $S' = \frac{dS}{df}$. S' can be considered as a frequency weighting factor.

dS From the foregoing, $dS = S' \, df$.

$\overline{Q(Z)}$ Weighted average of $Q(Z)$ which is related to the total loudness in a received speech signal.

λ Loudness of sound.

OLR, *SLR*, *RLR* Overall, sending and receiving loudness ratings.

1.4 *Loudness*

In considering speech transmission paths, it is necessary to define acoustical terminals of the paths. This can be done in terms of M.R.P. and E.R.P. There are no unique definitions of such reference points, but those shown in Figure 1.1 will serve for purposes of discussion (see P_M and P_E of Sections 1.3.1 and 1.3.2).

Curve 1 in Figure 1.1 b shows the spectrum density B'_S of speech emitted at a certain vocal level and measured at the M.R.P., P_M, in the absence of any obstruction in front of the mouth. The measurement may be thought of as made with the aid of a very small measuring microphone. When the speech reaches the ear of the other participant in a telephone conversation, it will have been subjected to transmission loss and distortion in the telephone speech path and the spectrum density may then be as shown in Curve 2; the E.R.P., P_E, to which Curve 2 is referred can, for explanation, be thought of as located at the opening of the ear canal, but might equally well be the tympanum, i.e. eardrum of the listener's ear. The interval, L_{ME}, between Curves 1 and 2 represents the "mouth-to-ear" transmission loss and is, in general, frequency-dependent.

The received spectrum represented by Curve 2 does not contribute uniformly to loudness, i.e. those portions of the spectrum lower in level than the listener's threshold of hearing contributes very little compared with those well above the threshold. Account is taken of this by defining a quantity termed "sensation level" (symbol Z) which is the interval between the received spectrum, Curve 2, and the threshold of audibility for continuous spectrum sounds (β_0-K) shown in Curve 3. Loudness of the received speech sound thus depends upon Z, which is, in general, frequency-dependent.

Studies have shown that the loudness, λ, can be expressed approximately as a function of Z in the following manner:

$$\lambda = C \int_{f_1}^{f_2} Q(Z)\, S'\, df \tag{1.1}$$

where C is a constant $Q(Z)$ is a "loudness growth function" which transforms Z so that equal increments of the transformed values represent equal increments in loudness, S' is a "frequency weighting function" which weights the transformed values of Z according to their positions along the frequency scale and f_1 and f_2 correspond to the lower and upper frequency limits for the band of interest.

If desired the frequency scale can be transformed to a scale of S, equal increments of which have the same "importance" so far as loudness is concerned.

Thus:

$$S' = \frac{dS}{df} \tag{1.2}$$

$$\text{and so } \lambda = C \int_{S_1}^{S_2} Q(Z) dS \tag{1.3}$$

where S_1 and S_2 are points on the scale of S that correspond respectively to f_1 and f_2.

The basic elements of the loudness rating process are shown in the flow diagram of Figure 1.1c. The flow diagram depicts a "reference" spectrum decreased by the loss of a telephone connection resulting in a received spectrum which together with the threshold of hearing produces Z, the values of which (as a function of frequency) are effective in producing the sensation of loudness.

$$\text{Thus: } Z = B'_S\text{-}L_{ME}\text{-}(\beta_0 - K) \tag{1.4}$$

and Z as a function of frequency is converted to loudness, λ, according to the equations explained above in which Z is transformed to loudness numerics which are then weighted by the frequency weighting function to produce $\overline{Q(Z)}$; a constant applied to $\overline{Q(Z)}$ produces λ, the loudness of the received speech expressed on some suitable scale.

The flow diagram of Figure 1.1c represents only basic elements in the loudness rating process. These elements require further specification in order to render them unique. For example, R'_S depends on the particular speaker and his vocal level, the test phrase used, and the location of the talker's lips with respect to the telephone microphone defined by his individual method of usage and by the somewhat arbitrarily defined M.R.P. Similarly, the received spectrum level depends on the particular listener and his characteristics, e.g. fit between his ear and the telephone earphone when the handset is held in a prescribed manner, whether or not he has a hearing loss, and on the E.R.P.

Furthermore, transmission planning studies require subdivision of the connection loss, L_{ME}, into component parts, e.g. a sending component, a receiving component and an interconnecting component.

The function $Q(Z)$ can, in part, be specified in terms of a parameter m which is the slope of the logarithm of $Q(Z)$ when plotted against Z. m does, however, depend upon the listening level (or Z) in the general case but may be considered constant over a wide and useful range of Z.

Those additional factors considered at present to be of importance are included in the more detailed flow diagram of Figure 1.2b which is an expansion of Figure 1.1c. The influence of these factors can be appreciated from the previous discussion and from review of the definitions given in Section 1.3. Figure 1.2a supplements these definitions.

1.5 *New definitions of loudness ratings*

1.5.1 *General*

In general, loudness ratings are not expressed directly in terms of λ but are expressed in terms of the amount of transmission loss independent of frequency that must be introduced into the "unknown" speech path to secure the same loudness of received speech as that defined by a fixed reference speech path. This implies that some interface exists in the "unknown" speech path at which the transmission loss can be introduced; in practice the "unknown" speech path is composed of a sending local telephone circuit coupled to a receiving local telephone circuit through sufficient transmission loss provided by a 600 ohm attenuator to render the termination of each local telephone circuit constant at 600 ohms (see Figure 1.2a). If the loss inserted in the "unknown" speech path is denoted by x_U and the setting of the reference speech path needed to define the desired constant loudness (λ_R) is x_R, then the loudness rating of the "unknown" is given by $x_R - x_U$. Figure 1.3a shows the method; all features other than those defining the speech paths themselves are common to both paths. The sending and receiving ends of the reference speech path are defined later; the value of loss, x_R, in the "junction" of the reference speech path is chosen so that a comfortable level of speech is produced at the ears of the listeners at the receiving end (the precise value is indicated later). The adjustable transmission loss, x_U, is varied during subjective loudness balancing tests or evaluated by calculation or objective measurement to determine the value that renders $\lambda_U = \lambda_R$; the "overall loudness rating" (O.L.R.) is then given by the difference $x_R - x_U$ dB.

In order to separate a sending loudness rating and a receiving loudness rating, it is necessary to define the characteristics of the sending and receiving ends of the reference speech path separately. The sending and receiving ends of the "intermediate reference system" (I.R.S.) to be used here are specified later in manners that fix their sending and receiving sensitivities S_{RMJ} and S_{RJE}.

Figure 1.3b shows the method used to determine sending loudness ratings (S.L.R.). A composite speech path is made up of the "unknown" sending L.T.C. coupled through a distortionless attenuator, x_{UR}, to the receiving end of the reference system. The complete reference speech path remains unchanged (x_R is fixed) and the "junction" loss x_{UR} is varied to secure loudness balance; then the sending loudness rating (S.L.R.) is given by $x_R - x_{UR}$ dB.

Figure 1.3c shows the corresponding arrangements for determining receiving loudness ratings (R.L.R.). The sending end of the reference speech path is associated with the "unknown" receiving L.T.C. and compared with the complete reference speech path by adjustment of x_{RU}. x_R remains fixed and the value of receiving loudness rating (R.L.R.) is given by $x_R - x_{RU}$ dB when λ_{RU} has been adjusted to equal λ_R.

1.5.2 *Requirements*

The proposals outlined below are intended to offer, in a practically useful manner, the following five desirable features:

1. The criterion of loudness of received speech with arbitrarily fixed talking conditions ought to be retained but should not be interpreted excessively strictly.

2. In view of the large body of information now in use which has been assembled on the basis of reference equivalents, it is important that the numerical values associated with existing items of equipment should not be unduly changed.

3. So far as is possible, the loudness rating of any complete speech path must be equal to the sum of the loudness ratings allocated to the separate component parts.

4. The methods used for assessing loudness ratings should be as realistic as possible so that real and genuine improvements in performance are properly reflected in the results.
5. The methods chosen ought to be simple to understand and be capable of being employed in any reasonably well-equipped telephonometric laboratory without serious disagreements in test results; this implies that the principles must be based on proper theoretical foundations so that the mode of implementation does not affect the numerical results.

Naturally, to a certain degree, these requirements are incompatible with each other and so some compromises are necessary; where appropriate the reasons for the choice are indicated briefly.

1.5.3 *Proposals*

The items needing to be specified are listed in Table 1.1 which indicates also the choices proposed under each heading. Particular attention should be directed to items 4, 5 and 13 of the table because these involve very important

TABLE 1.1

PROPOSALS FOR UNIFIED DEFINITIONS OF LOUDNESS RATINGS

No.	Item requiring to be specified	Specification
1	Fundamental reference system (complete speech path)	N.O.S.F.E.R. or A.R.A.E.N. might be retained. Choice not important for subjective tests. Not relevant for objective determinations once an Intermediate Reference System has been defined (see Item 13 below)
2	Vocal level	Reference vocal level (see Appendix 4-1)
3	Speech material	Chosen to produce a spectrum representative of normal telephone users in service (see Appendix 4-1)
4	Handset talking distance from lips	Defined by a new speaking position (see Appendix 1-1 for details)
5	Direction of speech	
6	Handset mounting for sending	Suitably clamped consistent with 4 and 5
7	Conditioning of carbon microphones	Some suitable rotational procedure.[a] The mounting arrangements must be designed to facilitate this
8	Hearing loss	Normal hearing
9	Handset mounting for receiving	This should correspond to fairly careful holding of the handset with the earphone placed comfortably against the ear
10	Design of experiment	Only important for subjective tests; any suitable that is consistent with 11 and 12
11	Listening level	Listening level constant at loudness equivalent to, say, 25 dB setting of N.O.S.F.E.R.
12	Balancing method	Margin method to satisfy Item 11
13	Method of separating sending and receiving ratings	Using an Intermediate Reference System with sending and receiving parts defined as shown in Table 1.2

[a] See Part 3.

changes from the practice used to determine reference equivalents according to Recommendation P.72. The remainder, although requiring careful consideration, do not raise crucial issues.

N.O.S.F.E.R. is unsuitable for the Intermediate Reference System (Item 13 of Table 1.1) but it could be retained as the Fundamental Reference System (Item 1 of Table 1.1). Alternatively, and perhaps preferably, A.R.A.E.N. could be used as the Fundamental Reference System; the sensitivity of A.R.A.E.N. is practically independent of frequency which is not the case with N.O.S.F.E.R. Whichever of these is used, it will be necessary to determine its setting to correspond with the desired value of λ_R. The characteristics of the I.R.S. must be decided so that Requirement 2 is satisfied before the "reference" setting of the Fundamental Reference System can be determined.

Reference Vocal Level (R.V.L.) is based on the definition given in C.C.I.T.T. Recommendation P.45 for the conduct of articulation rating tests. For greater convenience, it has been redefined in terms of the mean power of speech signals while the talker is active; this form is necessary in any calculations based on such theoretical treatment as is given in Part 4. Appendix 4-1 gives the spectrum density for speech emitted at R.V.L., the information applying to the average for males and females. It will be seen that, at the mouth reference point used here, the level of the total intensity of speech at R.V.L. is +89.3 dB relative to 1 pW/m^2 or a level of r.m.s. sound pressure of —4.7 dB relative to 1 N/m^2. The total intensity of speech at the level used for reference equivalent determinations in accordance with C.C.I.T.T. Recommendation P.72 is 5.6 dB lower than R.V.L. and its spectrum may be taken to differ somewhat from that shown in Appendix 4-1.

The method of holding the handset (Items 6 and 7 of Table 1.1) is particularly important when carbon microphones are involved; the design of the mounting arrangement must be such that the appropriate conditioning movements can be made without removing the handset from its clamp.

During subjective tests the earphone will be acoustically terminated by a real ear with a certain amount of leakage; it is important that the artificial ear should simulate this, perhaps rather poor, degree of coupling or that suitable allowance should be made for it. The magnitude of such allowances cannot be stated at present for all types of earphone and further information is required on the matter (see Section 3.3).

The level at which listening takes place should be fixed, independently of the sensitivity of the item being measured. For subjective determinations, the actual value affects the results when commercial telephone equipment is being balanced against a wide-band reference system; for objective determinations, the assumed listening level determines certain of the parameters in the computation programme or of the corresponding features of the instrumentation which performs the processing of the sensitivity/frequency characteristics.

1.5.4 *Speaking position*

The speaking position, including the distance from lips to microphone opening, specified for determinations of reference equivalent is not very typical of normal, natural use of telephone handsets; in fact, with very short handsets, it is extremely difficult to comply with, the talker having to place his lips against or even within the mouthpiece opening of certain commercial types of telephone handset. A much more representative speaking position was specified by the C.C.I.T.T. for use when determining articulation ratings; this is defined in C.C.I.T.T. Recommendation P.45. The same recommendation contains the definition of Reference Vocal Level referred to above (see Appendix 4-1). The new speaking position is derived from the definition in Recommendation P.45 by adding certain details that serve to locate the orientation of the handset in space so that the correct attitude is assumed by the granule chamber of a carbon microphone and so that the sound from the mouth strikes the mouthpiece at an appropriate angle. A detailed description of the new speaking position is given in Appendix 1-1.

If all telephone handsets were of very similar dimensions so that the mouth-to-microphone distance was constant and the obstruction effect of the mouthpiece was the same, this proposed change in speaking position would be comparatively unimportant because differences between sending ratings of different telephone sets would remain practically the same. In fact there is considerable variation in dimensions and shapes of telephone handsets and the effects on sending ratings of changing the speaking position from the reference equivalent speaking position to the new speaking position range from −2.0 (e.g. for a very long handset) to −3.6 (e.g. for a very short handset). More extreme values can be found when the amplitude distortion of the carbon microphone is severe.

1.5.5 *Intermediate reference system*

The primary function of the Intermediate Reference System (I.R.S.) is to provide a defined means for separating sending and receiving loudness ratings in such a manner that additivity is preserved, i.e. so that:

Overall loudness rating = Sending rating + Receiving rating + Intervening transmission loss.

This requirement will be fulfilled only if the frequency responses of the sending and receiving parts of the I.R.S. correspond in shape with the respective parts of typical real commercial local telephone circuits. Naturally the bandwidth must be clearly defined to lie within the nominal limits of 300–3400 Hz set by channelling equipment.

The I.R.S. must also conform to ordinary telephone sets by using a microphone and earphone in a handset. The shape and size of the handset must also result in reasonably "average" distances between talkers' lips and microphone opening.

TABLE 1.2

SENDING AND RECEIVING SENSITIVITIES[a] OF THE INTERMEDIATE REFERENCE SYSTEMS (I.R.S.)

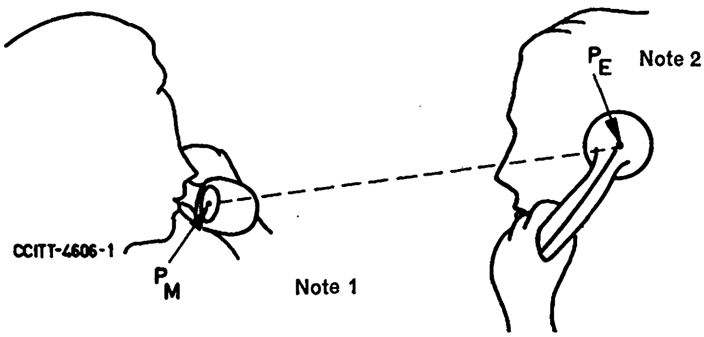
Frequency Hz	Sending S_{RMJ} dB[b]	Receiving	
		IEC/AE S_{RJE} dB[c]	Real ear S_{RJE} dB[d]
(1)	(2)	(3)	(4)
200	−18.8	+12.4	+4.0
300	−14.6	+10.5	+9.2
400	−11.6	+9.9	+10.6
500	−8.9	+9.0	+11.2
600	−6.2	+9.0	+11.5
800	−1.2	+8.5	+11.7
1000	+0.1	+9.4	+11.7
1250	+0.7	+10.5	+11.7
1600	+1.1	+11.6	+11.7
2000	+1.6	+15.3	+11.7
2500	+1.9	+19.1	+11.7
3000	+2.2	+18.0	+11.7
3500	−11.4	+6.7	−0.8
4000	<−28	<−11	<−20

[a] The sensitivities are referred to the mouth reference point and ear reference point defined in Section 1.3. The electrical interfaces are of 600 ohms impedance. The sending sensitivities may be determined either with the United Kingdom Post Office or the Bruel and Kjaer Type 4216 artificial mouth which have shown concordant results when used to measure sensitivities of actual telephone microphones at the new speaking position. The receiving sensitivities shown in Column (3) apply when the C.C.I.T.T./I.E.C. artificial ear is used and the earphone is sealed direct to the artificial ear, without any leak.

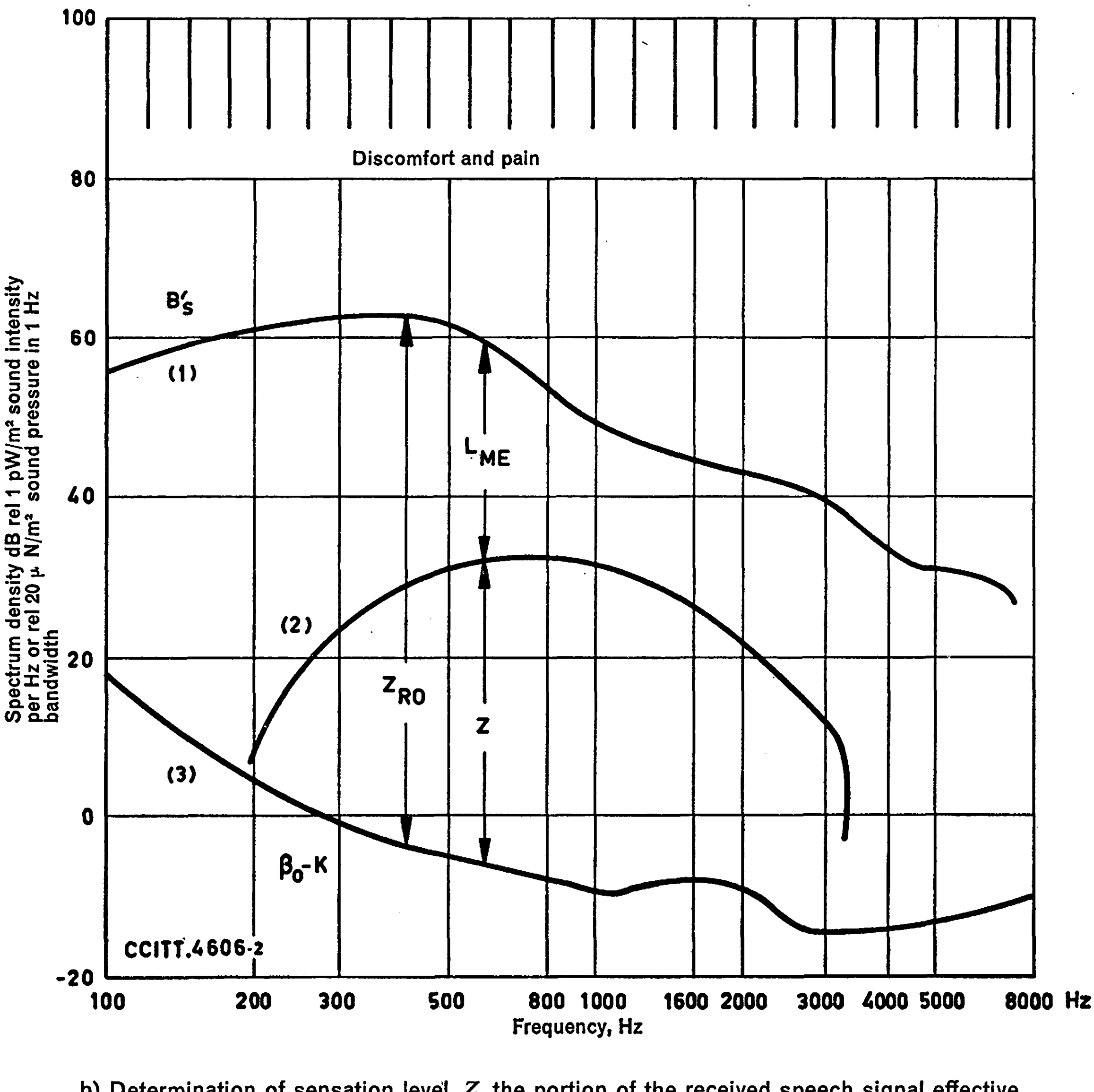
[b] New speaking position: see Appendix 1.1.

[c] The I.R.S. is, in principle, specified in terms of real-ear sound pressures (Column (4) measured on a median subject when he holds the handset with the earphone applied to his ear in a careful manner. The artificial ear sensitivities shown here apply when the I.R.S. is realized using the United Kingdom Post Office Receiver Inset No. 4T mounted in a United Kingdom Post Office Handset No. 3.

[d] See Section 3.5 for the corrections used to convert from artificial ear to real ear sensitivities.



a) Definitions of mouth and ear reference points



b) Determination of sensation level, Z, the portion of the received speech signal effective in producing the sensation of loudness

FIGURE 1.1. — **Definitions concerned with the loudness of speech heard over a transmission path**

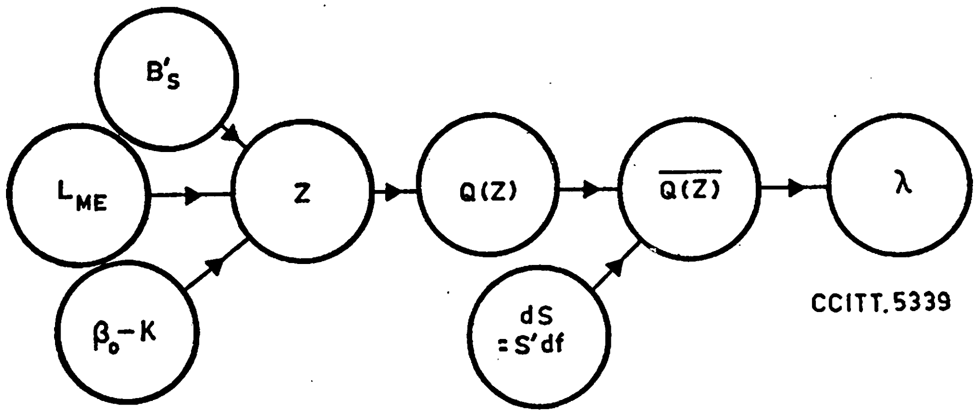


FIGURE 1.1. — *(concluded)*

c) Simplified flow diagram showing how loudness, λ, is related to sensation level, Z

Notes to Figure 1.1

Graph a)

(1) P_M is located at a specified position in front of the lips, e.g. 25 mm

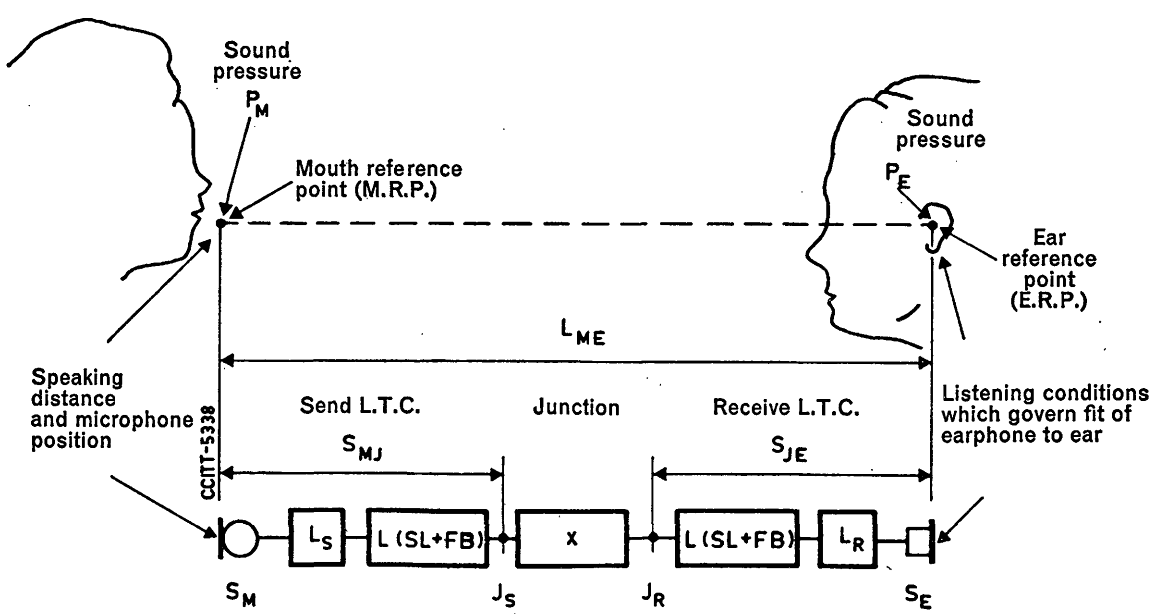
(2) P_E is located at a specified position relative to the listener's ear, e.g. entrance to the ear canal

Graph b)

Curve (1) — Spectrum density of speech at mouth reference point

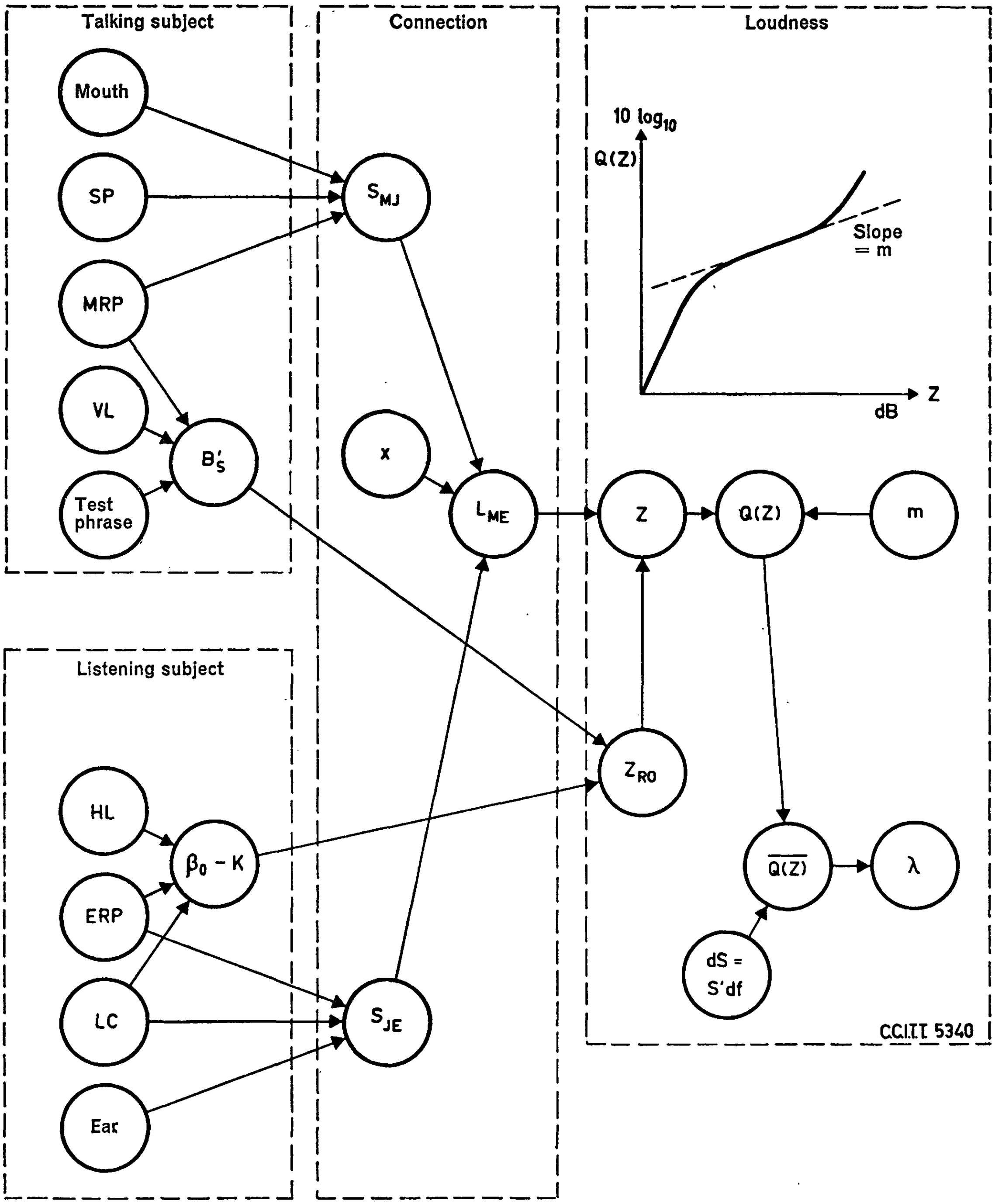
Curve (2) — Spectrum density of speech at ear reference point received over an approximately limiting telephone speech path

Curve (3) — Hearing threshold for continuous spectrum sounds



a) Definitions of transmission losses and sensitivities

FIGURE 1.2. — **Factors affecting loudness of received speech**



b) Flow diagram

FIGURE 1.2. — *(concluded)*

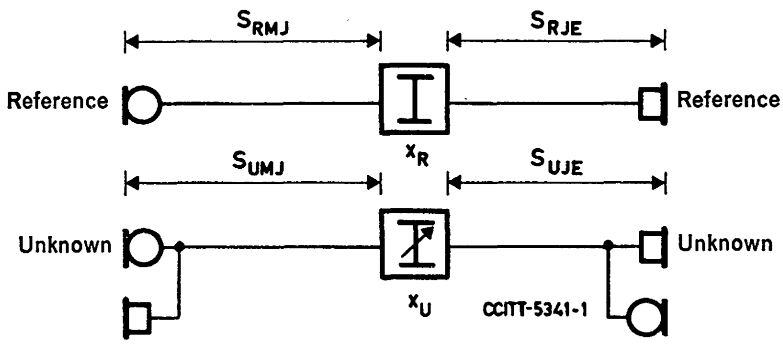
The absolute sensitivities, S_{RMJ} and S_{RJE}, of the sending and receiving parts of the I.R.S. can be chosen so that the following approximate equalities will apply when practical commercial local telephone circuits are rated relative to the I.R.S.:

a) The numerical values of sending loudness rating (S.L.R.) agree very closely with S.R.E.

b) The numerical values of receiving loudness rating (R.L.R.) agree very closely with R.R.E.

c) The numerical values of overall loudness rating (O.L.R.) agree very closely with the sum S.L.R. + R.L.R. and with nominal overall reference equivalent (N.O.R.E. = S.R.E. + R.R.E.). (True O.R.E. differs from S.R.E. + R.R.E. by about 3 dB, when two L.T.C.s are interconnected with distortionless loss.)

a) Overall loudness rating (O.L.R.)



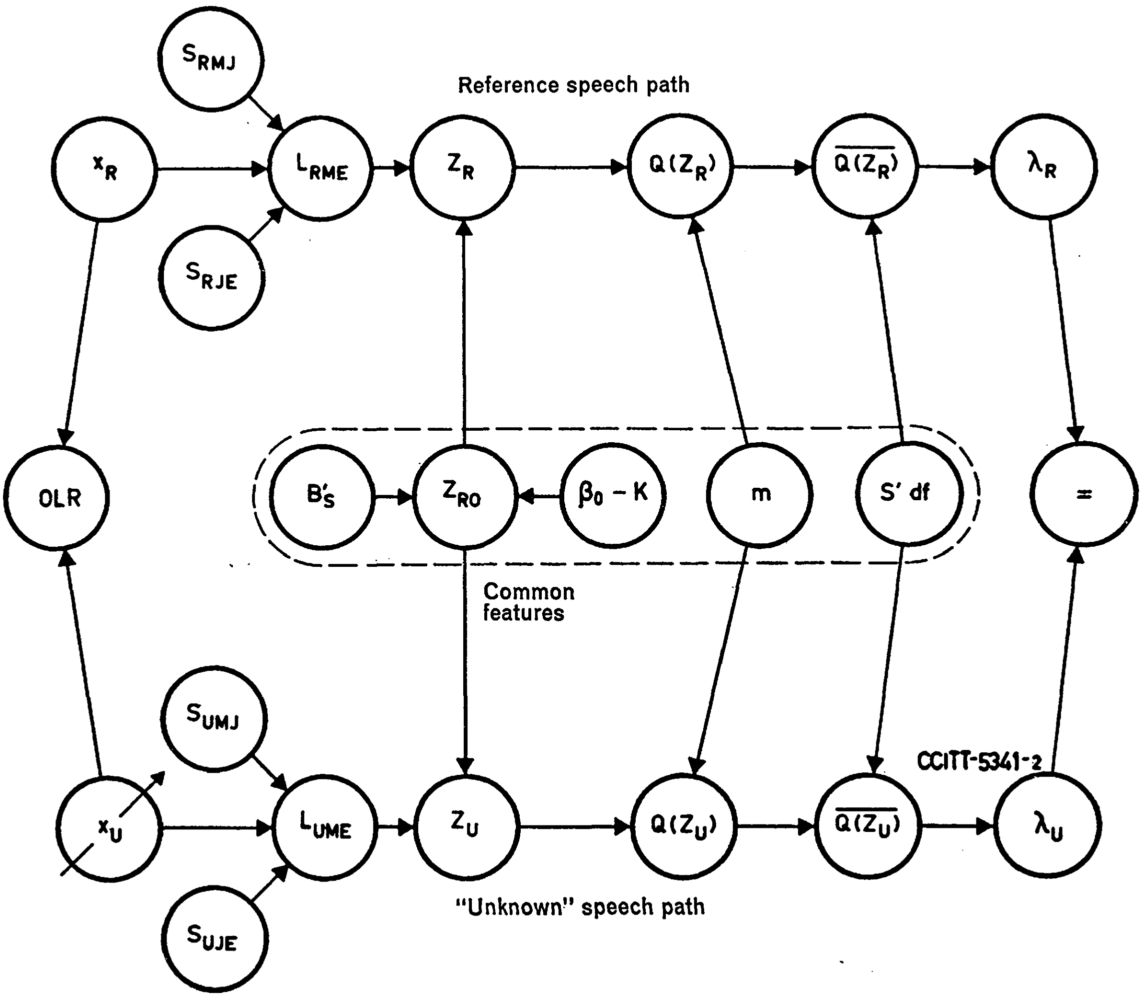


FIGURE 1.3. — **Flow diagrams illustrating determination of loudness ratings**

Unfortunately achievement of c) precludes agreement between sidetone loudness rating and sidetone reference equivalent and so it must be accepted that S.T.L.R. = S.T.R.E. + 2.5 dB approximately and so new values must be set for sidetone. Echo and crosstalk paths are usually characterized in practice by their values of *Nominal* O.R.E. and so they will remain almost unchanged in numerical values if based on S.L.R. and R.L.R. instead of S.R.E. and R.R.E.

b) Sending loudness rating (S.L.R.)

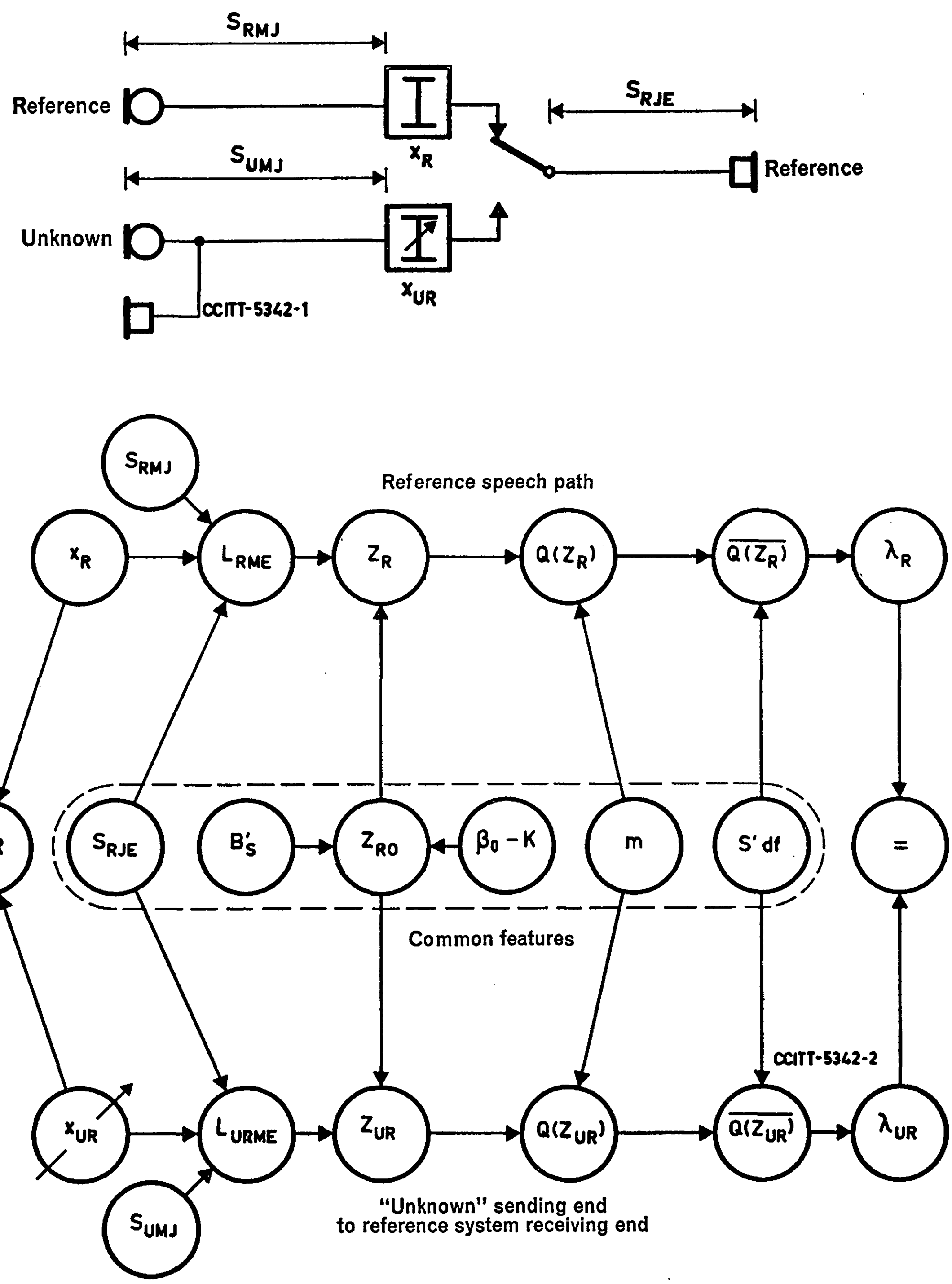
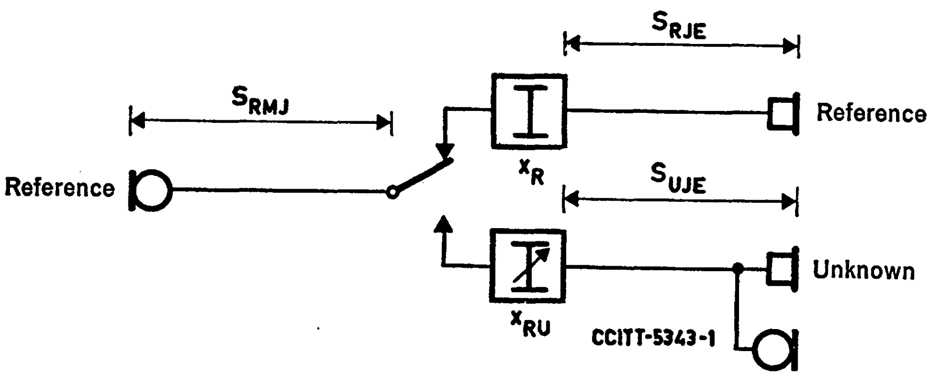


FIGURE 1.3. — *(continued)*

Columns (2) and (4) of Table 1.2 show the theoretical values of sending and receiving sensitivities that have been chosen provisionally by the United Kingdom Post Office for an I.R.S. intended to satisfy the above equalities.

Column (3) shows the values of receiving sensitivity likely to be found when such a receiving system is measured using an artificial ear; the symbol S_{RJE} is used to distinguish the values from those applicable when real-ear sensitivities are being discussed.

c) Receiving loudness rating (R.L.R.)



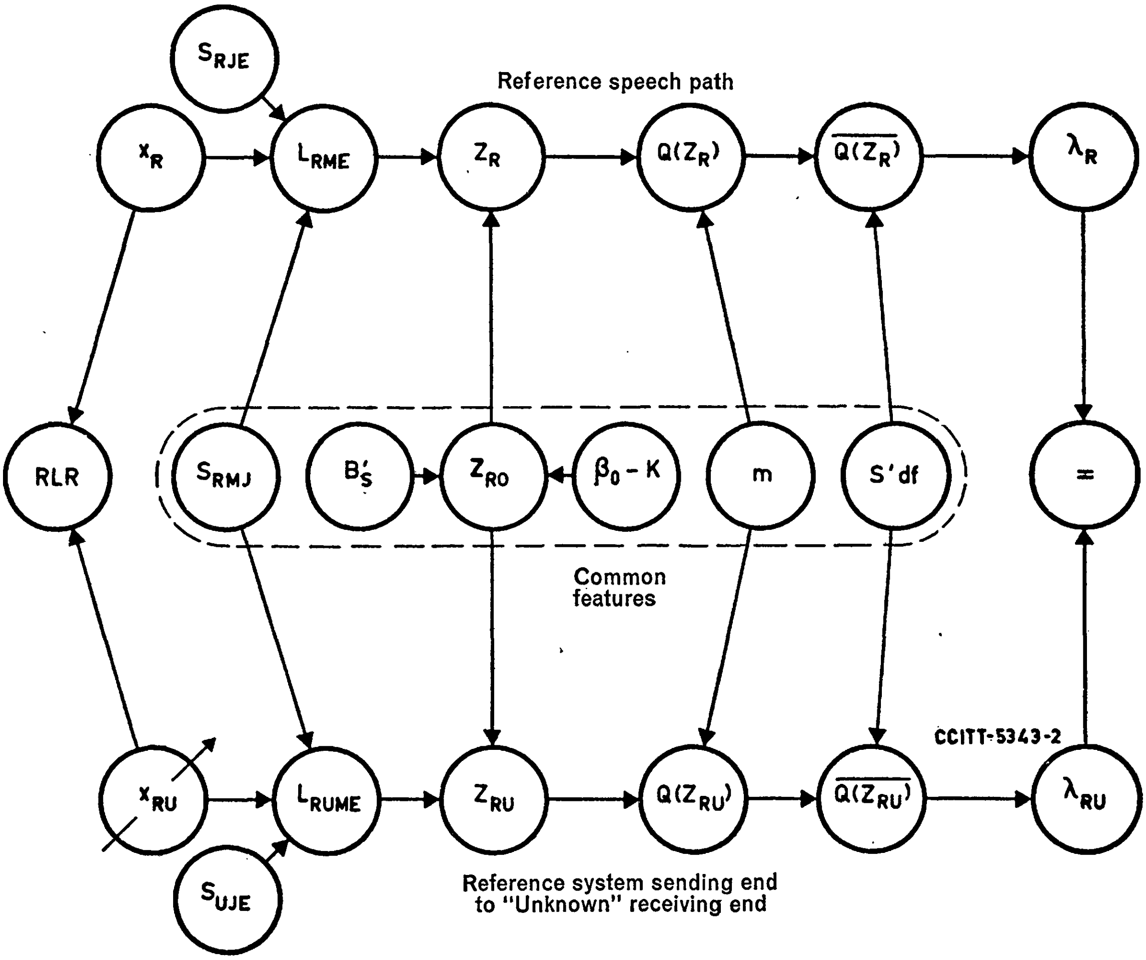


FIGURE 1.3. — *(concluded)*

APPENDIX 1-1

Definition of a new speaking position for testing telephone handset microphones

A method is described in Section 8 of Reference 12 (in Appendix 2 of the Preface) for measuring the sensitivities of commercial telephone sets; this method is suitable for the present purposes and has been used to determine all the sensitivities referred to in this paper. This Appendix describes in more detail the speaking position which is referred to in the C.C.I.T.T. Laboratory Reports as the special guard ring position.

The definition of a speaking position falls into two parts; description of the relative positions of mouth opening and ear-canal opening on a "modal" human head; and description of the angles that define the attitude in space of telephone handsets held to a "modal" head. For any given telephone handset, these descriptions together describe the relative special disposition of the microphone opening and the talker's lips, and hence the direction in which speech sound waves arrive at the mouthpiece and the distance they have travelled from a "virtual point source".

The relative positions of the centre of the opening of the mouth and that of the ear-canal can be described in terms of a distance δ and an angle α as shown in Figure 1.4. Point R in that figure represents the centre of a lip-ring located at the reference equivalent speaking position in accordance with C.C.I.T.T. Recommendation P.72. Position A is that used to determine ratings by the articulation method defined in C.C.I.T.T. Recommendation P.45. The approximately elliptical area encloses about 80% of the lip positions found in a sample of 3889 heads in the United States of America before 1930; averages of more recent, but less extensive and more diversified, results cluster round the point A.

A sound angle is required to define the direction in which speech is emitted from the mouth into the mouthpiece of the microphone. It is difficult to measure this precisely and it may be defined in various ways. In C.C.I.T.T. Recommendations P.45 and P.72, reference is made to an angle β which lies in the plane passing through both ear openings and the centre of the lips; β is defined in Recommendation P.72 as:

$$\beta = \arcsin \frac{\varepsilon}{\delta} - \alpha$$

where ε is the "semi-interaural" distance, i.e. half the length of a line joining the centres of the two ears; this line is, of course, perpendicular to the medial plane of the head and so ε is the length of a perpendicular from the centre of the plane of the earcap (Point 0 in Figure 1.4) on the medial plane of the head.

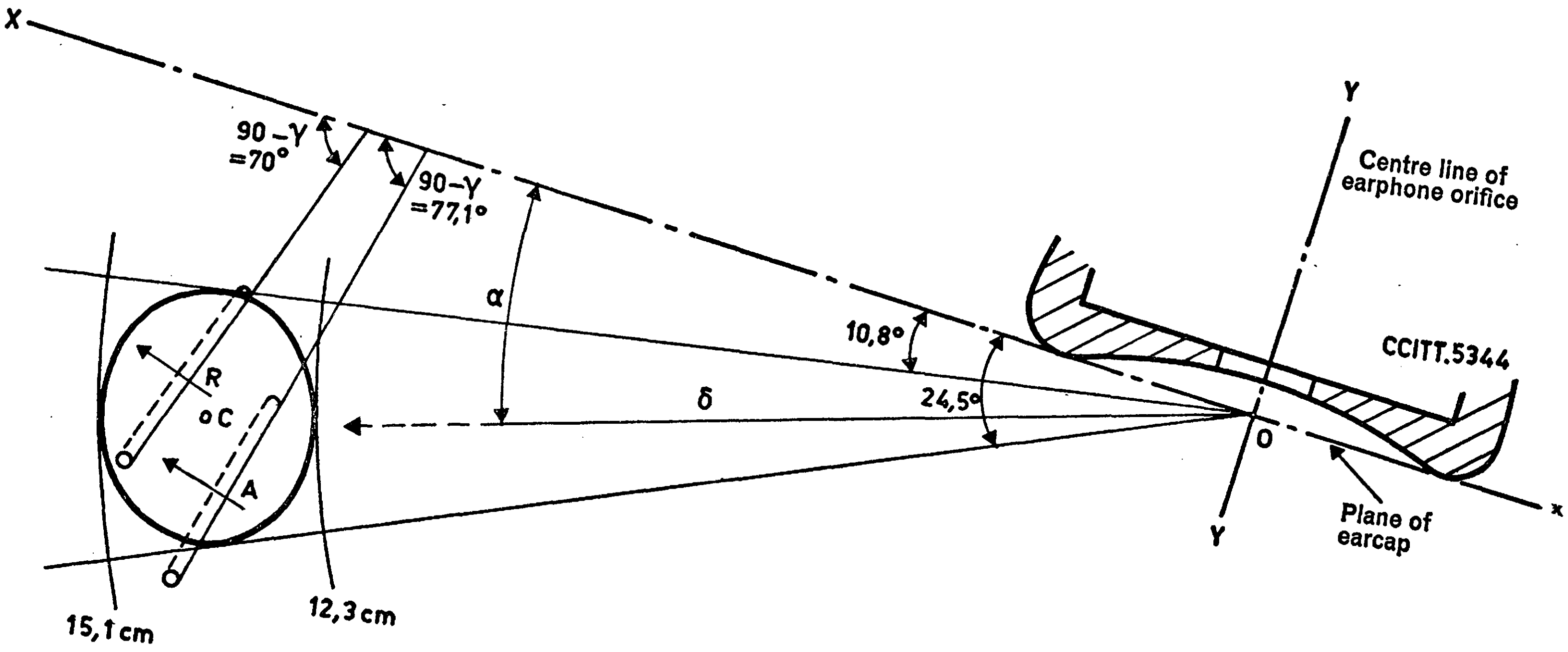
The plane referred to in defining β does not coincide with the plane of symmetry of the handset[1] which must be used in practice for defining the position of a lip-ring for attachment to the mouthpiece of the microphone. It is therefore more convenient to use an angle γ lying in the plane of symmetry of the handset as shown in Figure 1.4. The direction of speech may be taken to lie in the medial plane of the head, although the precise direction in this plane is difficult to determine; for many test purposes it can be assumed to be horizontal. The line, in the plane of symmetry of the handset, formed by the intersection of this plane with the (vertical) medial plane of the head can be specified by its angle with the line XX in Figure 1.4 and this angle is denoted by γ. This line that defines γ can therefore be considered as being the vertical projection of the direction of speech on the plane of symmetry of the handset.

Although β has been defined in this manner and appears in the descriptions of speaking positions given in C.C.I.T.T. Recommendations P.45 and P.72, actual setting-up procedures usually adopted treat the values quoted for β as though they applied for γ. If we treat the matter strictly and take $\beta = 12.9°$ (Recommendation P.45), $\varepsilon = 77.8$ mm, γ would be about 15°. Taking $\gamma = 12.9°$, as has been customary, when $\Theta = 19°$, $\varepsilon = 73.8$ mm and $\beta = 10.9°$.

Fortunately this inconsistency is of no practical importance because such small differences affect the sensitivity values of telephone handset microphones to a negligible extent. In order to avoid continuing this confusion, the symbol γ is employed here and is defined as shown in Figure 1.4, i.e. as lying in the plane of symmetry of the handset and not in a plane that is perpendicular to the medial plane of the head.

The position of the centre of the lips as defined by A in Figure 1.4 is used also to define the new speaking position, but two additional angles must also be defined, namely: the earphone rotational angle Φ and the handset

[1] These two planes are inclined to each other at the angle Θ referred to hereafter.



Notes:

1. — Points R and A are located as follows:
   A δ = 13.6 cm α = 22° γ = 12.9°
   R δ = 14.0 cm α = 15.5° γ = 18°
2. — Area shown includes about 80% of the sample of 3889 lip positions.
3. — A straight line tangential to the boundary of the area shown excludes 5% of lip positions.
4. — Solid line shows plane of lips; broken line shows lip-ring 1.6 mm thick.

FIGURE 1.4. — **Location of lip position relative to opening of ear canal**

rotational angle Θ. Earphone rotation is considered about an axis through the centre of the earcap (YY in Figure 1.4); handset rotation is taken about a longitudinal axis of the handset (XX in Figure 1.4); both angles are zero when the plane of symmetry of the handset is horizontal. Naturally, the earphone rotational angle is positive when the handle is pointed downwards away from the earphone and the handset rotational angle is positive in the sense that the upper part of the earphone is moved farther from the medial plane of the head.

The new speaking position is described by the following values for the distance and angles defined above:

$$\alpha = 22°, \gamma = 12.9°, \delta = 136 \text{ mm}, \Phi = 37° \text{ and } \Theta = 19°.$$

γ cannot be determined very precisely and is not convenient for use when setting up a handset for test in front of an artificial mouth and so the "semi-interaural" distance ε may be used in its place. For the new speaking position $\varepsilon = 73.8$ mm.

PART 2. — Subjective testing method

In the subjective comparisons, the Fundamental Reference System (F.R.S.) is used as a datum for comparing the following speech paths:

Path 1. — The fundamental reference system always provides the speech path against which each of the others is balanced. The N.O.S.F.E.R. or the A.R.A.E.N. could be used.

Path 2. — Is the send end of the intermediate reference system connected through an adjustable attenuator to the receive end of the intermediate reference system (I.R.S.)?

Path 3. — Is the send end of the test item ("unknown" local telephone circuit) connected through an adjustable attenuator to the receive end of the I.R.S.?

Path 4. — Is the send end of the I.R.S. connected through an adjustable attenuator to the receive end of the test item ("unknown" local telephone circuit)?

Path 5. — Is the send end of the test item ("unknown") connected through an adjustable attenuator to the receive end of the test item (another example of the "unknown")?

In these subjective comparisons, the junction of the fundamental reference system is fixed at a value to be decided[1], i.e. the level of speech sounds received via the fundamental reference system is kept constant, the loudness balance being obtained by the so-called "margin" method, the balance attenuator being that inserted in the telephone (or I.R.S.) path being tested.

Figure 2.1 shows the composition of the telephone paths to be compared. The balances should be conducted using the vocal level defined in Appendix 4-1.

The speaking position used with both the I.R.S. and the test telephone sets should be as defined in Appendix 1-1.

Paths 2, 3, 4 and 5 of Figure 2.1 are in turn balanced with respect to the fundamental reference system (Path 1) with attenuations of x_2, x_3, x_4 and x_5 respectively (x_1 is fixed).

The loudness ratings relative to the Fundamental Reference System are given by:

$$\begin{aligned}
(1) &= x_1 - x_2 = i_2 \\
\text{A} &= x_1 - x_3 = i_3 \\
\text{B} &= x_1 - x_4 = i_4 \\
\text{AB} &= x_1 - x_5 = i_5
\end{aligned}$$

The subjective ratings of the "unknown" local telephone circuit relative to the I.R.S. are given by:

Send rating

$$a = \tfrac{1}{2}[\text{A} + \text{AB} - (1) - \text{B}] = \tfrac{1}{2}[i_3 + i_5 - (i_2 + i_4)] = \tfrac{1}{2}[x_2 + x_4 - (x_3 + x_5)]$$

---

[1] This should be equivalent, in received speech level, to the setting of 25 dB in N.O.S.F.E.R. when the talker is speaking at the vocal level used to determine reference equivalents (Recommendation P.72).

Receive rating

$$b = \tfrac{1}{2}[B + AB - (1) - A] = \tfrac{1}{2}[i_4 + i_5 - (i_2 + i_3)] = \tfrac{1}{2}[x_2 + x_3 - (x_4 + x_5)]$$

An interaction may be defined as:

$$ab = \tfrac{1}{2}[AB + (1) - A - B] = \tfrac{1}{2}[i_2 + i_5 - (i_3 + i_4)] = \tfrac{1}{2}[x_3 + x_4 - (x_2 + x_5)]$$

If the interaction is zero, alternative simpler forms of send and receive ratings can be used as follows and will be identical:

$$\begin{aligned} a' &= [A - (1)] = i_3 - i_2 = x_2 - x_3 \\ \text{or } a'' &= (AB - B) = i_5 - i_4 = x_4 - x_5 \\ \text{and } b' &= [B - (1)] = i_4 - i_2 = x_2 - x_4 \\ \text{or } b'' &= (AB - A) = i_5 - i_3 = x_3 - x_5 \end{aligned}$$

It will be seen that the value of x_1, the setting of the Fundamental Reference System, does not affect the evaluation of these ratings.

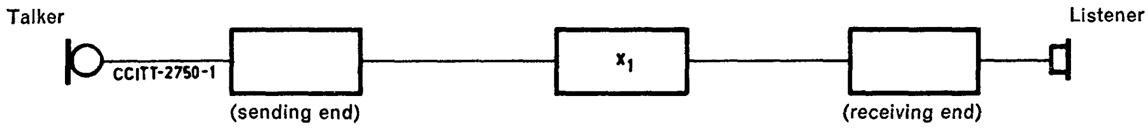
Tables 2.1, 2.2 and 2.3 show the results of certain tests conducted by the C.C.I.T.T. Laboratory in the manner described. The A.R.A.E.N. was used as the Fundamental Reference System and was set at 30 dB. The tables are arranged to facilitate calculation of the various ratings according to the foregoing expressions. The results from these tables may be summarized as follows:

TABLE 2.4

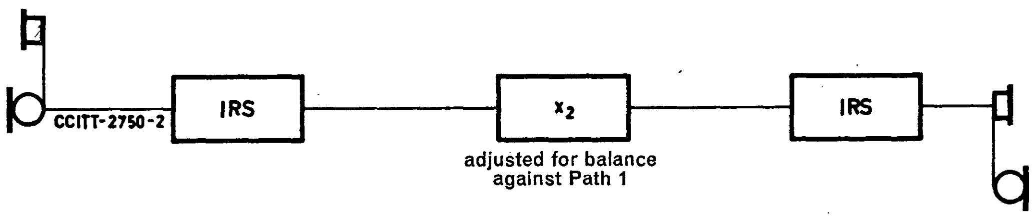
SUMMARY OF LOUDNESS RATINGS OBTAINED BY THE PROPOSED SUBJECTIVE TESTING METHOD[a]

"Unknown" local telephone circuit	Loudness rating		Interaction *ab* dB
	Sending *a* dB	Receiving *b* dB	
(1)	(2)	(3)	(4)
A.T.T. 1	+17.9	+4.2	−0.4
2	+17.1	+2.4	−0.1
U.K. 1	+ 8.8	−2.3	−0.1
2	+ 8.2	+0.3	−0.1
Swiss 1	+ 2.1	+0.2	−0.2
2	+ 2.7	+0.3	−0.5

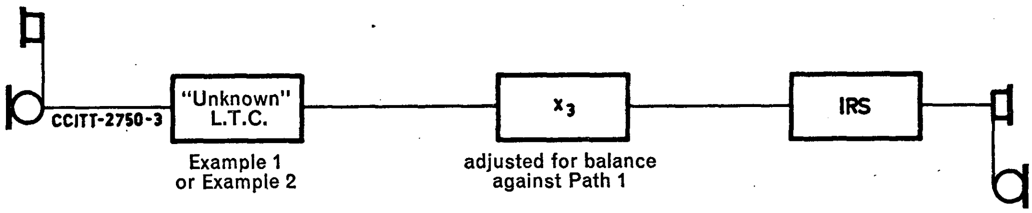
[a] These experiments used an intermediate reference system (I.R.S.) that differed slightly from that proposed in Section 1.5.



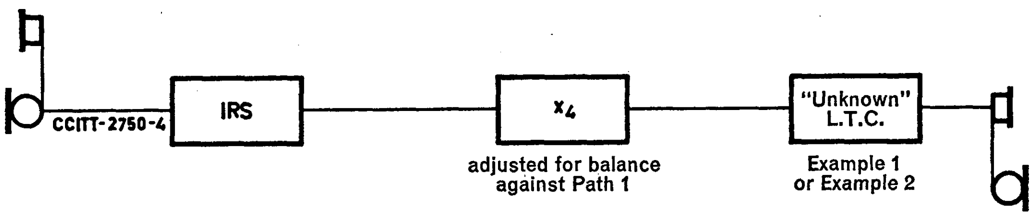
Path 1. — Fundamental reference system



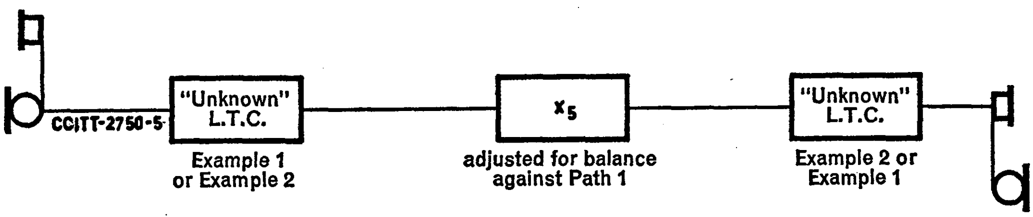
Path 2. — Intermediate reference system (I.R.S.)



Path 3. — "Unknown"/I.R.S.



Path 4. — I.R.S./"Unknown"



Path 5. — "Unknown"/"Unknown"

FIGURE 2.1. — **Arrangement of paths for subjective loudness ratings of two examples of one type of telephone system**

TABLE 2.1

A.R.A.E.N./30 dB RATINGS OF THE A.T.T. TELEPHONE SYSTEM (ASSOCIATED WITH I.R.S.)

Paths	i	A.R.A.E.N/ 30 dB rating	Subjective ratings relative to I.R.S.						Interaction
			Sending			Receiving			
			$i_3-i_2=a'$	$i_5-i_4=a''$	$\frac{1}{2}[i_3+i_5-(i_2+i_4)] = a$	$i_4-i_2 = b'$	$i_5-i_3 = b''$	$\frac{1}{2}[i_4+i_5-(i_2+i_3)] = b$	$\frac{1}{2}[i_2+i_5-(i_3+i_4)] = ab$
			1	2	3	4	5	6	7
S.R.I./S.R.I.	i_2	+ 7.72							
C_1/S.R.I.	i_3	+26.00	+18.28	+17.57	*+17.92*				
S.R.I./C_2	i_4	+10.43				+2.71	+2.00	*+2.36*	
C_1/C_2	i_5	+28.00							−0.35
C_2/S.R.I.	i_3	+24.93	+17.21	+17.03	*+17.12*				
S.R.I./C_1	i_4	+12.05				+4.33	+4.15	*+4.24*	
C_2/C_1	i_5	+29.08							−0.09

C.C.I.T.T.-4429

With filter in the commercial system and the I.R.S.

TABLE 2.2

UNITED KINGDOM TELEPHONE SYSTEM A.R.A.E.N./30 dB RATINGS (ASSOCIATED WITH I.R.S.)

Paths	i	A.R.A.E.N/ 30 dB rating	Subjective ratings relative to I.R.S.						Interaction
			Sending			Receiving			
			$i_3-i_2 = a'$	$i_5-i_4 = a''$	$\frac{1}{2}[i_3+i_5-(i_2+i_4)] = a$	$i_4-i_2 = b'$	$i_5-i_3 = b''$	$\frac{1}{2}[i_4+i_5-(i_2+i_3)] = b$	$\frac{1}{2}[i_2+i_5-(i_3+i_4)] = ab$
			1	2	3	4	5	6	7
S.R.I./S.R.I.	i_2	+8.63							
C_1/S.R.I.	i_3	+17.48	+8.85	+8.70	*+8.78*				
S.R.I./C_2	i_4	+ 9.00				+0.37	+0.22	*+0.30*	
C_1/C_2	i_5	+17.70							−0.08
C_2/S.R.I.	i_3	+16.87	+8.24	+8.12	*+8.18*				
S.R.I./C_1	i_4	+ 6.35				−2.28	−2.40	*−2.34*	
C_2/C_1	i_5	+14.47							−0.06

C.C.I.T.T.-4429

With filter in the commercial system

TABLE 2.3

A.R.A.E.N./30 dB RATINGS OF THE SWISS TELEPHONE SYSTEM (ASSOCIATED WITH I.R.S.)

Paths	i	A.R.A.E.N. 30 dB rating	Subjective ratings relative to I.R.S.						Interaction
			Sending			Receiving			
			$i_3-i_2=a'$	$i_5-i_4=a''$	$\frac{1}{2}[i_3+i_5-(i_2+i_4)]=a$	$i_4-i_2=b'$	$i_5-i_3=b''$	$\frac{1}{2}[i_4+i_5-(i_2+i_3)]=b$	$\frac{1}{2}[i_2+i_5-(i_3+i_4)]=ab$
			1	2	3	4	5	6	7
S.R.I./S.R.I.	i_2	+7.62							
C_1/S.R.I.	i_3	+9.92	+2.30	+1.83	*+2.06*				
S.R.I./C_2	i_4	+8.17				+0.55	+0.08	*+0.31*	
C_1/C_2	i_5	+10.00							−0.23
C_2/S.R.I.	i_3	+10.80	+3.18	+2.15	*+2.66*				
S.R.I./C_1	i_4	+8.30				+0.68	−0.35	*+0.16*	
C_2/C_1	i_5	+10.45							−0.51

C.C.I.T.T.-4429

With filter in the commercial system and I.R.S.

PART 3. — Determination of sensitivity/frequency characteristics

3.1 *General*

The sensitivity/frequency characteristics described herein relate to local telephone circuits (telephone set, subscriber's line and feeding bridge) and are required for the purposes of calculating sending and receiving loudness ratings in accordance with the principles described in Part 1.

The sending sensitivity is measured from the sound pressure under free-field conditions at a mouth reference point (defined relative to the lips or equivalent position of the lips) to the voltage across a 600 ohms load representing the "junction" that terminates the local telephone circuit.

The receiving sensitivity is measured from the "junction", represented by a source having 600 ohms internal resistance, to an ear reference point located at a defined position at the opening of the ear canal. The input voltage is taken as one half the emf of the source, i.e. the voltage that would be obtained when the source is terminated with a matched load.

In principle, these sensitivities ought to be determined using real mouth, real voices and real ears; in practice these must be simulated by an artificial mouth, a test signal and an artificial ear. Desirable features of these are given below.

3.2 *Artificial mouth*

The following properties are required:

a) The distribution in sound pressure around the orifice must be a good approximation to that around a human mouth.

b) The acoustical impedance looking back to the mouth must simulate that for human mouths so that the pressure increase caused by the obstruction effect of telephone microphones will be representative.

c) It must be possible to establish at the mouth reference point definite sound pressures as a function of frequency. A convenient feature to embody in a practical artificial mouth is to ensure that the ratio of sound pressure at the mouth reference point to the voltage input to the artificial mouth is linear over a suitable range of sound pressure and must be independent of frequency at least over the range 200 to 4000 Hz.

For the present purposes the mouth reference point is defined by the point on the axis of the artificial mouth located 25 mm in front of the equivalent lip position.

3.3 *Artificial ear*

The following properties are required:

a) The acoustical impedance presented to telephone earphones must simulate that presented by real ears under practical conditions of use of telephone handsets.

b) The sensitivity of the artificial ear, namely the ratio of voltage output to sound pressure in the coupler of the artificial ear, shall be uniform with frequency at least from 200 to 4000 Hz.

For a human ear, the ear reference point is defined as the centre of the plane of a circular telephone earcap when it is placed comfortably against the ear. The corresponding point when the earcap is fitted to an artificial ear will usually differ from the place at which the sound pressure is measured and for this and other reasons certain corrections are necessary when a telephone earphone is measured on an artificial ear. The amount of such corrections is at present the subject of study but it is usually found necessary to decrease sensitivities measured with the C.C.I.T.T./I.E.C. artificial ear by amounts that are frequency dependent but average (on a loudness basis) about 3.5 dB. (The symbol L_E will be used for this "earphone coupling loss".)

3.4 *Definition of sending sensitivity of an L.T.C.*

The sending sensitivity of a local telephone circuit (L.T.C.) depends upon the location of the handset relative to the equivalent lip position of the artificial mouth. For the present purposes the "special guard ring" speaking position as defined in Appendix 1-1 shall be used.

The sending sensitivity of a local telephone circuit is expressed in dB as follows:

$$S_{MJ} = 20 \log_{10} \frac{\text{voltage, } V_J\text{, across 600 ohms termination}}{\text{sound pressure, } p_M\text{, at mouth reference point}}$$

Note that p_M must be measured in the absence of the test microphone of the L.T.C.

The units of S_{MJ} are dB relative to 1 V per N/m^2.

When measuring L.T.C.s that contain no non-linear items (e.g. without carbon microphones), it does not matter at which sound pressure the measurements are made so long as it is known; when, for example, carbon microphones are present, different sensitivities will be obtained depending upon the sound pressure used. For calculation of sending loudness rating, these must be reduced to single values at each frequency and the method of reduction must take account of the characteristics of human speech. Various methods have been suggested but the following simple method seems to yield acceptable results:

a) Determine the sensitivity as a function of frequency at the sound pressure level of —4.7 dB relative to 1 N/m^2; this corresponds to the mean power while active of a talker emitting speech at reference vocal level (see Recommendation P.45). Note that R.V.L. is 5.6 dB higher than the vocal level used to determine reference equivalents in accordance with Recommendation P.72.

b) Repeat a) but with the sound pressure level increased by 10 dB.

c) Repeat a) but with the sound pressure level decreased by 10 dB.

d) Select from a), b) and c) the highest sensitivity at each frequency.

It goes without saying that carbon microphones must be given appropriate conditioning treatment at suitable intervals during the measurements.

3.5 *Definition of receiving sensitivity of an L.T.C.*

The receiving sensitivity of a local telephone circuit is expressed in decibels as follows:

$$S_{JE} = 20 \log_{10} \frac{\text{sound pressure, } p_e\text{, in the artificial ear}}{\frac{1}{2}\text{emf in the 600 ohms source } (E_J/2)} - L_E$$

Note that p_e differs from p_E, the sound pressure at the ear reference point, P_E, in a median human ear; L_E is the allowance to correct for this difference, L_E is a function of frequency and probably depends upon the type of earcap for information, the following values were found for the present type of earphone used in the U.K.:

200 Hz +8.4	1250 Hz —1.2
300 Hz +1.3	1600 Hz —0.1
400 Hz —0.7	2000 Hz +3.6
500 Hz —2.2	2500 Hz +7.4
600 Hz —2.5	3000 Hz +6.3
800 Hz —3.2	3500 Hz +7.5
1000 Hz —2.3	4000 Hz +8.8

These values are at present under review.

For information, a local telephone circuit having 0 dB S.L.R. and 0 dB R.L.R. will yield, at 1000 Hz, values of sending and receiving sensitivities of the order:

$$S_{MJ} = 0 \text{ dB relative to 1 } V \text{ per } N/\text{m}^2$$

$$\text{and } S_{JE} = +12 \text{ dB relative to 1 } N/\text{m}^2 \text{ per } V$$

3.6 *Methods for determining S_{MJ} and S_{JE}*

When the sending and receiving sensitivities of an actual local telephone system are required, the measurements according to the definitions given in Sections 3.4 and 3.5 can be made as illustrated in Figures 3.1, 3.2 and 3.3. These methods have been used by the C.C.I.T.T. Laboratory.

Figure 3.1 shows the method of setting up the artificial mouth so that the sound pressure p_M at the mouth reference point is known at each test frequency.

Figure 3.2 shows the measurement of output V_J from the local telephone circuit when the microphone is placed at the appropriate position in front of the artificial mouth and the artificial mouth is energized in the same manner as when the sound pressure p_M was set up in the absence of the test microphone (Figure 3.1).

Figure 3.3 shows the measurement of the sound pressure p_e in the artificial ear when the local telephone circuit is connected to a 600-ohm source of internal e.m.f. E_J. Note that the definition of S.J.E. is in terms of $\frac{1}{2}E_J$ and not the potential difference across the input terminals of the local telephone circuit; this potential difference will, of course, differ from $\frac{1}{2}E_J$, if the input impedance of the local telephone circuit is not 600 ohms.

The correction term L_E must be added to the sensitivity obtained using a conventional artificial ear such as the C.C.I.T.T./I.E.C. artificial ear (see Recommendation P.51). If the measurement is made on the receiving end of A.R.A.E.N. or N.O.S.F.E.R., and the special artificial ear described in Recommendation P.41 is used, the result will be practically the same as would be obtained with a sample of real ears and so L_E would be zero. If suitable modifications to conventional artificial ears are found from study of artificial ears so that they provide a proper representation of real ears, the term L_E would become unnecessary also when commercial local telephone circuits are measured.

If the complete local telephone circuit is not actually available, it will be necessary to estimate the sensitivity, sending and receiving, by combination of the sensitivities and transmission losses of the component parts. For example, in Figure 1.2a, S_{MJ} is shown as made up of the following components:

S_M = sensitivity of the microphone;

L_S = transmission loss, sending, of the electrical part of the telephone set;

$L\ (SL + FB)$ = transmission loss of the combination of subscriber's line and feeding bridge.

Similarly, S_{JE} is shown as composed of the following:

S_E = sensitivity of the earphone;

L_R = transmission loss, receiving, of the electrical part of the telephone set;

$L\ (SL + FB)$ = see above.

Suitably defined, the sensitivities and losses of the separate components can be combined algebraically to yield the sending and receiving sensitivities, S_{MJ} and S_{JE} that were defined in Sections 3.4 and 3.5; proper allowances must be made for any impedance mismatches.

The decomposition described above is convenient for the treatment of most current types of telephone set containing a transformer induction coil and transducers having relatively low electrical impedances which are approximately matched to the circuit and are without any amplification (except that provided by the operation of the carbon microphone).

New types of telephone set containing, for example, amplification in the electrical paths, will require to be treated differently. Each case will have to be treated on its merits to ensure that the overall sensitivities, S_{MJ} and S_{JE} of the local telephone circuits conform to the definitions given in Sections 3.4 and 3.5.

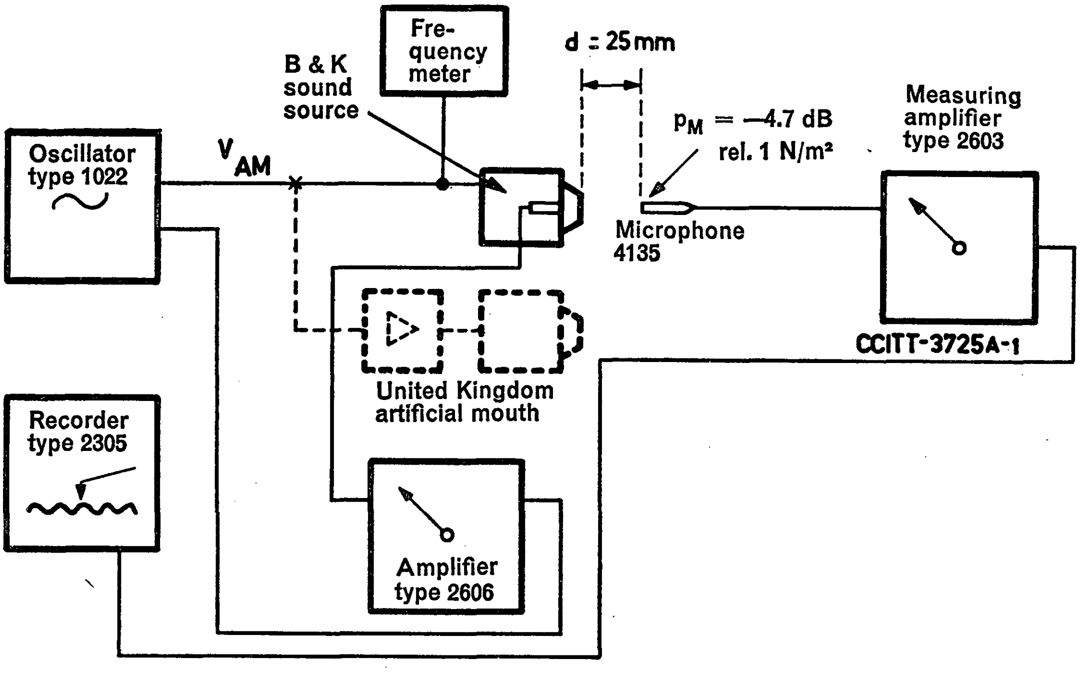


FIGURE 3.1. — Measurement of acoustic pressure p_M at the reference point 25 mm from the "artificial lip" plane of either the B and K 4216 sound source (as modified by the C.C.I.T.T. Laboratory) or United Kingdom artificial mouth

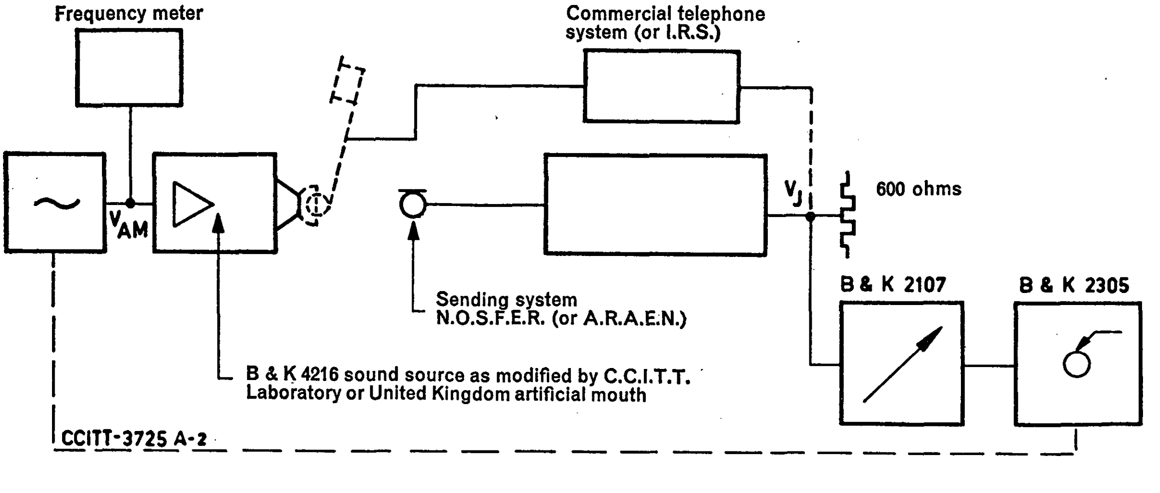


FIGURE 3.2. — Voltages V_J measured across the terminals of a 600 ohms pure resistance connected to the output of the N.O.S.F.E.R. (or A.R.A.E.N.) sending system (or of a commercial sending system) in dB relative to 1 volt

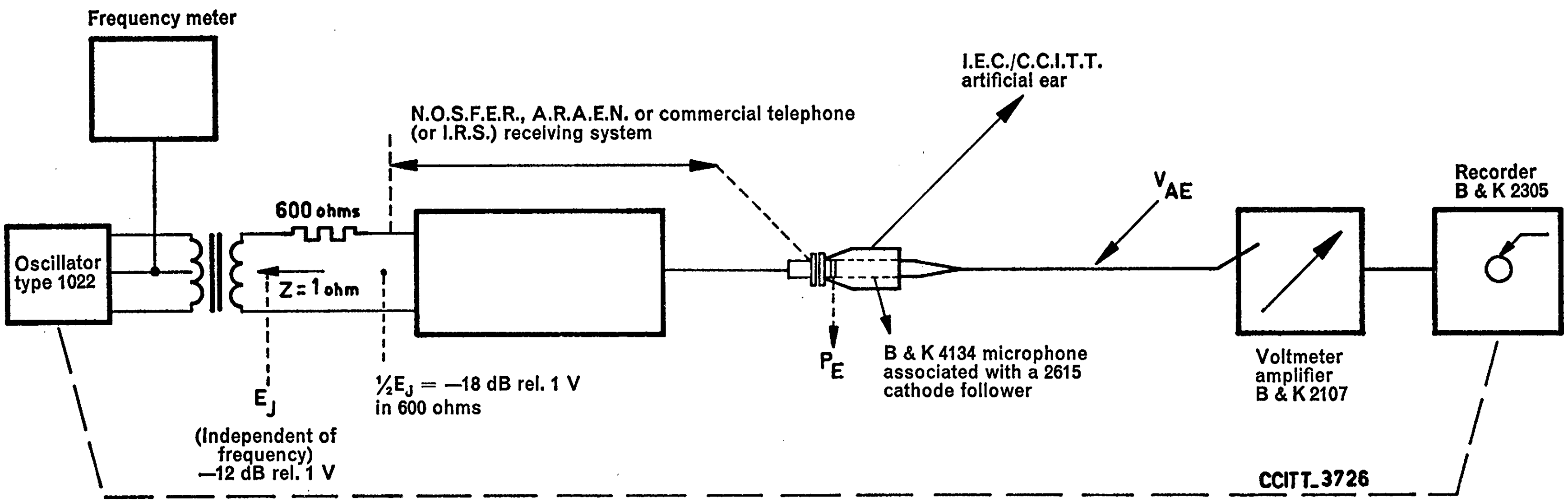


FIGURE 3.3. — Calibration of the N.O.S.F.E.R., A.R.A.E.N. or commercial telephone receiving system

PART 4. — Calculation of loudness ratings

It will be clear from the information given in Parts 1, 2 and 3 that loudness ratings could be determined without recourse to subjective tests provided that the following conditions were fulfilled:

a) The mathematical model were accepted and all the parameters were known;

b) The sending and receiving sensitivities, S_{UMJ} and S_{UJE}, have been determined under the appropriate conditions.

The general principles explained in Part 1 together with the specific definitions given in Section 1.5 can be used as the basis for calculation of loudness ratings. The relationships illustrated in Figures 1.3 (a), 1.3 (b) and 1.3 (c) can be adapted into forms more convenient for computation as shown below.

Equation 1.1 can be written:

$$\lambda_U = C \int Q\,(Z_U)\, S'\, \mathrm{df} \quad \dots \tag{4.1a}$$

$$\text{and } \lambda_R = C \int Q\,(Z_R)\, S'\, \mathrm{df} \quad \dots \tag{4.1b}$$

where λ_U and λ_R represent the loudness of speech received through the "unknown" and reference speech paths respectively and Z_U and Z_R are the corresponding values of sensation level (which are functions of frequency).

The calculation method to be described depends upon the assumption (largely verified for restricted ranges of listening level) that the function $Q(Z)$ can be put in the form:

$$Q\,(Z) = \text{constant} \cdot 10^{m\,(1/10)\,Z} \quad \dots \tag{4.2}$$

(These base 10 and the multiplier 1/10 are used merely to preserve the analogy to the decibel, in which unit Z is expressed.)

$$\text{Let } Z_{RO} = B'_S - (\beta_0 - \mathrm{K}) \quad \dots \tag{4.3}$$

and substitute in Equation 1.4 to obtain:

$$Z_U = Z_{RO} - L_{UME} \quad \dots \tag{4.4a}$$

$$Z_R = Z_{RO} - L_{RME} \quad \dots \tag{4.4b}$$

By substituting Equations (4.4a) and (4.4b) in Equations (4.1a) and (4.1b) and rearranging:

$$\lambda_U = C \int 10^{-m\,(1/10)\,L_{UME}}\,[10^{m\,(1/10)\,Z_{RO}}\, S']\, \mathrm{d}f \quad \dots \tag{4.5a}$$

$$\lambda_R = C \int 10^{-m\,(1/10)\,L_{RME}}\,[10^{m\,(1/10)\,Z_{RO}}\, S']\, \mathrm{d}f \quad \dots \tag{4.5b}$$

The loudness rating can be considered to be the loss Δx (independent of frequency) removed from the "unknown" speech path to render $\lambda_U = \lambda_R$.

Using the substitution:

$$G = [10^{m\,(1/10)\,Z_{RO}}\, S'] \quad \dots \tag{4.6}$$

and inserting $L_{UME} - \Delta x$ in Equation (4.5a) in place of L_{UME}, we obtain equality of the λ's.

$$\text{Therefore } \int 10^{-m\,(1/10)\,(L_{UME} - \Delta x)}\, G\, \mathrm{d}f = \int 10^{-m\,(1/10)\,L_{RME}}\, G\, \mathrm{d}f \quad \dots \tag{4.7}$$

$$10^{-m\,(1/10)\,\Delta x} = \frac{\int 10^{-m\,(1/10)\,L_{UME}}\, G\, \mathrm{d}f}{\int 10^{-m\,(1/10)\,L_{RME}}\, G\, \mathrm{d}f} \quad \dots \tag{4.8}$$

$$\text{and } \Delta x = -\frac{1}{m}\, 10 \log_{10} \int 10^{-m\,(1/10)\,L_{UME}}\, G\, \mathrm{d}f - \left\{ -\frac{1}{m}\, 10 \log_{10} \int 10^{-m\,(1/10)\,L_{RME}}\, G\, \mathrm{d}f \right\} \quad \dots \tag{4.9}$$

Without affecting the equality, G can be scaled by multiplying by a suitable constant to render $\int G\, \mathrm{d}f = 1$; G can then be treated as a weighting factor and each term on the right-hand side takes the form:

$$\Phi^{-1}[\int \Phi\,(L)\, G\, \mathrm{d}f] = \overline{L}$$

$$\text{Then the loudness rating} = \Delta x = \overline{L_{UME}} - \overline{L_{RME}} \quad \ldots \tag{4.10}$$

The terms $\overline{L_{UME}}$ and $\overline{L_{RMR}}$ can be considered as the "weighted average mouth to ear loss" of the "unknown" and reference speech paths respectively. In each of the foregoing equations, integration (and therefore averaging) is over the range between lower and upper frequency limits of interest.

For computation, it is convenient to divide the audible range of frequency into a number (N) of contiguous bands; for telephone bandwidths, 14 such bands have been chosen and these are represented by the set of frequencies shown in Table 4.1. Averaging the values of L_{UME} is then performed by summations of the form:

$$\overline{L_{UME}} = -\frac{1}{m} 10 \log_{10} \sum_{i}^{N} 10^{-m(1/10)L_{UME}} \, G\Delta f \quad \ldots \tag{4.11}$$

The acoustical transmission loss of a speech path is, in general, a function of frequency and can be defined as:

$$L_{UME} = 20 \log_{10} \frac{p_M}{p_E} \quad \ldots \tag{4.12}$$

where p_M and p_E are as defined in Sections 1.3.1 and 1.3.2.

It is necessary to know the values of L_{UME} at each frequency together with $G\Delta f$; naturally L_{UME} depends upon the telephone speech path under consideration but $G\Delta f$ and other information common to all speech paths is assembled in Table 4.1.

For rating telephone speech paths, m can be taken as 0.25 and this permits use of the substitution:

$$W_i = -40 \log_{10} G\Delta f \quad \ldots \tag{4.13}$$

Equation (4.11) can then be simplified in appearance to:

$$\overline{L_{UME}} = -40 \log_{10} \sum_{i}^{N} 10^{-(1/40)(L_{UME} + W_i)} \quad \ldots \tag{4.14}$$

For the present purposes, the reference speech path will be taken as the "intermediate reference system" (I.R.S.) defined in Section 1.5 of Part 1; having fixed the reference speech path, $\overline{L_{RME}}$ becomes constant, i.e. independent of i. Therefore Equations 4.10 and 4.14 can be combined to form:

$$\text{Loudness rating} = -40 \log_{10} \sum_{i}^{N} 10^{-(1/40)(L_{UME} - \overline{L_{RME}} + W_i)} \quad \ldots \tag{4.15}$$

When rating commercial local telephone circuits, the values of L_{UME} can be obtained for any given "unknown" speech path combining appropriate sending and receiving sensitivities, S_{MJ} and S_{JE}, in appropriate combinations.

For determining a sending loudness rating (*S.L.R.*) of a local telephone circuit,

$$L_{UME} = -(S_{UMJ} + S_{RJE}) \quad \ldots \tag{4.16a}$$

For determining a receiving loudness rating (*R.L.R.*) of a local telephone circuit,

$$L_{UME} = -(S_{RMJ} + S_{UJE}) \quad \ldots \tag{4.16b}$$

and for determining an "overall loudness rating" (*O.L.R.*) for a local telephone circuit,

$$L_{UME} = -(S_{UMJ} + S_{UJE}) \quad \ldots \tag{4.16c}$$

Substituting these in Equation 4.15:

$$SLR = -40 \log_{10} \sum_{i}^{N} 10^{(1/40)(S_{UMJ} + S_{RJE} + \overline{L_{RME}} - W_i)} \quad \ldots \tag{4.17a}$$

$$RLR = -40 \log_{10} \sum_{i}^{N} 10^{(1/40)(S_{UJE} + S_{RMJ} + \overline{L_{RME}} - W_i)} \quad \ldots \tag{4.17b}$$

$$OLR = -40 \log_{10} \sum_{i}^{N} 10^{(1/40)(S_{UMJ} + S_{UJE} + \overline{L_{RME}} - W_i)} \quad \ldots \tag{4.17c}$$

The terms $\overline{L_{RME}}$ and W_i are common to each of the Equations 4.17 and so further computational simplification is possible by making the following substitutions:

$$W_S = W_i - S_{RJE} - \overline{L_{RME}} \ \ldots \qquad (4.18a)$$

$$W_R = W_i - S_{RMJ} - \overline{L_{RME}} \ \ldots \qquad (4.18b)$$

$$W_O = W_i - \overline{L_{RME}} \ \ldots \qquad (4.18c)$$

Table 4.1 shows the values for these "weighting factors" together with certain other information useful in calculating loudness ratings relative to the intermediate reference system (I.R.S.) described in Section 1.5.3.

TABLE 4.1

WEIGHTING FACTORS AND OTHER INFORMATION USEFUL FOR CALCULATING LOUDNESS RATINGS

Band No. $i =$	Frequency Hz	Band width f Hz	$10 \log_{10} G\Delta f$	W_i	W_S	W_R	W_O
(1)	(2)	(3)	(4)	(5)	(6)	(7)	(8)
1	200	100	−13.9	55.6	56.8	79.6	60.8
2	300	100	−12.3	49.2	45.2	69.0	54.4
3	400	100	−11.9	47.6	42.2	64.4	52.8
4	500	100	−11.7	46.8	40.8	60.9	52.0
5	600	150	−10.6	42.4	36.1	53.8	47.6
6	800	200	−10.7	42.8	36.3	49.2	48.0
7	1000	225	−11.3	45.2	38.7	50.4	50.4
8	1250	300	−11.1	44.4	37.9	48.9	49.6
9	1600	375	−11.7	46.8	40.3	50.9	52.0
10	2000	450	−11.8	47.2	40.7	50.8	52.4
11	2500	500	−11.9	47.6	41.1	50.9	52.8
12	3000	500	−12.4	49.6	43.1	52.6	54.8
13	3500	500	−13.9	55.6	61.6	72.2	60.8
14	4000	750	−13.6	54.4	>79.6	>87.6	>59.6

Notes:

1. — The above 14 bands represent the frequency range from 150 to 4500 Hz; the boundaries of these bands are located mid-way between the frequencies shown in Col (2).

2. — The values of G used in Col (4) are scaled so that $\int G \, df = 1$ when integrated over the whole audible frequency range and not just that covered by the above 14 bands.

3. — As can be seen from the very large values of W_S and W_R, Band No. 14 makes very little contribution to *SLR* and *RLR* and may usually be ignored; this is a consequence of using a reference speech path very severely limited in response at the upper limit of frequency bandwidth.

4. — $\overline{L_{RME}} = -5.25$ dB.

An example of the application of these weighting factors to the calculation of loudness ratings will be found in Reference [14] given in Appendix 2 to the Preface. It was based on the results of sensitivity measurements made by the C.C.I.T.T. Laboratory relating to the local telephone circuits referred to in Part 2; these sensitivities were determined in accordance with the proposals given in Part 3.

The calculation procedure illustrated in the reference cited above can readily be used to write a rather simple computer programme for converting S_{UMJ} and S_{UJE} as functions of frequency into sending and receiving loudness ratings. Alternatively, a more flexible programme can be based on Equations (4.17a), (4.17b) and (4.17c) which enables the characteristics of the reference speech path to be chosen for each calculation. Programmes of both types are available and in use by the United Kingdom Post Office both for planning and design purposes.

APPENDIX 4–1

Certain characteristics concerning speech and hearing

To determine loudness ratings, it is necessary to specify certain items which are indicated in Table 1.1. The speaking position is treated in Appendix 1.1; the mouth reference point and the ear reference point used here are defined in Part 3; the present Appendix deals with the vocal level, the speech spectrum, the hearing threshold for continuous-spectrum sounds and the degree of coupling to be assumed between earphone and ear.

The reference vocal level used here is derived from C.C.I.T.T. Recommendation P.45 (Section b), "Acoustical speech power to be used during the tests". The description given there has been converted to apply at the mouth reference point used here, i.e. increased by 20.8 dB, and allowance made for the use of a speech voltmeter that gives the level of the rms voltage averaged over the duration of speech while the talker is active. The speech voltmeter allowance amounts to −5.5 dB which is the difference between the readings of a speech voltmeter of the above type and that used in the definition appearing in Recommendation P.45. Reference Vocal Level (R.V.L.) therefore corresponds to a sound pressure level of −4.7 dB relative to 1N/m² (rms while active).

A speech spectrum is shown in Col. (2) of Table 4.2 which applies for talkers speaking a short phrase ("Joe took father's shoe bench out") at R.V.L.; the average of males and females has been taken.

The hearing threshold for continuous-spectrum sounds can be taken as the hearing threshold for pure tones reduced by the quantity K which depends on the width of "critical frequency bands" at each part of the frequency range; Col. (6) shows the information used here. Col. (7) in the table shows the sensation level corresponding to zero mouth-to-ear loss, i.e. Col (2) minus Col. (6).

TABLE 4.2

CERTAIN CHARACTERISTICS CONCERNING SPEECH AND HEARING USED IN CALCULATING LOUDNESS RATINGS

Frequency Hz	Speech spectrum B'_S dB	Hearing threshold β_0 dB	Width of critical band, Hz $(\Delta f)_C$	$K = 10 \log_{10} (\Delta f)_C$	$\beta_0 - K$ dB	$Z_{RO} = B'_S - (\beta_0 - K)$ dB
(1)	(2)	(3)	(4)	(5)	(6)	(7)
100	+58.3	+37	87	19.4	+17.5	40.8
200	62.4	+22	52	17.2	+5	57.4
300	62.8	+17	50	17.0	0	62.8
400	61.6	+14.5	50	17.0	−3	64.6
500	60.2	+12	50	17.0	−5	65.2
600	57.1	+11	52	17.2	−6	63.1
800	51.6	+10	57	17.6	−8	59.6
1000	48.2	+9	63	18.0	−9	57.2
1250	45.8	+10	71	18.5	−8.5	54.3
1600	43.3	+11	83	19.2	−8	51.3
2000	41.4	+11	98	19.9	−9	50.4
2500	40.0	+9.5	120	20.8	−11.5	51.5
3000	38.6	+7.5	141	21.5	−14	52.6
3500	36.2	+9	170	22.3	−13.5	49.7
4000	33.9	+10	204	23.1	−13	46.9
5000	31.2	+12	275	24.4	−12.5	43.7
6000	30.8	+14	371	25.7	−11.5	42.3
8000	30.6	+18.5	588	27.7	−9	39.6

Notes:

1. — The speech spectrum is taken from the information published by Loye and Morgan (1939) from which the shape appropriate to reference vocal level has been determined.

2. — The hearing threshold for pure tones (Col. 3) has been taken from British Standard 2497: 1954 and applies for earphone listening.

3. — The 18 frequencies have been chosen arbitrarily to cover the frequency range from 70 to 9000 Hz. According to Fletcher this range is related to 89.7% of the total length of the basilar membrane; the remaining 10.3% is concerned with the very high frequencies up to 20 kHz which are audible only to young persons of "normal" hearing.

4. — The values of $\beta_0 - K$ in Col. (6) have been rounded to 0.5 dB.

PART 5. — Objective instrumentation

5.1 *General*

The theoretical principles explained in Part 4 have been embodied in certain types of objective instrumentation intended to determine loudness ratings without recourse to subjective tests. References [1], [2], [3], [4], [5], [6], [7] and [13] (given in Appendix 2 of the Preface) give examples. The theory underlying these equipments can be explained as follows:

From Equation 4.9

$$\Delta x = -\frac{1}{m} 10 \log_{10} \frac{\int_{f_1}^{f_i} 10^{-m(1/10)L_{UME}} G \, df}{\int_{f_1}^{f_2} 10^{-m(1/10)L_{RME}} G \, df} \quad \ldots \qquad (5.1)$$

From Equation 4.12

$$L_{UME} = 10 \log_{10} \frac{p_M^2}{p_E^2} \quad \ldots \qquad (5.2)$$

If we assume p_M, the sound pressure at the mouth reference point under free-field conditions, to be provided from a suitable artificial mouth in such a manner that p_M is independent of the speech path being measured, we can substitute Equation 5.2 in Equation 5.1 with the following result:

$$\Delta x = -\frac{1}{m} 10 \log_{10} \left[\frac{\int_{f_1}^{f_i} p_E^{2m} G \, df}{p_M^{2m} \int_{f_1}^{f_2} 10^{-m(1/10)L_{RME}} G \, df} \right] \quad \ldots \qquad (5.3)$$

Suppose also that the sound pressure p_M consists of a sinusoidal tone varying smoothly and cyclicly with frequency over the frequency band of interest (say 200 to 4000 Hz) without varying in amplitude. Assume that the reference speech path is fixed so that the integral in the denominator containing L_{RME} becomes a constant.

These assumptions also permit the variable in the numerator to be changed from frequency to time thus obtaining the expression:

$$\Delta x = -\frac{1}{m} 10 \log_{10} \left[\text{constant} \int_{0}^{T} p_E^{2m} F(t) \, dt \right] \quad \ldots \qquad (5.4)$$

where T is the period of cyclic variation in frequency of the tone from the artificial mouth and $F(t)$ is a function of time analogous to G (which is a function of frequency).

The instrumentation will be simplified if $F(t)$ is eliminated from the integral by making it constant. $F(t)$ will be constant if the variation in frequency of the tone is made such that df/dt is proportional to $1/G$. When this has been done, the bracketed term in Equation 5.4 becomes proportional to the mean value, over an integral number of periods of the tone, of p^{2m}; m can be taken as 0.25 and so we must average $p^{0.5}$.

Naturally, agreement between such objective instrumental loudness ratings and those obtained by subjective tests or calculation will only be secured if:

a) The artificial mouth and artificial ear included in the equipment are each sufficiently good representations of the corresponding human organs. The appropriate speaking position and listening conditions (see Figure 1.2b) must also be reproduced.

b) The instrumentation accurately represents a suitable structure such as that defined in Part 4 and the parameters of the various functions are correct.

c) The calibration of artificial mouth and artificial ear and the gain/frequency characteristics of certain elements are such that the sensitivities of the sending and receiving ends of the appropriate reference system are properly represented.

Figure 5.1 will be used to explain the principle and Figure 5.2 shows the arrangement of a practical realization of such equipment. The signal generator produces a tone which sweeps in frequency from 200 to 4000 Hz and back to 200 Hz at such an instantaneous rate that $|\mathrm{d}f/\mathrm{d}t|$ is proportional to the frequency f; i.e. $\log f$ is proportional to the elapsed time from the beginning of the cycle at 200 Hz or, when f is decreasing, to the time that must elapse until the end of the cycle at 200 Hz. The amplitude of the signal is independent of frequency and so the mean power per unit bandwidth, averaged over an integral number of cycles of the "warble tone" will fall at 3 dB per octave as f increases from 200 to 4000 Hz. As explained in Reference [12], such a signal when associated with a suitable form of voltmeter will represent, to a good approximation from 200 to 4000 Hz, a desirable form of frequency weighting, G. The cycling rate is, typically, once per second and the voltmeter must be sufficiently damped to integrate over several seconds.

The equipment is provided with an artificial mouth having, in principle, the properties described in Part 3. An artificial ear is also included and this ought to represent accurately the acoustic coupling conditions present when subjects apply telephone earphones to their ears. At present the available artificial ears do not perform this task very accurately and so an allowance, shown as L_E in Figure 5.1, must be made for the rather degraded coupling usually present between earphone and real ear as compared with the sealed conditions that apply when artificial ears of the existing types are employed (see also Parts 3 and 4).

When measuring sending ratings, the signal generator is connected to the artificial mouth and, when measuring receiving ratings, is used to provide the electrical input to the "unknown" local telephone circuit.

A special voltmeter having the following characteristics is provided:

The time-constant of the indicating instrument must be sufficiently long to average over several cycles of the signal generator.

A given deflexion will be produced when the following function of the instantaneous waveform at the input terminals remains the same:

$$y = \frac{1}{\tau} \int_0^\tau [|f(t)|]^{2m} \, \mathrm{d}t$$

where

$f(t)$ is the instantaneous amplitude (voltage) of the waveform, m is defined in Part 1,

τ is equal to the duration of any integral number of cycles of the signal generator.

The scale of the indicating instrument is such that changes in y are marked negatively in decibels relative to a datum input level to the voltmeter of, for example, 0.285V (−10.9 dBV) which is equivalent to −1 Npm in 600 ohms. When pure tones are applied to the voltmeter, the indication is independent of frequency over the range at least from 200 to 4000 Hz.

For measuring S.L.R., the "unknown" local telephone circuit is connected to the voltmeter, and for measuring R.L.R., the voltmeter is connected to the artificial ear and associated amplifier as shown in Figure 5.1.

Figure 5.1a shows an Objective Loudness Rating Measuring Instrument connected to the sending and receiving ends of a reference system. The figure also serves to illustrate definitions of the various sensitivities S_{AM}, S_{AE}, S_{RMJ}, S_{RJE}, etc. (which are expressed in dB). The acoustical interfaces (mouth reference point, *M.R.P.*, and ear reference point, E.R.P.) are as defined in Part 3. The earcap leakage transmission loss, L_E, is shown; this represents the impairment in acoustical coupling between earphone and ear when the earphone is used on real ears as compared with the perfect seal usually present when earphones are fitted to artificial ears for measurement purposes (see Part 3). S_{RJE} therefore represents the sensitivity of the Reference System Receive End when it is applied directly to the artificial ear; $S_{RJE} = S_{RJE} + L_E$. It is important to note that S_{RMJ} and S_{RJE} are assumed to apply, respectively, under real-voice and real-ear conditions.

The elements SW and $-SW$ represent, respectively, gain which is frequency-dependent and is introduced to change the shape of the sound spectrum applied to telephone microphones; this is only important when non-linearity is present, such as in carbon microphones. To preserve the overall frequency response from signal generator to voltmeter, complementary loss must be inserted in front of the voltmeter. When non-linearity is absent, it is immaterial to the theory and the principles whether SW and $-SW$ are both present or both absent. Appropriate values for SW, as a function of frequency, are shown below in Table 5.1; these will ensure that the shape of the long-term spectrum of the sound emitted from the artificial mouth will comply with that of speech as indicated by B_S in Table 4.2.

When measuring S.L.R., the Reference System Send-End is replaced by the "unknown" local telephone circuit (L.T.C.); the elements from point J ("junction") to the voltmeter can be replaced by the purely electrical arrangement shown in Figure 5.1b. Similarly, when measuring R.L.R. the Reference System Receive End is replaced by the "unknown" L.T.C. and the elements from the signal-generator to J can be replaced by the purely electrical elements shown in Figure 5.1c.

5.2 *Characteristics required to satisfy the proposals made in Part 1*

To ensure that the readings of an objective loudness rating measuring instrument will agree with those obtained subjectively and by calculation relative to the sending and receiving ends of the Intermediate Reference System defined in Part 1, it is necessary that C_S and C_R shall be consistent with the values of S_{BMJ} and S_{RJE} shown in Table 1.2.

It is convenient to arrange that the sensitivities of the artificial mouth and artificial ear (S_{AM} and S_{AE}) shall be practically independent of frequency and to introduce any-frequency-dependent elements necessary to comply with the above conditions by means of separate electrical networks (C_S and C_R). It is also convenient to retain −10.9 dBV as the datum electrical level shown in Figure 5.1, and as used in the O.B.D.M. and O.R.E.M.-A. To avoid excessive departure of the levels at various points in the instrument from those used in the O.B.D.M. and O.R.E.M.-A, the choice for S_{AM} and S_{AE} might be as follows:

$$S_{AM} = +5 \text{ dB relative to } 1\ N/\text{m}^2 \text{ per } V$$
$$\text{and } S_{AE} = -15 \text{ dB relative to } 1\ V \text{ per } N/\text{m}^2$$

The corresponding values of C_S and C_R, which satisfy the proposals in Part 1 (i.e. are consistent with the values of S_{RMJ} and S_{RJE} in Table 1.2), must then be as shown in Table 5.1. If the loss L_E is to be included in the same network as C_R, the values shown in Col. (4) should be used.

In Figure 5.2, C_R is shown transposed from the signal-generator end of the path to the voltmeter end. (Compare Figure 5.2b with Figure 5.1c.) This will have no effect upon the results provided that the "unknown" receiving end is linear. This transposition is for convenience by locating both C_S and C_R near the voltmeter

TABLE 5.1

GAIN/FREQUENCY CHARACTERISTICS REQUIRED FOR C_S, C_R AND SW TO COMPLY WITH PROPOSALS IN PART 1 ASSUMING CERTAIN VALUES, INDEPENDENT OF FREQUENCY, FOR S_{AM} AND S_{AE}

Frequency Hz	C_S dB	C_R dB	$C_R - L_E$ dB	SW dB
(1)	(2)	(3)	(4)	(5)
200	−11.0	−13.8	−22.2	+7.2
300	−5.8	−9.6	−10.9	+9.4
400	−4.4	−6.6	−5.9	+9.4
500	−3.8	−3.9	−1.7	+9.0
600	−3.5	−1.2	+1.3	+6.7
800	−3.3	+3.8	+7.0	+2.4
1000	−3.3	+5.1	+7.4	0
1250	−3.3	+5.7	+6.9	−1.4
1600	−3.3	+6.1	+6.2	−2.9
2000	−3.3	+6.6	+3.0	−3.8
2500	−3.3	+6.9	−0.5	−4.2
3000	−3.3	+7.2	+0.9	−4.8
3500	−15.8	−6.4	−13.9	−6.6
4000	<-35	<-23	<-32	−8.3

Notes:

1. — $S_{AM} = +5$ dB rel. 1 N/m^2 per V and $S_{AE} = -15$ dB rel. 1 V per N/m^2.

2. — The values of L_E used in Col. (4) have been taken from Section 3.5. They require to be revised in the light of results from current studies.

3. — The values of SW are derived from those of B'_S shown in Appendix 4-1.

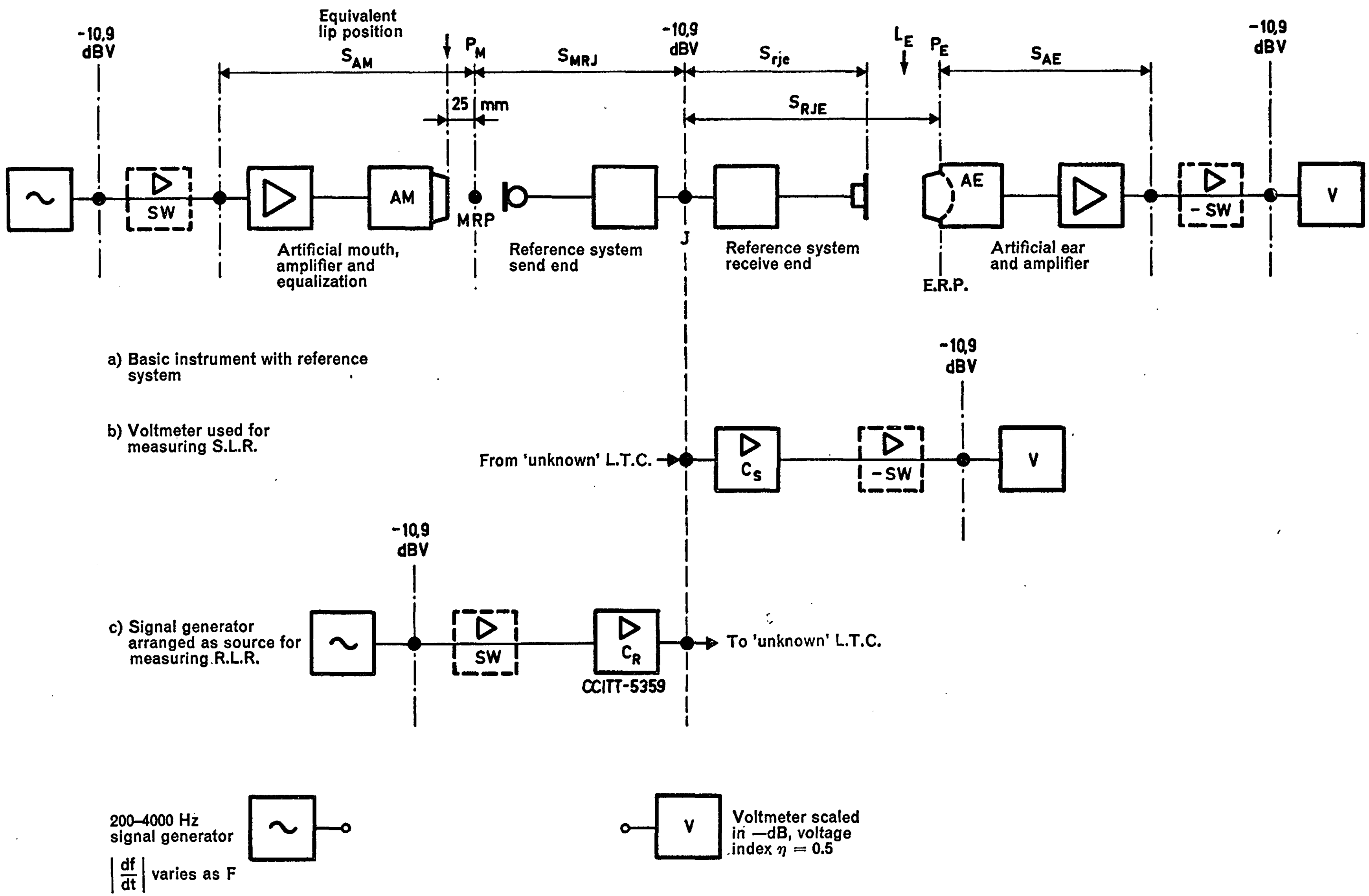
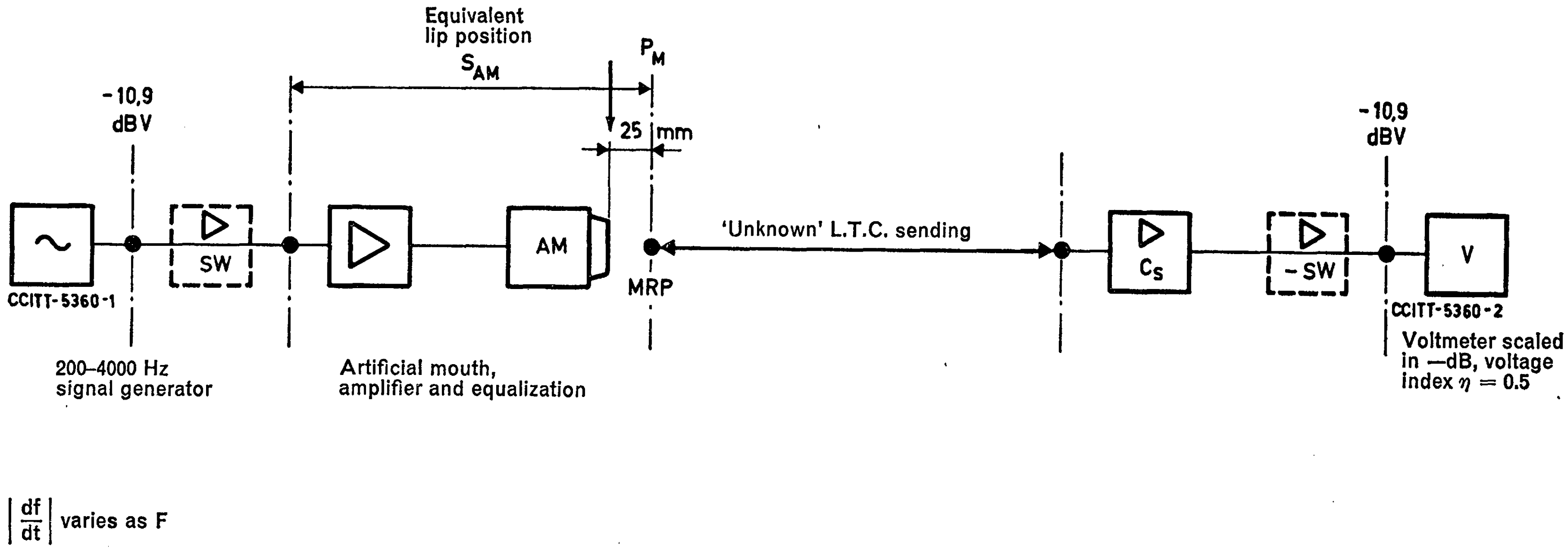
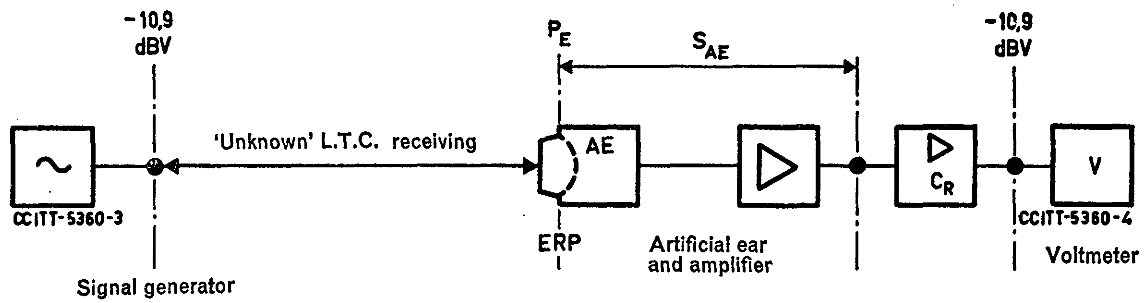


FIGURE 5.1. — Measurement of S.L.R. and R.L.R. using an objective measuring instrument



a) *Arrangement for measuring S.L.R.*



b) *Arrangement for measuring S.L.R.*

FIGURE 5.2. — **Objective loudness rating measuring instrument**

5.3 *Characteristics of present types of objective loudness rating measuring instruments*

The arrangements actually used in the O.B.D.M. and O.R.E.M.-A differ in several important respects from those described in Section 5.2. An analysis of their characteristics is given in Appendix 5–1 from which can be determined the theoretical amounts by which the readings can be expected to disagree with true measurements of reference equivalent, as defined in Recommendation P.72. Note that telephone microphones are tested at the reference equivalent talking position and not the new position proposed in Part 1. Some information on the extent of disagreement between readings by O.B.D.M. or O.R.E.M.-A and true values of reference equivalent is given in Appendix 5–2.

The analysis given in Appendix 5–1 assumes that the artificial mouth and the artificial ear are perfect representations of human mouths and ears at least in respect of the properties listed in Part 3. Artificial mouths may require some small corrections to be made to sensitivities measured with their aid, especially when "short" handsets are tested, but it is believed that these can be ignored for the present purposes. Artificial ears, however, have serious deficiencies when used to measure the sensitivities of telephone earphones, as explained in Part 3.

In Appendix 5–1, it is assumed that the earphone of any "unknown" receiving end is coupled to the artificial ear of the O.B.D.M. or O.R.E.M.-A with the same earcap coupling loss as would be present when used on human ears. When using an artificial ear, it is usual practice to seal the earcap directly to the coupler of the artificial ear, i.e. the earcap coupling loss L_E is thereby undesirably eliminated (see Figure 5.1a). In consequence, the readings of sensitivity would be some 3.5 dB too high (averaged over the frequency band) and so about 3.5 dB ought to be added to the loudness ratings obtained from the objective loudness rating measuring instrument. The O.B.D.M. includes provision of a 3.4 dB attenuator wich can be inserted when receiving loudness ratings are being measured.

The networks representing C_S, C_R, etc., indicated above could be readily inserted into existing types of objective loudness rating measuring instruments but some changes would be required in the methods of calibrating the various elements.

APPENDIX 5.1

Analysis of the O.B.D.M. and O.R.E.M.-A to determine the virtual sensitivities S_{RMJ} and S_{RJE}

The arrangement of the O.B.D.M. and the O.R.E.M.-A (as it is usually employed at present) differs from the proposed arrangement described in Section 5.2 and illustrated in Figure 5.1. The arrangement of O.B.D.M. and O.R.E.M.-A can be described in terms of Figure 5.2; when sending loudness ratings are measured, the network SW is present but the complementary $-SW$ network is absent. When receiving loudness ratings are measured neither SW nor $-SW$ is present. Unlike the proposed arrangement, the sending and receiving ends of O.B.D.M. and O.R.E.M.-A are not physically complementary to each other and so, for the purposes of analysis, certain of the elements included in Figure 5.2, namely those represented by C_S, C_R and $-SW$ do not exist as separate entities and must be considered as virtual. C_S and $-SW$ are exactly complementary and so cancel each other in reality; C_R is independent of frequency and so cannot be distinguished in reality from the sensitivity, S_{AE}, of the artificial ear, i.e. only $(S_{AE} + C_R)$ can be measured directly.

The various steps leading to the determination of values for S_{RMJ} and S_{RJE} are illustrated by Tables 5.2 and 5.3.

In the analysis of O.B.D.M. and O.R.E.M.-A, it is necessary to distinguish between the sensitivities, S_{CMJ} and S_{CJE}, of devices used to standardize the sensitivities of the artificial mouth and artificial ear and the sensitivities, S_{RMJ} and S_{RJE}, of the equivalent reference system. S_{CMJ} and S_{CJE} are real and relate specifically to the means used for calibration and are chosen so that this can be carried out at spot frequencies as well as with the signal generator sweeping the frequency range 200–4000 Hz. S_{RMJ} and S_{RJE}, however, are the sensitivities of sending and receiving ends of a reference system (which may be only virtual) which would reproduce (whatever the "unknown") the same transmission loss, at each frequency, from signal generator to voltmeter as is the case in the actual O.B.D.M. or O.R.E.M.-A. equipment. In fact, S_{RMJ} and S_{CMJ} are equivalent only "on average" and not frequency-by-frequency; corresponding equivalence applies for S_{RJE} and S_{CJE}.

TABLE 5.2

CALIBRATION SENSITIVITIES OF ONE EXAMPLE OF O.B.D.M. (C.C.I.T.T. LABORATORY)

Frequency Hz	"S.F.E.R.T." network *SW* dB (gain)	Total sens. of sending calib. device S_{CMJ} dB	S_{AM} dB	(2) + (3) + (4) dB	Actual values of C_S (= *SW*) dB	Values of C_R corresp. to S_{CMJ} (S_{AM} + S_{CMJ}) dB	S_{RMJ} dB	Virtual values of C_R (S_{AM} + S_{RMJ}) dB	S_{CJE} = − (S_{AE} + C_R) dB	S_{RJE} dB
(1)	(2)	(3)	(4)	(5)	(6)	(7)	(8)	(9)	(10)	(11)
200	− 0.3	(−10.3)	+10.6	0	− 0.3	+ 0.3	(−5.2)	+5.4	(+11.5)	(+16.1)
300	− 0.5	−10.1	+10.6	0	− 0.5	+ 0.5	−5.2	+5.4	+11.5	+15.9
400	− 0.6	−10.0	+10.6	0	− 0.6	+ 0.6	−5.2	+5.4	+11.5	+15.8
500	− 0.8	− 9.8	+10.6	0	− 0.8	+ 0.8	−5.2	+5.4	+11.5	+15.6
600	− 1.2	− 9.4	+10.6	0	− 1.2	+ 1.2	−5.2	+5.4	+11.5	+15.2
800	− 2.5	− 8.1	+10.6	0	− 2.5	+ 2.5	−5.2	+5.4	+11.5	+13.9
1000	− 4.1	− 6.5	+10.6	0	− 4.1	+ 4.1	−5.2	+5.4	+11.5	+12.3
1250	− 4.8	− 5.8	+10.6	0	− 4.8	+ 4.8	−5.2	+5.4	+11.5	+11.6
1600	− 6.6	− 4.0	+10.6	0	− 6.6	+ 6.6	−5.2	+5.4	+11.5	+ 9.8
2000	− 8.7	− 1.9	+10.6	0	− 8.7	+ 8.7	−5.2	+5.4	+11.5	+ 7.7
2500	−10.6	0.0	+10.6	0	−10.6	+10.6	−5.2	+5.4	+11.5	+ 5.8
3000	−11.3	+ 0.7	+10.6	0	−11.3	+11.3	−5.2	+5.4	+11.5	+ 5.1
3500	−11.4	+ 0.8	+10.6	0	−11.4	+11.4	−5.2	+5.4	+11.5	+ 5.0
4000	−10.9	(+ 0.3)	+10.6	0	−10.9	+10.9	(−5.2)	+5.4	(+11.5)	(+ 5.5)
		SRE = +5.7					*SRE* = +5.7		*RRE* = −2.4	*RRE* = −2.4

Notes:

1. — The characteristics of the "S.F.E.R.T." network, *SW*, were supplied by the C.C.I.T.T. Laboratory and apply to the example of O.B.D.M. located there.

2. — Col. (4) is the result of a special measurement made by the C.C.I.T.T. Laboratory. Col. (3) was obtained by subtraction assuming Col. (5) to be 0 dB at each frequency, see Note 3.

3. — Col. (5) is shown as 0 dB at each frequency although there are slight departures due to mismatches in shape between the "S.F.E.R.T." network and the Calibration Device. The equipment is set up so that this quantity "averages" 0 dB when the warble-tone signal generator is used and the level is measured with the special voltmeter.

4. — For explanation of Col. (6) see text.

5. — If S_{CMJ} applied for the sending end of the equivalent reference system, C_R would have to be equal to $S_{AM} + S_{CMJ}$ [see Col. (7)]; in fact this is not so and the true virtual values are shown in Col. (9).

6. — S_{RMJ} [see Col. (8)] is the sensitivity of a sending end having the following properties:

a) Restricted in frequency to the range 200–4000 Hz;

b) Independent of frequency within this band;

c) The value of S.R.E. is the same as that for S_{CMJ}, namely +5.7 dB.

7. — S_{CJE} is the sensitivity of a hypothetical Receiving Calibrating Device which would, if inserted into the arrangement shown in Figure 5.2b, produce a reading of 0 dB at each frequency. Note that $S_{AE} + C_R = -11.5$ dB relative to 1 *V* per N/m^2.

8. — S_{RJE} is the sensitivity of a receiving end that has the same shape of response as that of C_S [= *SW*, see Col. (6)] and whose *R.R.E.* is the same as that corresponding to S_{CJE}, namely −2.4 dB.

9. — The first and last entries in the columns showing S_{CMJ}, S_{RMJ}, S_{CJE} and S_{RJE} are bracketed as a reminder that they apply only within the range 200–4000 Hz outside which complete cut-off takes place.

The sending-end calibration of the O.B.D.M. and O.R.E.M.-A is based on the use of a measuring microphone inserted in a solid cylindrical baffle of certain dimensions which represent the arrangement of the now-obsolete S.F.E.R.T.[1] This device causes an increase in sound pressure, which is considerably frequency-dependent, so that for the present analysis the sensitivity of the calibration device in terms of the unobstructed sound pressure at the mouth reference point requires special measurement not included in the normal calibration; one set of results obtained by the C.C.I.T.T. Laboratory on their O.B.D.M. and another by the Post Office for an example of the O.R.E.M.-A are given in Tables 5.2 and 5.3. Each equipment is normally standardized with a network (termed the

[1] When the O.B.D.M. was originated (in 1940), S.F.E.R.T. was the only international fundamental loudness-rating reference system; it was superseded by N.O.S.F.E.R. in 1960.

TABLE 5.3

CALIBRATION SENSITIVITIES OF ONE EXAMPLE OF O.R.E.M.-A (UK-PO/R13)

Frequency Hz	"S.F.E.R.T." network SW dB (gain)	Total sens. of sending cal. device S_{CMJ} dB	S_{AM} dB	(2) + (3) + (4) dB	Actual values of C_S (= SW) dB	Values of C_R needed for S_{RMJ} = S_{CMJ} dB	S_{RMJ} dB	Virtual values of C_R (S_{AM} + S_{RMJ})	S_{CJE} dB	S_{RJE} dB
(1)	(2)	(3)	(4)	(5)	(6)	(7)	(8)	(9)	(10)	(11)
200	0.0	(−9.1)	+ 8.1	−1.0	0.0	− 1.0	(−4.4)	+4.6	(+11.5)	(+16.0)
300	− 0.1	−8.5	+ 8.4	−0.2	− 0.1	− 0.1	−4.4	+4.6	+11.5	+15.9
400	− 0.3	−8.2	+ 8.6	+0.1	− 0.3	+ 0.4	−4.4	+4.6	+11.5	+15.7
500	− 0.5	−8.5	+ 8.5	−0.5	− 0.5	0.0	−4.4	+4.6	+11.5	+15.5
600	− 1.0	−8.1	+ 8.7	−0.4	− 1.0	+ 0.6	−4.4	+4.6	+11.5	+15.0
800	− 1.9	−7.6	+ 9.5	0.0	− 1.9	+ 1.9	−4.4	+4.6	+11.5	+14.1
1000	− 3.1	−7.0	+ 9.4	−0.7	− 3.1	+ 2.4	−4.4	+4.6	+11.5	+12.9
1250	− 4.3	−6.4	+ 9.4	−1.3	− 4.3	+ 3.0	−4.4	+4.6	+11.5	+11.7
1600	− 6.3	−4.4	+ 8.7	−2.0	− 6.3	+ 4.3	−4.4	+4.6	+11.5	+ 9.7
2000	− 8.2	−1.9	+11.3	+1.2	− 8.2	+ 9.4	−4.4	+4.6	+11.5	+ 7.8
2500	−10.0	+1.4	+ 9.1	+0.5	−10.0	+10.5	−4.4	+4.6	+11.5	+ 6.0
3000	−11.4	+4.1	+ 9.1	+1.8	−11.4	+13.2	−4.4	+4.6	+11.5	+ 4.6
3500	−11.6	+0.8	+ 9.3	−1.5	−11.6	+10.1	−4.4	+4.6	+11.5	+ 4.4
4000	−11.4	(−4.3)	+ 9.1	−6.6	−11.4	+ 4.8	(−4.4)	+4.6	(11.5)	(+ 4.6
		SRE = +4.9	Average = +9.0				*SRE* = +4.9		*RRE* = −2.4	*RRE* = −2.4

Notes:

1. — The characteristics of the "S.F.E.R.T." network, *SW*, are taken from the Bruel and Kjaer Handbook associated with O.R.E.M.-A.

2. — Cols. (3) and (4) are the results of special measurements.

3. — In principle Col. (5) ought to show 0 dB at each frequency; the departures are due to mismatches in shape between the "S.F.E.R.T." network and Calibration Device characteristics. The equipment is set up so that this quantity "averages" 0 dB when the warble-tone signal generator is used and the levels are measured with the special voltmeter.

4. — See Notes 4, 5, 6, 7, 8 and 9 appended to Table 5.2, except *SRE* = +4.9 dB instead of +5.7 dB.

"S.F.E.R.T." network and indicated by *SW* in Figure 5.2) introduced at the electrical input to the artificial mouth; the gain/frequency characteristic is intended to cancel, in shape, the attenuation/frequency distortion caused by the obstruction effect of the calibrating device. The insertion gain of *SW* (actually, of course, a loss) is shown in Col. (2) of Tables 5.2 and 5.3 together with the measured sensitivity, S_{AM}, of the artificial mouth and of the calibrating device, S_{CMJ}, when this is referred to the sound pressure p_M at the mouth reference point (instead of being expressed as a pressure sensitivity of −11.5 dB relative to 1 V per Nm^{-2}, as stated in the published descriptions). More detailed measurements were made in the case of the O.R.E.M.-A which enable the actual departures of S_{AM} from constancy to be determined (see Col. (4) in Table 5.3). The "average" sensitivity has been taken as +9.0 dB relative to 1 Nm^{-2} per V.

Calibration of the receive end of O.B.D.M. and O.R.E.M.-A is achieved by standardizing the sensitivity of the measuring microphone which forms part of the artificial ear and setting the associated amplifier so that the total acousto-electric sensitivity up to the voltmeter, i.e. $S_{AE} + C_R$, is −11.5 dB relative to 1 V per Nm^{-2}. Col. (5) in each table shows the total sensitivity from the signal generator to the point J in Figure 5.2a. (See also Note 3 in each table.)

Col. (7) in each table shows what C_R ought to have been if it were desired to render S_{RMJ} equal to S_{CMJ}; C_R is, in fact, independent of frequency. The shape (as a function of frequency) of S_{RMJ} is the same as that of C_R and so, to satisfy this condition and to ensure that S_{RMJ} "averages" the same as S_{CMJ}, these two sensitivities have been matched in sending reference equivalents relative to N.O.S.F.E.R., by calculation as described elsewhere. Col. (8) in each table shows the necessary values which, like S_{CMJ} in Col. (3), corresponds to *SRE* = +5.7 dB for O.B.D.M. and +4.9 dB for the O.R.E.M.-A.

Col. (9) in each table shows the virtual values of C_R using, for the O.R.E.M.-A, a constant value of +9.0 dB for S_{AM}.

From Figure 5.2a, it can be seen that $C_S = SW$ (see Col. (6) in Tables 5.2 and 5.3). S_{CJE} is the sensitivity of a (virtual) receiving calibrating device which, when connected in place of the "unknown" in Figure 5.2b, will give a reading of 0 dB on the voltmeter of each frequency. Clearly, $S_{CJE} = -(S_{AE} + C_R) = +11.5$ dB relative to 1 V per Nm^{-2}. Such a receiving end would have $RRE = -2.4$ dB (relative to N.O.S.F.E.R.). The required shape of frequency response for S_{RJE} must be as shown for C_S (see Col. (6) of Tables A 5/1 and A/2) but it must have the same "average" sensitivity as S_{CJE}. The required values have been calculated and are shown in Col. (11).

Cols. (8) and (11) therefore show the required sets of values of S_{RMJ} and S_{RJE} which can be used in a computation programme based on Figure 1.3 and should yield the same values of *S.L.R.* and *R.L.R.* as the O.B.D.M. or O.R.E.M.-A instrumentation, subject, of course, to instrumentation accuracy. (Note especially the defects of present artificial ears and the need to introduce the allowance shown in Figure 5.1 as L_E.)

APPENDIX 5.2

Differences between readings by O.B.D.M. or O.R.E.M.-A and true values of reference equivalent

Table 5.4 summarises theoretical and empirical determinations of the amounts of disagreement between readings by O.B.D.M. or O.R.E.M.-A and true values of reference equivalent (sending and receiving) determined in accordance with C.C.I.T.T. Recommendation P.72. The theoretical values are from Appendix 5.1 and the measurements are from reported information as indicated in the notes to the table.

It will be seen from the table that different corrections have to be used for different types of telephone sets. In extreme cases, the differences between corrections could be at least 6 dB.

TABLE 5.4

AMOUNT BY WHICH THE READINGS OF THE O.B.D.M. AND O.R.E.M.-A ARE LESS THAN THE TRUE VALUES OF SENDING AND RECEIVING REFERENCE EQUIVALENT

Instrument	Source of information	Sending dB	Receiving dB
O.B.D.M.	Theory Appendix 5-1	+5.7	−2.4
O.B.D.M. $L_E = 3.4$ dB	Measurements on three types of L.T.C. [a]	+2.9 to +6.2	−1.7 to −2.4
O.R.E.M.-A	Theory Appendix 5-1	+4.9	−2.4
O.B.D.M. $L_E = 3.4$ dB	Measurements by C.C.I.T.T. Lab. (RT 371) [b]	+0.8 to +11.6	−3.8 to +4.3
O.R.E.M.-A $L_E = 0$ dB	Measurements by Spain [c]	+3.1 to +5.8	+0.8
O.R.E.M.-A $L_E = 0$ dB	Measurements reported by Cragg and Barnes [d]	+4.9 to +6.5	+1.8

[a] These results appear in C.C.I.T.T. Laboratory Technical Report No. 415.
[b] These results appear in C.C.I.T.T. Laboratory Technical Report No. 371; they are the extreme values for a large number of telephone sets.
[c] These results appear in COM XII - Nos. 5 and 35, 1968-1972 period, and are the mean values for ten different telephone sets each associated with three different subscribers' lines. The smaller values for sending are associated with the longest subscribers' lines and the larger values with the zero-line condition.
[d] These results are the mean values for 42 different types of telephone set, each measured with zero and 1200 ohms subscribers' lines. The smaller values for sending are associated with the 1200 ohms lines.

Question 16/XII — Impedance variations in subscriber lines and telephone sets

(Continuation of Question 19/XII, studied in 1968-1972)

(Documentary question)

a) What is the range of variation of the impedances of subscriber lines and telephone sets measured at the terminal of the local system and in the primary centre ?

What is the corresponding range of variation of the return loss measure in relation to a purely resistive impedance of 600 ohms or any other fixed value impedance used as a balancing circuit in the terminating sets ?

b) What method can be used in the design of telephone sets of subscribers' lines and supply bridges to reduce this range of variations ?

Note. — In addition to the return loss/frequency characteristic of local telephone circuits (i.e. subscriber's instrument, local line and feeding bridge) versus the appropriate balance network, it would be useful if the return loss from the points of view of echo and of stability could also be presented for each of a number of significant configurations (e.g.: maximum, median, and zero-length line).

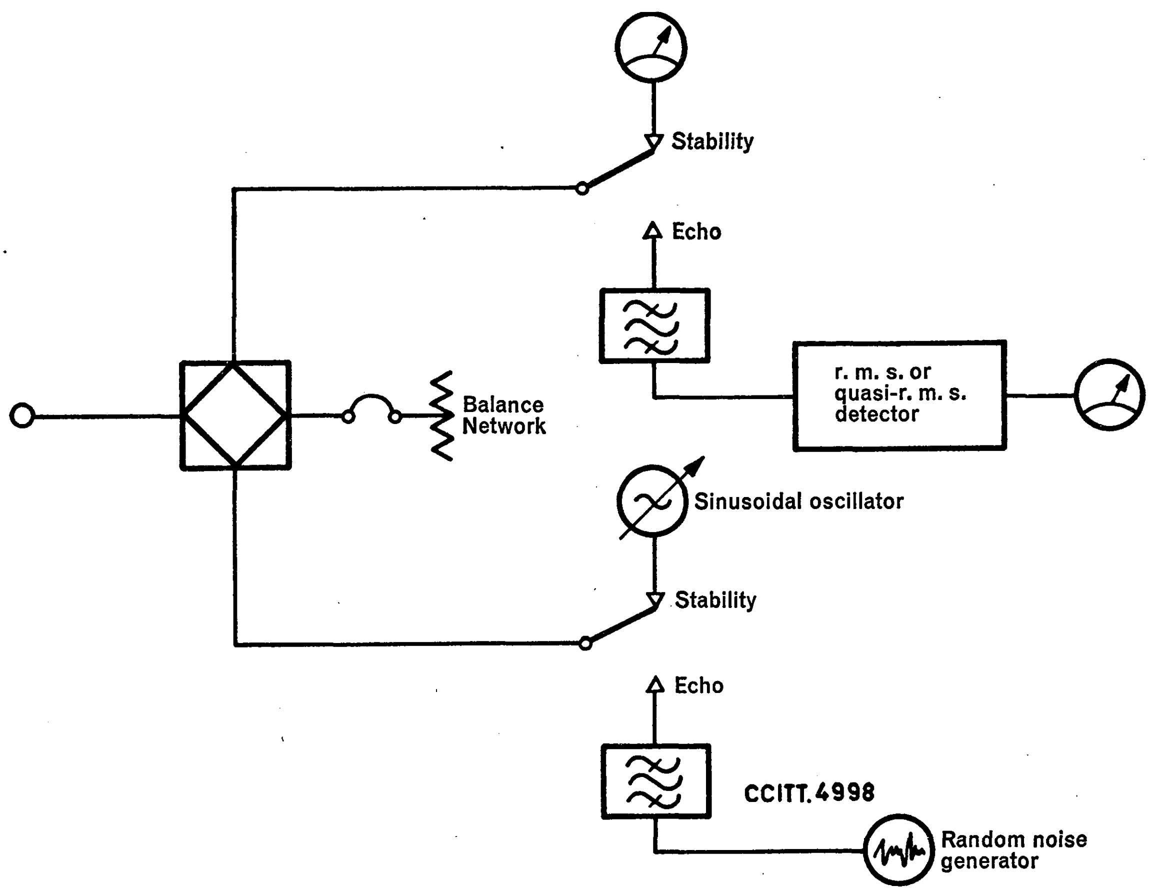


FIGURE 1. — **A two-wire return loss measuring set**

Notes to Figure 1. — It is suggested that a terminating set representative of good modern national practice be used in the test set.

Naturally, arrangements for setting up and holding calls must also be included, as well as means for calibrating the device.

It may also be convenient if the instrument were also capable of measuring the return loss of one impedance against another. Standard balance networks (e.g. 600 ohm, 900 ohm resistors, each in series with appropriate capacitors, if required) may be included, but provision should be made for the connection of an external balance network.

Examples of the loss/frequency characteristic likely to be encountered in practice will be found in the Handbook: *Transmission Planning of Switched Telephone Networks.*

A definition of echo return loss may be based upon the provisional definition of the transmission loss of the path a-t-b from the point of view of echo which is to be found in Recommendation G.122-B (b).

The definition of stability return loss is given in Recommendation G.122-A (a) and extends over the whole range 0–4 kHz. In practice the somewhat more restricted band 200–3500 Hz is adequate. The fact that the minimum value of the return loss at the local exchange may not coincide with minimum value of transmission losses in the four-wire portion of the national system provides some planning margin, and in any case cannot be relied upon. Figure 1 illustrates the principle of an instrument to measure these quantities directly at two-wire points.

Note 2. — Opportunity should also be taken to measure the various return losses presented by PBX exchange lines when PBX extensions are in the speaking condition.

Note 3. — The information collected together during 1968–1972 has been incorporated in the C.C.I.T.T. Handbook: *Transmission Planning of Switched Telephone Networks.*

Question 17/XII — Loudspeaker telephones

(Continuation of Question 17/XII, studied in 1968–1972)

What conditions (from the point of view of telephone transmission) should be satisfied by subscribers' telephone stations which may be used for international calls and which include loudspeakers or broadcasting type microphones with amplifiers ?

Note 1. — Annex 1 sets out the principles adopted for studying the conditions which telephone sets with loudspeakers must satisfy from the point of view of transmission performance. Annex 2 is a report of studies carried out in 1968–1972.

Note 2. — Further information can be found in annexes to earlier revsions of Question 17/XII as follows:

Federal Republic of Germany	:	*Red Book*, Volume V, pp. 705–706
Swedish Administration	:	*Red Book*, Volume V, pp. 706–708
Polish Administration	:	*Red Book*, Volume V*bis*, pp. 284–286
United Kingdom Post Office	:	*White Book*, Volume V, Question 17/XII pp. 2–3
International Telegraph & Telephone Co	:	*White Book*, Volume V, Question 17/XII pp. 4–5

Note 3. — The study of this question should also take note of loudspeaking telephones in connection with videophone systems, being studied under Question 4/XV and 5/XV.

ANNEX 1

(to Question 17/XII)

Principles adopted for studying the conditions which telephone sets with loudspeakers must satisfy from the point of view of transmission performance

1. *Transmission quality*

The transmission quality (at the sending and receiving ends), from the aspects of efficiency and of the response curve as a function of the frequency, should if possible be as good as that of the modern subscriber's handset.

Note. — In general, during measurements and use by the subscriber, allowance should be made for the room-effect when the distance between the mouth and the loudspeaker exceeds about 50 cm.

2. *Input impedance*

When a loudspeaker station is connected instead of the telephone apparatus, the input impedance of the station should correspond to the impedance of a normal telephone apparatus.

3. *Stability*

Self-excitation by electro-acoustic reaction should be avoided. To avoid its occurrence in unfavourable room or line conditions, an adjustment in the balance network or the reception sensitivity should be made.

4. *Restriction of self-excitation*

When self-excitation occurs care should be taken to see that the singing voltage at the terminals of the local line does not exceed, for example, 800 mV, or the voltage which will be defined subsequently under paragraph 6 below.

5. *Power adjustment*

If there is a manual device for adjusting the power, this should regulate the power of the speech received, and in such a manner that the received power cannot be reduced to zero. If an Administration considers it essential to use another adjusting device, this should be coupled to the device mentioned above in such a way that any adjustment carried out by the subscriber possessing a loudspeaker set does not disturb his correspondent.

6. *Vocal power sent to line*

As a precaution against the possibility of overloading and crosstalk when the subscriber speaks very close to the microphone, the power sent to line should be automatically limited to a maximum value, the study of which should be undertaken.

7. *Automatic control*

When using devices controlled by vocal voltages, care must be taken to see that these devices are so constructed that they do not cause difficulties or disturb the telephone communication either between a loudspeaker subscriber's station and an ordinary subscriber's station, or between two loudspeaker subscribers' stations, regardless of the type of circuit used between them. The control should not cause a total blocking in either direction. It is preferable to obtain only a simultaneous displacement of the gain of both directions.

8. *Talking condition indicator*

A distinctive mark should be provided for the talking condition so that the subscriber may have a sign (for example a visual indicator) helping him to remember to replace the loudspeaker station in the ringing condition after the conversation.

9. *Possible return to the use of an ordinary telephone set*

If, for any reason, the subscriber wishes to suspend use of the loudspeaker set, he must have an ordinary telephone set, or, if necessary, a handset associated with the loudspeaker telephone installation. If this suspension occurs during a call, it must not cause the call to be cut off or interrupted.

10. *Crosstalk*

Conversations over adjacent telephone lines should not be heard. For this reason, in the case of a loudspeaker telephone line, these conversations should not be intelligible at the smallest possible distance between the ear and the loudspeaker for the maximum adjustment of the receiving sensitivity.

Questionnaire for the further study of Question 17/XII

It is desirable that the information be given in a uniform manner, as set out below, together with an indication of the methods of measurement used to obtain it.

1. *Introduction*

2. *General aspects*

3. *Conditions of use*

3.1 Do Administrations provide for the use of a normal handset telephone in addition to a loudspeaking set and, if so, do they prefer a separate set or an incorporated subset?

3.2 Is there a requirement for a handset telephone with additional loudspeaking facilities on the receive portion to enable several listeners to follow the conversation?

4. *Facilities*

4.1 Power supply: if the instrument is provided with power from a separate source, what action is taken if the power supply fails?

4.2 Volume control: should manual volume control of the receive section be provided?

4.3 Stability: is instability permissible in extreme cases, e.g. when dialling?

It has been noted that several Administrations have specified values of singing voltage in excess of 800 mV, the value specified in paragraph 4 of Annex 1 to Question 17/XII (*Red Book*, Volume V). Do Administrations agree that a voltage above this value is tolerable?

5. *Transmission characteristics*

5.1 *Test conditions*

a) electrical conditions;
b) acoustical conditions;
c) room conditions, including room noise values (information on the effects of differing room characteristics on the transmission characteristics would be helpful—see Annex 3 to this Question in *White Book*, Volume V).

5.2 *Sending and receiving characteristics*

a) Sending-sensitivity (subjective and objective assessments), frequency response, distortion and vocal power sent to line in service conditions.
b) Receiving-sensitivity (subjective and objective assessments), frequency response and distortion.
c) Comparison with the standard handsets installed by the same Administration.

Note 1. — Although Study Group XII agreed upon a physical test arrangement (see Figure 1) for objective determination of reference equivalents and similar parameters of loudspeaker telephones, no complete method for the (subjective or objective) determination of these parameters has been internationally recommended. Therefore, Administrations are asked to give a detailed description of the method they use to determine the (send and receive) indices based on loudness.

Note 2. — It is of interest to determine whether articulation is affected by the voice-controlled switching device, if any. However, the customary articulation tests (such as those described in Volume V of the *Red Book* for A.E.N. measurements) are not suitable in every case and methods which simulate the conditions of a two-sided conversation have to be used.

5.3 *Gain and control characteristics*

a) Gain switching characteristics.
b) Means of avoiding singing.
c) Switching times and thresholds.
d) Upper and lower limits for manual gain control.

Note. — Administrations are also requested to state whether they take special precautions to prevent any inconvenience to the occupants of neighbouring premises from a loudspeaker telephone placed in a room.

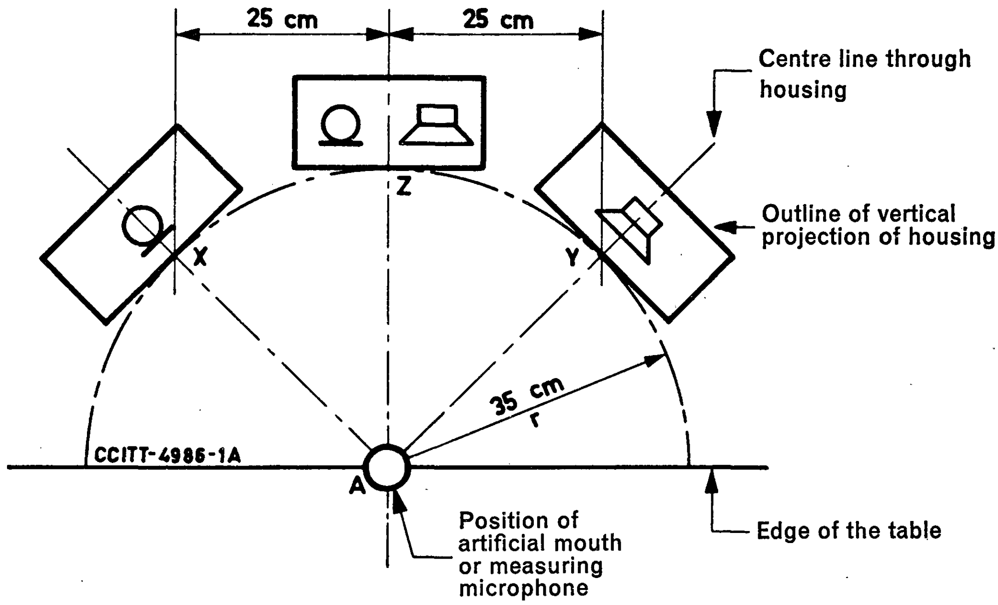


FIGURE 1

Notes to Figure 1:

1. — Table should have a hard surface (e.g. polished hardwood) and be not less than 1 square metre in area.
2. — All dimensions shown in the figure are parallel to the table surface.
3. — Single housing loudspeaking telephones should occupy position Z.
4. — The centre of the edge of the front of the box should be tangential to a circle of 35 cm radius.
5. — The equivalent lip position of the artificial mouth at A is in alignment with the table edge and 35 cm above it. It should be positioned such that its axis passes through the centre of the box housing the microphone.
6. — The acoustic screen of the measurement microphone at A should be in alignment with the table edge and 35 cm above it. It should be positioned such that its axis passes through the centre of the box housing the loudspeaker.

5.4 *Signal-to-noise characteristics*

5.5 *Impedance* in terms of a locus on an R-X plane with frequency as parameter. If the mode of operation is such that the impedance for sending is different from that for receiving, both values should be quoted.

Note. — The impedance presented by the instrument has a definite effect on the stability and echo of the long-distance network. This should not be overlooked when drawing up national specifications for such instruments. Due account should be taken of the range of permissible line lengths, feeding arrangements, etc.

6. *Balance to earth, overvoltage and surge protection* (including information on lightning protection)

ANNEX 2

(to Question 17/XII)

Reply by Study Group XII at the end of the 1968–1972 period

In 1972 Study Group XII agreed on a physical test arrangement for the objective determination of reference equivalents and similar parameters (see Figure 1). The aim was to specify a realistic arrangement to keep the sum of the sending and receiving distances close to 1 m, and to have a convenient way of positioning the sets under test.

Results of further measurements are required—both objective and subjective—which have been carried out in rooms having different acoustic characteristics including anechoic chambers (with and without test table) and ordinary rooms. The acoustic characteristics of these rooms should be described by giving details such as dimensions, reverberation time (as a function of frequency), noise, position of test set-up. With this information it may be possible to correlate the results of subjective and objective measurements. Administrations are invited to contribute on what an "ordinary room" should be, considering the data given in COM XII-Nos. 8 and 83 (period 1968–1972). Do they think that the acoustic room might be sufficient to characterize rooms for the purpose of

constant R, defined as $$R = \frac{S\bar{a}}{1-\bar{a}}$$

in which S = total surface area of the room
$\bar{a}$ = average absorption coefficient

loudspeaking telephones? Attention is also drawn to the effect of different types of noises (e.g. Hoth vs typewriter) and subscriber behaviour (e.g. actual speaking levels and preferred receiving levels).

It may be helpful to give some additional references on this topic:

GARDNER, M. B.: A study of talking distance and related parameters in hands-free telephony. *B.S.T.J.*, November 1970, page 1529.

GARDNER, M. B.: Effect of noise, system gain, and assigned task on talking levels in loudspeaker communication. *J. Acous. Soc. Am.*, Vol. 40, No. 5, November 1966, page 955.

GARDNER, M. B.: Effect of noise on listening levels in conference telephony. *J. Acous. Soc. Am.*, Vol. 36, No. 12, December 1964, page 2354.

BARNES, G. J.: Voice switching parameters in telephony. *I.T.T. Electrical Communication*, Vol. 47 (1972)—No. 3

Question 18/XII — Transmission performance of pulse-code modulation systems

(Continuation of Question 21/XII, studied in 1968–1972)

a) What recommendation might be made by the C.C.I.T.T. concerning the standard of transmission performance assessment that ought to be achieved for a single audio-link provided by a practical engineered PCM system bearing in mind the conditions under which such a link may form part of an international connection ?

b) So that acceptable values of fundamental parameters for an economical design may be recommended, the effects of the various factors contributing to the quantization distortion of an ideal PCM system, e.g. peak clipping and centre clipping, should be assessed in the same units as used for a).

c) Since any economically engineered PCM system will be liable to other forms of degradation due, for example, to transmission errors, quantization inaccuracies, synchronization difficulties, jitter, etc., how should such systems be measured in ordinary working conditions, to ensure that they comply with the requirements set forth under a) above ?

Note. — Annex 1 covers continuing studies of the question. Annex 2 reproduces a contribution communicated by Study Group XV; further information may be found in Annex 4 to Question 2/D and its appendices (Volume III of the *White Book*). Annexes 3 and 4 should serve as a basis for the method of discussion and for recourse to subjective tests; subjective tests results should be presented in such a way as to enable easy comparison to results contained in these annexes. Attention is also drawn to Annex 5 which discusses factors to be considered in answering the question, and comments on inter-relationships of the important factors. Annex 6 is included for study.

ANNEX 1

(to Question 18/XII)

Report on the study of Question 21/XII approved by Study Group XII in November 1972

Study Group XII agreed on a further subdivision of the question and made some progress as follows:

Part a) — Standard of transmission performance assessment

The matter of establishing a standard performance assessment can be divided into four basic elements:

1) *The reference scale*

A reference scale, readily defined in physical terms, is needed for measuring the transmission performance of PCM systems. Two reference scales that have been used are continuous random noise and random noise proportional to the instantaneous signal amplitude.

It was decided that the use of random noise proportional to the instantaneous signal amplitude be proposed to those Administrations conducting further evaluation of quality so that there will be a common basis for the comparison of results.

2) *The comparison method*

The comparisons necessary involve the following three classes of telephone connection:

a) speech links containing the reference device;

b) speech links containing PCM systems;

c) speech links containing no PCM systems but with injected circuit noise. These are included to relate the results to those available on the effects of loss and circuit noise (Question 4/XII).

Comparisons between pairs of connections from different classes can be made by pair-comparisons, listening-only tests or indirectly by opinion (rating scale) tests. Other listening-only methods are possible, including articulation tests, etc.

It is necessary to supplement listening-only tests by conversation tests because there is evidence that relative performance may not always be correctly assessed by listening-only tests.

3) *The framework*

Transmission quality can only be assessed within a framework of complete telephone connections and these must be representative of actual connections. Hypothetical reference connections of various compositions are therefore needed for studying different particular problems. No general statement can be made rigidly specifying such hypothetical reference connections but some guidance will be obtained by noting the hypothetical reference connections which have been used for related studies (see, for example, Recommendation G.103). However, in the study of reference connections, it should be noted that actual connections may include national circuits with 7-bit speech encoding.

Reference connections for studying quantizing distortion must allow the speech levels at the inputs to the PCM systems to range fully over those levels that could be found in practice. When listening-only tests are being conducted, this presents no difficulty but, in a conversational test, the variation of speech level at the input to a PCM system is governed only by the range of individual talking levels of the subjects and the location of the PCM system in the connection.

4) *The standard value*

The value of the reference quantity considered acceptable under particular circumstances must be established through subjective tests. Opinion tests, paired comparisons with more familiar impairments, or other methods can be used.

The United Kingdom Post Office has established an acceptable value of 22 dB for the signal-to-noise ratio of the special reference signal recommended under a),1) above. It reports that distortion of this level is undetectable to 50% of subjects over a range of listening levels from −5 to −25 dBm (mean power when active) at input to a zero reference equivalent receiving end.

Administrations are asked to furnish additional information of this kind so that this important link between PCM impairments and other transmission objectives can be established.

Part b) — Working measurement methods

Two types of measurements that might be applied to working PCM systems to ensure that they live up to the intent of the standards outlined in a) above have been defined.

1) *Load capacity*

A method which has been proposed is to determine when an increase in sine wave input produces a less than proportional increase in output. For example, the United Kingdom Post Office considers that a +2 dBm0 overload requirement has been met "when an increase in input level from +2 dBm0 to +4 dBm0 results in an increase in output level of not less than 1 dB nor more than 2 dB".

2) *Quantizing distortion*

Two methods have been established for measuring this effect:

Sine wave: The PCM system is excited with a signal with a frequency near to 1000 Hz but not a subharmonic of the sampling rate. A meter rejects the existing signal and measures the distortion content of the rest of the voice band.

Gaussian wave: The PCM system is excited with a band of noise from 450–550 Hz. The distortion products in the band above 850 Hz are measured and correction provided for the band below 850 Hz.

No information has been provided to Study Group XII on the relationship between these types of measurement and performance standards. Administrations are asked to provide appropriate data so that the study of this part can advance along lines consistent with parts a) and b) above.

APPENDIX

(to Annex 1)

Practical measurement of quantizing distortion and transmission loss of a PCM channel

(Contribution of the United Kingdom Post Office)

a) *General*

The method which has been used for calculating the transmission performance of a speech channel in a PCM system is described in [1]. The important parameters are signal to quantizing distortion ratio (R) and loss (L) of the channel.

The practical measurement of R is described in outline in [2]. In the following, a more complete description of the method for measuring both R and L and definition of the essential components of the measuring apparatus are given.

b) *Test equipment*

The arrangement of the test apparatus is indicated in Figure 1. The test signal is band-limited Gaussian noise. It should have a Gaussian distribution of amplitude over a range of at least four standard deviations at the output of the filter F1. The level of the signal is monitored with a voltmeter reading r.m.s. with a time constant of 5 s to ensure that the fluctuation of the reading is not more than ±0.1 dB. A calibrated attenuator A and amplifier G1 enable the level of test signal to be varied at the input to the channel over a range −60 to −3 dBm0.

The output of the channel is filtered by F2 to remove any out-of-channel frequency components. For the various stages of the tests, alternative paths are provided through band filter F3, through high-pass filter F4 or with no filter. The output level is measured with a voltmeter with similar characteristics to those of the meter at the input to the channel.

The filter characteristics are given in Tables 1 to 3.

c) *Measurement of R*

An appropriate level of test signal s (see Table 4) is applied to the system under test by testing the reading of V_1 and adjusting attenuator A accordingly. With switches K_1 and K_2 in position 2, measurement of the received level of signal plus distortion products is made (reading 1). The switches K are then set to position 3 to connect high-pass filter F4 into circuit and the output level is measured. This reading is then corrected for the mean loss of the filter over the range 850 to 3400 Hz to give the level of distortion products in that band (reading 2). From readings 1 and 2, the signal/distortion ration R_M may be deduced. For most of the input levels involved, there is little error involved by making the simplification of taking the difference between reading 1 and reading 2 (the maximum error over the range to be considered would be only 0.3 dB).

To obtain the values of R, the values of signal to distortion ratio R_M, obtained as above, should be decreased by 0.8 dB (i.e.: $10 \log_{10} (3100/2550)$) to compensate for the reduced bandwidth in which the distortion products are measured.

TABLE 1

CHARACTERISTICS OF 450–550 HZ BAND-PASS FILTERS (F_1 AND F_3)

Frequency (Hz)	⩽240	340	420	450–550	580	700	780	⩾ 820
Insertion loss (dB)	>60	>45	>15	4 ± 1	>10	>40	>50	>55

TABLE 2

CHARACTERISTICS OF 3400 HZ LOW-PASS FILTER (F_2)

Frequency (Hz)	⩽3000	3400	3750	5000	⩾ 5800
Insertion loss (dB)	<1	<5	>50	>55	>60

TABLE 3

CHARACTERISTICS OF 850 HZ HIGH-PASS FILTER (F_4)

Frequency (Hz)	⩽760	800	850	⩾ 1000
Insertion loss (dB)	>60	>35	<5	<1.5

TABLE 4

VALUES OF R_M FOR 8-DIGIT 13-SEGMENT A-LAW SYSTEMS, $T_{MAX} = +3$ dBm0

Test signal level $s\Delta$ (dBm0)	−4	−8	−28	−35	−40	−45	−50	−55
Signal-to-quantizing distortion ratio in 850–3400 Hz band, R_M (dB)	>31.0	>35.0	>34.6	>32.5	>28.4	>23.4	>18.4	>13.4

Minimum allowable values of R_M corresponding to a set of values of s which is proposed for test purposes are shown in Table. 4.

d) *Measurement of L*

An appropriate level of test signal *s* is applied to the system under test, as for the measurement of *R*. With the switches K_1, K_2 in position 1, to restrict the bandwidth of the output signal to that of the sent signal, the received level is measured. Generally, for the measurement of the performance of the codec, it is the variation of loss with differing input level which is of main interest, therefore the loss of the filters F2, F3 need not be accurately known. If absolute values of loss are required to be measured, the allowance to be made for the filters can be determined by direct connection between the sending and receiving parts of the measuring apparatus.

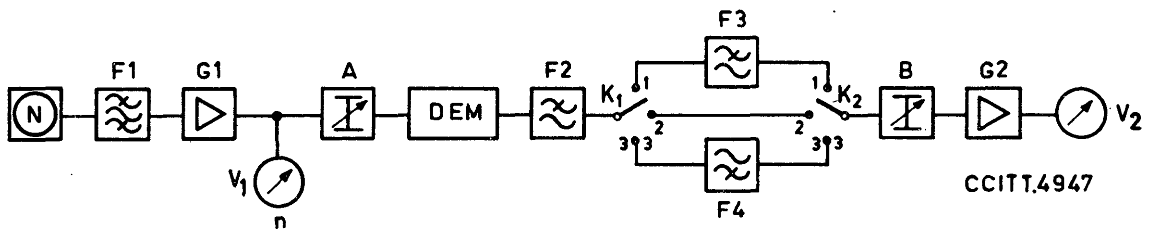


FIGURE 1

REFERENCES

[1] RICHARDS, D. L.: Transmission performance of telephone networks containing P.C.M. links, *PROC I.E.E.* 115, No. 9, Sept. 1968.

Note. — This text has been reproduced (with the exception of the table of calculated values) as Annex 1 to Recommendation G.712 (*Green Book*, Volume III).

[2] RICHARDS, D. L.: Telecommunication by speech (Butterworths, London, 1973).

Note. — See also *White Book*, Volume III, Question 2/D, Annex 6, Appendix 2 and Contribution COM XII-No. 84, period 1964–1968.

ANNEX 2

(to Question 18/XII)

Status of noise reference unit instrumentation

(Contribution of the American Telephone and Telegraph Co.)

This Annex gives a description of a noise reference unit which produces signal-dependent noise for use in subjective tests of compandored PCM systems and outlines preliminary plans for its use.

1. *Description of the unit*

The Modulated-Noise Reference Unit[1] described was developed by the United Kingdom Post Office and is a device that produces distortion similar in character to that of a PCM system using a logarithmic companding law; over an extremely wide range of signal amplitudes, the signal-to-distortion power ratio is constant. The instrument will handle input signals with peak levels of up to +6 dB relative to 1 volt r.m.s., and the residual noise level at the output is −55 dBm. The unit has input and output impedances of 600 ohms, is mains-operated and housed in a cabinet 50 × 30 × 25 cm approximately, weighing approximately 13.6 kg. A block schematic diagram of the unit is shown in Figure 1.

The input signal is divided into two paths, one of which passes to the output without distortion; this is termed the signal path. The other passes through a modulator which reverses the polarity of the signal according to the polarity of the output of a wide-frequency-band noise source; the noise acts as the carrier input to a ring modulator.

[1] Reference: LAW, H. B. and SEYMOUR, R. A. A reference distortion system using modulated noise, *PROC. I.E.E.*, *109B*, 1962, pp. 484–485.

The output of the modulator then consists of a waveform having an envelope of amplitude equal or proportional to that of the speech signal but devoid of intelligibility. This "modulated noise" is then added to the undistorted signal in a proportion that can be controlled by an attenuator (A2) in the modulated noise path. The combination is then filtered to remove modulated-noise products beyond the frequency range of the signal. The device is calibrated so that, when the control attenuator A2 is set at 0 dB, the mean power output due to modulated noise is equal to that due to the signal alone. The pre-set attenuator A1 and the switches S1 and S2 are provided to facilitate setting up. The effect of this process on speech is subjectively very similar to that produced by quantizing distortion in a PCM system.

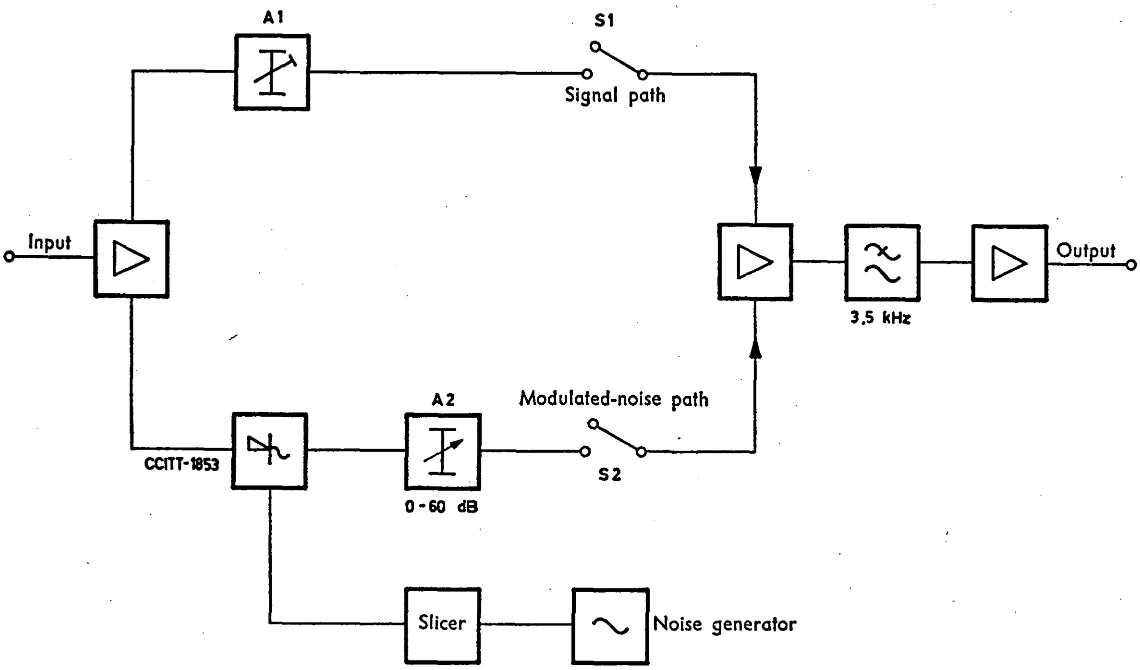


FIGURE 1. — **Modulated-noise reference unit**

2. *Plans for its use*

Initial sinusoidal tests indicate that a signal-to-noise ratio constant to within 0.5 dB can be maintained over at least a 40-dB range.

For setting up and for live testing when desirable, connections will be made through S1 and S2 to the output hybrid. It is expected that tape recordings will be used for the subjective tests; in which case, connections from S1 and S2 will be made to a dual-track tape recorder. The outputs of the tape recorder may then be combined with various conditions obtained by changing the relative gain in the noise and signal paths. When needed, an additional noise source will be added in the noise path to represent noise in the absence of signal.

Before beginning subjective tests, characteristics of the noise reference unit to be measured are the following:

1) Dynamic range.

2) Amplitude distribution and frequency spectrum of output with flat and voiceband weighted noise inputs.

3) Frequency response.

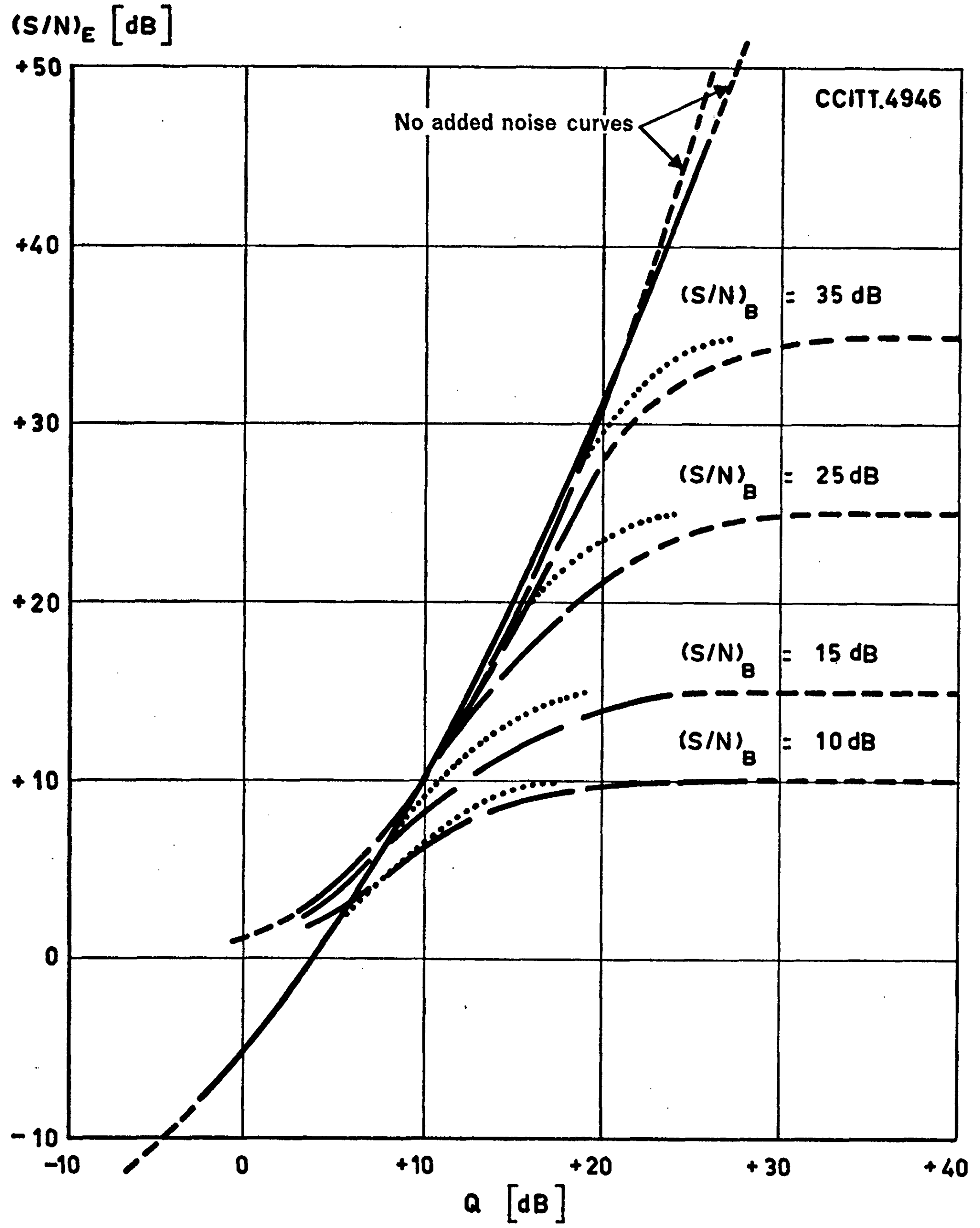
ANNEX 3

(to Question 18/XII)

Subjective evaluation of speech correlated noise

(Contribution of the American Telephone and Telegraph Company)

This Annex summarizes results of laboratory experiments to determine the subjective effects of speech correlated noise. A Modulated Noise Reference Unit (M.N.R.U., see Annex 2) was used in experiments to deter-



——— A. T. & T., $(S/N)_E = (0.027\,Q^2 + 1.26\,Q - 5.08)$ dB

—— —— U.K. Post Office $(S/N)_E = (0.0602\,Q^2 + 0.285\,Q + 1.07)$ dB

— — — — Curve extension beyond data points

. . . . A. T. & T. estimate based on limited data

Q = Speech-to-speech-correlated-noise ratio
$(S/N)_B$ = Speech-to-idle-circuit-noise ratio for background noise
$(S/N)_E$ = Equivalent speech-to-idle-circuit-noise ratio

FIGURE 1. — **Idle circuit noise equivalent of combined speech correlated and idle circuit noise**

mine 1) the threshold of detectability for speech correlated noise, and 2) the subjective idle circuit noise equivalent of speech correlated noise combined with background idle circuit noise. Both the threshold tests and the equivalence tests utilized background idle circuit noise levels covering a range of 41 dBt to 61 dBt and a range of speech (S) levels from 67 dBt to 91 dBt.[1]

Analysis showed that the relations given below closely approximated the average observer threshold responses. The standard deviation was approximately 4 dB,

$$Q = (0.252\ S/N + 10.62)\ \text{dB for } S/N \leqslant 39.4\ \text{dB},$$

$$Q = 20.3\ \text{dB for } S/N > 39.4\ \text{dB},$$

where Q = speech-to-speech-correlated-noise ratio,

S/N = speech-to-background-idle-circuit-noise ratio.

These results are very similar to those of another study which showed that the threshold value for Q was between 20 and 21 dB over a range of speech levels from about −45 dBm to about −10 dBm at the input to a 0 dB reference equivalent receiving system [1]. The latter study does not mention background idle circuit noise; comparison of results from that study with the functions given above suggest that the idle circuit noise was probably quite low.

Curves obtained from analysis of the equivalence test results are given in Figure 1 together with curves obtained in another study [2] reported in Annex 4 to this question. The two sets of curves are in close agreement for Q values greater than about 5 dB.

References

[1] Richards, D. L.: *Impairment of Telephone Speech by Quantizing*, Fourth International Symposium on Human Factors in Telephony, September 1968, VDE Verlag, Berlin.

[2] Richards, D. L.: *Telecommunications by Speech* (Chapter 4.2.2.1) (Butterworths, London, 1973).

ANNEX 4

(to Question 18/XII)

Transmission performance of pulse code modulation systems

(Contribution of the United Kingdom Post Office)

1. *General*

Information on several aspects of the transmission performance of PCM systems is contained in the following references:

[1] Richards, D. L.: *Speech Transmission Performance of PCM Systems. Electronics Letters*, 1965, *1*, No. 2, p. 40.

[2] Richards, D. L.: *Distortion of Speech by Quantizing. Electronics Letters*, June 1967, *3*, No. 6, pp. 230 ff.

[3] Richards, D. L.: *Transmission Performance of Telephone Networks containing P.C.M. Links, Proc. I.E.E.* 1968 *115*, pp. 1245–1258.

[4] Richards, D. L. and Berry, R. W.: *Impairment of Telephone Speech by Quantizing*, Fourth International Symposium on Human Factors in Telephony, September 1968, VDE Verlag, Berlin, pp. 29–46.

The aspects of these references (in particular Ref. 4) that have special relevance to Question 18/XII are summarised very briefly below. The following comments on the specific parts of the question are also brief, but may be expanded in detail if required from the content of the references.

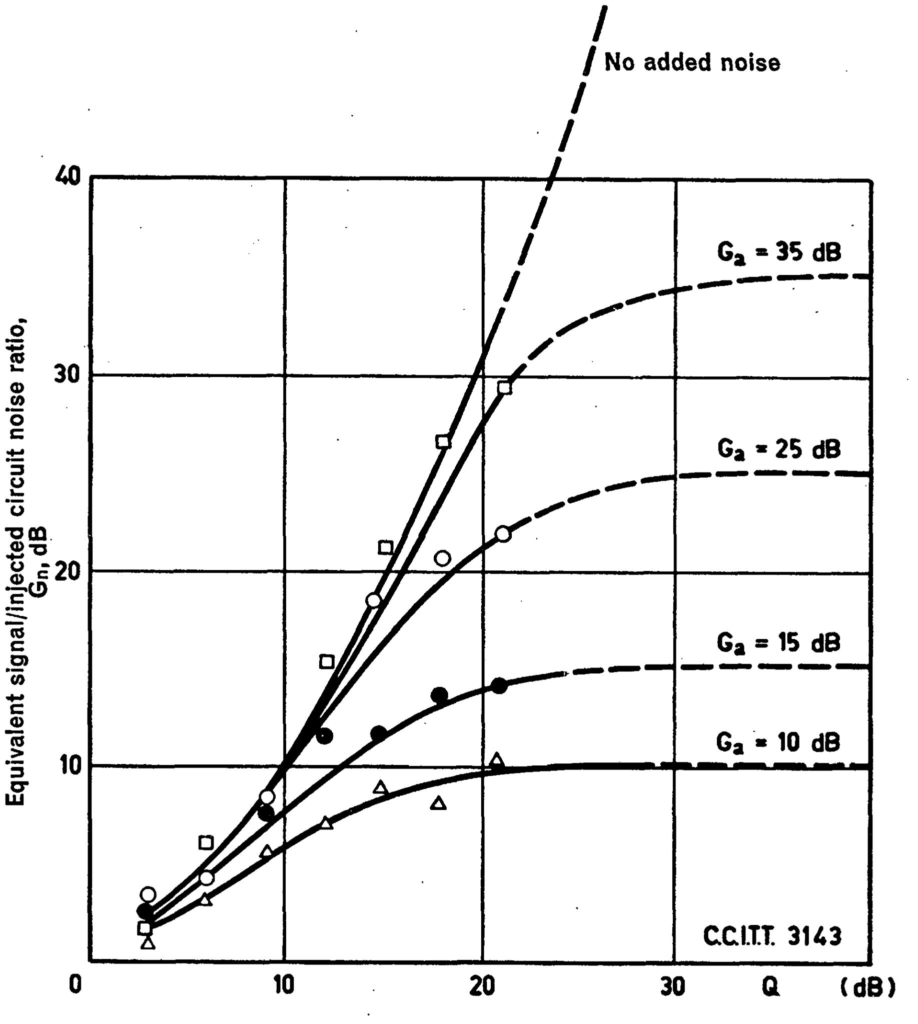
2. *Summary of references*

Mathematical analysis, using linear regression techniques, and utilising the amplitude probability density function of the (statistically stationary) input signal and the non-linear transfer function of an encoder/decoder system, yields R (dB) and L (dB) of the system as functions of the input level s (dBm0), where

[1] dBt = r.m.s. pressure, as measured in a 6 cm³ coupler, relative to 20 μPa (0.00002 Newtons/square metre). For noise, 50 dBt corresponds to −58 dBmp at the input to a 0 dB reference equivalent receiving system. Corresponding values for speech are 80 dBt and −29 VU.

R = signal-to-distortion ratio
L = system loss to the input signal

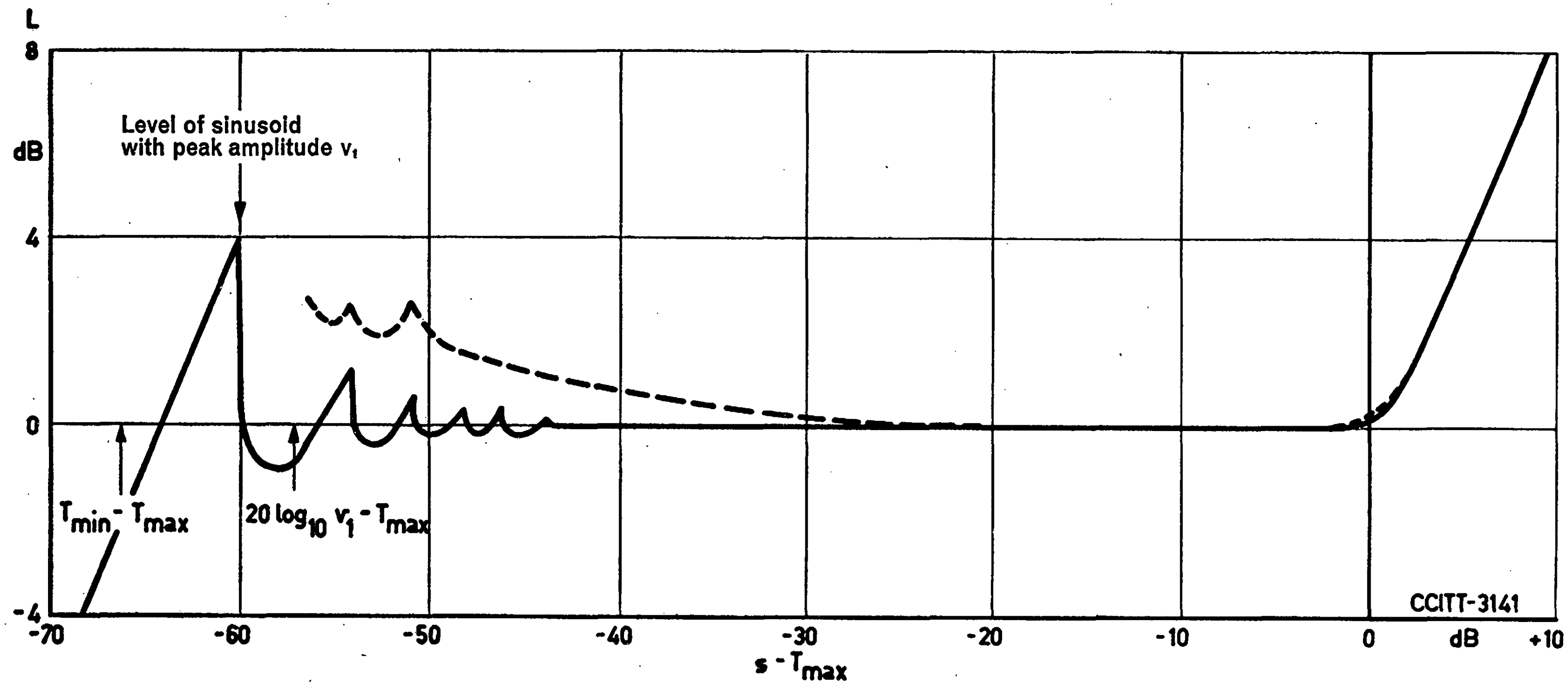
The analysis applies for any given input signal distribution, but if a Gaussian distribution is used, representing syllabic segments of speech, the derived R—characteristic may be transformed to a Q characteristic—a function of the total input speech level (s dBm0)—representing a subjectively averaged long-term speech-to-distortion ratio. The quantity Q is used to define the transmission performance of a PCM system in terms of the level of input speech.



Subjectively measured points: G_a = 10 dB △ 15 dB ● 25 dB ○ 35 dB □

Note. — G_a is the expression in decibels of the signal to the circuit noise added at the output of the M.N.R.U.
Equivalent signal/injected circuit noise ratio, G_n, dB

FIGURE 1. — **Variation in the equivalence between Q and signal/I.C.N. ratio, (G_n), in the presence of added noise**



SYSTEM A

——— measured

– – – – theoretical

Notes:

1. Input signal was 1 kHz sine wave.
2. Measurement at output was made with a transmission measuring set (non-selective).
3. The theoretical curve is for the fundamental component of the output. The curve for the total r.m.s. value of the output is practically identical between $s - T_{max}$ values of -58 and $+10$ dB.

FIGURE 2a. — **Measured and theoretical L curves for sinusoid**

If R is constant over the range of syllabic segment powers present in speech, then $Q = R$. An instrument — the Modulated Noise Reference Unit (M.N.R.U.) — has been designed to produce a constant R characteristic, and is calibrated in terms of Q. Subjective tests with the M.N.R.U. have produced a threshold value for Q, corresponding to detectability of distortion, of approximately 20 dB over a range of speech levels from -15 to -45 dBm at the input to a 0 dB R.R.E. receive end. By means of pair comparison tests with speech degraded by the addition of circuit noise, an equivalence between Q and signal-to-injected circuit noise ratio has been established (see Figure 1). Because the M.N.R.U. has a low basic noise level, it is necessary to add injected circuit noise at the output of the M.N.R.U. to simulate the real performance of a PCM system during "silent" periods of speech. The manner in which the Q to signal/I.C.N. ratio equivalence varies in the presence of this added noise has also been determined.

For a given Hypothetical Reference Connection of known Overall Reference Equivalent, the mean speech level at the input to the first PCM system may be determined. From the L characteristics of each PCM system in the connection, and the losses between them, an overall R, and hence the Q, characteristic of the non-linear elements in the connection may be derived. The total noise power of the connection may also be determined, allowing the connection to be represented by an M.N.R.U. of given Q setting, and a given noise level added to the output. From the established Q to signal/I.C.N. ratio equivalence, an equivalent connection containing loss and injected circuit noise may be determined, allowing the conversational mean opinion score or proportion of users who experience difficulty to be determined from known information.

This method of analysis has been applied to a range of H.R.C.s containing PCM systems using the 7-digit 13-segment A-law. The results have been expressed in terms of percentages of unsatisfactory opinions — a scale directly related to the conversation mean opinion scale. Work is proceeding with the M.N.R.U. (and later with a laboratory PCM system) to validate the results by means of conversation tests.

3. *Comment on Question 18/XII*

3.1 *Standard of transmission performance assessment*

1) *The reference scale*

The proposal to use a reference scale of random noise proportional to the instantaneous signal amplitude is accepted by the Post Office.

2) *The comparison method*

Work intendend to validate the performance of systems as estimated in accordance with reference [4] suggests that subjects are somewhat more critical of circuit performance under listening conditions than they are under conversation conditions. It is considered that, while the recommended comparison method is essential to obtain uniform presentation of results, and to assemble data on the required large number of conditions, it is also desirable to check some of the results by conversation tests.

3) *The framework*

The transmission performance of reference connections has been determined as described in reference [4].

4) *The standard value*

Determinations of values of the reference quantity have been made as follows:

a) threshold of detectability of distortion over a range of listening levels — threshold test using M.N.R.U.;

b) evaluation of Q for 6 H.R.C.s — analysis;

c) equivalence between Q and signal-to-injected circuit noise ratio; Pair comparison tests using M.N.R.U.;

d) evaluation of 6 H.R.C.s in terms of percentages of unsatisfactory opinions — analysis. (These evaluations may also be expressed in terms of percentage difficulty: see Question 2/XII.) Details of the results are given in Reference [4].

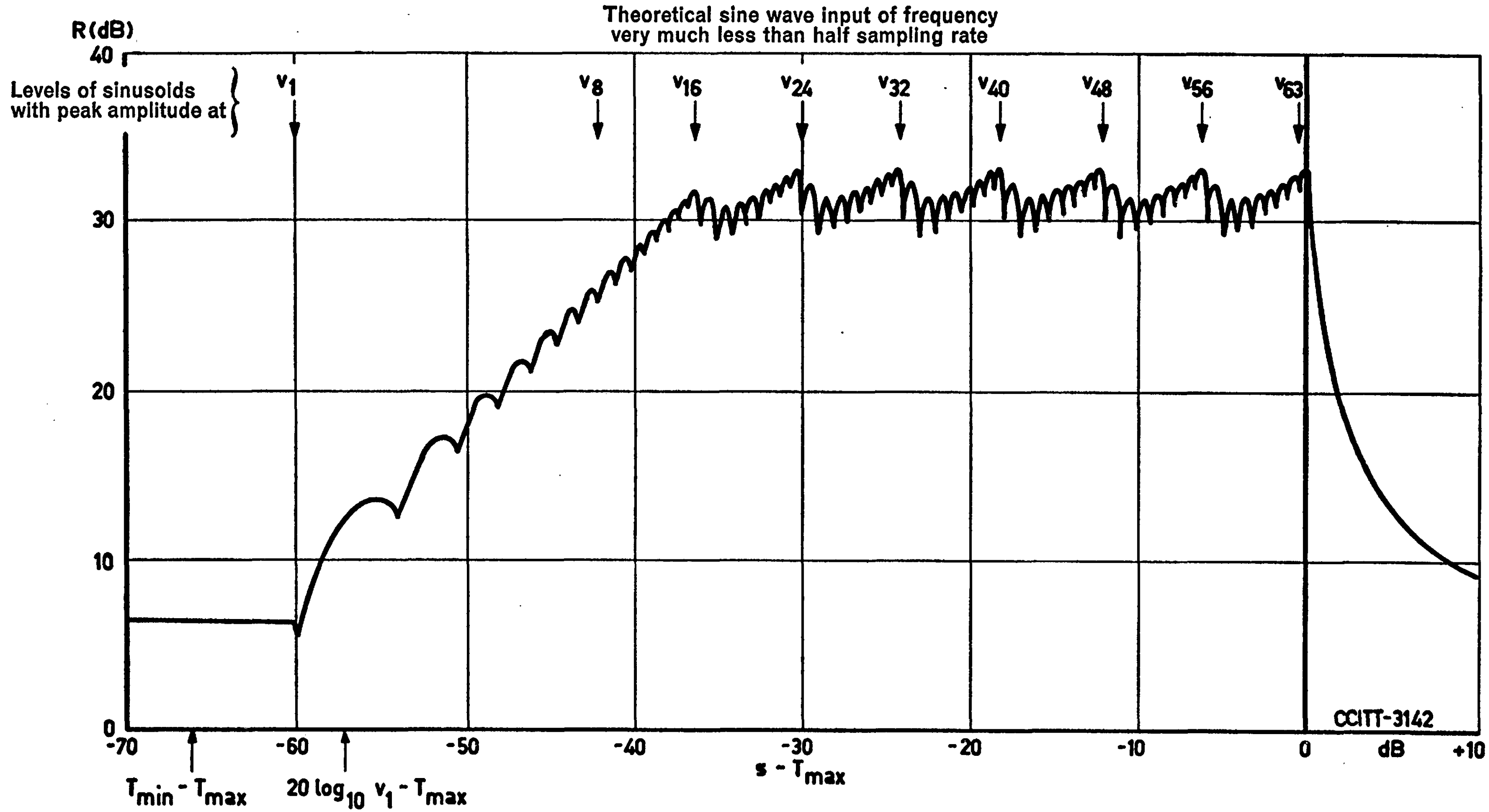


FIGURE 2b. — Theoretical R curve for sinusoid

Gaussian test signal 450–550 Hz

System A

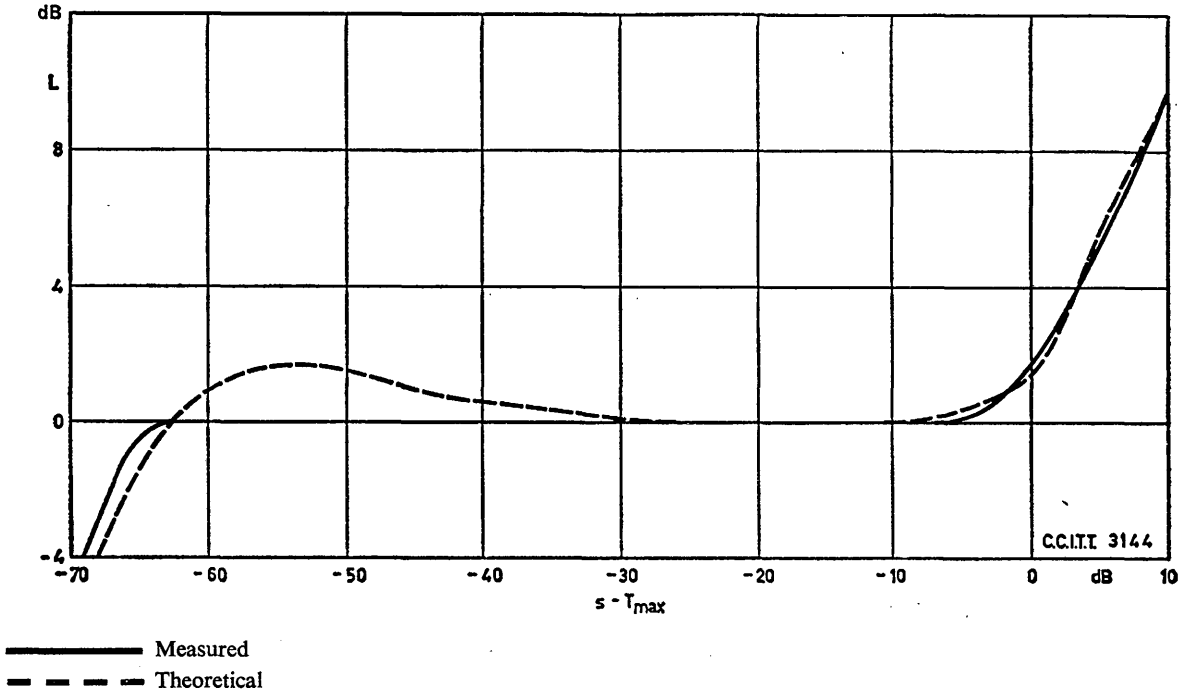


Figure 3a. — **Measured and theoretical L curves-Gaussian signal**

Gaussian test signal 450–550 Hz

System A

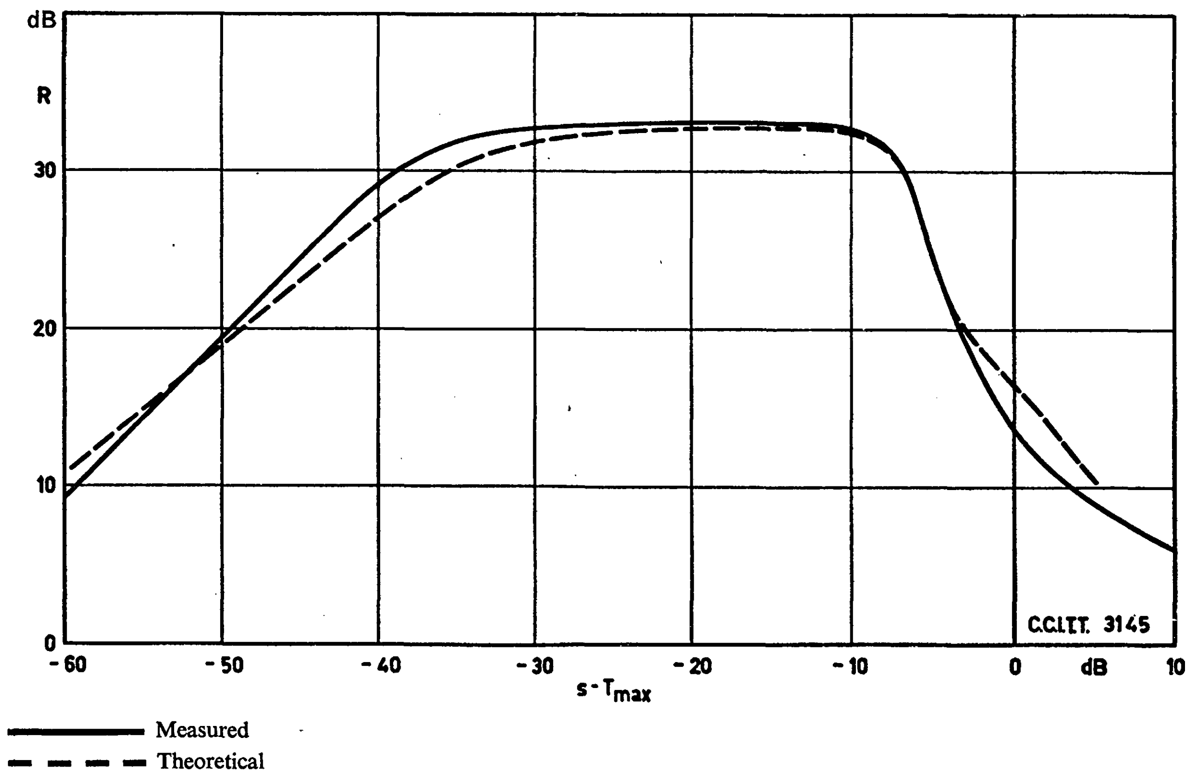


Figure 3b. — **Measured and theoretical R curves-Gaussian signal**

3.2 *Working measurement methods*

1) *Load capacity*

The United Kingdom Post Office now uses the following method for determining load capacity. A sinusoidal tone in the frequency range 300–3400 Hz and of specified harmonic content is injected at the 4-W input to the system The level is increased until the maximum amplitude code is just produced by the PCM encoder. T_{max} of the system is then this level plus a quantity (0.6 dB for a 7-digit 13 segment A-law system) to allow for the difference in an ideal system between the maximum real decision amplitude and the virtual decision amplitude defining T_{max} for the compression law used. The value of T_{max} so determined for an ideal system is required to be within ± 0.5 dB of the specified value.

2) *Quantizing distortion*

The associated Figures 2b and 3b show the significant advantages to be gained by the use of a Gaussian signal (450–550 Hz) over a sinusoid (approx. 500 Hz) for R measurements. The sinusoid measurement is so critical with respect to input level that practical measurements must either employ very small level intervals (of the order of 0.1 dB), or else result in a number of scattered points presenting difficulty in interpretation.

It is considered that an essential working measurement is that of variation in L, the system loss (Figures 2a and 3a). The figures show the theoretical and practical results of such a measurement using both Gaussian and sinusoidal signals. Again the Gaussian signal result is simpler to interpret, but the figures illustrate the effects at low levels of too wide manufacturing tolerances. If the loss at low levels were allowed to increase still further, the change in the R characteristic could still be negligible. But if four such systems were connected in tandem, severe centre-clipping of speech would result, making the overall connections performance quite unsatisfactory. It is for this reason that consideration is being given to the introduction of a loss requirement for future systems (measured by means of a Gaussian signal at low level) that permits only ± 0.5 dB variation in loss down to input levels of approximately 6 dB above T_{min}.

ANNEX 5

(to Question 18/XII)

Effects on customer difficulty in conversation of introducing PCM transmission systems into the telephone network

(Contribution of the United Kingdom Post Office)

1. *Introduction*

The general problem of assessing the transmission performance of the telephone network in the United Kingdom is discussed in Reference [3] and it is explained that fundamental planning data have been assembled in the form of relationships between percentage of customers likely to experience difficulty in conversation and the values of three quantities, namely:

Nominal overall reference equivalent (N.O.R.E.), dB.
Level of injected circuit noise referred to 0 dB receiving reference equivalent (ICNO), dBmp.
Impairment attributable to attenuation/frequency distortion in excess of that present in the local telephone circuits (I), dB.

The definitions of these quantities are given in Reference [1] which also contains the basic data now in use. For convenience in application for network planning, the information is now arranged in two tables, extracts from which are given here as Tables 1 and 2. The information in paragraph 2 explains how this procedure has now been extended to include also the effects of quantizing distortion of the type introduced into telephone connections by PCM systems.

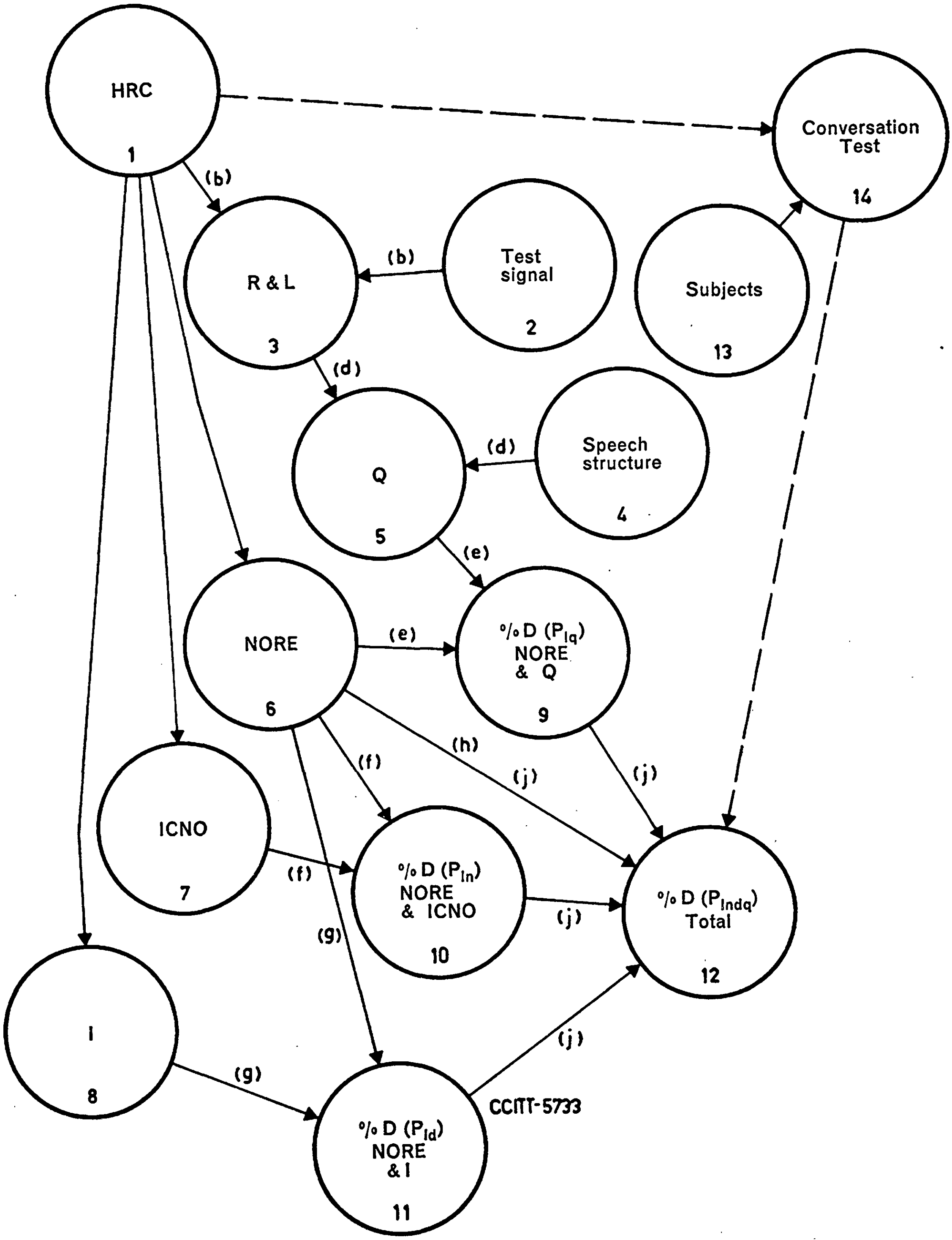


FIGURE 1. — Flow diagram to illustrate the estimation of percentage "difficult" conversations over telephone connections containing one or more PCM links

2. *Determination of "percentage difficult" when quantizing distortion is present*

When a transmission link (say a junction) in a telephone connection is formed by a PCM system instead of the type of line plant used formerly for that link (say unamplified, loaded cable), the following changes in the end-to-end transmission characteristics must be considered:

A. Quantizing distortion and other additional non-linear effects are introduced.

B. The level of circuit noise reaching the customer (I.C.N.O.) may be changed.

C. The overall transmission loss (N.O.R.E.) of the connection may be changed.

D. The total impairment (I) caused by attenuation/frequency distortion and suffered by speech in being transmitted between local exchanges may be changed.

Provided that the appropriate values of N.O.R.E., I.C.N.O. and I can be determined for the changed circumstances, the problem of estimating the "percentage difficult" (%D) can be solved using the data referred to above. The treatment of quantizing distortion will now be explained. Figure 1 will be used to explain the principles. The relevant links between nodes of the flow diagram are lettered to indicate the steps described below.

a) One or more hypothetical reference connections (H.R.C.s) are chosen as representative of an important class of real connections that could be established in the telephone network (node 1 in Figure 1).

b) The total power of the quantizing distortion products is computed or measured in the frequency range 300–3400 Hz when a special, specified test signal is applied at a suitable electrical interface in the connection (e.g. at the input to the first PCM system). The results are expressed in terms of the numbers R and L. As explained in Reference [2], R is the ratio (in dB) of the power of the test signal (without distortion) to that (measured without frequency weighting) of the quantizing distortion products and L is the transmission loss to the test signal (without distortion). R and L must be expressed as a function of the level, s (dBm), of the test signal at the chosen input interface.

c) Speech, which is much more complex statistically than the test signal chosen under b), is represented by a suitable statistical distribution of values of s, the level of the test signal.

d) The equivalent subjective degradation caused by quantizing distortion when speech at a given level of speech voltage is transmitted over the H.R.C. is expressed in terms of the quantity Q (dB) which can be computed from the R-characteristic by suitable convolution as explained in Reference [2]. Q is then expressed as a function of the level of speech voltage at the chosen interface.

TABLE 1

PERCENTAGE "DIFFICULT" OPINIONS AS A FUNCTION OF NOMINAL OVERALL REFERENCE EQUIVALENT AND LEVEL OF CIRCUIT NOISE
(50 dBA room noise present)

Nom. ORE	Level (dBmp) of circuit noise												
dB	None	−70	−68	−66	−64	−62	−60	−58	−56	−54	−52	−50	−48
16	4.0	4.2	4.2	4.4	4.4	4.6	4.8	5.0	5.1	5.5	5.9	6.2	7.0
18	5.1	5.5	5.5	5.5	5.7	5.9	6.0	6.2	6.6	7.2	7.4	8.4	9.6
20	6.2	6.8	6.8	7.0	7.2	7.4	7.6	8.1	8.6	9.4	10.2	11.5	13.5
22	7.9	8.6	8.6	8.8	9.1	9.6	10.2	10.7	11.2	12.4	13.9	16.0	19.0
24	10.0	11.0	11.2	11.5	11.8	12.4	13.2	13.9	15.3	16.8	18.6	21.8	25.6
26	12.8	14.2	14.6	14.9	15.7	16.4	17.2	18.2	19.6	22.1	24.7	29.0	34.6
28	16.4	18.2	18.6	19.0	19.6	20.9	22.1	23.4	25.6	29.0	32.4	37.8	44.4
30	20.5	22.5	23.4	23.8	25.1	26.5	28.0	30.5	33.0	37.3	41.2	47.8	56.5
32	25.6	28.5	29.5	30.0	31.9	33.6	35.7	38.2	41.2	46.2	51.5	59.3	68.5
34	31.4	35.7	36.8	37.8	39.2	41.2	43.9	46.7	50.5	56.5	62.4	71.0	79.5
36	38.6	43.4	44.4	45.6	47.8	50.0	52.7	56.5	60.8	67.0	73.3	81.1	89.0
38	46.7	51.7	52.7	54.4	57.1	59.2	62.4	66.0	71.0	76.9	82.7	89.6	
40	54.9	60.3	61.9	63.4	66.0	69.0	71.9	75.6	80.3	85.5	90.4		

Note. — The level, N, of circuit noise is referred to the input to a receive end of 0 dB receive reference equivalent.

TABLE 2

PERCENTAGE "DIFFICULT" OPINIONS AS A FUNCTION OF NOMINAL OVERALL REFERENCE EQUIVALENT AND "IMPAIRMENT" DUE TO BANDWIDTH LIMITATION OR ATTENUATION/FREQUENCY DISTORTION

(50 dBA room noise present but no circuit noise)

Nom. ORE	Impairment, I, dB										
dB	0	1	2	3	4	5	6	7	8	9	10
16	4.0	4.8	5.7	6.6	7.4	8.4	9.6	11.0	12.1	13.5	15.3
20	6.2	7.2	8.4	9.6	11.0	12.4	14.2	16.0	17.9	19.3	21.4
24	10.0	11.5	13.2	14.9	17.2	19.3	21.4	23.4	25.6	28.0	30.5
28	16.4	18.6	20.9	23.4	25.6	28.0	30.9	34.0	38.8	39.2	41.8
32	25.6	28.5	31.9	35.1	38.2	41.2	48.9	46.8	50.0	53.0	55.7
36	38.6	42.3	46.2	50.0	53.0	56.5	59.0	61.7	64.8	68.0	70.5
40	54.9	58.5	62.7	65.8	69.5	72.3	74.5	77.3	79.9	82.3	84.1

Note. — "Impairment" is measured in a listening opinion test (effort scale) at a mean opinion score of 2.5; the corresponding score in a conversation would be obtained, in the absence of circuit noise and distortion, at a nominal overall reference equivalent of about 32 dB.

e) Information has been prepared in the form of a table showing %D as a function jointly of N.O.R.E. and Q and an extract is reproduced as Table 3. From this information, taking the value of N.O.R.E. appropriate for the H.R.C. and Q corresponding to the median value of speech voltage to be expected at the chosen interface, the value of %D is estimated for a telephone connection as the H.R.C. but without any circuit noise or impairment attributable to attenuation/frequency distortion in the junctions and trunks. This percentage is denoted by P_{lq}.

f) Using the information exemplified by Table 1, %D is estimated for a telephone connection having the desired combination of N.O.R.E. and I.C.N.O. but having no attenuation/frequency distortion and without any quantizing distortion. This percentage is denoted by P_{ln}.

g) Using information such as that shown in Table 2, %D is estimated for a telephone connection having the desired combination of N.O.R.E. and I but having no circuit noise or quantizing distortion. This percentage is denoted by P_{ld}.

h) %D is estimated for a telephone connection having the desired value of N.O.R.E. but without any circuit noise, attenuation/frequency distortion or quantizing distortion. This percentage is denoted by P_l and can be obtained from any one of Tables 1, 2 or 3.

j) Using the results from e), f), g) and h) above, and making certain assumptions, the value of %D is calculated for the situation when all four of the sources of degradation are present as in the H.R.C. under consideration. This percentage is denoted by P_{lndq}. The procedure embodying these assumptions is explained in Reference [3].

Special features

The foregoing procedure caters for the commonly occurring practical situations that require treatment in transmission performance studies. In some cases, special features are present and these require treatment not included in the above. Examples of special features are the following:

1. Off-set of the central decision value from zero.
2. Presence of errors in the digital transmission stream.
3. Presence of large-amplitude but low-frequency interference (e.g. from 50 Hz power system) entering the PCM system with the speech input.
4. Presence of very small amplitude speech signals from another channel (i.e. crosstalk) which might be enhanced by the non-linearity of the PCM process.
5. Existence of mismatch between compression and expansion characteristics of encoder and decoder.
6. Need for facilities to enable transmission loss or gain to be introduced by numerical processing of the digital signals between encoder and decoder.

Information on certain effects associated with the above features is available.

It must be admitted that the procedure adopted for estimating percentage difficulty is based on several assumptions which are very difficult to examine independently in detail. It has been demonstrated by conversation tests that the predictions based on the procedure described above for dealing with the effects of quantizing distortion are consistent with those now in use for treating loss, circuit noise and attenuation/frequency distortion. As has been

TABLE 3

PERCENTAGE "DIFFICULT" OPINIONS AS A FUNCTION OF NORE AND Q

(Room noise 50 dBA; Circuit noise none)

Nom. ORE	Value of Q							
dB	12	14	16	18	20	22	24	00
16			7.6	5.9	5.0	4.4	4.2	4.0
18			8.8	7.0	6.0	5.5	5.3	5.1
20			11.0	8.6	7.4	7.0	6.6	6.2
22		18.6	13.5	10.7	9.4	8.6	8.1	7.9
24		21.8	16.4	13.5	11.8	11.0	10.4	10.0
26	34.6	24.7	19.6	16.8	14.9	13.9	13.5	12.8
28	39.2	29.5	23.4	20.1	18.6	17.9	17.2	16.4
30	43.4	34.0	28.0	24.7	22.5	21.4	20.9	20.5
32	47.8	39.2	34.0	30.0	28.0	27.0	26.0	25.6
34	53.3	45.1	39.6	36.8	34.0	33.0	31.9	31.4
36	59.2	51.7	46.7	43.4	41.2	40.2	39.2	38.6
38	65.0	58.7	53.8	50.5	48.3	47.2	46.7	46.7
40	71.5	66.0	61.3	57.6	56.0	54.9	54.9	54.9

Note 1. — This table is derived from Table 1 using subjectively-determined equivalence between Q and G_n (see Figure 1 of Annex 4) and a relationship between conversational mean opinion score (Y_C) and speech voltage.

Note 2. — It is important to note that, like the corresponding information in Tables 1 and 2, this information is based on conversational mean opinion scores (Y_C) obtained in 1957 using telephone connections with 13-2P-27 telephone sets and converting these scores to %D by use of a conversion based on rather limited data.

indicated from contributions by the United Kingdom Post Office to Questions 4/XII, 7/XII and 15/XII, all planning data of the type exemplified by Tables 1, 2 and 3 are being revised so that modern and future types of telephone sets can be properly catered for and the relative advantages and disadvantages fairly assessed.

REFERENCES

[1] RICHARDS, D. L.: *Transmission performance assessment for telephone network planning, Proc. I.E.E.*, 1964, *111*, pp. 931–940.

[2] RICHARDS, D. L.: *Transmission performance of telephone networks containing pcm links, Proc. I.E.E.*, 1968, *115*, pp. 1245–1258.

[3] RICHARDS, D. L.: *The planning process and assembling fundamental information*, International Seminar on National Telephone Transmission Planning, Melbourne, Australia, 1970.

ANNEX 6

(to Question 18/XII)

Quantization distortion and its effect on telephone transmission performance, by J. Lalou, C.C.I.T.T. Senior Counsellor

SUMMARY

Various quantities proposed for characterizing quantization distortion are compared with one another and with the results of subjective tests. A possible method for this purpose is described.

In the conclusions, reference is made to those features of the study which might concern Study Group XII and Special Study Group D.

1. *Introduction*

This document supplements and replaces [13].

Table 1 is a table of concordance between the notations used in various articles and the system employed in this document.

TABLE 1

CONCORDANCE OF NOTATIONS

Symbols [1]	Symbols [12]	Symbols Here	Definition
N	2B	2B	Number of steps per signal character ($2B = 2^b$)
		d	$d' = I'_{max} = -\Sigma\, p_i \ln (p_i)$
		D	Most significant part of d
		Δ	Signal-to-equivalent noise ratio
		e	r.m.s. voltage of a Gaussian signal
$\sqrt{\overline{e}^2}$		E	r.m.s. voltage of input speech signal
$(\Delta e)_j$	h_j	h_i	Amplitude of step j or i
		h	$h = \frac{V}{B}$, value of h_i in the case of uniform steps
		m	Slope of the compression characteristic, $m = \frac{dv}{du}$
		M	Companding advantage in terms of R_g
$P(e)$	$p(v)$	$p(u)$	Probability density of the speech signal
P_j		p_i	(Discrete) probability of use of step j or i
		$P(u)$	Gaussian probability density of a test signal
		$P(X)$	Probability density of e, given E
		$p(x)$	$P(X)$ — as a function of x
σ		q^2	Sum of the squares of the quantization errors
		Q	Average of R_g, as defined in [9]
		R	Signal to distortion ratio $R = 20 \log_{10} (E/q)$
		R_g	R for a Gaussian signal $R_g = 20 \log_{10} (e/q)$
V	V	V	Maximum (positive) input voltage. The r.m.s. value of the sinusoidal signal attaining this value is $\frac{V}{\sqrt{2}}$
e	v	u	Instantaneous input voltage
v		v	Instantaneous voltage at compandor output (maximum V)
		U_0	(Positive) input voltage corresponding to the lowest decision amplitude
		X	$X = \frac{e}{E}$
		x	$x = 20 \log_{10} X$ ($S' - S$ in [9])
C		$Y\sqrt{2}$	$\frac{\text{overload voltage of compandor}}{\text{r.m.s. voltage of input signal}} = V/\sqrt{\overline{e}^2}$
	$S - T_{max}$	y	Level of the long-term mean power of the signal for a given talker at the input of the system (in dB in relation to the power of the fully loading sinusoidal signal) ($S' - T_{max}$ in [9]) $y = -20 \log_{10} Y$
		z	$z = x + y$

[16]	[4]	[5]	here	
	C		C	Information capacity of a channel
	W		F	Bandwidth (in Hz)
	S		$\overline{S}$	Mean power of signal
	N		$\overline{N}$	Mean noise power
		p_i	p_i	Probability of symbol of rank i
C			I_{max}	Maximum mutual information per use of a channel

Note. — The primes (1) indicate quantities expressed in nepers.

The following quantities proposed for evaluating the signal-to-quantization noise ratio as a function of y have been compared:

$R = 10 \log_{10} \frac{E^2}{q^2}$ (R_g is the value of R calculated or measured when a Gaussian wanted signal is applied);

$R'' + 5$ dB, R'' being defined in [12];

Q, calculated from R as indicated in [9];

Δ and D, both calculated as described below.

The use of quantity R in assessing transmission performance has been criticised as it represents with its mean value a speech signal the components of which have very variable powers ([3], [12]). It will be seen from Figures 1 and 2 and Table 4 that for high signal levels R differs considerably from the other quantities.

According to [12], $R'' + 5$ dB constituted at the time a good approximation to the results of the subjective tests. According to [9], Q gives a good characterization (but only as a relative quantity) of transmission performance as assessed by subjective tests; [10], [22] and [23] show relations between Q and the signal-to-equivalent noise ratio according to subjective tests.

Table 2 shows the cases studied.

TABLE 2

No.	Coding law			Voice-frequency bandwidth (kHz)	Curve reference
	Parameter	Form	b		
1	A = 1	Uniform steps	7	4	[9], Figure 2 (R_g), Figure 3 *(Q)*, curves B = 64, A = 1
2	A = 87.6	13 segments	7	3.1	[10], Figure 5a (R_g), Figure 5b *(Q)*
3	A = 87.6	13 segments	8	3.1	[10], Figure 1a (R_g), Figure 1c *(Q)*
4	μ = 255	15 segments	8	3.1	[10], Figure 1a (R_g), Figure 1c *(Q)*
5	A = 100	Continuous	7	4	[9], Figure 2 (R_g), Figure 3 *(Q)*, curves B = 64, A = 100; [12], Figure 1 *(R)*, Figure 2 *(R'')*, with law of probability according to [2] curves 64 + 64.

2. *Application of information theory*

2.1 *Concept of mutual information*

Let us consider an input random variable u and an output random variable v; by definition ([16], page 16), the mutual information is:

$$I(u; v) = \log \frac{P_i(u/v)}{P_i(u)}$$

where $P_i(u)$ is the *a priori* probability of $u = u_i$

$P_i(u/v)$ the *a posteriori* probability of $u = u_i$ if the received symbol is v_j.

From this definition, mutual information seems to be a good criterion of the extent to which v gives faithful information on u [14].

It can be shown that the average mutual information (over all possible values of u and v)

$\overline{I(u;v)} = \overline{I(v;u)} = H(v) - H(v/u)$

where $H(v)$ is the entropy of the ensemble of values of v

$H(v/u)$ the conditional entropy of v, given u.

In the case of pulse code modulation (PCM), if u is the analogue input voltage and v the quantized output voltage in the absence of errors, $H(v/u) = 0$ since v is exactly known when u is given (a formal proof and additional explanations are given in [14]).

$$\text{Thus we have: } I(u; v) = H(v) = -\sum_{i=-B}^{i=+B} p_i \log(p_i) \qquad (1)$$

On the other hand, if we consider a memoryless channel with additive noise and if the mean power of the input (analogue) signal is limited to $\overline{S}$, it can be shown ([16], from page 335 onwards for discrete time channels and from page 407 onwards for continuous channels) that the mutual information between the received signal (affected by noise) and the input signal is maximum if the input signal has a Gaussian probability distribution. If the noise is also Gaussian, with a mean power $\overline{N}$, the following formula ([16], page 32) is found:

$$\mathrm{I}_{\max} = \frac{1}{2} \log \left(1 + \frac{\overline{S}}{\overline{N}}\right) \qquad (2)$$

Now the "capacity" C of such a channel, with bandwidth F, as defined in [4] and [5], is given by:

$$C = F \log \left(1 + \frac{\overline{S}}{\overline{N}}\right) \qquad (3)$$

Combining (2) with (3), we may write

$$C = 2F\,\mathrm{I}_{\max}$$

The capacity can therefore be said to be the product of the maximum mutual information for a channel utilization ($\mathrm{I}_{\max}$) and the maximum number of utilizations per second (2 F). This is an illustration of formula (3) and not a mathematical proof, which would cause difficulties in the case of channels transmitting continuous signals ([16], pages 383–384).

The above formulae are valid irrespective of the logarithm base used. In what follows, we shall use *Naperian* logarithms; Appendix 1 contains comments on the use of other logarithms and various units.

2.2 *Definition of the signal-to-equivalent noise ratio*

It is a well-known fact that quantization distortion appears as a noise in PCM

If we use $\overline{N}$ to denote the mean quantization noise power (hypothetical) and Δ' to denote the signal-to-equivalent noise ratio as might be determined by subjective tests, we obtain by definition

$$\Delta' = \frac{1}{2}\, ln \frac{\overline{S}}{\overline{N}} \quad \mathrm{Np} \qquad (4)$$

The simplest assumption which can be made is that the listening subject unconsciously attempts to reconstitute the continuous variation signal which he expects to receive and attributes deviations from this signal to an additive noise which would be equal to the quantization distortion. We would thus write $\Delta' = R'$. However, this assumption is only borne out by subjective tests in certain particular conditions [8].

According to a more general hypothesis, the listening subject unconsciously evaluates the maximum quantity of information which the type of signal received can transmit (allowing for the constraints to which the signal is subject). We showed in [15] that the calculations based on formula (3) and the properties of the human ear coincide satisfactorily with the results of the conventional methods used to evaluate telephone transmission performance for analogue signals, despite the fact that the speech signal is far from being Gaussian.

Reverting to p.c.m., we shall assume that the sampling frequency is $2F$ (F being the highest voice frequency to be transmitted), then the power corresponding to the quantization error lies entirely within the signal band. (The clipping which in fact occurs at low frequencies is not due to the principle of the modulation method and will be accounted for separately.) Moreover, the noise of mean power $\overline{N}$ must have a uniform spectrum if formula (2) is to apply: the quantization error meets this condition if the wanted signal has a complex spectrum like that of a speech signal [6]. However, the quantization distortion is not a Gaussian noise and the information theory cannot be used to show that formula (2) is applicable in this case. We may nevertheless consider it probable that the listening subject is basing himself on the experience acquired with analogue signals and that he confuses the quantities g' and d' defined below in his subjective evaluation of Δ'.

$$\text{By convention } g' = \frac{1}{2} \ln \left(1 + \frac{\overline{S}}{\overline{N}}\right) \quad \mathrm{Np} \qquad (5)$$

which tends towards Δ' when $\frac{\overline{S}}{\overline{N}}$ increases.

d' (expressed in nepers) is numerically equal to I'_{max}, expressed in nats by formula (1); we therefore have:

$$d' = - \sum_{i=-B}^{i=+B} p_i \ln (p_i) \quad \text{Np} \tag{6}$$

However, only the results of subjective tests, which we shall examine in section 5, will tell us whether Δ', calculated on the assumption that $g' = d'$ in fact represents the signal-to-equivalent noise ratio. For the time being, we shall calculate d' from formula (6).

Any amplitude lower in absolute value than a certain amplitude U_0 (which may be zero) is coded as if it were zero (centre-clipping effect) and any amplitude greater than V as if it were equal to V (peak-clipping effect). We can therefore write

$$d' = d'_1 + d'_B + D' \tag{7}$$

where

$$d'_1 = -p_1 \ln (p_1) - p_{-1} \ln (p_{-1})$$

$$d'_B = -p_B \ln (p_B) - p_{-B} \ln (p_{-B}) \tag{8}$$

$$D' = -\Sigma p_i \ln (p_i) \ (i \text{ varying from } -B + 1 \text{ to } -2 \text{ and from } +2 \text{ to } B - 1) \tag{9}$$

d'_1 and d'_B do not depend on the compression law $v(u)$ used between U_0 and V; p_1 and p_B represent the probability of occurrence of the instantaneous speech voltage in the corresponding intervals.

d'_1 corresponds to a very specific information (presence or absence of a signal) and it would not seem logical to take it into account in the expression of quantization noise as this occurs only in the presence of a signal; d'_B corresponds to an information indicating only whether the peak of the signal is clipped. It has been observed that with uniform steps and for $b = 5$, the values calculated for $d'_B + D'$ tally well with the results of the articulation tests described in [11]. However, articulation may still be good for a fairly badly mutilated signal (see references [21] - [23] in [1] and does not therefore seem to be a criterion to adopt in the case of PCM

In what follows the values of D' are indicated in place of d'. This is tantamount to ignoring the information conveyed by the peak-clipped signals, which, from the standpoint of the naturalness of the transmitted speech, is only very rough.

3. *Laws of probability p (u)*

3.1 An exponential law of the form shown below has frequently been employed:

with

$$\left. \begin{array}{l} p(u) = G \exp(-\lambda u) \text{ for } u \geqslant 0 \\ p(-u) = p(u) \end{array} \right\} \tag{10}$$

Taking the total probability to be equal to 1 and the r.m.s. value to E, we obtain

$$\lambda = \sqrt{2}/E \qquad G = 1/E\sqrt{2} \tag{11}$$

This law is a simplification of the law indicated from tests by *Davenport* [7]; *Smith* [1] and a number of other authors have used it in theoretical calculations. It must be said that it may lead to contradictions, since it would give a maximum probability density for $u = 0$, whereas this law is applicable only during periods of speech activity.

An arbitrary convention introduced in [13] to avoid such contradictions made it possible to calculate D. The peak clipping, however, produces a subjective noise-like effect which occurs even in the experiments where there is no quantization effect. This effect should therefore be dealt with separately, although its value is still uncertain.

In any case, the calculations based on this exponential law can only be applied to perfect systems, and its results cannot be expressed in terms of measurements of distortion in real systems.

The law enunciated by *Purton* [2] and applied in [12] is identical with the preceding one during periods of activity and assumes that periods of silence account for 30% of the time.

3.2 For certain theoretical calculations, a rectangular law is assumed:

$$\left\{ \begin{array}{l} p(u) = \dfrac{1}{2e\sqrt{3}} \quad \text{for } -e\sqrt{3} \leqslant u \leqslant e\sqrt{3} \\ p(u) = 0 \text{ for } u < -e\sqrt{3} \text{ and } u > e\sqrt{3} \end{array} \right.$$

e being the r.m.s. value.

3.3 The Gaussian probability density is:

$$P(u) = \frac{1}{e\sqrt{2\pi}} \exp\left(-\frac{u^2}{2e^2}\right) \tag{12}$$

3.4 According to [9] or [20], experience shows that a Gaussian probability density distribution represents any syllabic segment of a speech wave considered over a duration of about 10 to 20 milliseconds; during such an interval of time, the statistical parameters of speech may be considered constant.

It may be noted that such a duration is long enough for the human ear to distinguish between successive syllabic segments on the basis of variation in time, whereas within a segment the perception of sounds is accomplished mainly by a sort of harmonic analysis (see for example [17], Chapter 14).

During such a period of 10 to 20 ms hundreds of bits are transmitted; therefore the limiting value defined as the channel capacity by formula (3) could be reached by some type of coding, with, in principle, a Gaussian noise [18]. The coding actually used is very different and highly redundant; however, we are concerned here with the type of information defined in [15] and not with the information corresponding to the meaning of speech.

In addition, as indicated in [14], section 3.0, in formula (1) above $H(v)$ is the entropy of the ensemble of all possible values of v and *not* the entropy of the sequence of random variables emanating from the coder; we must therefore ignore the correlation between successive samples of speech.

For the calculation of D, it is then legitimate to consider as independent (and then add up) the amounts of information calculated by putting —

$$p_i = \iint P_E \, de \cdot P(u) \, du \tag{13}$$

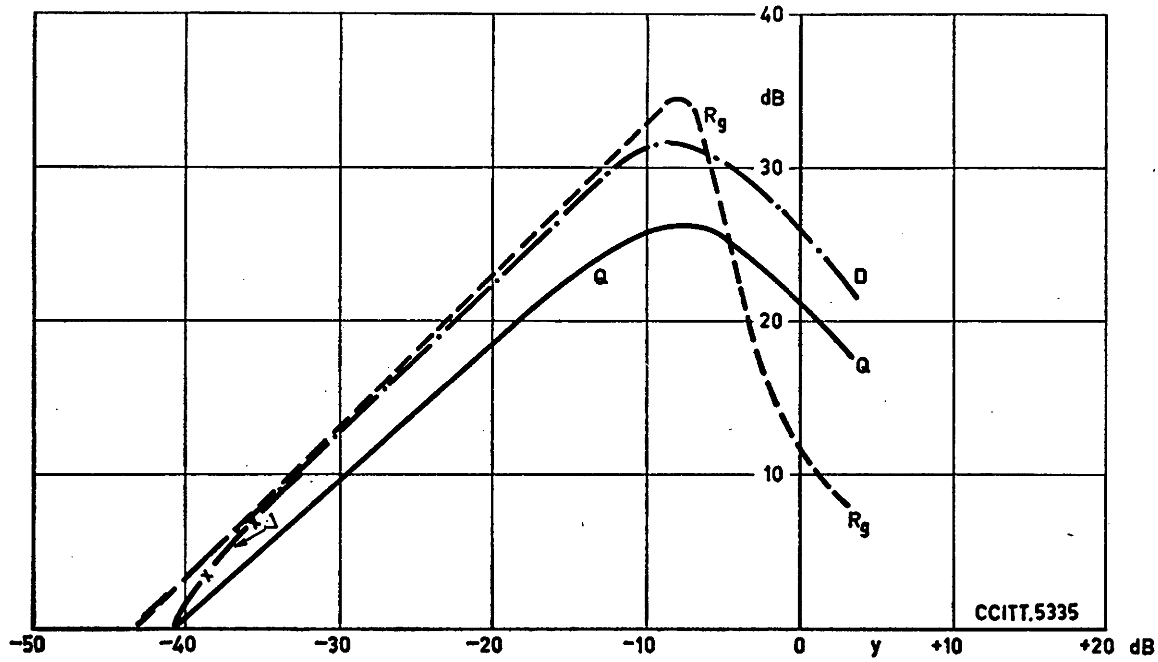
where $P(u)$ is the Gaussian probability density corresponding to a speech segment,

du extends over step h_i

TABLE 3

SYLLABIC PROBABILITY DENSITY AS A FUNCTION OF SYLLABIC POWER LEVEL RELATIVE TO THE MEAN SPEECH POWER LEVEL

x (dB)	Probability density $p(x)$	x (dB)	Probability density $p(x)$
15	0.0001	−13	0.0240
14	0.0002	−14	0.0212
13	0.0006	−15	0.0186
12	0.0012	−16	0.0163
11	0.0024	−17	0.0141
10	0.0043	−18	0.0123
9	0.0071	−19	0.0106
8	0.0108	−20	0.0091
7	0.0154	−21	0.0078
6	0.0207	−22	0.0067
5	0.0265	−23	0.0057
4	0.0325	−24	0.0049
3	0.0381	−25	0.0042
2	0.0432	−26	0.0036
1	0.0474	−27	0.0030
0	0.0505	−28	0.0026
− 1	0.0524	−29	0.0022
− 2	0.0532	−30	0.0019
− 3	0.0529	−31	0.0016
− 4	0.0517	−32	0.0013
− 5	0.0497	−33	0.0011
− 6	0.0471	−34	0.0009
− 7	0.0440	−35	0.0008
− 8	0.0407	−36	0.0007
− 9	0.0372	−37	0.0006
−10	0.0337	−38	0.0004
−11	0.0303	−39	0.0004
−12	0.0271	−40	0.0003



A = 1 b = 7

FIGURE 1

P_E (e) is the probability density of e for a given r.m.s. voltage E of speech and the integration should be carried out over all possible values of e.

In fact $P_E(e)$ is only a function $P(X)$ of $X = \frac{e}{E}$; Table 3 reproduces the values of $p(x)$ published by the United Kingdom Post Office, whence we can derive:

$$p(x)\, d\,[8.686 \ln (X)] = P(X) dX$$

$$P(X) = 8.686\, \frac{p\,(x)}{X}$$

4. *Relations between R, D and Q*

4.1 *D expressed in terms of* R_g

Formulae (1), (2) and (3) of Appendix 2, when converted into decibels, give:

$$D(y) = \int_{X=0}^{X=\infty} R_g\,(z)\; P(X)\; dX + K \tag{14}$$

with

$$K = a - c - 7.78 \text{ dB} \tag{15}$$

$$a = \frac{8.686 \times 2}{\sqrt{\pi}} \int_0^\infty X P\,(X)\, dX \tag{16}$$

$$c = 8.686 \int_0^\infty \ln\,(X)\, P\,(X)\, dX = \int_{X=0}^{X=\infty} x \cdot P\,(X)\; dX = \int_{X=-40}^{X=+15} x \cdot p\,(x)\; dx \tag{17}$$

The proof given in Appendix 2 assumes that there is no centre clipping and no peak clipping.

TABLE 4

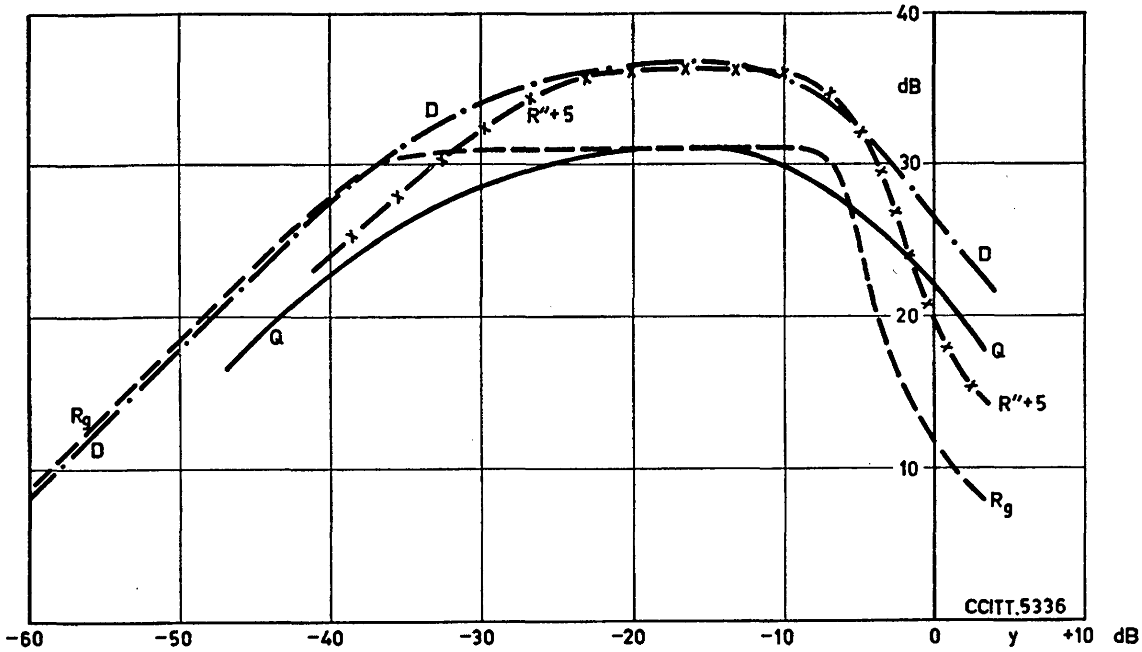
VALUES OF D (IN dB)

y (dB)	Encoding law		
	A = 87.6/13 segments		μ = 255/15 segments
(1)	$b = 7$ (2)	$b = 8$ (3)	$b = 8$ (4)
−45	24.03	29.89	33.87
−44	24.93	30.82	34.54
−43	25.83	31.74	35.17
−42	26.70	32.64	35.77
−41	27.56	33.51	36.34
−40	28.39	34.36	36.89
−39	29.19	35.17	37.40
−38	29.96	35.95	37.89
−37	30.69	36.69	38.35
−36	31.38	37.38	38.78
−35	32.02	38.03	39.18
−34	32.63	38.64	39.56
−33	33.19	39.20	39.91
−32	33.70	39.72	40.24
−31	34.17	40.19	40.55
−30	34.60	40.62	40.83
−29	34.99	41.01	41.09
−28	35.34	41.36	41.33
−27	35.65	41.67	41.56
−26	35.93	41.95	41.76
−25	36.18	42.19	41.95
−24	36.40	42.41	42.12
−23	36.59	42.60	42.28
−22	36.76	42.77	42.42
−21	36.90	42.91	42.55
−20	37.03	43.03	42.67
−19	37.13	43.13	42.76
−18	37.21	43.20	42.84
−17	37.27	43.24	42.89
−16	37.30	43.24	42.90
−15	37.30	43.19	42.86
−14	37.25	43.07	42.76
−13	37.14	42.87	42.58
−12	36.96	42.57	42.30
−11	36.70	42.16	41.91
−10	36.35	41.63	41.39
− 9	35.90	40.96	40.74
− 8	35.33	40.16	39.96
− 7	34.66	39.23	39.05
− 6	33.89	38.17	38.01
− 5	33.01	37.00	36.86
− 4	32.05	35.74	35.61
− 3	31.01	34.41	34.29
− 2	29.92	33.02	32.91
− 1	28.78	31.59	31.50
0	27.61	30.15	30.07
1	26.43	28.71	28.65
2	25.26	27.30	27.24
3	24.10	25.91	25.86

Numerical calculation using formulae (15) to (17) gives:

$a = 7.60, \qquad c = -5.33, \qquad K = 5.15$ dB

Table 4 gives the values of D calculated for encoding laws 2, 3 and 4 in Table 2; the D curves in Figures 1 and 2 were determined (with a lower accuracy) for laws 1 and 5.



A = 100 b = 7

FIGURE 2

4.2 *Relation between $D(y)$ and $R_g(y)$*

For values of z such that peak clipping can be disregarded, $R_g(z)$ is a linear function of z, identical with the expression given in Appendix 1. At values of y which are sufficiently low for z not to go beyond this interval, we shall have

$$R_g(z) = R_g(y) + x$$

$$D(y) = R_g(y) + \int xp(x)\,dx + K = R_g(y) - 0.18 \text{ dB} \tag{18}$$

4.3 *Relation between D and Q*

Q is defined in [9] or [20]; it is clear that for values of Q corresponding to an acceptable quality of transmission the curve $D(Q)$ tends towards an asymptote with the following equation:

$$\Delta(y) = D(y) = Q(y) + K$$

Figure 3 represents the relation between D and Q obtained by eliminating y between the calculated values of D and the already known values of Q (see Table 2). Although expression (14) of D has only been proved in the absence of centre clipping and peak clipping, the points calculated for the descending part of the characteristics $D(y)$ (hence with peak clipping effect) are practically on the same curve as the points calculated for the ascending part of these characteristics and a single curve $D(Q)$ is found for all the encoding laws.

For low values of Q and D, where the difference between D and Δ is not negligible, Δ is fairly close to Q.

In practice, the value of Q must be determined for a chain of circuits [21] by combining the noise powers existing at various points of the chain and the quantization "noise" powers (characterized by R), taking account of the relative levels.

To obtain D by a similar process, it is noted that formula (14) can be written

$$D(y) = \int_{X=0}^{X=\infty} [R_g(z) + K]\,P(X)\,dX$$

with $K = 5.15$ dB

By way of example, D was calculated by combining an added noise of power B (characterized by the signal-to-noise ratio G_a) and the power N, by writing

$$R_g + K = 10 \log_{10} \frac{S}{N} \text{ and } G_a = 10 \log_{10} \frac{S}{B}$$

G_a is given and S is arbitrary, but the same in both cases.

We calculate

$$T = 10 \log_{10} \frac{S}{N + B}$$

for all the possible values of z and for two values of G. We then calculate

$$D(y) = \int_{X=0}^{X=\infty} T(z)\, P(X)\, dX$$

After eliminating y between D and Q, we obtain the relations between D and Q which have also been included in Figure 3, for $G_a = 25$ dB and $G_a = 35$ dB.

4.4 *Note on the calculation of R, Q and D*

R_g can be measured on a real system. For the numerical calculations we used R_g values calculated by a regression method, as described in [19]. For the theoretical calculations in Appendix 2 we assumed that R_g was given by the theoretical law in [1], which gives the same results in the interval of y values in which formula (14) was proved.

All these formulae are based on the assumption that the voice frequency band is 4 kHz, which is the case for Figures 1 and 2. For the 3.1 kHz band used in practice, we have to increase R_g by 1.1 dB. The same increase is applied to D (which was done in Table 4) and Q.

All the numerical values of D and Q were thus increased in Figure 3.

5. *Comparison with the results of subjective tests*

5.1 *Case where* $D = R$

The subjective tests described in [8] were carried out with y values low enough to ensure that there was no appreciable peak clipping effect, but high enough so that Δ was virtually equal to D. Therefore, in accordance with paragraph 4.2 we substantially have $D = R$. These tests have shown that, in these conditions, R was equal (to within $\pm$ 0.5 dB) to the signal-to-equivalent noise ratio assessed subjectively.

For high values of y, where there is strong peak clipping, R decreases much more rapidly than D or Q and does not convey an accurate idea of the transmission quality.

5.2 *Comparison with* $R''+5$

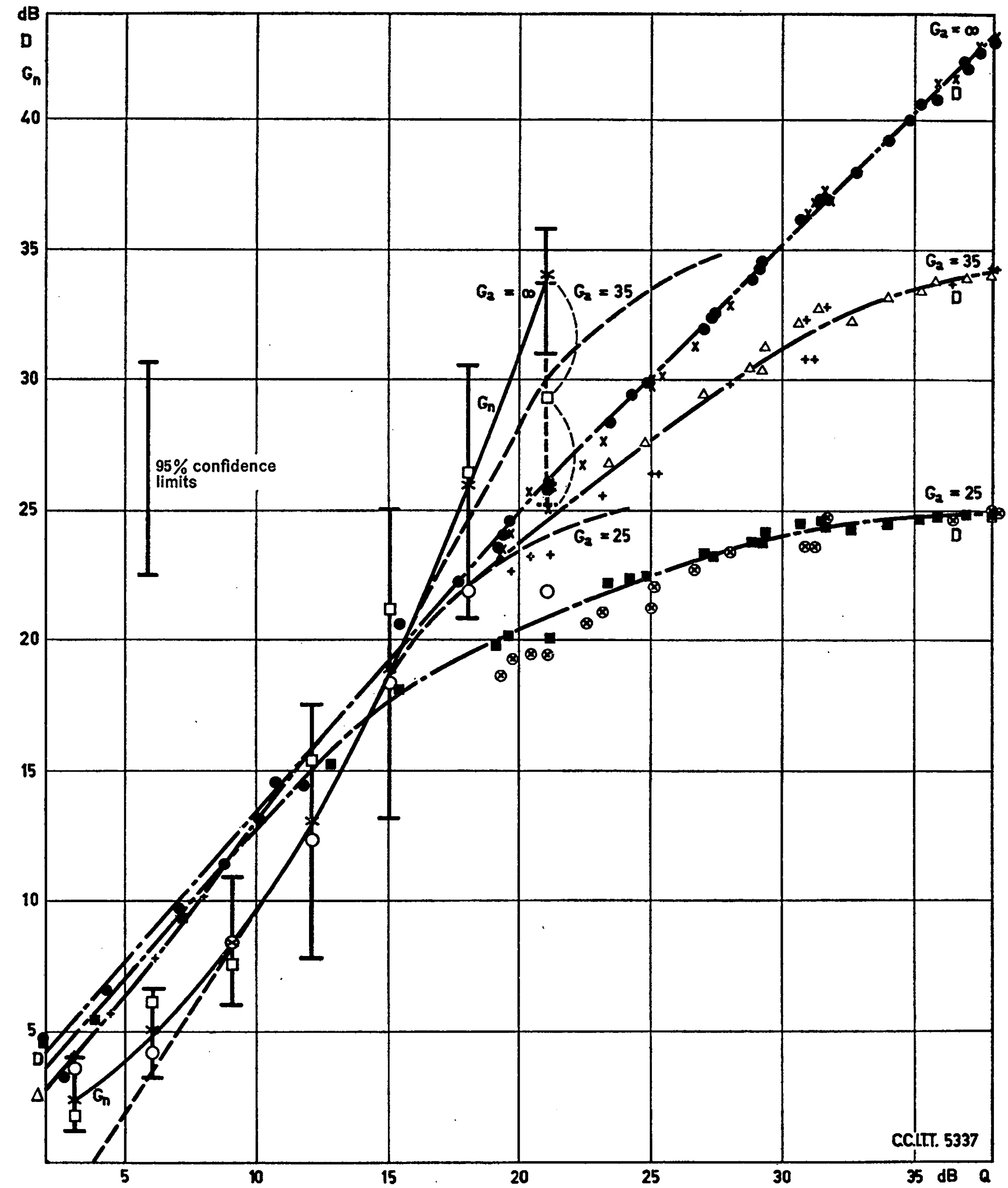
Figure 2 shows the values of D for encoding law 5 (the only one for which $R'' + 5$ values are available). D is clearly very close to $R'' + 5$ for the steps in the curves; for lower values, $R'' + 5$ comes closer to Q.

5.3 *Comparison with a speech-modulated noise*

Subjective tests in which the quantization noise was simulated by producing a random noise proportional to the voice currents by means of a device graduated in Q values are described in [10], [22] and [23]. In some of these tests, a constant noise was added which was characterized by the signal-to-noise ratio G_a. Figure 3 contains the signal-to-equivalent noise ratio G_n derived from these tests, as a function of Q, so that this ratio can easily be compared with D or Δ.

For Q values of practical utility (of the order of 10 to 20 dB), in the absence of inserted noise, D is in fairly good agreement with the signal-to-equivalent noise ratio G_n determined subjectively, which is, moreover, subject to an appreciable scatter. For $Q = 18$ and 21 dB, G_n increases more rapidly than D, but according to other subjective tests described in [22], when Q reaches 20 to 22 dB, (i.e. $D >$ approx. 25 dB), the quantization distortion becomes imperceptible (thus G_n is very large). Moreover, such values of D are obtained for almost all y values, with a single codec, and the recommended encoding laws (see Table 4).

The values obtained with an added noise are therefore of greater practical utility. Figure 3 shows that there is a good agreement between the calculated values of D (or, if applicable, of Δ) and the G_n values determined by the subjective tests, particularly for $G_a = 25$ dB.



Fitted curves G_n (Subjective tests) for $G_a = \infty$ $G_a = 35$dB $G_a = 25$dB

[10] ———— × □ ○

[23] — — — —

calculated points D (Q, G_a

$G_a = \infty$ $G_a = 25$ $G_a = 35$ dB

● ■ △ – in the increasing part

× ⊕ + – in the decreasing part

Curves ——×— — —×——△ D (Q, G_a)

FIGURE 3

6. *Conclusions*

From the standpoint of Study Group XII, which is concerned with the overall quality of hypothetical reference connections, Δ is the calculated quantity which is closest to the signal-to-equivalent-noise ratio evaluated by subjective tests. D, which is obtained in the first stage of the calculation, is practically equal to Δ when the transmission quality is acceptable.

There is a one-to-one correspondence between D and Q, irrespective of the compression law. This corroborates the value of Q for comparisons of relative transmission quality. There are several grounds for believing that D gives a better indication of the absolute quality, but when already calculated values of Q are available, they can readily be used to deduce D from Figure 3.

From the standpoint of Special Study Group D, R_g is the quantity which must be measured on a real codec in order to calculate D on a complete connection. The equipment specification must therefore indicate the lower limits of R_g; this points to the utility of measurement methods that yield the signal-to-quantization distortion ratio which would be measured with an ideal and perfectly Gaussian signal.

Bibliographical References

[1] Smith, Bernard: Instantaneous Companding of Quantized Signals, *B.S.T.J.*[1] May 1957, Vol. 36, p. 653–709.

[2] Purton, R. F.: A Survey of Telephone Speech-Signal Statistics and Their Significance in the Choice of a P.C.M. Companding Law. *Proceedings of the Institution of Electrical Engineers*, 1962, 109B, p. 60. (See also discussion, p. 485, of the same volume.)

[3] Velichkin, A. I.: Correlation Function and Spectral Density of Quantized Speech. *Journal of Acoustics of the Academy of Science of the U.S.S.R.*, 1963, 9, p. 13. English version: *Soviet Physics, Acoustics*, 1963, 9, p. 10.

[4] Shannon, C. E.: Communication in the presence of noise. *Proceedings of the Institute of Radio Engineers*, Vol. 37, pp. 10–21, January 1949.

[5] Shannon, C. E.: A mathematical theory of communication. *B.S.T.J.*, Vol. 27, pp. 379–423 and 623–656, July–October 1948.

[6] Bennett, W. R. Spectra of quantized signals. *B.S.T.J.*, Vol. 27, pp. 446–472, July 1948.

[7] Davenport, W. B. Jr.: An experimental study of speech-wave probability distributions. *Journal of the Acoustical Society of America*, Vol. 24, pp. 390–399, July 1952.

[8] Barducci, I., Bianchi, F., and Giacomelli, L.: Quantization noise in a P.C.M. system and its influence on the quality of the transmission. C.C.I.T.T. *Red Book*, Vol. V*bis* (Geneva, 1964, published by the I.T.U.) pp. 379–397.

[9] C.C.I.T.T., 1964–1968: Contributions COM XII—No. 32, and COM XV—No. 72. Choice of encoding law for p.c.m. transmission systems (United Kingdom) (Contained in part in [21]).

[10] United Kingdom Post Office: Annex 4, Question 18/XII, C.C.I.T.T. *Green Book*, Volume V.

[11] Chile Telephone Co.: C.C.I.T.T. *Red Book*, Vol. V*bis*, pp. 362–364, Annex 2 to Question 27/XII.

[12] United Kingdom Post Office: C.C.I.T.T. *Red Book*, Vol. V*bis*, pp. 372–379. Annex 6 to Question 27/XII.

[13] C.C.I.T.T. Secretariat: Effect of quantization distortion on telephone transmission performance (pp. 6–25 of 1968–1972—Contributions COM XII—No. 48, COM Sp. D–No. 55).

[14] Messerschmitt, D. G.: Quantizing for maximum output entropy (Memorandum — File 36674–12), 2 June 1970.

[15] Lalou, J.: Calculation of telephone transmission performance by the information theory, C.C.I.T.T. *Red Book*, Volume V*bis*, Geneva, 1964, pp. 168–199.

[16] Gallagher, R. G.: *Information theory and reliable communication*, John Wiley & Sons, 1968.

[17] Fletcher, H.: *Speech and hearing in communication*, D. van Nostrand, 1961.

[18] Slepian, D.: Bounds on communication, *B.S.T.J.*, Volume 42, pp. 681–707, May 1963.

[19] Richards, D. L.: Speech transmission performance of p.c.m. systems. *Electronics Letters*, 1965, *1*, No. 2, p. 40.

[20] Richards, D. L.: Distortion of speech by quantizing. *Electronics Letters*, June 1967, *3*. No. 6, p. 230.

[21] Richards, D. L.: Transmission performance of telephone networks containing p.c.m. links. *Proc. I.E.E. 1968*, 115, pp. 1245–1258.

[22] Richards, D. L. and Berry, R. W.: Impairment of telephone speech by quantizing. *Fourth International Symposium on Human Factors in Telephony*, September 1968, VDE Verlag, Berlin.

[23] American Telephone and Telegraph Co.: Annex 3, Question 18/XII C.C.I.T.T. *Green Book*, Volume V.

[1] *Bell System Technical Journal*

APPENDIX 1

(to Annex 6)

Notes on the units used

Let us see what happens to some of the foregoing formulae when decimal logarithms are used. We have

$$I_{\max} = \frac{1}{2} \log_{10}\left(1 + \frac{\overline{S}}{\overline{N}}\right) \tag{2}$$

Moreover, according to the normal conventions, we must put

$$\Delta = 10 \log_{10}\left(\frac{\overline{S}}{\overline{N}}\right) \quad \text{dB} \tag{4}$$

$$g = 10 \log_{10}\left(1 + \frac{\overline{S}}{\overline{N}}\right) \quad \text{dB} \tag{5}$$

and formula (6) becomes

$$d = 20\, I_{\max} = -20 \sum p_i \log_{10}(p_i)$$

If we use logarithms to base 2, formulae (1) to (3) remain unchanged. In this case, no convention is established for expressing a signal-to-noise ratio, but it is found that, if $I_{\max}$ varies by 1 shannon, D varies by 6.02 dB and D' by 0.693 Np.

Sample application

$$\text{According to [1] } q^2 = \sum \frac{h_i^3}{12} p(u_i)$$

For a given law $p(u_i)$, R' is maximum if all the h_i are equal.

Furthermore, for given e, D' is maximum if all the p_i are equal.

The only way to meet both conditions is to consider encoding with uniform steps and a rectangular probability distribution as described in paragraph 3.2.

Provided that $e\sqrt{3} \leqslant V$, there is no peak clipping, and each probability p_i has the same value

$$p_i = h_i\, p(u) = \frac{V}{B} \times \frac{1}{2e\sqrt{3}}$$

we then obtain

$$D' = -\ln(p_i) = \ln(2B) + \ln\left(\frac{e\sqrt{3}}{V}\right) \quad \text{Np}$$

$$D = 6.02\, b + y + 1.76 \quad \text{dB}$$

which is identical with the well-known expression of R.

For $e = \dfrac{V}{\sqrt{3}}$, D' and R' reach the same supremum (over all possible values of e)

$$D'_M = R'_M = \ln(2B)$$

According to formula (4), the channel capacity in this case is $2\,F \times \log_2(2B) = 2\,F\,b$ shannons/seconds and is equal to the bit rate of the digital signal, which clearly constitutes an upper limit to the information which can be transmitted on average.

APPENDIX 2

(to Annex 6)

Calculation of R'_g and D'

We assume that the probability density of the speech signal is that defined in section 3.4 above.

Between U_0 and V, the symbols of probability p_i correspond to equal steps $h = \frac{V}{B}$ of the *output* voltage v of the compressor. We have as a first approximation $p_i = p(u_i)\, h_i$ with $h_i = h \frac{du}{dv} = \frac{h}{m}$.

According to Table 3, P_E is practically zero outside a region which can be subdivided into n intervals δ. The probability $P_E\,(e_k)\,\delta$ is assigned to a Gaussian law $P\,(u/e_k)$ having a standard deviation e_k, whose contribution to the probability p_i is $h_i\, P(u_i/e_k)$, so that we have

$$p_i = \sum_{k=1}^{k=n} P_E\,(e_k)\,\delta\, h_i\, P\,(u_i/e_k)$$

To abbreviate, let us write $l_i = -\ln\,(p_i)$

Since $\ln\,(p_i)$ varies much more slowly than p_i, we may use the exponential law of section 3.1 (although it is only approximate) to evaluate $\ln\,(p_i)$.

$$l_i = \lambda u_i + \ln\,[m\,(u_i)] - \ln\,(hG)$$

$$\text{with } -\ln\,(hG) = \ln\,(2B) + \ln \frac{E}{\sqrt{2}}$$

Since $v_0 \leqslant |u_i| \leqslant V$, the double integral of formula (13) is reduced to the sum of a finite number of terms, some of which are given in Table 5 (for $u > 0$), so that the addition can be made in any order. If the addition is first made for each column in Table 5, we obtain

$$D = \sum_{K=1}^{K=n} P_E\,(e_k)\,\delta\, f_k$$

$$\text{with } f_k = 2 \sum_{i=2}^{i=B-1} P\,(u_i/e_k)\, l_i\, h_i$$

TABLE 5

$p_2\, l_2 = P_E\,(e_1)\,\delta\, h_2\, P(u_2/e_1)\, l_2 + \ldots + P_E\,(e_k)\,\delta\, h_2\, P(u_2/e_k)\, l_2 + \ldots + P_E\,(e_n)\,\delta\, h_2\, P(u_2/e_2)\, l_2$
$p_i\, l_i = P_E\,(e_1)\,\delta\, h_i\, P(u_i/e_1)\, l_i + \ldots + P_E\,(e_k)\,\delta\, h_i\, P(u_i/e_k)\, l_i + \ldots + P_E\,(e_n)\,\delta\, h_i\, P(u_i/e_n)\, l_i$
$p_{B-1}\, l_{B-1} = P_E\,(e_1)\,\delta\, h_{B-1}\, P(u_{B-1}/e_1)\, l_{B-1} + \ldots + P_E\,(e_k)\,\delta\, h_{B-1}\, P\,(u_{B-1}/e_k)\, l_{B-1} + \ldots$ $+ P_E\,(e_n)\,\delta\, h_{B-1}\, P(u_{B-1}/e_n)\, l_{B-1}$

We can consider f_k as the calculation result of a definite integral by the trapezoidal method, with intervals h_i which happen to be equal to the quantization steps. Provided that all the h_i are small, this result is practically equivalent to the value of the integral derived from theoretical considerations; the fact that there are finite steps is reflected only by the presence of h and m in the formulae. We can thus write:

$$D' = \int_E P(X)\, f\,(e)\, dX$$

$$\text{where } f(e) = 2 \int_{U_0}^{V} l(u)\, P(u)\, du$$

and we can write:

$f(e) = f_1 + f_2 + f_3$

with

$$f_1 = 2 \int_{U_0}^{V} \lambda u\, P(u)\, du = - \frac{2\,\lambda e^2}{e\sqrt{2\pi}} \left[\exp. \left(\frac{-u^2}{2e^2} \right) \right]_{U_0}^{V}$$

$$f_2 = 2 \left[\ln (2B) + \ln \frac{E}{V\sqrt{2}} \right] \int_{U_0}^{V} P(u)\, du$$

$$f_3 = 2 \int_{U_0}^{V} \ln (m)\, P(u)\, du$$

For the same Gaussian signal

$$R'_g = \ln (e) - \ln (q)$$

$$\text{with } q^2 = 2 \int_{U_0}^{V} \frac{V^2}{12\, m^2\, B^2} P(u)\, du$$

$$\text{or } R'_g = \ln (2B) + \ln \frac{e\sqrt{3}}{V} + M'$$

$$\text{where} \quad M' = -\frac{1}{2} \ln \left[2 \int_{U_0}^{V} \frac{P(u)}{m^2}\, du \right]$$

Case of uniform steps

In this case, $m = 1$, then $f_3 = 0$. If, in addition, we can ignore centre clipping and peak clipping, $M = 0$ and we have simple expressions:

$$f(e) = \frac{2}{\sqrt{\pi}} \frac{e}{E} + \ln (2B) + \ln \frac{E}{V\sqrt{2}}$$

$$R'_g = \ln (2B) + \ln \frac{e}{E} + \ln \frac{E\sqrt{3}}{V}$$

When integrating over X, we obtain:

$$D' = \int_0^{\infty} R'\, P(X)\, dX + a' - c' - \frac{1}{2} \ln 6 \qquad \text{Np} \tag{1}$$

$$\text{with } a' = \frac{2}{\sqrt{\pi}} \int_0^{\infty} X P(X)\, dX \tag{2}$$

$$c' = \int_0^{\infty} \ln (X)\, P(X)\, dX \tag{3}$$

General case

In general, m is a function of u. For the μ companding law, for instance:

$$m = \frac{m_0}{1 + \mu \dfrac{u}{V}} \qquad \text{with } m_0 = \frac{\mu}{\ln (1 + \mu)}$$

then, if we can ignore centre clipping,

$$M' = \ln (m_0) - \frac{1}{2} \ln \left[\int_{-V}^{+V} P(u)\, du + \frac{2\mu}{V} \int_{-V}^{+V} P(u)\, u\, du + \frac{\mu^2}{V^2} \int_{-V}^{+V} P(u)\, u^2\, du \right]$$

$$f_3 = \ln (m_0) \int_{-V}^{+V} P(u)\, du + \int_{-V}^{+V} \ln \left(1 + \mu \frac{u}{V} \right) P(u)\, du$$

If z is low enough so that $\mu \frac{u}{V} \ll 1$, there is certainly no peak clipping. We then have

$$\int_{-V}^{+V} P(u)\, du = 1$$

$$\int_{-V}^{+V} P(u)\, u\, du = 0$$

$$\int_{-V}^{+V} P(u)\, u^2\, du = e^2$$

and we can use expansions of the logarithms, which gives

$$M' = \ln (\mathrm{m}_0) - \frac{1}{2} \ln \left(1 + \frac{\mu^2 e^2}{V^2}\right)$$

$$M' = \ln (\mathrm{m}_0) - \frac{1}{2} \frac{\mu^2 e^2}{V^2} + \ldots$$

$$f_3 = \ln (\mathrm{m}_0) + \int_{-V}^{+V} \frac{\mu u}{V} P(u)\, du - \frac{1}{2} \int_{-V}^{+V} \frac{\mu^2 u^2}{V^2} P(u)\, du$$

$$f_3 = \ln (\mathrm{m}_0) - \frac{1}{2} \frac{\mu^2 e^2}{V^2} + \ldots$$

The companding advantages for $f(e)$ and R'_g are then equal and formulae (1), (2) and (3) of this Appendix remain valid at least for such low levels.

List of Questions allocated to Study Group XII

No.	Short title	Remarks
1/XII	National system reference equivalents in the new transmission plan	
2/XII	Assessment of service transmission quality	
3/XII	Reference equivalents of operators' headsets	
4/XII	Effect of circuit noise on transmission performance	
5/XII	Hourly noise clause	Question 8/XVI [a]
6/XII	Users' tolerance of echo and propagation time	
7/XII	Determination of transmission quality by objective measurement	
8/XII	Measuring the efficiency of a microphone of a receiver	
9/XII	Sidetone	
10/XII	Increase in the sensitivity of local systems	
11/XII	Limits for intelligible crosstalk	
12/XII	Artificial voices, mouths and ears	
13/XII	Non-linear distortion of telephone apparatus	
14/XII	Effect of attenuation distortion	
15/XII	Measuring of loudness ratings	
16/XII	Impedance variations in subscriber lines and telephone sets	
17/XII	Loudspeaker telephones	
18/XII	Transmission performance of pulse code modulation systems	

[a] Results of the study to be transmitted to Special Study Group C.