THE INTERNATIONAL TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE
(C.C.I.T.T.)

IVth PLENARY ASSEMBLY

MAR DEL PLATA, 23 SEPTEMBER - 25 OCTOBER 1968

WHITE BOOK

VOLUME V

Telephone transmission quality, local networks and telephone sets

Published by
THE INTERNATIONAL TELECOMMUNICATION UNION
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CONTENTS OF THE C.C.I.T.T. BOOKS
APPLICABLE AFTER THE IVth PLENARY ASSEMBLY (1968)

WHITE BOOK

Each volume contains, where appropriate, extracts from contributions received on the subject of the volume concerned whenever their interest is such as to warrant publication.

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NOTICE TO VOLUME V OF THE WHITE BOOK

Volume V of the *White Book* supersedes Volumes V (New Delhi, 1960) and V*bis* (Geneva, 1964) of the C.C.I.T.T. *Red Book.*

It has been indicated (immediately after the titles of Recommendations or Supplements) whether the texts are new ones approved by the Plenary Assembly of Mar del Plata, 1968, 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

Some of the recommendations in this present publication give quantities expressed in more than one system of units [1]. The following conventions apply in such cases:

1. A quantity in brackets shows a conversion to a suitable degree of accuracy.

Examples: 2500 km (1550 miles); 2.1 nepers (18.2 decibels).

2. When there is a choice of one of two values, depending on the unit used, they are given with the word "or" between them.

Examples: 9.5 mm or 0.375 inch; −10 dB or −1.2 Np.

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

dBm (Nm): the absolute (power) level in decibels (in nepers);

dBm0 (Nm0): the absolute (power) level in decibels (in nepers) referred to a point of zero relative level;

dBr (Nr): the relative (power) level in decibels (in nepers);

dBm0p (Nm0p): the absolute psophometric power level in decibels (in nepers) referred to a point of zero relative level.

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[1] *Note by the C.C.I.T.T. Secretariat*

In issuing Recommendation B.4 (*White Book*, Volume I), which advocates the exclusive use of the decibel in certain cases, the IVth Plenary Assembly (Mar del Plata, 1968) recognized that this Recommendation could not be applied immediately to the present publication.

PART 1

SERIES P RECOMMENDATIONS

Quality of telephone transmission; local telephone installations and betworks

SECTION 1

GENERAL RECOMMENDATIONS ON THE TRANSMISSION QUALITY FOR AN ENTIRE INTERNATIONAL TELEPHONE CONNECTION

A booklet (common to Volumes III and V of the *White Book*), to be inserted immediately after this page, contains the Recommendations of this section.

P.11 Reference equivalents in an international connection [1].
P.12 Articulation reference equivalent (A.E.N.).
P.13 Transmission impairments and noise.
P.14 Mean one-way propagation time.
P.15 Group-delay distortion.

This booklet also contains the following Recommendations of section 2:

P.20 Transmission characteristics of national networks [2].
P.21 Reference equivalents of national systems [1].

[1] These two Recommendations replace former Recommendation P.11 (*Red Book*, Volume V).
[2] Replaces former Recommendation P.21 (*Red Book*, Volume V*bis*).

GENERAL RECOMMENDATIONS ON THE TRANSMISSION QUALITY FOR AN ENTIRE INTERNATIONAL TELEPHONE CONNECTION [1]

RECOMMENDATION G.111 (P.11) (Geneva, 1964; amended in Mar del Plata, 1968)

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 (for a country of average size).

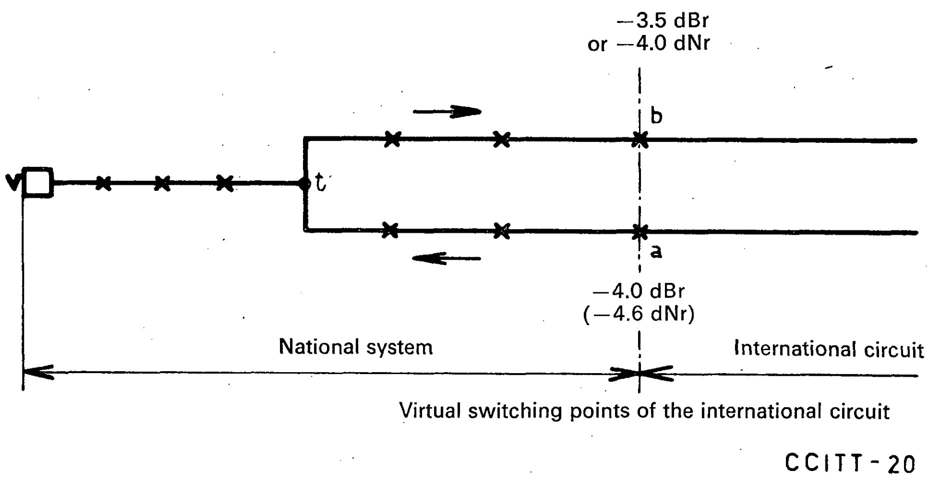

FIGURE 1. — **Definition of the virtual switching points**

[1] The Recommendations appearing under this heading constitute sub-section 1.1. of Section 1, *White 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 or —4.0 dNr, sending
—4.0 dBr (—4.6 dNr), receiving

The nominal transmission loss of this circuit at the reference frequency between virtual switching points is therefore 0.5 dB or 6 cNp.

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 20.8 dB (24 dNp); and

— the nominal reference equivalent of the receiving system between the same two points should not exceed 12.2 dB (14 dNp).
(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 or 6 cNp 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 or 6 cNp 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

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, *White Book*, Volume V).

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

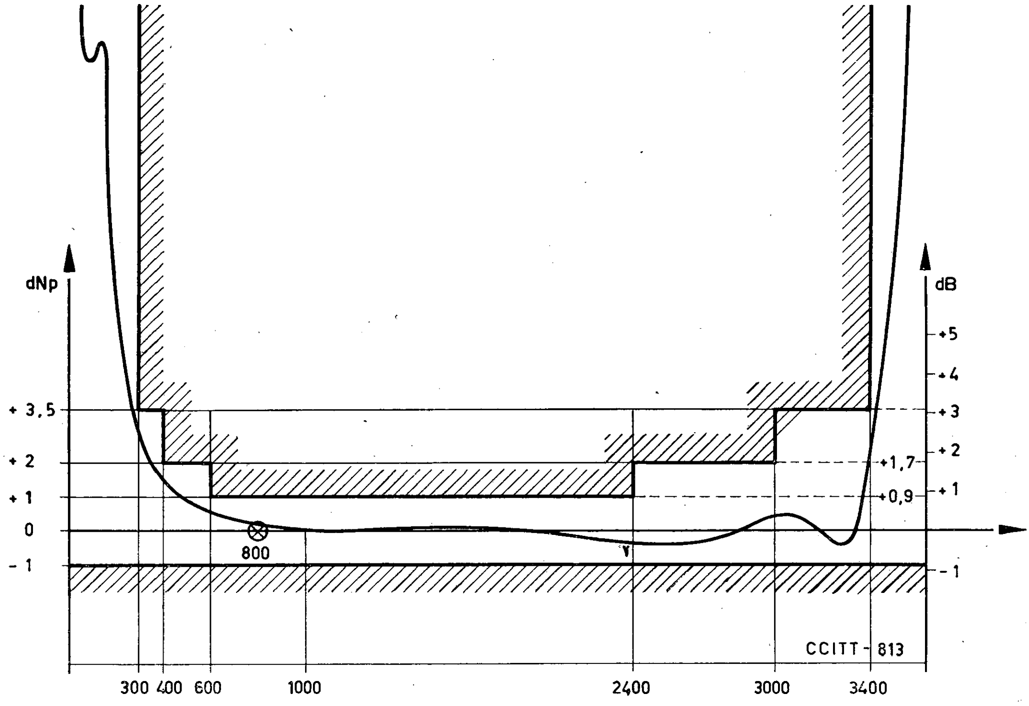

FIGURE 2. — **Characteristic of A.R.A.E.N. filter**

(The staircase diagram is Diagram No. 2 B 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.

[1] Figure 2 shows the actual characteristic of these filters and reproduces Diagram No. 2 B of Recommendation G.232.

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 7 cNp per annum, a systematic increase due to ageing of the microphone. This point is being studied by the C.C.I.T.T. within the framework of Question 1/XII.

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

These 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, are not applicable to the transmission plan now recommended by the C.C.I.T.T.

RECOMMENDATION G.112 (P.12) (modified in Geneva, 1964 and in 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 *White 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 or nepers,
L = length of the toll circuit in kilometres,
K = coefficient which depends on the type of toll circuit considered, in decibels per kilometre or nepers 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 or nepers. 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 or 1 dNp 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.

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[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.

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 (*White Book*, Volume V).

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 or 2.8 Np;

— the nominal A.E.N. of the national receiving system should not exceed 18 dB or 2.1 Np.

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 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 or 7 Np 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 or private operating agencies 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*, 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 toll circuits

A toll circuit 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 British Administration, 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].

[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.

Therefore, the A.E.N. of the toll circuit 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 toll circuit, 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 toll circuit (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:

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

where

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

L = length of toll circuit in kilometres;

K = coefficient, which depends on the type of toll circuit considered, in decibels per kilometre or in nepers 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 toll circuit 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 toll circuit in the national network considered.

RECOMMENDATION G.113 (P.13) (amended in 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 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 decibels.

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.

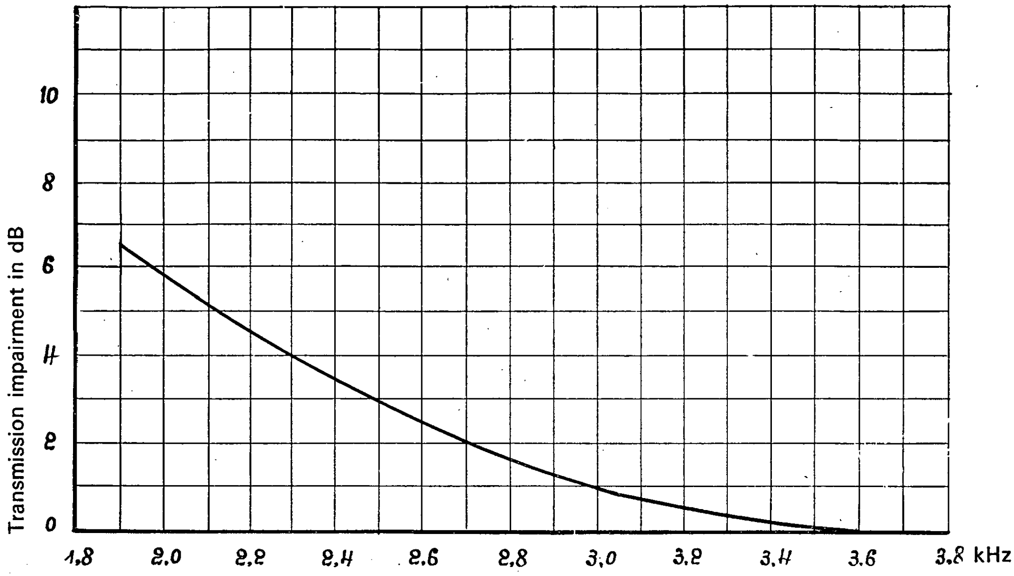


FIGURE 1. — **Transmission impairment due to bandwidth limitation (cut-off impairment)**

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.

b) *due to room noise*

The method of measuring A.E.N. takes account of 60 dB of room noise (Hoth spectrum) 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 mean values of noise power of the connections, should not exceed −43 dBm0p or −5 Nm0p referred to the input of the first circuit in the chain of international circuits.

[1] The power density spectrum of the room noise used in A.E.N. measurements is given in Figure 2. 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.

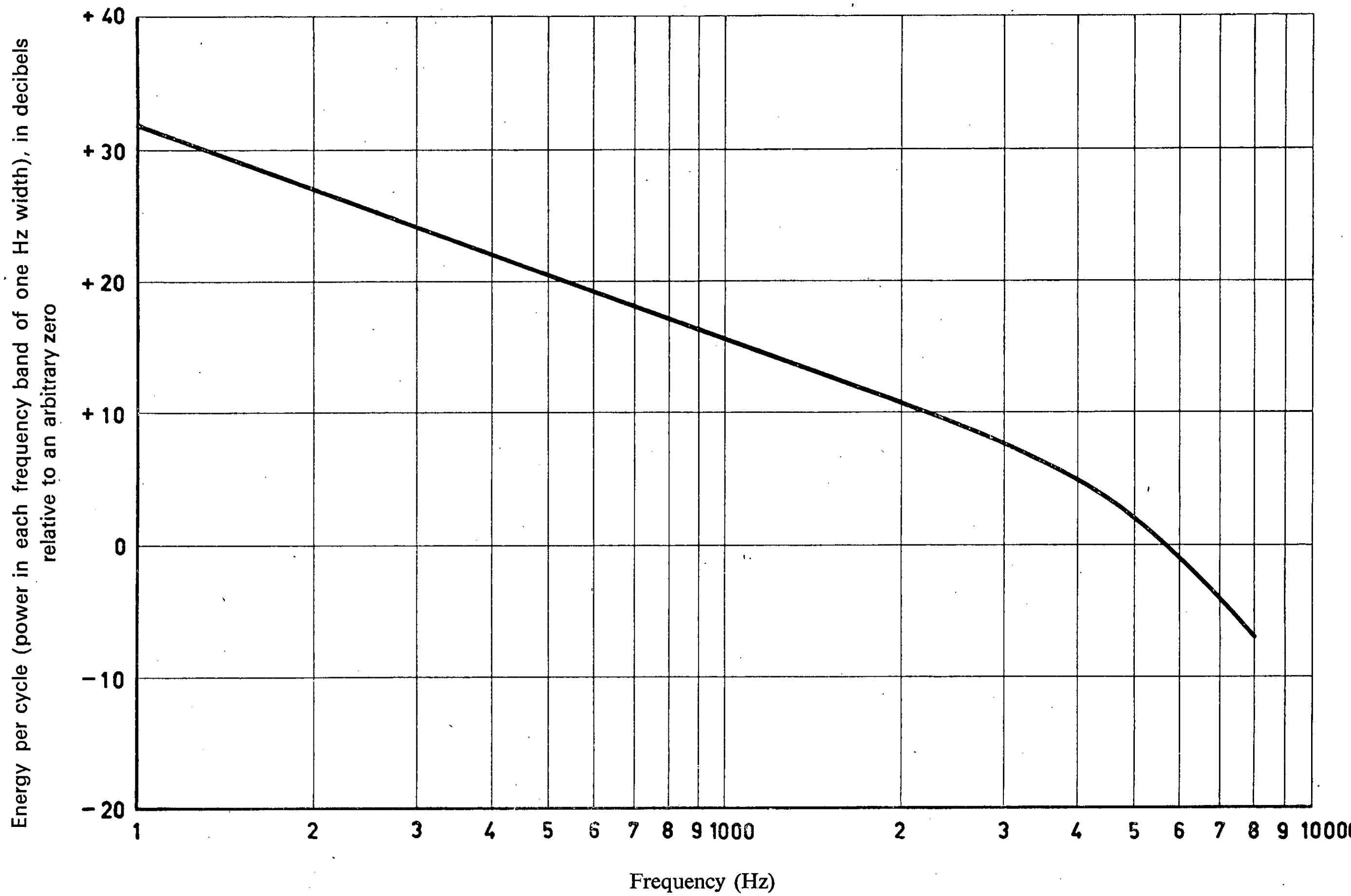


FIGURE 2. — **Power density spectrum of the room noise produced in the listening cabinet of the C.C.I.T.T. Laboratory**

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

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).

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 in 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 bibliography below, particularly with reference 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 bibliography in selecting, from alternatives, plans involving delays in range b) above.

Bibliography

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. ½, 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:

$12+(0.0064\times\text{distance in miles})$ ms

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

Here the factor 0.0064 (or 0.004) is based on the assumption that national trunk circuits will be routed over high-velocity plant (155 miles/ms or 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 100 miles (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)

100 miles/ms (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 8700 miles or 14 000 km altitude	110 ms
Satellite at 22 500 miles or 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 3.

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 in 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

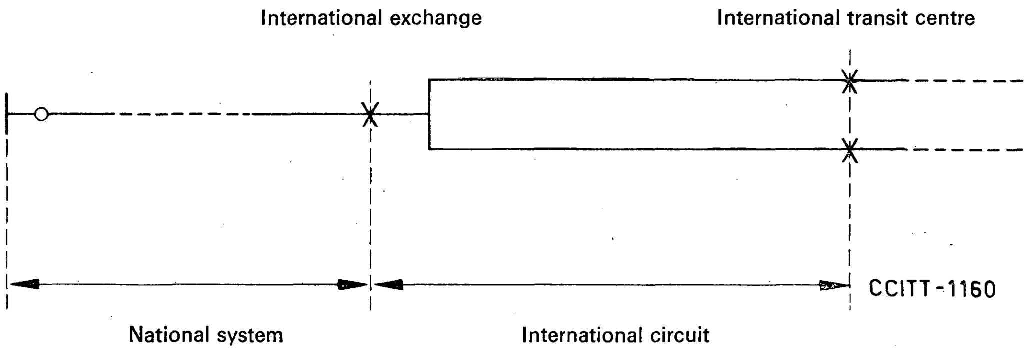
Typical group delays at various frequencies for a chain of 12 circuits in tandem are given in Recommendation G.232 (*White Book*, Volume III).

[1] Same as Recommenation G.133 (*White Book*, Volume III).

1.2 General characteristics of national systems forming part of international connections [1]

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) [2]

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.11) 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.

[1] Recommendations G.120 (P.20) and G.121 (P.21) in this sub-section also form part of Volume V.

[2] Former Recommendation P.21 of Volumes V and V*bis* of the *Red Book* amended at Mar del Plata, 1968; did not appear in Volume III of the *Blue Book*.

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

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

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 systems over very short distances to Recommendation G.125.

3. National trunk circuits should have characteristics enabling them to conform to Recommendations G.131, G.132 and G.134 in the *White 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 G.142 in the *White Book*, Volume III.

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 Part II of Chapter V (Transmission) of the handbook on "*National Telephone Networks for the Automatic Service*".

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. Supplement No. 7, *White Book*, Volume V, describes the transmission plans used in various countries. The Annex below gives some information on methods that may be applied to draw up such plans. Information on methods of planning national networks is also given for information purposes in Chapter V (Transmission) of the handbook on *National telephone networks for the automatic service.*

ANNEX

(to Recommendation G.120 (P.20))

Information on the organization of a national telephone network

a) *General organization and nomenclature* (see Chapter V (Transmission) of the handbook on *National Telephone Networks for the Automatic Service*, pp. 4-5).

b) *Choice of method for specifying transmission performance*

Different methods are used in some countries to ensure satisfactory transmission performance for national calls. For example:

— Supplement 8, *White Book*, Volume V (former Annex 1, *Red Book*, Volume V, pp. 167-173) describes the methods, based on opinion tests, employed by the United Kingdom Administration;

— "North-American practice for transmission requirements of the national network" is described under this heading in Supplement 7, *White Book*, Volume V;

— another section of the same Supplement explains how the A.E.N. method described in Recommendation P.12 (G.112) is applied in the Japanese national network.

The simplest procedure, however, which is used by many administrations, is to set reference equivalent limits for national calls since, in any case, this must be done for international calls. Once the sending and receiving reference equivalents for every type of subscriber set used in the country are known, the reference equivalent of any connection (or part of a connection) can be calculated by the methods outlined in Chapter V of the handbook on *Local Telephone Networks*, (Section 5 and Annex 3).

c) *Improvement of performance in existing networks*

Within existing telephone networks it is important to improve the transmission quality for unfavourably situated telephone sets which handle considerable traffic and especially international traffic. Several methods can be used for this purpose, for example:

1) Repeaters may be used on subscriber lines, junction circuits in the networks of large towns and toll circuits.

Note. — These repeaters may be either two-wire repeaters of the standard type or negative impedance repeaters (two- or four-wire). In each case it should be verified that the stability of the transmission remains adequate.

2) The transmitting and receiving insets may be graded in several qualities and the better insets may be fitted in the telephone sets served by lines having the greatest attenuation and vice versa.

3) Telephone sets specially designed for particularly long subscriber lines may be used. They may include an amplifier at the sending end.

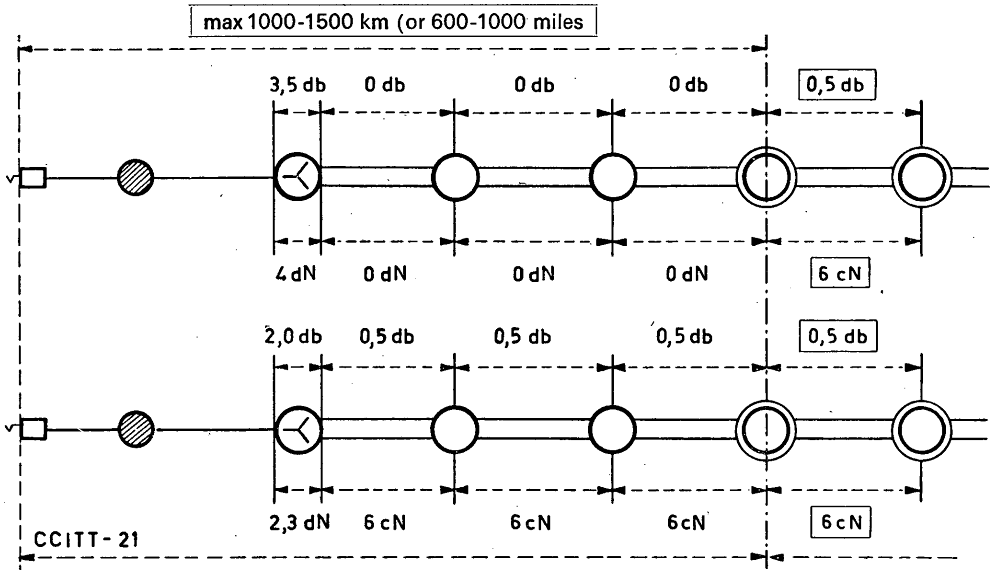
RECOMMENDATION G.121 (P.21) (Geneva, 1964; amended at Mar del Plata, 1968)

REFERENCE EQUIVALENTS OF NATIONAL SYSTEMS

A. DEFINITION

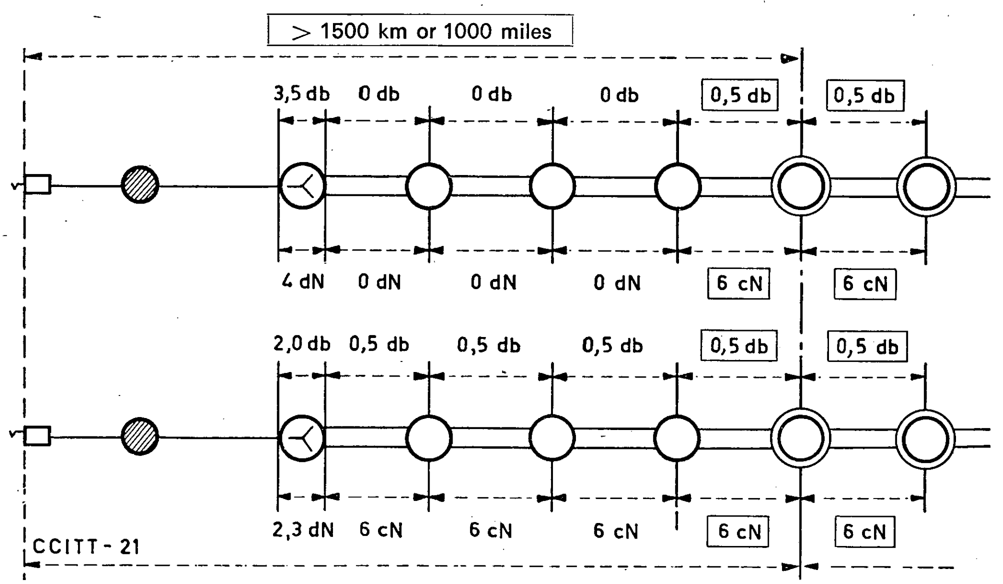
By definition, the virtual switching points of the national system are the theoretical points at which the system is interconnected to the virtual switching points of the international telephone circuits—i.e. points a and b of Figure 1 of Recommendation G.111 (P.11) and the figure appearing in Recommendation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



Maximum reference equivalents for 97% of calls		
	Sending	20.8 dB (24 dNp)
	Receiving	12.2 dB (14 dNp)

FIGURE 1. — **Distribution of equivalents for an international call, in a country of average size**



Maximum reference equivalents for 97% of calls		
	Sending	21.3 dB (24.6 dNp)
	Receiving	12.7 dB (14.6 dNp)

FIGURE 2. — **Distribution of equivalents for an international call, in a large country**

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

B. Maximum nominal sending and receiving reference equivalents

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:

— the nominal reference equivalent of the sending system between a subscriber and the first international circuit should not exceed 20.8 dB (24 dNp);

— the nominal reference equivalent of the receiving system between the same two points should not exceed 12.2 dB (14 dNp).

In a large country, these limits shall be, respectively: 21.3 dB (24.6 dNp) and 12.7 dB (14.6 dNp) if a fourth national circuit is part of the four-wire chain, or 21.8 dB (25.2 dNp) and 13.2 dB (15.2 dNp) if five national circuits form part of the four-wire chain.

In Figures 1 and 2, 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), the limits of 20.8 dB and 12.2 dB cannot be met immediately, but an attempt should be made to abide by them when the networks are reorganized or when telephone sets of a new type are introduced.

Note 2. — The 97% limit is provisional, and it is desirable to use a higher percentage when planning new networks.

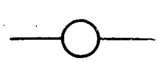
Note 3. — The nominal reference equivalents given for national systems include the systematic differences between the performances of the subscriber set at the sending and receiving ends and their nominal values; however, they do not include the variations of loss with time in the various parts of the national system, nor fortuitous variations of the reference equivalents assessed by subjective methods.

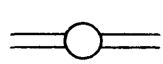
C. Minimum reference equivalents

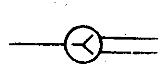
Administrations must take care not to overload the international transmission systems if they reduce the attenuation in their national trunk network. This aspect of the problem must be studied separately before any precise recommendation can be prepared.

Legend for Figures 1 and 2

 Subscriber's set

 Two-wire switching exchange

 Four-wire switching exchange

 Exchange with terminating unit [1]

 Local exchange

 International exchange

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

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 and private operating agencies 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, junction lines 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 (NOSFER) or with a working standard system already compared with NOSFER 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).

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 Annex 3 to Chapter V of the handbook on *Local Telephone Networks* (the method described in paragraph 6 of this annex is also applicable to junctions and toll circuits).

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

$$q = Q - Q_0 \qquad (1)$$

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 assumed that the required precautions have been taken to assess separately the effect of the variations in the feed current.

Note 2. — Part b) of Question 7/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 NOSFER 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 (*White 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 or 2 Np.

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 (0.8 and 1.2 Np) 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 which penetrates through the acoustic leak between the earcap and the human ear is picked up by the microphone and is transmitted as sidetone to the earcap and the ear of the listener, thus increasing the total noise received.

Note 2. — Even when the value 17 dB or 2 Np is attained, administrations may consider it advisable to set a limit for room noise (see Recommendation G.113 (P.13)).

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 insertion 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 operators' set while being used on an international telephone call should not cause, in the silent listening position (microphone out of circuit), an insertion loss greater than 0.43 decibel (0.05 neper) 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 or private operating agencies put any new type of operator's set into service they must check that this is still so.

Note 2. — On an international telephone call the operators' positions should, so far as the reference equivalent between two operators or between an operator and a subscriber is concerned, not exceed the limits specified in Recommendation P.11.

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 insertion 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 decibel (0.03 neper) 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 insertion 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 and private operating agencies 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 to the circuits concerned, in the normal manner, the subscribers of the local network, 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 decibels (1.3 neper);

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 subscriber's 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 subscriber's 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 RENTED TEMPORARILY 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. Until the results of this study become available, administrations and private operating agencies should refer to the recommendation below, which lays down the conditions which should be satisfied by subscribers' stations used with international circuits rented temporarily for private purposes.

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

considering

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

unanimously recommends

that it is desirable for the rented 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 rented circuits should satisfy the same electrical conditions as commercial connections between subscribers,

unanimously recommends

1. that it is desirable for administrations and private operating agencies 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 and private operating agencies to reserve themselves the right to verify by means of volume meters, that the volume transmitted over rented telephone circuits does not reach an excessive level;

3. that, where administrations and private operating agencies 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 to all international telephone circuits.

RECOMMENDATION P.32

DEVICES FOR RECORDING MESSAGES OR TELEPHONE CONVERSATIONS

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

considering

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

that, where certain administrations or private operating agencies 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 out 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 insertion loss does not exceed 0.5 decibel (0.06 neper) 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 decibels (2.3 nepers) above the background noise. 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 decibels (2.3 nepers) 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 28.7 decibels (3.3 nepers), 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 decibels or 6.3 nepers 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 disk 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 neces-

sary 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)

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,

the C.C.I.T.T. considers it necessary to expedite studies so that a final reply to Question 17/XII may be drawn up by the end of the 1968-1972 study period.

However, 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 absolute power level, corrected to a zero relative level point, 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 disks 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 this volume 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 by means of the junction switching panel.

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)

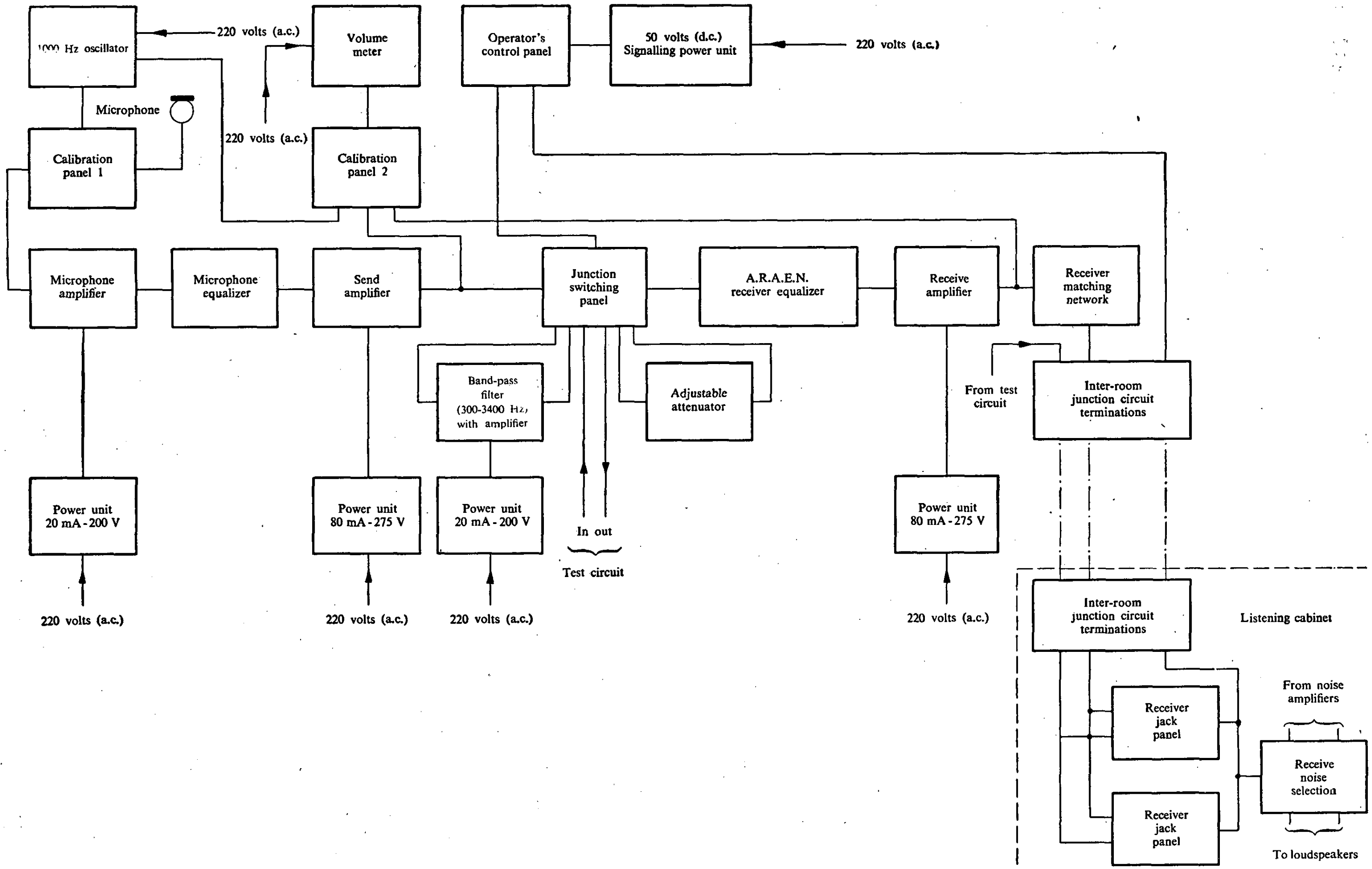


FIGURE 1. — **Schematic diagram of the reference equipment for the determination of A.E.N.**

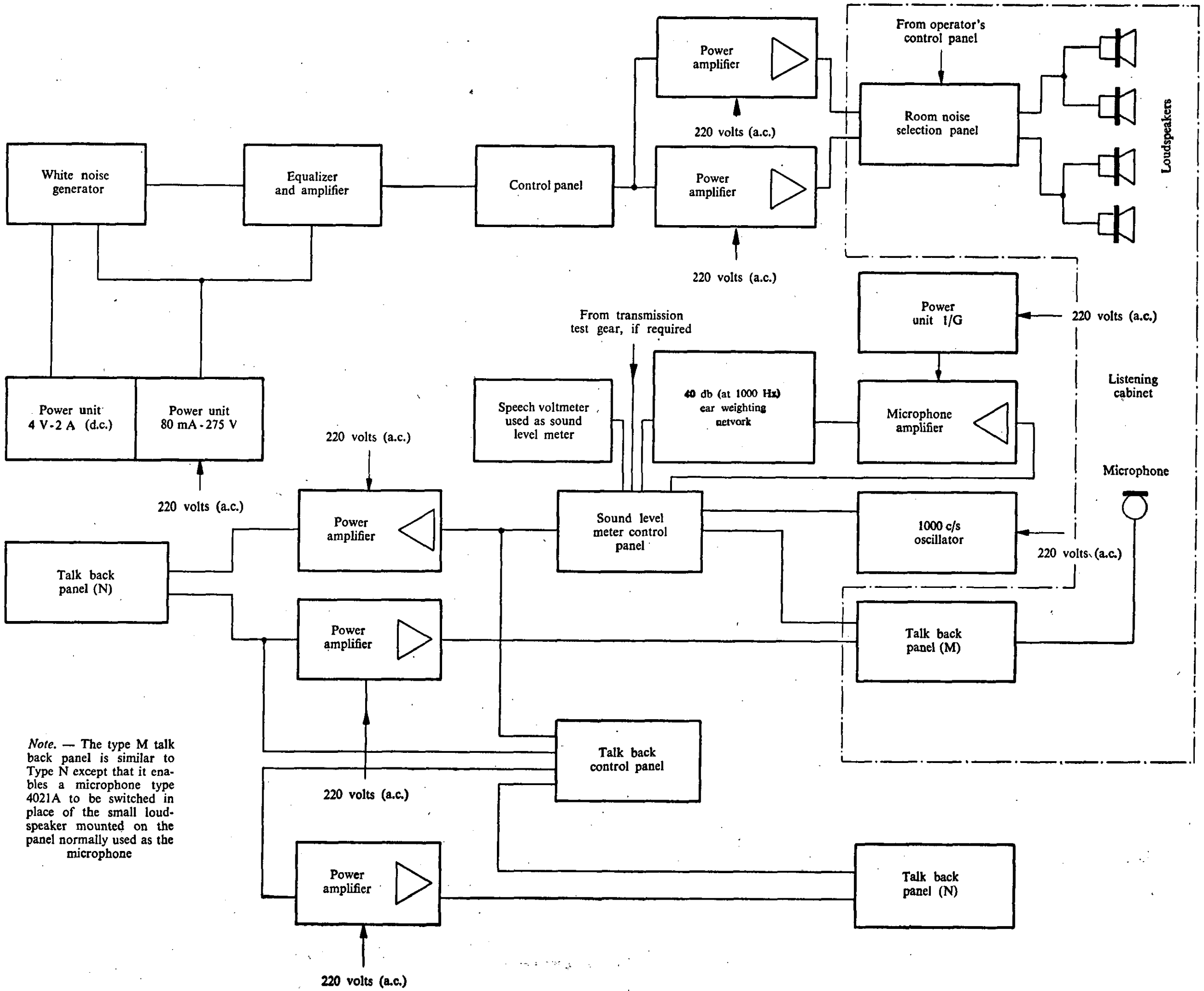


FIGURE 2. — **Noise generation, level measurement and talk-back equipment**

B. Equipment for supply of room noise and intercommunication circuit

This equipment, of which Figure 2 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. The method of using this equipment at the C.C.I.T.T. Laboratory is described in Supplement No. 9 of this volume.

The Rayleigh disk 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 disk 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.

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.

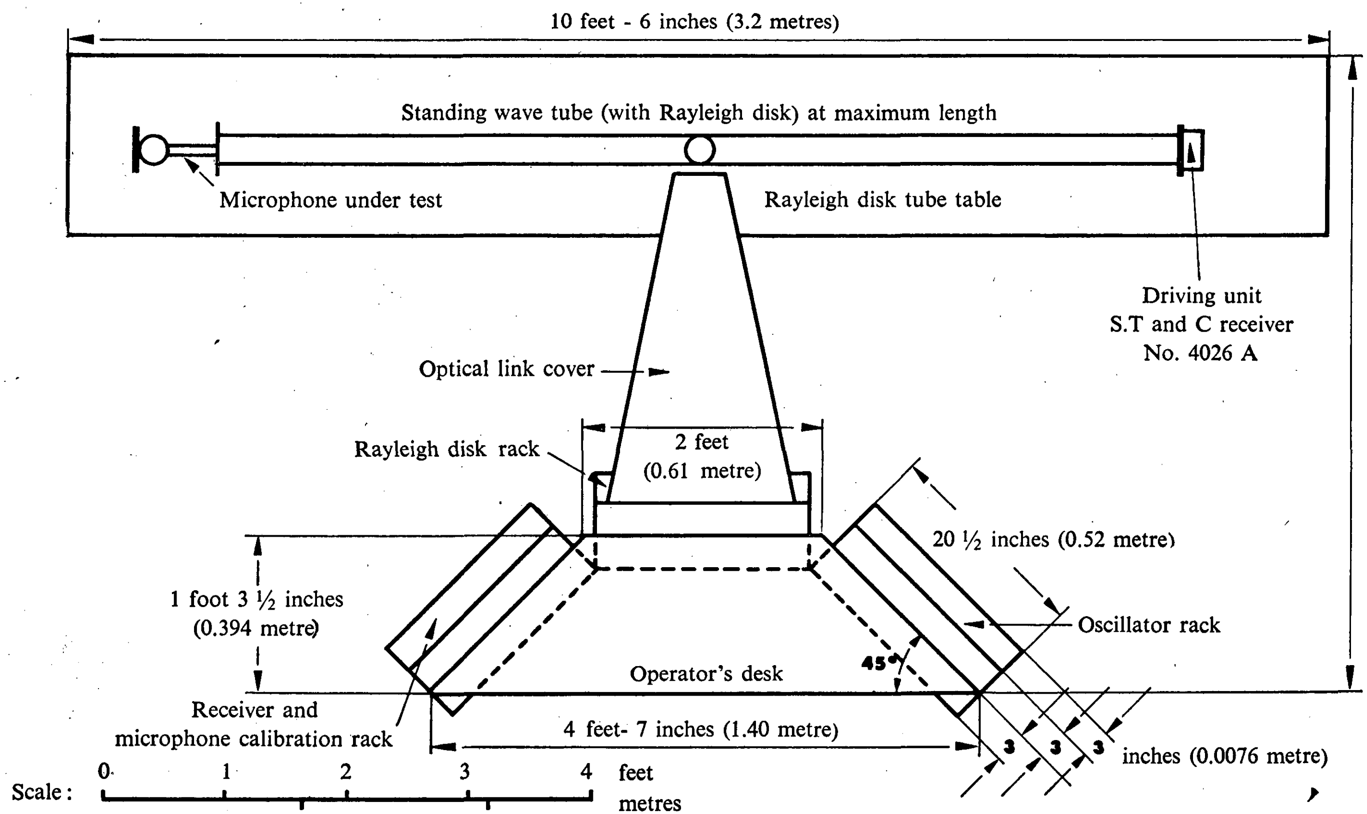


FIGURE 3. — **Plan of microphone and receiver calibration equipment A.R.A.E.N.**

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 the C.C.I.T.T. Laboratory. This equipment should not be regarded as forming a specific part of the reference system.

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 13.25 inches (about 33 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.

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:

$$20 \log_{10} \frac{39.37}{13.25} = 20 \log_{10} \frac{100}{33.5} = 9.5 \text{ decibels.}$$

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

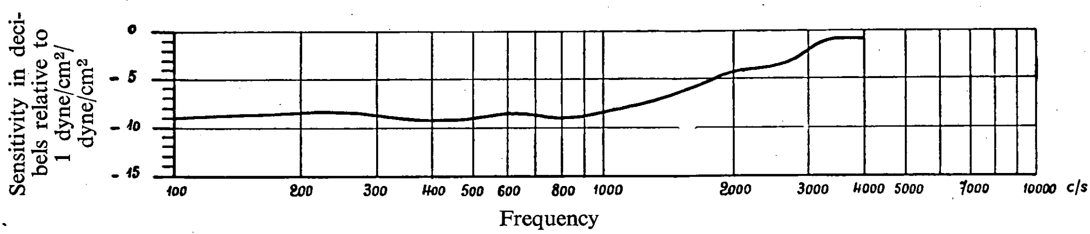
TABLE 2

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

[1] The rim of the baffle plate of the microphone is situated at 12 inches (1 foot) from the talker's lips.

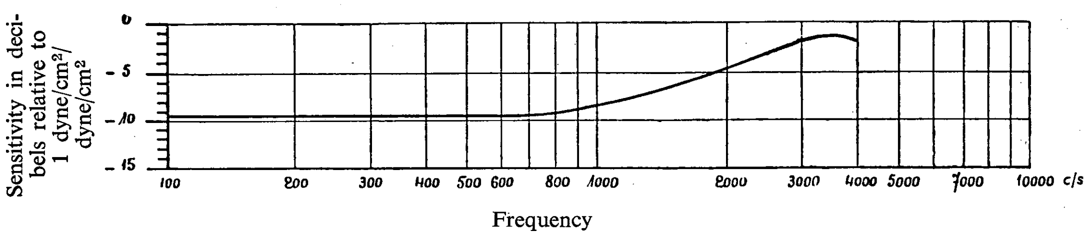
Sensitivity of the A.R.A.E.N. sending end. — The sensitivity of the sending end of the A.R.A.E.N. has been fixed at a value permitting the control of the speaking level by means of a specified speech voltmeter (see Recommendation P.52) connected to the output of the sending system.

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 1 dyne/cm².



(*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 { Send amplifier " normal "; Junction : 30 dB; Receive amplifier " normal " +1 dB



(*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. — **A.R.A.E.N.**

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 decibels.

Table 3 gives the values of acoustic pressure (in decibels relative to 1 dyne/cm²) produced by a receiver when a level of −30 decibels relative to 1 volt is applied to the input of the receiving system — i.e. when an acoustic pressure of 1 dyne/cm² is applied to the microphone.

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[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.

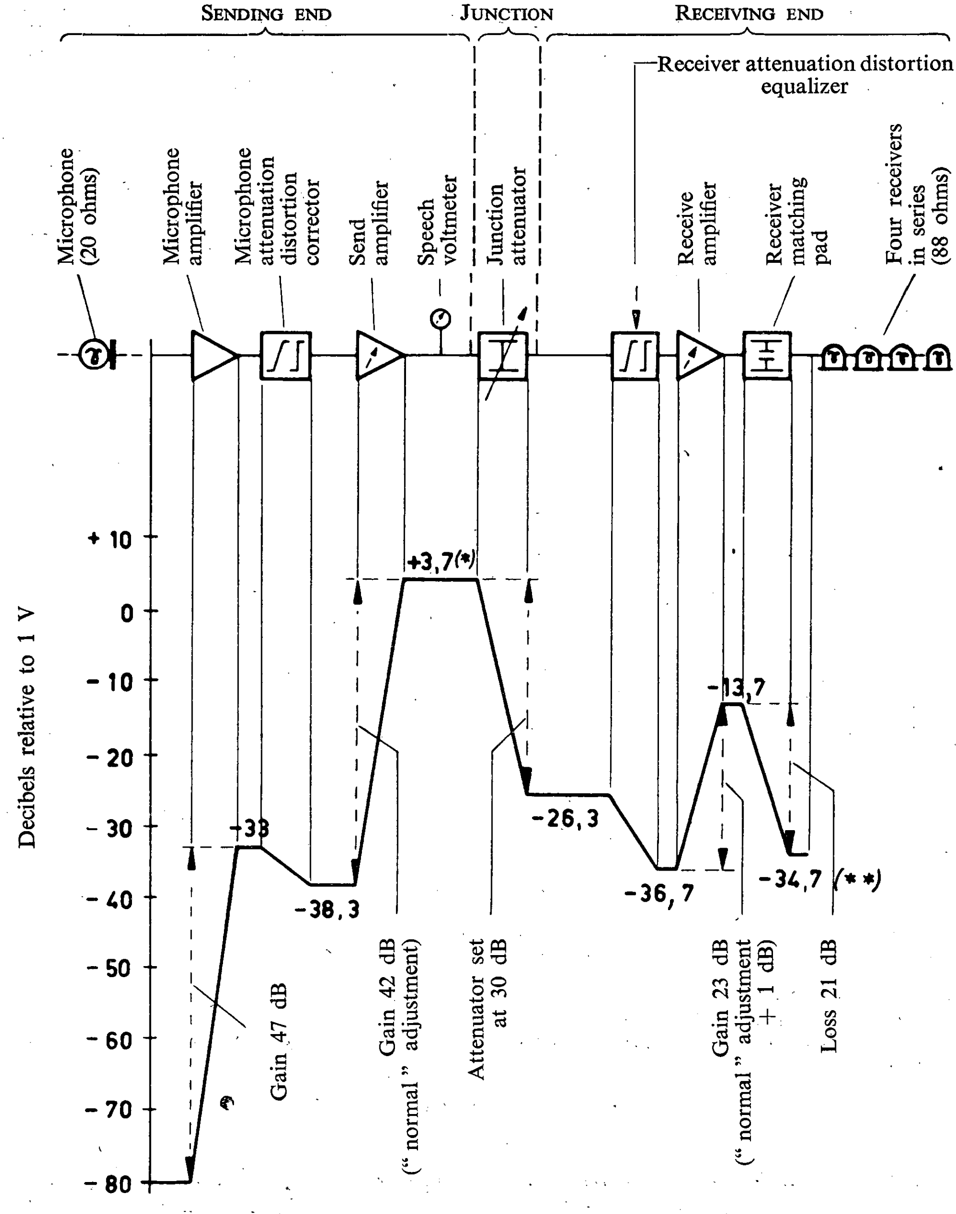

FIGURE 5. — **Diagram showing the levels at various points in the A.R.A.E.N. when a pure tone of 1000 Hz at a level of — 80 dB relative to 1 V is applied to the microphone sockets, in the following conditions of adjustment**

Send amplifier : " normal "

Receive amplifier : " normal " + 1 dB

Junction attenuator : 30 dB

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

** 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	−45.3	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 [1]
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

[1] 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 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 centimetres.

An equalizer defined by Figures 1 and 2 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 of this volume, 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 and 4 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).

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[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 :

1) it corrects the distortion of the A.R.A.E.N. receivers, and
2) 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.

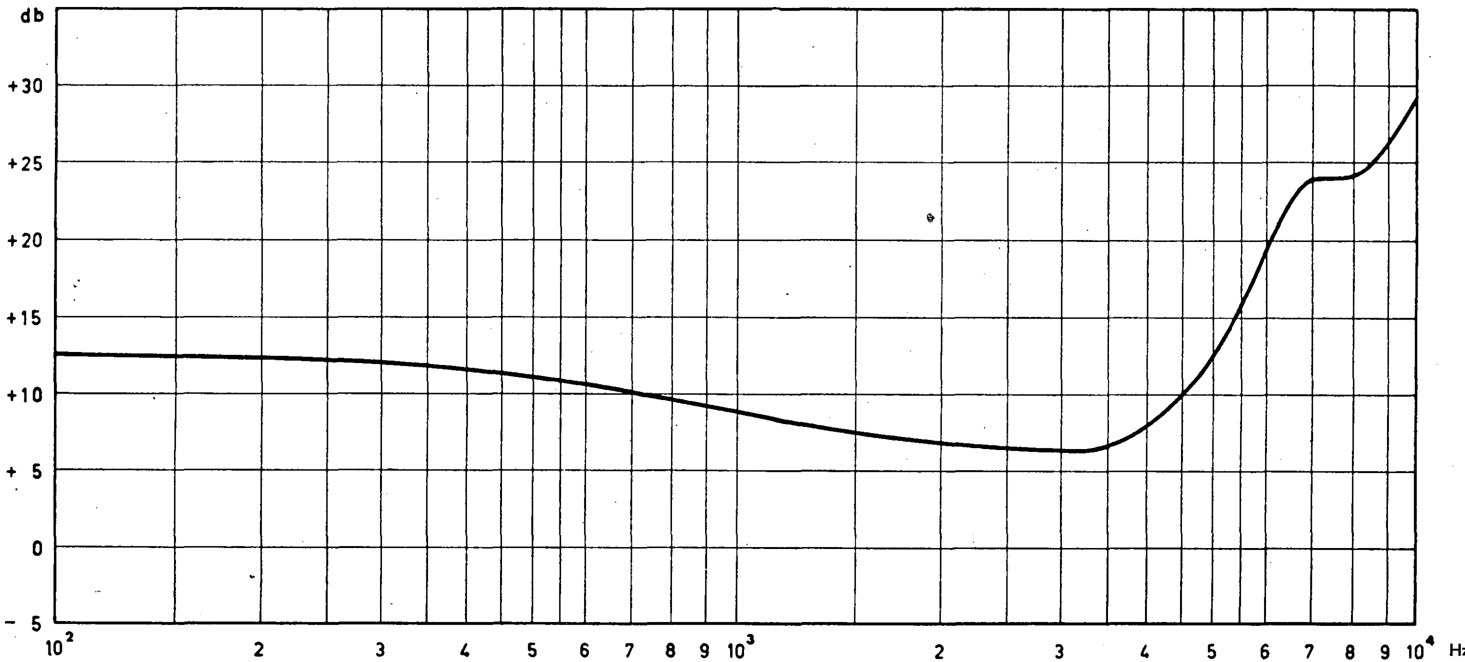


FIGURE 1. — **Insertion loss characteristic of the N.O.S.F.E.R. sending end equalizer (measured between 600-ohm terminations)**

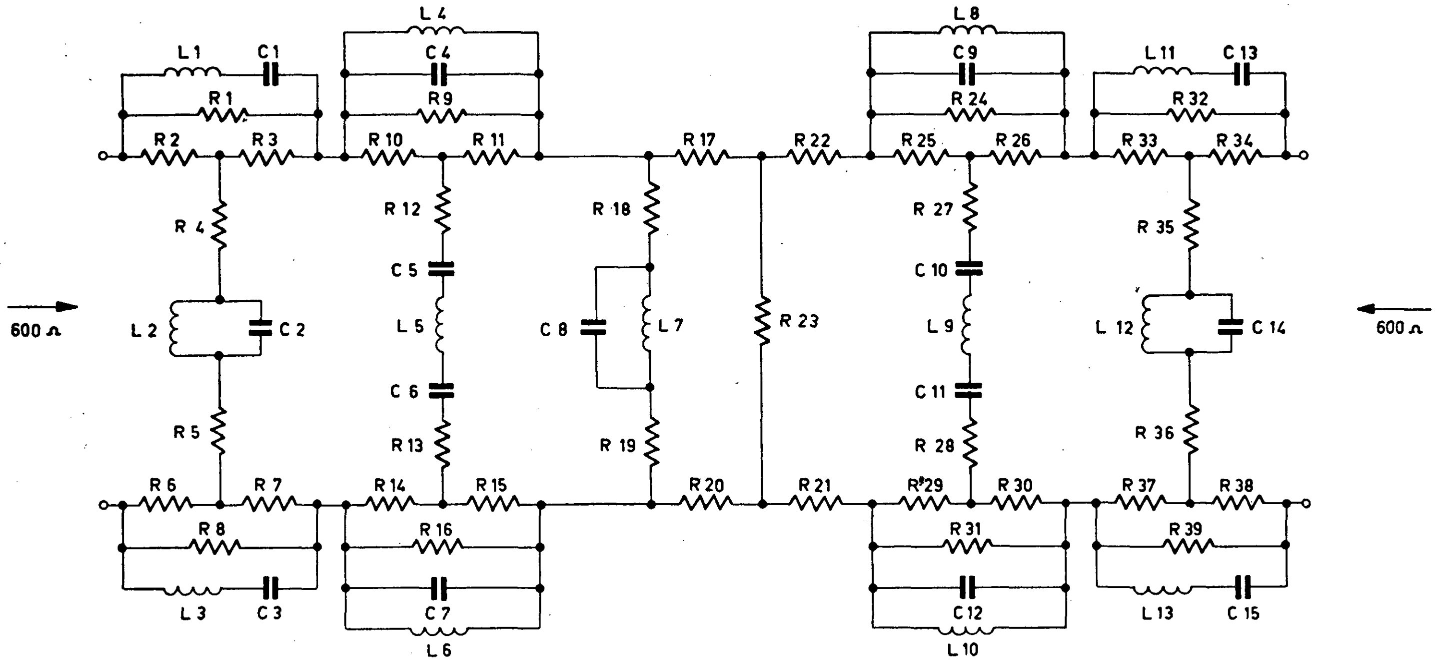


FIGURE 2. — **Circuit of the N.O.S.F.E.R. sending end equalizer**

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 pure 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		

TABLE 2

Values of the components used in the N.O.S.F.E.R. sending end equalizer (Figure 2)

R		L					C	
(non-inductive)								
	ohm		mH	d.c resistance in ohms	“ Q ” 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%

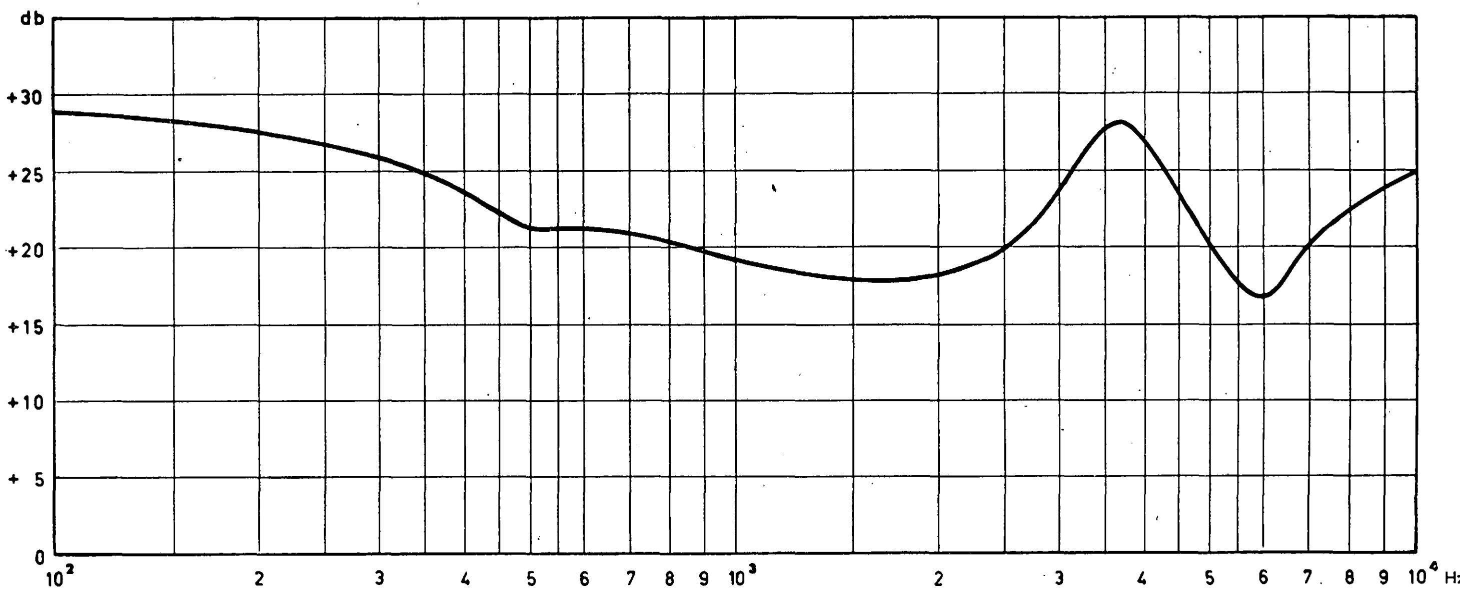


FIGURE 3. — **Insertion loss characteristic of the N.O.S.F.E.R. receiving end equalizer (measured between 600-ohm terminations)**

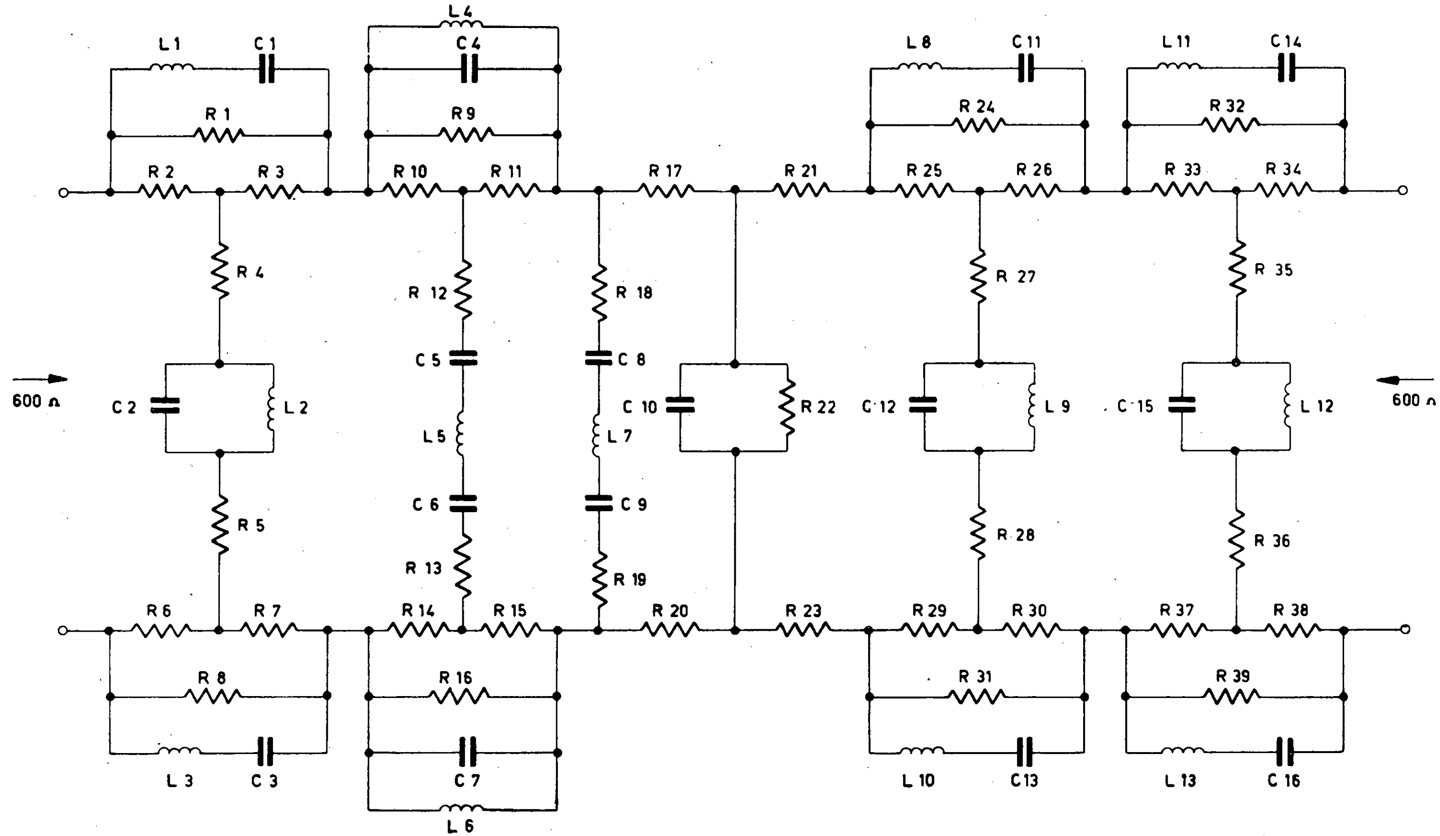


FIGURE 4. — Circuit of the N.O.S.F.E.R. receiving end equalizer

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 pure 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		

TABLE 4

Values of the components used in the N.O.S.F.E.R. receiving end equalizer (Figure 4)

R (non inductive)		L					C	
	ohm		mH	d.c. resistance in ohms	"Q" f_r	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%

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.

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.

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

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

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

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/cm^2) 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	
300	89.0	−81.1	8.0	−89.1	−(−89.2)−89
900	89.0	−83.0	5.8	−88.8	
Average	89.0	−83.1	6.1	−89.2	+0.2

These corrections were determined by the British Administration. 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 * (dB relative to 1 volt/dyne/cm^2) 2	Sensitivity of the sending system in free acoustic field (dB volt/1 dyne/cm^2) (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

* Values extracted from Research Report No. 13200 of the British Administration (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

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		

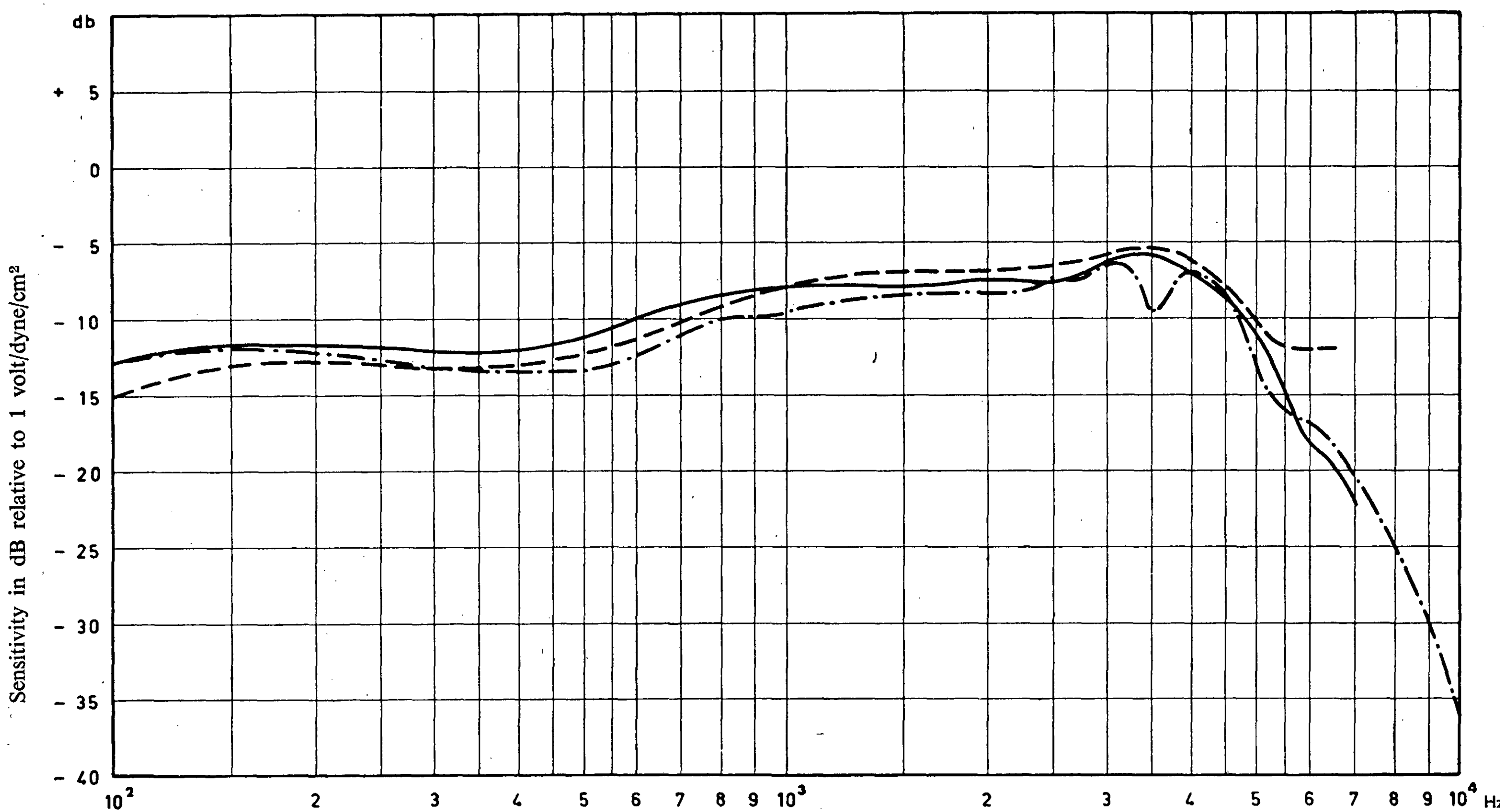
British Administration) and also on a closed coupler. Transmitter amplifier gain is adjusted to the value corresponding to this microphone (" normal " + 0.2).

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 grill 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 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.



——— 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. — **Response curve of the N.O.S.F.E.R. sending end**

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/cm^2) 2	Sensitivity of the sending system in free acoustic field (1+2) (dB relative to 1 volt/dyne/cm^2) 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

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 British Administration.

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/cm^2 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.

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 $\pm$ 0.4 dB in relation to 1 dyne/cm^2 per volt.

TABLE 8

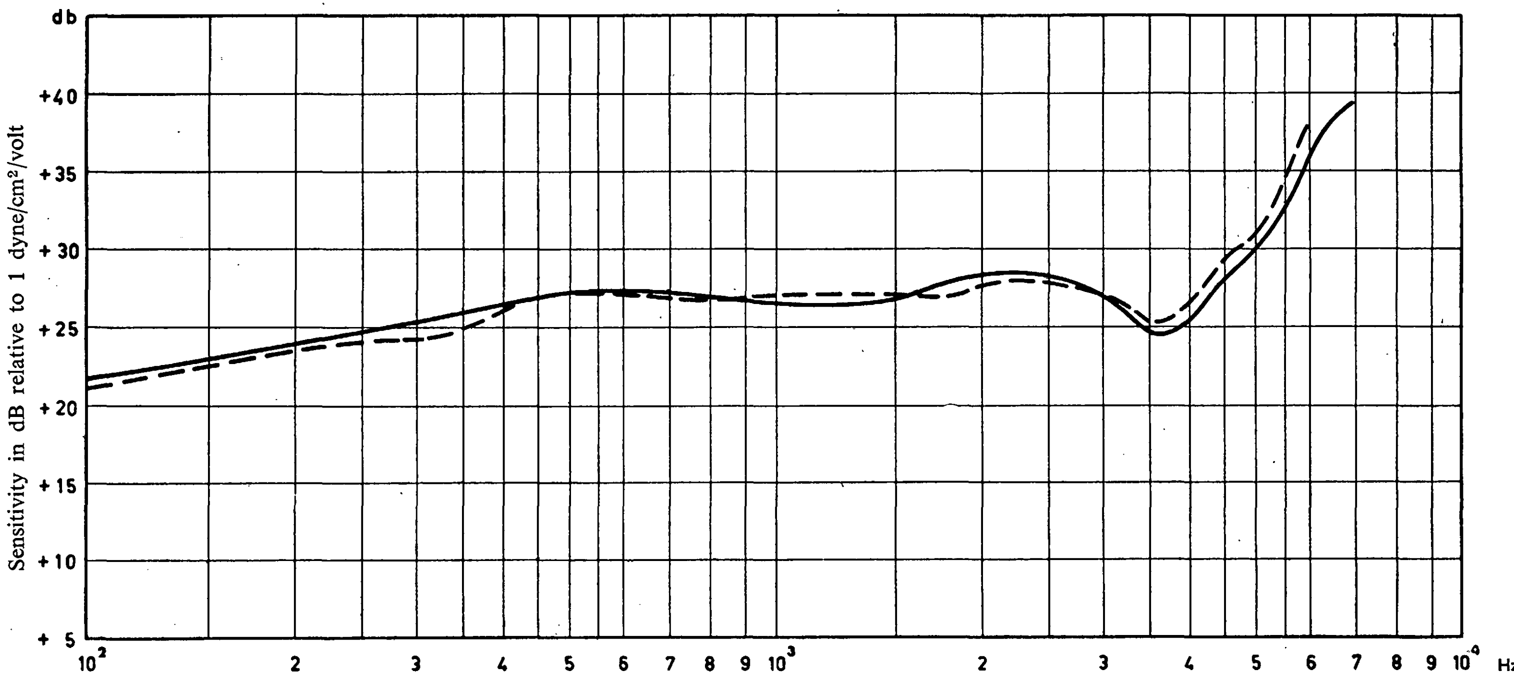
Hz	Gain of the electric part of the sending system (terminated on 88 ohms)	Correction of 12 dB at the output of the impedance adapter of the receivers (four receivers in series)	Average sensitivity −1 dB, of a receiver (db relative to 1 dyne/cm^2/volt)	Nominal sensitivity of the receiving system (dB relative to 1 dyne/cm^2/volt)
	1	2	3	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

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 gives the " sensitivity/frequency " characteristics of the N.O.S.F.E.R. receiver system.

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					+43.5	+43.6	+43.2	+43.5	+25.4	+25.4	+24.3	+24.8	+25.0
	Supplementary attenuator					b=1.5dB	b=1.5dB	b=2.0dB	b=2.0dB					



——— 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. — **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)**

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

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

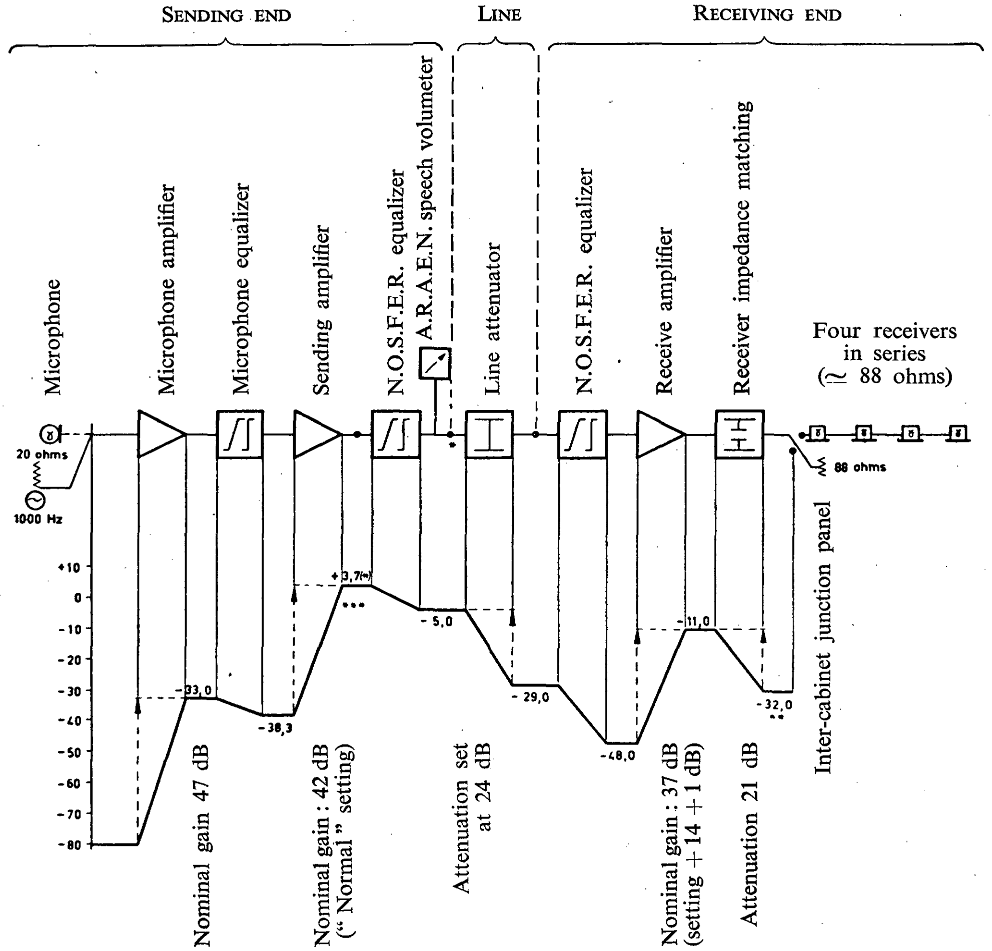


FIGURE 7. — **Level diagram for the N.O.S.F.E.R. for a 1000-Hz tone at a level of −80 dB relative to 1 volt supplied by a generator of 20-ohm internal impedance and applied to the jack of the microphone : the following settings being used :**

send amplifier " normal "

receive amplifier : " 14 dB + 1 dB "

line attenuator : 24 dB

---

* 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.

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

*** 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 of this volume, gain adjustment of the sending amplifier.)

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 decibels 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 ".

Note 1. — In the same conditions, a vu measuring set (see Supplement No. 11 of this volume), 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 of this volume.

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. 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 transmission units (decibels or nepers) 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 length 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 and 9 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 decibels at the C.C.I.T.T. Laboratory (in any case the value of this attenuator should be greater than 15 decibels or 1.8 neper) 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 a " hidden " attenuation; 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 and 9), 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)

Each individual balance is carried out between two operators. The talker repeats

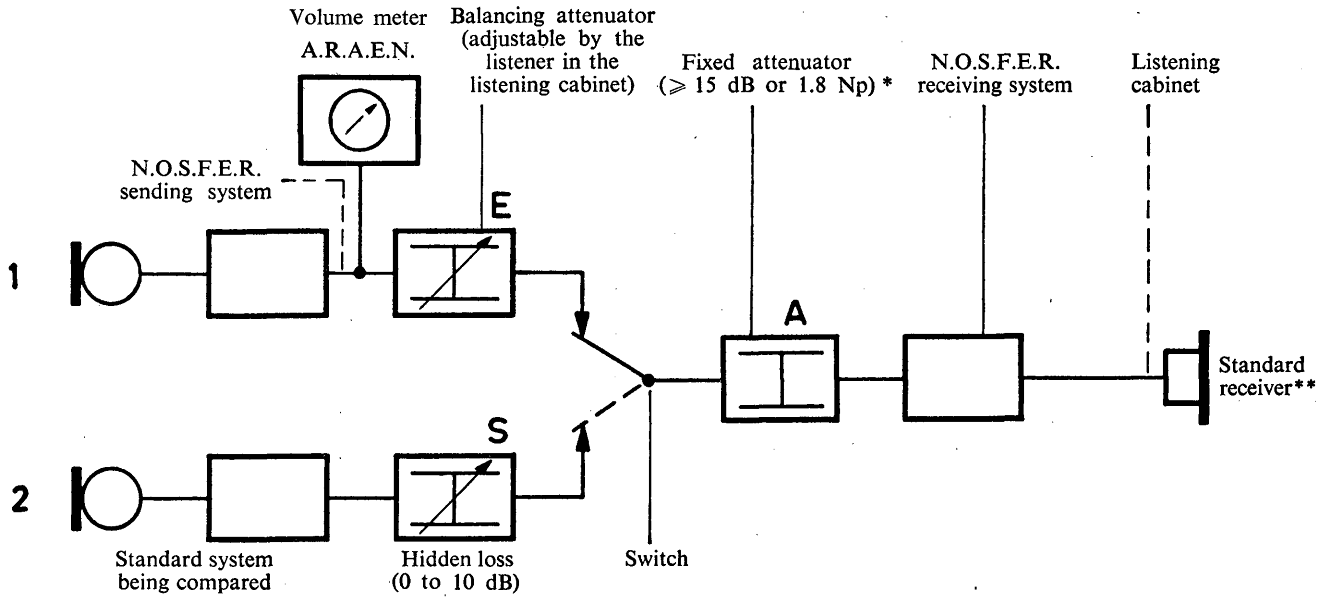


FIGURE 8. — **Comparison of a standard sending system with a N.O.S.F.E.R. sending system (method termed " two-operator-hidden-loss method ")**

* In the C.C.I.T.T. Laboratory, this value is fixed at 24 dB.

** 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.

a predetermined sentence [1] in front of each microphone alternately; the hidden length attenuation is set at a particular value.

In Figure 8 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 " length). 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 for attenuation of the " balancing line ", 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 attenuation of the " hidden length " 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)

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 $S + A$.

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 12 at the C.C.I.T.T. Laboratory with a normal crew of six operators which 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 length 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

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

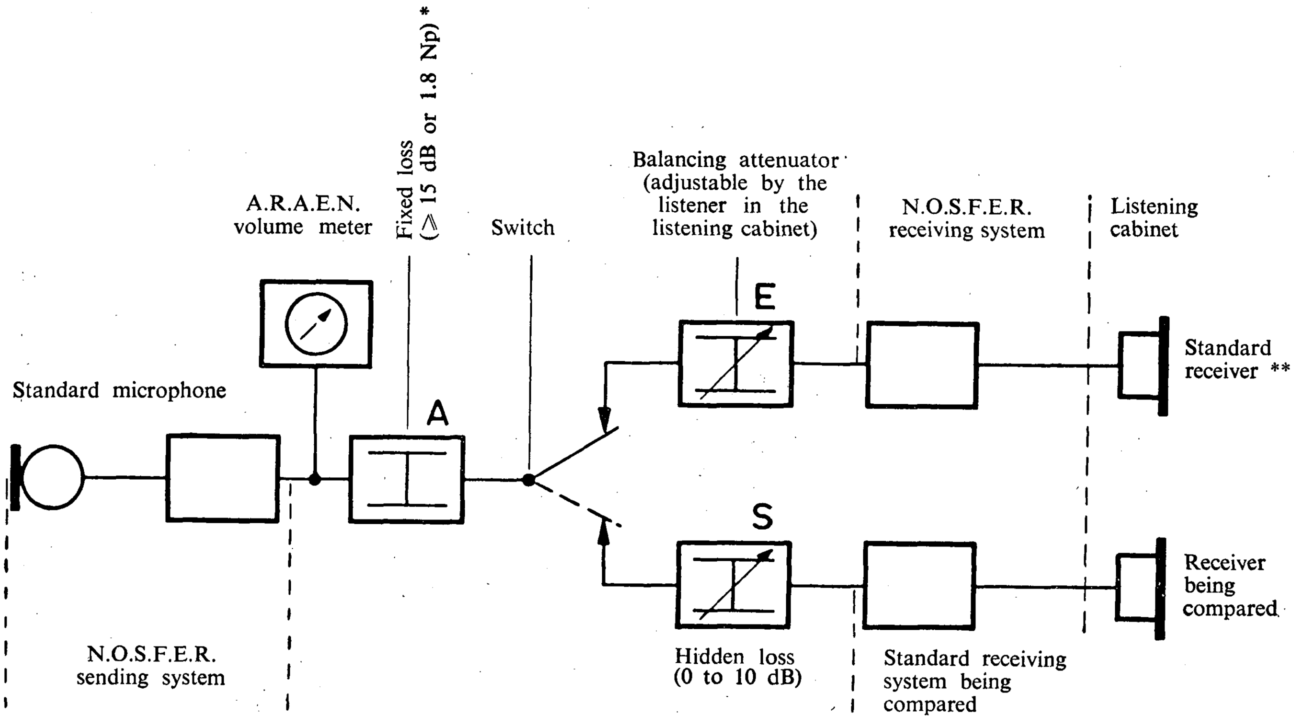


FIGURE 9. — **Comparison of a standard receiving system with a N.O.S.F.E.R. receiving system (method termed " two-operator-hidden loss method ")**

*
** } (See footnotes on Figure 8).

analysis are sent to administrations and recognized private operating agencies 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 decibels or 1.8 neper 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 and private operating agencies 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 electro-dynamic microphone and receiver (S.E.T.E.D.)

Even in a working standard with modern instruments (S.E.T.A.B.), the stability obtainable is less than is desirable and there remains the necessity of calibrating the standards by periodical comparison with the N.O.S.F.E.R. or another system of greater stability.

A working standard has therefore been designed to use a moving coil microphone and receiver furnishing a high-quality circuit with stable characteristics.

The system has been designed for high-quality speech because it is considered that the standard with which commercial systems are compared should always be better in performance than the commercial systems tested; there is also another reason, namely the provision of a stable system with which to compare commercial systems on an articulation basis since this seems likely to become the basis of transmission planning instead of planning on the loudness basis of reference equivalents.

The new working standard is provided with means for absolute calibration of its microphones and receivers. It is sufficiently simple to be installed in many laboratories, and will serve as a primary standard on the spot, making it seldom necessary for those that have this working standard to send transmitters and receivers elsewhere for calibration.

The necessary equipment is available to check the acoustic characteristics of the complete system.

The S.E.T.E.D. working standard comprises:

1. a rack with 19-inch panels of the following dimensions:

6 ft. 5½ in. high

2 ft. 2 in. wide

2 ft. 6 in. deep

On this rack are mounted a vu meter; a microphone amplifier and its equalizer; a control panel which contains an attenuator; a filter; push buttons and relays to change the test circuit and jacks for external connections; a receiver amplifier with equalizer and a mains-operated power unit.

2. a standard piezo-electric quartz microphone for objective calibration of the transducers (microphones and receivers);

3. a moving coil microphone and a receiver;

4. accessories.

The working standard includes a volume meter of the vu meter type for the adjustment of the vocal power of the talker and also as an indicating voltmeter for objective calibrations.

An oscillator for calibration over the range 100 to 5000 Hz is required; this is not included as part of the equipment.

The moving coil microphone is of a special type designed for close-talking. It is substantially protected against the effects of breath moisture.

To standardize the lip position a guard-ring is fitted to the microphone, which is mounted on an adjustable stand.

The frequency characteristic of the microphone and its equalizer is such that the ratio of its output voltage to the sound pressure at the speaker's lip position is constant over the working frequency range.

The receiver is furnished with an equalizer such that the ratio of the sound pressure in a human ear to the voltage across the input of the equalizer varies as a function of frequency according to the " orthotelephonic " response curve.

The gain of the amplifiers is adjustable so as to obtain normal values of 62.1 decibels for the microphone amplifier and 17.5 decibels for the receiver amplifier.

The calibration of an amplifier is carried out by comparing its gain against the loss of a pad of fixed resistors using a single-frequency signal from an external oscillator and the volume indicator as a meter indicating equality. The amplifier has generous negative feedback and there is ordinarily no need to specify stabilized mains supply.

A band-pass filter, 300-3400 Hz, is introduced to restrict the transmission to a band similar to that of an ordinary telephone circuit. This is necessary because the new working standard in fact transmits a wide frequency band extending from 50-9000 Hz and is in this form unsuitable for comparison with circuits of narrow band width.

The piezo-electric quartz microphone used for calibration is described below. The moving coil microphone calibration consists of a direct comparison between its output voltage and that of the quartz crystal assembly, which is not frequency-dependent. The moving-coil receiver is calibrated by measuring the voltage fed to it in order to excite the quartz crystal assembly to a constant output voltage, i.e. to produce a constant sound pressure.

The quartz crystal is associated with a three-stage battery-operated pre-amplifier followed by the 62.1 dB amplifier already mentioned. The volume meter is used throughout the calibrations to indicate equality of voltage only.

The gain-frequency characteristic of the pre-amplifier is automatically eliminated from the calibration because the calibrating signal is injected directly in series with the crystal element. The only " unknown " in the system is therefore the piezo-electric activity, and this can be calculated from the physical dimensions and first principles.

The standardization of the pre-amplifier, calibration of microphone and receiver and setting-up of other combinations of circuits are easily made by a switch which operates the appropriate relays. These relays are fitted with platinum or gold contacts in all circuits which carry speech currents. Other switches operate the other relays necessary to set up the test circuits, adjust the gains of the amplifiers and set up the complete system as required for tests based on comparisons of volume or for A.E.N. measurements.

In the microphone and receiver calibrations, the attenuator is connected in the circuit limb containing the transducer under test. This attenuator is used to bring the two readings of the volume meter to equality so that the attenuator settings are a direct measure of the electro-acoustic calibration in question.

Some features of the acoustic calibrations are:

1. The relationship between the sound pressure applied to the crystal microphone and its output voltage is independent of frequency and temperature.

2. The acoustic medium is air at the pressure and temperature of normal operation.

3. A few measurements have been made to ascertain the extent to which the moving coil microphone and receiver and the calibration equipment are affected by temperature. The entire calibrating equipment and moving-coil microphone and receiver were raised from 16° to 40°C. The microphone sensitivity fell generally about 0.7 dB. The receiver sensitivity was practically unchanged except for an octave either side at the diaphragm resonance frequency, the maximum change being a fall of under 2 dB. Both calibrations returned to their original values when the apparatus was restored to room temperature.

4. The acoustic couplers are small enough to ensure uniformity of sound pressure.

5. The calibration measurements are repeatable to ± 0.5 dB (or better in the middle range).

6. The receiver coupler is in the form of an artificial ear having an acoustic impedance compounded of stiffness and acoustic resistance terms.

7. The microphone is calibrated at constant pressure independent of frequency. This calibration curve is related to the free field response mentioned earlier by means of a correction depending on the geometry; the correction is established once for all by means of a free field calibration measured with an artificial mouth in a room of controlled acoustics.

The S.E.T.E.D. working standard has been designed with a view to its use as a reference system for articulation tests.

It may also be employed with loudness balancing for the determination of reference equivalents.

In both cases appropriate circuits can be set up for determining the relative equivalent (or the A.E.N. value) of a commercial telephone system for receiving, sending, or for the complete system.

The reference equivalents of the S.E.T.E.D. for sending, receiving and overall obtained by direct determinations made in the C.C.I.T.T. Laboratory are as follows:

the sending reference equivalent of the S.E.T.E.D. is 3.3 dB better than the S.F.E.R.T. (1953);

the receiving reference equivalent of the S.E.T.E.D. is 1.1 dB better than the S.F.E.R.T. (1953);

the sending reference equivalent of S.E.T.E.D. is 0.1 dB better than a N.O.S.F.E.R. (1967).

the receiving reference equivalent of S.E.T.E.D. is 2.8 dB better than N.O.S.F.E.R. (1967);

the overall reference equivalent of S.E.T.E.D. is 4.5 dB better than N.O.S.F.E.R. (1967).

The A.E.N. values have also been determined by comparison with the A.R.A.E.N. for the complete system, for the sending system and for the receiving system:

the sending A.E.N.	is 3.3 $\pm$1.9 dB better than the A.R.A.E.N.,
the receiving A.E.N.	is 10.2 $\pm$2.3 dB better than the A.R.A.E.N.,
the overall A.E.N. of the complete system	is 17.3 $\pm$2.2 dB better than the A.R.A.E.N.

The S.E.T.E.D. is provided with arrangements at the input of the receive amplifier to allow noise to be injected or sidetone to be provided, but the values of A.E.N. given above have been determined with airborne room noise which was used with the A.R.A.E.N. at the period when these tests were made.

Calibration of the S.E.T.E.D. depends upon the quartz crystal microphone which uses the direct piezo-electric effect in a stack of X-cut quartz crystal (see Figure 10); one face of each crystal element is exposed to sound pressure, the opposite face is securely attached to a massive block. This ensures that the driven face is stiffness-controlled up to the resonant frequency of the combined mechanical system (estimated at 20 kHz). Six similar crystal elements are stacked with alternately poled electric faces adjacent. The advantage over a single rectangular block of the equivalent size is that the source impedance is lower. This facilitates design of the valve input stage. An incidental advantage is a small improvement in the signal to noise when the limiting factor is thermal noise in the associated grid leak resistor.

It is imperative in using the direct action principle with quartz to ensure that the acoustic drive is limited to a single surface. At the same time the crystal assembly must be free from mechanical constraints so that its sensitivity can be calculable from first principles. The design adopted meets these requirements, a non-hardening compound being used to seal an air gap between the crystal and the surrounding metal. This prevents the access of sound to the side faces of the crystal assembly without constraining the motion.

The crystal having been very carefully selected and prepared, it is possible to calculate the microphone sensitivity on the reasonable assumptions of crystal homogeneity and of simple compressional stress parallel to the Y = axis which is uniform and equal throughout the crystal to the applied acoustic pressure, i.e.

$$\frac{e}{p} = \frac{4 \pi L_1 d_{21}}{n K_{11}} \times 300$$

where e = open circuit e.m.f. in volts between adjacent crystal interfaces;

p = sound pressure in dynes per cm^2;

n = number of crystal sections;

K_{11} = dielectric constant of quartz in direction of X-axis (4.55);

L_1 = dimension of crystal stack parallel to X-axis (in cm) (see Figure 10);

d_{21} = piezo-electric constant of quartz relating compressional stress parallel to Y-axis to polarization parallel to X-axis. The units are e.s.u. charge per dyne (6.9×10^{-8}).

It is seen that the quantity $\frac{e}{p}$ is independent of frequency, so that the frequency response is truly flat. This gives the microphone its useful character as a standard.

The source impedance of generated e.m.f. is a capacitance where

$$C = \frac{1}{9 \times 10^{11}} \cdot \frac{n^2 K_{11} a}{4 \pi L_1} \text{ farads}$$

where a = area of the part of the crystal face (perpendicular to the X-axis) covered by metal, slightly less than the area L_2 L_3 of the crystal face.

Upper limits to the sensitivity $\frac{e}{p}$ and to C are set by the following requirements:

1. the resonant frequency of crystal system must be higher than the highest frequency in use (this determines L_2);
2. the linear dimensions of the exposed face must be small compared to the wavelength of the sounds in air (this determines L_1 and L_3).

The values adopted are

L_2 = 3.92 cm

L_1 = L_3 = 1.34 cm

a = 4.77 cm^2

n = 6

from which the sensitivity is calculated at

$$20 \log_{10} \left(\frac{e}{p}\right) = -97.9 \text{ dB rel. 1 volt per dyne/cm}^2,$$

and the capacitance C = 51.5 pF.

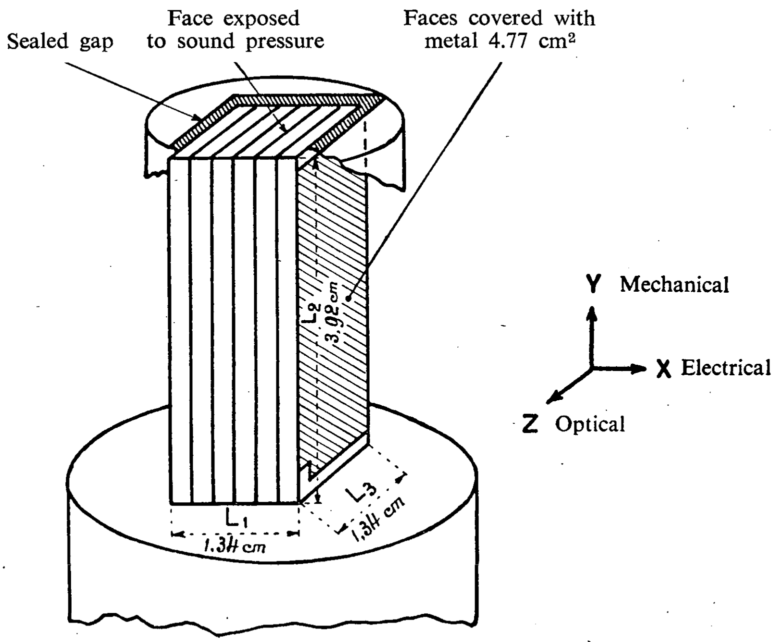


Figure 10

Response of the pre-amplifier is maintained down to about 100 Hz by using a 100-megohm grid leak resistor and Type 155 amplifying pentode input stage with underrun heater, having an input capacitance about 4 pF. The exact circuit constants however are unimportant because they are eliminated in the inject calibrating method, which introduces a known e.m.f. directly in series with the crystal.

Secondary calibrations of the quartz microphone have been made using probe microphones coupled to it by a specially designed fixture. The three probe microphones used had been compared directly with primary standards. The deduced quartz microphone sensitivities were −98.2 −98.0 and −98.4 dB rel. 1 volt per dyne/cm^2, compared to −97.9 dB from calculation.

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 $\pm L_p \%$.

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 analysed.

(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 (Part III of the present volume).

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 or recognized private operating agencies 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 or a private operating agency 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 of this volume), 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 or private operating agencies 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 or private operating agency 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 or private operating agency 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 systems concerned comprise 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 or private operating agency 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 periodical re-calibrations by means of reference equivalent determinations where the object is mainly to verify the stability of the microphones and receivers the administration (or private operating agency) 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 or private operating agencies 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 paragraphs A and B

The object of the general recommendations given above is to guide administrations or private operating agencies. When an administration or private operating agency 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 or private operating agency 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) at a psophometric e.m.f. of 2 mV.

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

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

The band-pass filter has cut-off frequencies of 300 and 3400 Hz; it stimulates 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). For frequencies

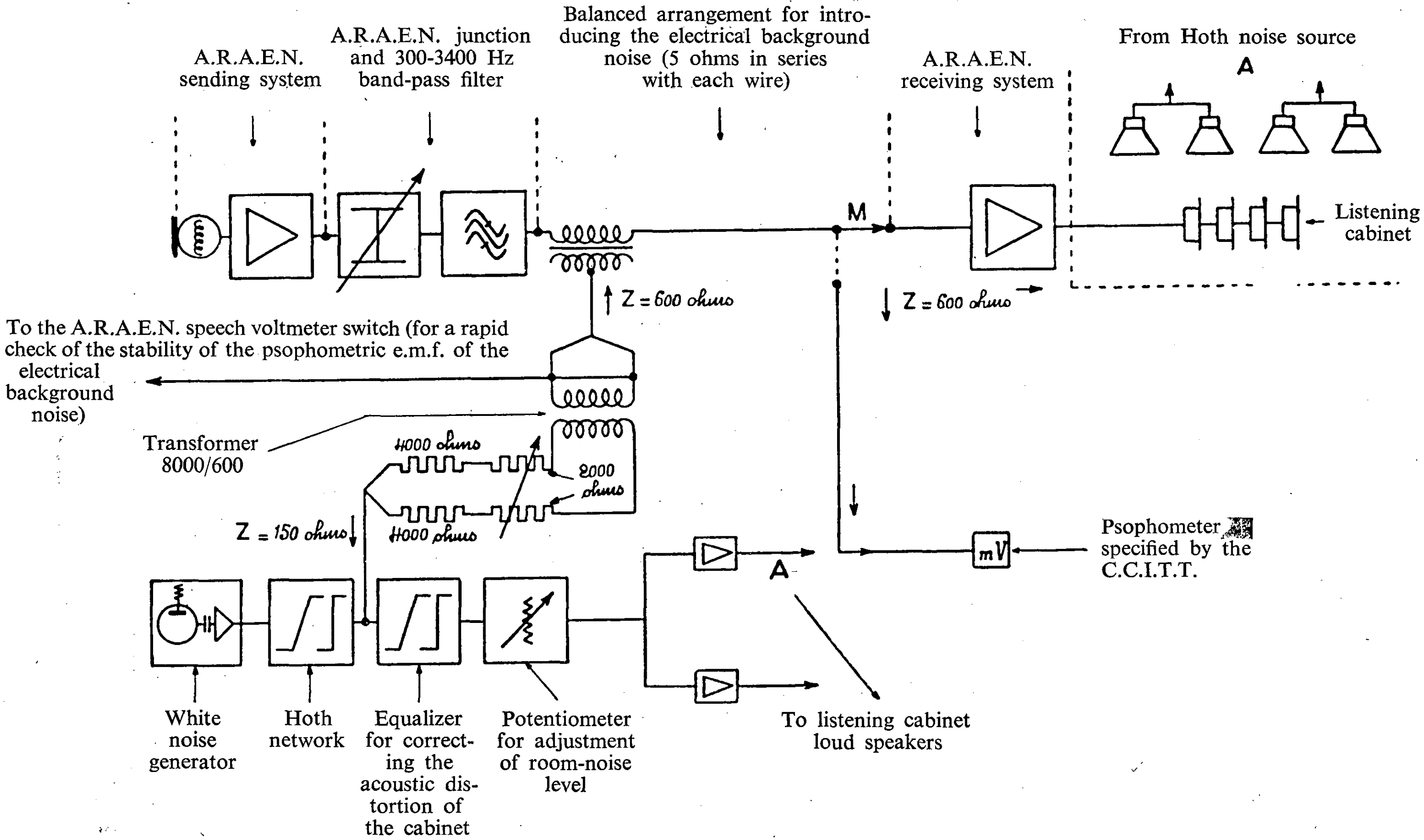


FIGURE 1. — **Schematic diagram of the S.R.A.E.N. including the arrangement for injecting the " electrical background noise " into the A.R.A.E.N., and the measurement of the psophometric voltage of this noise**

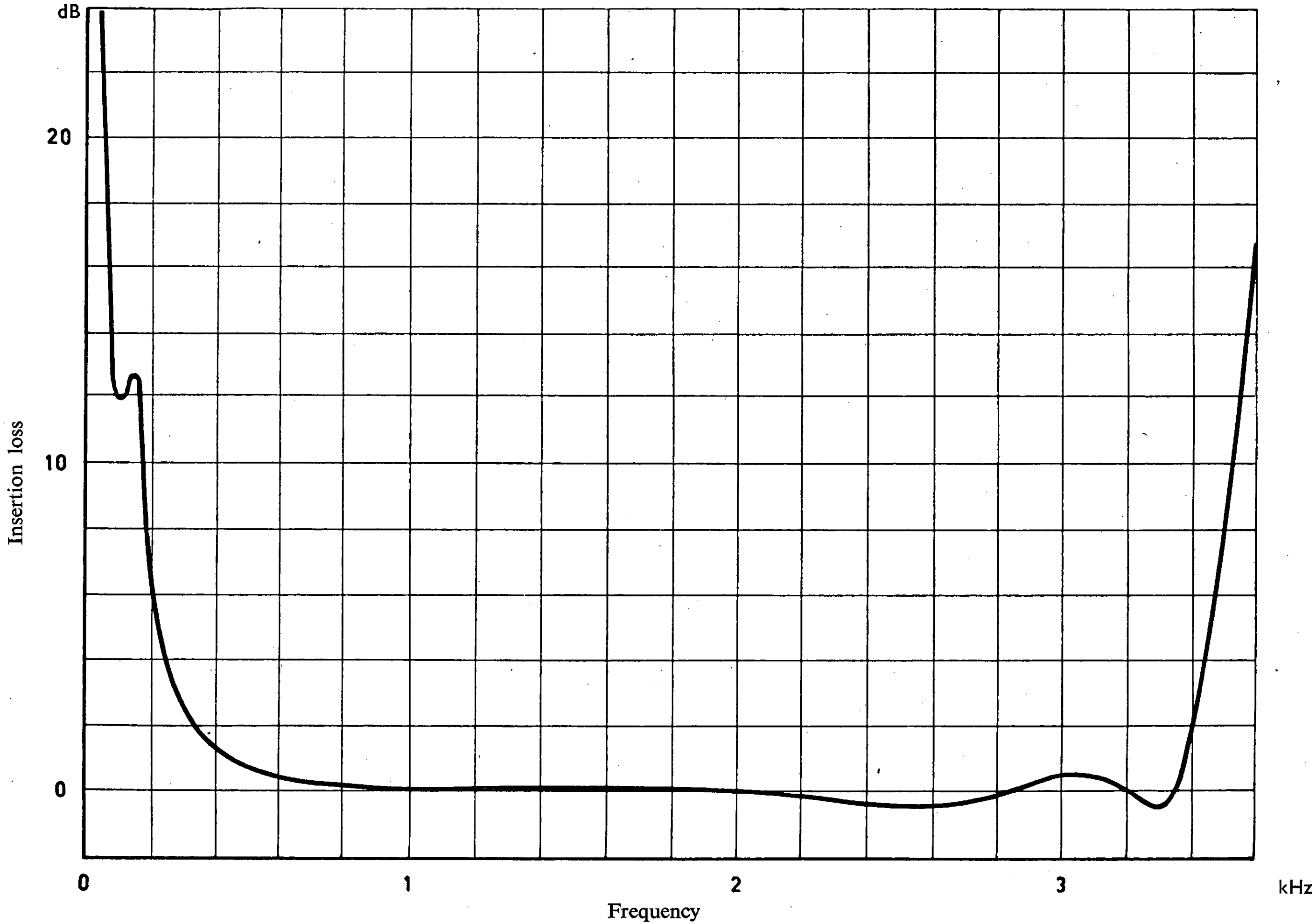


FIGURE 2. — **Insertion loss (between 600-ohm terminations) of the 300-3400-Hz band-pass filter**

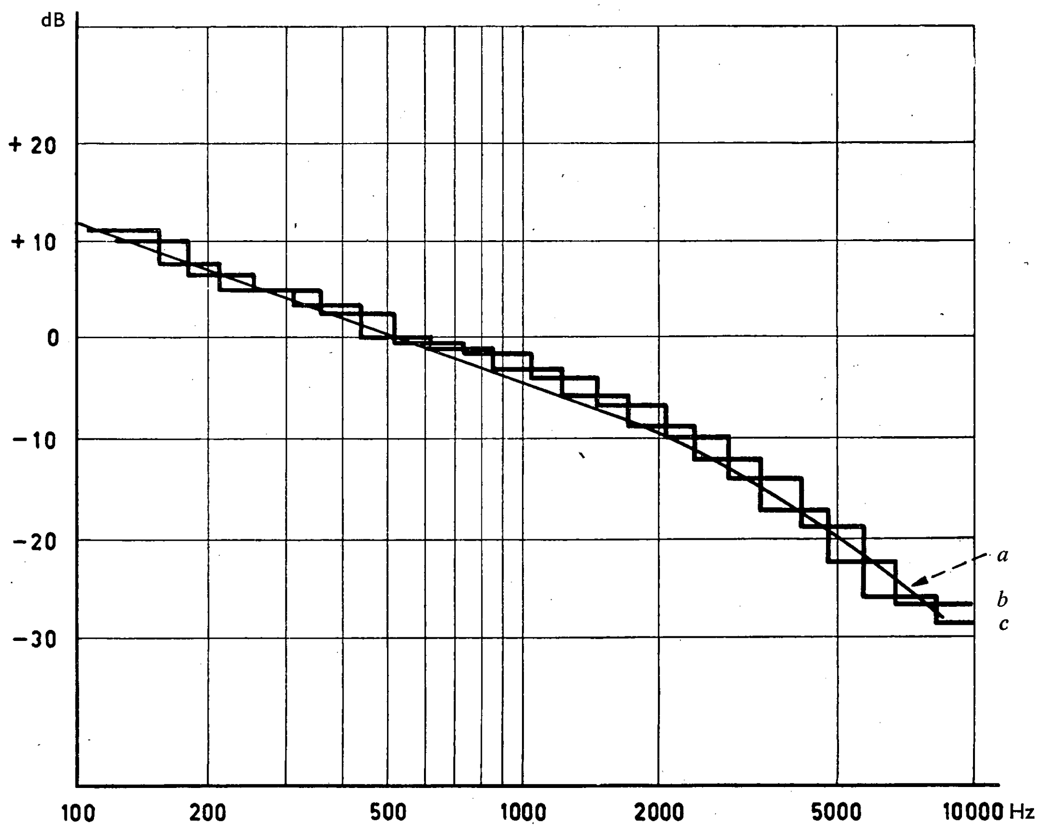
S.R.A.E.N.

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 gives the mean power density spectrum observed at subscriber's 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 and private operating agencies can 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.



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

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

FIGURE 3. — **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 or private operating agency 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 of this volume) 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 δ. 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 or private operating agency 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 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*

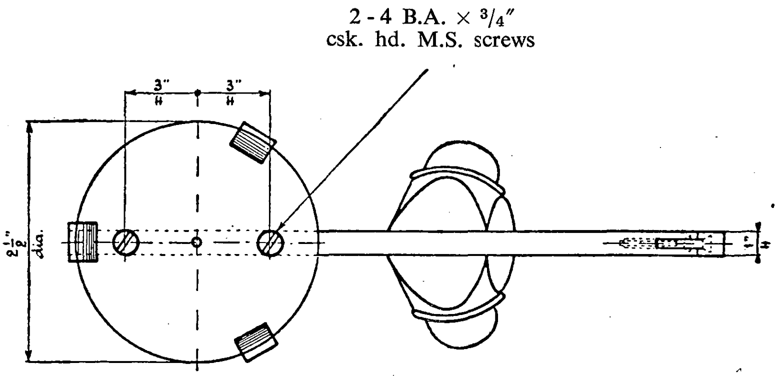
Before each talk and after the handset has been fixed in its support in the appropriate manner, the feeding current is applied and the microphone is rotated gently, once forward and once back, about $^3/_4$ of a circle and is then fixed in position while avoiding any mechanical shock.

(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 an " 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 the " 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* (the case of circuit 3, see Introduction above), 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 which also shows the spectral distribution of a typical room



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

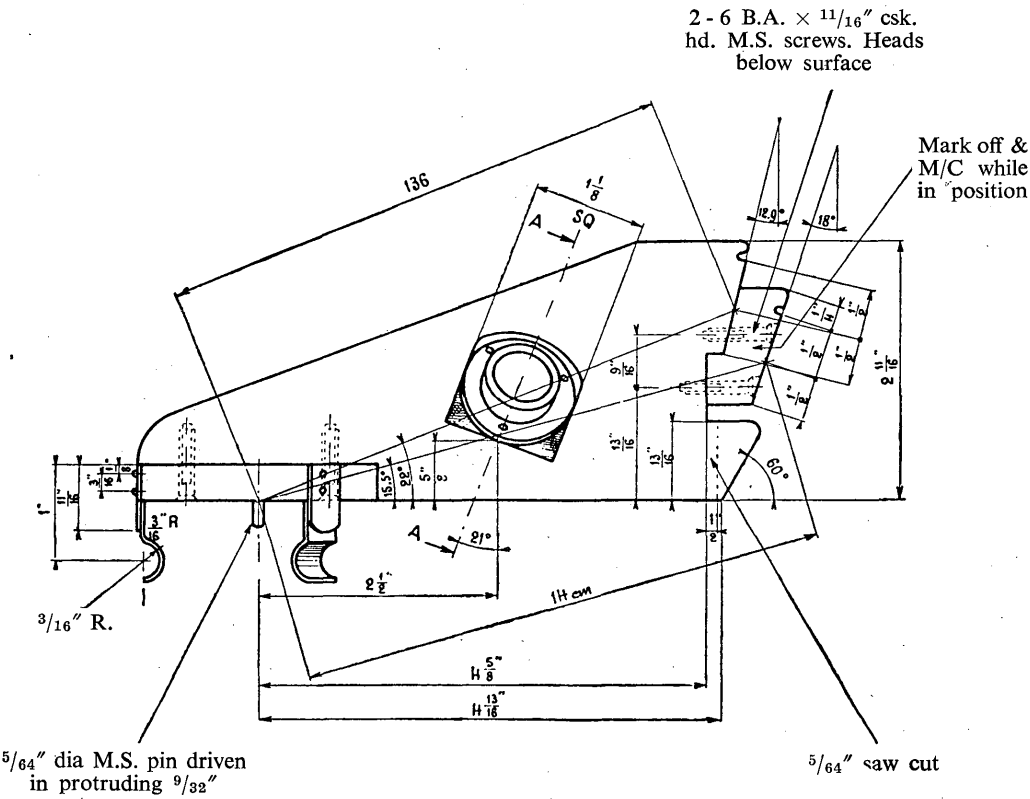


FIGURE 1. — **Type of gauge used for setting the handsets in articulation tests at the C.C.I.T.T. Laboratory**

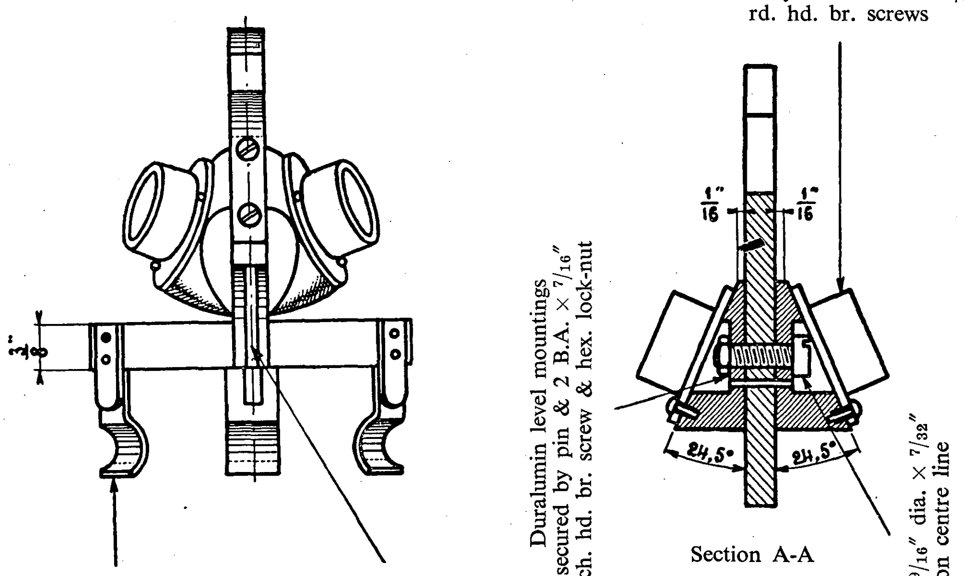


FIGURE 1 (contd.)

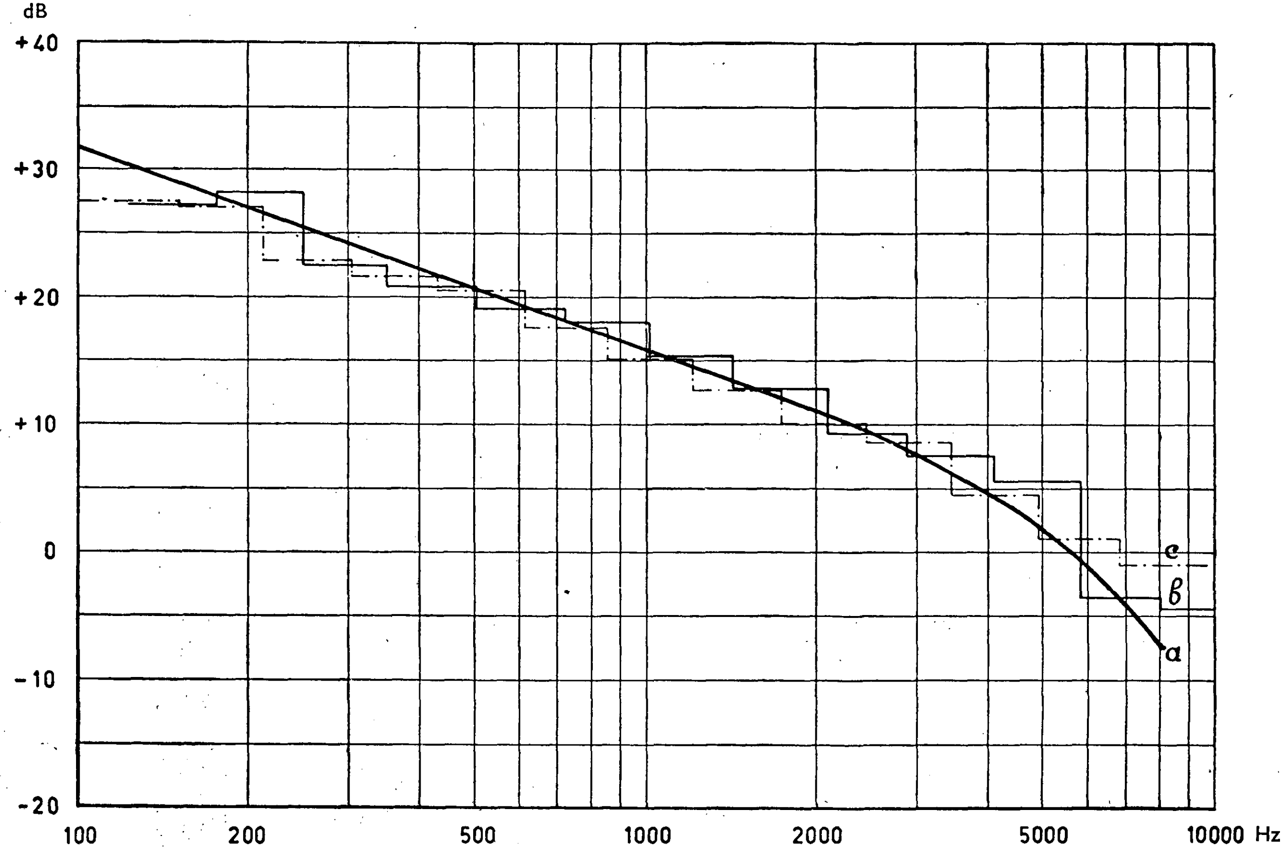
noise measured in the listening cabinet of the C.C.I.T.T. Laboratory; graphs *b* and *c* represent respectively the results of measurements on this noise made with two sets of half-octave filters.

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 decibels 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 subscriber's stations (published by Hoth)

b, *c* Typical graphs obtained at the C.C.I.T.T. Laboratory with two sets of filters

FIGURE 2. — **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).

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 sidetone), 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.

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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

SECTION 5

OBJECTIVE MEASURING APPARATUS

RECOMMENDATION P.51 (amended at Mar del Plata, 1968)

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.,

provisionally recommends the use of the artificial ear described in section B of this Recommendation.

Note 1. — The C.C.I.T.T. cannot issue a definitive Recommendation concerning this artificial ear until it has been finally standardized by the I.E.C.

Note 2. — 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 3. — Telephone administrations and private operating agencies can, if they wish, use in the future devices which they may 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 4. — 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 mean human ear, for use in laboratory.

Artificial ear type 1 (or reference coupler) is the subject of I.E.C. publication ...; 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 mean human ear, after which the Working Group defined an electrical network equivalent to the mean 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. specifications [1], 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 mean 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

[1] *Note by the C.C.I.T.T. Secretariat* — The Plenary Assembly at Mar del Plata had before it I.E.C. Document 29C (Central Office) 3, which was circulated in September 1968 for approval under the six-months rule. The Secretariat has replaced the extracts from this document by the corresponding passages of the approved text, which will be issued as an I.E.C. Publication with the name *An I.E.C. artificial ear, of the wideband type, for the calibration of earphones used in audiometry.*

Figure 1. The lumped parameter values of the coupling elements shall be adjusted as follows:

$$L_2 = 5 \times 10^2 \text{ Ns}^2\text{ m}^{-5}$$
$$L_3 = 1 \times 10^4 \text{ Ns}^2\text{ m}^{-5}$$
$$R_2 = 6.5 \times 10^6 \text{ Ns m}^{-5}$$
$$R_3 = 2 \times 10^7 \text{ Ns m}^{-5}$$

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.

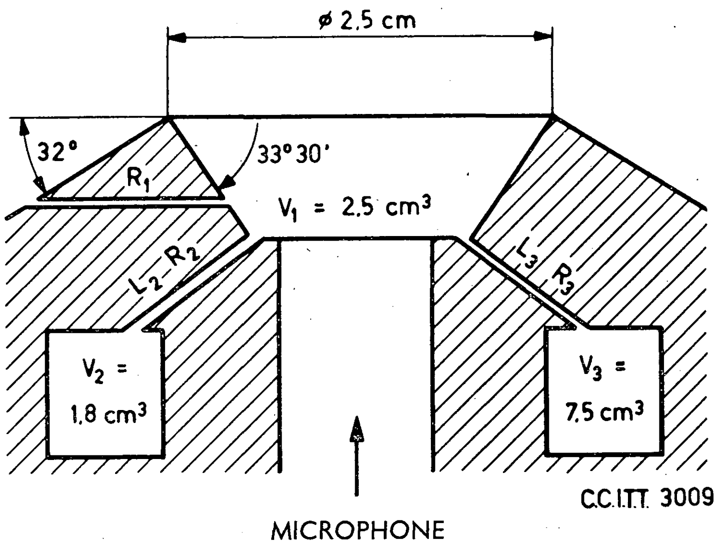


FIGURE 1

(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 ± 1% and the magnitude of the coupling elements within ± 5%. The angular dimension 33°30′ shall have a tolerance of ± 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 $\pm$ 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*

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 an 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 overall stability in an instruction manual.

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.

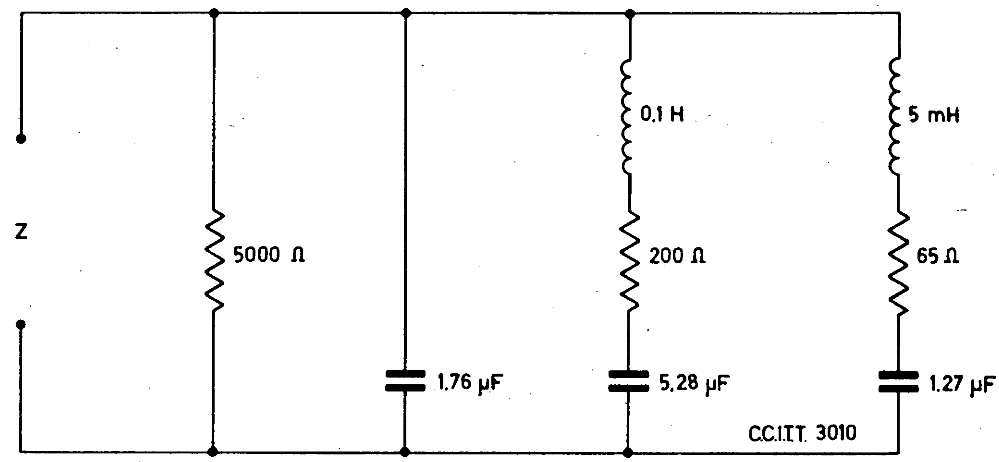
Ithell, A. H. : A determination of the acoustical input impedance characteristics of human ears; *Acustica 13* (1963), 311.

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 a

Lumped-parameter electrical network analogue of the artificial ear

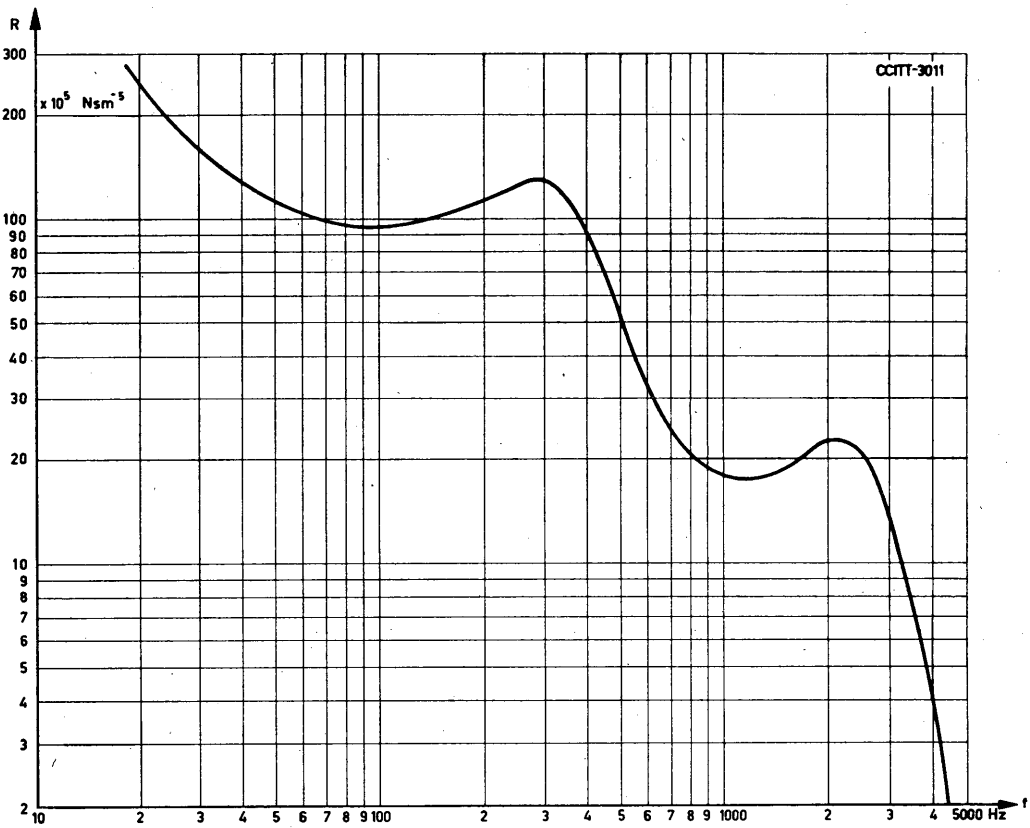
In this analogue, one electrical ohm corresponds to 10^5Ns m^{-5}.



Three independent determinations of the acoustical impedance of the mean human ear under no-leak conditions were available (see bibliography) covering various earcap contours used on audiometric earphones. In each case an analogue network of the type shown in the figure 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 the figure are average values corresponding to a plane earcap.

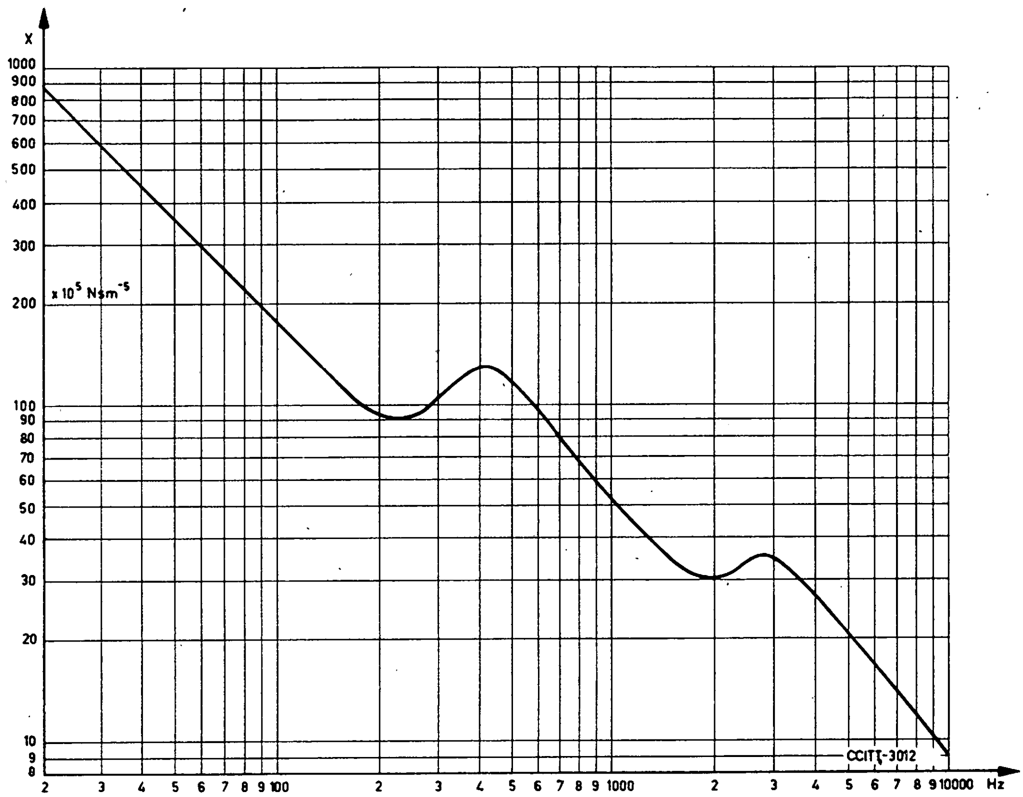
ANNEX 1 b1

Real component of the impedance of the electrical analogue network



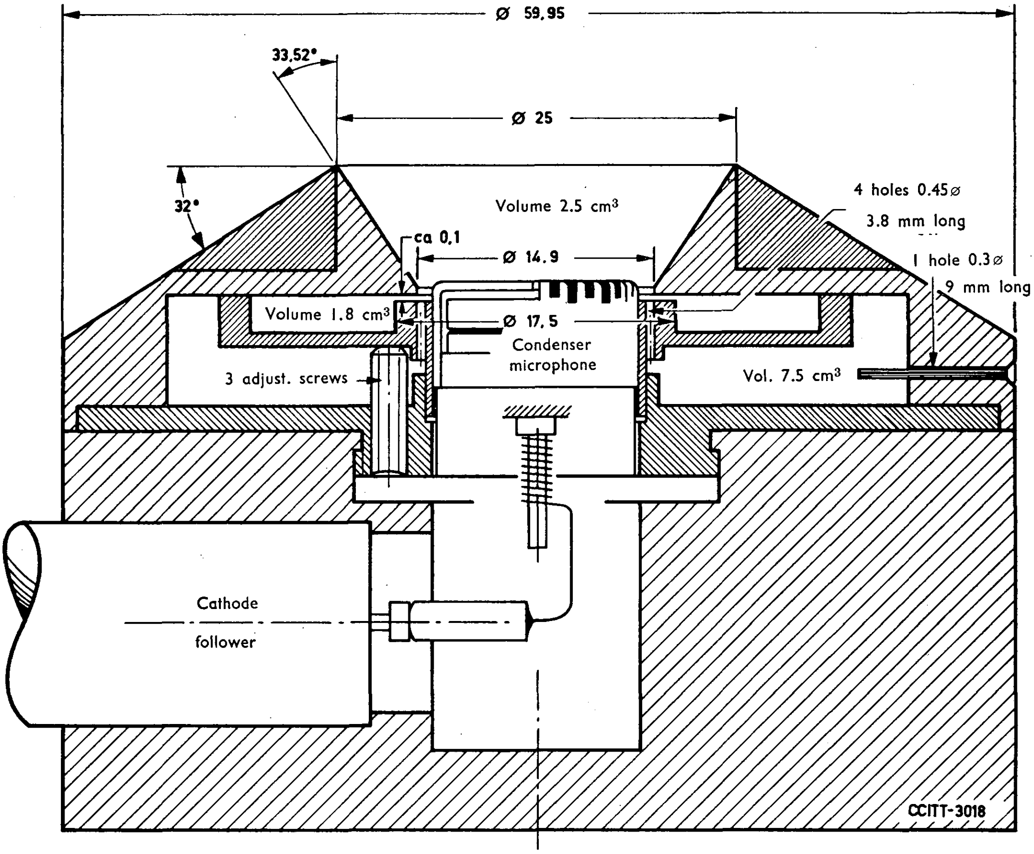
ANNEX 1 b2

Imaginary component of the impedance of the electrical analogue network



ANNEX 2

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 keeping a watch on the volume or peak values during telephone conversations or radio broadcast transmissions.

Principal characteristics of the various instruments used for monitoring the volume or peaks during telephone conversations or programme 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 " British 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 "	2	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 type B sound meter		less than 300 ms for meters with pointer indication and less than 150 ms for meters with light indication	10±5 60±10	1.5 to 2 seconds from " 0 dB " point at 30% of the length of the operational section of the scale

Note. — Descriptions of the following devices are contained in the Supplements to this Volume :

A.R.A.E.N. volume meter or speech voltmeter : Supplement No. 10.

Volume meter standardized in the United States of America, termed the " vu meter " : Supplement No. 11.

Peak indicator used by the British Broadcasting Corporation : Supplement No. 12.

Maximum amplitude indicator Types U21 and U 71 used in the Federal Republic of Germany : Supplement No. 13.

The Volume Indicator — S.F.E.R.T. Volume Indicator 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*)

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 British Administration appear in Supplement No. 14 to this volume.

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 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 $[V_{(\text{output})} = V_{(\text{input})}{}^n]$ 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.

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); 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 neper or 0 decibel to the frequency 800 Hz.

Permissible tolerances

The following permissible tolerances are:

50 to 300 Hz	±2 decibels	or ±0.23 neper
300 to 800 Hz	±1 decibel	or ±0.12 neper
800 Hz	0 decibel	or 0 neper
800 to 3000 Hz	±1 decibel	or ±0.12 neper
3000 to 3500 Hz	±2 decibels	or ±0.23 neper
3500 to 5000 Hz	±3 decibels	or ±0.35 neper

TABLE 1

Table of commercial telephone circuit psophometer weighting coefficients

Frequency Hz	Weight			
	Numerical value	Numerical value squared	Value in nepers	Value in decibels
16.66	0.056	0.003136	−9.79	−85.0
50	0.71	0.5041	−7.25	−63.0
100	8.91	79.3881	−4.72	−41.0
150	35.5	1 260.25	−3.34	−29.0
200	89.1	7 938.81	−2.42	−21.0
250	178	31 684	−1.73	−15.0
300	295	87 025	−1.22	−10.6
350	376	141 376	−0.98	− 8.5
400	484	234 256	−0.73	− 6.3
450	582	338 724	−0.54	− 4.7
500	661	436 921	−0.41	− 3.6
550	733	537 289	−0.31	− 2.7
600	794	630 436	−0.23	− 2.0
650	851	724 201	−0.16	− 1.4
700	902	813 604	−0.10	− 0.9
750	955	912 025	−0.046	− 0.4
800	1 000	1 000 000	0.000	0.0
850	1 035	1 071 225	+0.034	+ 0.3
900	1 072	1 149 184	+0.069	+ 0.6
950	1 109	1 229 881	+0.103	+ 0.9
1 000	1 122	1 258 884	+0.115	+ 1.0
1 050	1 109	1 229 881	+0.103	+ 0.9
1 100	1 072	1 149 184	+0.069	+ 0.6
1 150	1 035	1 071 225	+0.034	+ 0.3
1 200	1 000	1 000 000	0.000	0.0
1 250	977	954 529	−0.023	− 0.20
1 300	955	912 025	−0.046	− 0.40
1 350	928	861 184	−0.075	− 0.65
1 400	905	819 025	−0.100	− 0.87
1 450	881	776 161	−0.126	− 1.10
1 500	861	741 321	−0.150	− 1.30
1 550	842	708 964	−0.172	− 1.49
1 600	824	678 976	−0.193	− 1.68
1 650	807	651 249	−0.214	− 1.86
1 700	791	625 681	−0.234	− 2.04
1 750	775	600 625	−0.255	− 2.22
1 800	760	577 600	−0.275	− 2.39
1 850	745	555 025	−0.295	− 2.56
1 900	732	535 824	−0.311	− 2.71
1 950	720	518 400	−0.329	− 2.86
2 000	708	501 264	−0.345	− 3.00
2 050	698	487 204	−0.359	− 3.12
2 100	689	474 721	−0.373	− 3.24
2 150	679	461 041	−0.386	− 3.36
2 200	670	448 900	−0.400	− 3.48
2 250	661	436 921	−0.414	− 3.60
2 300	652	425 104	−0.428	− 3.72
2 350	643	413 449	−0.442	− 3.84

TABLE 1 (contd.)

Table of commercial telephone circuit psophometer weighting coefficients

Frequency Hz	Weight			
	Numerical value	Numerical value squared	Value in nepers	Value in decibels
2 400	634	401 956	−0.456	− 3.96
2 450	626	390 625	−0.470	− 4.08
2 500	617	380 689	0.484	− 4.20
2 550	607	368 449	−0.499	− 4.33
2 600	598	357 604	−0.513	− 4.46
2 650	590	348 100	−0.528	− 4.59
2 700	580	336 400	−0.544	− 4.73
2 750	571	326 041	−0.560	− 4.87
2 800	562	315 844	−0.576	− 5.01
2 850	553	305 809	−0.593	− 5.15
2 900	543	294 849	−0.610	− 5.30
2 950	534	285 156	−0.627	− 5.45
3 000	525	275 625	−0.645	− 5.60
3 100	501	251 001	−0.691	− 6.00
3 200	473	223 729	−0.748	− 6.50
3 300	444	197 136	−0.812	− 7.05
3 400	412	169 744	−0.886	− 7.70
3 500	376	141 376	−0.979	− 8.5
3 600	335	112 225	−1.09	− 9.5
3 700	292	85 264	−1.23	−10.7
3 800	251	63 001	−1.38	−12.0
3 900	214	45 796	−1.54	−13.4
4 000	178	31 684	−1.73	−15.0
4 100	144.5	20 880.25	−1.93	−16.8
4 200	116.0	13 456	−2.15	−18.7
4 300	92.3	8 519.29	−2.38	−20.7
4 400	72.4	5 241.76	−2.62	−22.8
4 500	56.2	3 158.44	−2.88	−25.0
4 600	43.7	1 909.69	−3.13	−27.2
4 700	33.9	1 149.21	−3.38	−29.4
4 800	26.3	691.69	−3.64	−31.6
4 900	20.4	416.16	−3.89	−33.8
5 000	15.9	252.81	−4.14	−36.0
>5 000	<15.9	<252.81	<−4.14	<−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 à 6 000	<15.9	<252.81	<−4.14	<−36.0
≥6 000	<7.1	< 50.41	<−4.95	<−43.0

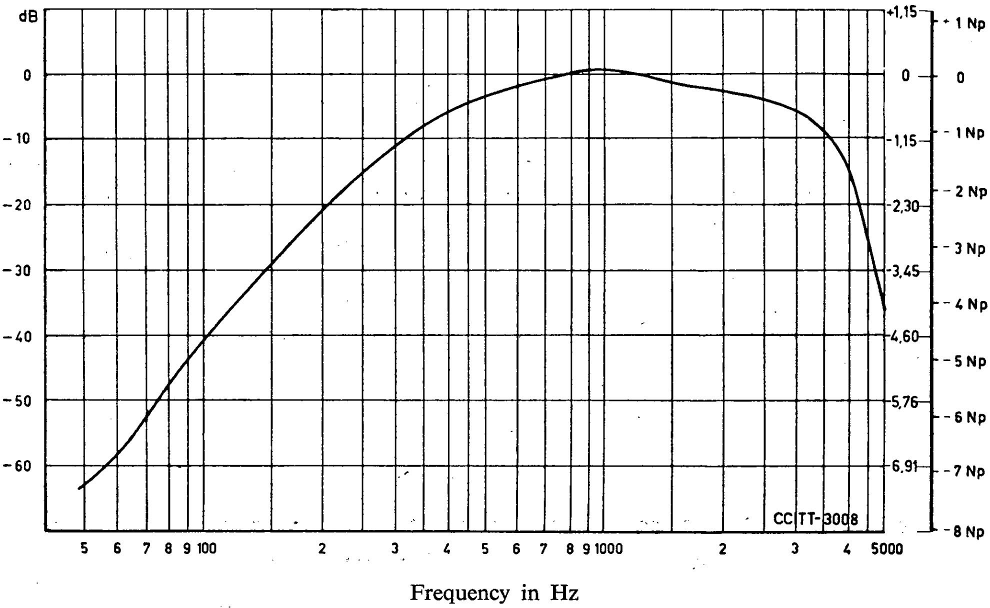


FIGURE 1. — **Characteristic curve of the psophometer filter network used for measurements at the terminals of a commercial trunk telephone circuit**

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.

Correspondance 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 psophometer, 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 decibels or 0.35 neper with a tolerance of ± 0.5 decibel or 0.05 neper.

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 or ± 0.05 neper).

Second method. — The British Telephone Administration 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 or ± 0.05 neper).

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) 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.

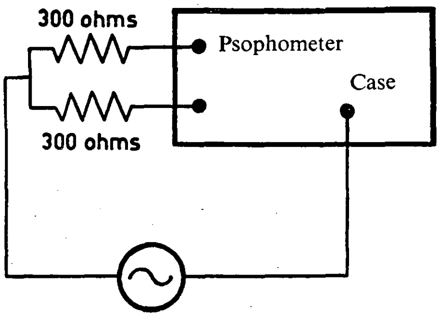


FIGURE 2

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 British Telephone Administration 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, 40″;

— cross section not greater than 1 sq. inch 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 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 circuits for programme transmission 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 programme circuit

(See curve in Figure 3)

Frequency Hz	Weighting relative to 1000 Hz			
	Nominal		Tolerance	
	Np	dB	Np	dB
20 and below	⩽−4.6	⩽−40	—	—
50	−3.95	−34.3	±0.17	±1.5
60	−3.70	−32.2	”	”
100	−3.00	−26.1	”	”
200	−2.00	−17.3	”	”
400	−1.01	− 8.8	”	”
800	−0.22	− 1.9	”	”
1 000	0	0	—	—
2 000	+0.61	+ 5.3	±0.17	±1.5
4 000	+0.94	+ 8.2	”	”
5 000	+0.97	+ 8.4	”	”
6 000	+0.94	+ 8.2	”	”
7 000	+0.84	+ 7.3	”	”
8 000	+0.59	+ 5.1	”	”
9 000	−0.03	− 0.3	±0.35	±3.0
10 000	−1.12	− 9.7	”	”
13 000	⩽−3.5	⩽−30	—	—
20 000 and above	⩽−4	⩽−35	—	—

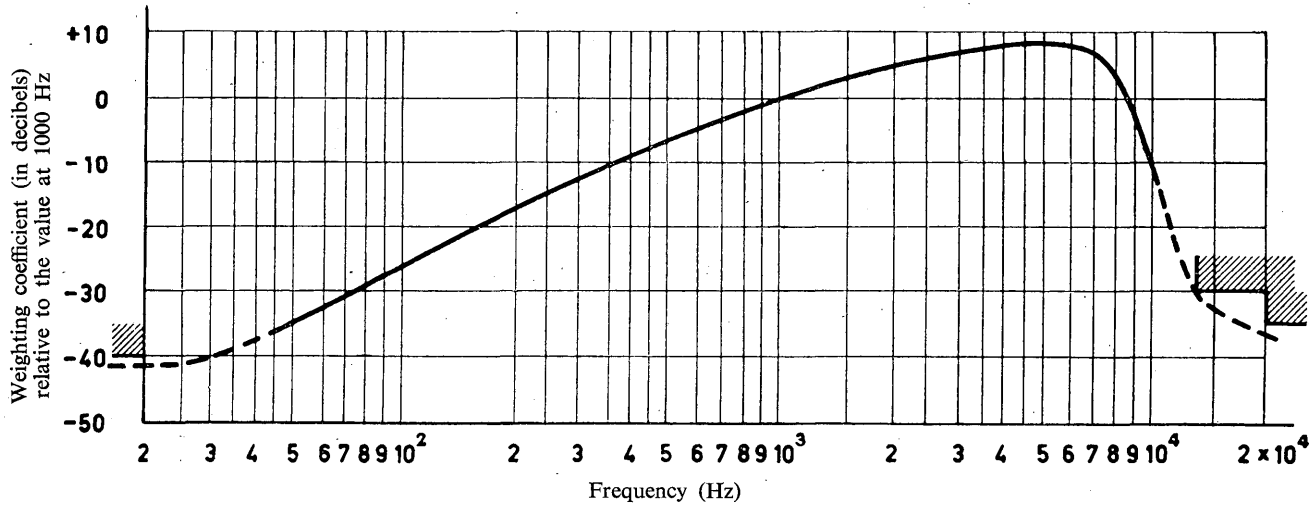


FIGURE 3. — **Characteristic curve of the filter network for the psophometer used for measurements on radio broadcasting circuits**

RECOMMENDATION P.54 (amended at Mar del Plata, 1968)

SOUND LEVEL METERS

(APPARATUS FOR THE OBJECTIVE MEASUREMENT OF ROOM NOISE)

The C.C.I.T.T. draws the attention of administrations to the following I.E.C. publications:

Publication 123 — Recommendations on sound level meters.

Publication 179 — Precision sound level meters.

Note. — Pending international standardization, the C.C.I.T.T. provisionally adopted, as the specification of equipment for the objective measurement of room noise, the specification given in Standard Z.24.3.1944 of the American Standards Association (Sound level meter for measurement of noise and other sounds). This specification is reproduced in Annex 24 (Part II of Volume V of the *Red Book*); it was superseded as the USASI (United States of America Standards Institute) standard by Document S 1.4-1961 (American Standard Specification for General Purpose Sound Level Meters).

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

APPARATUS FOR THE MEASUREMENT OF CLICKS

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 pulse meter defined in Recommendation H.13 (Volume III of the *White Book*) for measuring the occurrence of series of pulses on circuits for both speech and data transmission.

Note. — At national level, administrations might continue to study whether the use of these pulse meters 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.

[1] Former Recommendation P.55 (*Red Book*, Volume V, page 134) is withdrawn.

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*.

b) *Rayleigh disk 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 this volume.

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 functuon 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 favoured 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) So 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) So 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, is now being studied by the American Telephone and Telegraph Company. The basic measuring equipment is described in Annex 1 to Question 15/XII, Part II of this volume.

SECTION 7

SUBJECTIVE VOICE-EAR MEASUREMENTS

RECOMMENDATION P.71 (amended at Mar del Plata, 1968)

MEASUREMENT OF SPEECH VOLUME [1]

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.),

[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.

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:

α) *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:

α.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 transmission units (nepers or decibels) "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.

α.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.

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, Leon, 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 "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.

[1] The position of the guard-ring is defined in section C of the present Recommendation.

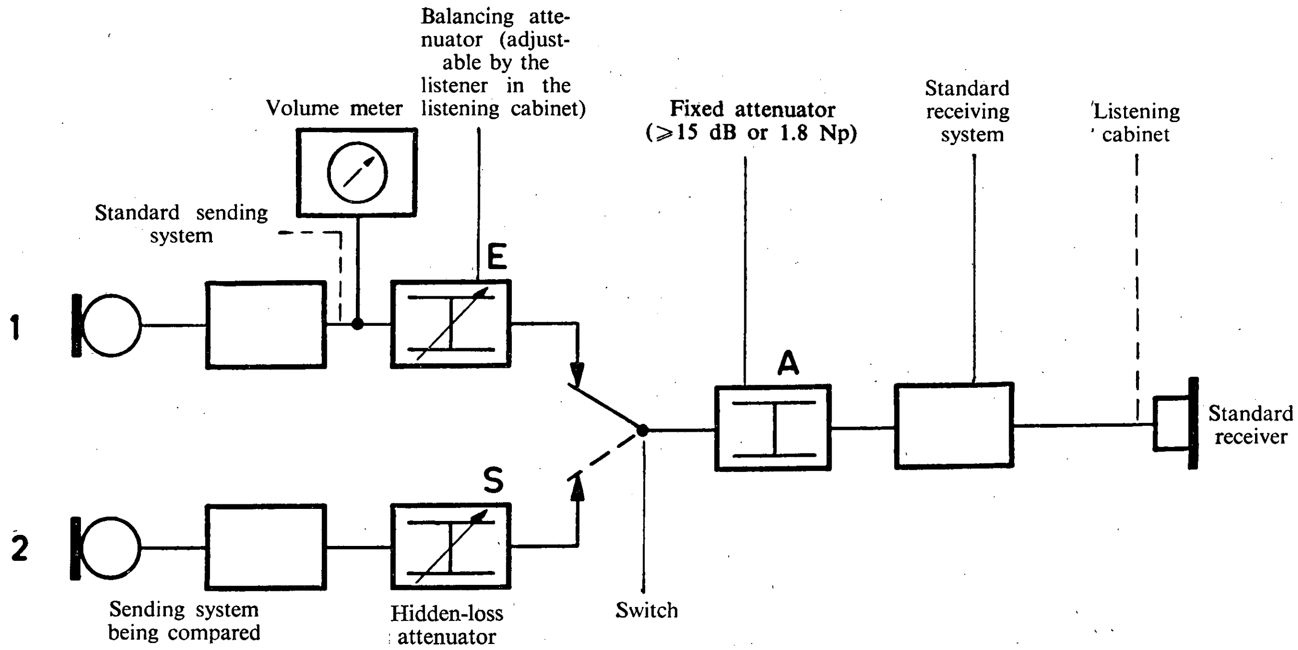


FIGURE 1. — **Comparison of a given sending system with a standard sending system (method termed “ Two-operator with hidden-loss method ”)**

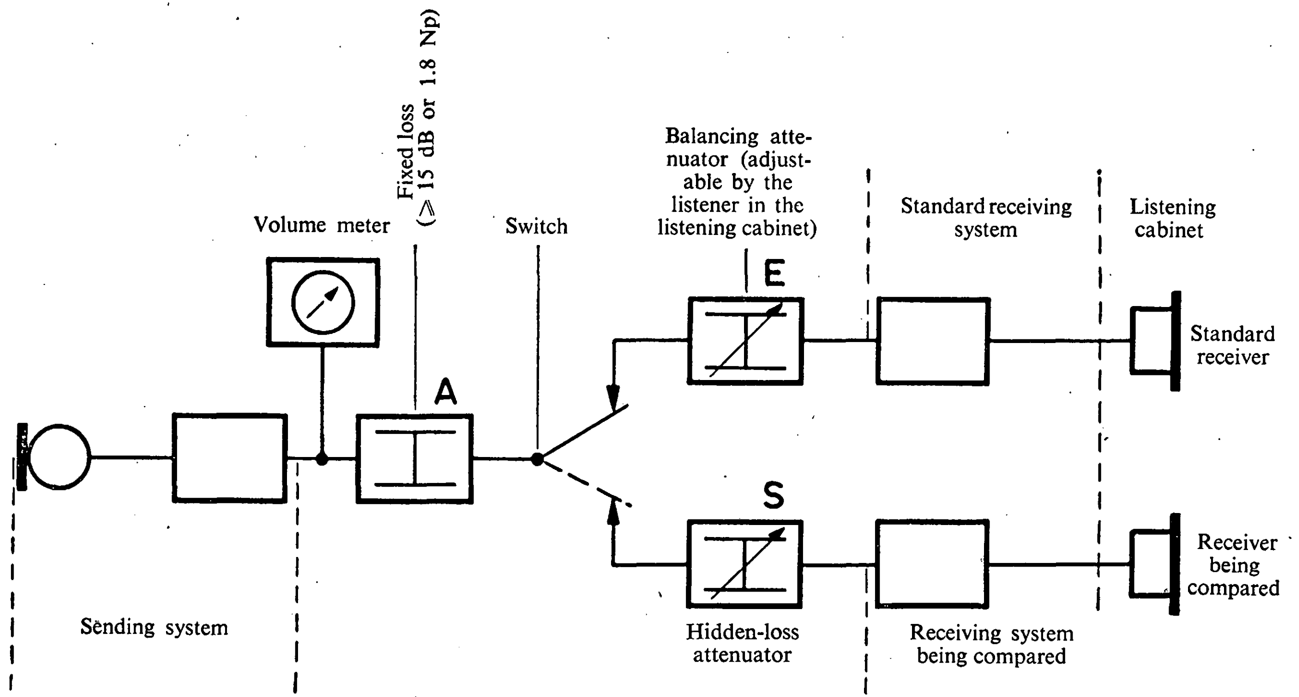


FIGURE 2. — **Comparison of a given receiving system with a standard receiving system (method termed “ Two-operator with hidden-loss method ”)**

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.

α.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.

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 (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)

α.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:

α.2.1 *Comparison of a sending system with a standard sending system* (Figure 3)

Operator C adjusts the balancing attenuator to a preliminary value a_1, he then signals by lamp, by buzzer, or verbally 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.

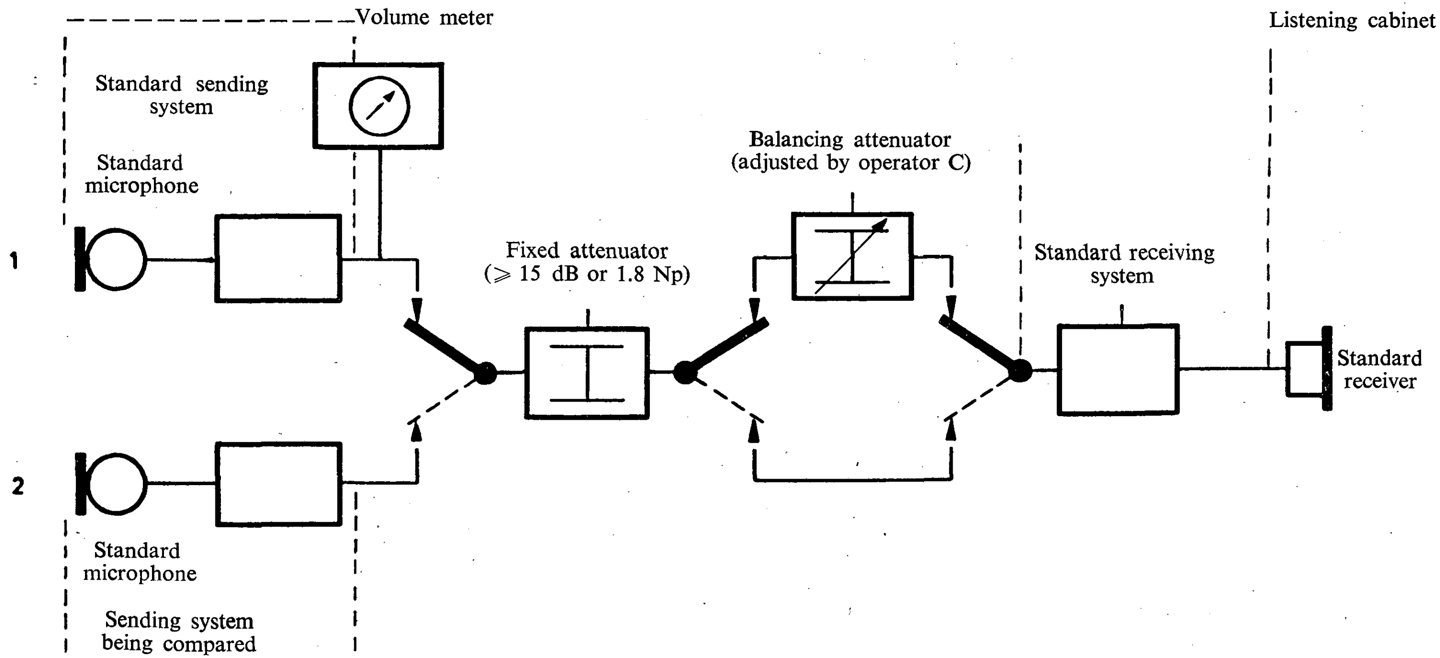


FIGURE 3. — **Comparison of a sending system with a standard sending system (three-operator without hidden-loss method)**

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.

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

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 latter is in series with an attenuation of a_2 units; 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 one or two decibels, or by 0.1 or 0.2 neper, 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 and private operating agencies 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, point 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. 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) 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.

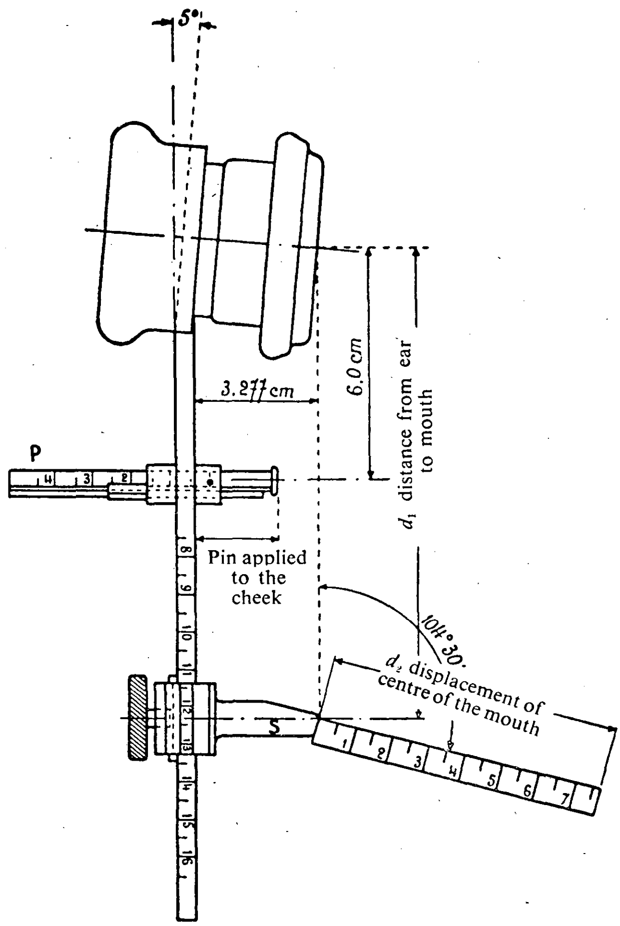


FIGURE 4. — **Device for measuring the dimensions of the head**

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:

$$\beta = \arc \sin \frac{l}{2\,\delta} - \alpha$$

The C.C.I.T.T. recommends the following values for α, β and δ in the case of reference equivalent measurements:

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

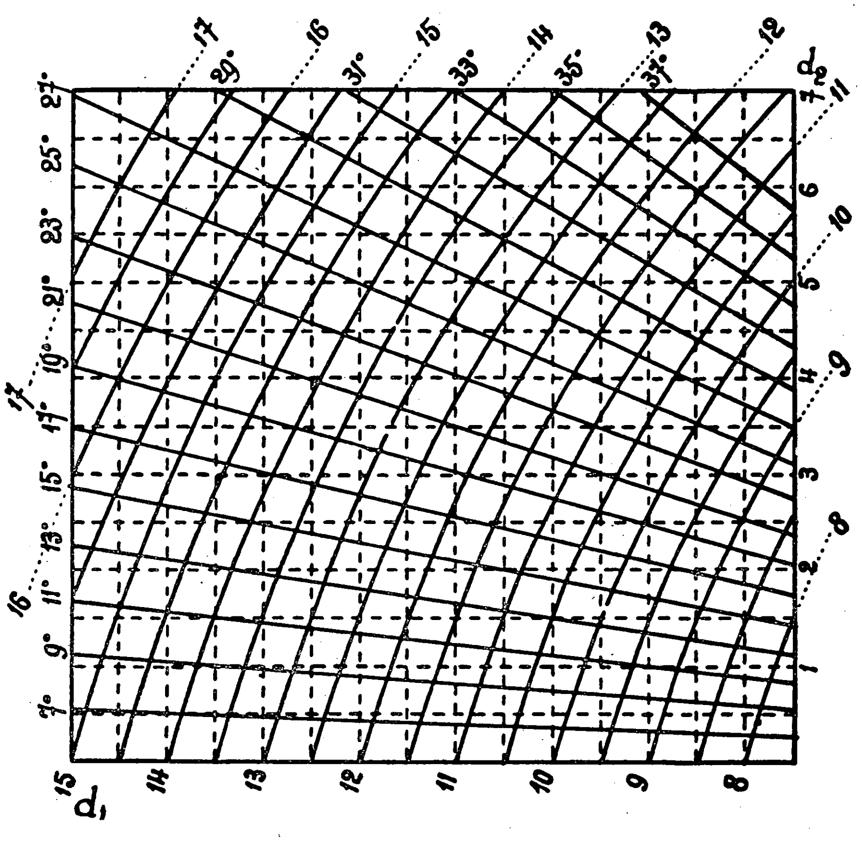


FIGURE 5. — **Abac used with the device for measuring the dimensions of the head**

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

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.

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.

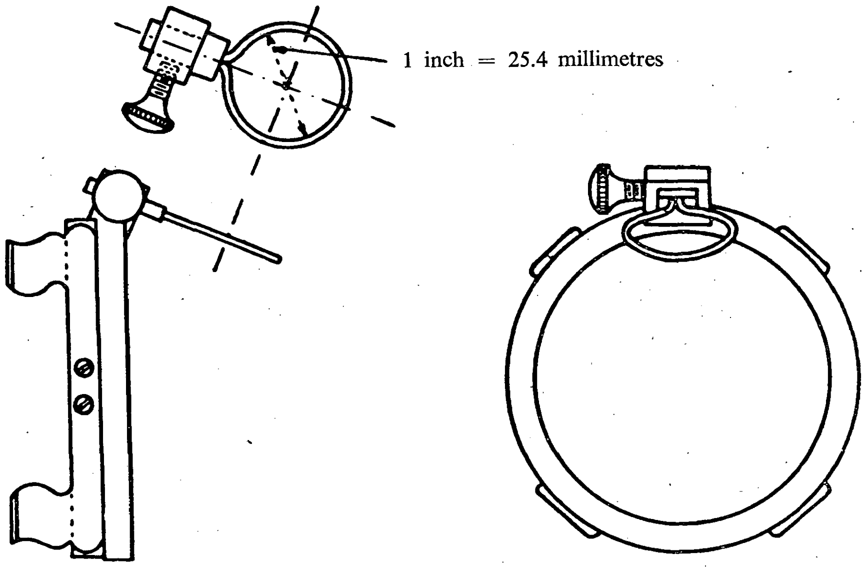
It should be noted that the position of the guard-ring, thus defined, has been fixed without reference to the inclination of the diaphragm of the microphone and does not necessarily correspond to the best operating conditions of the latter.

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.

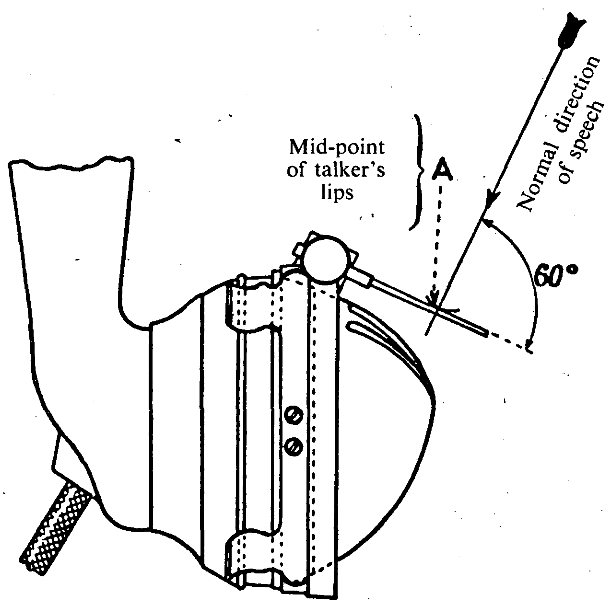
In order to avoid this trouble the receiver of the handset should be disconnected and is not to be applied to the operator's ear; in addition, in the test arrangement a similar receiver should be inserted in place of the disconnected receiver which should be placed face downwards on the table so as to 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 and Figure 7 is recommended.

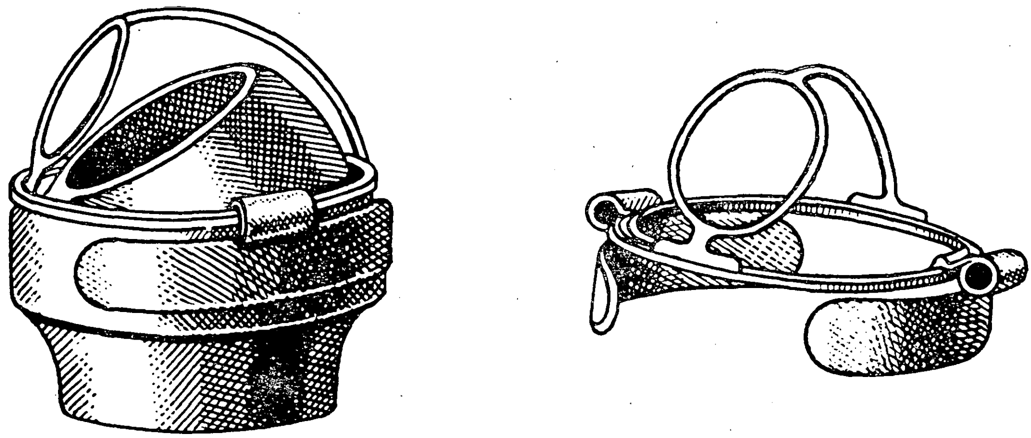


1. Example of guard-ring for tests of handsets



2. Attachment of guard-ring to a handset

FIGURE 6



Attachment of guard-ring
to a handset

Perspective sketch
of the guard-ring

FIGURE 7. — **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 and private operating agencies 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:

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

while the values retained for reference equivalent measurements are:

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

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.

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

[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.

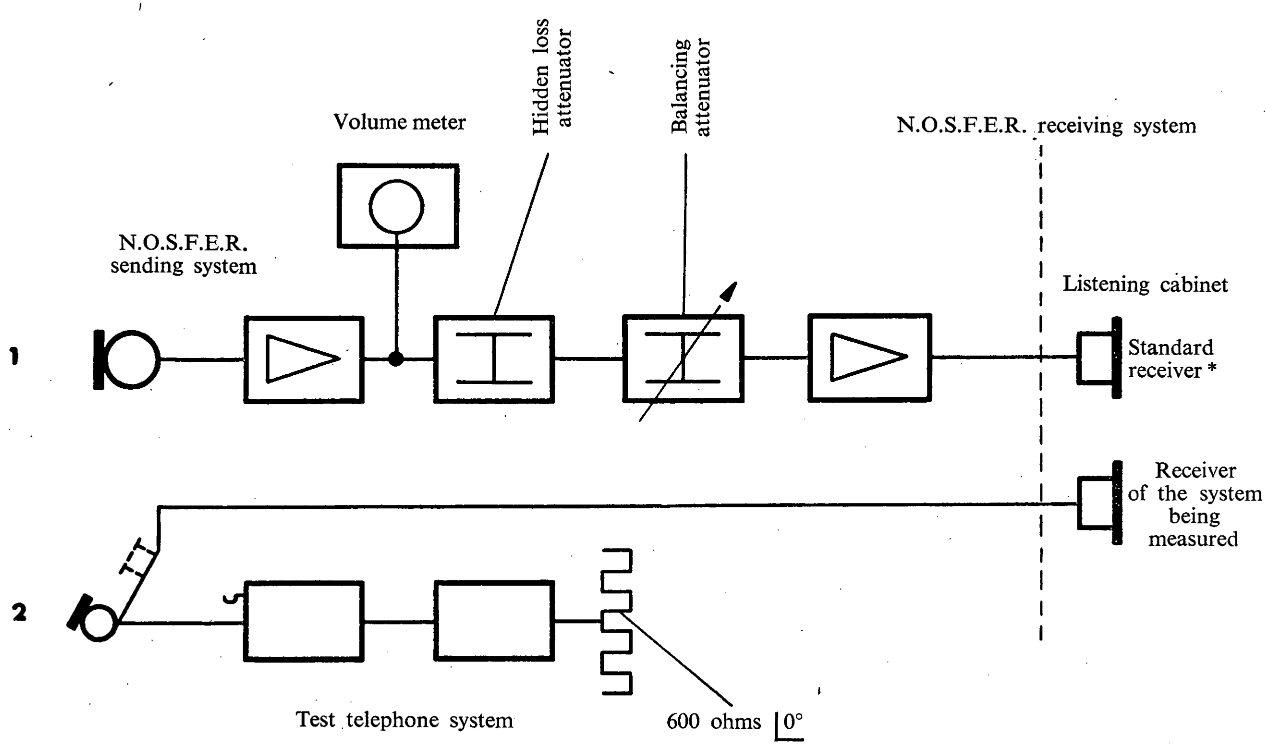


FIGURE 1. — **Measurement of the sidetone reference equivalent of a commercial telephone system**

* 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.

of the telephone system is equal to the sum $S + Q$ of the attenuation 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 a " 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 disk 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).

The value of the reference equivalent of sidetone for room noise is equal to $S + Q - 17$ decibels, where S is the loss of the " hidden length " attenuator, Q is the loss of the balance attenuator, the correction of 17 decibels 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

[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.

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 transmission units (nepers or decibels).

The test circuit used for measurements by this method should reproduce the average conditions of a typical commercial trunk telephone call. Each administration or private operating agency 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, Supplement No. 8 to this volume describes a method used by the British Administration for subjectively determining telephone transmission quality.

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 periodical 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 Great Britain, 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

[1] The United Kingdom Administration considers that the cost of applying preventive maintenance to telephone sets other than public call-boxes and private branch exchanges would not be justified.

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 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. 0.0002 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. A 0 to 100 scale on the same meter is used to read 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 plans to assess subscriber sets with an electroacoustic rating system (E.A.R.S.)[1]. This system is used in the laboratory to determine relative 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 Federal German Administration 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 Part III of this volume, under " Federal Republic of Germany " in Supplement No. 7).

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

[1] See Annex 1 to Question 15/XII in Part II of this volume.

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 decibels 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 below. 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 of Great Britain and Northern Ireland

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 new type of microphone (the 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 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.

PART II

QUESTIONS CONCERNING TELEPHONE TRANSMISSION QUALITY AND LOCAL NETWORKS ALLOCATED TO STUDY GROUP XII IN 1968-1972

Important notes

1. The Plenary Assembly having set up Special Study Group D, all questions relating to pulse code modulation (p.c.m.) have been assigned to this Study Group for the time being (except for a study which Study Group XII has been asked to carry out within the specific context of its work).

The Chairman of Special Study Group D will contact the other Chairmen to make arrangements for liaison with the other Study Groups concerned as work progresses.

2. 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 co-ordination within their national administrations, in accordance with the decision taken by the Plenary Assembly.

Question 1/XII — Reference equivalents of national systems in the new transmission plan

(continuation of Question 1/XII, studied in 1964-1968)

a) What is the percentage of international calls for which it seems possible to satisfy the limits of 20.8 dB (24 dNp) and 12.2 dB (14 dNp) for the national reference equivalents [1] ?

Note. — A minimum of 97 % is provisionally recommended in Recommendation P.11. Annex 2 describes how the real value of this percentage is determined.

b) How should allowance be made for variation with time of the equivalents of national circuits (other than four-wire circuits interconnected to the international chain on a four-wire basis), and for the ageing of microphones and telephone receivers ?

[1] When studying these parts of the question, attention will be paid to the information to be collected on the transmission quality of complete telephone connexions and national extensions during the studies indicated in supplements X and Y of Volume IV of the *White Book*.

c) What is the minimum value[1] to be recommended for the nominal reference equivalent of the national sending system ?

Note. — It is impossible to reply to part c of Question 1/XII until a reply has been given to part b of Question 10/XII.

d) What is the minimum value to be recommended for the nominal reference equivalent of the national receiving system ?

Note. — The replies to parts c and d will determine the minimum reference equivalent of an international connection.

e) What, according to customer preferences, should the short- and long-term planning objectives be for the reference equivalent of an international connection and how should the overall reference equivalent be apportioned between national sending and receiving systems ?

Note. — Annex 3 sets out the reasons for studying part e.

f) What maximum and minimum values should be advocated for the nominal reference equivalent of a connection comprising an operator's telephone set ?

Note. — In the reply to part d, account will be taken of Annex 3.

General note. — Annex 1 below reproduces the reply made to the various parts of this question in 1967.

ANNEX 1

(to Question 1/XII)

Report (approved by Study Group XII in October 1967) by Mr. D. L. Richards' Working Party

Part a of Question 1/XII

The percentage of international calls for which it seems possible to satisfy the limits of 24 dNp and 14 dNp for the national reference equivalents is at least 95 % and, for some administrations or operating companies, the percentage reaches 97 %.

Study Group XII proposes that section B.a in Recommendation P.11 shall be amended accordingly.

This part of the question requires further study to clarify the section, including note 3, 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[2]. 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

[1] The documentation collected by Study Group XVI concerning minimum send reference equivalents will be included in the documentation of Study Group XII in 1968-1972.

[2] 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. NÅSELL : 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.

took place on the possibility of re-wording Recommendation P.11 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.

It was pointed out that statistical variations and certain planning practices could result in differences in overall reference equivalent between the two directions of transmission of more than 10 dB. This could be limited by requiring that the difference between the sending and the receiving reference equivalents of every national system was a constant quantity, for example 8.7 dB (1 Np). The suggestion was made that a clause should be added to Recommendation P.11 to this effect. Some evidence was quoted from the article by Mr. Boeryd to support the proposition that such asymmetrics in transmission are important.

The Chairman ruled that the evidence concerning the importance of the effect should be examined before proceeding to draft any qualifying clause for association with Recommendation P.11. The Working Party was generally of the opinion that the evidence concerning the subjective effects of asymmetry was not, at present, conclusive and that it would be premature to consider now any qualifying clause. It was considered, however, that the matter ought to be studied further as Question 3/XII.

Part b

The Working Party considers that telephone receivers do not deteriorate appreciably with time and so there is no need to take account, when planning networks, of any dispersion in their sensitivities. Telephone microphones show greater variations than telephone receivers but it seems possible to take account of this in planning only by allowing a suitable margin when choosing planning rules.

No information has been supplied on the variation of loss of the lines connecting terminal local exchanges to the primary switching centre.

Part c

Discussion showed that there is no longer any need to set a minimum value for the nominal send reference equivalent of the national system provided administrations carry out, from time to time, statistical surveys of the volume of speech sounds transmitted in their networks.

The Working Party considered that reference to a minimum value for the mean of the send reference equivalent also should be deleted. The reply to this part of the question made by Study Group XII in 1966 is therefore cancelled. See also the report by the Working Party annexed to Question 10/XII.

Former Part d (now Part f)

There are at present no descriptions of the manner in which the reference equivalents of operators' telephone sets should be measured so that the following maximum values suggested in 1966 cannot be confirmed :

18 dNp (15.6 dB) for the sending reference equivalent, and 8 dNp (7 dB) for the receiving reference equivalent when the subscriber's line is connected, the sending and receiving reference equivalents being calculated at the " virtual switching points " of the international circuit.

There is need for information concerning the talking distance and vocal levels that should be adopted for such measurements and the arrangements for terminating the line terminals of the sets. A statement of the problem is given in Annex 4.

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 1233 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 would also be respected by more than 97 % of connections, thus complying with the provisional recommendation in Recommendation P.11 (Volume V of the *White Book*).

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.

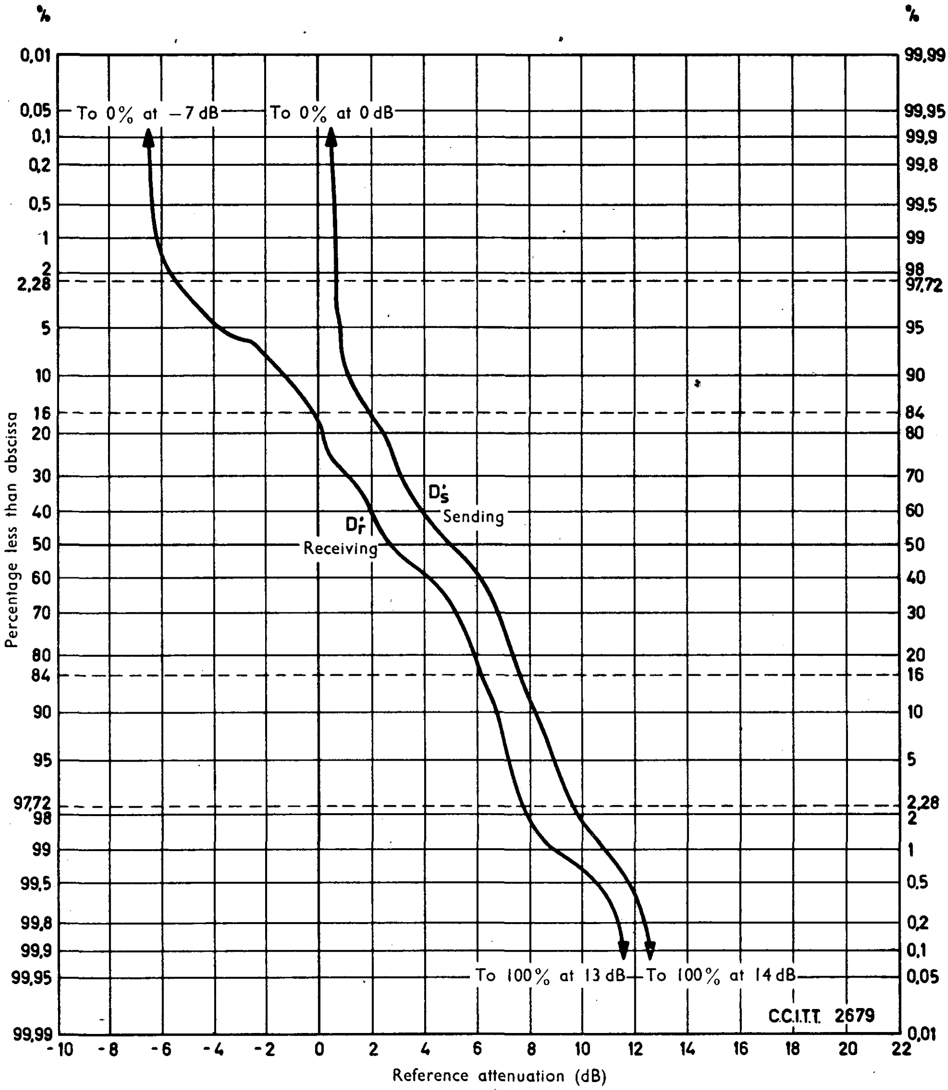


FIGURE 1. — **Reference attenuation of inter-exchange network between terminal exchange and international virtual switching points. (International outgoing calls—10 May 1967)**

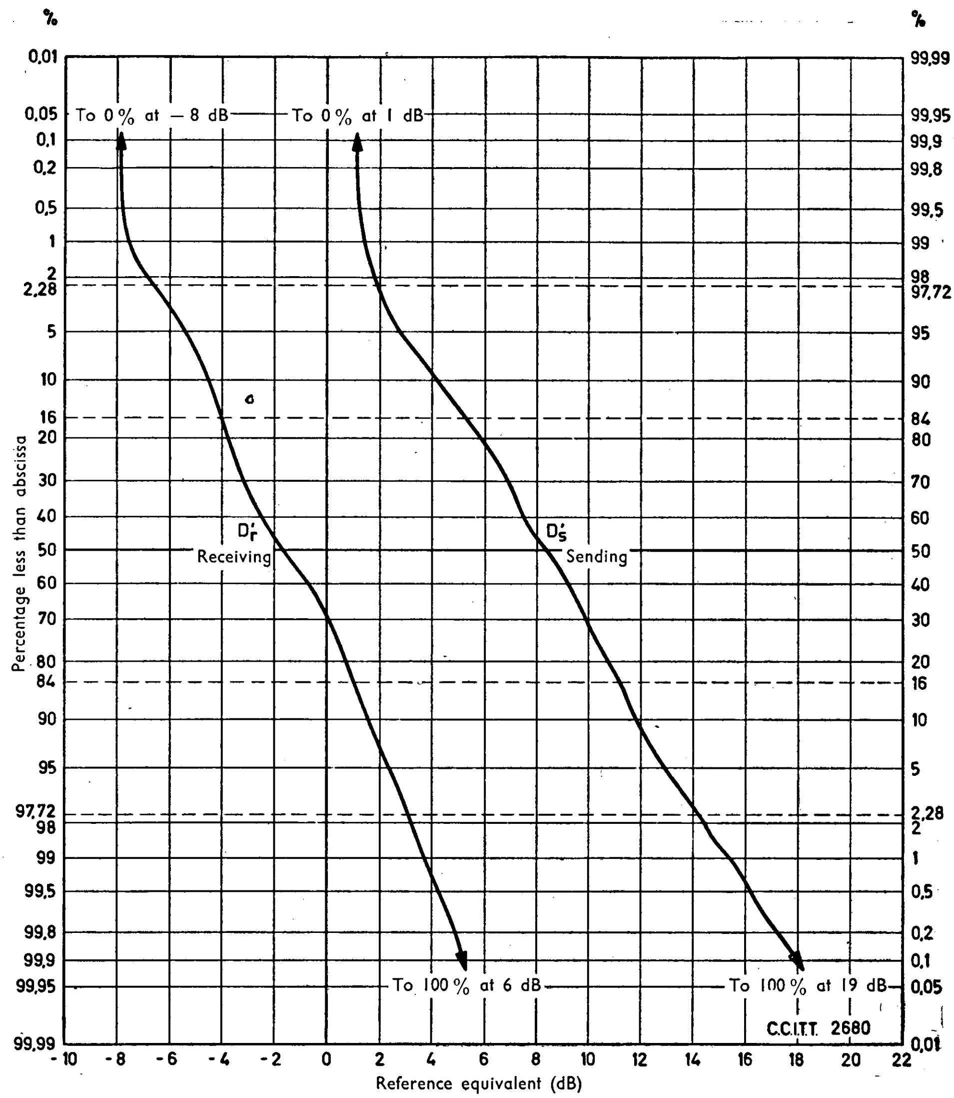
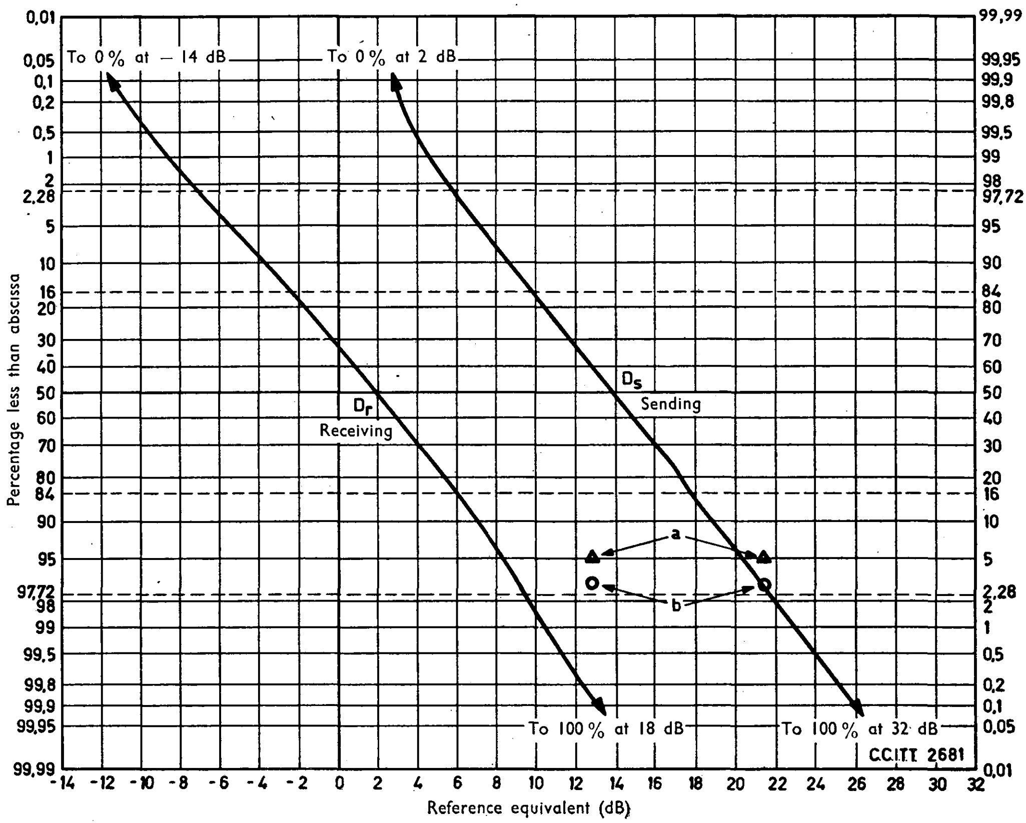


FIGURE 2. — **National subscribers' sending and receiving reference equivalents**
(Survey of subscribers' network, 1965)



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**

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.

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	590 miles (950 km)
Queensland	75	63.5	43	36.5	118	9.55	650 miles (1045 km)
South Australia	43	79.6	11	20.4	54	4.35	1090 miles (1750 km)
Western Australia ...	44	89.8	5	10.2	49	3.96	2540 miles (4080 km)
Tasmania	15	79	4	21	19	1.54	990 miles (1590 km)
Totals ...	1063	86.1	170	13.9	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)

Reply to part d of Question 1/XII in 1967

(Contribution by the United Kingdom Administration)

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 deviations for subscribers are 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 with 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 (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 the United Kingdom Administration's operators' sets 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 Administration 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 2/XII — Assessment of service transmission quality

(continuation of new Question A, 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 question 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 describes this kind of method and Annex 2 gives an example of test programmes.

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). 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 part a of Question 7/XII 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 Group XIII studies questions relating to those aspects of service appraisal other than speech transmission quality. An example of co-ordination occurred in the past when questions relating to transmission were included in a trial questionnaire used in a study carried out by Study Group XIII; this study is being pursued within the scope of Question 12/XIII and it would be desirable to co-ordinate the study of questions 2/XII and 12/XIII very closely.

ANNEX 1

(to Question 2/XII)

Methods for the evaluation of service from the standpoint of speech transmission quality

(Contribution by the American Telephone and Telegraph Company)

1. *General*

Methods used to evaluate speech transmission quality experienced by customers should provide information representative of actual use conditions. There are two ways in which this can be achieved. The first, and most direct, approach involves quality assessment on actual intercontinental telephone calls; the assessment is made and reported by the customer. A second, indirect, approach makes use of measured transmission parameters on intercontinental connections and independently determined relationships between these parameters and quality assessments to predict the quality on intercontinental calls. This, of course, requires that the necessary relationships be known.

Both approaches are useful when applied in an appropriate manner. However, the second requires a greater knowledge of the impairments and their effect, i.e. the relationships referred to above. Such knowledge is often not available and, therefore, the method is not always applicable. For this reason, the discussion in this document is restricted to the direct approach. This is not intended to minimize the importance of the indirect approach in applications where the transmission parameters of interest are well understood, measurement methods and equipment exist, and subjective test data are available. For example, the indirect approach may be particularly appropriate in assessing the speech transmission quality on circuits on which the impairment is caused by loss and noise.

Before any proposed method of measurement is accepted as an adequate measure of transmission quality, the method should be evaluated with respect to the criteria of reliability and validity. Some of the commonly-used measures of transmission quality fail one or both of these tests. Some methods are insensitive to known changes or differences in transmission, some show excessive variation over different people making the measurements and some methods have not produced consistent results even when the conditions are unchanged.

The next section defines the criteria for evaluating measurement methods. The third section discusses methods which have been used by the A. T. & T. Company and the final section illustrates the evaluation of several measurement methods according to the criteria defined in section 2.

2. *Necessary and desirable properties of the methods for measuring transmission quality*

Any measurement method which is used to obtain data on speech transmission quality based on user reaction should be evaluated for reliability and validity as described below :

Reliability — The overall method must give the same or similar results or values from repeated measurements under the same conditions. Fluctuation in the reported values due to random variations should be small compared with variations due to changes in transmission. Any method should satisfy these two requirements.

In addition, methods proposed for general use should satisfy more severe requirements. If comparisons are to be made between measurements made at different times or places the method must give the same or similar results or values when the essential transmission parameters are unchanged even though non-transmission conditions are changed, e.g., different observers, interviewers, or users. That is, the method should be relatively insensitive to factors other than transmission quality. This broader definition of reliability might be termed " robustness " in keeping with the use of this term in statistics.

Validity — The results or values produced by the method must reflect known changes in transmission quality and correlate with other good measures of transmission quality. The method must be sensitive to known changes in transmission quality if it is to be used as a measure of an unknown transmission quality.

3. *Methods and types of measurement*

The A. T. & T. Company has used a number of different measures of transmission quality based on naturally occurring conversations followed by interviews or questionnaires. Specific items or questions are listed in Table 1 along with some comments about the factors which have governed the choice of particular questions.

The methods by which the distributions of answers by users are reduced to single-valued indicators of transmission quality are not discussed in this document in any detail. Two simple approaches frequently used are to calculate the percentage of calls affected by a given condition and to compute a mean opinion score from ratings. A number of more complicated methods may also be used.

4. *Examples of evaluation of methods and measurement*

The application of the criteria of reliability and validity to measurements made according to the methods of Table 1 are illustrated below with data from recent experiments conducted by the A. T. & T. Company.

Reliability can often be determined by comparing the results of odd versus even numbered calls or interviews on odd versus even days on which measurements were made. If the difference between these subsets is small compared with the differences between different types of circuits then the measurement method is reliable. For example, a series of measurements conducted on overseas calls involved customer answers to an interview question about conversational difficulty. The percentage of calls reported as causing difficulty was computed for odd serial numbered calls and for even serial numbered calls at each value of the transmission parameter varied. The difference between odd and even calls at each parameter value was found to be substantially less than percentage differences due to the parameter variation.

An example of robustness comes from the overseas tests referred to above. During these tests interviewers in Paris conducted interviews in French and English on the same sample of calls on which interviewers in New York were conducting interviews in English only. At each value of the transmission parameters under study the percentage of calls reported as causing difficulty was computed from the New York and from the Paris interviews. The differences in

TABLE 1

A. *General question concerning transmission. This has been used when the exact nature of degradations is unknown or of secondary importance.*

Did you or the person you were talking to have any difficulty in talking or hearing over that connection ? (If answer is " yes ". probe for nature of difficulty, but *without* suggesting possible types of difficulty), e.g. " Could you describe the difficulty a little more ? "

B. *Specific questions concerning transmission features. These have been employed when the types of difficulty can be specified, and a classification of the user's responses into these categories is desired.*

Was the connection affected by any of the following conditions : (For any item answered " yes ", add the following question :)

Did the .... bother you not at all, a little, a fair amount, or a great deal ?

— Low volume ?

— Noise and hum ?

— Distortion ?

— Variations in level, cutting on and off ?

— Crosstalk ?

— Echo ?

— Complete cut-off ?

C. *Overall rating by user has been used in almost all forms.*

Which of these four words comes closest to describing the quality of that connection :

Excellent, good, fair, or poor ? (If a large proportion of the population of circuits under test exhibit serious degradation, then a fifth category of bad or unsatisfactory may be added.)

percentage between the New York and Paris results were found to be small compared with variations due to the transmission parameter under study.

Validity of a measurement is much more difficult to determine than reliability because it requires the definition of a criterion measure or variable with which the measurement in question must correlate. Often the measurement in question (e.g., average rating by the users) has as much claim to be *the* criterion variable as any other measurement (when telephone transmission quality is being measured). Nevertheless, since there is no one criterion measure or variable, any measurement should be evaluated, at least in part, by its correlation with other measurements and its sensitivity to changes in circuits which are commonly acknowledged

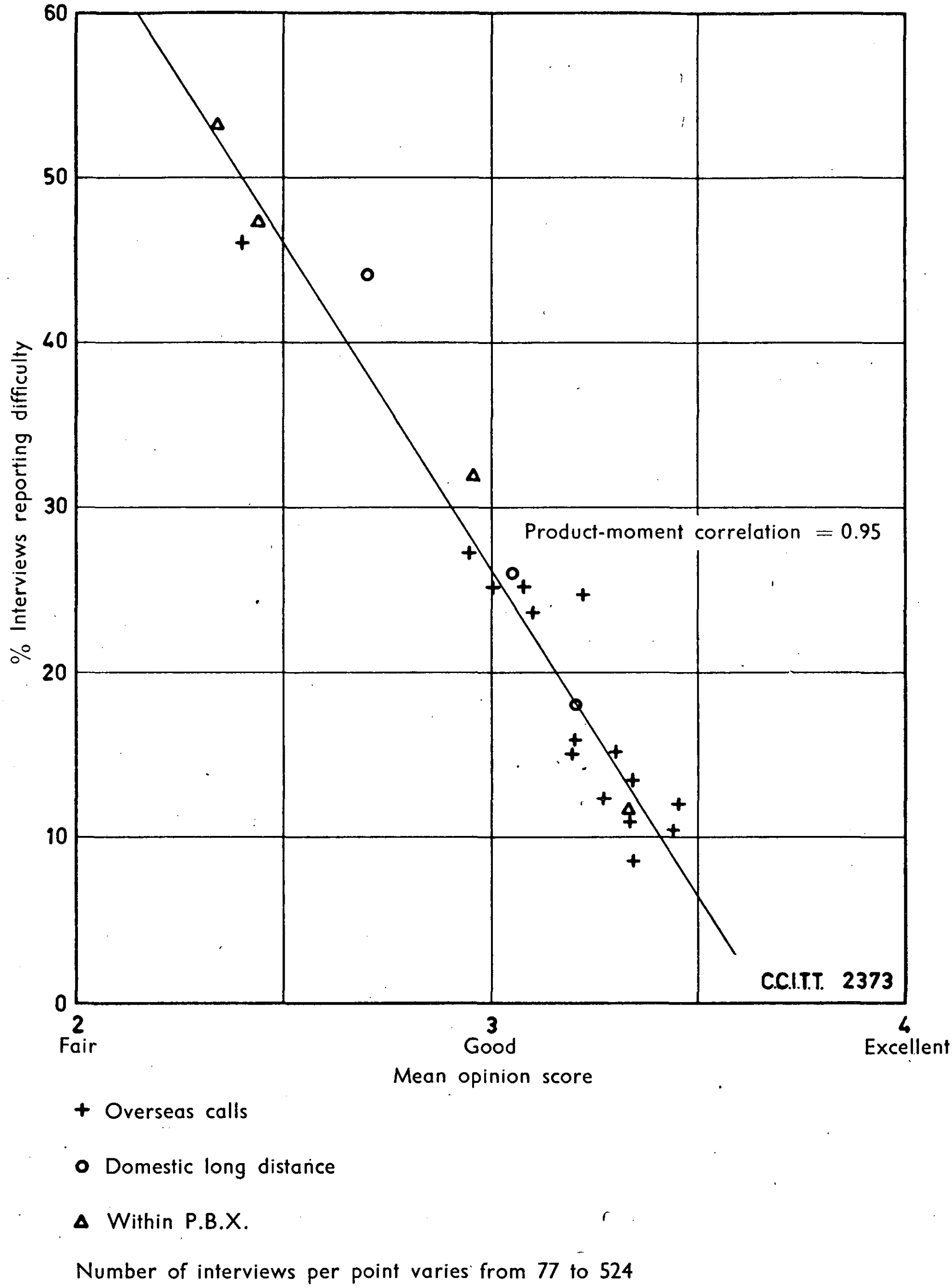

60
50
40
30
20
10
0
% Interviews reporting difficulty
Product-moment correlation = 0.95
C.C.I.T.T. 2373
2
Fair
3
Good
4
Excellent
Mean opinion score
+ Overseas calls
o Domestic long distance
Δ Within P.B.X.
Number of interviews per point varies from 77 to 524

FIGURE 1

to produce changes in quality. For example, mean opinion score based on the user's rating has often been shown to vary consistently with circuit loss and circuit noise. Recent experiments have shown these relations to hold when ratings are made after naturally occurring telephone calls over circuits, which have controlled noise and loss.

An example of the correlation between two measures of transmission quality is shown in Figure 1, which plots mean opinion score against percentage of interviews reporting difficulty with data drawn from several field and laboratory tests. Each point is based on many telephone interviews following naturally occurring conversations, but the types of circuits vary in length (within P.B.X.[1], domestic, overseas) and many points relate to circuits having different values of several different transmission parameters. The 0.95 correlation between these two measures of circuit quality indicates that they are measuring the same thing to a very large extent. The slope of the regression line shows that a change of 0.25 in mean opinion score is equivalent to a 10% change in the percentage of interviews reporting difficulty. An indication of the robustness of this relation is that the P.B.X., domestic and overseas calls all show about the same slope despite wide differences in length of conversation and expectation by the users.

Caution

The above examples are based on average values obtained for a large number of telephone calls under each condition. High reliability and validity for measures based on user reaction depends upon averaging over many calls and users to reduce the influence of random differences between circuits and between people. It can be shown that single ratings by users have very little reliability and therefore can have little validity in measuring circuit properties. Our rule of thumb is to require at least 50 calls or interviews or more often at least 100 for each condition being evaluated.

ANNEX 2

(to Question 2/XII)

Quality of service on multi-link telephone connections

(Contribution by the Telephone Association of Canada)

Introduction

Among the factors which contributed to the setting of Question 2/XII is the concern felt regarding the relatively large number of tandem connected links which might be encountered in a telephone connection under the International Routing Plan. This aspect of the International Routing Plan was recently discussed by Study Groups XIII and Special B in Tokyo in 1967 and the desirability of evaluating subscriber reaction to multi-link connections was reiterated.

The Telephone Association of Canada shares the concern regarding multi-link connections and is consequently proposing a test programme for evaluating subscriber reaction to

[1] Subscriber installation with additional stations.

such connections. This contribution presents details of the proposed test programme. Semi-automatic operation has been assumed.

Test programme

The study of routing principles, and of routing plans which permit a large number of links in tandem on a large percentage of calls, makes it necessary to assess the subscriber reaction to such a service. A test programme to obtain such data should be based on direct subscriber experience with the service and not on laboratory or other forms of simulated test conditions. Techniques of this sort are now well established and have been used in the evaluation of satellite circuits with long delay times. Study Group XII studied such methods at its 1966 meeting in Geneva, and this study is continuing. The central feature of the technique is to obtain subscriber opinion through a questionnaire by telephone immediately after he has completed a normal call over the facilities under test. Computer analysis of the replies has been found to give meaningful statistical data on subscriber reaction.

Normal traffic should be routed over the facilities under test in such a way that neither subscriber nor operator is aware of the purpose. This can be done through arbitrary codes which are dialled by the operator in semi-automatic service. Such codes will be in the position of the area code (trunk code) and will control the routing of the call. Machine access to the trunks at the originating terminal will be desirable. These arbitrary codes will be designed to set up one, two or more links in tandem, depending on the particular code used, for completion of the call. The calls established over these facilities will always be between subscribers in the same terminal cities and there will be no alternative routing, so that a direct comparison can be obtained of service on the routes corresponding to each special code. The translators in the switching machines at each transit point would be arranged to accept the special codes and take the proper action.

The following example will clarify the above explanation. In Canada, traffic between Montreal and Vancouver is handled over a direct high-usage group or over finals through the regional centre at Regina. Thus, a one- or two-link connection may be obtained depending on whether the direct or alternative route is utilized on a particular call. However, the 3000-mile route across Canada, between Montreal and Vancouver, has other switching points which can be used to set up three, four or five link connections without any significant increase in mileage. Through the use of special codes we can set up the following conditions on a normal call between Montreal and Vancouver :

1. Direct Montreal-Vancouver without overflow
2. One switch at Regina (2-link)
3. Switch at Regina and Winnipeg (3-link)
4. Switch at Regina, Winnipeg and Toronto (4-link)
5. Switch at Calgary, Regina, Winnipeg and Toronto (5-link)
6. Other less direct routes involving more mileage are also available to provide 6- and 7-link connections.

It should be noted that in each case, as above, normal circuits without any special preparation are used. The only special feature is a change in the translator of each switching machine to utilize the special codes.

In the above outline of a test programme, emphasis has been placed on the evaluation of subscriber reaction to service over semi-automatic multi-link connections. It is obvious, of course, that additional factors could be included in the programme which would appear as subscriber reaction in automatic service as distinct from semi-automatic. We refer to post-dialling delay, blocking probability, answer signal delay and switching errors. These could be measured and recorded by the operator who handles the call. An analysis of these additional factors combined with the analysis of subscriber reaction would approximate the conditions of automatic service.

ANNEX 3

(to Question 2/XII)

Improvement in quality of transmission with the duration of a telephone call

(Contribution by the Czechoslovak Administration)

When measuring the telephone transmission quality in local networks we repeatedly observed a decrease of repetition rate with increasing time of occurrence of repetitions in the course of conversation and with its increasing duration. As an example, the diagram in Figure 1 shows a typical interrelation between the total average number of repetitions Z and

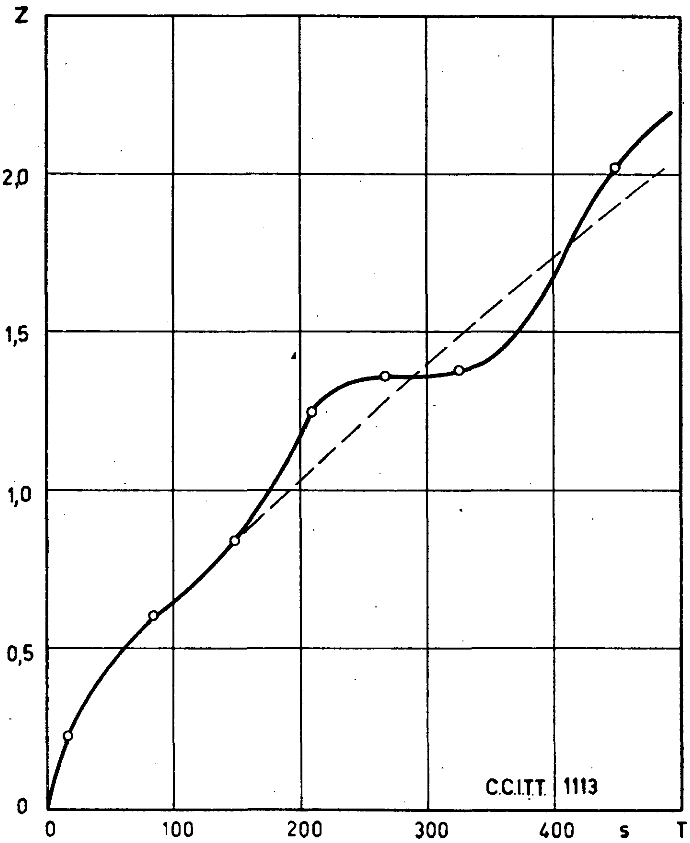


FIGURE 1. — **Relation between the total average number of repetitions *Z* and the duration of telephone conversation *T***

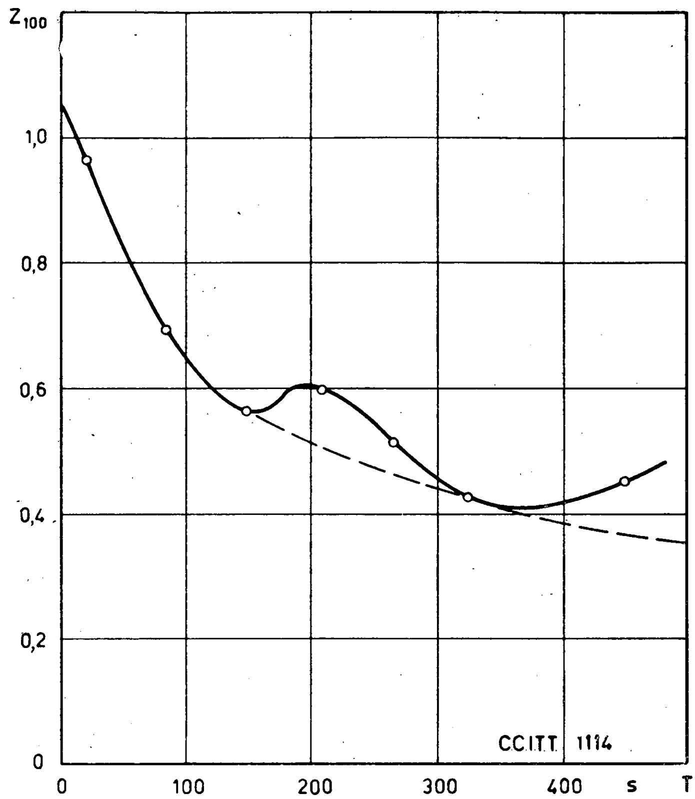


FIGURE 2. — **Relation between the average repetition rate z_{100} and the duration of telephone conversation T**

the duration of telephone conversation T during which these repetitions were observed. In Figure 2 a curve showing how the average repetitions rate z_{100} in a time unit of 100 seconds depends on the duration of telephone conversation T, is plotted and, finally, the diagram in Figure 3 illustrates the interdependence of repetition rate density ξ_{100} and the time t of repetition occurrence during telephone conversation.

The values Z, z_{100} and ξ_{100} are interrelated by the following equations :

$$z_{100} = 100 \frac{Z}{T}$$

$$\xi_{100} = 100 \frac{dZ}{dt}$$

The diagrams shown in Figures 1 to 3 were obtained by measurement in a local telephone network with approximately 400 main subscribers' stations and 800 extensions, connected to an automatic telephone exchange. The total time of measurement amounted to approximately 130 000 seconds, so that each point in diagrams Nos. 1 and 2 has been gained from a measurement at an average of 18 500 seconds.

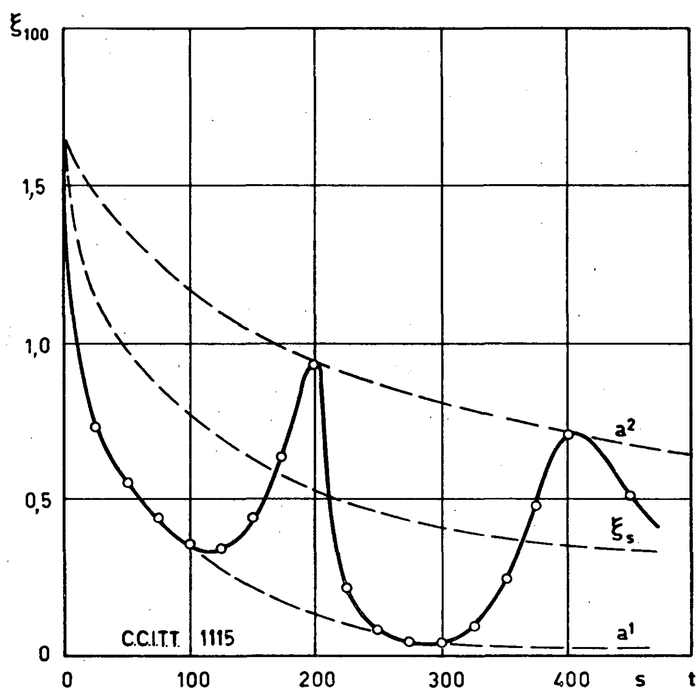


FIGURE 3. — Relation between the repetition rate density ξ_{100} and the time of occurrence t in the course of telephone conversation

The decrease of repetition rate with increasing length of telephone conversation and with increasing time of repetition occurrence indicates a gradual increase of redundance in the course of conversation which can be explained by conceptual and thematic narrowing of the subject of conversation after the initial phase in which this subject has been more or less clearly defined.

The graph in Figure 3 shows characteristic maxima adjacent to 200 and 400 seconds whose time of occurrence approximately agrees with the average period of carbon microphone packing. Thus the maxima can, at least to some extent, be explained as being due to a substantial increase of reference equivalent of the transmission channel resulting from the microphone packing.

The shape of the curve in Figure 3 could perhaps result from that the reference equivalent of carbon microphone gets temporarily improved by tapping or shaking the handset, this being a rather habitual reaction of subscribers to a distinct worsening of telephone transmission quality. Hereby the repetition rate density again decreases until a new microphone packing takes place which, in turn, involves a new increase of repetition rate density and, consequently, a more or less exact repetition of the whole cycle, described above.

The zone of repetition rate densities, given by the envelopes a_1, a_2 of the real curve, as well as the mean value ξ_s show a quite distinct decrease with increasing time co-ordinate.

ANNEX 4

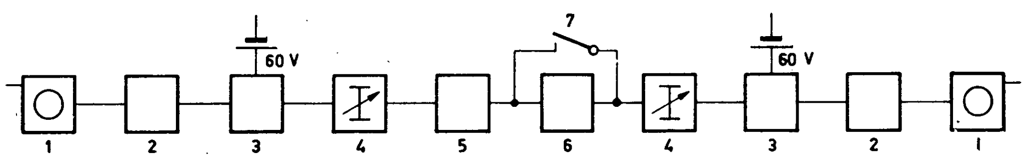
(to Question 2/XII)

Variation in the quality of a telephone call when the frequency bandwidth of the transmission channel is varied while the call is in progress

(Contribution by the Czechoslovak Administration)

The Czechoslovak Telecommunication Administration laboratories carried out tests to ascertain how the quality of a telephone call is affected when the frequency bandwidth of the transmission channel is reduced while the call is in progress.

When the quality of a telephone call is assessed in service conditions by the " repetition rate " method (see another Czechoslovak contribution), it is found that soon after the start of the call, once the subject of conversation has been established, the proportion of repetitions drops very rapidly, the redundancy of the information exchanged increases. Advantage could be taken of this fact to reduce the frequency band in the transmission channel some time after the beginning of the conversation, but only in such a manner that the quality of telephone call is not appreciably diminished. The frequency band thus economized could be used to set up new temporary channels for telegraphy or data transmission.



1 Telephone handset
2 Subscriber line
3 Microphone feeding bridge
4 Variable attenuator
5 0-4 kHz low-pass filter
6 0-2.3 kHz low-pass filter
7 Filter cut-out switch

FIGURE 1. — **Mock-up of a telephone connection**

For the assessment of telephone call quality, a laboratory representation of a telephone connection was used (see Figure 1). The opinions of typical subscribers were sought according to the method recommended by the C.C.I.T.T.

The test circuit consisted of automatic telephone sets of modern design and of cable subscriber lines, represented by artificial lines each 5 km long, the wire (copper) diameter being 0.5 mm; it was also equipped with means for adjusting the line attenuation. The speech band was limited to 4 kHz. A low-pass filter with a cut-off frequency of 2300 Hz was inserted in the circuit 30 seconds or 1 minute after the call had begun and was retained until the end of the call. The attenuation of the filter was 1.051 dB in the pass-band and was 52.12 dB above 2400 Hz. The measuring team was submitted to a simple audiometric examination and was instructed in the 5-point test scale expressed by the scores 0 to 4 (B, P, F, G, E). It consisted of 40 persons, both men and women, from 18 to 25 years of age. Members of the team chose whatever subject of conversation they wished and spoke in a normal and natural manner for 3 minutes. The number of measurements was $20 \times 5 \times 3 = 300$.

The results are shown in graph form in Figures 2, 3 and 4.

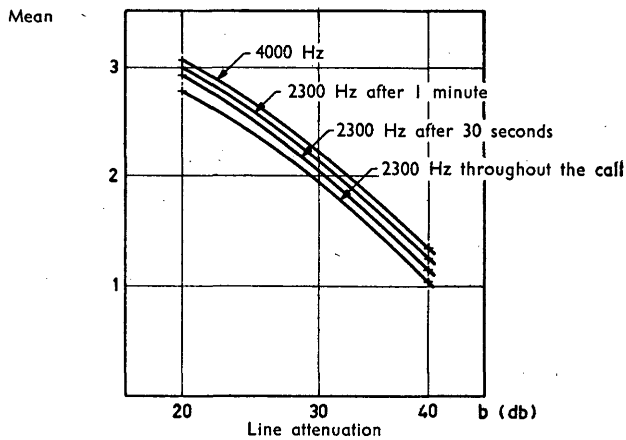


FIGURE 2. — **Variation of the mean in relation to the line attenuation**

Note. — The cut-off frequency and the insertion time delay of the low-pass-filter are used as parameters.

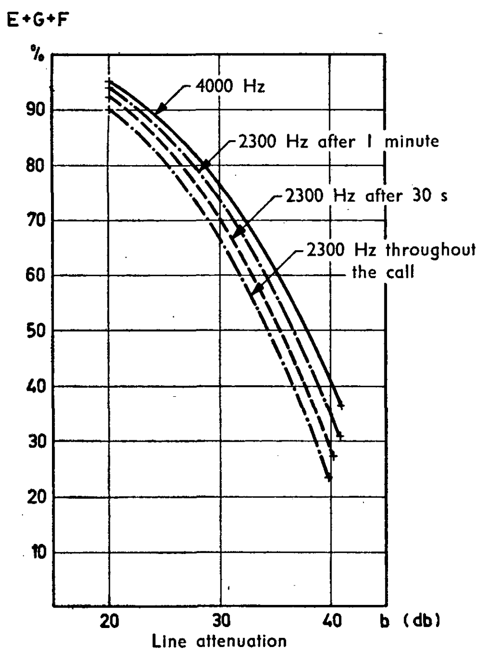


FIGURE 3. — **Percentage of " satisfied " users E + G + F in relation to the line attenuation**

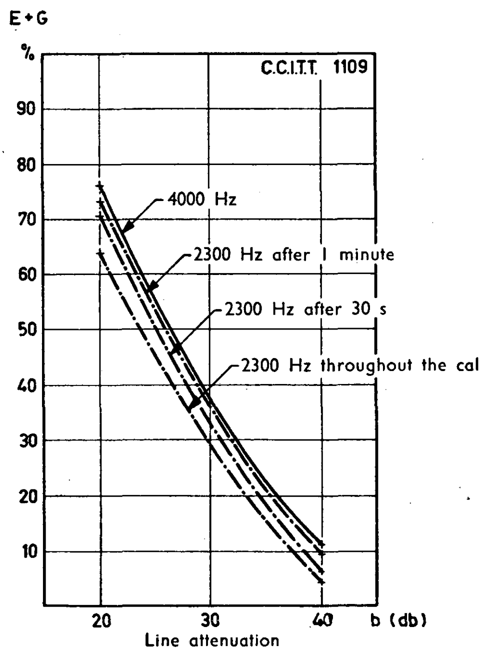


FIGURE 4. — **Percentage of " satisfied " users E + G in relation to the line attenuation**

Conclusion

The theoretical possibilities of setting up additional temporary channels were verified during a stereophonic conversation. We assume that smaller differences in transmission performance, as a function of the time when the 2300 Hz low-pass filter is inserted could be obtained at the expense of greater uncertainty as regards the subject of conversation even if the initial redundancy of the call were relatively low.

Question 3/XII — Asymmetry between the two directions of transmission

(new question)

Assuming that the existing form of Recommendation P.11 is satisfied, and

considering that various physical factors and planning practices can result in differences in overall reference equivalents for the two directions of transmission in excess of 10 dB,

a) are such asymmetries important from the point of view of subscribers using such connections ?

b) If the reply to a) is affirmative, what form of recommendation should be made to control the extent of such asymmetry ?

Note.— The reply to this Question should be transmitted to Study Group XVI for use in studying Question 5/XVI.

Question 4/XII — Effect of circuit noise on transmission performance

(continuation of Question 4/XII, studied in 1964-1968)

What is the family of curves which gives the noise transmission impairment as a function of the indications given by the psophometer standardized by the C.C.I.T.T. and for different values of the equivalent of the chain of national international circuits used ?

Note 1.— The study should be pursued with particular attention to the following points :

a) To find a simple method whereby, given the results obtained for a particular type of telephone set, it would be possible to calculate the results that would be obtained in the case of a telephone set having a different sensitivity. In this connection, attention is drawn to the article by F. Markman, published in *Ericsson Technics*, No. 2, 1960. It is desirable that administrations which may put forward such a method carry out some tests in order to verify its validity.

b) Administrations which have carried out opinion tests are asked to give the values of the speech volumes measured in these tests, if they did, in fact, measure them.

Note 2.— The method used by the United Kingdom Administration is described in the following article : D. L. Richards : Transmission performance assessment for telephone network planning, *I.E.E. Proceedings*, pages 931-940, May 1964.

Question 5/XII — Specification of sound level meters

(new question)

Are the sound level meters dealt with in I.E.C. Publications 123 and 179 satisfactory from the C.C.I.T.T. standpoint ?

If not, should the C.C.I.T.T. study a new specification ?

Note.— The C.C.I.T.T. has already issued Recommendation P.54 on the subject.

Question 6/XII — Subscribers' tolerance of echo and propagation time

(continuation of Question 6/XII, studied in 1964-1968)

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 ?

b) When using the echo tolerance curve, how should the echo balance return loss be calculated from the variation curve of the balance return loss as a function of the frequency in order to obtain an echo attenuation measurement that corresponds to the subjective effect produced by this echo ?

c) What is the maximum propagation time for which echo suppressors complying with Recommendation G.161, Volume III of the *Blue Book*, would be satisfactory ? The effects of practically occurring ranges of relevant factors such as transmission losses, return losses, circuit noise levels and types of subscribers' sets and lines, should be considered.

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 below.

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 [1].

5) Echo suppressors 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 1968.

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[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, page 1847.

Sondhi, M. M. : An adaptive echo canceller; *B.S.T.J.*, March 1967, Volume XLVI, No. 3, page 497.

Thies, A. W. and Zmood, R. B. : New ways of echo suppression; *Australian Telecommunication Research*, November 1967, Volume 1, Nos. 1 and 2, page 14.

ANNEX 1

(to Question 6/XII)

Studies to be pursued within the framework of point d) 1 of Question 6/XII

(Extract from the report on the meeting of Study Group XVI in June 1966)

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.

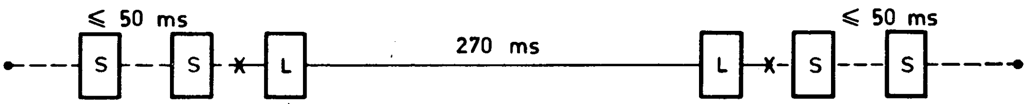


Figure 1

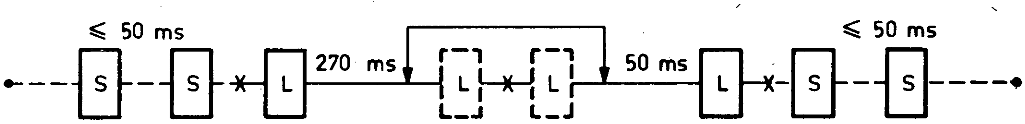


Figure 2a

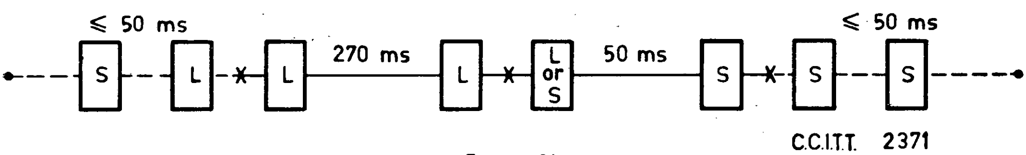
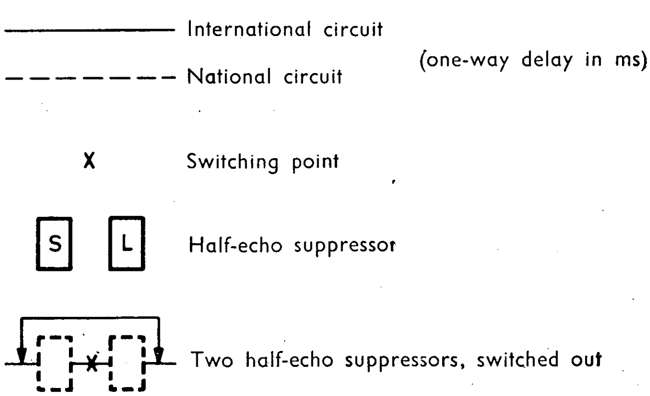


Figure 2b

Legend



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 C.C.I.T.T. Recommendation G.151.D.b.2 (43 dB or 5Np).

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)

Reply of Study Group XII in September 1968 to Question 6/XII

Part a

This part of the question must continue to be studied; for the time being Study Group XII has nothing to add to Annexes 1 and 2 to the Question in the *Red Book*, Vol. V *bis*, pages 153-158.

Part b

Study Group XII points out to Study Group XVI that the return loss for the echo can be calculated provisionally as corresponding to the non-weighted mean of the power ratios in the 500-2500 Hz band for the application of Recommendation G.122.B and the calculations described in Annex 2 of Volume III of the *Blue Book*. This can also be done by taking a quarter of the sum of the power ratios at the following frequencies, weighted as follows :

500 — weight $= \frac{1}{2}$
1000 — weight $= 1$
1500 — weight $= 1$
2000 — weight $= 1$
2500 — weight $= \frac{1}{2}$

Annex 3 to Question 6/XII in the *Red Book*, Volume V *bis*, pages 158-162, describes another calculation method, the results of which have been compared with those obtained with subjective tests; administrations are invited to carry out comparative measurements and calculations to enable Study Group XII to recommend a single method.

Part c

Study Group XII considers that, for propagation times not exceeding 50 ms in one direction, existing echo suppressors having the characteristics specified in Recommendation G.161.B (*Blue Book*, Volume III) can be used unchanged.

For propagation times between 50 and 150 ms, these echo suppressors can continue to be used provided that their circuit arrangements are modified in particular so as to increase the blocking attenuation and sensitivity.

Above 150 ms, use should always be made of echo suppressors conforming to the new specifications drawn up by Study Group XV (Recommendation G.161, Volume III of the *White Book*). Study Group XII notes that, in view of the direction taken by the work of Study Group XI, it should also be possible to use these long-propagation-time echo suppressors for medium delay transmission times.

Part d[1]

1. *Presence of several circuits equipped with echo suppressors*

Tests have been made by the Telephone Association of Canada [1], the United Kingdom [3] and the A. T. & T. Co. [4] in which two circuits were connected in tandem, each circuit having a separate pair of echo suppressors.

In each test, values of mean one-way propagation time for one circuit were taken up to that of a synchronous-orbit satellite and these circuits were equipped with echo suppressors designed for such conditions. The second circuit had modest values of mean one-way propagation time up to 30 ms and was, in each case, equipped with echo suppressors intended for such shorter propagation times. Under these conditions there was no statistically significant degradation over that due to the long propagation time circuit alone.

Information on three or more interconnected circuits with separate echo suppressors has not been contributed.

2. *End-delay*

The specification of a new echo suppressor, formulated by Study Group XV, provides for end-delays up to 25 ms round-trip and return losses as low as 0 dB.

If both parameters are appreciably less favourable than this in a given application degradation will result.

Trade-offs can be made between end-delay and return loss, less favourable values of one being acceptable if the other is correspondingly improved. Numerical values defining this relationship should be established with the assistance of Study Group XV, particularly for the 36 ms round-trip end-delay case of interest to Study Group XVI.

3. *Losses on circuit extensions*

Studies either with controlled speech levels or with controlled losses [2] have shown that asymmetry of these parameters increases the degradation.

Further information is needed on this point and a practical range of speech volumes from the two ends should be covered. A method which should be considered for these investigations is described in a contribution of the United Kingdom [5].

4. *Return loss*

The relationship between end-delay and return loss has been treated under heading 2 above. It can be said, further, that higher return loss might permit the design of more effective echo

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[1] Points 1 to 6 correspond to the former wording of this part of the Question (*Red Book*, Volume V *bis*, page 153).

control. Appropriate methods are being explored by various administrations and further information is invited from administrations on this point.

5. *Circuit noise*

The new echo-suppressor specification formulated by Study Group XV will ensure satisfactory operation for the levels of noise likely to be encountered on international circuits designed in accordance with C.C.I.T.T. Recommendations.

6. *Echo suppressors of different types*

The joint tests by the United Kingdom [3] and the A. T. & T. Co. [4] have demonstrated that echo suppressors designed independently for application to circuits having long propagation times can be applied at opposite ends of an international circuit with performance which is not significantly different from that obtained with pairs of the same type.

The objective of Study Group XV work on new echo suppressors is that such compatibility be achieved. It is desirable that there be methods for measuring the performance of combinations of echo-suppressors under laboratory conditions to establish that this has been accomplished.

Two means that have been designed for such measurements are described in contributions to Study Group XII [5], [6].

7. (*Additional point*)

The Working Party had nothing to add to the earlier conclusion that customer experience with satellite circuits had no statistically significant effect on customers' judgement of the quality of the circuits they are using; this was stated in document COM XII—No. 39 (1964-1968), page 5, last paragraph, and is supported by evidence in references [3] and [4].

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 RICHARD, 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 1964-1968)

a) What criterion should be adopted to determine equality of performance of different transmission systems ?

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 gives directives on the method to be followed in studying this question.

Note 2.— Attention is drawn to the importance of using suitable methods for preparing test programmes and for analysing 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 new Question 2/XII.

ANNEX

(to Question 7/XII)

Directives for the study of Question 7/XII

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 21/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, 16 and 21/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). Methods of this type are also described in the Annexes to Question 2/XII.

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]). 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 [3], [10] and Annex 6 to Question 15/XII, 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 Contribution COM XII—No. 87 (period 1964-1968).

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 British Administration 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]. 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.

REFERENCES

[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; article published in the *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.

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

Question 8/XII — Measurement of the sensitivity of a carbon microphone

(continuation of Question 8/XII, studied in 1964-1968)

The shape of the sensitivity/frequency characteristic of a sending system depends greatly 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 1.— By way of information, the Annex below gives the principal experimental conditions which ought to be observed when measuring the " sensitivity-frequency " characteristic of a transmitter system comprising a carbon microphone.

Note 2.— The various methods described in Annexes 27 to 31 (*Red Book*, Vol. V, Part II) should also be applied to telephone systems other than those to which they have already been applied.

Note 3.— Tests with various artificial mouths and ears were carried out in the C.C.I.T.T. Laboratory in connection with the study of Question 12/XII (see Annex 2 to Question 12/XII).

Note 4.— The attention of administrations is drawn to the following documents (period 1964-1968):

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 Company)

The provisional standard appearing in document No. 269 of the *I.E.E.E.*, May 1966, entitled " Proposed method for measuring transmission performance of telephone sets ".

Note 5.— The attention of administrations is also drawn to the importance of the period elapsing between the treatment of the carbon microphone and the beginning of the measurements. The experience of the A.T.T. in this respect is summarized in the following extract from the standard mentioned in Note 4 (extract reproduced in COM XII—No. 95 (1964-1968), p. 7) :

" 5.2.2 The measurement of transmit should comprise the following steps :

. . . .

3) rotate the transmitter, clockwise and counterclockwise, through a 180-degree arc, returning to the test position,

4) measure the telephone set output for the specified input sound pressure level; any or all of three test conditions may apply :

. . . .

b) slow-sweep measurements : here the measurement should begin within 30 seconds of the completion of conditioning of the transmitter, see paragraph 5.2.2.3. Not more than 4 minutes should be allowed for the completion of the test run. In case of question the transmitter should be reconditioned and the test run repeated . . . . . "

Administrations are invited to indicate their experience after taking such precautions.

ANNEX

(to Question 8/XII)

Principal experimental conditions which should be taken into account when determining the sensitivity-frequency characteristic of a sending system having a carbon microphone

(Contribution from the Administration of the German Federal Republic)

The microphone attenuation distortion (sending system) of a subscriber's set (sensitivity-frequency characteristic) is the distortion of the e.m.f. of the subscriber's set in relation to the frequency in case of acoustic radiation. Since we are here dealing with a carbon microphone, this attenuation distortion is influenced by :

1. the acoustic field (the artificial mouth),

2. the magnitude of the acoustic pressure,
3. the recording speed,
4. the preliminary conditioning of the microphone,
5. the measuring system used.

1. So that repeated measurements and measurements made at various places give the same results, the acoustic field to which the microphone is subjected must have definite, exactly defined characteristics. This calls for a spherical acoustic field. This necessitates a small size for the sound source and a short distance from the measurement point. As a source, and to create the spherical acoustic field, an artificial mouth is used. A room with high acoustic absorption is not needed to make measurements with the artificial mouth, but care must be taken to avoid nearby sound-reflecting surfaces. It is important that the acoustic pressure should vary in inverse proportion to the distance from the source, for all frequencies within the range of measurement and within an angle of radiation of about 90°.

Measurement with an artificial mouth offers, relative to that in a flat acoustic plane, the advantage that it reproduces, with reasonable accuracy, the conditions obtaining when somebody speaks, because the sound radiated by the human mouth, too, forms an acoustic field which may be assumed to be spherical for all practical purposes.

For measurement, the microphone is placed in the handset (with the mouthpiece) and is introduced into the acoustic field in such a fashion that the diaphragm is vertical.

2. Since there exists a non-linear relation between the e.m.f. of a carbon microphone and the acoustic pressure, the measurement must be made at an acoustic pressure of ten dynes/cm^2, constant for all frequencies. Hence the artificial mouth should be so constructed that it gives the same acoustic pressure at all the frequencies in the measurement range or else means of making the requisite adjustment of the voltage applied should be available. To measure the acoustic pressure, a condensor microphone is used; this must be very small, so that there is no notable accumulation of acoustic pressure in front of the diaphragm, even at the highest measurement frequencies. A pressure of ten dynes/cm^2 more or less corresponds to the volume of the vocal sounds in determinations of the reference equivalents made with the S.F.E.R.T.

3. In order to determine exactly all the resonances and all the anti-resonances in the sensitivity-frequency characteristic, this latter must be recorded with a continuous variation of the frequency on some recording device or made visible on a cathode ray tube (and even be photographed). The time during which the entire band is scanned must not be too short, so that no transient phenomena may occur. Nor must it be too long, or else the microphone sensitivity may vary.

4. To get reproducible measurement results, the microphone must first be subjected to a definite pre-conditioning. In so doing, the object is to ensure that the grains of carbon offer the same electrical resistance and exert the same influence on the electrodes and the diaphragm at each measurement. The following procedure has proved a reliable one :

a) the microphone capsule is placed in the handset;
b) the feeding current is connected;
c) the handset is turned through 270°, thrice forward and backward from the starting position (the diaphragm being vertical) and this in such a way that the vertical position is followed first of all by the horizontal position in which the layer of granulated carbon

is below the diaphragm. The handset must be turned in such a manner that the force of gravity acts only on the carbon layer. Accelerations and shocks are to be avoided;

d) the sound source is then switched on. Before a) takes place, the acoustic pressure is to be adjusted to ten dynes/cm^2, so that the microphone is subjected to a higher acoustic pressure;

e) the measurement frequency is made to vary several times from its lowest to highest value, and vice versa;

f) the sensitivity-frequency characteristic is recorded;

g) the sound generator is then switched off;

h) the operations c) to g) are then repeated.

With this method, it has been observed that there is, in general, never more than 2 dB difference between two extreme curves.

5. Measurement is effected with the normal arrangement of the subscriber's set and exchange. Normally speaking, a subscriber's cable must not be interposed.

Question 9/XII — Limits applied to national trunk and local networks

(continuation of Question 9/XII, studied in 1964-1968)

(documentary question)

What are the limits applied by your Administration to the national trunk and local networks of your country in order to ensure satisfactory quality for national calls, it being understood that the recommendation of the C.C.I.T.T. relating to reference equivalents is satisfied for international calls ?

Note.— The documentation already received is mentioned in Recommendation P.21 and reproduced in a supplement in Volume V of the *White Book.*

Question 10/XII — Increase in the sensitivity of local systems

(continuation of Question 10/XII, studied in 1964-1968)

(documentary question)

a) 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.

If transmitting and receiving sensitivities of subscribers' apparatus are increased above levels at present prevailing, what will be the consequences in regard to :

1) the subscriber's behaviour;

2) the performance of transmission systems ?

b) What is the relationship between speech volume, measured at a suitable point in a telephone connection, and the send reference equivalent between the talking subscriber and the measuring point ?

Note 1.— In order that the replies to this question may be presented in a uniform way, it is recommended that the speech levels be expressed in v.u. These levels, and the statistical distribution law at present found acceptable for sending, should be established from measurements carried out at local exchanges and/or international exchanges [1].

Note 2.— It is desirable to study the effect upon volume in service and its distribution when subscribers are provided with telephone sets which are more sensitive than those to which they have become accustomed. By way of information, the transmission features which are liable to be affected by the introduction of more sensitive telephone instruments are given below :

a) non-linear distortion and cross modulation in multichannel transmission systems;
b) crosstalk in cables and lines;
c) inconvenience due to the increased audibility of echo;
d) signal imitation;
e) sidetone and singing due to coupling between transmitter and receiver;
f) room noise impairment.

Note 3.— Annex 1 below deals with the effect which increased sensitivity of subscribers' telephone sets might have on the subscribers' behaviour. The section (contributed by the Helsinki Telephone Company) in the supplement to Volume V of the *White Book* contains some comments on the use of telephone sets with automatic regulating devices.

Note 4.— Annex 2 gives the results of tests carried out by the Administration of the German Federal Republic.

Note 5.— Annex 3 reproduces the reply made in 1967 to this Question by Study Group XII. Annex 4 indicates the relationship between the work of Study Groups XII and XVI in this field.

ANNEX 1

(to Question 10/XII)

Effect of the increase in efficiency of telephone instruments on subscribers' behaviour

As far as the subscriber's behaviour is concerned, the effect of an increase in the transmitting and receiving efficiency of subscriber's telephone sets, when it exceeds the standards normally encountered, can be considered from two different aspects :

a) the effect produced by these increases on the volume of speech sounds in the line,

b) the effect on the opinion of the subscriber using that line.

Effect on the volume of speech sounds

The variations in the volume of speech sounds must be considered bearing in mind the effects of the following factors :

I. Microphone efficiency.

II. Sensitivity of the sidetone path.

Note.— This will depend on the efficiency of the microphone, the efficiency of the receiver, and on the attenuation of the anti-sidetone circuit. This latter will depend on the line impedance (on its composition and its length).

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[1] *Note by the Secretariat* — These measures will be co-ordinated when Question 11/C is studied (see Circular C.C.I.T.T. No. 10 of 28 January 1969).

III. Intensity of the speech sounds received.

Note.— This depends on the transmission efficiency of the talker's set, the total attenuation of the circuit, and the receiving efficiency of the listening subscriber's set.

IV. Circuit noise perceived by the listening subscriber.

V. Level of room noise, chiefly at that end of the circuit at which the speaking subscriber happens to be.

Effect on subscribers' opinions

An excessive sidetone, due to increase in transmission and reception efficiency, and an excessive increase in the volume of the speech sounds received, will unfavourably affect the opinion expressed by the users.

ANNEX 2

(to Question 10/XII)

Consequences of increased efficiency of microphone capsules

(Contribution by the Administration of the German Federal Republic)

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 German Federal Administration were used. The users of these sets were not employed in the technical telephone services.

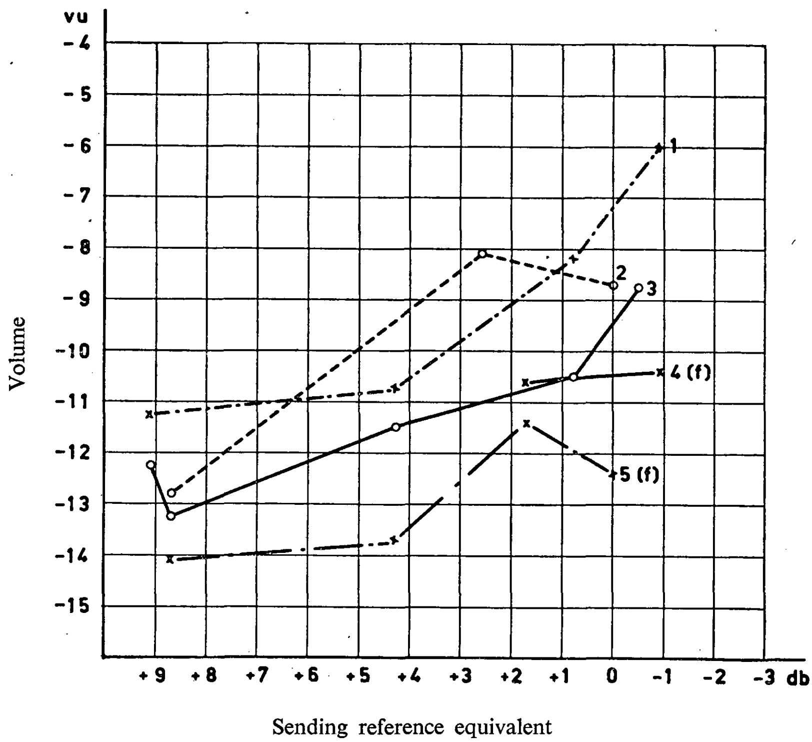
These telephone sets were equipped with microphone capsules having reference equivalents between — 1 dB and + 9 dB. The users of the sets in question usually made calls only within the building, or trunk calls. There was a percentage of about 40 to 60 % trunk calls (an average of about 55 %). The percentage 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

ANNEX 3

(to Question 10/XII)

Reply by a Working Party, approved by Study Group XII

(October 1967)

Increased sensitivity of telephone sets

Increased sensitivity of telephone sets tends to be used for increasing the subscribers' line losses that can be permitted without exceeding the maximum values of reference equivalent for the national system. An increase in line loss of Δa must however be accompanied by increases of 2 Δa in both sidetone balance return loss and crosstalk attenuation. This necessity will probably limit the extent to which it will be possible in future to increase the sensitivity of subscribers' sets.

Measurements of speech power

1. There is a need for more extensive information on the relationship between speech volume, measured at a suitable point in a telephone connection, and the send reference equivalent between the talking subscriber and the measuring point. This relationship is dependent to some extent upon other characteristics of the complete connection but information published and related to the United Kingdom network [1] shows the mean power level for a median talker while active to be -15 dBm when the send reference equivalent is $+7$ dB. (This applies when the overall reference equivalent of the connection is in the range 15 to 25 dB). Information expressed in vu applying to the Federal Republic of Germany is given in Annex 2 above. The Working Party considers that administrations should be asked to supply more information on this relationship. The information can be obtained from a survey of actual telephone calls. The information is needed in connection with the measurement programme for speech power [2], and for other purposes (see Question 4/XII, Note 1.b).

2. The results already collected—in connection with the measuring programme for speech power executed during the period 1964-1968—were studied (see document COM XII—No. 96 (1964-1968)). The Chairman of the Working Party explained that the results quoted for the United Kingdom were the levels of mean power for a talker corresponding to the median of the distribution law and not the levels of mean power averaged over the whole distribution of talker volumes. The Chairman asked representatives of the other administrations whose results are quoted in document COM XII—No. 96 to say whether or not this applied to their results also (there is still some uncertainty on this point [3]). To enable loading of multichannel systems to be estimated there is need also of the corresponding standard deviations of the speech volume distributions and of the activity factors, together with the definitions that had been used to distinguish " active speech " from periods of " inactivity ". The definition used by the A.T.T. for its measurements is given in document COM XII—No. 46 (1964-1968). The Chairman explained that this differed from that used by the United Kingdom, and that the activity factors were therefore different. It was agreed that there was need for administrations, etc., to provide additional information on these matters before the data could be relied upon for estimating speech loading of multiplex systems.

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[1] D. L. Richards, " Transmission performance assessment for telephone network planning ", *Proc. I.E.E.*, *111*, 1964, pages 931-940.

[2] See the note by the Secretariat at the foot of page Question 10/XII, p. 2, above.

[3] See Appendix below.

The problem of relating results of measurements of speech power to a common basis of comparison was discussed but no conclusions were reached (see, however, the suggestion made by the Chairman [1]).

Reference equivalent of the national sending system

It is suggested in document COM XII—No. 38 (1964-1968) that the mean nominal value of the reference equivalent of the national sending system should, desirably, be equal to or exceed 14 dB. This view is supported in document COM XII—No. 58 (1964-1968) by N. V. Philips, where it is estimated that such a value corresponds to a long-term mean power per channel of 22 microwatts for a multiplex system (see Recommendation G.223). On the other hand, document COM XII—No. 89 (1964-1968) includes some estimates of the mean value of send reference equivalent for the United Kingdom national system derived from speech volume measurements; these send reference equivalents are of the order 10-12 dB. There is no evidence that such mean values result in any overloading of multiplex equipment in the United Kingdom designed according to Recommendation G.223 and so it is concluded that such values are acceptable. The reason for this apparent discrepancy was discussed and it was concluded that it could not be resolved without further study of the methods employed in estimating the loading of multichannel systems. The mathematical and statistical framework used for estimating the required load capacity when given the results of measurements of speech power and activity factors involves several assumptions based on rather old information concerning the statistical structure of speech signals emitted by telephone microphones during telephonic conversation and concerning the characteristics of multichannel systems near their overload point. The working party considered that the basic information should be brought up-to-date and the whole mathematical and statistical framework reviewed, especially for cases where modest numbers of channels are involved.

Appendix (to Annex 3)

The following remarks by the Chairman of the Working Party may assist in the interpretation of speech power measurements.

Speech power measurements have been made by different administrations according to various methods, some of which are given below.

A. Speech volume transmitted by a given talker in a conversation. The result is expressed, in dBm, as the level of the mean power for that talker while active. The definition of " active " is necessary together with a statement of the " within-conversation " activity factor. The results of a series of measurements on a telephone channel used successively by different talkers, or of a series of measurements on different channels of a multiplex system, yields a distribution of speech volumes, which is commonly found to be Gaussian, defined by the mean and standard deviation. Separate determinations are needed of the proportion of time for which the channel is connected to subscribers.

B. Mean power of all speech signals occurring within a defined interval of time on a given telephone channel. The time interval may be taken as including only " channel-busy " time, " subscriber-conversation " time or " active-speech " time. In any case the results may be expressed in terms of the mean value of the readings expressed in dBm and the standard deviation of such readings. The appropriate activity factors are also required.

[1] See Appendix below.

C. Mean power of a multiplex system of several speech channels averaged over a given total time. If the total time is long enough or if the number of channels is very large this quantity is the same as that for which a value of 22 microwatts is quoted in Recommendation G.223. If the total time for which each reading is obtained is not very long (say a few minutes or less) and the number of channels is small, the readings will be appreciably dispersed and appropriate corrections will be necessary to obtain the long-term mean power.

The distribution of readings obtained by the methods A and B cannot be directly compared and one cannot be converted to the other without assumptions. Both types of results may be used, however, to estimate the long-term mean power of a multiplex system carrying speech signals, which is given directly by C. This requires, in both cases, certain further assumptions which can be expressed by means of mathematical and statistical formulae. An approximation valid for systems having a very large number of channels is given by adding, to the mean value of the levels expressed in dBm, the quantity 0.115 (standard deviation)^2 and subtracting an allowance for the appropriate overall value of activity factor. In applying this approximation to results by method A, the appropriate activity factor must include the proportion of total time for which the circuit is available for subscribers' conversation; this is approximately the proportion of revenue-earning time of the circuit and will be less than the time for which the circuit is marked as " busy ".

As an example, the results given in document COM XII—No. 89 (1964-1968), in accordance with method A, yield a mean level of − 15.8 dBm0 and a standard deviation of 5.7 dB. The " within-conversation " activity factor is about 0.43 and the fraction of total time during a busy-hour for which a given channel is actually in use by subscribers for conversation is about 0.62.[1] The fraction of total time that a channel is occupied by active speech is therefore 0.27. On the assumption that the multiplex system contains a very large number of channels, the level of the long-term mean power is equal to :

$$\begin{aligned} & -15.8 + 0.115\,(5.7)^2 + 10 \log_{10} 0.27 \\ = & -15.8 + 3.7 - 5.7 = -17.8 \text{ dBm0} \end{aligned}$$

which corresponds to 16.6 microwatts.

This figure does not include speech by operators.

It would seem prudent to convert the results of measurements by other methods to appropriate estimates of the long-term mean power per channel assuming multiplex systems having a very large number of channels.

The load capacity needed by a multiplex system intended to carry only a modest number of channels must take account also of the short-term fluctuations in mean load. In any case, allowance must be made for the fact that the peak load is considerably higher than even the higher values of the short-term mean load. Further mathematical and statistical framework (involving assumptions) is needed for such estimation. See, for example, D. L. RICHARDS : " Statistical properties of speech signals ", *Proc. I.E.E., 111*, 1964, pages 941-949.

ANNEX 4

(to Question 10/XII)

Extract from the reply of Study Group XVI to point 6 of Question 1/XVI

(Geneva, 24-31 October 1967)

1. Study Group XVI took note of document COM XVI—No. 86 (1964-1968), and the reply to Question 10/XII (Annex 3 above) which was made by Study Group XII in October 1967 and contains observations on the problem as a whole.

[1] The fraction devoted to operator speech was 0.12; to signalling, 0.13; complete inactivity, 0.13.

Measurements of the power of signals transmitted over telephone circuits can be carried out for various reasons, for example :

— to find out the effective load of wideband systems—this involves determining the long-term mean and the dispersion of the power *per channel*;

— to find out the power (mean and dispersion) *per call*, for example to try to establish a relationship between this power and the reference equivalent at the transmitting end.

The first of these aims should be pursued by Special Study Group C on behalf of Study Group XV; the second mainly concerns Study Group XII.

Study Group XVI should make use of the conclusions of these two other Study Groups (Special Study Group C and Study Group XII) in establishing or improving the transmission plan.

. . . .

4. In general, Study Group XVI is considering transferring this problem to Special Study Group C, which will be able to take a decision, with Study Group XII, on a measurement programme to be carried out under clearly defined conditions for the next C.C.I.T.T. period. It is suggested that this measurement programme should aim at providing data of specific interest to each Study Group [1].

Question 11/XII — Limits for intelligible crosstalk

(new question) [2]

A. Is it possible to establish a threshold value for crosstalk attenuation between two telephone circuits, below which crosstalk from a single interfering speech signal may be regarded as intelligible ?

Note 1.— The answer to the Question will be used to review the objectives for crosstalk attenuation (from the point of view of intelligibility) in cables, circuits, transmission systems and exchanges.

Note 2.— The answer should include consideration of at least the following variables :

a) the level of the wanted speech signal;
b) the level of psophometric noise associated with the wanted speech signal;
c) the level of the interfering signal;
d) the receiving reference equivalent of the listening subscriber's line and telephone;
e) the attenuation distortion in the crosstalk path;
f) the method of measurement or calculation which should be associated with the threshold value of crosstalk attenuation.

B. Pending a complete answer to part A, can the following values of crosstalk ratio *between international circuits* be regarded as acceptable, assuming the use of modern telephone instruments in local networks :

a) 58 dB in the presence of a psophometrically weighted noise power referred to a point of zero relative level of 500 pW (—63 dBm0p);

b) 62 dB in the presence of a psophometrically weighted noise power referred to a point of zero relative level of 200 pW (—67 dBm0p)?

If these values are not acceptable, what figures might be proposed ?

[1] See the note by the Secretariat at the foot of page Question 10/XII, p. 2, above.

[2] Reply to be transmitted to Study Group XVI.

Note 1.— The figure for crosstalk given in a) is based on Recommendation G.151, paragraph D.a. It is assumed that intelligible crosstalk from a single disturbing circuit must all arise within a single homogeneous section of a hypothetical reference circuit, and hence masking noise cannot be relied on to be greater than will be produced in this single homogeneous section. In the busy hour, in accordance with Recommendation G.222, this noise is allowed to have a maximum value of 10 000/9 pW = 1100 pW, but at times of light traffic and allowing for system design margins, the noise may in practice well fall to as low a value as 500 pW.

Note 2.— The figure in b) is derived by considering a very short circuit of the type envisaged in Recommendation G.125. Here the crosstalk may be assumed to be largely determined by the two terminal equipments. If these contain channel modulators each just complying with Recommendation G.232, paragraph J.a, and their contributions are added on a power basis, we obtain the figure quoted. The maximum noise during busy-hour conditions is governed by Recommendation G.125.a (as revised), Volume III of the *White Book*, but during periods of light traffic and allowing for design margins, it might be as low as 200 pW.

Question 12/XII — Artificial voices, mouths and ears

(continuation of Question 12/XII, studied in 1964-1968)

a) How should Recommendation P.51, concerning the artificial ear advocated by the C.C.I.T.T., be amplified ?

b) What general characteristics should be fixed for artificial voices and mouths ?

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).

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 *White 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)

Tests made with artificial mouths

The programme of tests on artificial mouths mentioned in paragraph 2.2 of the former Annex 2 to Question 12/XII (*Red Book*, Volume V *bis*, pages 212-215) was carried out by the C.C.I.T.T. Laboratory. The results are given in contributions COM XII—Nos. 54, 81, 90 and 106, and temporary documents Nos. 1 and 4 (1964-1968 period).

The increased acoustic pressure created by the presence of an obstacle placed in front of the lips of the United Kingdom artificial mouth is shown in Figure 1.

The same measurements carried out with other models of artificial mouths agree well with those obtained with the United Kingdom artificial mouth.

Study Group XII decided that, until an artificial mouth is standardized, the C.C.I.T.T. Laboratory should make such tests with the United Kingdom model, several copies of which are available, one being permanently assigned to the Laboratory.

Question 13/XII — Non-linear distortion of telephone apparatus

(continuation of Question 13/XII, studied in 1964-1968)

(documentary question)

Collection of information :

1. on the effects which the non-linear distortion of a subscriber's telephone apparatus has on the quality of telephone transmission;

2. on methods of measuring the non-linear distortion of a subscriber's telephone apparatus; and

3. 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 of Volume V of the *Red Book*). The following Annex contains a more recent contribution to the study of this question. For additional information, administrations may refer to Contribution COM 12 — 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.

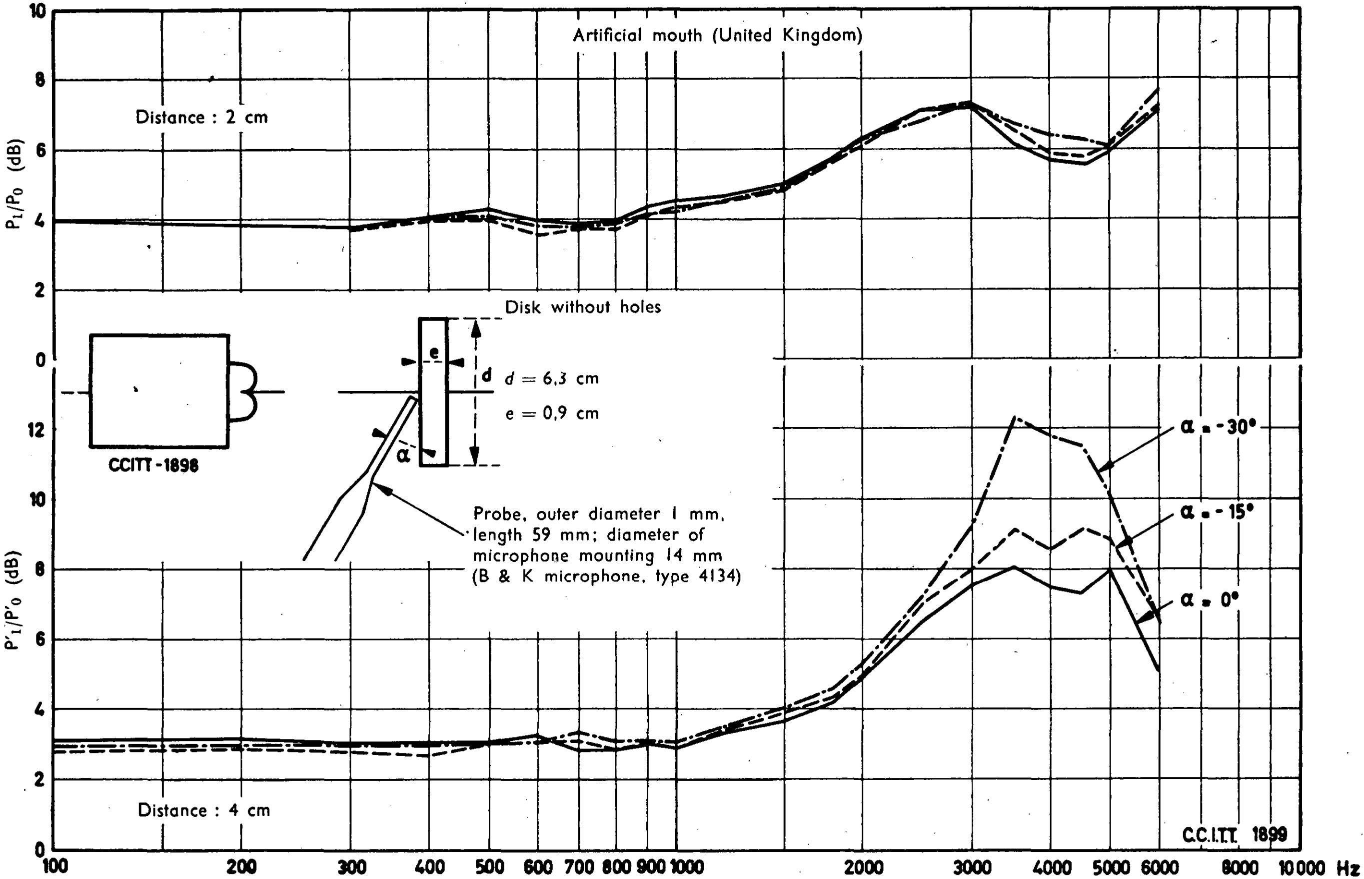


FIGURE 1. — Increase in sound pressure resulting from the presence of a disk, 6.3 diameter (20 $\log_{10} P_1/P_0$)

ANNEX

(to Question 13/XII)

Contribution by the Italian Administration

Item 1

Transmission performance tests were carried out on a complete telephone communication in which the non-linear distortion and reference equivalent could be modified at will. The circuit diagram is shown in Figure 1, where it will be seen that the carbon microphone was replaced by a receiver capsule suitably amplified and corrected.

Figure 1 also shows the circuit producing the distortion, brought about with a rectifier inserted in the third winding of a transformer. The distortion depends on the level of the signal, in accordance with the curves shown in Figure 2. An attenuator at the input of the distortion circuit, with an amplifier and an attenuator at the output of this circuit, renders it possible to modify the distortion independently of the value selected for the reference equivalent of the complete circuit.

Opinion tests were performed with a 50-dB room noise (Hoth spectrum), and the figures obtained are shown in Figure 3, which gives the mean score for various values of reference equivalent and distortion. Figure 4 shows the reduction in transmission performance, in dB, as a function of distortion for various values of the reference equivalent.

Item 2

This point was studied with particular reference to the non-linear distortion produced by four kinds of microphone capsules in common use at the present time with a feeding

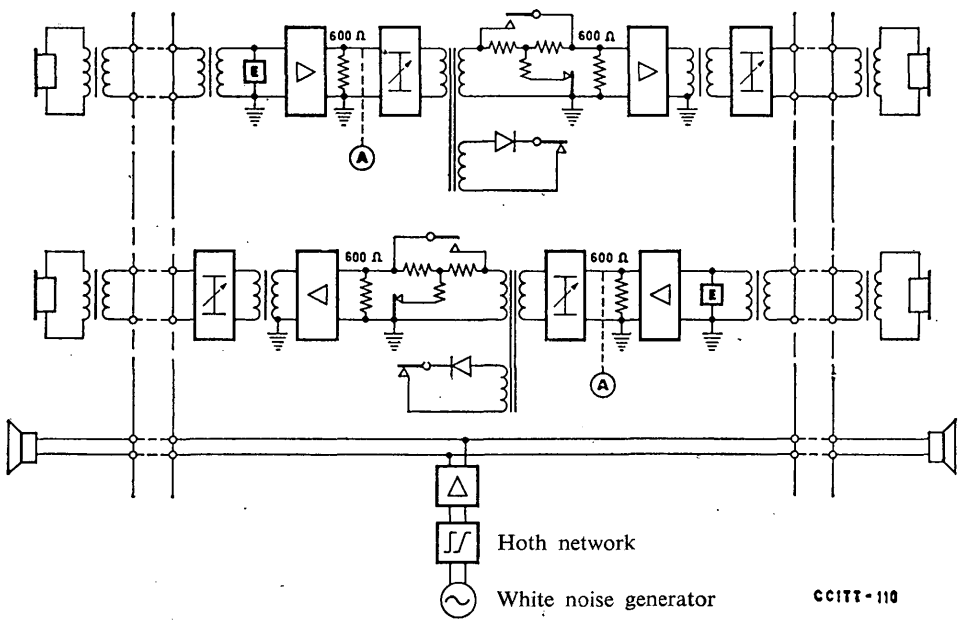


FIGURE 1

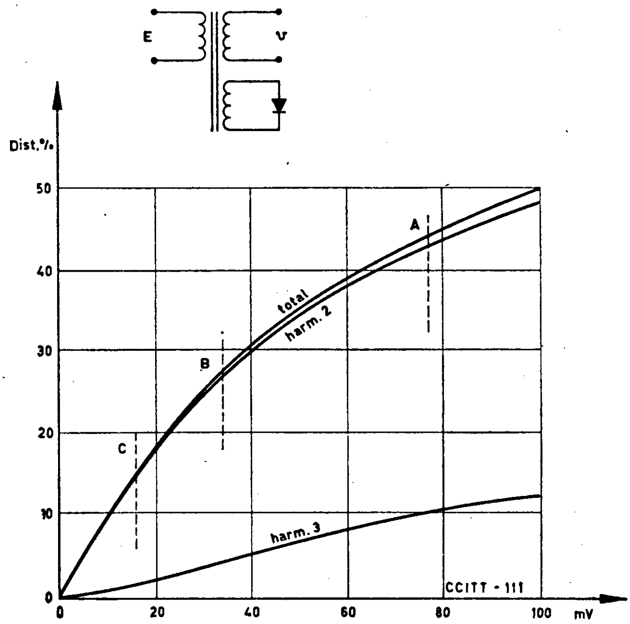


FIGURE 2

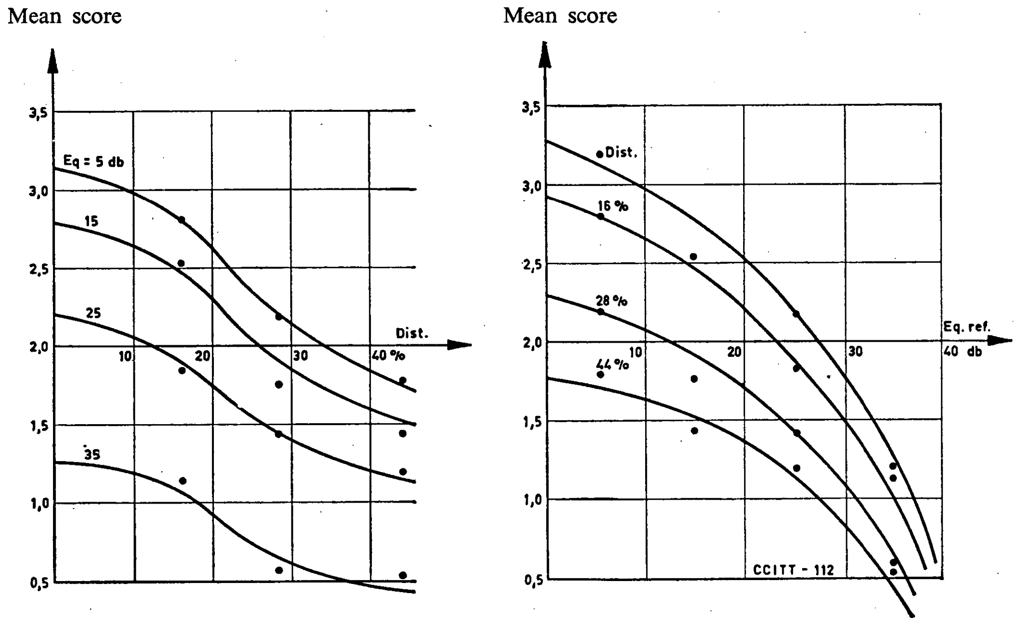


FIGURE 3

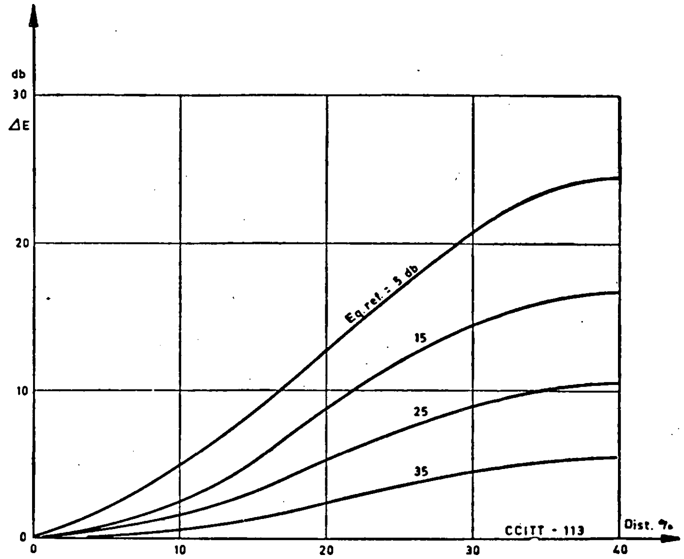


FIGURE 4

current corresponding to conditions of use in a telephone set connected to a feeding bridge by a zero line.

The microphones were energized by a source of constant acoustic pressure at various frequencies; the pressures used were 10, 30 and 100 dynes/cm^2. The experimental circuit is shown in Figure 5.

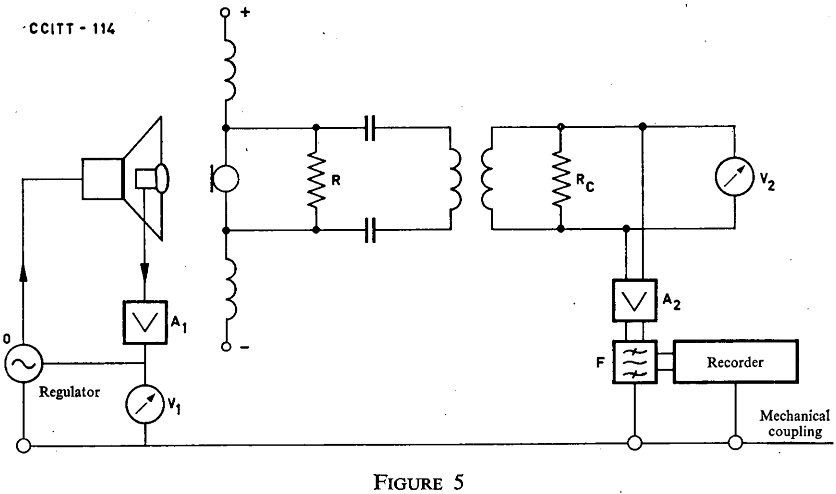


FIGURE 5

The following two methods were used :

1) Microphone energized at the fixed frequencies of 300, 500 and 1000 Hz, care being taken to tap the microphone in a uniform manner at the beginning of the measurement which was performed as rapidly as possible. At the output of the circuit, the voltage of the fundamental and the second and third harmonics were measured.

2) Microphone energized with continuous sweeping of frequencies between 100 and 5000 Hz lasting some 10 seconds. In this case no preliminary mechanical treatment of the microphone was necessary. The circuit output was connected to a level recorder preceded by a variable frequency band-pass filter, synchronized with the sweep rate, either in step with the sweep frequency or spaced from it by one octave or two octaves. Thus three representative curves were obtained for the fundamental, second harmonic, and third harmonic.

The figures obtained with these two methods are fairly constant and coincide fairly well for type A, while for types B, C, and D only the second method gave fairly constant results. Furthermore, type A gave higher distortion readings than types B, C, and D, which more or less coincide. The results are shown in Figure 6.

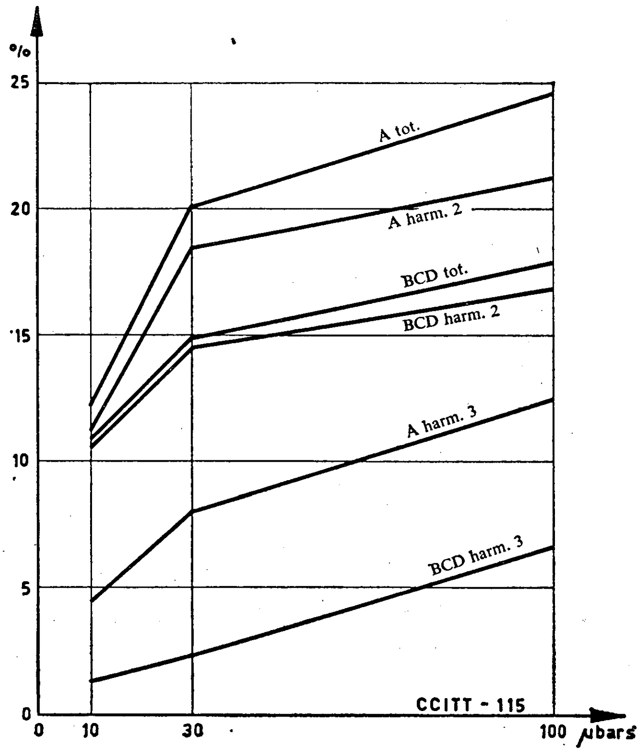


FIGURE 6

Item 3

The noise level produced by the carbon microphone in the above test conditions was measured and found to be, on the average, — 76 dBmp, with a standard deviation of 2 dB.

It is thought that, considering its effect on transmission performance, this noise, either on account of its level or on account of the speech level, may be regarded as a circuit noise.

In this context reference may be made to the previous contribution by Italy (Annex 1 to Question 4/XII, *Red Book*, Volume V, pages 579 to 594).

Question 14/XII — Extension of the bandwidth transmitted

(continuation of Question 244/XII studied in 1964-1968)

How much improvement in quality of transmission can be obtained by reducing the attenuation/frequency distortion in the lower-frequency range of the band of frequencies transmitted by international and national trunk circuits ?

Note.— In studying this Question, the following factors should be taken into consideration :

1. If such reduction in attenuation/frequency distortion were achieved, the susceptibility to interference by low-frequency noise would be increased, particularly so far as power-frequency harmonics are concerned.
2. The speech power to be transmitted by carrier systems would be increased and the statistical characteristics would be changed.
3. The adverse effects of phase/frequency distortion caused by the need for a very sharp cut-off when out-of-band signalling systems are used.
4. The increased power in the low-frequency region would coincide with those frequencies at which it is difficult to ensure good balance return losses.

Question 15/XII — Measurement of ratings based on loudness

(continuation of Questions 5/XII and 15/XII, studied in 1964-1968)

1. Several methods are used by various administrations to measure loudness ratings in their laboratories, for example :

a) the reference equivalent defined by the C.C.I.T.T. and used in the C.C.I.T.T. Laboratory;

b) the O.B.D.M. equipment used by the Administration of the German Federal Republic and in the laboratories of other administrations;

c) the E.A.R.S. equipment used by the A. T. & T. Company;

d) path through one metre air-path method, based on the A.R.A.E.N. and used by the United Kingdom Administration.

2. Can the results obtained from these diverse methods be related by theoretical considerations in respect of the following quantities :

a) overall rating of a subscriber's set and line connected through an attenuator to another subscriber's set and line;

b) sending rating of a subscriber's set and line;

c) receiving rating of a subscriber's set and line;

d) sidetone rating of a subscriber's set connected to a given line ?

If so, then what ought the relationship to be ?

3. If theoretical relationships can be derived it would be very desirable to check that they apply accurately enough in practice. This would be best achieved by co-opera-

tion between the laboratories of administrations and the C.C.I.T.T. Laboratory to secure measurement of various types of subscribers' sets by all these methods.

4. If no theoretical relationship can be derived, the only way of obtaining a useful empirical relationship would be by undertaking a series of measurements on different kinds of sets. A plan for the exchange of stable telephone sets by different laboratories was prepared in 1967 by Study Group XII.

5. The methods enumerated in 1 and other similar methods not specifically mentioned have each their various advantages and disadvantages and they should all be studied with the object of deriving a single method which embraces the best features of each. In seeking to determine the best features, it should be borne in mind that, although in many cases it might be necessary to measure ratings by subjective methods, the ultimate aim is an objective method. Annex 7 below gives some information concerning the features of a loudness rating method that should, in the opinions of L. M. Ericsson Co., the United Kingdom Administration and the Australian Administration, be given special attention.

6. It is desirable that the method ultimately chosen as the result of work described in paragraph 5 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 Notes 1 and 5 to Question 8/XII) 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 analysing the results obtained.

7. It has been proposed that, as an intermediate stage, a new method be defined for subjective tests that provides results easier to compare with objective measurements than the reference equivalents. In particular, the reference equivalent is now defined under test conditions characterized by variation of the level of the speech sounds received. A technical report by the C.C.I.T.T. Laboratory gives data on the possible effect of this situation on the values measured.

Note 1.— By way of information, Annexes 1, 2, 3 and 4 describe methods by which quantities analogous to reference equivalents may be determined by means of objective measurements. In Annexes 27 to 29 of Part II of Volume V of the *Red Book*, descriptions are given of apparatus used by the Administrations of France, the Federal Republic of Germany and Switzerland, respectively, for the objective measurement of reference equivalent. Annex H (Part II of Volume V *bis* of the *Red Book*) describes the portable telephonometer used by the Czechoslovak Administration.

Note 2.— Annex 5 describes a method developed by the Administration of the Federal Republic of Germany for measuring the image attenuation of microphone and receiver insets; it is supplemented by Annex 6 below.

ANNEX 1

(to Question 15/XII)

Contribution by the American Telephone and Telegraph Company

(See *Red Book*, Vol. V *bis*, pages 250-254)

ANNEX 2

(to Question 15/XII)

Contribution by the Italian Administration to the study of objective methods of measuring reference equivalent

(See *Red Book*, Vol. V, pages 624-637)

ANNEX 3

(to Question 15/XII)

The " loudness function " and the calculation of relative equivalents for Japanese speech sounds

(Note by the Japanese Telephone Administration)

(See *Red Book*, Vol. V, pages 637 to 662)

ANNEX 4

(to Question 15/XII)

Contribution by the French Administration

(See *Red Book*, Vol. V, pages 663-665)

ANNEX 5

(to Question 15/XII)

Image attenuations of microphone and receiver insets

(See *Red Book*, Vol. V *bis*, pages 255-265)

ANNEX 6

(to Question 15/XII)

Measurement of ratings based on loudness

(Contribution by the Administration of the Federal Republic of Germany)

In a telephone connection, the microphone and receiver are the terminal elements of a chain of electric quadripoles composed of circuits and switching equipment. The characteristic attenuations of quadripole theory are taken as a basis for planning and measuring the electric components.

The concept of reference equivalent was introduced to determine the electroacoustic transduction by the telephone set. The reference equivalent is obtained by tests in which the telephone set applied to the ear is compared with a reference system used as the normal standard system for the transmission of speech. The reference system used to be the one known as SFERT, but it was replaced some time ago by the NOSFER system, which is its equivalent as regards transmission technique. Whereas image attenuation is an abstract, objective measurement, the reference equivalent is a relative, subjective measurement, which is determined by comparison of voice and ear tests carried out by several persons on the reference system.

A method for determining the image attenuation of a transducer has already been described [1]. As in the case of the electric quadripoles, an electroacoustic transfer coefficient for transducers can also be defined on the basis of the ratio between the power provided and the power used, but the powers have different forms in the case of the transducer. One is electric, the other acoustic energy.

Let us assume that, on the electric side of the transducer, U is the voltage, J the current in amperes and $Z = \frac{U}{J}$ the electric impedance in ohms and that, on the acoustic side of the transducer, p is the sound pressure in $N/m^2 = 10$ μbars, q the sound flux in $m^3/s = 10^6 \ cm^3/s$ and $Z_a = \frac{p}{q}$ the acoustic impedance in new acoustic ohms $((\Omega^x) = 10^5$ old acoustic ohms (Ω). The image transfer coefficient $g = a + jb$, where a is the image attenuation constant and b the image phase-change constant, is then the logarithm of the ratio between the electric power UJ and the sound power pq.

For the receiver we have : $e^{2g} = \frac{UJ}{pq} = \frac{U^2}{p^2} \cdot \frac{Z_a}{Z_e}$;

and for the microphone : $e^{2g'} = \frac{p'q'}{U'J'} = \frac{p'^2}{U'^2} \cdot \frac{Z_e}{Z_a}$.

In general, only the attenuation a is of interest; it is obtained on its own when only the absolute values of U, p and Z are considered. The attenuation a in decibels is given by the equation :

$$10^{\frac{a}{10}} = \frac{U^2}{p^2} \cdot \frac{Z_a}{Z_e}$$

The attenuation a therefore depends not only on the transfer coefficient $\frac{U}{p}$, but also on the ratio between the acoustic impedance and the electric impedance. Two transducers having the same transfer coefficient but different impedances may therefore have different electroacoustic attenuations.

It will therefore not suffice, in determining an attenuation for a reference system, to fix the transfer coefficient (by the thermophone or Rayleigh method); the acoustic impedance must also be ascertained.

If, however, the acoustic impedance is known in addition to the transfer coefficient and the electric impedance, the attenuation a can be calculated from the above equation. It should be observed, though, that at the higher frequencies a difference that can be calculated [2] may occur between the input acoustic pressure and the acoustic pressure at the membrane of the microphone when the distance is no longer small compared with the wavelength. But to simplify matters, this difference has been ignored in what follows. The calculated attenuations may be checked experimentally by assembling two identical transducers acoustically in tandem, thus forming an electric quadripole of which the image attenuation can be measured. This electroacoustic attenuation is equal to half the image attenuation measured. As will be shown in a subsequent contribution, the experimental value agrees very well with the calculated attenuation.

Whereas the acoustic impedances are not known with the SFERT and NOSFER reference systems, fixed acoustic impedances are connected to the measurement microphone when the objective reference equivalent measuring set is calibrated. The microphone is fitted with the SFERT mouthpiece for sending, and with the coupler (according to Braun) for receiving. The acoustic impedance of the coupler has been measured in the C.C.I.T.T. Laboratory [3]. The acoustic impedance of the SFERT microphone mouthpiece can be calculated from its dimensions. It is therefore possible to calculate the electroacoustic attenuation for 0 reference equivalent. For reception, the following equation is valid :

$$10^{\frac{a}{10}} = \frac{U^2}{p^2} \cdot \frac{Z_a}{Z_e},$$

where $U = 0.285$ V, $p = 1.6$ N/m^2, $Z_e = 600$ ohms. We thus get

$$10^{\frac{a}{10}} = 5.23\, Z_a \cdot 10^{-5}.$$

The values are grouped in Table 1 for the various frequencies.

TABLE 1

f (Hz)	300	600	800	1000	1500	2000	2400	3000	3600
$Z_a \cdot 10^{-5}$	173	80	58,9	44.8	23	17.3	17.4	14.3	26.1
a (dB)	29.5	26.2	24.9	23.7	20.8	19.6	19.6	18.8	21.3

For 800 Hz, $a \approx 25$ dB.

The following equation applies for sending : $10^{\frac{a'}{10}} = \frac{p^2}{U^2} \cdot \frac{Z_e}{Z_a}$

where $p = 1.07$ N/m^2, $U = 0.285$ V, $Z_e = 600$ ohms. For a barometric pressure P = 760 torrs $= 1.013 \times 10^5$ N/m^2, $\kappa = 1.4$ and with a microphone mouthpiece volume $V = \pi \cdot 2.14^2 \cdot 1.4 = 20.3$ cm^3, we get Z_a from the simplified equation :

$$Z_a = \frac{\kappa P}{\omega V} = \frac{1.42 \cdot 10^{-5}}{2\pi f \cdot 20.3 \cdot 10^{-6}} = \frac{1.1 \cdot 10^6}{f \text{ in kHz}}$$

We thus find $10^{\frac{a'}{10}} = 0.776 \cdot 10^{-2} f \text{ in kHz}$.

The calculated values are grouped in Table 2.

TABLE 2

f (Hz)	300	600	800	1000	1500	2000	2400	3000	3600
a' (dB)	−26.3	−23.3	−22.1	−21.1	−19.3	−18.1	−17.3	−16.3	−15.5

For 800 Hz, $a' = -22$ dB.

The reference equivalent of modern telephone sets is negative, i.e. the electroacoustic attenuation is less than 25 or −22 dB. If the set is not assembled, the electroacoustic attenuation

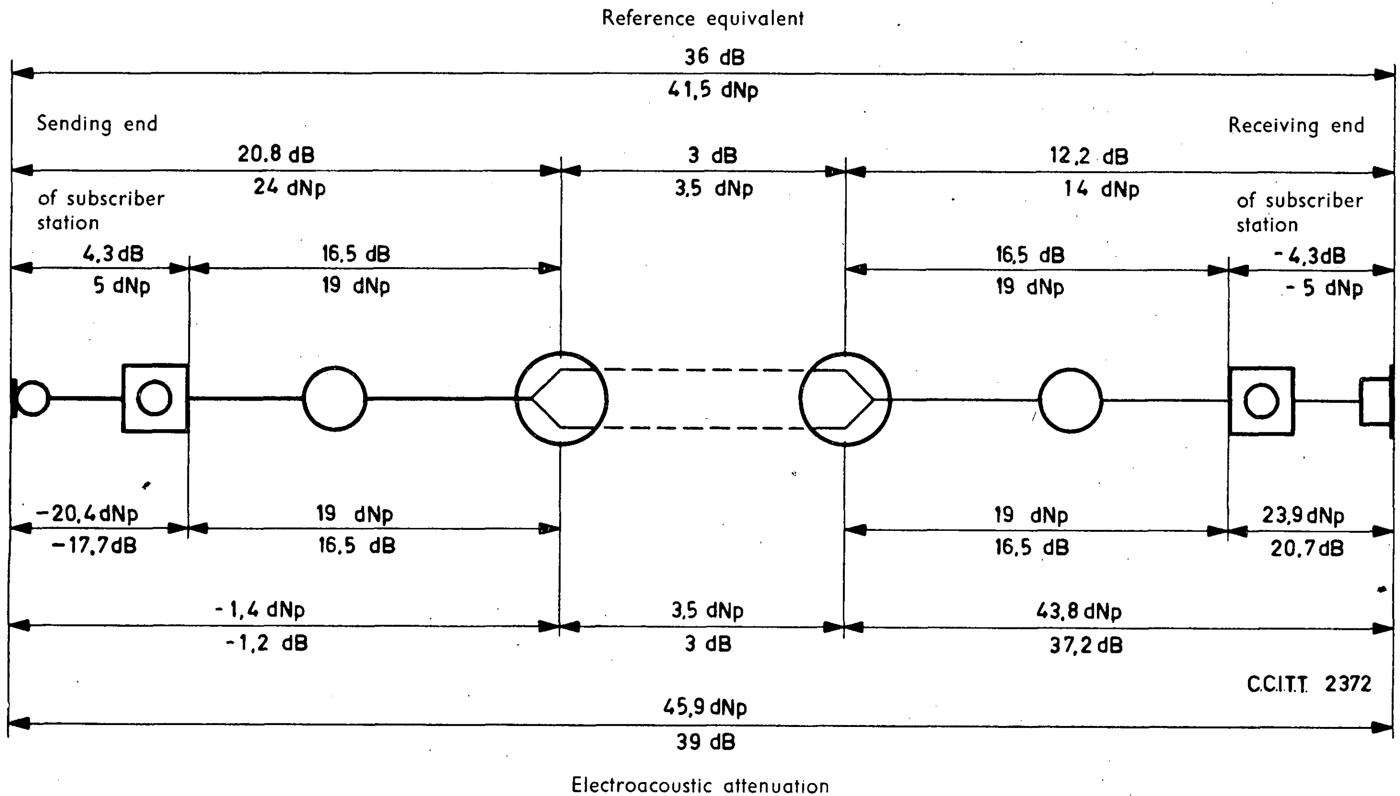


FIGURE 1. — **Distribution of reference equivalents and electroacoustic attenuations**

of a sensitive microphone can only fall below -35 dB and that of a receiver below 20 dB. The minus sign for the electroacoustic attenuation does not mean an electroacoustic attenuation but an electroacoustic gain—which reflects the physical reality but does not appear in the notion of " sending reference equivalent ". With the electroacoustic attenuation or gain, the attenuation plan can be suited to the method of designation that is usual in line technique. Since the sending reference equivalent is obtained by adding 22 dB or 25 dNp to the electroacoustic attenuation and the receiving reference equivalent by subtracting 25 dB (29.8 dNp) from the electroacoustic attenuation, the connection with the fixed reference attenuation distribution is guaranteed. Figure 1 shows the attenuation plan of an international connection set up between two telephone sets with a sending reference equivalent of $+5$ dNp and a received reference equivâlent of -5 dNp. An electroacoustic attenuation of -20.4 dNp (-17.7 dB) (a gain, in other words) corresponds to the sending reference equivalent $+5$ dB (4.3 dB), and an electroacoustic attenuation of 23.8 dNp (20.7 dB) to the received reference equivalent -5 dNp (-4.3 dB).

The electroacoustic transfer coefficient of a telephone set can therefore be determined by the concept of electroacoustic attenuation. The mean electric and acoustic terminal impedance occurring in telephone transmission should be chosen for this purpose. In line technique, it is usual—in measuring the equivalent—to take as a basis a resistance $Z_e =$ 600 ohms $\angle\, 0°$ as terminal impedance. This value could well be taken also as a basis for the electric impedance to determine the electroacoustic impedance, as is done for measuring the reference equivalent, although the impedance of subscriber lines differs from it in most cases.

When a telephone set is in practical use, the receiver is terminated by the acoustic impedance of the ear. This acoustic impedance of the ear has now been standardized by the I.E.C. [4] and for this purpose a coupler has been indicated although it simulates only the ear input impedance accurately. The principle of simplicity applied in line technique in the choice of an electric resistance of 600 ohms has been abandoned here; a simple acoustic capacitance would also suffice for the determination of the electroacoustic attenuation. This fact is shown by a comparison of the electroacoustic attenuations of the I.E.C. coupler with those which result from the different acoustic impedance values between the coupler of the objective reference equivalent measuring set. The difference in attenuation at audio frequencies is less than 1 dB. The electric input impedance necessary to determine the received electroacoustic attenuation is generally known, and can also be measured in a simple manner. Since the electric power absorbed by the transducer still depends on the internal resistance of the generator, and hence on the resistance of the subscriber line, it is advisable to feed the telephone set with a generator having an internal resistance Z_e = 600 ohms and e.m.f. U_0. The received electroacoustic attenuation is then the ratio between the maximum power that the generator can produce $\frac{U_0^2}{4Z_e}$ and the power reaching the ear with an acoustic impedance Z_0, i.e. $\frac{p^2}{Z_0}$. This ratio is therefore

$$10^{\frac{a}{10}} = \frac{U_0^2}{p^2} \cdot \frac{Z_0}{4Z_e} \cdot$$

The acoustic impedance of the microphone contributing to the sending electroacoustic attenuation depends on the frequency and is not known. Moreover, the distance between the mouth and the microphone mouthpiece varies with different handsets. An artificial mouth simulating the human mouth might well be used instead of a normal generator. To calibrate the artificial mouth, a calibrated acoustic impedance in the form of a normal microphone is necessary. Similarly the same acoustic impedance could well be used for this purpose as for

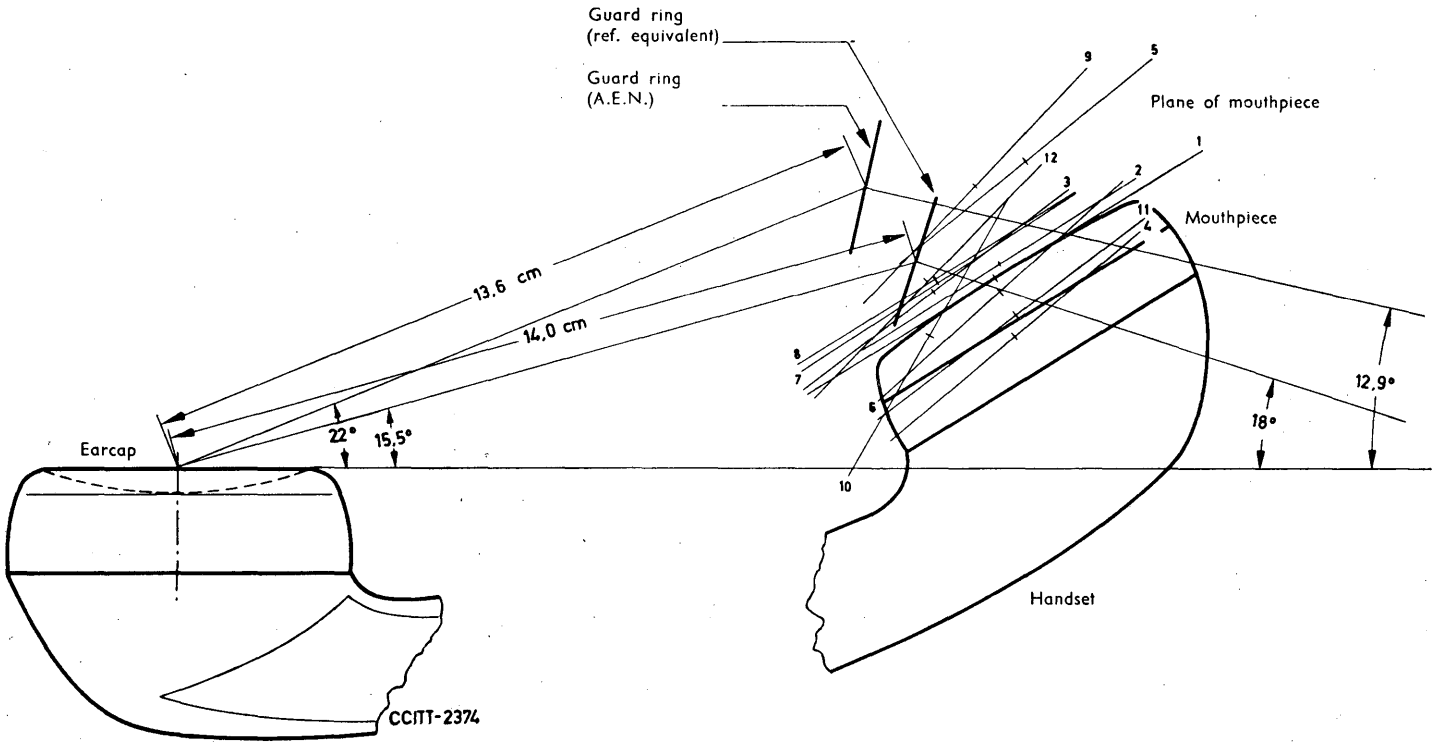


FIGURE 2. — **Position of mouthpieces**

the artificial ear, since the attenuation of the whole connection is equal to the ratio of the sound pressures. In general, the acoustic impedances of the microphone mouthpieces of handsets are much smaller.

The distance between the mouth and the normal microphone should be the mean distance imposed by the dimensions of modern handsets and based on the position of the mouth fixed for the determination of A.E.N.s. Figure 2 shows the position of microphone mouthpieces for a large number of modern handsets. The circumference of the normal microphone should correspond to the microphone mouthpiece of a handset. This condition is met by the I.E.C. coupler.

The sending electroacoustic attenuation a' may usefully be defined as being the ratio between the acoustic power $\frac{p'^2}{Z_0}$ received by the normal microphone when the artificial mouth is used and the electric power $\frac{U^2}{Z_e}$ delivered at the electric terminal resistance $Z_e = 600$ ohms; this ratio is therefore

$$10^{\frac{a'}{10}} = \frac{p'^2}{U^2} \cdot \frac{Z_e}{Z_0}.$$

The equations for a and a' give without ambiguity the electroacoustic attenuation of a telephone set at reception and transmission. The e.m.f. U_0 and the acoustic pressure p' must be so chosen as to correspond roughly with the values occurring in the telephone set, i.e.

$$\frac{U_0}{2} \approx 0.285 \text{ V and } p' \approx 1 \text{ N/m}^2.$$

Speech is a mixture of frequencies which is not perceived by the ear as an r.m.s. value but in accordance with a square-root law. To indicate the mean attenuation applicable to speech transmission, we must therefore consider the speech spectrum and the law for the addition of sound power with the standard coefficient 0.6. It is fairly easy to simulate the speech spectrum and the sound power ratio if speech is replaced by a warbling sound, the frequency of which varies more or less logarithmically at constant amplitude. It is possible in this way to make a direct recording of the frequency characteristics of a telephone set.

The relevant principles can be defined by using a normal microphone and modern receiver and varying the transmitted band by inserting various low-pass and high-pass filters. Such measurements have already been carried out [5], and have been taken into account in the O.B.D.M.

Bibliography

[1] C.C.I.T.T. *Red Book*, Vol. V *bis*, pages 255-265, Annex 5 to Question 15/XII.

[2] K. Braun: *N.T.Z.* 1964, pages 191 to 196; 230 to 236.

[3] C.C.I.T.T. Laboratory technical report No. 315, page 26.

[4] See Recommendation P.51 in Part 1 of this document.

[5] K. Braun: *T.F.T.* 1939, pages 311 to 318.

ANNEX 7

(to Question 15/XII)

Considerations in choosing a method for measuring loudness ratings

Note.— This material has been taken from information contained in the following documents (period 1964-1968) :

COM XII—No. 71, Annex 2 to the reply to point D (page 15) (Note presented by the United Kingdom Administration and L. M. Ericsson),

COM XII—No. 91, page 2 (Contribution by the Australian Administration).

In choosing a method to be standardized for making loudness rating measurements, particular attention ought to be given to the following desirable characteristics.

1. *Sensitivity/frequency characteristic of the reference system*

The reference system (or, in the case of an instrumental method, the measuring arrangement) should be such as will ensure, when measuring sending or receiving ratings, that no undue emphasis is given to particular parts of the frequency band normally transmitted by telephone channels, and that components lying outside this band should be excluded.

2. *Distance between the lips and the microphone of the reference system*

The distance between the lips of the talker (or, in the case of an instrumental method that between the artificial mouth and the point at which the reference sound pressure is established) should be such that, on the one hand, small inaccuracies in positioning are not important and, on the other, the acoustical environment has little influence.

3. *Specification of the reference system*

Having decided, in principle, upon particular characteristics for the reference system, these should be specified in sufficient detail to enable actual systems to be set up without any uncertainty. Readily available electro-acoustical elements should be used and, preferably, the requirements of the specification ought to be capable of being satisfied by using items from different manufacturers and of slightly different external form. The calibration arrangements should, as far as possible, be based on internationally standardized measuring methods and equipment (such as artificial mouths and ears). The type of earphone and any rubber pad associated with it should be chosen to reduce as much as possible the variability when used on real ears. It should also ensure that the receiver calibration procedure gives the same sensitivity on the artificial ear as that obtained on real ears.

4. *Vocal level, speech material and enunciation*

The local level should be chosen with due regard to the need for talkers to continue to maintain a constant value for long periods. The words and sounds used should be suitable for talkers of different native languages and the style of enunciation should be specified.

5. *Distance between talker's lips and handset under test*

The talking distance, when a handset is being tested, should be suitable for use with telephone handsets of modern design. In particular this distance ought to conform approximately with that normally used by telephone users when conversing normally.

6. *Method of holding the telephone handset*

The position in which the handset is held while being tested needs to be specified to ensure that an appropriate attitude is used for carbon microphones. Clamping to a fixed stand ensures precise control but holding in the hand may obviate or reduce the need for pre-conditioning procedures.

Question 16/XII — Maintenance of subscribers' sets

(continuation of Question 25/XII studied in 1964-1968)

(Question Africa H from Plan Sub-Committee for Africa)

a) How should a satisfactory service be organized for the maintenance of telephone equipment and lines, particularly as regards the qualities covered by voice-ear tests : types of apparatus to be used — possibility of making simple voice-ear tests, particularly the sensitivity of sending systems (including the microphones), measurements of line noise, etc. ?

b) Definition of measurement methods.

Note.— Contributions to the study of this Question should take the form of additions to or modifications in Recommendation P.81.

Question 17/XII — Loudspeaker telephones

(continuation of Question 17/XII, studied in 1964-1968)

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.

Note 2.— The following two points seem to merit especial attention :

1. A method to be recommended for measuring the sensitivity of a loudspeaker telephone. (A method used by the United Kingdom Administration is described in Annex 2 below.)

2. The effect of noise and echo time in the room in which the loudspeaker telephone is installed. Should limits be laid down beyond which neither the loudspeaker nor the broadcasting type microphone should be used ? Some comments on this point appear in Annex 3.

Note 3.— Recommendation P.33 answers this Question in part. Study Group XII considers it necessary to expedite studies so that a final reply may be drawn up by the end of the study period 1968-1972.

To this end, the attention of administrations is drawn to the method for measuring the sensitivity of a loudspeaker telephone set described in Annex 2 below. Administrations are invited to carry out tests with this method or with other methods giving comparable results, for example, using the NOSFER reference system instead of ARAEN. It is particularly important to determine the line-transmitter volume corresponding to a given value of the sending sound rating, measured by this method.

Note 4.— Annexes 4, 5 and 6 give additional information supplied by the Administrations of the German Federal Republic, Poland and Sweden.

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

(See *Red Book*, Vol. V, pages 703 and 704.)

ANNEX 2

(to Question 17/XII)

A method for measuring the sensitivity of a loudspeaking telephone set

(Contribution of the United Kingdom Administration)

The United Kingdom Administration has in hand studies which will determine the transmission performance characteristics needed for successful loudspeaking telephones. The factors being studied include the following :

1. Vocal level used while speaking into a microphone more distant than the normal handset. It has been found that subjects talk some few dB louder.

2. Preferred sound pressure when listening to speech reproduced by a loudspeaker. The sound pressure needed in a subject's ear, when listening without room noise, is considerably lower for loudspeaker listening (some 20 dB) than that needed when speech is reproduced from an earphone. On the other hand loudspeaker listening performance, in terms of opinion score, is much degraded by the presence of, say, 50 dB room noise. Furthermore, it does not seem possible to compensate this degradation by any increase in sound pressure; the speech would first be judged too loud.

3. Preferred shape for over-all air-to-air sensitivity-frequency characteristic is also being studied. The preferred shape is by no means a level response; some 6 to 8 dB per octave rise from 200 to 3200 Hz seems preferable but may not be optimum. It would seem desirable to incorporate this shaping in the sending part and leave the receiving part substantially level. This ensures satisfactory operation in combination with the usual shapes of frequency response of modern handset telephones.

If the preliminary conclusions mentioned in 1 and 2 above are substantiated, it is clearly undesirable to aim at sensitivities, sending and receiving, which are as high as those of handset telephones. Such sensitivities are, in any case, probably unattainable for a loudspeaking telephone.

The experimental results are being collected with the aid of what might be termed a " working reference loudspeaking telephone ". This comprises high-quality components, no voice switching but reasonably adequate over-all sensitivity by means of directional microphones and careful matching.

Sensitivities, sending and receiving, are being measured and expressed in terms of the corresponding parts of the A.R.A.E.N. (without filter). The principle is as follows :

High-quality recorded speech material is used. For sending, it is reproduced from an artificial mouth and the electrical output from a loudspeaking telephone is compared with that from the A.R.A.E.N. send end, comparison being made by means of a speech voltmeter which measures long-term r.m.s. (with pauses excluded). For receiving, the speech signal is injected into the loudspeaking telephone line terminals and the sound pressure at a defined point not too far from the loudspeaker compared with that which would be found in the ear of a listener had the same electrical input been applied to the A.R.A.E.N. receive end. Figures 1 and 2 show the conventional positions and distances that have been adopted. An A.R.A.E.N. send end can conveniently be used in conjunction with a speech voltmeter to measure the sound pressure from the loudspeaking telephone. All measurements are made with the loudspeaking telephones standing on a table in an ordinary room so as to correspond to normal conditions of use.

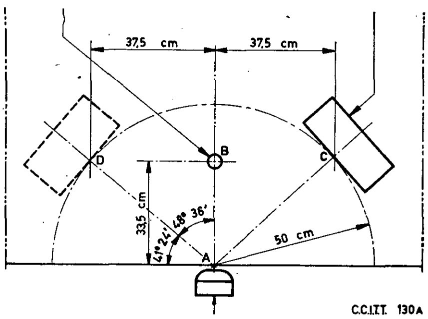


FIGURE 1. — **Sending sensitivity**

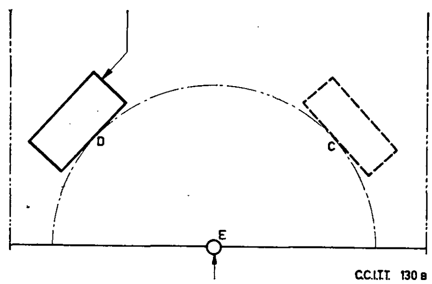


FIGURE 2. — **Receiving sensitivity**

1. All dimensions are parallel to table surface.
2. The point source of the artificial mouth at A is in alignment with the table edge and 35 cm above it.
3. The height of the acoustic screen of the reference microphone at B or E is 35 cm above the table surface.

Typical measured results are as follows :

Loudspeaking telephone	With or without voice switching	Sensitivity dB relative to A.R.A.E.N.	
		Sending	Receiving [1]
A	Without	– 30.0	– 12.2 to + 1.7
B	With	– 19.0	– 10.8 to + 6.9
C	With	– 20.0	– 6.1 to + 1.1

[1] Extreme positions of manual gain adjustment which is under the direct control of the user.

Corresponding sensitivities for handset telephones with no subscribers' lines relative to the A.R.A.E.N. send and receive ends are about –9.5 dB for sending and + 14.5 dB for receiving. (These figures were obtained in a somewhat different manner and so need to be checked; they are given as rough indications only.)

It appears from the foregoing that desirable sending and receiving sensitivities could be specified in terms of the A.R.A.E.N.

ANNEX 3

(to Question 17/XII)

Effect of room noise and reverberation time in the room in which the loudspeaking telephone set is used

I. Contribution by the United Kingdom Administration

Preferred sound pressure when listening to speech reproduced by a loudspeaker.

The sound pressure needed in a subject's ear, when listening without room noise, is considerably lower for loudspeaker listening (some 20 dB) than that needed when speech is reproduced from an earphone. On the other hand loudspeaker listening performance in terms of opinion score is much degraded by the presence of, say, 50 dB room noise. Furthermore, it does not seem possible to compensate this degradation by any increase in sound pressure; the speech would first be judged too loud.

II. Contribution by the Swedish Administration

In the light of its experience with various kinds of loudspeaker telephones, the Swedish Administration considers that especial attention should be given to the effect of room noise with loud-speakers with voice-operated switching. To reduce this effect, the set must be so designed that the directional effect throws the speaker's voice into relief. Otherwise room noise will cause pointless blockages.

III. Contribution by the American Telephone and Telegraph Company

The American Telephone and Telegraph Company considers that customers' loops or other plant used in the regular switched network should not be changed to fit loudspeaking (distant talking and listening) telephone set requirements. Therefore, the objective is that such sets should provide the same grade of transmission and meet the same impedance requirements as hand telephone sets used by the Bell System.

Early Bell System implementations could not meet this objective. These employed fixed amplification in both the transmit and receive branches to compensate for the distance losses of the microphone and loudspeaker. It was found to be unpractical to provide sufficient hybrid balance, for the wide range of impedances existing in the loop plant, to permit full compensation for these losses without creating a singing, or near singing, condition through the air path coupling the microphone and loudspeaker. Under favourable room conditions, transmission with sets of this design would be satisfactory for perhaps 10 to 15 % of Bell System customer loops.

Realization of these shortcomings led to a much improved design using the switched gain principle. (The switched gain loudspeaking telephone set and its design principles are described in bibliographical references [1] and [2]). This new design proved highly satisfactory, and has replaced practically all units of the earlier type. With the new design, amplification is switched, by voice control, from the receiving branch for incoming speech to the transmitting branch for outgoing speech. The receiving branch amplification is under control of the loudspeaking telephone user, but in no case is the total loop gain (through the transmit, receive, hybrid and air coupling paths) allowed to be sufficient to create singing or near singing conditions. Under favourable room conditions, it is estimated that the new design provides satisfactory transmission with better than 90 % of Bell System customer loops.

Any distant talking set will provide less satisfactory transmission than a close talking set if used in a reverberant room. Similarly, high ambient room noise will have a more adverse effect on loudspeaking telephone set transmission than in the case of a hand telephone set both because of the greater effect on received speech at practical listening levels. Thus, loudspeaking telephone set installations are limited as far as practical to rooms characterized by favourable ambient noise and reverberation conditions, e.g., individual offices of the executive or junior executive level and home locations. Customers are instructed to speak within a reasonable distance of the microphone, about 2 feet or less, for best performance. Also, a hand telephone set is provided with the loudspeaking unit to better ensure privacy when required and to meet the needs of other unfavourable conditions.

Bibliographical references

[1] Busala, A. : Fundamental Considerations in the Design of a Voice-Switched Speakerphone, *B.S.T.J.*, Vol. 39, p. 265.

[2] Clemency, W. F. and Goodale, W. D. Jr. : Functional Design of a Voice-Switched Speakerphone, *B.S.T.J.*, Vol. 40, p. 649.

ANNEX 4

(to Question 17/XII)

Contribution by the Administration of the Federal Republic of Germany

(See *Red Book*, Vol. V, pages 705-706.)

ANNEX 5

(to Question 17/XII)

Contribution by the Swedish Administration

(See *Red Book*, Vol. V, pages 706-708.)

ANNEX 6

(to Question 17/XII)

Contribution by the Polish Administration

(See the *Red Book*, Vol. V *bis*, pages 284-286.

Question 18/XII — Study of the implications of Spanish phonetics for telecommunication systems

(continuation of Question 28/XII studied in 1964-1968)

Study of the implications of Spanish phonetics for telecommunication systems:

a) statistical data;

b) clearness of sentences, words and sounds on telephone circuits [1];

c) relation between clearness and loudness rating.

Question 19/XII — Impedance variations in subscriber lines and telephone sets

(continuation of Question 19/XII, studied in 1964-1968)

(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 these studies the return loss should be studied from the point of view of echo as well as stability.

Question 20/XII — Synthetic speech and frequency compression systems

(continuation of Question 20/XII studied in 1964-1968)

a) What quality can at present be obtained from synthetic speech ?

Note.— Synthetic speech is defined as a group of currents similar to voice currents produced by a system of generators, themselves controlled by signals transmitted according to a suitable code.

b) What is the quality of speech transmitted over a channel whose frequency band is compressed ?

Note.— Administrations are asked to give a description of the operating principles of the system whose quality has been studied.

[1] See the study published in the C.C.I.F. *Green Book* (Paris, 1928, pages 139-171) which is drawn from an article by J. Collard entitled : " A theoretical study of the articulation and intelligibility of a telephone circuit ", *Electrical Communication*, Volume 7, p. 168, January 1929.

c) Under what circumstances could systems of the types studied under a) and b) be used in the international telephone network in the near future ?

Question 21/XII — Transmission performance of pulse-code modulation systems

(continuation of Question 27/XII, studied in 1964-1968)

a) What recommendation might be made by the C.C.I.T.T. about the standard of transmission performance assessment that ought to be achieved for a single audio-audio link provided by a practical engineered p.c.m. 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 p.c.m. system, e.g. peak clipping and centre clipping, should be assessed in the same units as used for a).

c) Since any economically engineered p.c.m. 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 live up to the requirements set forth under a) above ?

Note.— In the study of this Question the results already achieved in 1967, as indicated in Annex 1, will be taken into consideration. Annexes 2, 3 and 4 reproduce contributions communicated by Study Group XV; further information may be found in Annex 4 to Question 2/D and in its Appendices (Volume III of the *White Book*).

ANNEX 1

(to Question 21/XII)

Extracts from the report of a Working Party, approved by Study Group XII in October 1967

The Working Party 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 p.c.m. 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 evaluations of quality so that there will be a common basis for the comparison of results.

2) *The comparison method*

A reference scale is used to measure the performance of a p.c.m. system by comparing that system with a reference device. Methods of comparison that have been used by administrations include articulation tests, preference tests (paired comparison or balancing type), and analysis validated by preference tests.

While the particular comparison approach used may not have a major effect on results, the Working Party suggests the use of preference methods or analysis validated by preference methods. This does not mean that articulation tests are not suitable but is simply an attempt to have more uniformly presented results for comparison purposes.

3) *The framework*

To be meaningful, the evaluations of p.c.m. quality by comparison with the reference device must be within the framework of real system applications. The Working Party has defined four appropriate reference connections, shown in Figures 1-4.

Three of the reference connections involve four tandem p.c.m. systems; one reference connection involves eight tandem p.c.m. systems. These connections are for the specific purpose of evaluating p.c.m. effects and situations with extreme masking effects were avoided.

It should be understood that the reference connections provided are just a starting-point and should not be considered as rigid. In particular, a range of speech signals from − 2 dBm to −22 dBm (mean power of a talker when active) should be considered to occur at the input to the first p.c.m. system in the sending direction. This is, of course, not consistent with any single fixed sending reference equivalent.

4) *The standard value*

The value of the reference quantity considered acceptable under particular circumstances must be established through subjective tests. Opinion tests, threshold tests, paired comparisons with more familiar impairments, or other methods can be used.

The United Kingdom 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 p.c.m. impairments and other transmission objectives can be established.

Part b: Effects of various factors

While a great deal of information has been provided in the past on the effects of various p.c.m. factors, this has involved such a range of alternatives and rating methods that conclusive comparisons are not possible. Now Study Group XV has drawn attention to certain significant encoding/decoding characteristics which need to be studied, particularly from the standpoint of the interconnection of systems with different characteristics. Both direct inter-

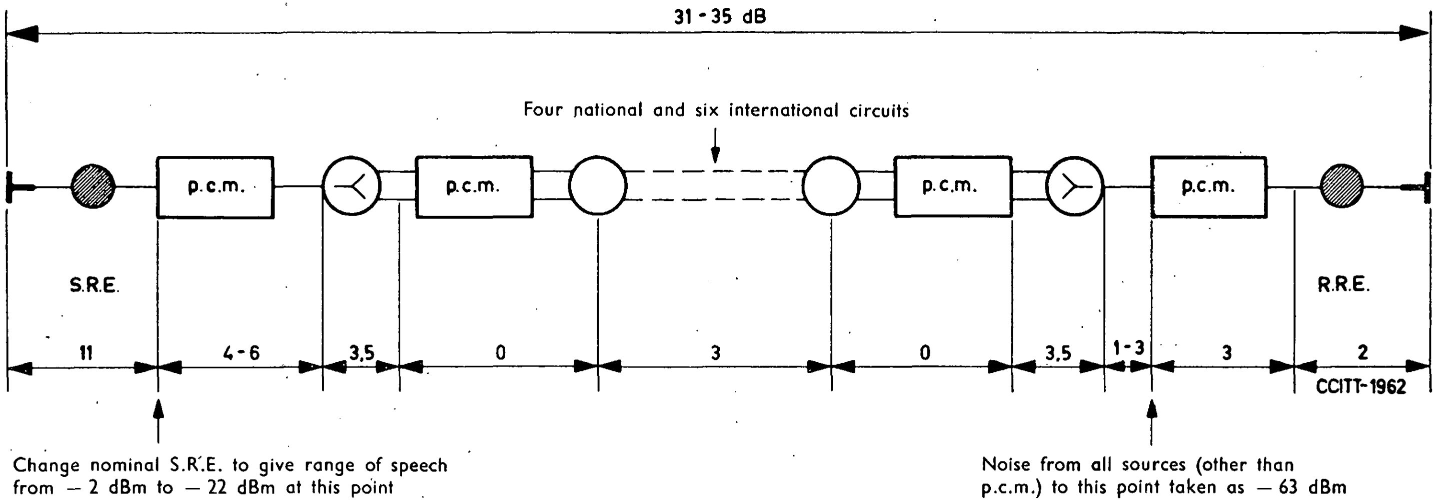


FIGURE 1. — **Long international reference connection with four p.c.m. systems**

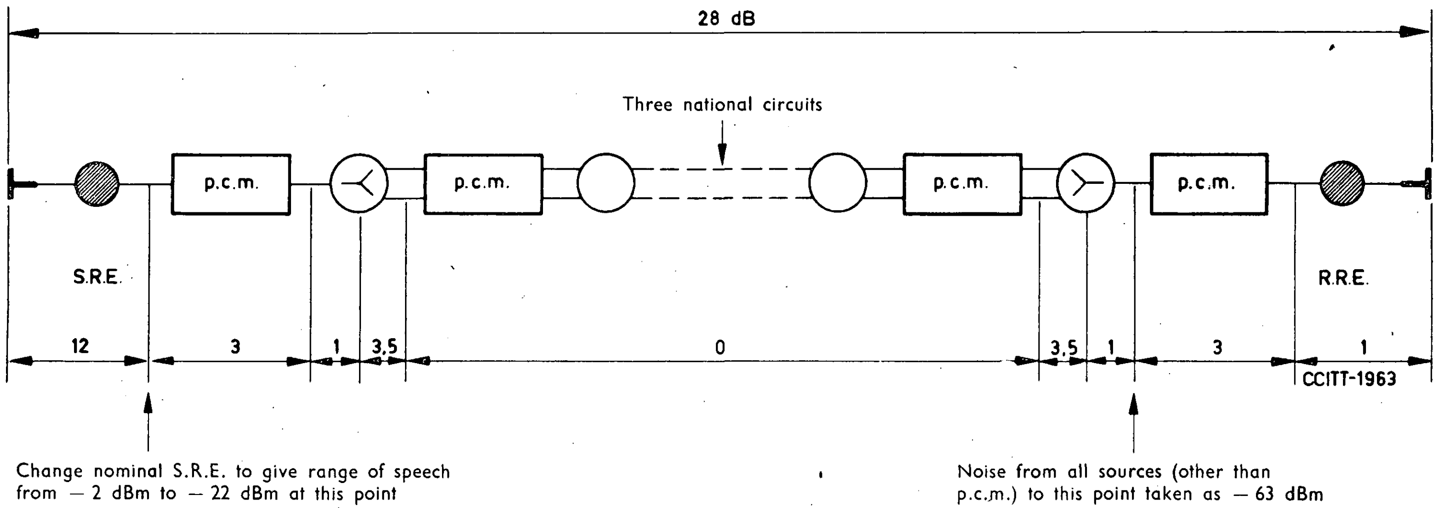


FIGURE 2. — **Long national reference connection with four p.c.m. systems**

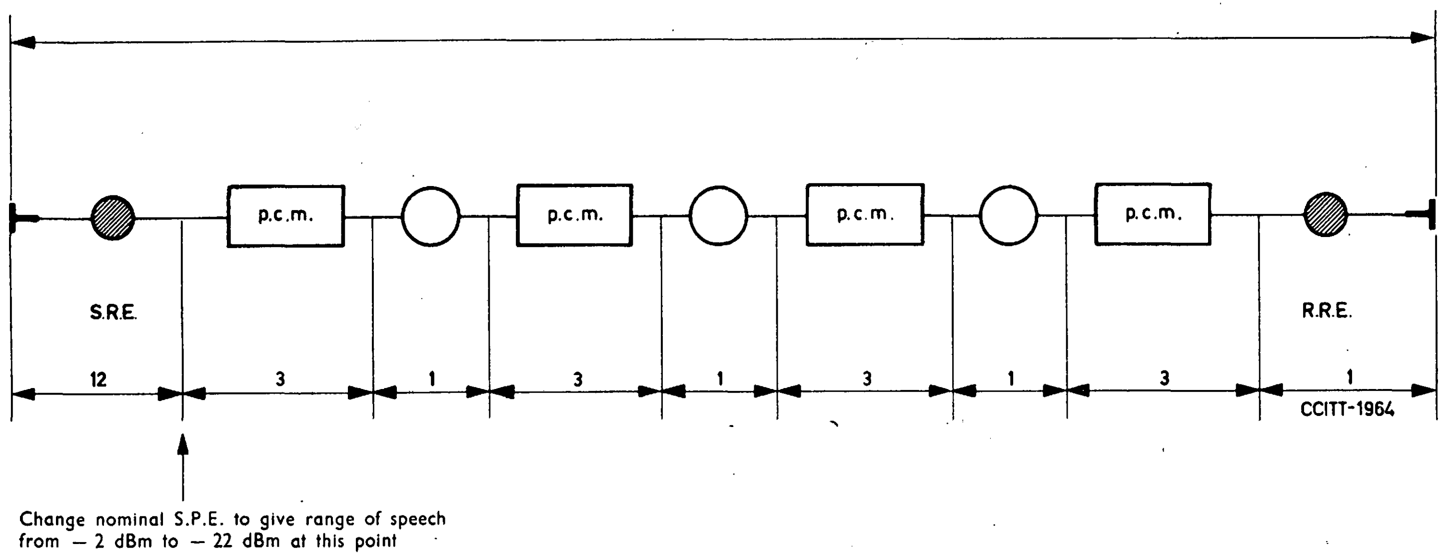


FIGURE 3. — Short reference connection in which all links are set up on p.c.m. systems

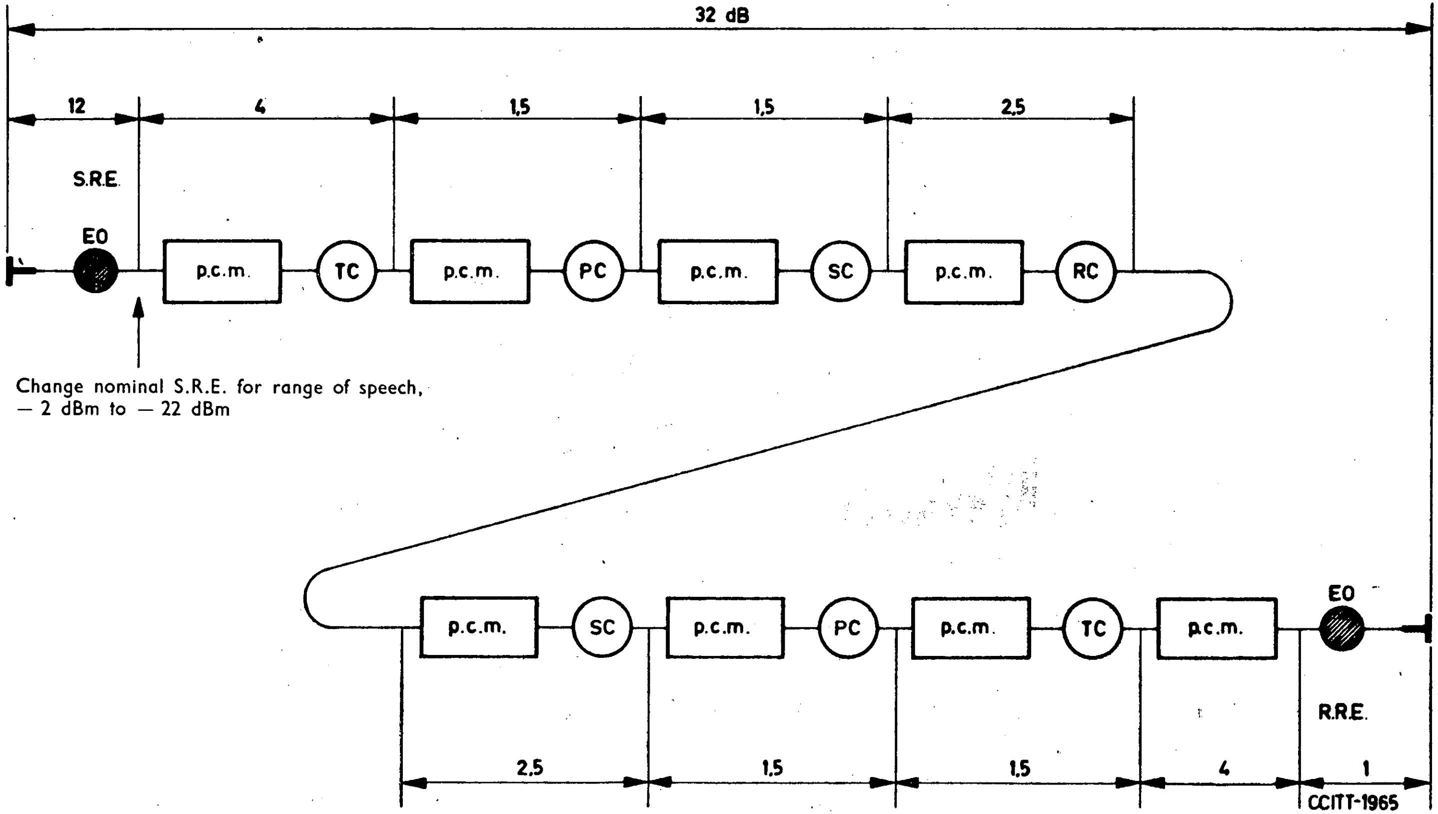


FIGURE 4. — Long reference connection in which all links are set up on p.c.m. systems

connection and interconnection with suitable numerical manipulation are considered within the scope of this request. Table 1 below summarizes the characteristics to be considered:

TABLE 1

	7 binary digits	8 binary digits
$A = 87.6$ **	$\boxed{\text{W}}$ 13 segments + 2 dBm0 *	$\boxed{\text{X}}$ 13 segments + 2 dBm0 *
$\mu = 100$ **	$\boxed{\text{Y}}$ Continuous + 3 dBm0 *	$\boxed{\text{Z}}$ 31 segments + 3 dBm0 *

* These overload values were added by the Working Party to more completely specify the characteristics.
** The definitions of encoding characteristics denoted by A and μ are as follows :

A-characteristic

$$0 \leqslant v \leqslant V/A \quad y = \frac{Ax}{1 + \log A}$$

$$V/A \leqslant v \leqslant V \quad y = \frac{1 + \log (Ax)}{1 + \log A}$$

μ-characteristic

$$y = \frac{\log (1 + \mu x)}{\log (1 + \mu)}$$

n these formulae : $x = v/V$
$y = i/B$

where v is the instantaneous input voltage,
V is the maximum input voltage for which peak limitation is absent,
i is the number of the quantization step starting from the centre of the range,
B is the number of quantization steps on each side of the centre of the range.

The out-of-band suppression characteristics of the p.c.m. filters should be part of the information provided since this can have a marked effect on the performance of p.c.m. links in tandem.

Care must be taken to ensure that systems used in these evaluations approach the performance inherent in these fundamental encoding/decoding characteristics. Factors affecting the performance of real p.c.m. systems are enumerated in Annex 1 to Question 33/XV) (Volume III of the *Blue Book*)

Note.— A Working Party of Study Group XV considers that in addition to the hypothetical reference connections represented by Figures 1, 2 and 3, the quality should also be evaluated of connections containing more than four p.c.m. systems, in accordance with Table 2. This table shows a selection of the possible combinations of the four types of p.c.m. system given in Table 1 which now require to be studied. If this proposal is accepted, Figure 4 will become superfluous. In any case, the results obtained should be compared with the results the Study Group already has for connections of similar composition but comprising only circuits set up on frequency-division multiplex systems.

Part c: Working measurement methods

Two types of measurements that might be applied to working p.c.m. 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 Admi-

TABLE 2

	Similar compression laws	Mixtures of compression laws
Figure 1	2Y + 4Z + F + F + 4Z + 2Y 2W + 4X + F + F + 4X + 2W 4Y + 2Z + F + F + 2Z + 4Y 4W + 2X + F + F + 2X + 4W 14W, 14X, 14Y, 14Z	4W + 2X + 2F + 2Z + 4Y 4W + 3X + 3Z + 4Y 7X + 7Z
Figure 2	7W, 7X, 7Y, 7Z 2W + 3X + 2W 2Y + 3Z + 2Y	None
Figure 3	4W, 4X, 4Y, 4Z W + 2X + W Y + 2Z + Y	None

F = f.d.m.

nistration 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 p.c.m. system is excited with a signal near 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 p.c.m. 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.

ANNEX 2

(to Question 21/XII)

Status of noise reference unit instrumentation

(Contribution by the American Telephone and Telegraph Company)

The following Appendix (extracted from document COM XII—No. 84, of 24 August 1967) contains the description of a noise reference unit which produces signal-dependent noise for use in subjective tests of compandored p.c.m. systems. A similar unit has been assembled and is being tested. A description of the unit and preliminary plans for its use are presented in this contribution.

A block diagram of the unit is shown in Figure 1. Operation is as described in the Appendix. 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.

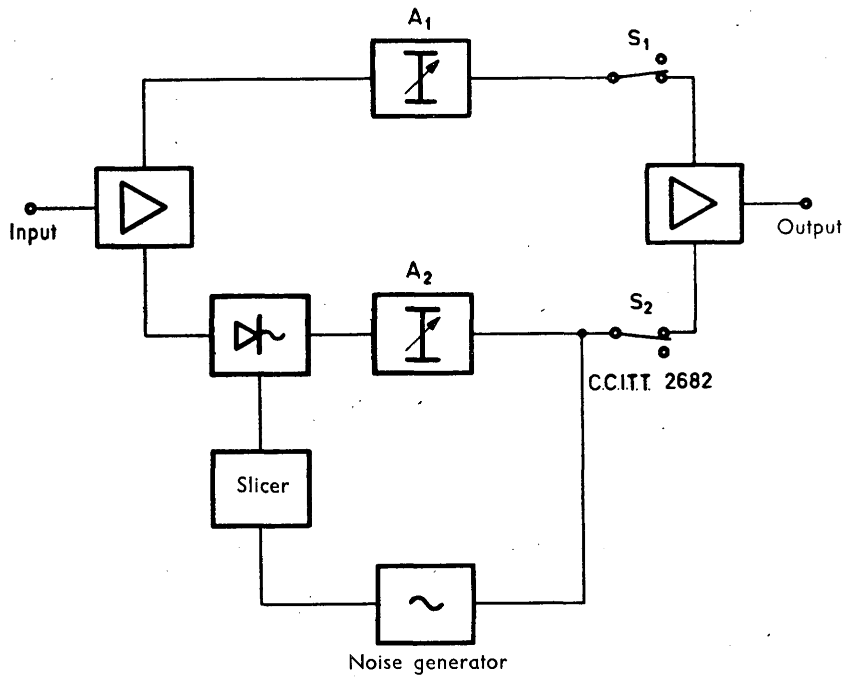


FIGURE 1. — **Noise reference unit**

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.

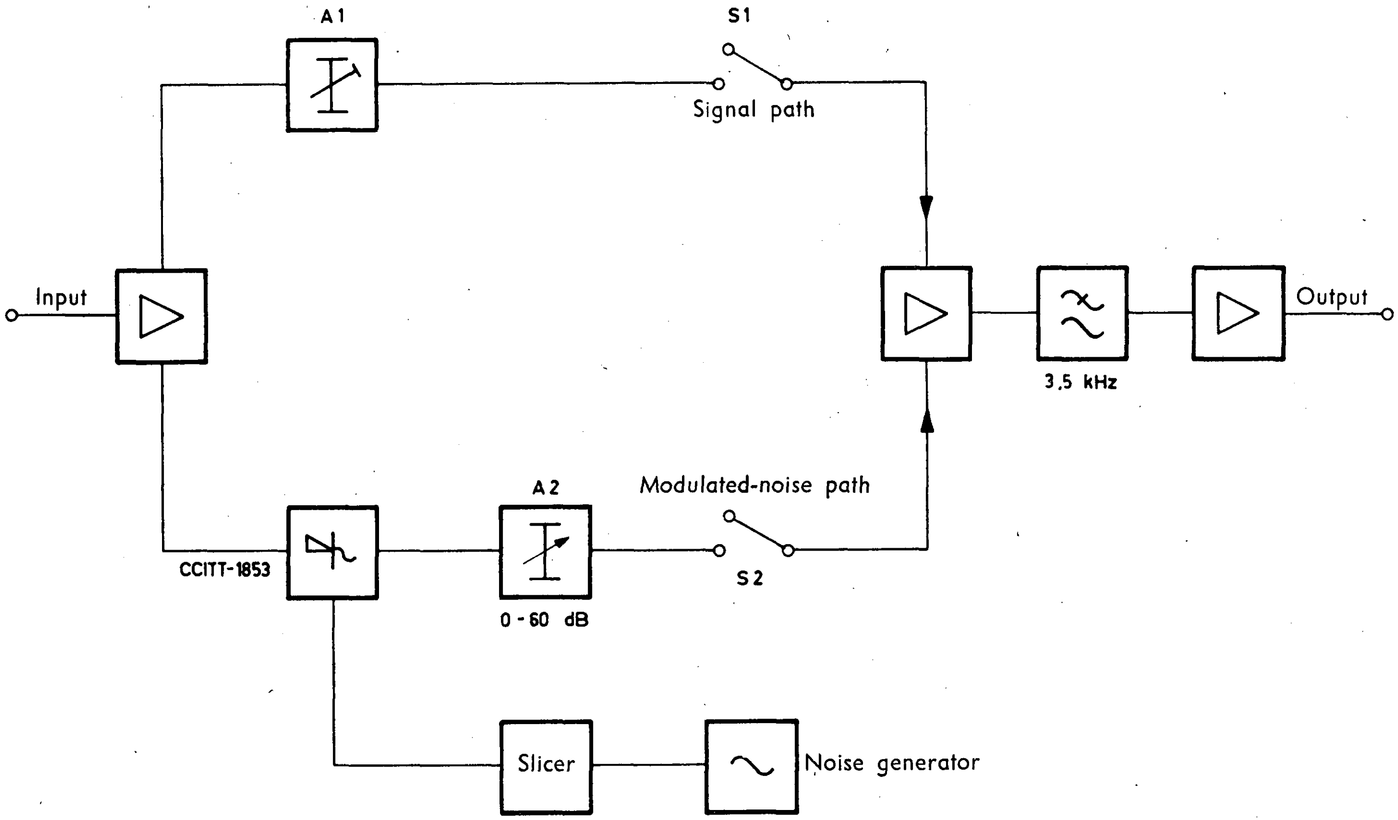


FIGURE 2. — **Modulated-noise reference unit**

APPENDIX to Annex 2

Description of the modulated-noise reference unit used in the United Kingdom

The Modulated-Noise Reference Unit described is a device that produces distortion which is similar in character to that of a p.c.m. 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 20″ × 12″ × 10″, weighing approximately 30 pounds. A block schematic diagram of the unit is shown in Figure 2 on the preceding page.

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. 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 p.c.m. system.

ANNEX 3

(to Question 21/XII)

Analysis on the performance of several cascaded p.c.m. systems

(Contribution by Telettra Sp.A.)

Introduction

This contribution deals with the problem of the objective noise produced at the output of a chain of several p.c.m. systems by the quantization of the individual systems and by the possible insertion of thermal noise along the chain.

This problem, which is quite complex without any simplifying assumption, can be much more easily treated if it is assumed that, in any section of the chain, the noise statistics are gaussian and the noise and signal are statistically independent processes.

Using this hypothesis the problem of the whole chain performance can be reduced to the determination of the noise at the output of a single system, with a signal at its input affected by gaussian noise, this noise being statistically independent of the signal itself.

The purpose of this contribution is, after having shown and discussed numerical results obtained for a single system for different quantizing laws (analysing at the same time the validity of the simplifying hypotheses made), to give some examples of calculation of noise at the output of some p.c.m. systems chains.

Gain and signal-to-noise ratio of a single p.c.m. system with noisy input signal

Computations were made of the attenuation Γ and signal-to-noise ratio $\left(\frac{S}{N}\right)_0$ at the output of a p.c.m. system, assuming as input signal the sum of a random process $i(t)$ having an exponential probability density function[1], and gaussian noise $n_i(t)$ statistically independent of the signal.

The following definitions have been adopted[2] :

$$\Gamma = \frac{E(i^2)}{E(i \cdot u)} = \frac{1}{k_0} \qquad \left(\frac{S}{N}\right)_0 = \frac{1}{\dfrac{E(u^2)}{k_0^2\, E(i^2)} - 1}$$

where : $E(x)$ is the expectation of the random variable x and $u(t)$ is the output of the system.

Figure 1 explains the meaning of the symbols used.

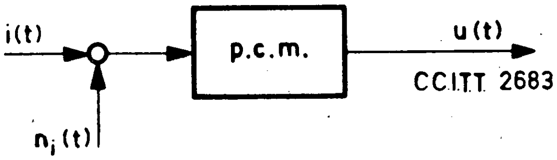


FIGURE 1

The probability density functions of i and n_i being:

$$p_i(x) = \frac{1}{\sqrt{2}\, I} \exp\left(-\frac{\sqrt{2}}{I}\,|x|\right)$$

$$pn_i(x) = \frac{1}{R\sqrt{2\pi}} \exp\left(-\frac{x^2}{2R^2}\right)$$

The resulting expressions for $E(i \cdot u)$ and (Eu^2) are:

$$E(i \cdot u) = -R\sqrt{\frac{2}{\pi}} \sum_{-N^k}^{+N} u_k(\Phi_{k+1} - \Phi_k); \qquad E(u^2) = \sum_{-N^k}^{+N} \frac{u_k^2}{2}(\Psi_{k+1} - \Psi_k)$$

where :

$$\Phi_k = \exp\left(\frac{-i_k^2}{2R^2}\right) + \frac{\sqrt{\pi}}{4}\exp\left(\frac{R^2}{I^2}\right)\left\{\left(\frac{I}{R} - 2\frac{R}{I}\sqrt{2}\,\frac{i_k}{R}\right)\exp\left(\sqrt{2}\,\frac{i_k}{I}\right)\cdot\right.$$

$$\left.\cdot\left[1 - \operatorname{erf}\left(\frac{i_k}{\sqrt{2}R} + \frac{R}{I}\right)\right] + \left(\frac{I}{R} - 2\frac{R}{I} + \sqrt{2}\,\frac{i_k}{R}\right)\exp\left(-\sqrt{2}\,\frac{i_k}{I}\right)\left[1 + \operatorname{erf}\left(\frac{i_k}{\sqrt{2}R} - \frac{R}{I}\right)\right]\right\},$$

$$\Psi_k = \exp\left(\frac{R^2}{I^2} + \frac{i_k}{\sqrt{2}I}\right)\left[1 + \operatorname{erf}\left(-\frac{i_k}{\sqrt{2}R} - \frac{R}{I}\right)\right] - 2\operatorname{erf}\left(-\frac{i_k}{\sqrt{2}R}\right) +$$

$$+ \exp\left(\frac{R^2}{I^2} - \frac{i_k}{\sqrt{2}I}\right)\left[1 \operatorname{erf}\left(-\frac{i_k}{\sqrt{2}R} + \frac{R}{I}\right)\right]$$

and where i_k and u_k are the decision and reconstruction levels, and 2 N-1 is the number of steps.

[1] It is well known that the speech signal can be regarded as a random process with such a probability density function.

[2] A. BELLMAN & F. VAGLIANI (Telettra Sp.A.) : " Quantizzazione ottimale nei sistemi p.c.m."; *Alta Frequenza*, No. 10, Vol. XXXVI, 1967, pages 942-951.

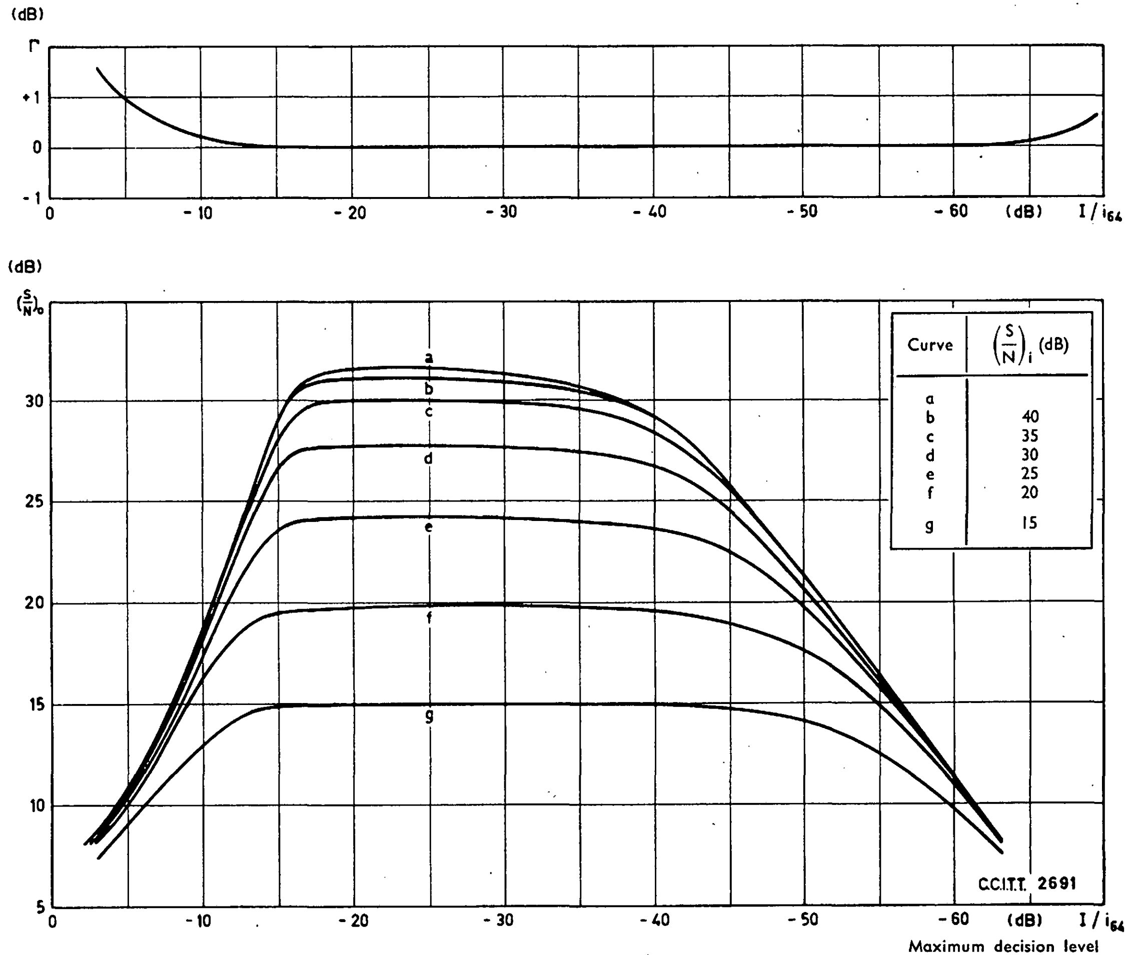


FIGURE 2. — **Attenuation Γ and output signal-to-noise ratio $\left(\frac{S}{N}\right)$ versus input signal level for several input signal-to-noise ratios $\left(\frac{S}{N}\right)_i$** (13 segments quantizing law $A = 87.6$; 127 steps)

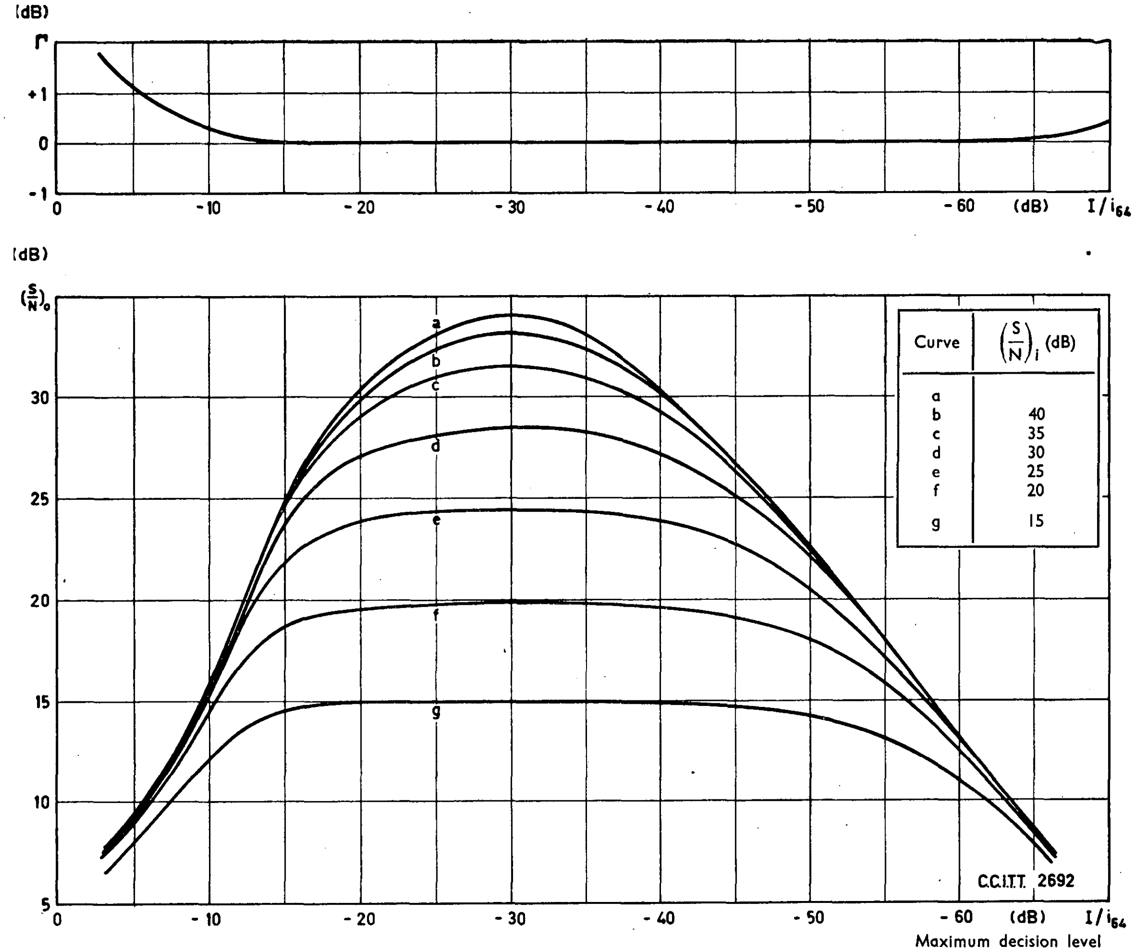


FIGURE 3. — **Attenuation Γ and output signal-to-noise ratio $\left(\frac{S}{N}\right)_0$ versus input signal for several input signal-to-noise ratios** $\left(\frac{S}{N}\right)_i$ (Hyperbolic quantizing law *h* 20)

The results of the numerical computations are shown in Figures 2 and 3 as graphs of Γ and $\left(\frac{S}{N}\right)_0$ at the output of each system versus the input signal level for several values of the input signal-to-noise ratio $\left(\frac{S}{N}\right)_i = \frac{I^2}{R^2}$.

Figure 2 refers to a logarithmic quantizing law approximated by 13 segments ($A = 87.6$) with 127 quantizing steps; Figure 3 refers to a hyperbolic quantizing law ($h = 20$) with 127 steps.

By performing the relevant calculations it can be seen that in each case and under all conditions considered, the resulting noise power is, to a very good approximation, equal to the sum of the quantizing noise power that would be obtained without input noise, and the input noise power itself. Therefore the transformation shown in Figure 4 can be regarded as valid for a noise power evaluation at least for the levels of signal and noise considered. In this figure $n_q(t)$ is the quantization noise caused by the signal $i(t)$ alone.

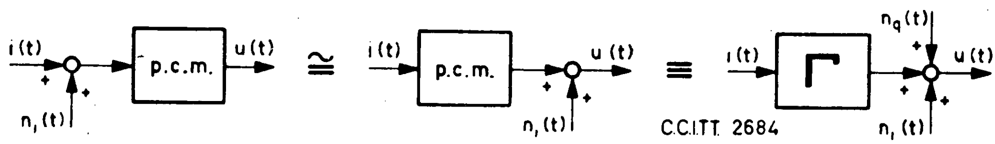


FIGURE 4

This is an important result in practice because it simplifies very much the evaluation of noise in a p.c.m. system with a noisy input signal and therefore also the evaluation of noise for a whole chain. It should be noted also that the gain is practically unaffected by the input noise.

Validity of the hypotheses

Such simple results could raise some doubt about the assumptions made.

Let us consider therefore the validity of these assumptions. The hypothesis of gaussian noise along the chain is equivalent to admitting that in a single p.c.m. system, at whose input signal $i(t)$ and noise $n_i(t)$ are present in any ratio, and for any level of the signal, the output noise $n_0(t)$ probability density function is gaussian.

This is true to a very good approximation even in the absence of noise $n_i(t)$ at the input provided that :

a) The input signal level is not so high as to drive the system into saturation (which would produce noise far from gaussian).

b) The input signal level is not so low as to operate the system principally in the linear zone of the compression-expansion characteristic (the noise probability density would then tend to a rectangular distribution).

The presence of noise at the input of the system, of course, further supports the hypothesis of gaussian noise at the output, particularly when the input noise is almost gaussian.

Analysing a long chain of p.c.m. systems the assumption of gaussian noise along all the sections can therefore be valid in all cases except when the first systems of the chain go into heavy saturation.

Ignoring this possibility, the first systems work in a zone of the compression-expansion characteristic which results in noise with an intrinsically almost gaussian distribution and the last systems, which work at very low levels, receive signals already affected by almost gaussian noise and therefore their output noise is even closer to gaussian distribution.

The limitation of the validity of this hypothesis to the case of systems which do not saturate is not very important for it is well known that in heavy saturation the objective noise bears little relation to intelligibility and consequently an exact computation of noise in these conditions could not give meaningful results.

The second hypothesis of statistic independence between signal and noise along the chain, although untrue in general, does not significantly affect the accuracy of the calculations. The following gives an outline of the justification of this statement.

The calculations were performed assuming statistical independence, thus implying that the probability density of the noise n_i at a given moment is independent of the value of the input signal i at the same moment.

This is generally not true if noise and signal come from another p.c.m. system. A much more accurate (and complex) computation would relate the noise variance at any moment to the value of the input signal at the same moment, according to a relationship depending upon the level of the signal and upon the characteristics of the chain at the input of the system.

This accurate computation would not cause an appreciable improvement of the final results. The reason for this can be found in the fact that the input noise power n_i can be directly added to that of the quantization noise n_q to obtain the output noise power.

Noise in a chain of systems

Let us consider now as an example the chain shown in Figure 5a, formed by four p.c.m. systems, each having a 3-dB loss, and a central conventional transmission system which can be characterized by a 12-dB attenuation and a noise insertion at the centre.

To calculate the total output noise, this chain can be replaced by the equivalent chain shown in Figure 5b.

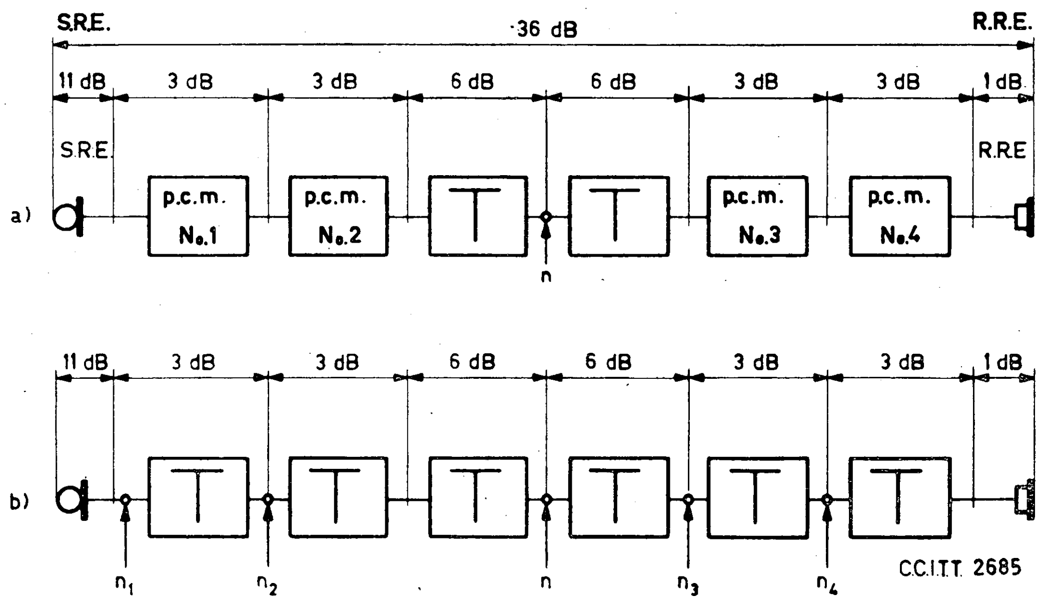


FIGURE 5

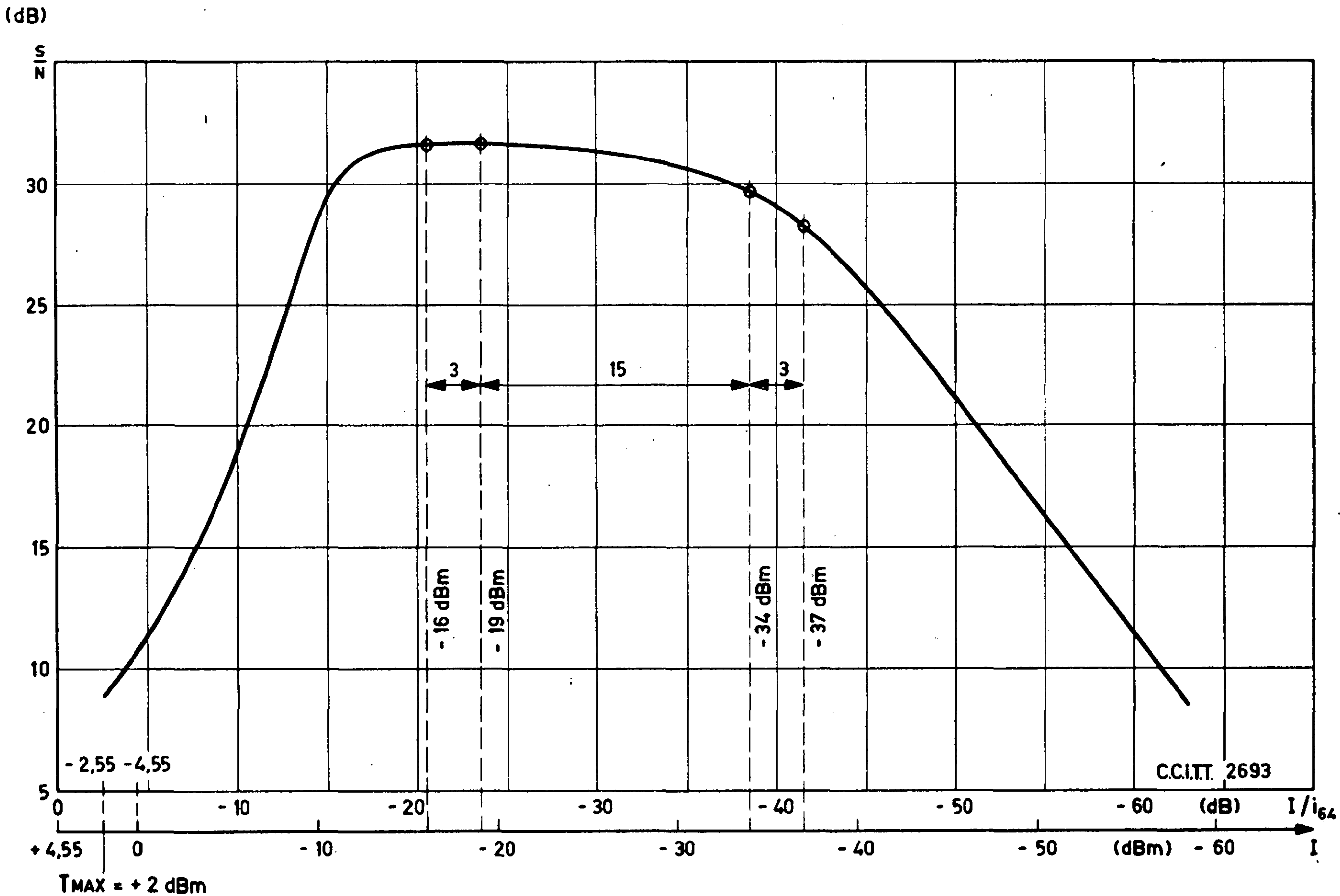


FIGURE 6. — **Evaluation of the noise contributions along the chain** (13 segments law $A = 87.6$)

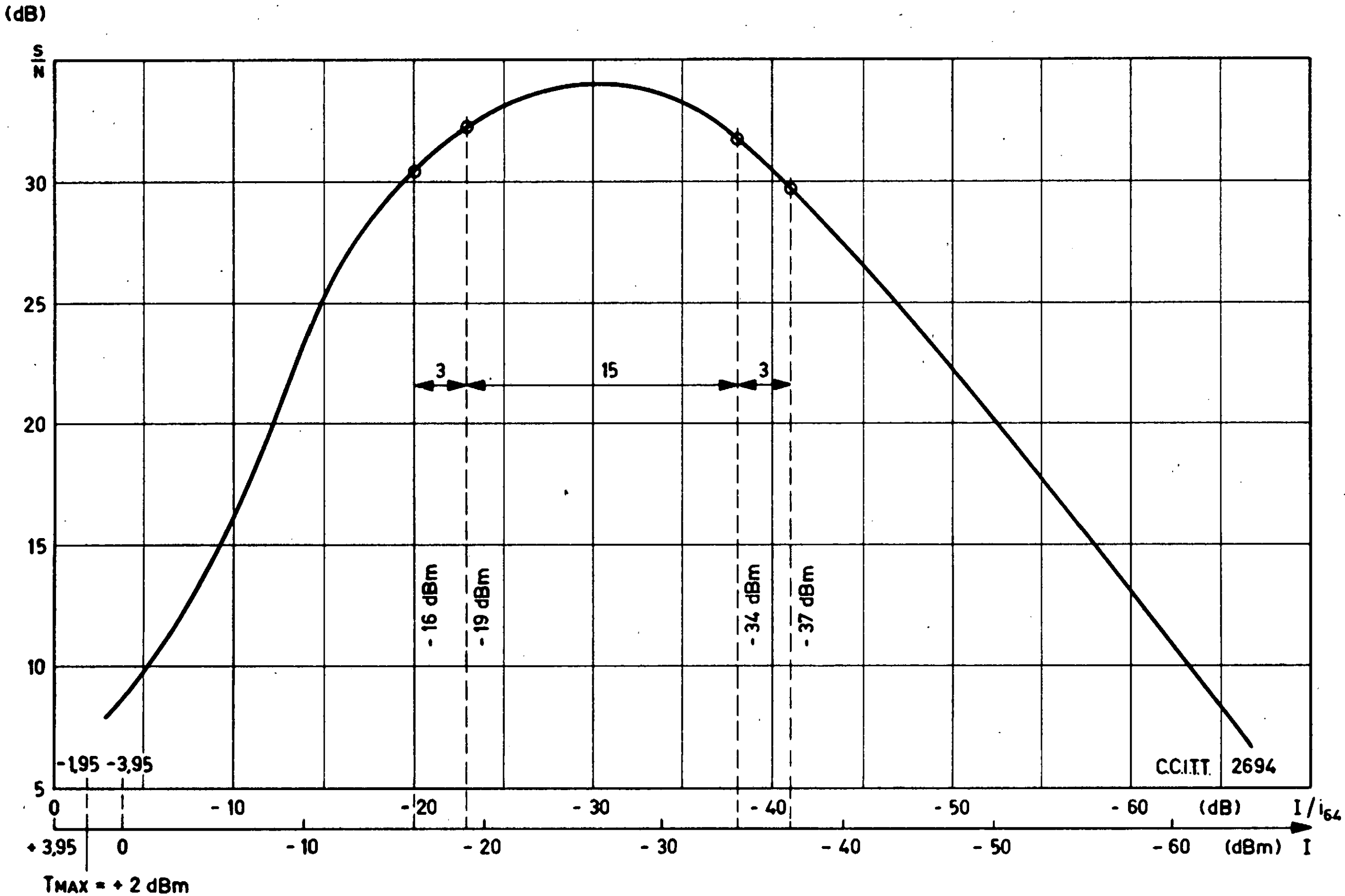


FIGURE 7. — **Evaluation of the noise contributions along the chain** (Hyperbolic law $h = 20$)

where the noise powers n_1, n_2, n_3, n_4, from the discussion above, have to be evaluated as functions of the level of the signal at the input of the four p.c.m. systems respectively as if the signal were noise-free.

Those powers can then be summed together and also summed with the noise power of the conventional system, taking into account the relative levels of the sections when they are inserted.

With 11 dB S.R.E. and 30 dB total R.E. the input level in the first system can be assumed to be −16 dBm, this level does not cause appreciable saturation.

In Figure 6 the graph of $\left(\frac{S}{N}\right)_0$ versus signal level is shown for a system with a 127-quantizing-steps logarithmic law approximated by 13 segments (A = 87.6), and the working points of the four systems are indicated.

From this graph the absolute powers given by the individual systems and then the total power are obtained.

The same calculation is repeated for a hyperbolic quantizing law (h = 20, 127 quantizing steps). (See Fig. 7.)

The noise values (partial and total) in both cases are listed in the table below, where n is assumed = − 60 dBm0.

		System 1	System 2	System 3	System 4	Conv. system	Total	Unity
	Abs. i/p signal level	− 16	− 19	− 34	− 37			dBm
13 segments	S/N	31.5	31.6	29.7	28.3			dB
13 segments	Noise level	n_1 − 50.5 0.9 × 10	n_2 − 50.6 0.9 × 10	n_3 − 48.7 1.3 × 10	n_4 − 47.3 1.9 × 10	n − 60 0.1 × 10	− 42.8 5.1 × 10	dBm0 pW0
Hyperbolic	S/N	30.3	32.2	30.7	29.7			dB
Hyperbolic	Noise level	n_1 − 49.3 1.1 × 10	n_2 − 51.2 75 × 10	n_3 − 49.7 1 × 10	n_4 − 48.7 1.3 × 10	n − 60 0.1 × 10	− 43.7 4.2 × 10	dBm0 pW0

ANNEX 4

(to Question 21/XII)

Compatibility of 7- and 8-bit p.c.m. systems: Choice of the encoding law and the code

1. *Introduction*

So far no agreement has been reached about the number of bits in which a speech sample should be coded. In all probability both 7- and 8-bit systems will be realized.

For the compatibility between the systems it is important that in any case the same encoding law is chosen. This also holds for the compatibility between 7- and 8-bit systems. However, if 7- and 8-bit systems are interworking, another small extra distortion will arise causing the quality of such a connection to be somewhat worse than a pure 7-bit one. In the following this compatibility problem will be further discussed. Finally, something is said about the coding to be applied. A proposal of the Netherlands Administration is presented concerning the code and the encoding law to be used.

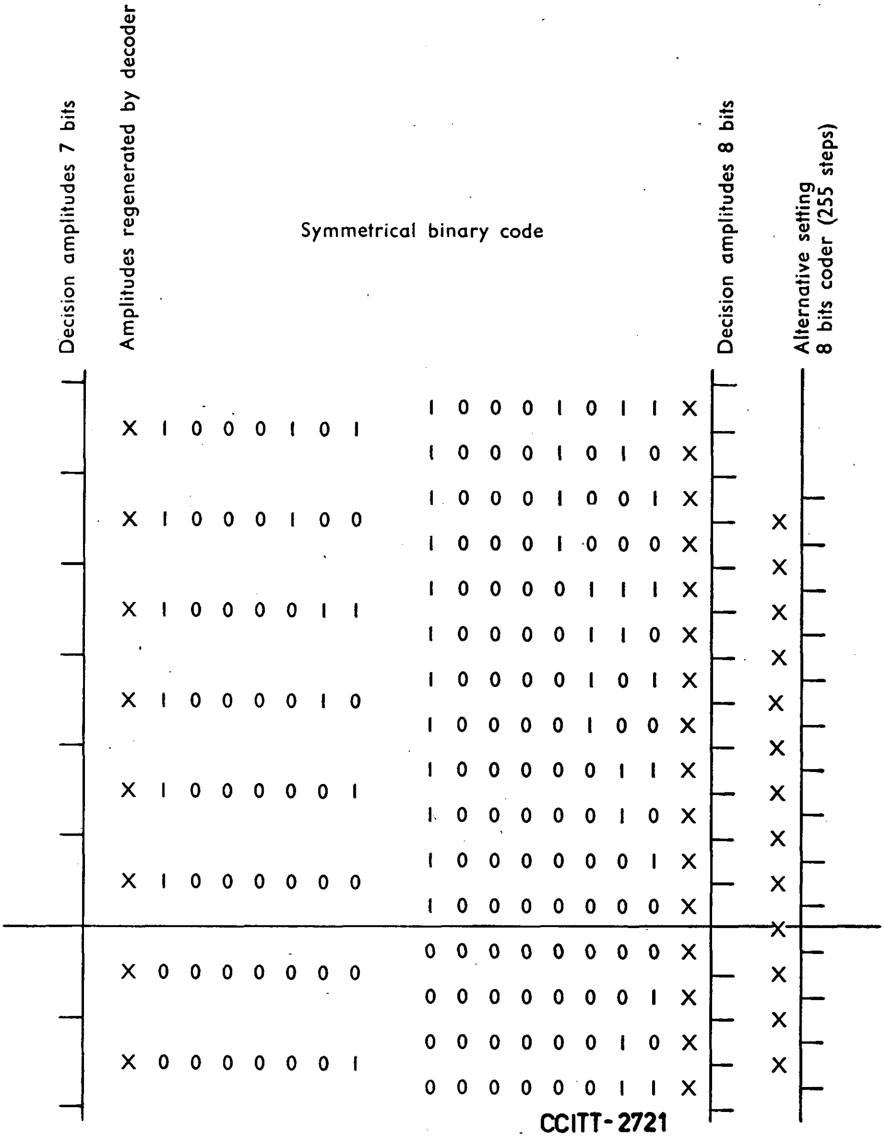
2. *Compatibility of 7- and 8-bit coders*

2.1 *Linear coding*

The compatibility problem between 7- and 8-bit equipment can be illustrated by means of a linear coder. Figure 1 shows that the levels regenerated by a 7-bit decoder are not available in an 8-bit one. If a 7-bit coder and an 8-bit decoder are connected, the amplitude regenerated will be a little too low if a zero is added to the 7-bit code, this level being somewhat too high in case a one is added. The difference is a quarter of the step of the 7-bit coder.

In the reverse direction there is no such problem in the example given, for if an 8-bit coder is connected with a 7-bit decoder, this corresponds with leaving out the 8th bit. The amplitude range from which a code can be originated will then cover the corresponding area of the 7-bit equipment. In a linear coder with symmetrical binary coding the effect described would therefore result in somewhat extra positive or negative centre clipping (dependent on the interpretation as " 0 " or " 1 " of the missing bit) when a 7-bit coder is coupled to an 8-bit decoder. The coupling of an 8-bit coder to a 7-bit decoder by no means leads to a worse result than would have been obtained with 7-bit systems.

The centre clipping would not occur if no symmetrical binary coding was applied, but a continuous normal binary coding. In that case the effect would only be a slight shift of the signal (see Figure 2). The same result could be obtained with the symmetrical binary coding, if the 8th bit would be coupled to the 1st bit (polarity bit). However, in the case of non-linear coding this is not the most favourable solution, as will appear later on. Another possibility that could be considered to obtain improvement is to shift the decision amplitudes of the 8-bit coding half of the step of the 8-bit coder. This is given as an alternative in Figure 1. Then an exact representation of a 7-bit code can be obtained in an 8-bit decoder, but here we are confronted with the same problem in the reverse direction. If an 8-bit coder is now connected with a 7-bit decoder, the amplitude-range from which a code can be originated does no longer cover the corresponding range of the 7-bit equipment. This also leads to extra distortion.



Note. — The symmetrical binary code is indicated above. The transmitted code is this code with alternate inversion of the bits.

FIGURE 1

2.2 *Non-linear coding*

In case of non-linear coding the shift, as explained in the aforegoing paragraph, leads to an amplitude-dependent deviation and so, in principle, to distortion. Let us assume $y = f(x)$ is the compression law normed to 1 ($y = 0$ in case $x = 0$, $y = 1$ in case $x = 1$), in which y represents the compressed signal. The shift of $\frac{1}{4}$ of a step of the 7-bit coder on the y-axis will have a constant value $\Delta y = \pm \frac{1}{4} \times \frac{1}{64} = \pm \frac{1}{256}$, independent of the value of the signal, as the y-axis is divided linearly in the desired number of steps.

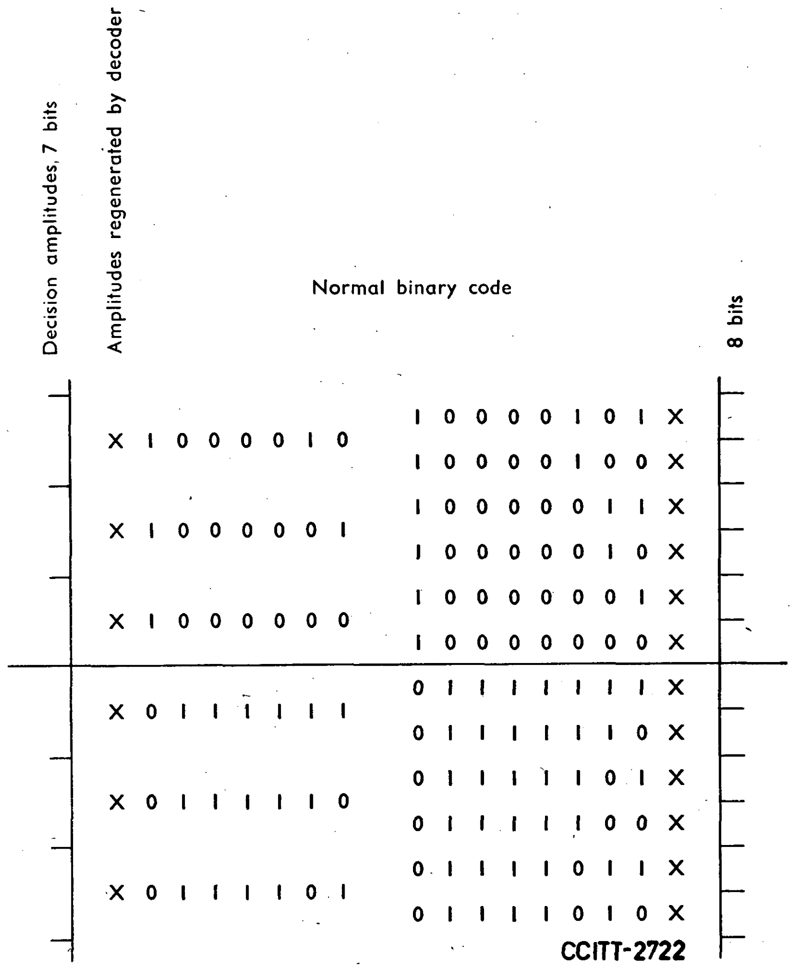


FIGURE 2

Then the following value is found for Δx :

$$\Delta x = \frac{\mathrm{d}x}{\mathrm{d}y} \Delta y = \frac{\Delta y}{\mathrm{d}y/\mathrm{d}x}$$

If for $y = f(x)$ the logarithmic A-characteristic is taken :

$$y = \frac{1 + \ln Ax}{1 + \ln A} \quad \text{with } A = 87.6, \text{ we obtain :}$$

$$\frac{\mathrm{d}y}{\mathrm{d}x} = \frac{1}{(1 + \ln A)\, x}, \text{ so that :}$$

$$\Delta x = (1 + \ln A) \cdot \Delta y \cdot x$$

$$\Delta x = \pm \frac{5.47}{256}\, x = \pm\, 0.0213\, x$$

So in the case of a logarithmic characteristic the extra deviation Δx is proportional to x, which can be interpreted as an amplification (if Δy is positive) or an attenuation (if Δy is negative), of 0.183 dB (0.0211 Np). This interpretation implies that in case of a symmetrical binary coding with a logarithmic encoding law the compatibility-deviation from 7- to 8-bit equipment can for the greater part be interpreted as a slight attenuation or amplification. It is only for signal values for which $x < \frac{1}{A}$, where the characteristic passes over into a linear one, that this interpretation is not applicable. Then the compatibility-error results in a slight positive or negative centre clipping (see Figure 3). If, as suggested for the linear coder, we wish to avoid this centre clipping by coupling the sense of the shift Δy to the polarity-bit, it can easily be seen from the

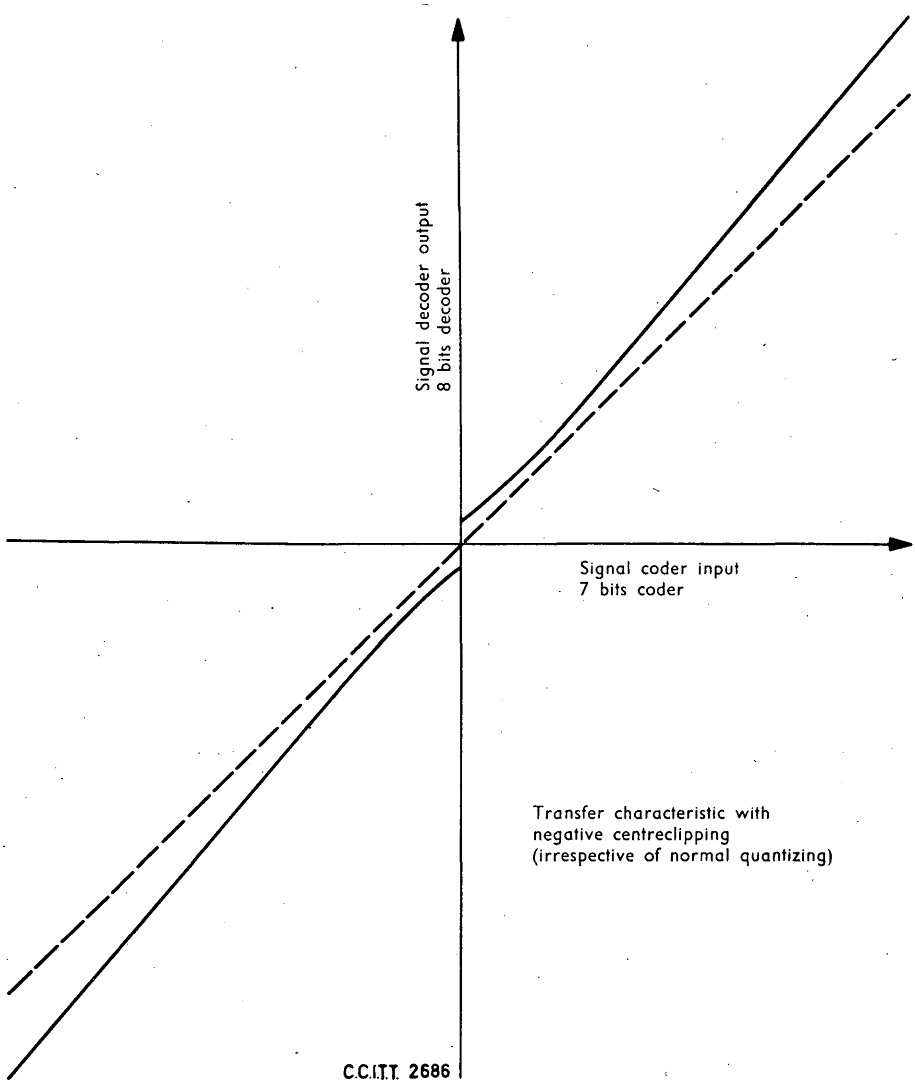


FIGURE 3

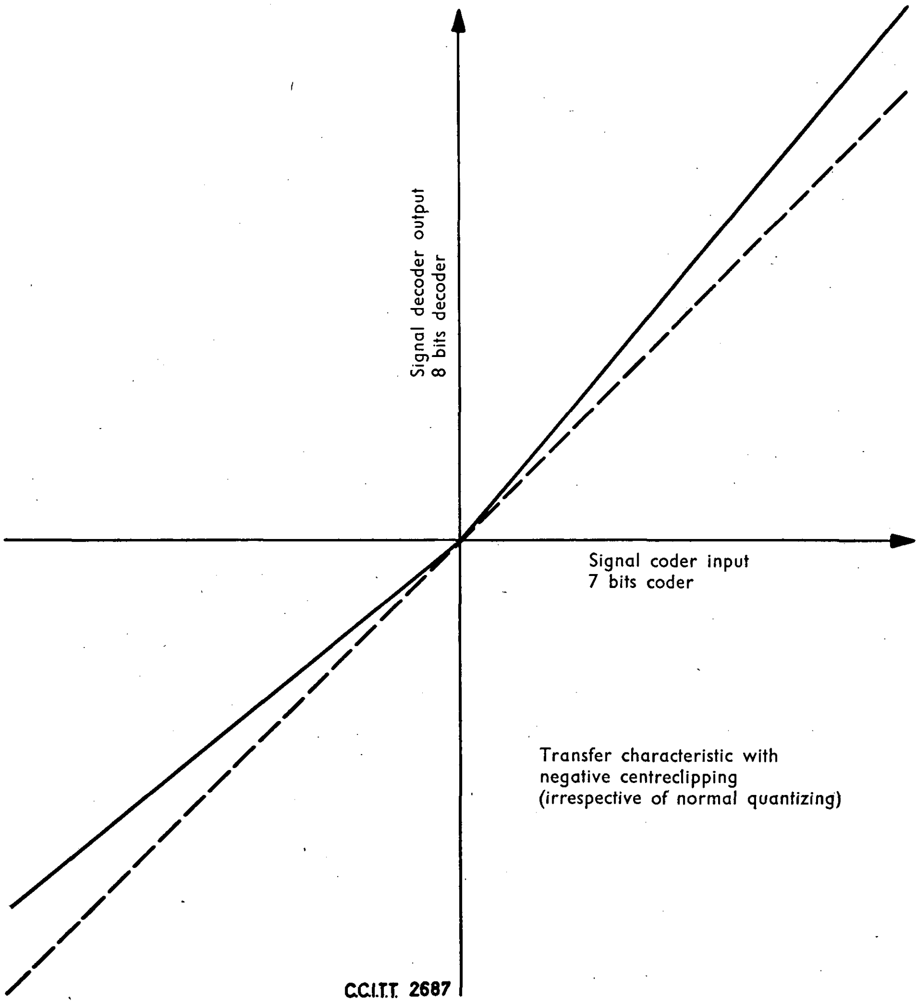


FIGURE 4

aforegoing that, globally, a broken characteristic is obtained as indicated in Figure 4. The difference in transfer between the positive and negative halves of the characteristic amounts to 2 × 0.183 dB = 0.37 dB. It can easily be shown that this leads to a distortion of about 0.9 % for a sinusoidal tone. Although this is still a rather slight distortion, preference should be given to the admission of the very small centre clipping without further distortion for signals in the normal amplitude range. In that case we still have the choice between positive and negative centre clipping. Here negative centre clipping leads to a greater step around the signal area " 0 ", which may result under certain circumstances in background noise or crosstalk that is 3.5 dB more intensive than with pure 7-bit systems. Positive centre clipping is to be preferred, because it has no such disadvantages. So, when a 7-bit coder is coupled to an 8-bit decoder, the 8th bit should be frozen on 0 in a symmetrical binary code.

2.3. *13-segment approach of the A-characteristic*

The aforegoing theory applies accurately to the continuous A-characteristic. At an approach in 13 segments the deviation Δx will not be exactly proportional to x, so that a small rest-effect will be noticeable as a quantizing distortion. Starting from the interpretation that over the whole signalling range the amplification has decreased by a factor $B = 1 - \left(\frac{\Delta x}{x}\right)$ mean $=$ $= 1 - 0.0213 = 0.9787$, the S/N-ratio has been calculated for the 13-segments approach of the A-law for three amplitude distributions, viz. a flat probability distribution, an exponentional probability distribution and a sine wave probability distribution. In the calculation no account has been taken of any filter-effects below 4 kHz.

The results are shown in Table 1.

TABLE 1

x		Flat probability distribution		Exponential probability distribution		Sine-wave probability distribution	
		7→7 8→7	7→8	7→7 8→7	7→8	7→7 8→7	7→8
	α	0.5	0.75	0.5	0.75	0.5	0.75
	β	1	0.9787	1	0.9787	1	0.9787
0.700		20.74936	20.47693	8.49578	8.44889	33.64466	33.32799
0.600		31.06823	30.67095	9.90602	9.85132	32.86902	32.95699
0.500		31.92974	31.88485	11.87742	11.81176	31.10156	31.09355
0.400		30.97069	30.86297	14.81645	14.73428	29.67171	29.64119
0.300		31.52734	30.98764	19.60220	19.49221	32.86895	32.95690
0.250		31.92963	31.88470	23.18939	23.05704	31.10148	31.09344
0.200		30.97051	30.86272	27.60311	27.44109	29.67160	29.64104
0.170		31.02738	30.68606	29.95036	29.77399	33.15309	33.35067
0.150		31.52685	30.98704	30.91417	30.73442	32.86842	32.95615
0.130		31.94923	31.96524	31.35318	31.17359	32.20678	32.17241
0.100		30.96905	30.86075	31.49423	31.31500	29.67075	29.63988
0.070		32.37335	32.38271	31.49217	31.30998	32.61437	32.62063
0.060		31.52622	31.51405	31.48546	31.30090	31.54793	31.55623
0.050		30.95739	30.84501	31.47149	31.28226	29.66396	29.63057
0.040		31.77031	31.39170	31.44114	31.24111	32.99399	33.13528
0.030		31.46520	31.43002	31.36246	31.13397	31.49932	31.48922
0.025		30.86522	30.72112	31.26866	31.00669	29.60930	29.55656
0.020		31.55627	31.12188	31.07424	30.74321	32.76854	32.81767
0.017		31.73365	31.52567	30.84977	30.44103	31.22819	31.17625
0.015		31.00562	30.81066	30.61052	30.12223	31.12566	30.98366
0.013		30.47012	30.13710	30.24773	29.64530	30.40944	29.84301
0.010		30.14267	29.41396	29.28198	28.41022	31.24487	30.82858
0.007		27.87110	27.02591	27.28416	25.98027	27.40924	26.75958
0.006		26.61436	25.58819	26.20350	24.71448	27.12035	26.04851
0.005		25.06057	23.84882	24.80292	23.10595	24.55946	23.47093
0.004		22.95488	21.58650	22.96848	21.03343	23.58897	22.31419
0.003		20.41462	18.70121	20.50130	18.27611	20.37677	18.52456

S/N in dB

The r.m.s. values of the signal are varied from 0.003 corresponding to −45.4 dBm0 up to 0.700, corresponding to +2 dBm0 if the overload point is +2 dBm0.

For r.m.s. values of x larger than 0.5 an overload effect can be observed for the flat probability distribution. With the exponential probability distribution this already begins at much lower values of x. In the sine-wave probability distribution no overload effect occurs. Since the flat probability distribution and the sine-wave probability distribution are abruptly limited, the results of the S/N-ratio are clearly non-monotonous; for certain signal values the results from the 7-bit coder to the 8-bit decoder are even more favourable than from 7-bits to 7-bits.

The smooth results of the exponential probability distribution are most appropriate for a direct comparison of the $7 \rightarrow 8$ bit results with the $7 \rightarrow 7$ or $8 \rightarrow 7$ bit results.

The extra degradation with a coupling of a 7-bit coder to an 8-bit decoder with regard to a 7-7 system has been indicated for exponentionally distributed speech in Table 2.

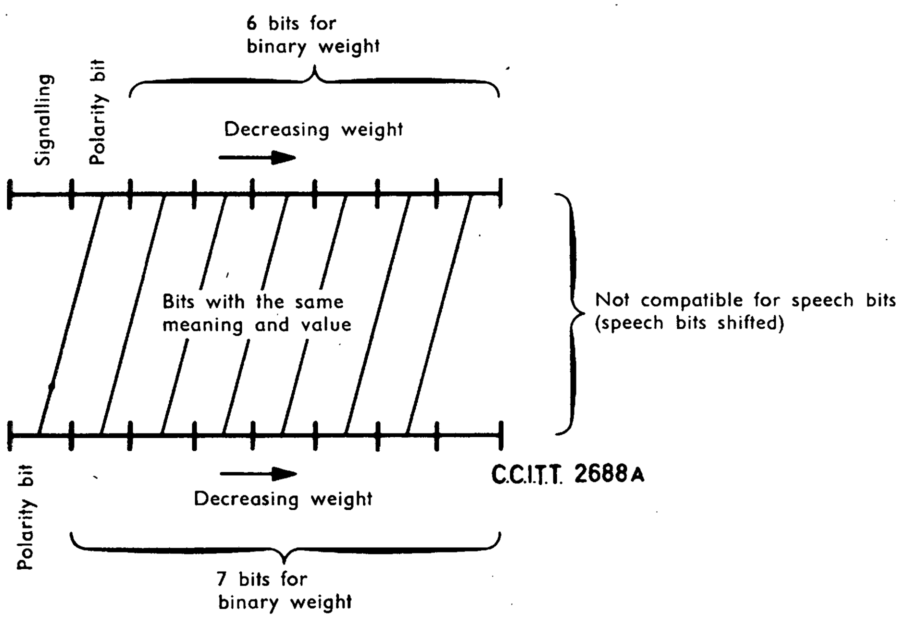
TABLE 2

Number of dB under the overload point	Extra degradation in dB
0	0.05
− 10	0.15
− 20	0.18
− 30	0.30
− 40	1.3

The deterioration in the S/N-ratio is slight and by all means acceptable. It is to be observed in this connection that the centre clipping effect in the exponential distribution is rather strongly represented owing to the character of this distribution. In the appendix it has been indicated in detail how the calculation of Table 1 was made.

2.3.1 *Sinusoidal signals*

The results in the columns relating to a flat probability distribution can also be interpreted as valid for saw-tooth or triangular signals. The columns relating to the sine-wave probability distribution can also be applied to sinusoidal signals. However, one must bear in mind that the results will only be exactly valid for sinusoidal signals, if the ratio between the frequency of the sinusoidal signal and the sampling frequency is not rational. In practice this means that one has to be cautious with the use of the resulting figures in the event of periodical signals. If there is a rational ratio between these frequencies indeed, certain various distortion components which are formed by reflection in respect of n times the sampling frequency will exactly overlap other products. The mutual phase relation between signal and sampling frequency is then of importance. From measurements in an experimental system it has appeared that products can be measured which are a number of dB stronger than the table indicates. If the frequency ratios are not rational, it is possible that beat effects occur, by which the result may be less favourable than the simple theory says, if the distortion in its r.m.s. value is not taken as an average for an interminably long time.


6 bits for
binary weight
Signalling
Polarity bit
Decreasing weight
Bits with the same
meaning and value
Not compatible for speech bits
(speech bits shifted)
Polarity bit
Decreasing weight
7 bits for
binary weight
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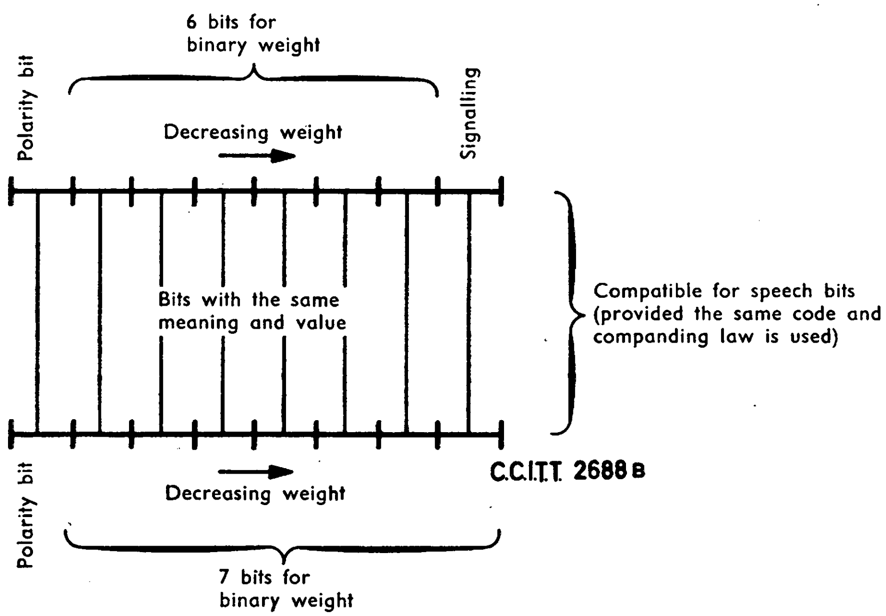

6 bits for
binary weight
Polarity bit
Signalling
Decreasing weight
Bits with the same
meaning and value
Compatible for speech bits
(provided the same code and
companding law is used)
Polarity bit
Decreasing weight
7 bits for
binary weight
C.C.I.T.T. 2688B

FIGURE 5

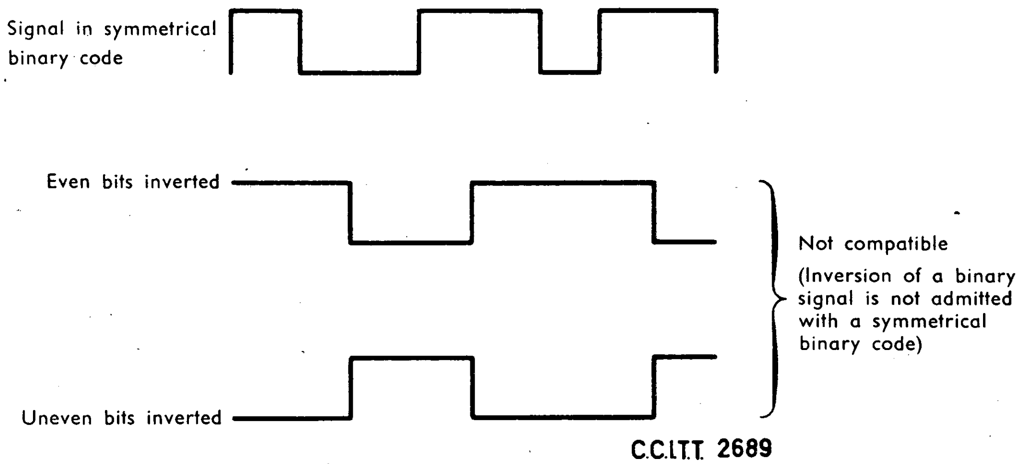


FIGURE 6

3. *Code*

A question of practical importance is the meaning and value which must be ascribed to the 8 bits in a time-slot in chronological order. Notably it is important whether a symmetrical binary coding or another type is applied, but also, whether in the 7-bit systems the 1st or the 8th bit in the time-slot is used for other purposes. The intercoupling of 7- and 8-bit systems will require considerably less modification in the equipment if the 8th bit and not the 1st bit is made a multi-purpose bit (see Figure 5).

In order to reduce the risk of a long occurrence of the zero signal the possibility has been indicated to send or transmit every second digit inverted. The manner in which this take places should also be fixed (see Figure 6).

In connection with the aforegoing the Netherlands PTT-Administration suggests to recommend the following :

a) symmetrical binary code in which uneven bits are inverted;

b) to incorporate this code in the first 7-digit places or in all 8 digit places of the time-slot;

c) if one of the 8-digit places is to be used for other ends the 8th digit place should be allocated for that purpose;

d) in the event of a coupling of a 7-bit coder to an 8-bit decoder the 8th bit should be frozen for the decoder (at the transmitting or receiving side) on the binary value 0.

4. *Encoding law*

In Europe a general agreement has been reached regarding the encoding law to be applied. The logarithmic characteristic with $A = 87.6$ combined with an overload point of +2 dBm0 leads to a background noise of maximum −61.2 dBm0 for the 7-bit systems and can lead to a S/N-ratio of approximately 31.5 dB for signals in the normal level range. For 8-bit systems 6 dB less background noise and crosstalk can be achieved, under certain circumstances provided that coders and decoders are used that are twice as accurate. At any rate a S/N-ratio can be achieved that is 6 dB better in the normal level range of the signals.

Also in view of the favourable properties of a logarithmic characteristic in connection with the compatibility between 7- and 8-bit systems the Netherlands Administration suggests to recommend the logarithmic characteristic with $A = 87.6$. An approach in 13 segments as proposed in document COM XV—No. 72 (1964-1968) is acceptable in view of the results of Table 1.

APPENDIX à l'Annex 4

Calculation of the signal-to-quantization distortion ratio with the use of the 13-segment " A "-coder and decoder

In the calculation two cases were distinguished:

a) 7-bit coding—7-bit decoding, also corresponding to 8-bit coding—7-bit decoding;

b) 7-bit coding—8-bit decoding.

The case 8-bit coding—8-bit decoding gives results for all values that are 6 dB better than case a).

Figure 7 shows the various amplitudes which occur in the coder and in the decoder. For case a) $a = \frac{1}{2}$, that is to say that the decoding amplitude lies in the middle of the coding interval. For case b) $a = \frac{3}{4}$ or $\frac{1}{4}$, dependent on the state of the 8th bit. If $a = \frac{3}{4}$, a slight amplification is noticed, since the decoding amplitude is always a quarter of a step above the middle of the coding interval. This amplification is 1/0.9787 at an average. Since this effect does not lead to quantization distortion, an attenuator $\beta = 0.9787$ has been assumed to be connected in series with the 8-bit decoder.

The quantization distortion which occurs per interval $a_n < x < a_{n+1}$ will then be :

$$\int_{a_n}^{a_{n+1}} \left[x - \beta \left\{ a_n + a\,(a_{n+1} - a_n) \right\} \right]^2 p(x)\,\mathrm{d}x$$

in which $p(x)$ is the probability density of the signal to be coded.

The total quantization distortion is :

$$p = \sum_{n=1}^{64} \int_{a_n}^{a_{n+1}} \left[x - \beta \left\{ a_n + a\,(a_{n+1} - a_n) \right\} \right]^2 p(x)\,\mathrm{d}x$$

$$+ \int_{a_{65}}^{\infty} \left\{ x - \beta\,(a_{64} + a a_{65} - a a_{64}) \right\}^2 p(x)\,\mathrm{d}x$$

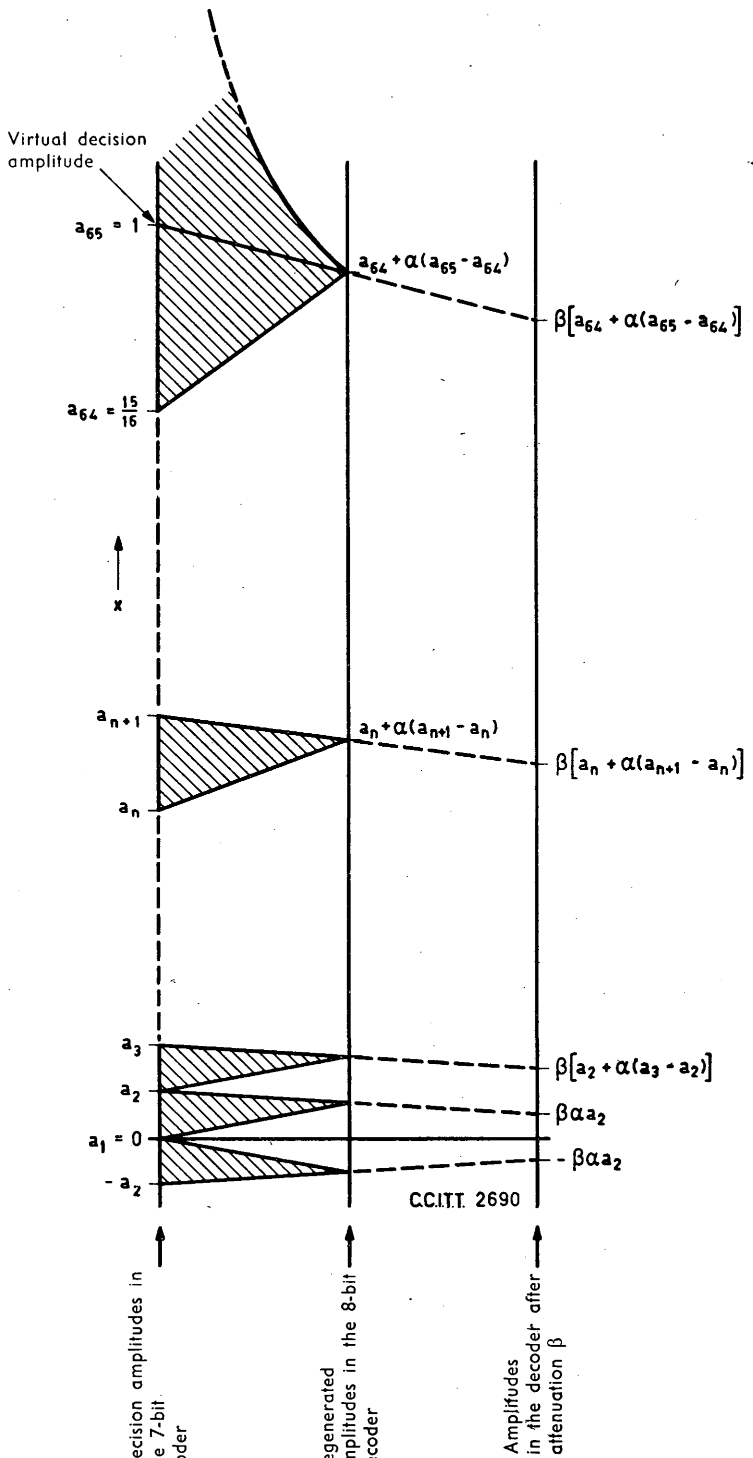
The last integral represents the distortion which occurs owing to peak limiting ($x < a_{65}$). The signal-to-distortion ratio has now been calculated by means of a computer for various distributions $p(x)$.

1. *Flat probability distribution*

$$p(x) = \frac{1}{\sigma\sqrt{3}} \qquad 0 \leqslant x \leqslant \sigma\sqrt{3}$$

$$p(x) = 0 \qquad x > \sqrt{3}$$

σ is the r.m.s. value of x.



Virtual decision amplitude
$a_{65} = 1$
$a_{64} = \frac{15}{16}$
x
a_{n+1}
a_n
a_3
a_2
$a_1 = 0$
$-a_2$
$a_{64} + \alpha(a_{65} - a_{64})$
$a_n + \alpha(a_{n+1} - a_n)$
$\beta[a_{64} + \alpha(a_{65} - a_{64})]$
$\beta[a_n + \alpha(a_{n+1} - a_n)]$
$\beta[a_2 + \alpha(a_3 - a_2)]$
$\beta\alpha a_2$
$-\beta\alpha a_2$
C.C.I.T.T. 2690
Decision amplitudes in the 7-bit coder
Regenerated amplitudes in the 8-bit decoder
Amplitudes in the decoder after attenuation β

FIGURE 7

2. *Exponential probability distribution*

$$p(x) = \frac{\sqrt{2}}{\sigma} e^{-\frac{\sqrt{2}}{\sigma}x} \quad x \geqslant 0$$

3. *Sine-wave probability distribution*

$$p(x) = \frac{\sqrt{2}}{\pi \sigma \sqrt{1 - \frac{1}{2}\left(\frac{x}{\sigma}\right)^2}} \qquad 0 \leqslant x < \sigma\sqrt{2}$$

$$p(x) = 0 \qquad x < \sigma\sqrt{2}$$

For the three distributions the following was calculated:

$10 \log \frac{\sigma^2}{p}$ as a function of σ, with $a = \frac{1}{2}, \frac{3}{4}$

and $\beta =$ 1, 0.9787

The decision levels a_n are, in accordance with the 13-segment compression law:

$$\begin{array}{ll}
a_n = (n - 1)\ 2^{-10} & 1 \leqslant n \leqslant 17 \\
a_n = (n - 9)\ 2^{-9} & 18 \leqslant n \leqslant 25 \\
a_n = (n - 17)\ 2^{-8} & 26 \leqslant n \leqslant 33 \\
a_n = (n - 25)\ 2^{-7} & 34 \leqslant n \leqslant 41 \\
a_n = (n - 33)\ 2^{-6} & 42 \leqslant n \leqslant 49 \\
a_n = (n - 41)\ 2^{-5} & 50 \leqslant n \leqslant 57 \\
a_n = (n - 49)\ 2^{-4} & 58 \leqslant n \leqslant 65
\end{array}$$

Question 22/XII — Revision of the Manual on Local Telephone Networks

(new question)

Revision of Chapter V of the Manual on *Local Telephone Networks*

Summary of Questions allocated to Study Group XII in 1968-1972

Question No.	Short title	Remarks
1/XII	National system reference equivalents in the new transmission plan	Of concern to S.G. XVI
2/XII	Assessment of service transmission quality	Of concern to S.G. II and XIII

Summary of Questions allocated to Study Group XII in 1968-1972 (continued)

Question No.	Short title	Remarks
3/XII	Asymmetry between the two directions of transmission	Reply to be transmitted to S.G. XVI (Question 5/XVI)
4/XII	Effect of circuit noise on transmission performance	
5/XII	Specification of sound level meters	
6/XII	Users' tolerance of echo and propagation time	Reply to be transmitted to S.G. XVI (Question 3/XVI)
7/XII	Determination of transmission quality by objective measurement	
8/XII	Measuring the efficiency of a microphone or a receiver	
9/XII	Limits applied in national trunk and local networks	
10/XII	Increase in the sensitivity of local systems	Of concern to S.G. XVI (Question 1/XVI, point 6). Reply to be transmitted to S.G. XVI
11/XII	Limits for intelligible crosstalk	
12/XII	Artificial voices, mouths and ears	
13/XII	Non-linear distortion of telephone apparatus	
14/XII	Extension of the bandwidth transmitted	
15/XII	Measuring of loudness ratings	
16/XII	Maintenance of subscriber sets	Question Africa H
17/XII	Loudspeaker telephones	
18/XII	Statistical study of the implications of Spanish phonetics for telecommunication systems	Question Latin America 5
19/XII	Impedance variations in subscriber lines and telephone sets	Of concern to S.G. XVI (Question 1/XVI, points 3 and 4)
20/XII	Synthetic speech and frequency compression systems	
21/XII	Transmission performance of pulse code modulation systems	Linked with Question 2/D
22/XII	Revision of the Manual on Local Telephone Networks	

PART III

SUPPLEMENTS TO RECOMMENDATIONS SERIES P

SUPPLEMENT No. 1

(Mar del Plata, 1968; referred to in Recommendation P.14)

SUBSCRIBER TOLERANCE OF LENGTHENED PROPAGATION TIME, ECHO AND ECHO SUPPRESSORS

(Contribution by the Telephone Association of Canada)

Introduction

The Telephone Association of Canada carried out a test programme during a 3-month period in 1965, to determine the probable reaction of Canadian subscribers to telephone connections involving intercontinental facilities with different propagation times (transmission delays). The intercontinental facilities considered involved submarine cable, medium altitude satellite systems and stationary orbit satellite systems, in combination with national extension circuits ranging up to 6000 km in length.

The test programme was conducted on the basis of simulating the various intercontinental facilities and national extension circuits by building-out a national telephone circuit between Montreal and Toronto to exhibit the appropriate transmission delays and noise. In addition, the test circuit was equipped with echo suppressors to simulate those included in the intercontinental facilities and, in some cases, with two echo suppressors in tandem to simulate situations involving an intercontinental circuit in tandem with a long national extension circuit equipped with an echo suppressor.

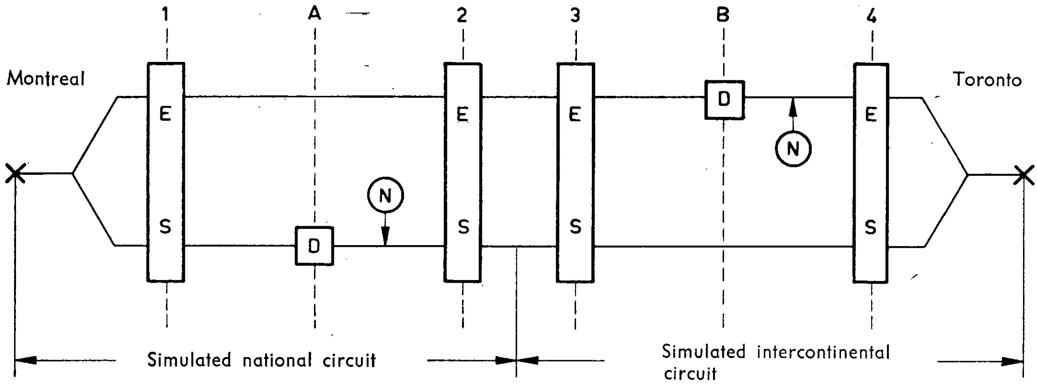
The built-out telephone test circuit was arranged to carry regular telephone traffic between Montreal and Toronto. Subscribers making calls over this test circuit were interviewed shortly after completing their call and requested to reply to specific questions concerning the quality of transmission rendered.

Details regarding the test arrangements, the results obtained and conclusions drawn are given below.

Test arrangements

The Montreal-Toronto circuit selected for the test programme is 550 km in length. The circuit was arranged at different times to simulate the 10 test conditions shown in Figure 1.

With reference to Figure 1, the intercontinental portion of the test connection was arranged to simulate submarine cable and satellite circuits of transatlantic length. The national extension circuits were arranged to simulate conditions in Canada involving service to centres located



Test cond.	Equipt. 1	A *	2	3	B *	4	Test circuit noise	I/C facility
1 2 3	0 1A 1A	20 ms 40 ms 60 ms	0 1A 1A	1A 1A 1A	70 ms 70 ms 70 ms	1A 1A 1A	−47 dBm0p −47 dBm0p −47 dBm0p	Submarine cable
4 5 6	0 1A 1A	20 ms 40 ms 60 ms	0 1A 1A	2A 2A 2A	300 ms 300 ms 300 ms	2A 2A 2A	−47 dBm0p −47 dBm0p −47 dBm0p	Medium altitude satellite
7 8 9	0 1A 1A	20 ms 40 ms 60 ms	0 1A 1A	2A 2A 2A	600 ms 600 ms 600 ms	2A 2A 2A	−47 dBm0p −47 dBm0p −47 dBm0p	Synchronous satellite
0	0	0	0	0	0	0	−57 dBm0p	

E/S = denotes echo suppressors, Bell System type 1A or 2A
D = delay apparatus
N = noise generator
* All values of delay reflect round-trip conditions.

FIGURE 1. — **Test conditions**

2000 km, 4000 km and 6000 km from the intercontinental circuit terminal. In the case of service to centres more than 2000 km from the intercontinental circuit terminal, an echo suppressor is involved in addition to the one included in the intercontinental circuit.

The noise on the test circuit was artificially set for the first nine test conditions at −47 dBm0p (43 dBrnc0). This is about 10 dB higher than the noise on the circuit in its normal condition (test condition 0). It is estimated that the value of −47 dBm0p (43 dBrnc0) would be representative of the noise conditions on transatlantic circuits in combination with Canadian extension circuits ranging in length from 2000 km-6000 km.

The variables in the test programme consisted of round-trip transmission delay, type of echo suppressor, and the arrangement of echo suppressors as indicated in Figure 1.

To facilitate administration of the test programme, only person-to-person calls between Montreal and Toronto were placed over the test circuit. Following each call, an attempt was made to interview the calling and called parties involved. Some 200 interviews were completed for each test condition to provide statistically adequate samples. In all, over 2300 interviews were conducted during the 3-month period between February-April 1965. It is to be noted that subscribers were not aware of the special character of the test circuit either before or after an interview.

Results

The measure of subscriber reaction to the type of facilities employed was based on replies to interview questions concerning difficulties in talking or hearing and the quality rating of the connection in terms of " excellent ", " good ", " fair " and " poor ".

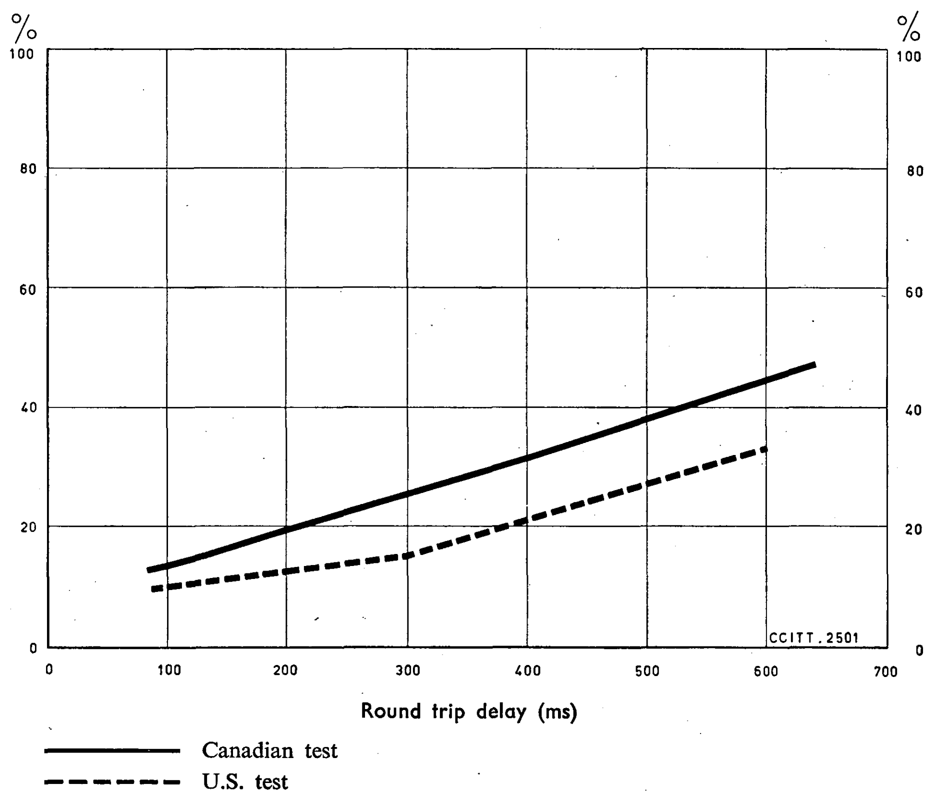


FIGURE 2. — **Percentage of interviews reporting difficulty combined over test conditions**

The results obtained indicate that national extension circuits with or without echo suppressors, as used in the test programme, have little effect on subscribers. The test results also show that the percentage of subscribers having difficulty in talking or hearing is governed largely by the magnitude of the transmission delay in the intercontinental facilities.

Figure 2 is a summary of the results obtained from the test programme in terms of subscriber difficulties (talking or hearing) vs. transmission delay. Figure 2 also contains the results of tests carried out between the U.S.A. and Europe, during 1964, in which satellite circuit transmission delays were simulated on actual submarine cable circuits [1].

Figure 2 indicates that the results of the Canadian and U.S.A.-Europe test programmes are similar with respect to the slope of the curves. The difference between the curves, in terms of absolute values, is attributed to subscriber expectation of service between Montreal and Toronto, as compared with that for the intercontinental service as rendered at the time of the U.S.A.-Europe tests.

The ratings of transmission quality obtained during the Canadian test programme in terms of " excellent ", " good ", " fair " and " poor " as a function of round-trip propagation time (transmission delay) is given in Table 1.

TABLE 1

Round-trip propagation time	Excellent	Good	Fair	Poor
100 ms	41 %	43 %	14 %	2 %
300 ms	37 %	40 %	16 %	7 %
600 ms	26 %	35 %	25 %	14 %

The mean opinion score with reference to the quality of transmission of the calls established during the test programme ranged from 3.3 for a round-trip delay of 100 ms to 2.7 for a round-trip delay of 600 ms. In developing the mean opinion scores, an " excellent " rating was given a weight of 4, " good " 3, " fair " 2 and " poor " 1.

Conclusions

The main conclusions to be drawn from the test programme are as follows :

a) A long transmission delay on a telephone connection (such as the delay introduced by a stationary orbit satellite circuit) constitutes a real and significant impairment to the quality of transmission rendered. This conclusion is based on subscriber opinions regarding difficulty in talking or hearing and the considerable increase in " fair " and " poor " ratings of calls at the expense of " good " and " excellent " ratings.

b) From the point of view of echo suppressor usage, a circuit which has a long round-trip transmission delay (up to 600 ms) and is equipped with an echo suppressor can be placed in tandem with a circuit which has a relatively short round-trip transmission delay (up to 60 ms) and is also equipped with an echo suppressor. In such a case, the transmission impairment introduced by the circuit with the long transmission delay is not significantly increased by the circuit which has a relatively short transmission delay.

The results of the test programme reflect the present attitude of Canadian subscribers towards national and regional (North-American) long-distance connections having different transmission delays. However, the results are also considered to be indicative of the future attitude of subscribers towards intercontinental connections as world-wide calling becomes more prevalent, and subscribers making such calls become more demanding.

[1] C.C.I.T.T. *Red Book*, Volume V *bis*, Annex E : Contribution by the Delegation of the United States of America.

SUPPLEMENT No. 2

(Mar del Plata, 1968; referred to in Recommendation P.14)

SUBJECTIVE EVALUATION OF TRANSMISSION PERFORMANCE ON TELEPHONE CONNECTIONS WITH LONG PROPAGATION TIME

(Contribution by K. D. D. — Japan)

A series of laboratory tests to evaluate the transmission performance on telephone connections with long propagation time were carried out on the test system shown in Figure 1 with a separate pair of echo suppressors especially designed for long propagation time. In the test, return losses were changed by means of the balancing network of the terminating set, and extra losses were added to each two-wire circuit equally or unequally. End-delay was introduced in only one test condition. The evaluation was based on the rejection rate given by the 26 subjects.

The end-delay of 17.5 ms (one-way) was introduced in the test circuit having 300 ms transmission delay and 8 dB return loss. The rejection rate increased from 12 to 24% when no protection was provided against end-delay.

Curve A in Figure 2 shows the rejection rate in the symmetric feature of losses on the two-wire extension. To check the soundness of judgement by the subjects, tests were conducted on the zero delay system (curve B). Table 1 shows the rejection rate in the asymmetric feature of losses on the two-wire extension. The impairment which gives the higher rejection rate is produced by the increase of the speech mutilation inevitably caused by a differential type echo suppressor.

TABLE 1

Test result

Reference equivalent approximately 25 dB

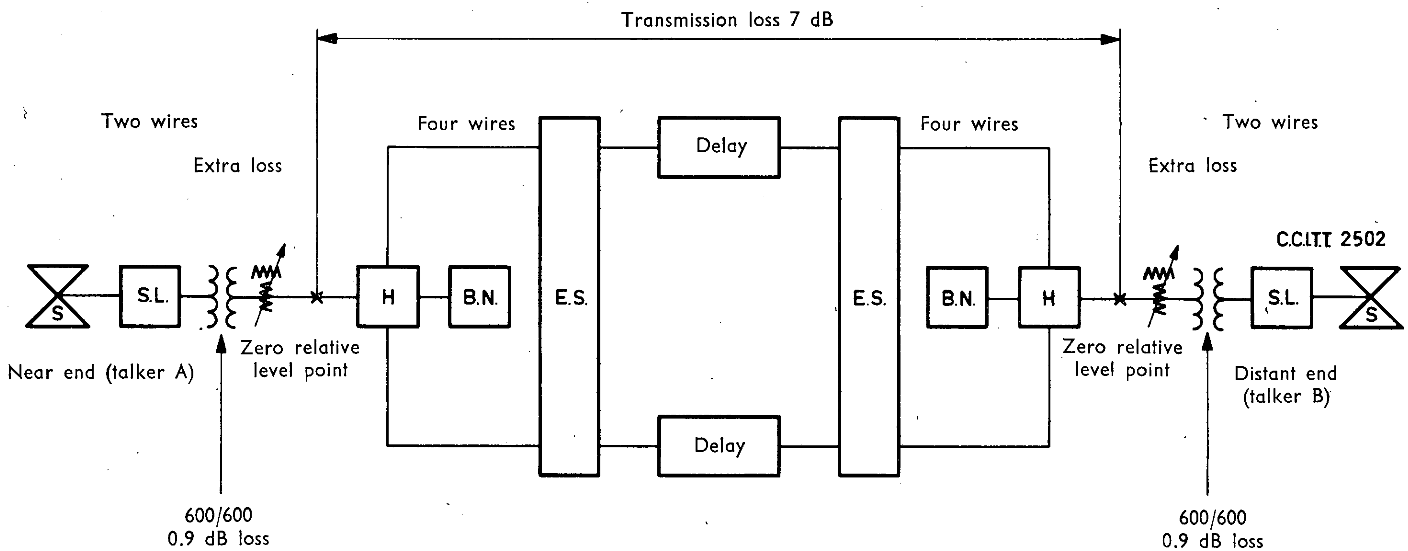
Subjects	On whose side 10 dB extra loss was added	On whose side 0 dB extra loss was added
Rejection rate	8 %	36 %

Reference equivalent approximately 35 dB

Subjects	On whose side 20 dB extra loss was added	On whose side 0 dB extra loss was added
Rejection rate	28 %	36 %

Figure 3 shows the deterioration of conversation affected by return losses of various magnitudes. The impairment is produced by residual echo in a double talk situation and clicks generated by the received signal sent back to a distant talker during the short period before the echo suppression takes place.

The following conclusions are drawn from these test results. Since the quality of transmission is impaired by the presence of the end-delay of as much as 18 ms (which seems to be maximum



S = Telephone set (type 600)
S.L. = Artificial standard subscriber's line (0.5 mm N.L. 3 km cable)
H = Terminating set (hybrid)
B.N. = Balancing network
E.S. = Echo suppressor

Room noise = 50 phones (Hoth)
Circuit noise = – 50 dBm0p
Reference equivalent = approximately 15 dB

FIGURE 1. — **Test system**

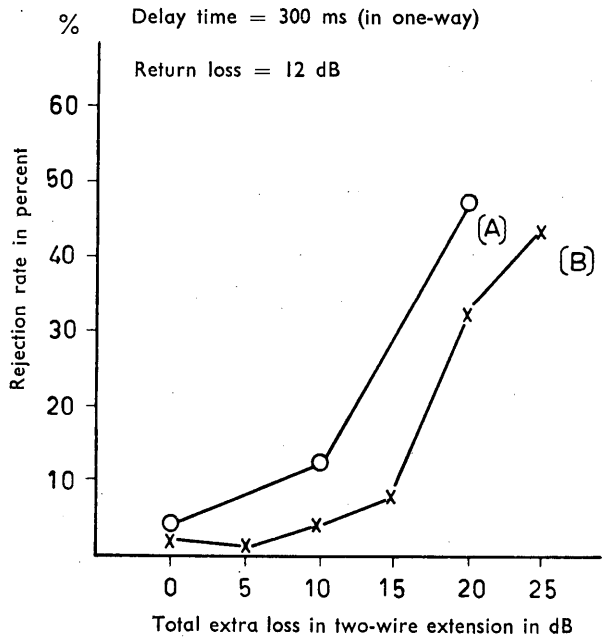


FIGURE 2. — Test result

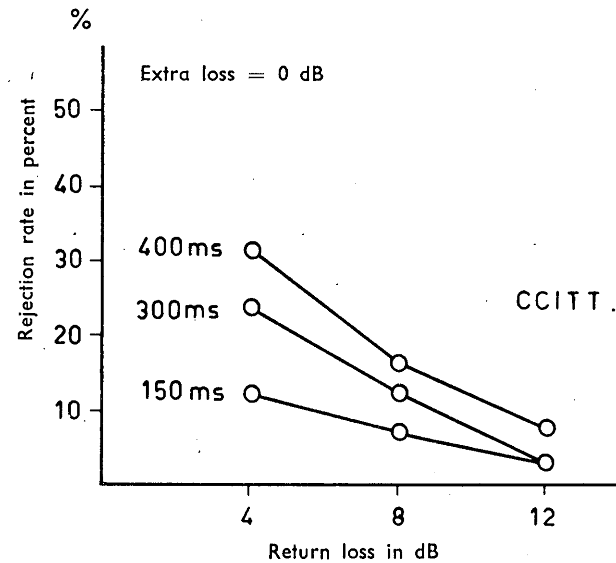


FIGURE 3. — Test result

CCITT.2503

for an average-sized country), care should be taken to design the echo suppressors so that some protection against end-delay is obtained. It is also noticed that the transmission quality observed in one side affected more appreciably by the circuit conditions such as return loss, two-wire extension loss and end-delay on the other side than by those circuit conditions on its own side, as far as the combined effect of delay and echo suppressor is concerned.

SUPPLEMENT No. 3

(Mar del Plata, 1968; referred to in Recommendation P.14)

SUBSCRIBERS' REACTION TO " EARLY BIRD " CIRCUITS

(Contribution by the Administrations of Denmark, Norway, Sweden)

Introduction

When commercial traffic via " Early Bird " commenced on 28 June 1965, the Scandinavian countries had three circuits with New York, terminating in Oslo, Stockholm and Copenhagen respectively, and routed via London.

The three Administrations found it desirable to obtain at least a subjective evaluation of how calls via " Early Bird " were accepted by the public, the idea being to make as close a comparison as possible with calls via cable circuits with New York.

The special echo suppressors were placed on the European side in the three capitals from where connections were established to all parts of Scandinavia. The satellite calls were restricted to the 500-mile (800 km) area from New York, thus avoiding the connection with circuits equipped with the normal type of echo suppressors in the United States national network. The satellite and the cable circuits were all operated manually.

Organization of tests

The tests were carried out in each of the three countries in accordance with an agreed plan and were divided into two parts. In the period 28 June to 11 September, the Scandinavian operators listened in on a total of 1624 satellite calls and made notes on the quality of the connection and on subscribers' reactions or comments. The subscribers were not told that they would be talking over a satellite circuit, and they were not interviewed. For the sake of comparison it was decided to establish the same arrangements for cable calls to or from the 500-mile region in the United States. This was done for 844 calls in the period 20 July to 11 September. The results of this first part of the tests are given in Annex 1 and commented on below.

In the second part of the tests (period 15 August to 11 September) subscribers were interviewed immediately after the calls in accordance with an agreed questionnaire. Each of the three terminal stations made interviews after 50 satellite calls and 50 cable calls, making a total of 300 calls. The subscribers were not told whether they had a satellite or a cable call. The same subscriber was never interviewed more than once, and in order to avoid any publicity about the current investigations and thereby any influence on other interviews, such subscribers as newspapers, news agencies, magazines, broadcasting organizations, etc., were not interviewed. In Annex 2 a summary is given of the results of the more important questions.

Conclusions

It is obvious from the foregoing that the statistical material available is very limited. Nevertheless it is not likely that investigations over a longer period and with a large number of circuits would produce substantially different results.

The analysis of the observed or reported difficulties should be taken with caution as most of these difficulties are attributable to factors having nothing to do with the satellite link or the transatlantic cable section.

The different categories of assessment have been entirely subjective and were justified only because they were employed on satellite calls and cable calls at random.

The main conclusion of the tests is that calls via " Early Bird " have been generally satisfactory to the Scandinavian public, and that the results of the subjective evaluation of such calls are very much the same as for cable calls. The number of reported difficulties which might be due to the delay peculiar to satellite transmission is quite small, and observations or inquiries have shown that the difficulties have been in no way serious.

ANNEX 1

Summary of operators' observations

Satellite circuits/cable circuits between Scandinavia and New York 500 miles (800 km) region

1624 satellite calls in period 28 June to 11 September
844 cable calls in period 20 July to 11 September

		Satellite calls/cable calls		
		Very good	Good	Fair
Signal strength	number	1365/640	244/190	15/14
	%	84.1/76.1	15.0/22.5	0.9/1.7
Freedom from noise	number	1358/645	248/179	18/20
	%	83.6/76.5	15.3/21.2	1.1/2.3
Overall merit	number	1366/641	237/190	21/13
	%	84.1/75.7	14.6/22.5	1.3/1.5

		Yes	No
Echo	number	21/3	1603/841
	%	1.3/0.4	98.7/99.6
Crosstalk	number	14/0	1610/844
	%	0.9/0	99.1/100.0
Signal level variations	number	25/3	1599/841
	%	1.5/0.4	98.5/99.6

Repetition asked for	number	47/14
	%	2.9/1.6

Observed subscribers' statements about quality of connection

Positive statements	number %	45/8 2.8/0.9
Negative statements	number %	19/12 1.2/1.4

ANNEX 2

Summary of replies from subscribers' interviews

50 interviews per country immediately after satellite calls
50 interviews per country immediately after cable calls

Question 3

Had you or your correspondent any difficulties in speaking or listening during the call ?

Reply

	Satellite	Cable
No difficulties	114	122
Some fading	6	2
Some echo	6	2
Difficult to hear the correspondent when speaking at the same time	5	0
Connection weak	3	11
Crosstalk and noise	10	7
Syllables disappear	6	3
Disconnected	0	3
Total ...	150	150

Question 4

Which of the following four terms does best describe the quality of the connection ?

Reply

		Satellite	Cable
Very good	number	70	81
	%	46.7	54.0
Good	number	61	56
	%	40.8	37.4
Fair	number	17	13
	%	11.3	8.6
Poor	number	2	0
	%	1.2 [a]	0
Total	number	150	150
	%	100	100

[a] Terrestrial part of circuit found faulty.

One of the questions in the interviews was : " How often do you have calls with the United States ? ".

The information thus obtained was related to the subscribers' evaluation of the calls just completed. It appeared that the actual evaluation of the calls, by satellite or by cable, was hardly dependent upon how frequently the particular subscriber had calls with the United States.

SUPPLEMENT No. 4

(Mar del Plata, 1968; referred to in Recommendation P.14)

TELEPHONE TESTS ON SATELLITE HS 303

(Contribution by Telespazio, Italy)

The experimental period of operating satellite Hs 303 started on 28 June and terminated on 19 November 1965.

The Intercontinental Telephone Centre of Italcable had during this period operated eight telephone channels via satellite, in addition to 15 cable channels, and has occasionally operated four radio channels.

In accordance with the agreements reached by A. T. & T. and the four European Administrations interested in the tests (the United Kingdom Administration, the French P. T. Administration, the German P. T. Administration and Italcable) testing carried out during this period aimed at ascertaining the :

1) Quality of transmission via the satellite,
2) Performance of the new echo suppressors, planned for satellite connection with a long propagation time,
3) Customers' reaction to the new means of communication, with special reference to the effects of a long propagation time.

The results of the experiment can well be said to have been most satisfactory in every way.

From analysis of the traffic, together with observations made by our operators, it emerges that :

1) The majority of subscribers did not notice any qualitative difference between the submarine cable system and the satellite one,
2) The voice quality during conversations via satellites was estimated by our operators as being better than that transmitted via cable,
3) The signal-to-noise ratio via the satellite is higher,
4) The new echo suppressors met the requirements of the system very well, and offered the customer a vivid impression of the other party's presence, besides avoiding the speech clipping that may sometimes occur.

A comparative assessment of the performance of the satellite system with that via cable is very difficult as far as Italy is concerned; this is because their efficiency is greatly influenced by imperfections of the distant interconnecting lines.

In other words, Italcable is of the opinion that a juxtaposition of the two systems of communications would ultimately be tantamount to comparison of the quality of interconnecting telephone circuits linking Rome and Paris for cable system, with those linking Rome and Frankfurt for satellite system.

Such an opinion is confirmed by the fact that on Saturdays and Sundays when the Fucino earth station was operated and connected to gateway by a short SHF connection, the quality of transmission was far higher.

ANNEX 1

Investigation of the acceptability of circuits via Early Bird
Outgoing calls New York area code

	On week-days	*On Saturdays/Sundays*	*Tota*
Via satellite	4254-40 %	736-39 %	4990-40 %
Via cable	6332-60 %	1153-61 %	7485-60 %
	10586	1889	12475

ANNEX 2

Investigation of the acceptability of circuits via Early Bird

The following data have been made available from the investigation of the acceptability to users of the service provided via the Early Bird satellite.

The period covered runs from 28 June to 19 November 1965, on the eight circuits operated.

	Outgoing calls	Incoming calls	Total
Number of calls investigated	4990	2970	7960
Calls affected by " poor transmission " without complaint after call or request for charge adjustment	15 (0.36 %)	36 (1.20 %)	51 (0.64 %)
Calls affected by cut-off, without complaint after call	143 (2.85 %)	46 (1.54 %)	189 (2.5 %)
Calls affected by " poor transmission " and by cut-off, without complaint after call	6 (0.12 %)	—	6 (0.08 %)
Calls with complaint after call by the users	4 (0.08 %)	17 (0.57 %)	21 (0.26 %)

No data are available on the incidence of transmission delay and echo suppressors on the percentages of the poor performance. However, the overall standards of acceptability can be considered as quite satisfactory.

SUPPLEMENT No. 5

(Mar del Plata, 1968; referred to in Recommendation P.14)

RESULTS OF MEASUREMENTS AND TRAFFIC OBSERVATIONS OF SATELLITE CIRCUITS

(Contribution by the Federal Republic of Germany)

In June 1965 the Federal Republic of Germany started participating in commercial traffic via the telecommunication satellite HS 303 " Early Bird " with the secondary group link Frankfurt (Main)-New York 6001 containing the primary group links Frankfurt (Main)-New York 1201 and 1202 with 14 telephone circuits and 2 VF telegraphy systems.

During the period 28 June 1965 to 12 November 1965 a number of service observations of the semi-automatically switched calls via satellite circuits Frankfurt (Main)-New York were carried out in agreement with the A. T. & T. schedule.

4622 connections were established during these months, each call lasting 7.7 minutes on the average. 95.7% of the total number of calls were without any reclamation, 4.3% being reclaimed (1.7% due to interruption during conversation, but immediately reconnected by the operator; 2.6% due to a poor intelligibility during conversation, but no complaint after the call).

As determined by former investigations made by A. T. & T. on 5000 telephone calls on cables 95.6% remained without any complaint, the remaining 4.4% giving rise to complaints.

The distribution of calls with respect to their duration is indicated in the following table. These figures are very similar to those encountered in cable communications :

Duration of calls (minutes)								Total
<3	4 to 6	7 to 9	10 to 12	13 to 15	16 to 30	31 to 60	>60	
18.1 %	29.9 %	18.1 %	10.9 %	7.3 %	13.6 %	1.9 %	0.2 %	100 %

SUPPLEMENT No. 6

(Mar del Plata, 1968; referred to in Recommendation P.14)

CORRELATION OF THE TELEPHONE TRANSMISSION IMPAIRMENT DUE TO LONG PROPAGATION TIME AND THAT DUE TO NOISE

(Contribution by the Telephone Association of Canada)

Introduction

A test programme to evaluate the subjective effect of long propagation time on telephone connections was carried out by the Telephone Association of Canada in 1965. These tests were made on a national telephone circuit which was modified to simulate the propagation time of different intercontinental circuits in tandem with national circuits of the type and length encountered in Canada. The propagation times selected for the simulated intercontinental portion of the test circuit were those of a submarine cable circuit of transatlantic length, a medium altitude satellite circuit and a stationary orbit satellite circuit.

The test circuit carried ordinary national traffic between the terminal cities involved which were Montreal and Toronto. Subscribers making calls on the test circuit were interviewed immediately after making their calls and their opinion of the calls solicited. The subscribers involved were not made aware of the fact that they were on a test circuit.

The results of the test programme were presented in terms of percentage of interviews reporting difficulty (talking or hearing) as a function of propagation time. They were also presented in terms of subscriber rating of the calls as " excellent ", " good ", " fair " or " poor ", as a function of propagation time. Details of the results and the test arrangements are given in Supplement No. 1 above.

The present supplement analyses the results of the 1965 test programme for the purpose of establishing a correlation between the transmission impairment caused by long propagation time and the amount of noise which would be necessary to provide an equivalent impairment.

Correlation of the transmission impairment caused by long propagation time and that due to noise

Results from the 1965 test programme concerning subscriber rating of calls are given in Table 1.

TABLE 1

Round-trip propagation time	Good or better (G + E) [1]	Fair or better (F + G + E) [1]
100 ms (submarine cable case)	84 %	98 %
300 ms (medium altitude case)	77 %	93 %
600 ms (stationary orbit case)	61 %	86 %

[1] F, G, E, refer to fair, good and excellent ratings respectively.

During the test programme, the noise on the test circuit was artificially set and fixed for all test conditions at 43 dBrnc0 (−47 dBm0p). This amount of noise was considered to be representative of the noise conditions on transatlantic circuits in combination with Canadian circuits ranging from 2000-6000 km in length. The noise received at a subscriber's telephone set in the majority of cases involved in the test was estimated to be about 32 dBrnc (−58 dBm0p). The average received talker volume was estimated to be about −28 vu (−29.4 dBm).

Data concerning the subjective effect on subscribers of received noise under conditions where propagation time is not a factor have been made available by the American Telephone and Telegraph Company in Contribution COM XII-No.77 (1964-1968 study period). Figure 1 is reproduced from this A. T. & T. contribution. Table 2 below indicates that the Canadian test results are in good agreement with the A. T. & T. data under conditions of relatively small propagation time (100 ms).

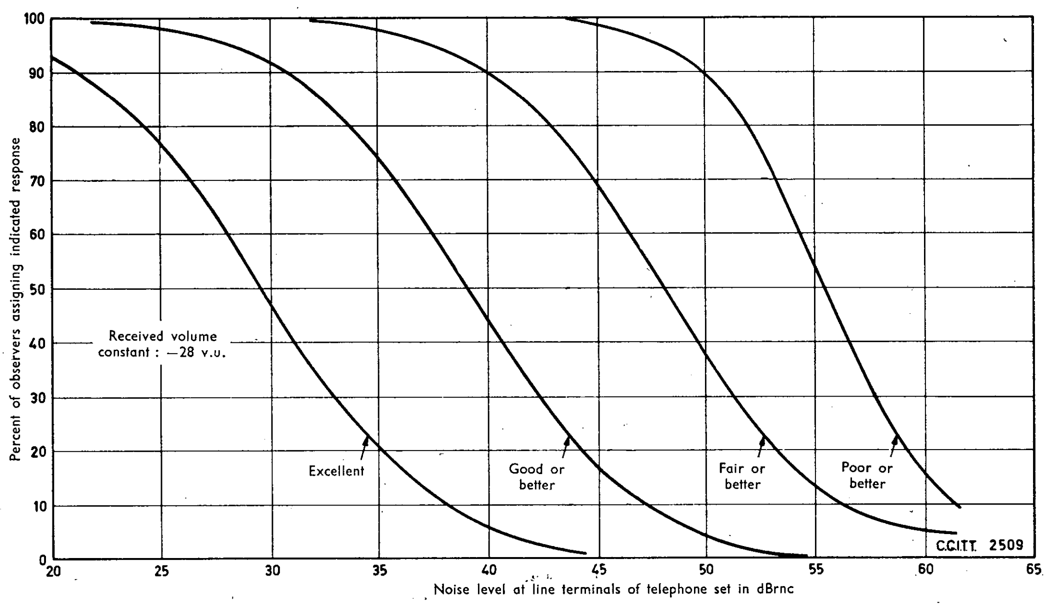


FIGURE 1. — **Noise opinion curves**

TABLE 2

Subjective effect of 32 dBrnc of noise at a subscriber telephone set

	Good or better (G + E)	Fair or better (F + G + E)
TAC test (Table 1) (100 ms condition)	84 %	98 %
A. T. & T. data (Figure 1)	87 %	99 %

Figure 1 and the data of Table 1 have consequently been used to determine the transmission impairment caused by long propagation time in terms of an equivalent noise impairment. The results are given in Tables 3 and 4.

TABLE 3

300-ms round-trip propagation time

Subscriber reaction (from Table 1)	Equivalent noise (from Figure 1)	Equivalent noise impairment (equiv. noise −32 dBrnc)
77 % good or better	34 dBrnc	2 dB
93 % fair or better	38 dBrnc	6 dB

TABLE 4

600-ms round-trip propagation time

Subscriber reaction (from Table 1)	Equivalent noise (from Figure 1)	Equivalent noise impairment (equiv. noise −32 dBrnc)
61 % good or better	37 dBrnc	5 dB
86 % fair or better	42 dBrnc	10 dB

Table 3 indicates that the equivalent noise impairment due to 300 ms round-trip propagation time ranges from 2 to 6 dB. Table 4 indicates that the equivalent noise impairment due to 600-ms round-trip propagation time ranges from 5 to 10 dB.

It can be argued that the higher end of each of the above ranges should be taken as the equivalent noise impairment since the larger values are derived from " fair or better " ratings which are also a measure of the " poor or worse " ratings (100% — fair or better). From a service standpoint, the " poor or worse " ratings are often the most critical and therefore controlling. However, if the " good or better " ratings are given the same weight as the " poor or worse " ratings, the equivalent noise impairment due to long propagation time would be at about the middle of each range as indicated below.

Round-trip propagation time	Equivalent noise impairment
300 ms	4 dB
600 ms	7 to 8 dB

Conclusions

The correlation developed above is based on present service expectations of Canadian subscribers making national and regional (North America) long-distance calls. However, this correlation should also be applicable to future intercontinental telephone service as world-wide calling becomes more prevalent and subscribers requiring this service become more demanding.

SUPPLEMENT No. 7

(formerly Annex 4, amended at Geneva, 1964, and Mar del Plata, 1968; referred to in Recommendation P.20)

METHODS APPLIED BY VARIOUS ADMINISTRATIONS IN INLAND LOCAL AND TRUNK NETWORKS, WITH A VIEW TO PROVIDING SATISFACTORY PERFORMANCE FOR NATIONAL CALLS

(it being understood that the C.C.I.T.T. Recommendation relative to reference equivalents is satisfied for international calls)

Argentine Republic
Australia
Austria
Canada and U.S.A.
Spain
Finland
France
Hungary
Indonesia
Italy
Japan
Malaysia
Norway
New Zealand
Pakistan
Netherlands
F. R. of Germany
United Kingdom
Singapore
South Africa (Republic of)
Sweden
Switzerland
Czechoslovakia
Zambia

Contribution by the Argentine Administration

Argentine is taking part in the work of the C.C.I.T.T.; its network will be brought into line with the standards laid down by that body for interconnections with networks of other countries and will be adapted to its national geographic and economic conditions.

The stability of the systems comprised in the network will finally determine the effective maximum attenuation between two subscribers. Preliminary analysis, based on normal equipment, shows that the maximum attenuation will not exceed 32 dB between the most separated subscribers, the C.C.I.T.T. limits regarding connection loss, reflection, balance, etc., being respected.

Contribution by the Australian Administration

1. *Introduction*

The methods adopted to provide satisfactory performance of a telephone network are intimately related to the size of the network and to the state of technical and other development reached. To enable the methods used in Australia to be appreciated in the light of these factors, the present state of the Australian network is described briefly below.

The telephone services in Australia are operated by a Federal Agency, the Postmaster-General's Department, and cover the entire continent of Australia. The area covered is similar in size to that of the U.S.A. A large part of this area is desert and the main areas of population are along the eastern and south-eastern seaboard and around Perth in Western Australia. A notable feature of the populated areas is the large distance (500 miles (800 km) or more) between larges cities. Approximately three-fifths of the telephones are located in five cities in the 500 000 to 2 500 000 population range. Cities exceeding a population of a few thousands are generally about 60 miles (100 km) or more apart.

The following data applied at June 1967 :

Telephones	3 178 000
Telephone services	2 234 000
Automatic telephone services	85.3%
Exchanges (automatic)	2 500
Exchanges (manual, mostly small)	4 000
Trunk lines	43 000
Calls	2 313 000 000
Trunk calls dialled by subscribers	18.6%

Up to 1964, local automatic exchanges were largely of the step-by-step type. Local areas were extended throughout Australia and unit-fee calls could be made over inter-exchange distances up to about 25 miles (40 km). Most trunk traffic was handled semi-automatically by one operator at the outgoing end over an automatic transit network with " two-wire " and " four-wire tail eating " switching and employing two v.f. signalling.

The switching equipment being provided at present (1968) is of the crossbar type employing two-wire switching of local traffic and true four-wire switching in the secondary and higher-order transit centres. This equipment facilitates alternate routing, and the switching pattern adopted is based on the hierarchical alternate routing principle.

Subscriber-trunk dialling (S.T.D.) has been introduced in 268 exchanges mostly over direct routes without transit switching. It is planned that at least 66% of all trunk calls will be directly dialled by the subscriber in 1975 over the crossbar transit network.

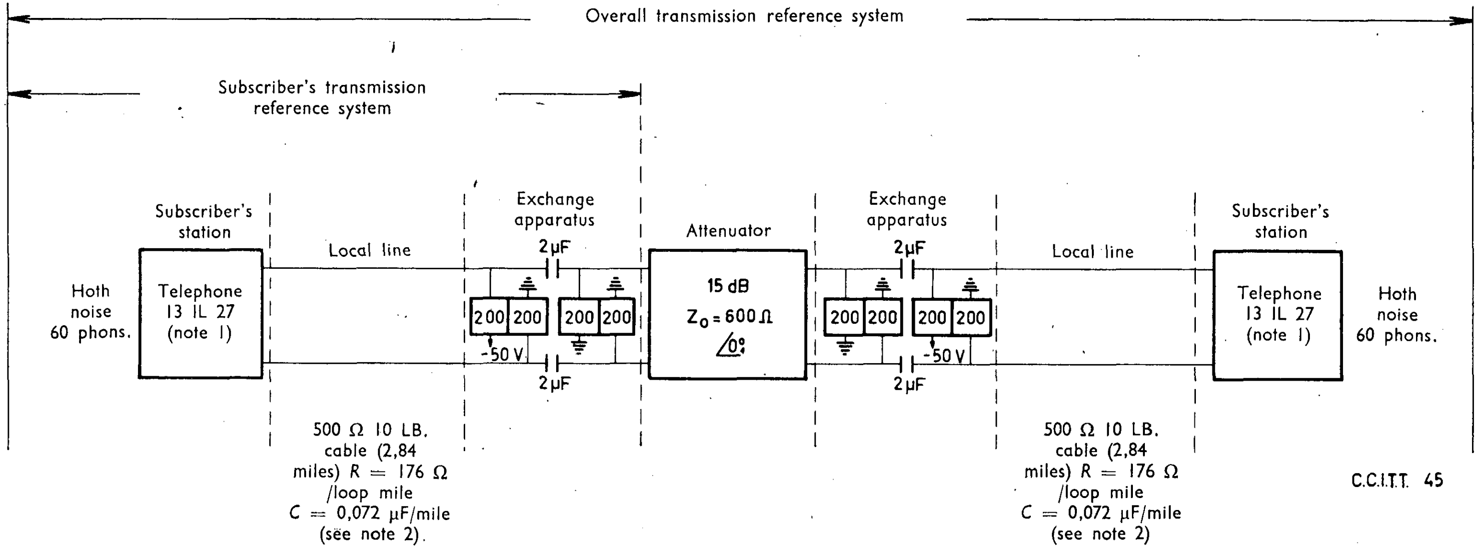
The increase in trunk traffic, the completion of the COMPAC and SEACOM cables facilitating intercontinental calls to and from Australia on good-quality cable circuits and the expansion of subscriber-trunk-dialling have increased the importance of echo-control, and more effort is now being directed to this feature.

Other factors which are taken into account in the network design are the introduction of more modern telephones with greater efficiency and automatic regulation, data transmission over the public telephone network, a variety of special services and the introduction of high-speed interregister signalling using multi-frequency code signals (MFC). The MFC signalling will operate on an end-to-end basis without regeneration (with some exceptions) rather than link-by-link, and this imposes certain minimum requirements on the performance of a chain of tandem-connected trunk channels.

2. *Objectives*

2.1 *Overall objective*

The transmission plan is based on an overall reference system comprising two telephone instruments with local lines and feeding-bridges joined by a 15-dB attenuator of 600 ohms impedance as shown in Figure 1. The network design ensures that overall transmission is better than the reference system for a very large percentage of connections and that only a very small



C.C.I.T.T. 45

Notes:

1. Telephone 13.IL.27. refers to handset type telephones having the following transmission components :

   Transmitter inset No. 13 }
   Receiver type IL } British Post Office designations
   Anti-sidetone induction coil No. 27 }

2. 0.635 mm dia, 4.56 km, 109.3 Ω/100p km, 44.7 nF/km
3. The telephones include transducers of minimum acceptable efficiency for new instruments

FIGURE 1. — **Overall transmission reference system (Australia)**

proportion of connections will be worse than the reference system. For design purposes the network is divided into two parts :

a) Local line network (subscriber lines)

b) Exchange-to-exchange network.

2.2 *Local line network (subscriber lines)*

The local line network is designed so that the transmission performance of exclusive exchange lines is at least equal to that of the local line section of the reference system which includes transducers of minimum acceptable efficiency for new instruments. This has been measured in Geneva and has a sending reference equivalent of 14 dB and a receiving reference equivalent of 6 dB. A receiving equivalent of 3 dB is exceeded on only 20% of existing services; no new service exceeds 3 dB. It will therefore be possible to reduce the overall reference equivalents by 3 dB at a later date.

Nearly half of the telephones in the Australian network are of the British Post Office 300-type, introduced in 1939. The 400-type telephone (of comparable sensitivity to the B.P.O. 700-type and the U.S. 500-type, but without automatic regulation) was introduced in 1957. The current standard telephone, introduced in 1963, is the 800-type, in which the transmission circuit is similar to that of the 400-type, with the addition of automatic regulation for loop length.

The conductors used in any given subscriber's cable are dependent on the more restrictive of two limits. The first is the transmission loss limit, and is a joint function of the line attenuation (expressed for planning purposes as the loss at 1.6 kHz) and the reduced transmitter feed current. The second is the signalling limit, beyond which the calling, dialling, or ring trip functions may be impaired. The signalling limit differs with different exchange equipment.

The present transmission limits for unloaded cables are shown in Table 1.

TABLE 1

Transmission limits for unloaded subscribers' cables

Conductor gauge * (lb/mile) Conductor diameter (mm)	4 0.4035		6½ 0.5075		10 0.6347		20 0.9025	
Telephone type	Modern	Obsolescent	Modern	Obsolescent	Modern	Obsolescent	Modern	Obsolescent
Resistance (ohms)	1150	670	920	600	770	500	610	400
Distance miles	2.62	1.52	3.41	2.22	4.37	2.84	7.0	4.55
Distance km	4.22	2.44	5.48	3.57	7.04	4.56	11.3	7.33
Attenuation at 0.8 kHz dB	6.55	3.8	6.58	4.28	6.73	4.37	7.49	4.87
Attenuation at 0.8 kHz dNp	7.52	4.37	7.57	4.92	7.73	5.02	8.62	5.59
Attenuation at 1.0 kHz dB	7.34	4.25	7.26	4.73	7.47	4.85	8.34	5.41
Attenuation at 1.0 kHz dNp	8.44	4.88	8.34	5.43	8.58	5.57	9.58	6.22
Attenuation at 1.6 kHz dB	9.16	5.32	9.1	5.88	9.32	6.05	10.2	6.64
Attenuation at 1.6 kHz dNp	10.55	6.12	10.48	6.77	10.7	6.97	11.9	7.65
Calculated line reference equiv. dB	6.98	4.08	6.73	4.38	6.59	4.28	6.58	4.32
Calculated line reference equiv. dNp	8.05	4.69	7.74	5.04	7.58	4.93	7.57	4.96

* Capacitance = 0.072 microfarad/mile = 44.7 nF/km.

Subscribers' cables may be loaded, using 88-mH coils spaced at 6000 ft. (1.83 km) where economically justified. The transmission limits for loaded subscribers' cables are shown in Table 2. Table 3 shows the present signalling limits in the Australian network.

TABLE 2

Transmission limits for loaded subscribers' cables (modern telephones only are used)

Conductor gauge (lb/mile) Conductor diameter (mm)	4 0.4035	6½ 0.5075	10 0.6347	20 0.9025
Resistance ohms	1500	1300	1200	1200
Distance miles	3.3	4.7	6.5	12.5
Distance km	5.3	7.55	10.45	20.1
Attenuation at 1.6 kHz dB	6.0	5.26	4.94	5.12
Attenuation at 1.6 kHz dNp	6.9	6.05	5.68	5.9
Calculated line reference equiv. dB	5.64	6.47	6.54	6.08
Calculated line reference equiv. dNp	6.48	7.44	7.52	6.98

TABLE 3

Signalling limits for direct exchange lines

Type of exchange	Line loop resistance (ohms)
Magneto and crossbar (automatic)	1500
2000-type; Siemens No. 17 (automatic)	1000
A.P.O. rural automatic exchange	750
Other exchanges	650

2.3 *Exchange-to-exchange network*

2.3.1 *General*

Each telephone exchange is classified according to its position in a hierarchical network. The principal exchange in each state is called a main trunk centre; other exchanges in descending order of switching importance are primary [1], secondary and minor trunk centres and terminal exchanges. The last type of exchange switches only for its own subscribers; it does not interconnect other exchanges. Figure 2 shows the six states of Australia and the five main exchanges. It also shows for one state only the hierarchical trunk exchange structure.

All main exchanges will be fully mesh-connected with final routes by the early 1970s.

In general, the transmission plan provides for a maximum nominal loss of 15 dB between any two exchanges, plus echo control losses (see para. 2.3.2). The two-wire loss between a

[1] Consideration is being given to elimination of this classification.

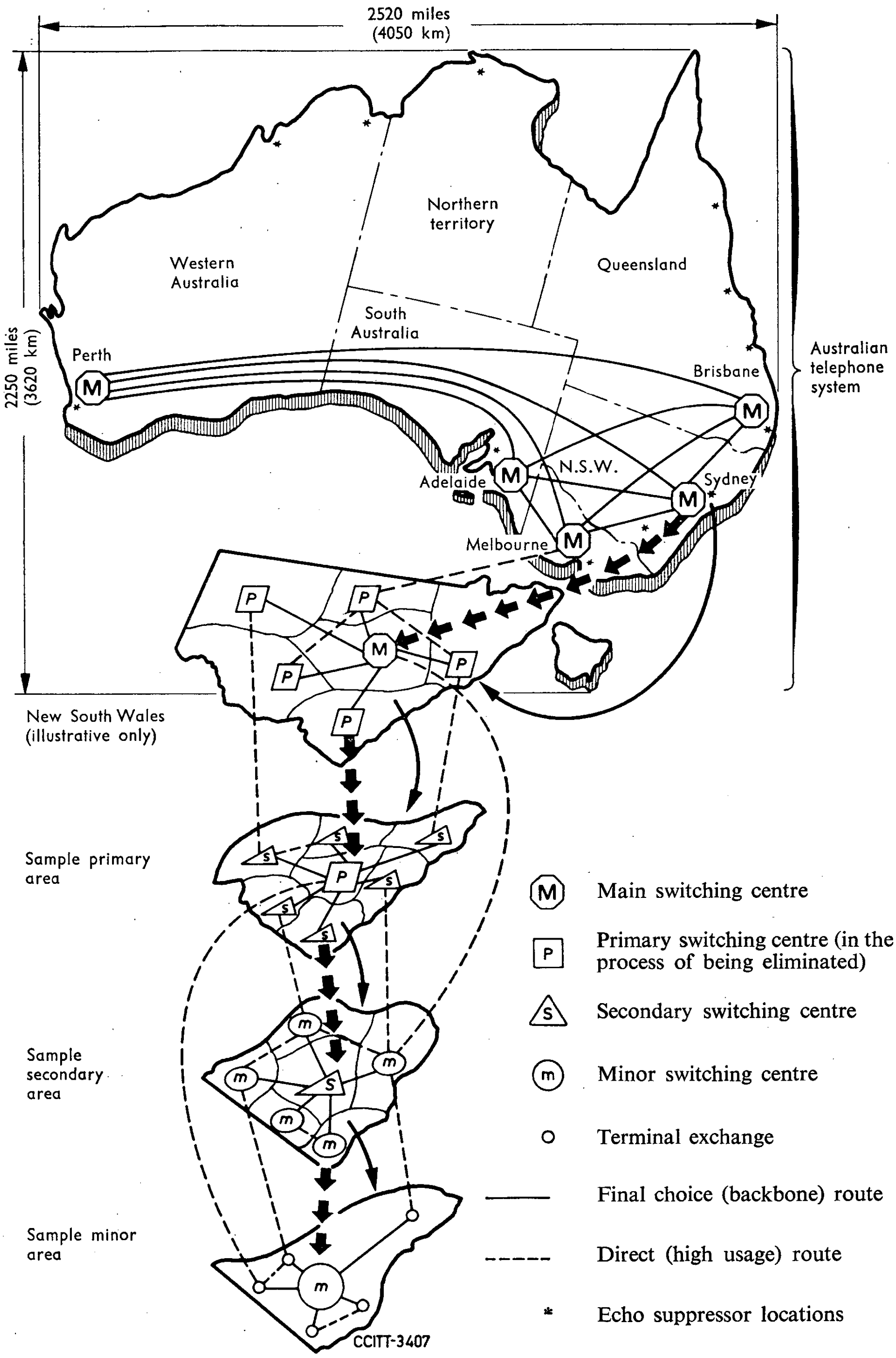


FIGURE 2. — **Australian trunk network structure**

terminating set and a terminal exchange has a minimum value of 3 dB to ensure adequate return loss.

In the following paragraphs the nominal design losses are those between virtual switching points and include line, transformer and switching equipment losses.

2.3.2 *Trunk network*

The design of the trunk network is based on the use of four-wire high-velocity circuits of low transmission loss between main trunk centres and between main, primary and secondary centres. These are provided over microwave and/or coaxial cable broadband bearers. The design requires echo suppressors to be fitted only on the longer-distance connections (longer than about 1500 miles) (2500 km). The nominal design loss of such circuits is 0.5 dB. All other four-wire circuits are assigned a nominal loss depending on three distance classifications in order to reduce the effects of talker echo. (See Table 4.)

TABLE 4

Echo-control losses

Distance		Loss
Miles	km	dB
0-350	0-563	0
351-750	564-1210	0.5
>750	>1210	1.0

The maximum nominal circuit loss between a terminal exchange and its switching centre (which may be any of the higher order centres) is 7.5 dB. Such circuits are generally provided by passive loaded cable pairs. Negative-impedance repeaters are used to a small but increasing extent in the Australian network.

Where the full 7.5-dB allowance is not required in the link between a terminal exchange and a minor trunk centre, a passive link may be provided from the minor trunk centre to its parent centre (i.e. secondary, primary or main trunk centre) provided that the total loss of the two links is between 3 and 7.5 dB. This implies that all the other terminal exchanges served by the minor trunk centre meet the same conditions; otherwise, the minor-to-parent link loss must be 0 dB.

Where direct circuits are provided between terminal exchanges the limiting circuit loss is 12 dB.

2.3.3 *Metropolitan networks*

In a metropolitan network the whole network is related to a single trunk exchange (generally a main trunk exchange) which connects to other trunk exchanges in the national trunk network. (See paragraph 2.3.2.)

Frequently a metropolitan area is served by a single numbering plan and calls within this area are switched clear of the trunk exchange via several tandem exchanges. In such cases, special rules have been drawn up to take advantage of the flexibility made possible by the availability of several tandem exchanges still using the same transmission allowances governing the design of the trunk network. With a single central tandem, the design methods are necessarily those of para. 2.3.2. However non-central tandems will be located closer to their dependent terminals and so have reduced losses on the interconnecting circuits. The balance of the transmission allowance is then available for direct circuits to foreign tandems, over lighter cables or shorter routes to give cheaper circuits. Calls may overflow via the final choice routes through the respective tandems.

The loss allocated to each type of circuit is not specified, but is determined for each network by distributing the allowable loss over the various junction groups in the most economical way. Most junctions are in 10 lb/mile cable (0.63 mm conductors) loaded with 88-mH coils spaced at 6000 ft (1.83 km). No allowance is made in the practical design procedures for mismatch losses. The transmission plan depends rather critically on the number and location of tandem exchanges in the network.

In practice, losses are typically as follows :

Tandem to terminal : 3 dB to 5.5 dB

Tandem to tandem : 6 dB down to 1 dB

Tandem to trunk : 1 dB to 3 dB

(Number of tandems : 17 down to 2)

2.3.4 *Stability*

The previous paragraphs have indicated the maximum nominal losses allowable in the Australian network. Where a connection includes four-wire circuits of low transmission loss it is also necessary to specify a minimum loss to ensure an adequate margin of stability on built-up connections. The minimum loss between terminal exchanges is 6 dB for connections including four-wire circuits. This minimum loss may be provided by the actual loss of the two-wire circuit between the four-wire circuit and the terminal exchange, or by building-out each two-wire circuit by means of a pad to a minimum loss of 3 dB, or (at a four-wire exchange) by building-out on the four-wire side of the terminating set. The trans-hybrid loss at the four-wire terminating set is improved to a satisfactory value by this added minimum loss between the set and the widely variable subscribers' lines.

Complementary to the provision of terminal links of minimum loss 3 dB is the provision of matching transformers on all loaded lines at the centres where these circuits connect to four-wire circuits. The standard matching impedance is 600 ohms.

At the centres where four-wire lines are switched two-wire, the effect of the matching transformers may be offset by the impedance unbalances contributed by signalling relay sets and different lengths of office wiring. The present plan provides for office balancing to reduce these effects, but little progress has so far been made in this direction.

2.3.5 *Echo suppressors*

Up to the present time (1968), echo suppressors have been in use on national calls only on circuits between Melbourne and Perth, a distance of 2180 miles (3500 km), the echo suppressors being associated with the circuit. On international calls, the echo suppressor is placed in the circuit at the international switching centre at Sydney regardless of the location of the terminal exchange in the national network.

An echo suppressor plan has been developed for the Australian national network and is in the process of being implemented. It requires the provision of echo suppressor installations at the five main switching centres and seven isolated lower-order switching centres, where half-echo suppressors will be connected by the switching equipment in such a way as to ensure that on a built-up connection no more than two half-echo suppressors will be in circuit. These will be located as near as possible to the terminal exchanges of the connection. Concurrently echo suppressors for international calls will be provided from the national installation nearest the terminal exchange for the call (in lieu of the present practice above).

3. *The future*

The Australian transmission plan is under continuous review assisted by actual performance checks of the network.

Recent surveys of subscribers' satisfaction have revealed the need for reduction of overall loss in the network to satisfy the more exacting desires of present-day subscribers. To meet this demand for lower losses on connections lower loss objectives are being set for achievement in a few years. The present and possible future overall loss objectives between terminal exchanges for total traffic originating and terminating in each secondary area (including own exchange traffic) are :

Estimated present achievement	*Possible objective in a few years time*
50 % less than 9 dB	50 % less than 6 dB
80 % less than 12 dB	70 % less than 7 dB
99 % less than 14 dB	85 % less than 8 dB
100 % less than 15 dB	100 % less than 9 dB

In addition, to minimize variations of loss between successive calls from any exchange, which may be connected over direct, alternate or backbone routes, a mean loss and a standard deviation of loss for the busy-hour traffic between each pair of terminal exchanges in a numbering area may be specified. Prescribed values are presently 12 dB mean loss and a standard deviation of 1.8 dB. Tentative objectives in a few years time are 7 dB and 1.5 dB respectively.

Reference. — R. G. KITCHENN. Telephone transmission objectives; *The Telecommunication Journal of Australia*, February 1968, Volume 18, No. 1, page 15.

CONTRIBUTION BY THE AUSTRIAN ADMINISTRATION

The development of Austria's automatic trunk telephone network is based on a transmission plan in keeping with the C.C.I.T.T. recommendations. In this transmission plan, the reference equivalent of the local system (subscriber's set + subscriber's line + feeding-bridge) and that of the trunk network have been fixed independently.

The local system

In the local system, the sending and receiving reference equivalent must not exceed 1.25 Np (10.8 dB) and 0.3 Np (2.6 dB), respectively. The maximum loop resistance of the subscriber's line has been fixed at 800 ohms. To allow for the different subscriber line lengths and the resulting differences in feeding current losses, the microphones and the telephone receivers in the subscribers' sets are grouped according to their sensitivity. In addition, the subscribers' sets are equipped with a pad.

The trunk network

Figure 1 gives a schematic survey of the Austrian automatic trunk telephone network, with an indication of the attenuation values between the switching centres. The local exchanges

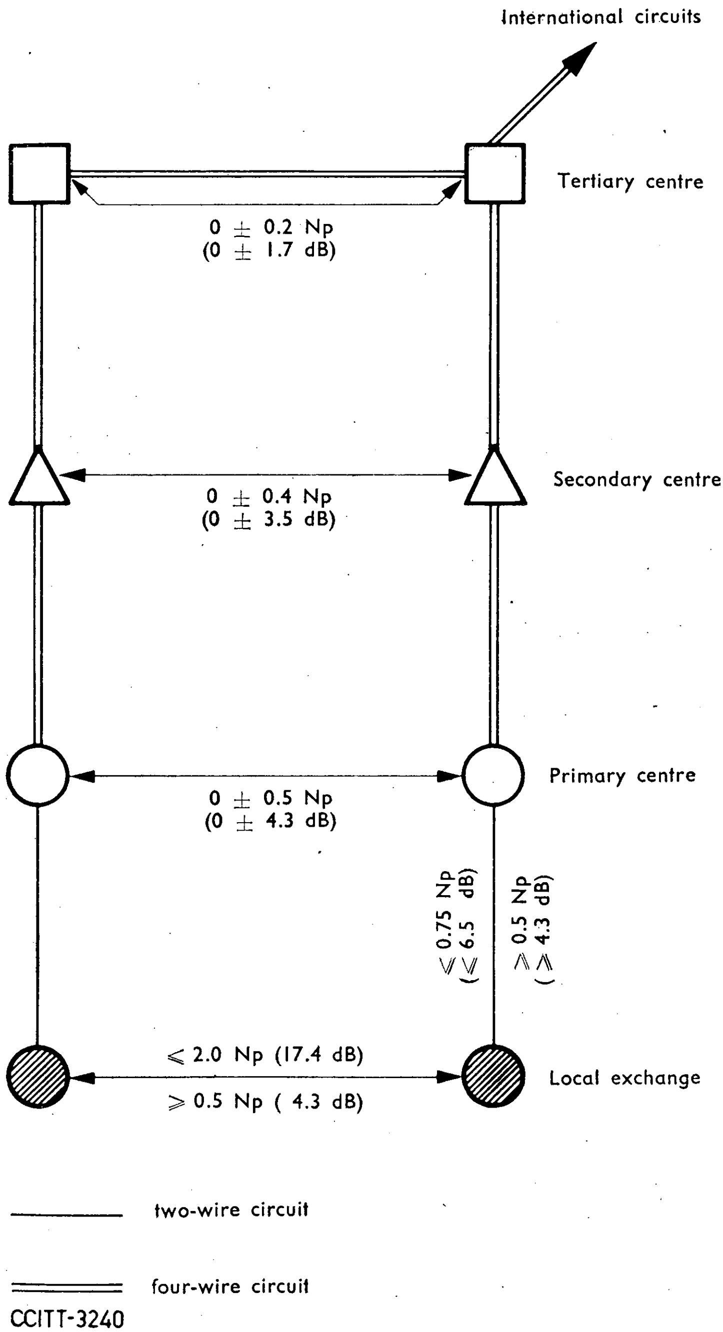


FIGURE 1. — **Attenuation diagram for the automatic trunk telephone network (Austria)**

are connected to the primary centres by two-wire circuits. For the other connections within the trunk network, four-wire circuits are used, with four-wire switching in the secondary and tertiary centres.

The maximum reference equivalent between any two local exchanges has been fixed at 2.0 Np (17.4 dB). The four-wire circuits are operated at zero loss. For reasons of stability and taking the attenuation variation of the four-wire sections into account, a minimum attenuation has been fixed for the two-wire circuits. This minimum attenuation is achieved, if necessary, by a pad.

In the national network, the maximum reference equivalent between any two subscribers' stations is 3.55 Np (30.8 dB) (including the attenuation variation).

In case of need, the network shown in Figure 1 can be supplemented by direct circuits between the various switching centres.

Contribution by the Telephone Association of Canada and by the American Telephone and Telegraph Co.

North-American practice for transmission requirements of the national network

This text outlines briefly the transmission practices at present in use on the North-American continent. Special mention is made of the major differences compared with transmission practices followed in other continents.

By way of general orientation the major differences between this method and other methods are :

1. The expression of transmission of other than " reference equivalent ";
2. The abandonment of the limiting loop concept in transmission design;
3. The substantial independence of the " loop and telephone set " transmission performance and that of the other links of a connection.

Transmission principles

The basic principle has been to give the subscriber the kind of transmission he himself finds satisfactory. Methods of doing this have changed from time to time, but fundamentally this approach to transmission design is based not on what subscribers will tolerate, but rather on what they prefer.

The general philosophy being applied in current reviews of transmission objectives is to provide a quality of service that will be rated by customers as GOOD in at least 95% of connections; FAIR in no more than 5% and POOR in a negligible percentage of cases due to specific trouble conditions. Guidance in achieving this goal is obtained by tests in which sample groups of subjects are asked to appraise a range of impairment conditions using categories of the GOOD, FAIR, POOR variety. Such subjective test results have begun to influence the design of the North-American network and they have been used to evaluate the present performance. As an example, it has been shown that the general service objective has not yet been fully achieved with respect to received talker volumes.

Philosophies as to how to apportion the permissible transmission losses among the various links of a connection have varied over the years. At one time, for example, there was an assigned value for " loop and trunk " portions; and extensive studies were made in individual cases to determine the most economical division of the value between loops and trunks.

The general method of transmission design outlined below is based on :

1. Great improvements in the efficiency of the telephone set;
2. Decreased cost of providing transmission facilities between local exchanges and for long-distance connections;
3. Considering both cost and time of utilization of individual links.

Subscribers' lines (loops)

1. The line conductors are designed on the basis of assigning the minimum amount of copper necessary to meet the limitations on d.c. resistance imposed by the signalling, pulsing, and supervision requirements of the central office switching circuits.

For the most widely used types of switching systems, satisfactory operation will be obtained with external loop d.c. conductor resistance up to 1200 to 1300 ohms. This is approximately equivalent to the lengths shown in the following tabulation :

Length of pair (feet)	Gauge	Diameter (mm)	Pounds per conductor mile
15 000	26	0.40	4.1
24 000	24	0.51	6.5
38 000	22	0.64	10.3

2. In order to reduce the spread in transmission performance between the longest loops and the average loops, loading coils generally of 88 mh are introduced at 6000 feet spacing on all loops longer than 18 000 feet. The first loading coil is located 3000 feet from the central office so as to ensure maximum effectiveness in connections between loaded loops and between loaded loops and trunks.

3. Bridged taps are kept to a minimum and in any case are usually limited to a maximum of 6000 feet.

With subscriber plant designed on this basis, calls between any two telephones connected to the same switching office would be rated GOOD with respect to volume by virtually all subscribers, and hence would be in line with the broad over-all objective.

This method obviously greatly simplifies engineering (see [1], [2], [3]). Obviously, too, it might need modification with changes in switching systems, telephone apparatus, and signalling requirements or customer demand for improvement in service. Its development involved trial application to various types of central office areas as regards length and distribution of loops, the comparison of the effective transmission performance with that of previous methods, judgement as to how much to allow for losses in trunks. The allowance leaned heavily on the trend toward very low losses in these circuits, and specifically the more widespread use of carriers in very short lines.

Trunks

In 1964, a change in definition of trunk terminal points was made in order to make trunk design losses consistent with measuring techniques. Previously, gains and losses between the centre of switch of the originating office to the centre of switch of the terminating office were considered in trunk design. A trunk is now considered to consist of all gains and losses between the outgoing switch appearance at the originating end and the outgoing switch appearance to which the trunk is connected at the terminating end.

In the present method, battery supply losses which were formerly considered to be part of the loop are now considered to be part of the trunk. Consequently, trunk design loss values were adjusted to take this into account. Therefore, while the new trunk loss objectives appear to be larger in magnitude than those previously stated when the change in terminal points is considered, they are actually coincident.

Small changes in maximum values of long-distance circuits have also been made. These changes were necessary in order to make design losses compatible with automatic transmission measuring equipment.

Direct and tandem trunks

These are facilities to provide connections between two local offices. This is shown diagrammatically in Figure 1. The small permissible spread between nominal and maximum should be noted. If the route distance between end offices is greater than 100 miles long-distance objectives as shown on Figure 2 should be applied.

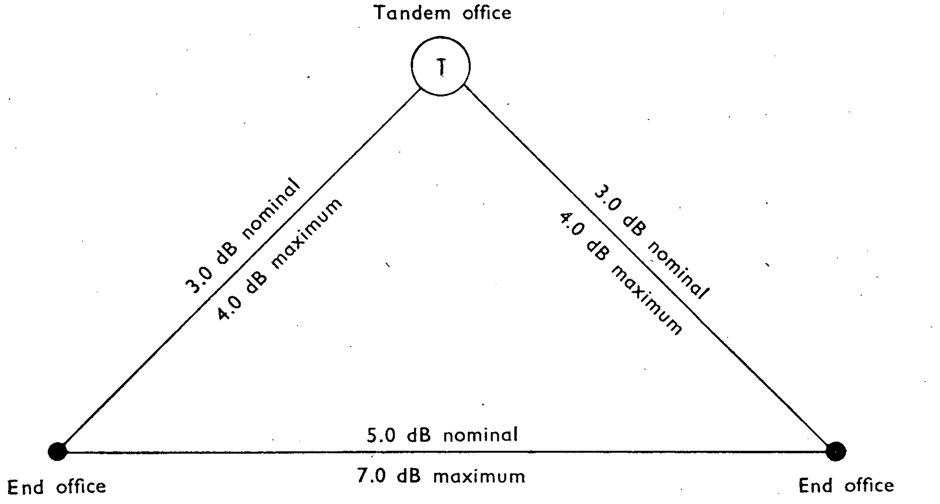


FIGURE 1. — **Local network (North America)**

Toll connecting trunks

These are facilities between local offices and long-distance switching systems. They are operated at losses between 3 and 4 dB. The objective is to provide a loss of V.N.L.+2.5 dB with a maximum of 4 dB. V.N.L. (or via net loss) is defined as the lowest loss in dB at which it is desirable to operate a trunk facility considering limitations of echo, crosstalk, noise and singing. From a knowledge of the various types of facilities employed, it has been possible

NORTH-AMERICAN NETWORK

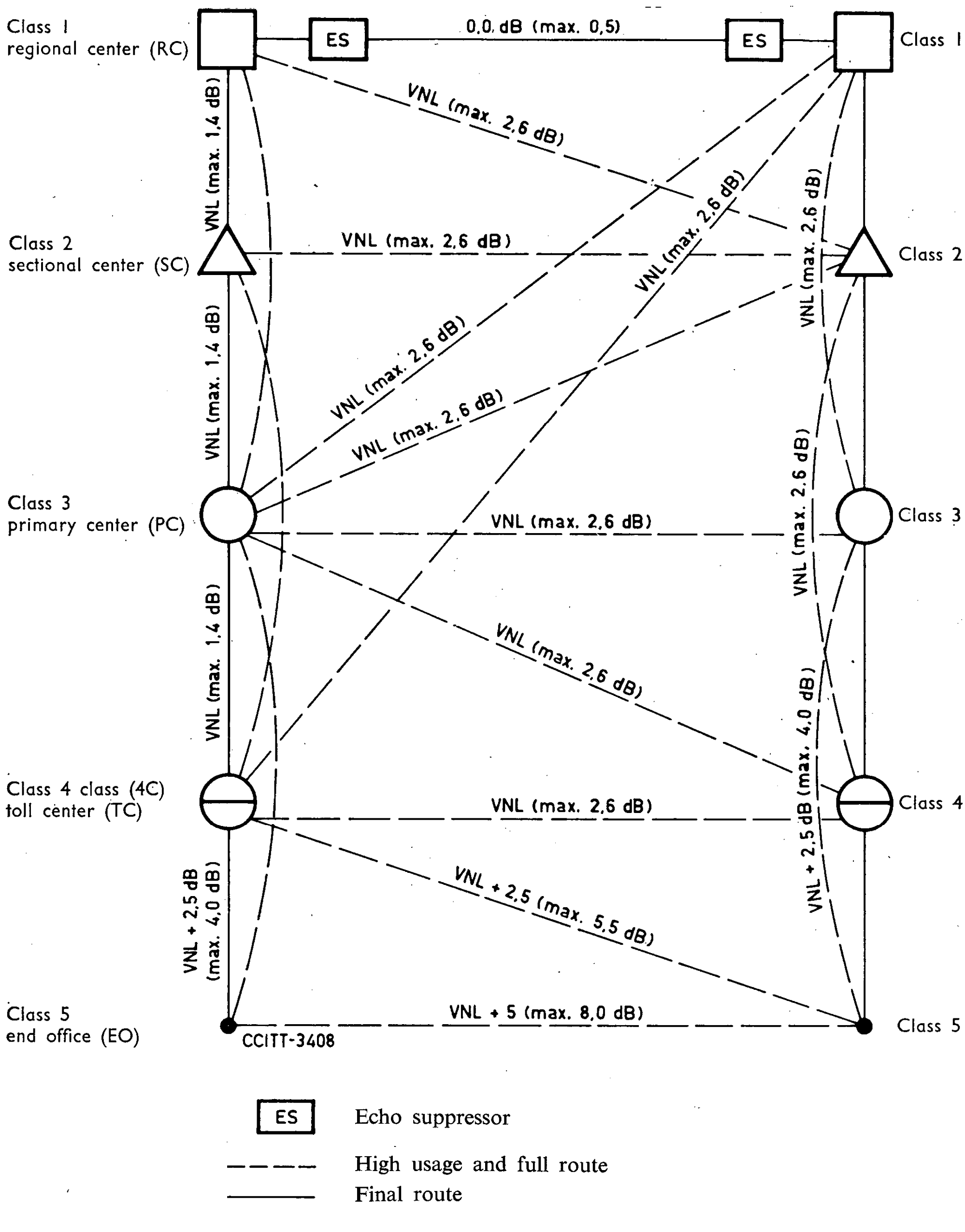


FIGURE 2. — **Trunk losses with via net loss (V.N.L.) design (North America)**

Note by the Secretariat.— This figure shows the designations of the various exchanges used in North America (with their abbreviations). These designations do not correspond to C.C.I.T.T. terminology; in particular the " primary centre (PC) " does not meet the definition of a primary centre applied by the C.C.I.T.T.

to specify the V.N.L.[1] for each type on a per-mile basis. A minimum loss of 2 dB is required (which in the case of short trunks is provided by inserting a resistive attenuator) to ensure stability of overall connections.

Long-distance circuits

These are the circuits between switching centres in the continent-wide long-distance network. The losses of final trunk groups are the V.N.L. losses, i.e., the smallest loss permissible in each case from standpoint of echo, singing and crosstalk. A further requirement of 1.4 dB maximum is imposed so as to ensure low loss; high velocity facilities are assigned for this service. Interregional high-usage trunks (see Figure 1) are permitted to have a maximum loss of 2.6 dB.

Thus it may be seen that the objective for the average loss in a connection between two local offices is about 5 dB if the connection is via direct or tandem trunks; or somewhat greater if via long-distance circuits, this varying with the number of links. In 83% of calls only a single long-distance circuit is involved; in 15% two; and three or more in the remaining 2%.

A recent survey * shows that the average end office to end office loss on all toll calls was 7.5 dB. Its breakdown by airline mileage categories is :

7.2 dB for 0 to 180 miles (0 to 290 km),

8.9 dB for 180 to 725 miles (290 to 1160 km),

9.6 dB for 725 to 2900 miles (1160 to 4650 km).

Discussion

As indicated, in North America, the transmission design of subscribers' loop and telephone sets has been largely divorced from that of trunks, and both are such that the concept of national transmitting and receiving " reference equivalents " now used in the C.C.I.T.T. is not applied. While this method was formerly used in North America it was abandoned when evolution of new telephone instruments decreased its meaning. For example, it was discovered that the very real quality improvement due to introducing a substantially flat frequency response in telephone instruments was not reflected in the loudness balance measurements of a reference equivalent. To circumvent this an effective loss method was introduced in which comparisons between the various loops and the various sets were made on the basis of the rate of repetitions asked for by subscribers during normal conversations.

Effective loss objectives were then established for the subscriber loop plant and data relating effective losses to loop make-up and sets were used for practical design purposes. The resistance design method now used, described above (see section 1), was later conceived as a more simple and practical method to use to stay within the effective loss limits and at the same time to meet loop signalling limitations.

Rating on the basis of repetitions has grown to be a rather insensitive measure of performance, and effective losses are by nature difficult to relate to reference equivalents. On the North-American continent, it again seems practical to revert to the principle of loudness ratings

[1] For a detailed discussion of V.N.L., see [4]

since telephone instruments and trunks are now of a uniformly high quality. The design of trunks has become a matter of 1000-Hz loss since this quantity is an adequate measure of the effect of modern facilities on transmission performance. An electro-acoustic rating system [6] which measures loudness loss objectively is being studied in order to put the evaluation of loops and sets on a similarly objective basis. It is expected that such ratings can be related to reference equivalents assuming the use of similar high-quality telephone instruments in both cases. It is appreciated that such a relation will not take into account the difference in performance due to side-tone, frequency response and similar factors which the basic rating systems do not evaluate.

References

[1] Simplified transmission engineering in exchange cable plant design; *Communication and Electronics (A.I.E.E.)*, November 1954.

[2] Transmission economics; *Telephony*, 23 August 1958.

[3] Subscriber loop design; *Telephone Engineer and Management*, 15 September 1961.

[4] Transmission design of intertoll telephone trunks, *Bell System Technical Journal*, September 1953.

[5] I. Nasell : Some transmission characteristics of Bell System toll connections; *Bell System Technical Journal*, Volume 47, July-August 1968.

[6] A revised telephone transmission rating plan; *Bell System Technical Journal*, May 1955. (reproduced in the C.C.I.T.T. *Red Book*, Vol. V, pages 607-624).

Contribution by the Compañia Telefónica Nacional de España (Spain)

1. *Principles*

The maximum reference equivalent admitted for a national call is 32 dB. In exceptional cases, a value of 35 dB can be admitted, as long as the national connections of this type do not exceed 5% of the total.

Figure 1 is a schematic diagram of the national automatic network, showing the maximum attenuations admitted for the various circuits of the network.

The automatic links between national exchanges, whether they are direct circuits or final circuits, are set up on four wires, with a gain of 1 dB on the four-wire sections.

Links between sector exchanges or tandem exchanges and the national exchange use four-wire circuits with a gain of 0 dB.

All four-wire trunk circuits have a nominal adjustment of 8 dB between the two-wire terminals, which leaves a maximum local equivalent of 12 dB, including the attenuation due to switching in the sector exchanges and the terminal exchanges.

2. *Reference equivalents*

2.1 *Reference equivalents assigned to a primary area (sector)*

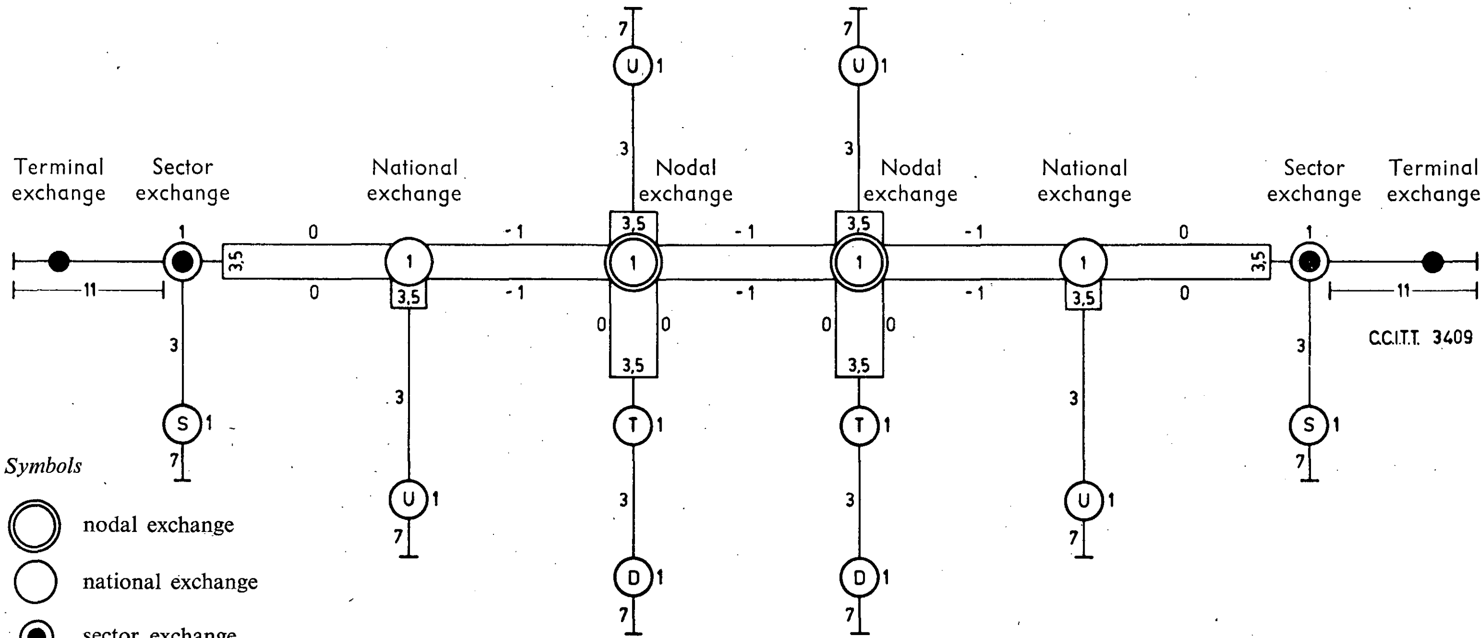
The subscriber line plus the local circuits are considered.

The maximum reference equivalent, measured at 800 Hz, is 12 dB.

The attenuations due to the local exchange and the primary centre have to be subtracted from these values.

2.2 *Nominal reference equivalent between two exchanges*

Between 22 dB and 18 dB.



Symbols

- nodal exchange
- national exchange
- sector exchange
- terminal exchange
- (T) tandem exchange
- (U) local exchange nodal of a national exchange
- (D) local exchange depending on a tandem
- (S) local exchange depending on a sector exchange
- 7 subscriber loop (7 term. ref. equiv.)

The numbers refer to the loss in dB.

FIGURE 1. — **Transmission plan for the national automatic networks (Spain)**

2.3 *Attenuations assigned to a local circuit*

These attenuations, measured at 800 Hz, lie between 4 and 6 dB, depending on the size of the primary exchange area.

The attenuation due to the primary exchange has to be subtracted from these values in order to obtain the value assigned to line attenuation [1].

CONTRIBUTION BY THE FINNISH ADMINISTRATION

The basic plan for the Finnish telephone network is shown in Figure 1. There are 79 secondary areas (numbering areas) which together form 9 tertiary areas. The inter-tertiary area net-

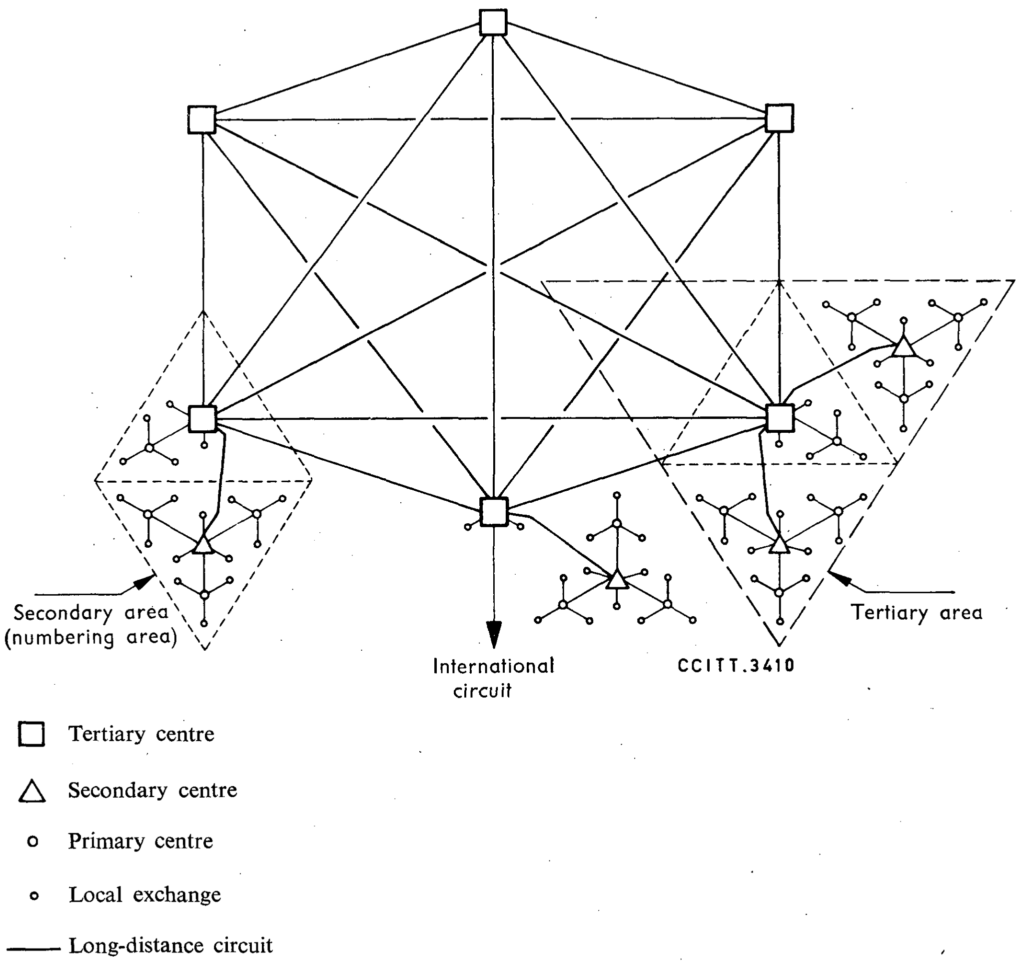


FIGURE 1. — The basic plan for the Finnish telephone network

[1] *Note by the Secretariat.*— In view of paragraphs 2.1 and 2.3, and as can be seen from Figure 1, there remain 7 dB for the terminal reference equivalent, which is the mean of the reference equivalents (the values of which have been supplied elsewhere) for the whole of the subscriber station, the subscriber line and the feeding-bridge :

12 dB (send),
2 dB (receive).

work is a star network. In the basic plan all the tertiary centres are interconnected by direct circuits forming thus a mesh network. The national network is connected to the international network by the tertiary centre in the capital of Finland.

If it is not possible to route a call the shortest way to its destination, it can be routed via alternate routes where the network is meshed. The number of circuits interconnected in tandem is limited. In a national call the number is 8 or less and in an international call in the national part 5 or less. There is an intention to try to limit the number in the second case to 4, by limiting the use of alternate routes, and thus meet the aims of the C.C.I.T.T. for an average-size country.

The transmission plan is based on the C.C.I.T.T. plan a) Recommendation P.11, *Red Book*, Volume V *bis*, Figure 2. The maximum value of the nominal total reference equivalent of a national call is 33 dB (38 dNp). All the long-distance circuits, i.e. circuits interconnecting secondary areas, are four-wire circuits and their nominal loss is 0 dB (0 Np) at the frequency of 800 Hz. In a secondary area the maximum value of the nominal sending reference equivalent from a subscriber to the four-wire switching point of a long-distance circuit is 20.8 dB (24 dNp). The corresponding value for the receiving reference equivalent is 12.2 dB (14 dNp). 11.3 dB (13 dNp) of these values has been reserved for the attenuation of exchanges and inter-exchange circuits and the rest for the subscriber's system which consists of the subscriber's instrument, subscriber's line and feeding-bridge. In the basic plan the four-wire chain extends to the primary centre. The nominal value of the attenuation from the switching point of a long-distance circuit to the two-wire side in a primary centre shall be at least 5.2 dB (6 dNp) and shall not exceed 7 dB (8 dNp).

There are two reasons for the minimum value being greater than the nominal attenuation of a hybrid. First, in an international call the limit of numbers of four-wire circuits interconnected in tandem in the national extension is four at present. Secondly, it should be possible to use in the secondary area carrier systems with greater variations in loss with time than specified by the C.C.I.T.T.

The maximum value of the nominal sending reference equivalent of the subscriber system of a local exchange is 9.5 dB (11 dNp) and the corresponding receiving reference equivalent 0.9 dB (1 dNp).

The Finnish telephone network will not be built according to the basic plan in every respect. The mesh-network between tertiary centres will not be complete and direct circuits will often be added to the star-network. The deviations from the basic plan are made when they are justified by technical, economical and traffic viewpoints. In a secondary area the four-wire chain can also reach the local exchange or be limited to the secondary centre.

These departures from the basic plan will cause changes in the transmission plan in that part of the network. Direct circuits that will not carry traffic to the long-distance network can be dimensioned less strictly. The maximum nominal reference equivalent value of 33 dB (38 dNp) in a complete call will be followed, though.

When the four-wire chain exceptionally extends through the primary centre to the local exchange (Figure 3), the nominal attenuation from the switching point of a secondary centre to the two-wire side of the local exchange shall be at least 6 dB (7 dNp) and shall not exceed 7 dB (8 dNp). When the four-wire chain ends in the secondary centre (Figure 4) the permissible nominal attenuation from the four-wire switching point of the secondary centre to the local exchange is the previously mentioned value of 11.3 dB (13 dNp).

There is an intention to employ throughout the network telephone sets with equalizers to reduce transmitting and receiving efficiency in short loops. The transmission characteristics of such a set are determined by connecting the set to a subscriber cable of 0.5 mm and 37 nF/km

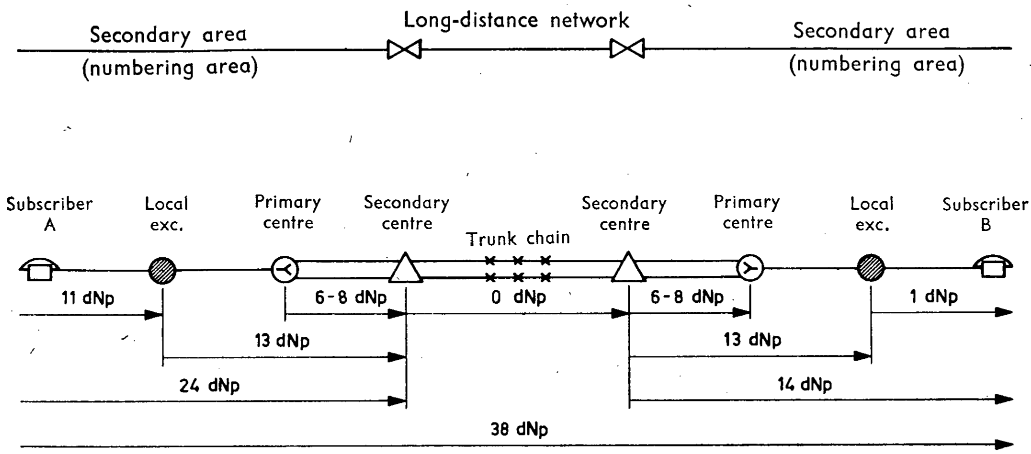


FIGURE 2. — **(Finland)**

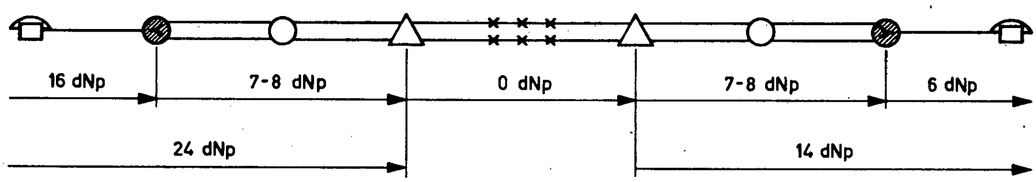


FIGURE 3. — **(Finland)**

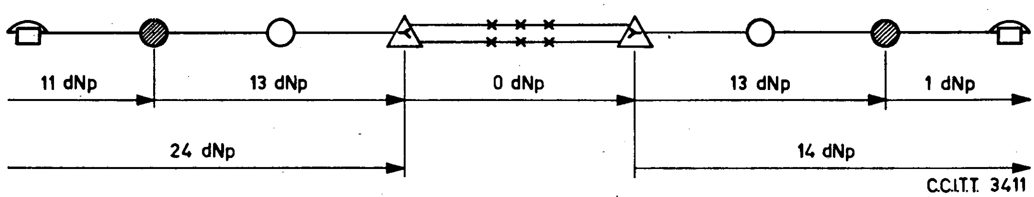


FIGURE 4. — **(Finland)**

and to a feeding-bridge of 60 V and 2 × 500 ohms in such a way that the mean value of a suitably large sample of a consignment must fulfil the specifications in Figure 5. The length of the cable is 0 to 5 km. The standard deviation shall not exceed at any length the value 1.3 dB (1.5 dNp) at sending and the value 0.9 dB (1 dNp) at receiving. The measurements are made with an objective reference equivalent measuring apparatus (O.B.D.M.) The apparatus is calibrated with N.O.S.F.E.R. by means of the national working standard.

As can be noted there is reserved for the subscriber system of every exchange a sending reference equivalent value of 9.5 dB (11 dNp) and a receiving reference equivalent value of 0.9 dB (1 dNp). In certain exchanges the corresponding values are 14 dB (16 dNp) and 5.2 dB (6 dNp). In both cases there has been specified the maximum allowable lengths of different types of subscribers' lines, by measurements with an O.B.D.M. apparatus.

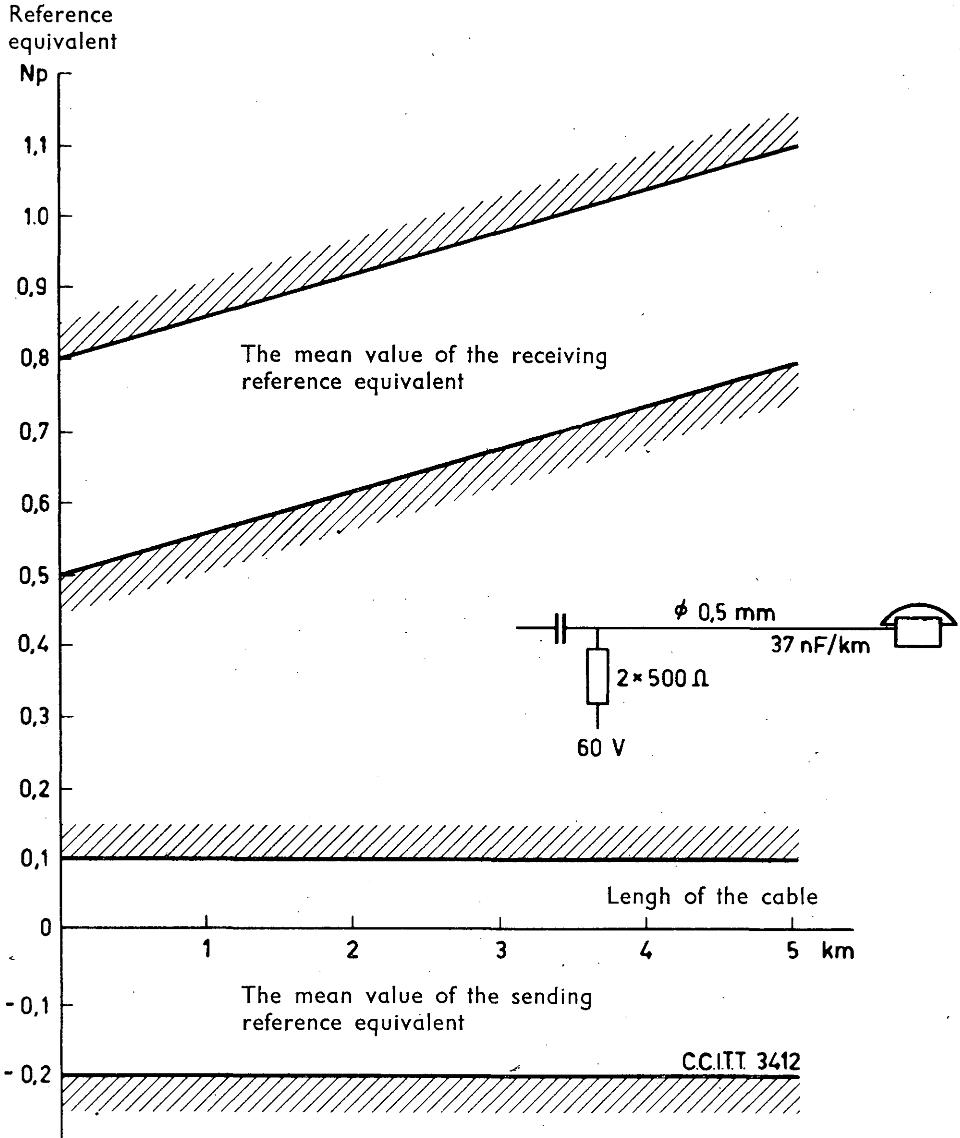


FIGURE 5. — **The specified mean values of the sending and receiving reference equivalents of a sample of telephone sets (Finland)**

CONTRIBUTION BY THE FRENCH ADMINISTRATION

1. *General*

The new transmission plan adopted by the French Administration takes account of the C.C.I.T.T. recommendations concerning the international telephone service, and particularly of Recommendation G.111 (or P.11), which established practical limits for the nominal reference equivalents of national sending and receiving systems.

As regards national communications, the following rules have been accepted :

1) The overall reference equivalent should be less than 4.15 Np for 90% of calls between any two subscribers in the network;

2) The overall reference equivalent should exceed 4.60 Np in quite exceptional cases;

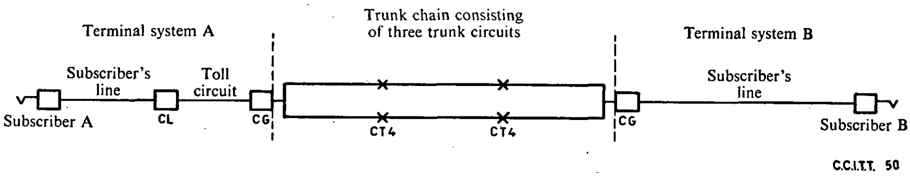
3) The nominal values of the equivalents at 800 Hz of the various circuits comprising the four-wire chain shall be selected in such a way as to ensure the stability of the chain for a negative variation of the overall loss equal to three times the standard deviation.

By definition :

the *terminal system* is the unit formed by the subscriber's set, the subscriber's line, the local exchange(s), the toll circuit(s) and primary centre; the *trunk chain* is the unit constituted by the trunk circuits and the transit exchanges situated between the two primary centres.

Figure 1 shows, as an example, a diagram of a trunk call broken down into its three component parts.

The new plan gives separate definitions for nominal reference equivalents of the trunk circuit and the terminal system (sending and receiving).



CL = local exchange
CG = primary centre
CT4 = transit centre with four-wire switching

FIGURE 1. — **(France)**

2. *Reference equivalent of the trunk chain*

The reference equivalent of the trunk chain was assumed to be equal to the sum of the equivalents at 800 Hz of the various circuits, possibly increased by the losses due to passage through transit centres equipped with two-wire switching (CT2).

The following values were taken :

1. The nominal attenuation of a chain of interconnected four-wire circuits, as measured between two-wire input and output points, is equal to:

$$4 + 0.5\ n \text{ dNp}$$

n being the number of four-wire trunk circuits in the chain (in practice n is between 1 and 6).

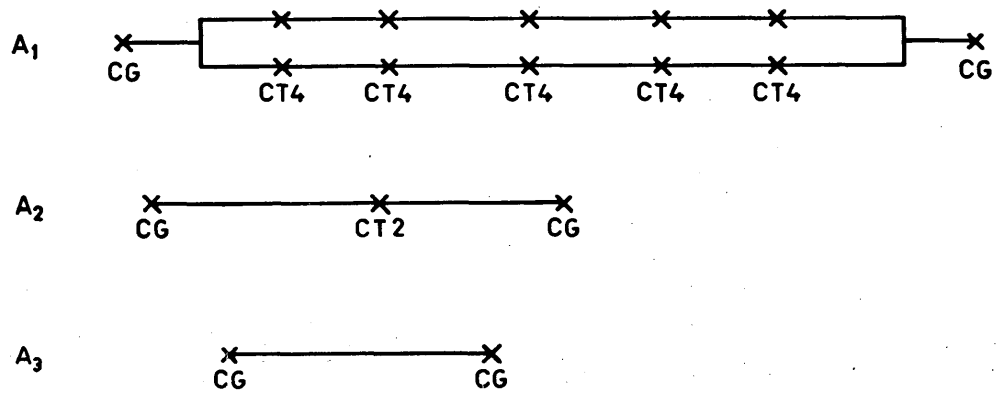
2. The trunk circuits switched on a two-wire basis at both ends are adjusted to 3 dNp (four-wire circuits) or 5 dNp (two-wire circuits).
3. The equivalent at 800 Hz of a circuit interconnecting two primary centres which do not provide transit facilities should not exceed 1 neper.
4. The insertion loss of a CT2 is assumed to be equal to 15 cNp.

The various types of trunk chains being planned are shown in Figure 2. When four-wire switching will have been installed in all the regional transit centres and in the busiest departmental transit centres, only cases A_1, A_2 and A_3 will have to be taken into account. During the interim period, cases A_4, A_5 and A_6 will also be taken into account.

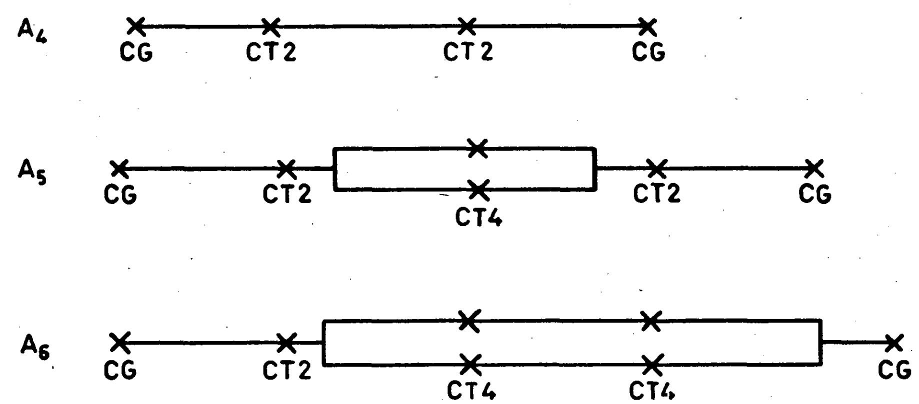
The standard deviation of the variations in the equivalent of a circuit is assumed to be 12 cNp for circuits on primary groups with automatic regulation, and 17 cNp for other circuits.

Longest four-wire chain consisting of n = six circuits

Final phase



Interim phase



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CG = group centre
CT2 = transit centre with two-wire switching
CT4 = transit centre with four-wire switching

FIGURE 2. — **Trunk chains (France)**

3. *Sending and receiving reference equivalents of the terminal system*

The terminal system reference equivalent is obtained by adding the equivalent at 800 Hz of the toll circuit(s), the loss through centres (other than the centre serving the subscriber's installation) and the reference equivalent of the subscriber's set.

Planning rules for local networks tend to establish the nominal maximum reference equivalent of the terminal system at 1.6 Np for the sending end, and at 0.6 Np for the receiving end. This value will be attained only for the least-favoured subscribers in the group.

Further, the possible variations of the reference equivalent of the terminal system (sending or receiving) have been estimated at ±2 dNp about its nominal value, due :

— to dispersion and variations with time of the sensitivity of the microphone and receiver insets, and

— to fluctuations in battery voltage.

Where appropriate some account should be taken of variations in the equivalent at 800 Hz of repeatered toll circuits.

4. *Overall results*

To verify that the rule of 4.15 Np applies to 90% of calls, it appears necessary to study a statistical model of the network. The basic data are being collected and detailed information will be supplied later.

It is comparatively simple, on the other hand, to verify that in almost all possible calls the reference equivalent is less than 4.6 Np.

For each of the cases previously mentioned, Table 1 shows maximum estimates for the reference equivalent of calls. The assumptions on which this calculation is based are :

a) That each of the two subscribers is one of the least-favoured subscribers in his group (reference equivalent of the terminal system 1.6 Np sending and 0.6 Np for receiving).

TABLE 1

Type of trunk chain	Reference equivalent (in Np)					
	Nominal of trunk chain	Nominal maximum of terminal system		Nominal total	Total (including positive variations)	
		Sending	Receiving		$\sigma = 0.12$ Np	$\sigma = 0.17$ Np
A_1 ($n = 3$)	0.55	1.6	0.6	2.75	3.57	3.75
A_1 ($n = 5$)	0.65	1.6	0.6	2.85	3.79	4.01
A_1 ($n = 6$)	0.70	1.6	0.6	2.90	3.90	4.14
A_2	1.15	1.6	0.6	3.35	4.09	4.23
A_3	1	1.6	0.6	3.20	3.84	3.94
A_4	1.60	1.6	0.6	3.80	4.62	4.80
A_5	1.80	1.6	0.6	4	4.88	5.08
A_6	1.20	1.6	0.6	3.40	4.28	4.48

b) That each of these figures has been increased by 0.2 Np, which represents the maximum variation of the equivalent in the unfavourable sense.

c) That the positive variation in the loss of a chain of n trunk circuits has been estimated at $2\,\sigma\sqrt{n}$ ($\sigma = 0.12$ or 0.17 Np) (with 2% probability that this will be exceeded for a Gaussian distribution of the equivalents about the mean).

Clearly, this procedure for calculating the reference equivalent of a call is very conservative and the probability that it will be exceeded is almost nil.

It can be seen that the condition of 4.60 Np is maintained with a comfortable margin in the case of configurations relating to the final phase of application of the new plan. However, the distribution is occasionally exceeded, in cases A_4 and A_5, which correspond to the very beginning of the interim phase.

Information on the telephone networks of the Hungarian Administration

Last year the Hungarian Administration prepared a new network plan to replace the 1957 plan. In doing so, it made a detailed study of the relevant C.C.I.T.T. statistics. However, it did not prove possible to deduce clear and easily usable instructions from the data on actual service distribution and facilities. Hence, the new attenuation plan contains instructions on the upper limit of the reference equivalent. However, it is expected that the networks projected under the plan will satisfy the statistical prescriptions. Estimates carried out on existing networks indicate that the probability distribution of the reference equivalent for connections is close to normal. The networks based on the new attenuation plan will likewise be checked in due course from the point of view of the statistical prescriptions.

The various planning criteria of the Hungarian telephone network are as follows :

1. *Transmission plan*

Three types of national circuits are distinguished from the point of view of the attenuation plan :

a) The subscriber line goes from the subscriber station to the first public exchange (local exchange) and also includes the feeding-bridge of the exchange ensuring the microphone supply. There is an exception in the case of an extension station connected to an automatic switching unit since in this case the local line connected to the local exchange also forms part of the subscriber line.

b) The chain of four-wire circuits formed by the interconnection of four-wire circuits is the most important part of the national system.

c) The connection between the terminal point of the four-wire chain and the local exchange is normally composed of relatively short two-wire trunk circuits without amplification. The junction circuit in local networks with several exchanges are two-wire circuits.

1.1 *Subscriber network*

The reference equivalent of the subscriber network is given by the total value of the telephone station, the subscriber line and the feeding-bridge to which a supplementary attenuation is added (in the sending direction), which depends on the value of the loop resistance, due to the reduction of the supply current. To make the reference equivalent of subscriber lines of different lengths uniform and to develop the local networks economically, telephone stations have been classified up to the time when stations equipped with automatic level regulating devices of good quality are generally available. Telephone stations are classified as follows :

Category	Reference equivalent	
	Send Np	Receive Np
I	+0.5 to +0.9	−0.2 to +0.2
II	+0.2 to +0.5	−0.5 to −0.2
III	−0.1 to +0.2	−0.8 to −0.5

Note. — The sets of the type formerly used come under category I.

The observed increase in sending reference equivalents due to a reduction in the supply current is 0.3 Np in the case of a loop resistance of 500 ohms, and 0.55 Np for a loop resistance of 1000 ohms.

According to the statistical data for a metropolitan subscriber network, 80% of lines are less than 2 km in length, 95% do not exceed 3.7 km and 99.5% do not exceed 6.2 km. Hence, the local network can be characterized by a mean length of 1.2 km and a standard deviation of 1.5 km. The attenuation of the exchange and the feeding-bridge was shown to be 0.1 Np on the average.

The basic planning values for the subscriber network are a reference equivalent (send) of 1.35 Np and one of 0.35 Np (receive). For extension stations of automatic switching units, an additional reference equivalent of 0.25 and 0.25 Np is admissible for sending and receiving. For subscribers situated at a great distance from the exchange, the reference equivalent limits may be increased in some exceptional cases by a maximum of 0.4 Np.

To respect the reference equivalence limits authorized in the subscriber network and to ensure a Gaussian distribution of attenuations in the network, telephone stations are used as follows :

A category I telephone station is associated with a line having a loop resistance of 0-250 ohms.

A category II telephone station is associated with a line having a loop resistance of 250-500 ohms.

A category III telephone station is associated with a line having a loop resistance of more than 500 ohms.

Note.— For subscriber lines attached to " rotary " exchanges, a loop resistance of a maximum of 750 ohms is admitted (without the set) and in the case of crossbar exchanges a resistance of a maximum of 1000 ohms.

1.2 *Chain of national four-wire circuits*

The attenuation of the chain of national four-wire circuits varies according to the number of four-wire circuits connected in tandem, taking into account the degree of attenuation variations. Some four-wire circuits are adjusted to a residual attenuation of 0 Np. The nominal equivalent of a chain of four-wire circuits with its terminating sets is

0.4 Np for two circuits

0.4 Np for four circuits

0.5 Np for six circuits.

For eight circuits—which is rather exceptional—the prescribed attenuation is 0.6 Np.

1.3 *Two-wire toll circuits*

The permissible reference equivalent for two-wire circuits varies according to whether the complete telephone connection comprises a chain of two, four, six (or exceptionally eight) four-wire circuits.

The reference equivalent for a complete connection between two local exchanges—taking account of the larger attenuation variations which can be observed—must not exceed 2 Np.

If the chain of four-wire circuits is prolonged by two-wire circuits via a four-wire transit point, and by eliminating an attenuation of 0.4 Np, the reference equivalent of the two-wire circuits can be :

0.85 Np in the case of two four-wire circuits,

0.8 Np in the case of four four-wire circuits,

0.75 Np in the case of six four-wire circuits.

1.4 *The reference equivalent of the complete connection*

The reference equivalent of a complete telephone connection between one subscriber and another may be 3.7 Np (maximum) considering the admissible value for the subscriber network, i.e. $1.35 + 0.35 = 1.7$ Np and 2 Np for regional cables in the case of trunk connections. This value for extension stations connected to automatic switching units is higher by $0.25 + 0.25 = 0.5$ Np, i.e. it may be equal to 4.2 Np. However, most connections already set up have a much lower reference equivalent and it is to be expected that the mean reference equivalent of calls set up over the national network will be about 2.4 Np, while 95% of connections already made do not exceed 3.7 Np.

2. *Interconnection plan*

2.1 *Types of exchanges*

The Hungarian national network consists mostly of a star-shaped system.
The following types of exchange exist (decreasing order) :

I. Main exchange (international switching centre) (there is only one such exchange in the country—in the capital)

II. Tertiary exchange (there are 18)

III. Secondary exchange (68)

IV. Primary exchange (130)

V. Terminal exchange (900).

All the exchanges of the higher category can also do the tasks of exchanges in the lower category (those indicated by the larger roman figures).

Switching method

In exchanges I, II and III, four-wire switching is used without exception. In general, the category IV exchanges also ensure four-wire transit but they may also be used exceptionally with two-wire switching. One of the objectives in the establishment of the network is to prolong the chain of four-wire circuits as far as the primary exchanges.

Four-wire switching makes it possible to make use of the advantages deriving from attenuation control, with an unfavourable influence on stability owing to the fact that the attenuators are installed in the four-wire branches.

2.3 *Use of transverse lines*

To make good use of the circuits, call routing may differ from an ideal star-shaped system. Category II exchanges may be connected together, then category III exchanges may also be connected to each other or to the main exchange by a direct first-choice circuit, if this is justified by the traffic volume. Also in the numbering zone it is possible to set up a first-choice circuit between the primary exchanges.

3. *Additional remarks*

In view of the small size of Hungary, it is not necessary to use echo suppressors, nor is it difficult to observe C.C.I.T.T. Recommendations on transmission delay. Interference due to phase distortion is also eliminated by setting up four-wire circuits on carrier systems and limiting the length of two-wire circuits (maximum of 30-50 km).

Two-wire trunk circuits with repeaters are installed only exceptionally and there are few of them, so that the provisions regarding uniform impedance and echo attenuation are respected throughout the network. When the network is planned, the following maximum balance return losses are laid down :

For DM (multiple twin) underground cables : maximum of 2.7 Np

For DM (multiple twin) overhead cables : maximum of 2.0 Np

For local underground cables : maximum of 2.0 Np

For local overhead cables : maximum of 1.5 Np.

These balance return losses enable attenuation control to be applied and, if necessary in exceptional cases, allow for the use of a terminal repeater.

Transmission plan for the automatic networks in Indonesia

For national calls, the attenuation from subscriber to subscriber should not exceed 34.8 dB (4 Np).

In the case of international calls, the nominal reference equivalent between a subscriber and the four-wire terminals of the international circuit should not exceed 20.8 dB (2.4 Np) sending and 12.2 dB (1.4 Np) receiving.

Contribution by the Italian Administration

Introduction

The transmission plan in the Italian telephone network forms part of the " National telephone regulatory plan " (P.R.T.N.) which has been prepared with the aim of giving the Italian network a rational configuration in keeping with the principles of modern technique.

The plan caters for a division of the network into three parts :

a) District network (star layout) comprising lines connected to a district centre (CD);

b) Secondary network (star layout) comprising connecting lines between CDs and " regional centres " (CC);

c) Primary network (mesh layout), interconnecting the CCs.

The country has been divided into 220 districts grouped in 21 regions, of which the CCs are generally the regional capitals. Of these, Rome and Milan, which are also equipped as international transit centres, are called " national centres (CN) ".

Each district is divided into sectors and the sectors into local networks. This arrangement was devised in order to concentrate :

Traffic of an exclusively local interest within the sectors;

Traffic for zones influenced by geographical factors, commercial relations, etc. within districts;

And essentially regional traffic within the regions.

The regional centres constitute nodal sorting centres for medium and long-distance traffic (between the various regions).

Transmission plan

The Italian automatic trunk network is now being established in consideration of the following attenuation plan;

a) The national equivalent (at 800 Hz) between two district centres must not exceed 0.8 Np, including any attenuation in intermediate transit circuits;

b) The nominal equivalent of the circuit from the telephone station to the district centre must not exceed 1.1 Np;

c) The attenuation, at 800 Hz, between automatic exchanges (excluding CCs) should not exceed a total of 0.5 Np;

d) The sum of the sending and receiving reference equivalents of subscribers' stations must not exceed a total of 0.7 Np;

By adding a mean equivalent variation of ± 0.3 Np for the chain of circuits serving two CDs, an attenuation of 4.5 Np is reached, which is less than the C.C.I.T.T. limit of 4.6 Np.

However, it should be observed that the value of 4.5 Np is reached only in a very small percentage of cases, as it is seldom that maximum values in one and the same connection will occur simultaneously.

The circuits in the primary network are all four-wire, chiefly of the carrier type. In the regional centres, the circuits are interconnected by selectors for four-wire circuits; four-wire connection is similarly used for interconnection with circuits of the secondary network or the district network (which may be two- or four-wire, whereas the modern trend is towards the constitution of four-wire carrier circuits). This helps to contribute corresponding improvement in transmission, particularly where it is necessary to use an alternative route; this is the method indicated in Volume VI of the *Red Book*, page 180, paragraph b).

REFERENCES

[1] Piano Regolatore Telefonico Nazionale; *Gazetta Ufficiale*, 30 December 1957, No. 321.

[2] FORNO, A.: Servizio Telefonico in Italia; *National Association of Italian Engineers and Architects*, Milan, 17-20 June 1962.

CONTRIBUTION BY THE NIPPON TELEGRAPH AND TELEPHONE PUBLIC CORPORATION (JAPAN)

1. Principles of the new national transmission plan are based on the new telephone transmission engineering standard which was established on 1 September 1965, in consideration of adopting the new type No. 600 telephone set.

Note 1.— The conventional national transmission plan was established in accordance with the old telephone transmission engineering standard, being based on the old-type No. 4 telephone set. Its outline has already been reported in contribution COM XI-No. 47, period 1961-1964.

The main points of this standard are as follows :

2. The transmission quality of a connection between any two subscribers in our country is evaluated by the A.E.N., but the transmission quality of an international connection from an international exchange to a subscriber in our country is evaluated by the RE (reference equivalent).

3. The objective for the national transmission quality is that 90% of the connections between any two subscribers should be better than the A.E.N. 49 decibels (which is equivalent to 43 decibels of the nominal A.E.N. recommended by the C.C.I.T.T.).

The quality of almost all connections becomes better than the A.E.N. 50 decibels (which is equivalent to 44 decibels of the nominal A.E.N. recommended by the C.C.I.T.T.).

Note 2.— The nominal A.E.N. recommended by the C.C.I.T.T. (*White Book*, Recommendation P.12), does not include the transmission impairment due to loss variation, line noises and frequency band limitation. In Japan, however, it has been considered reasonable from the viewpoint of the A.E.N. conception to include these impairments in treating the A.E.N.

Note 3.— Though the C.C.I.T.T. nominal A.E.N. is applied to the actual call, the A.E.N. is applied to the connection in Japan. This is considered reasonable because of the transmission network design.

The interpretation of "90% of the connections which satisfy the A.E.N. 49 decibels" is as follows : a subscriber having 90% subscriber's line loss in any local exchange area satisfies the A.E.N. 49 decibels if he is connected with a subscriber having 90% subscriber's line loss in any other local exchange area.

4. The objective for international transmission quality is that the R.E. between an international exchange and a subscriber for more than 97% of all international calls should be better than the R.E. of C.C.I.T.T. Recommendation P.11.

5. The total of transmission impairments, as mentioned in Note 2, is determined so as to satisfy the A.E.N. 49 decibels as shown in Table 1. Corresponding to these A.E.N. values, the design limits of the transmission performances are determined as shown in Figure 1, for the typical longest connection, which is called the national standard telephone connection.

6. Figure 2 shows the permissible transmission loss of each line or circuit for the national standard telephone connection.

TABLE 1

A.E.N. expressing national transmission performances (in dB)

	Almost all connections		90 % of the connections	
Factor of transmission performance	A.E.N.	C.C.I.T.T. nominal A.E.N.	A.E.N.	C.C.I.T.T. nominal A.E.N.
Subscriber's sending part	18.3	18.3	16.6	16.6
Subscriber's receiving part	8.5	6.5	8.5	6.5
Transmission loss of trunk	17.0	17.0	17.0	17.0
Variation of transmission loss and handset sensitivity	5.7	2.0	5.7	2.0
Effective transmission frequency band	0.5	—	0.5	—
Total ...	50.0	43.8<44	48.3<49	42.1<43

Note.— The A.E.N. values given in Table 1 for the local system are valid in the case of a subscriber's line with 0.4-mm conductors in unloaded cable, having an attenuation of 7 dB (at 1.5 kHz).

The transmission quality of the type 600 telephone set has been improved recently, resulting in a total A.E.N. of 24.8 dB for the local sending and receiving system, and a reference equivalent of 7.2 dB in the same line conditions as above. This represents improvements of about 2 dB and 7 dB respectively compared with the situation given in Table 1.

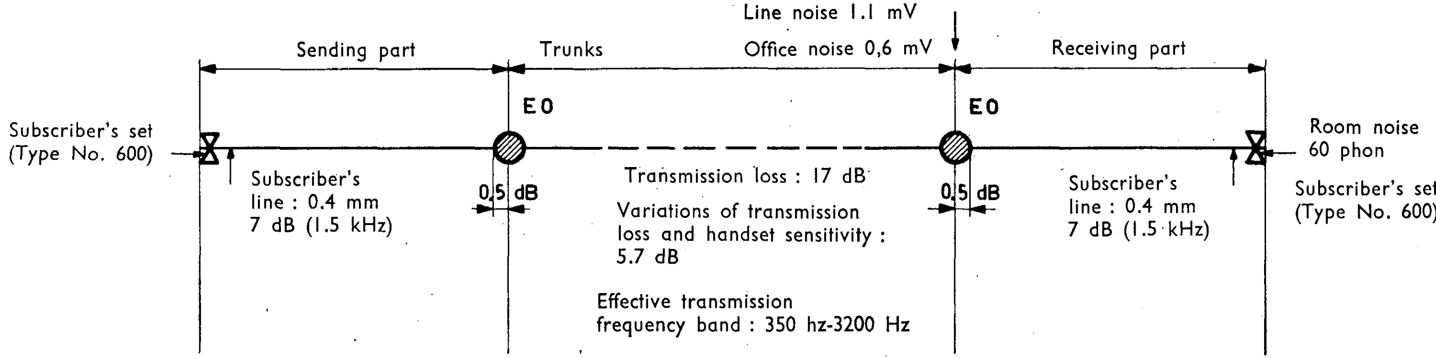
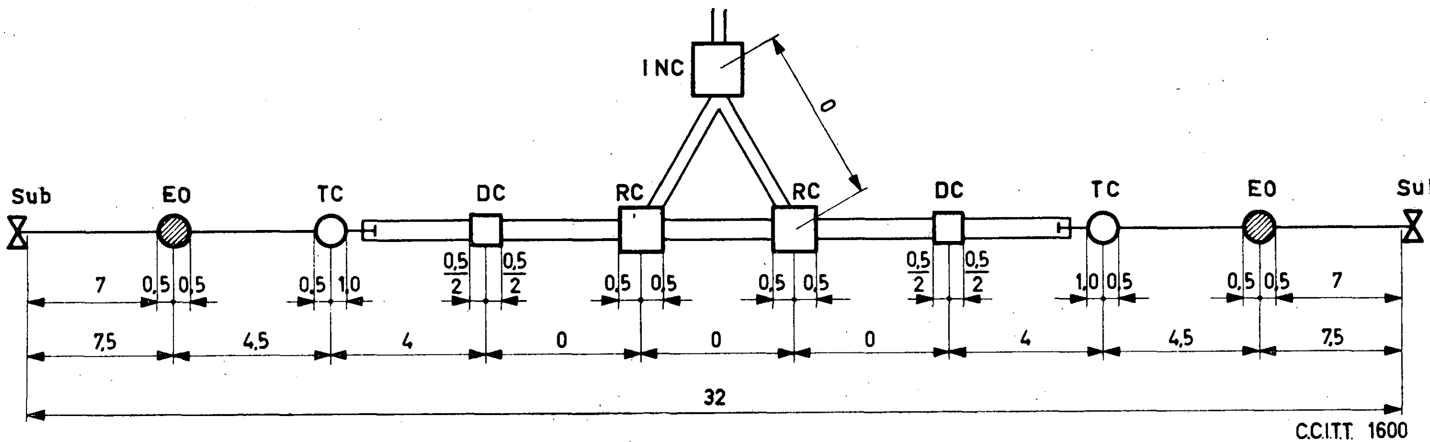


FIGURE 1. — **Design limits of transmission performances for the national standard telephone connection (Japan)**



Notes. — INC = international centre
RC = regional centre
DC = district centre
TC = toll centre
EO = end office
Sub = subscriber

FIGURE 2. — **Transmission losses of the national standard telephone connection (Japan)**

Malaysian national transmission plan

1. *General*

The national transmission plan adopted by the Malaysian Administration takes account of the relevant Recommendation of the C.C.I.T.T. and C.C.I.R. in order to arrive at an economical national telephone system.

2. *Matters studied*

A telephone transmission system, in its most general form, comprises the telephone transmitter, the links from transmitter to the terminal exchange, from terminal exchange to group exchange, from group exchange to zone exchange, from zone exchange to main exchange, from main exchange to international exchange and thence down to the distant end telephone receiver in the reverse order.

Figure 1 illustrates this :

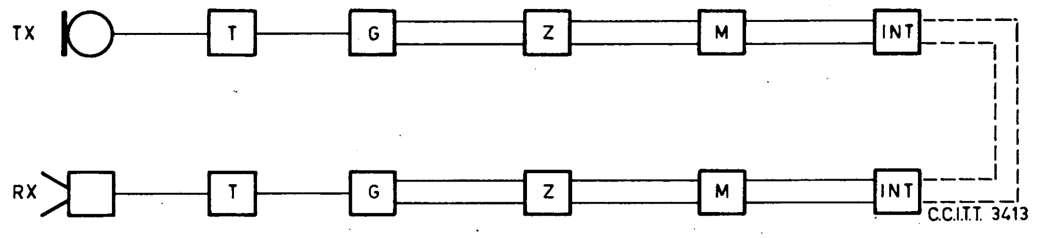


FIGURE 1. — (Malaysia)

In this study, each of the above links is considered in relation to the others and limits of transmission losses and noise are allocated to it.

3. *Reference equivalent (R.E.)*

The loss suffered in the transmission of telephone signals is known as the reference equivalent (R.E.). Reference equivalents are measured under certain C.C.I.T.T. Laboratory conditions and include the performances of the telephone transmitter and receiver. Under C.C.I.T.T. Recommendation G.111 (Geneva, 1964) the total nominal reference equivalent between two subscribers is not strictly limited; its maximum value results from the sum of the reference equivalents of the national and international circuits.

3.1 *Reference equivalent for international chain*

The nominal loss between the virtual switching points (see footnote 1) of each international link should be 0.5 dB (Recommendation G.111, Geneva, 1964).

[1] A virtual switching point in a four-wire circuit is a theoretical point at which the " GO " level is −3.5 dBr and the " RETURN " level is −4.0 dBr.

3.2 *Nominal reference equivalent of national system*

The C.C.I.T.T. has recommended (Recommendation G.111, Geneva, 1964) that the national sending and receiving reference equivalents for 95% of the calls should individually comply with the following conditions :

1) The nominal reference equivalent of the sending system between the subscriber and the first international link should not exceed 20.8 dB.

2) The nominal reference equivalent of the receiving system between the same two points should not exceed 12.2 dB.

3.3 *Budgeting of reference equivalents in the Malaysian national transmission plan*

Under the Malaysian national switching plan the switched network would be essentially as shown in Figure 1. In order to allocate the sending and receiving reference equivalents to the various links, so that 95% of the calls would meet the C.C.I.T.T. conditions given in (3.2), it is necessary to pre-determine the achievable stability of the transmission level in each four-wire link in terms of standard deviations. Bearing in mind the economic and practical aspects involved, it is recommended that the Malaysian transmission system should conform to the following stabilities :

Four-wire link	*Stability*
1) Group to zone exchange	1 dB std. deviation
2) Zone to main exchange (or to another zone)	0.5 dB std. deviation
3) Main to international exchange (or to another main)	0.5 dB std. deviation

Under C.C.I.T.T. Recommendation P.11, *Red Book*, Volume V *bis*, paragraph B e, page 13, the nominal reference equivalent given for national systems includes systematic difference between the performance of the subscriber's set at the sending and receiving ends and their nominal values. Hence, the overall standard deviation between the telephone instrument and the international exchange is therefore :

$$S = \sqrt{1^2 + 0.5^2 + 0.5^2} = 1.22 \text{ dB}$$

In order that 95% of the calls will meet the C.C.I.T.T. conditions of (3.2), the design sending and receiving reference equivalents (S.R.E. and R.R.E.) should therefore be as follows :

1) Design S.R.E. = 20.8 dB − 1.64 × 1.22 dB = 18.8 dB

2) Design R.R.E. = 12.2 dB − 1.64 × 1.22 dB = 10.2 dB

The multiplying factor 1.64 derives from the fact that in the normal distribution, 95% of the area under the curve falls between $-\infty$ and $+ 1.64$ standard deviation from the mean.

It is recommended that the design sending reference equivalent (S.R.E.) and receiving reference equivalent (R.R.E.) should not be worse than the values shown in Table 1.

TABLE 1

Components of transmission system	S.R.E. dB	International circuit R.E. dB	R.R.E. dB	Standard deviation dB
a) Telephone instrument ..	+2.4		−6.2	
b) Subscribers line	+8.0		+8.0	
c) Terminal to group ...	+5.0		+5.0	
d) Group to zone	+1.0		+1.0	1.0
e) Zone to main (or to another zone)	+0.5		+0.5	0.5
f) Main to international (or to another main) .	+0.5		+0.5	0.5
g) Switching loss (total) ..	+1.4		+1.4	
h) International (*n* links) .		0.5 *n*		
Total ...	+18.8 dB	+0.5 *n* dB	+10.2 dB	+1.22 dB

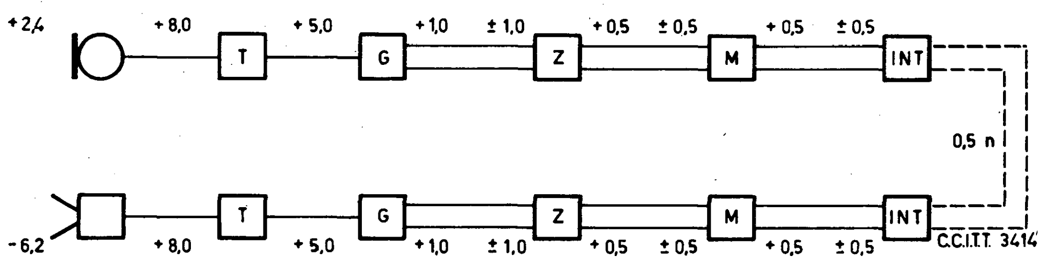
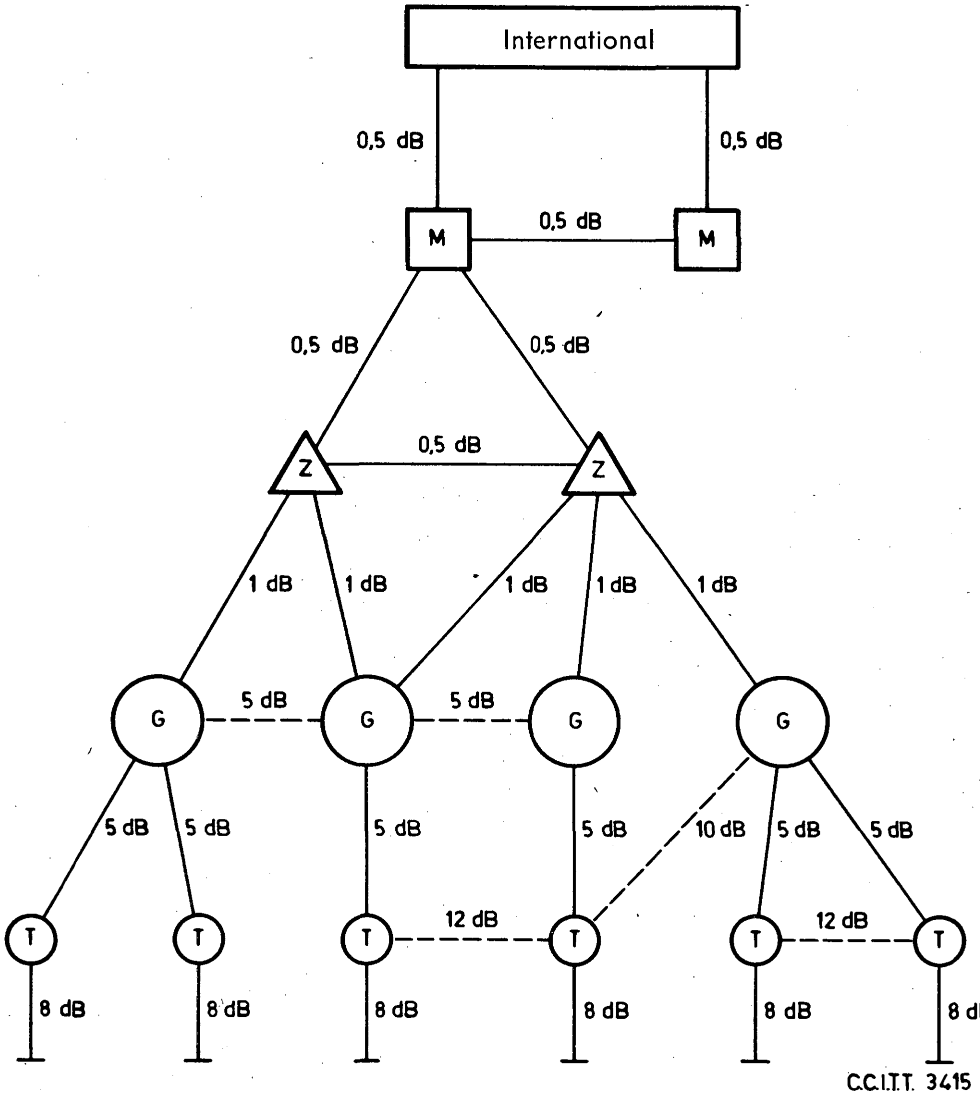


FIGURE 2. — (Malaysia)

The budgeting of the reference equivalent in Table 1 is based on the following considerations :

1) Transmission in the four-wire circuits can be controlled within close limits, with the use of pilot-regulating equipment if necessary. Hence a minimum loss is allocated to these circuits.

2) Transmission loss in the two-wire circuit between terminal and group exchange can be controlled to the maximum limit of + 5.0 dB by the use of two-wire negative impedance amplifiers if necessary. The possibility of utilizing pad switching techniques to introduce compensating gain in high-loss two-wire circuits have been carefully considered, but is not recommended because of the complexity of the switching equipment involved.



Notes.— Dotted lines are high-usage routes and will be used only as follows :

1) From group to group, group to terminal, or terminal to terminal exchanges where such high-usage routes exist;

2) No two high-usage routes will be used in tandem;

3) Group to group exchange high-usage routes will be used as second choice routes for traffic between terminal exchanges having their own high-usage routes.

FIGURE 3. — **Allocation of maximum permissible R.E. in Malaysian national transmission plan**

3) Telephone instruments with sensitivities no worse than those specified in Table 1 are now readily available in the market.

4) The maximum possible amount of loss is allocated to the subscribers' lines. Line plants are expensive assets. A large permissible loss means that smaller size conductors may be used or alternatively for the same size conductors longer lines can be tolerated.

3.4 *Reference equivalents in " high usage " circuits from group to group and terminal to terminal exchanges*

In the Malaysian national switching plan, it is not necessary always to go up the hierarchy of exchanges and down again to reach the required terminal or group exchange. Direct (high usage) group to group, group to terminal and terminal links can be established. When such links are used, it is usually because there is a strong community interest between the exchange areas. This being the case, it is considered desirable to design the high-usage links with performance better than that which the normal reference equivalent limit would allow. It is therefore recommended that the reference equivalent for high-usage circuits from group to group exchanges should not exceed 5 dB, from group to terminal exchanges it should not exceed 10 dB and from terminal to terminal exchanges it should not exceed 12 dB. Figure 3 gives some examples of such high-usage circuits.

These " high usage " circuits should not be used to carry transit traffic destined for an exchange of higher rank.

3.5 *Frequency response of trunk and junction circuits*

Junction circuits should be loaded such that the frequency attenuation characteristics are reasonably flat. Carrier-derived circuits in modern systems usually have a reasonably uniform frequency attenuation characteristic and no special precaution is necessary.

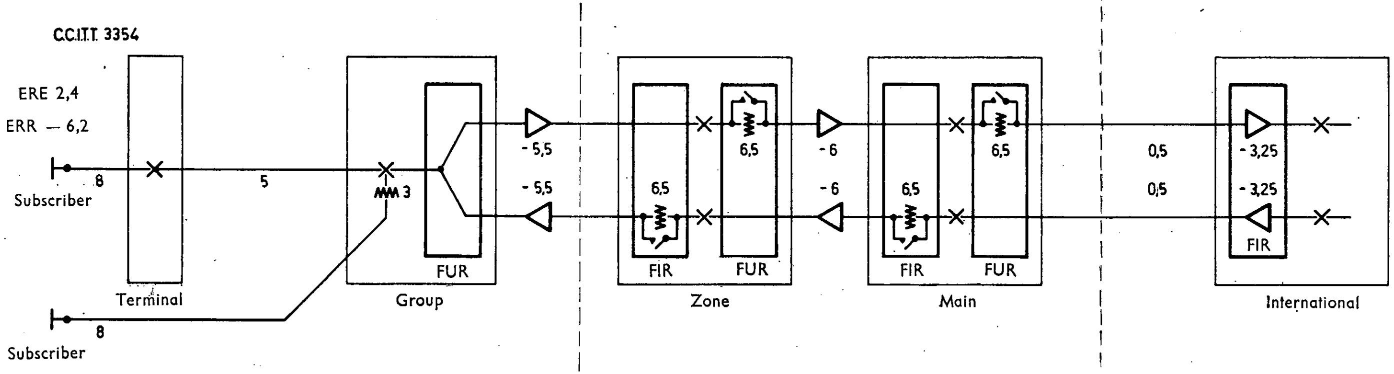
The reference frequency at which loss measurements are made shall be 1000 Hz.

4. *Pad switching and balance return loss at hybrids*

In the Malaysian national switching plan, four-wire pad switching facilities are included. Pads of 6.5 dB loss are located at the input to the modulator of a four-wire transmission link. The active part of the link has a gain of 6 dB (i.e. input at −3 dBm, output at +3 dBm), so that the overall loss in a four-wire transmission circuit with the pads switched in is 0.5 dB. If a four-wire circuit terminates at a hybrid, then, depending on whether the two-wire circuit is a junction circuit or a subscriber's line, the following pad switching conditions apply (see Figure 4).

a) *Junction circuit*

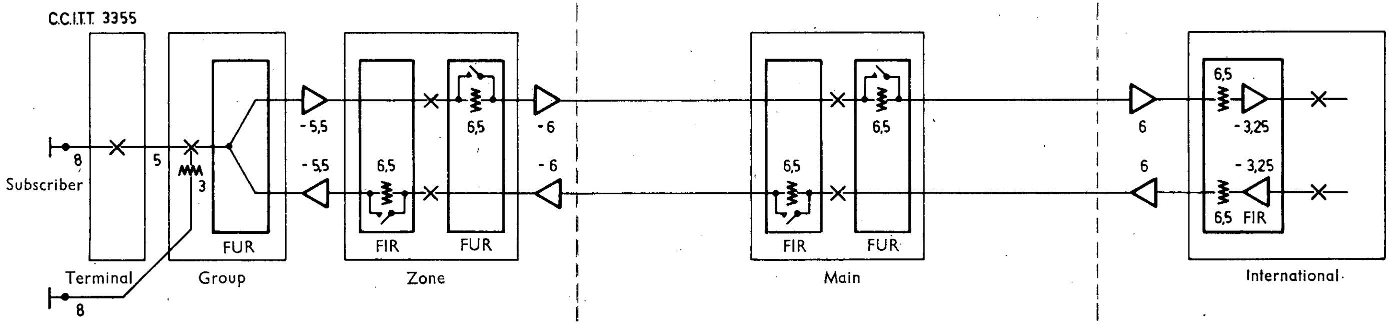
In a connection between two subscribers via a number of 4-wire active links, there will be always one amplifier in excess of pads which will exactly compensate the transmission attenuation of both hybrids. When a subscriber is connected to the international exchange, a 3.25-dB gain will be available at the international exchange to compensate for the transmission attenuation of the single hybrid. In order to meet stability and echo conditions, all junction circuits should have a minimum loss of 3 dB. Pads should be inserted in the junctions to achieve this if necessary. Further, junction circuits should have a loss not greater than 5 dB in order to meet transmission requirements. (See section 3.3.)



Note. — All terminal to group junctions must have a minimum loss of 3 dB.

Note. — The 3.25 dB amp can be adjusted in the carrier equipment by not compensating for the hybrid loss.

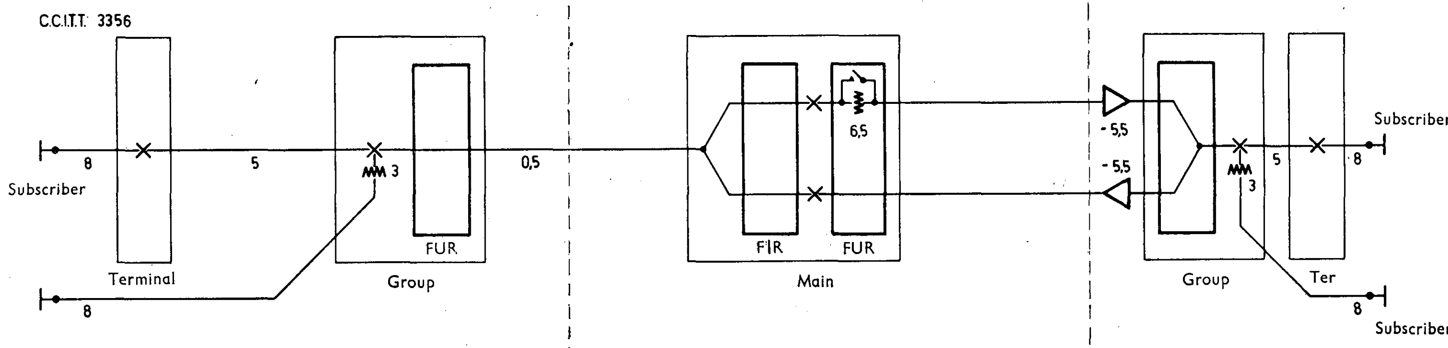
FIGURE 4. — **Pad switching to meet the Malaysian transmission plan**



Note. — All terminal to group junctions must have a minimum loss of 3 dB.

Note. — In the case of Jesselton and Kuching main exchange the four-wire line up level will have a four-wire to four-wire loss of 0.5 dB. This will be done by passive pad on the four-wire side.

FIGURE 4 (*cont.*)



Notes. — 1) The city tandem exchange should be treated in all cases transmission wise exactly like a group.

2) All terminal to group junctions must have a minimum loss of 3 dB.

Notes. — 1) The hybrid will have to be well balanced to the 2-wire line.

2) Amplifiers may be placed on the 4-wire side to compensate for losses on the 4-wire junction.

3) It is essential that proper termination be maintained during switching at the group centre to maintain stability.

Note. — All terminal to group junctions must have a minimum loss of 3 dB.

FIGURE 4 *(cont.)*

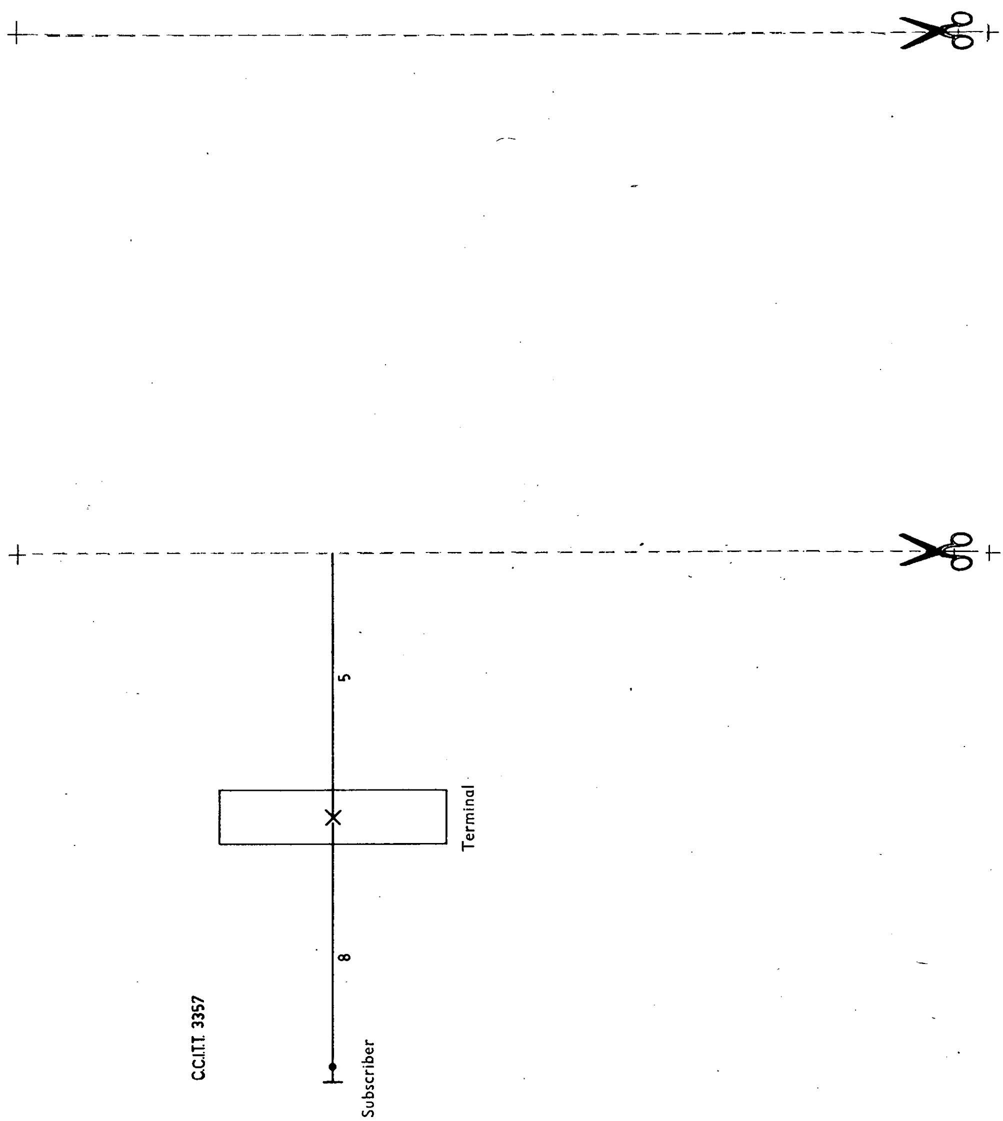


FIGURE 4 (*concl.*)

b) *Subscriber's line*

If the two-wire side is connected directly to a subscriber's line, a 3-dB pad will be inserted in the relay set connected to the subscriber. In other words, the losses between the two-wire and four-wire sides of the hybrid are not compensated for. This loss is required to meet stability and echo conditions.

From a) and b) above, it is clear that the minimum transmission loss between the " RETURN " and " GO " paths of a hydrid is 6 dB. In a practical telephone network, the balance return loss presented by the telephone instruments must be added to the above to arrive at the stability condition of the system. Tests with telephone instruments show that the mean balance return loss exceeds 6 dB when the hand-set is " off-hook ". However, for an incoming call, there will be a time when the circuit is connected but the hand-set is on-hook, i.e. during the time the bell is ringing. Under this condition, and taking into account that the majority of subscribers' line have some basic losses, the balance return loss should not be less than 3.5 dB for practically all cases.

Therefore, the balance return loss at the terminating hybrids in the Malaysian telephone network can be taken to have a mean value of at least 6 dB with standard deviation of not more than 2.5 dB.

5. *Stability*

The Malaysian national chain of an international connection appears, relative to the virtual switching points A and B of the first international circuit, as shown in Figure 5 of C.C.I.T.T. Recommendation G.122, Geneva 1964, the stability criteria that should be met by the national system are as follows :

1) The round-trip loss A-T-B should have a mean value of at least $(10 + n)$ dB.

2) The standard deviation of the round-trip loss A-T-B should not exceed $(6.25 + 4n)^{\frac{1}{2}}$ dB,

where n is the number of four-wire links in the national chain (e.g. $n = 3$ in Figure 5).

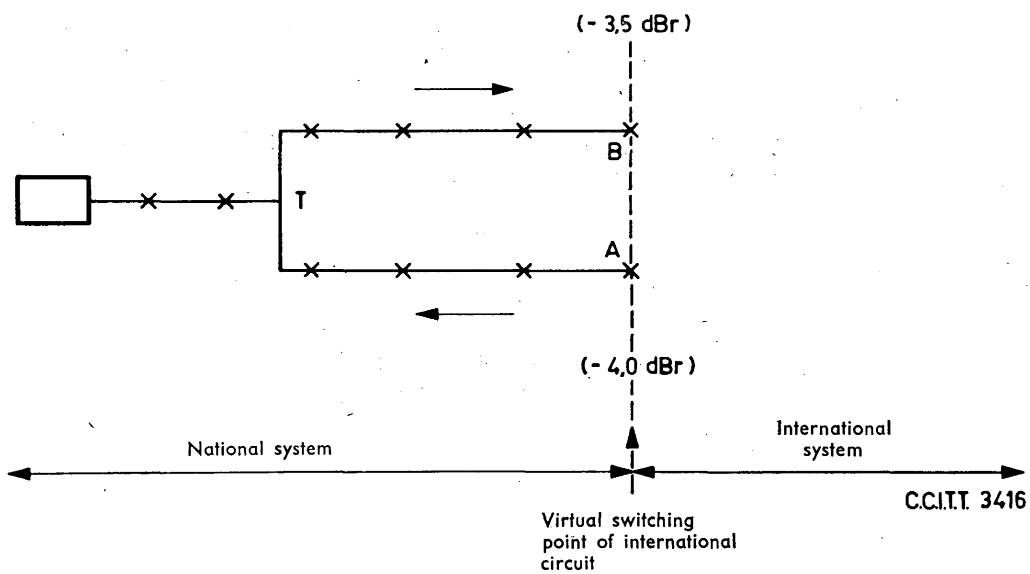


FIGURE 5. — (Malaysia)

It is now shown below that these criteria can be met :

Considering first criterion 1) above, the n dB minimum mean round-trip loss for n links is fully satisfied as each four-wire link has a minimum one-way loss of 0.5 dB. The 10-dB minimum mean loss can be exceeded at the hybrid as follows :

a) Trans-hybrid loss or junction circuit round-trip loss (section 4) .	6.0 dB minimum
b) Mean balance return loss	6.0 dB minimum
	12.0 dB

Referring to criterion a), as any four-wire link in the Malaysian national transmission plan has a standard deviation equal to or less than 1 dB, the sum of the variances for n links, even assuming unity correlation between the variations in the " GO " and " RETURN " paths, should therefore be less than 4 n. Further, as the standard deviation of the balance return loss at the hybrids does not exceed 2.5 dB, the variance therefore does not exceed $(2.5)^2 = 6.25$. Hence the standard deviation of the round-trip loss A-T-B should not exceed $(6.25 + 4\,n)^{\frac{1}{2}}$ dB.

6. *Echoes*

Echo arises from the reflection of voice energy from the distant end, and is always present in a practical telephone system. However, the value of the mean talker echo attenuation above which echo is not objectionable is a function of the round-trip propagation time delay—the longer the time delay, the greater is the echo attenuation required. If the echo attenuation cannot be made large enough, then echo suppressors should be fitted.

6.1 *Malaysian national network*

The longest possible Malaysian national connection would be from Kota Bahru to some remote part of Sabah or Central Sarawak, involving approximately 950 miles of " high " velocity microwave system and 1000 miles of " low " velocity SEACOM submarine cable. The one-way propagation time is given as follows :

a) 950 miles of microwave system at 190 miles/ms	5 ms
b) 1000 miles of cable system at 100 miles/ms	10 ms
Total ...	15 ms

From Figure 11 of C.C.I.T.T. Recommendation G.131, Geneva 1964, the mean talker echo attenuation for which the probability that 95% of the calls will not meet with objectionable echo, is given as 27 dB. In the Malaysian network, the worst possible connection from the echo point of view is one in which :

a) The subscriber is connected by a short line to a terminal exchange;

b) The loss in the junction circuit from the terminal exchange to the group exchange is 3 dB;

c) The four-wire connection is the longest possible, involving 15 milliseconds one-way delay.

Under the above circumstances and with reference to the Malaysian national transmission plan (see Figure 2), the mean talker echo attenuation is given as follows :

a) *" go " and " return " direction losses*

1) Junction circuit $+ 2 \times 3.00$ dB	=	$+6.0$ dB
2) Near end hybrid $+ 2 \times 3.25$ dB	=	$+6.5$ dB
3) Four-wire circuit gain 6.5 dB	=	-6.5 dB
4) Group to group four-wire transmission paths, $2 \times (1.0 + 0.5 + 0.5 + 0.5 + 0.5 + 0.5 + 1.0)$ dB	=	$+9.0$ dB

b) *Distant end hybrid*

1) Trans-hybrid loss	$+6.5$ dB
2) Balance return loss	$+6.0$ dB
Total talker echo attenuation ...	$+27.5$ dB

It is therefore concluded that in the Malaysian national network 95% of the calls connected in the longest possible national circuit will not suffer from objectionable echoes. It is therefore recommended that no echo suppressors will ever be required for intra-Malaysia transmission. Also no echo suppressors will be required for Singapore–Malaysia and Brunei–Malaysia traffic.

6.2 *International circuits*

While in section 6.1 it is shown that echo attenuation is adequate for intra-Malaysia calls, the margin of safety is quite small. It therefore follows that, as a rule, overseas circuits from the Kuala Lumpur Gateway exchange will require echo suppression. Exception to this might be Thailand–Malaysia communications via the proposed Bangkok–Penang–Kuala Lumpur microwave system, and Kuala Lumpur–Medan–Djakarta communications via the proposed VHF/microwave system across Sumatra. These cases will have to be considered on an individual basis.

7. *Microwave systems*

In the Malaysian transmission system, microwave equipment will be extensively used. Microwave systems may be classified into two categories as follows :

1) Main trunk systems, which link zone centres to zone centres, zone centres to main centres, and main centres to international exchange, or to other main centres. The capacity is usually 600 or more channels.

2) Spur systems which link group centres to zone centres and terminal centres to group or zone centres. The capacity is usually below 600 channels.

Microwave systems should be so designed that under median propagation conditions, the following apply :

a) noise arising from R.F. and I.F. interference and crosstalk is at most of the same order as R.F. and I.F. thermal noise;

b) the total telephone channel noise with white noise loading as in C.C.I.R. Recommendation 393, Geneva 1963, should be predominantly intermodulation noise. This is to ensure that, under moderate fading conditions (say 10 to 20 dB) the system performance is not degraded to any great extent;

c) for main trunk systems the total psophometrically weighted noise at a point of zero relative level of the worst telephone channel under white noise loading as in b) should not exceed 3 picowatts per kilometre length of the microwave route;

d) for spur systems, the total psophometrically weighted noise at a point of zero relative level of the worst telephone channel under white noise loading as in b) should not exceed 5000 picowatts (−53 dBm) in the overall spur system. The overall spur system may comprise one or more microwave links in tandem; in the latter case the performance of the individual link should be correspondingly improved in order to meet the overall spur system objective.

In order to meet the transmission stability requirements as recommended in section 3 and to reduce the need for group and supergroup pilot-regulating equipment wherever possible, the microwave system baseband stability over each terminal to terminal link should conform to the following :

a) ±0.5 dB peak over a period of one month (for spur systems this may be relaxed to ±1.0 dB);

b) ±0.5 dB peak for ambient temperative variation between 17° C and 45° C;

c) +0.5 dB peak for ±10% a.c. supply voltage variation.

Further, the overall baseband response should be better than ±0.5 dB with respect to mid-band response.

8. *Telephone instruments*

In section 3, it is recommended that telephone instruments should have minimum sensitivities corresponding to a sending reference equivalent (S.R.E.) of +2.4 dB and a receiving reference equivalent (R.R.E.) of −6.2 dB. These values are to be maintained when the instrument is connected at the end of a subscriber's line having a line reference attenuation of 8 dB and loop resistance of 1500 ohms, the reference equivalent values being those of the instrument itself and does not include the 8-dB line reference attenuation. In other words, the S.R.E. and R.R.E. at the telephone exchange under the above circumstances would be + 10.4 dB (2.4 + 8.0 dB) and + 1.8 dB (8.0 − 6.2 dB) respectively.

Telephone instruments may incorporate sensitivity regulating devices, in which case the S.R.E. and R.R.E. at the exchange should not exceed +10.4 dB and +1.8 dB respectively for any length of subscriber's line up to a maximum limit of 8 dB line reference attenuation and 1500 ohms loop resistance.

9. *Subscribers' lines*

Subscribers' lines should have a maximum line reference attenuation of 8 dB and maximum loop resistance of 1500 ohms. For any particular size of line conductor and for lines of different conductor sizes connected in tandem, these two limits may not be reached simultaneously. In these cases, either limit could be the design oriterion, whichever comes first.

CONTRIBUTION BY THE NORWEGIAN ADMINISTRATION

The main network will be a combined star and mesh network with up to five different types of exchanges, see Figure 1. (Tielines are established where profitable.)

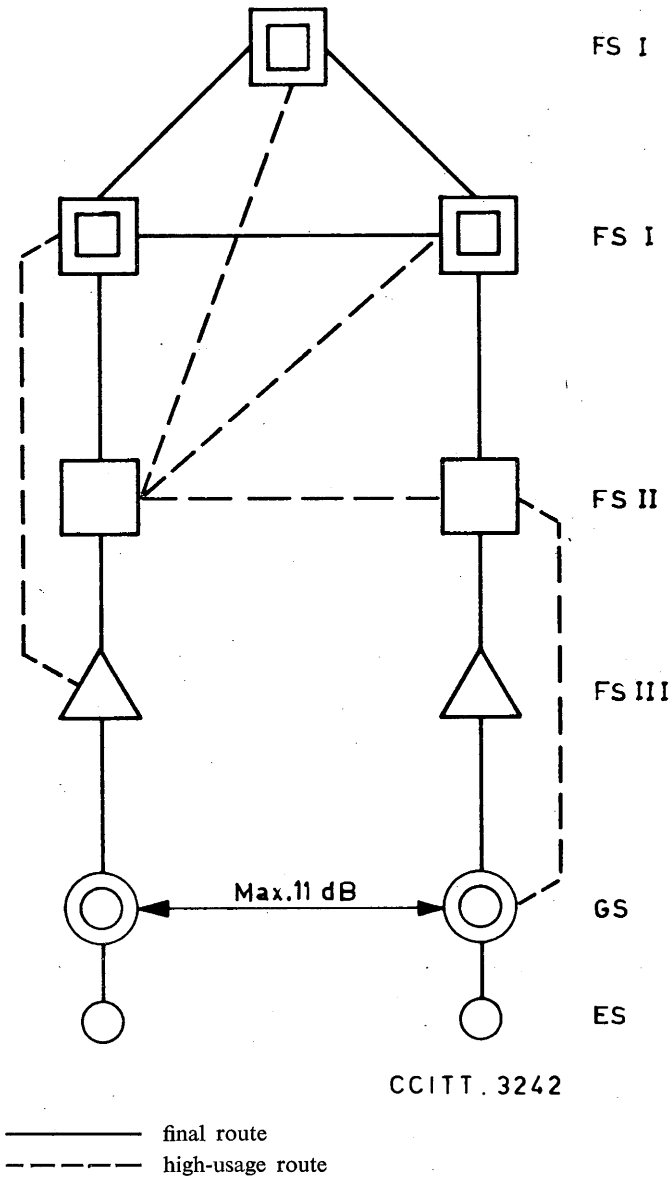


FIGURE 1. — (Norway)

There will be four-wire through switching in the central parts of the network. The network is based on distributed loss, with a fixed loss of 4 dB in the terminations and 1 dB in each circuit.

It is technically and economically an advantage to use overhead plastic-cable instead of open-air lines to subscribers far from the local exchange. For that reason, the future network planning will be based upon a bigger allotment of reference equivalence for the local network than has so far been usual.

This will be achieved by extensive use of four-wire switching exchanges, transistorized repeaters in two-wire trunk lines, loading of long subscriber lines and introducing of transistor-telephones.

CONTRIBUTION FROM THE ADMINISTRATION OF NEW ZEALAND

The New Zealand transmission plan for national calls has been revised and will be implemented progressively over a number of years.

The plan consists of the following elements :

a) Overall transmission which is defined in terms of—

1) a standard telephone and local line; and

2) attenuation in dB of the chain of junctions and trunk circuits connecting the two local exchanges, including switching and mismatch losses.

b) Standard nominal impedances at the two-wire and four-wire switching points.

c) Limits for return loss and stability of the trunk circuits.

d) Limitations on the use of compandors.

The plan is summarized in Figure 1 for a limiting connection through the long-distance trunk network.

Telephone and local circuit

The limiting subscriber's circuit is defined as follows :

Telephone : British 700 type with regulator

Local line : 1000-ohm loop of $6\frac{1}{2}$ lb per mile paper-insulated lccal type cable.

Transmission bridge : 200 + 200 ohms, 2 μF capacitor type, non-ballast.

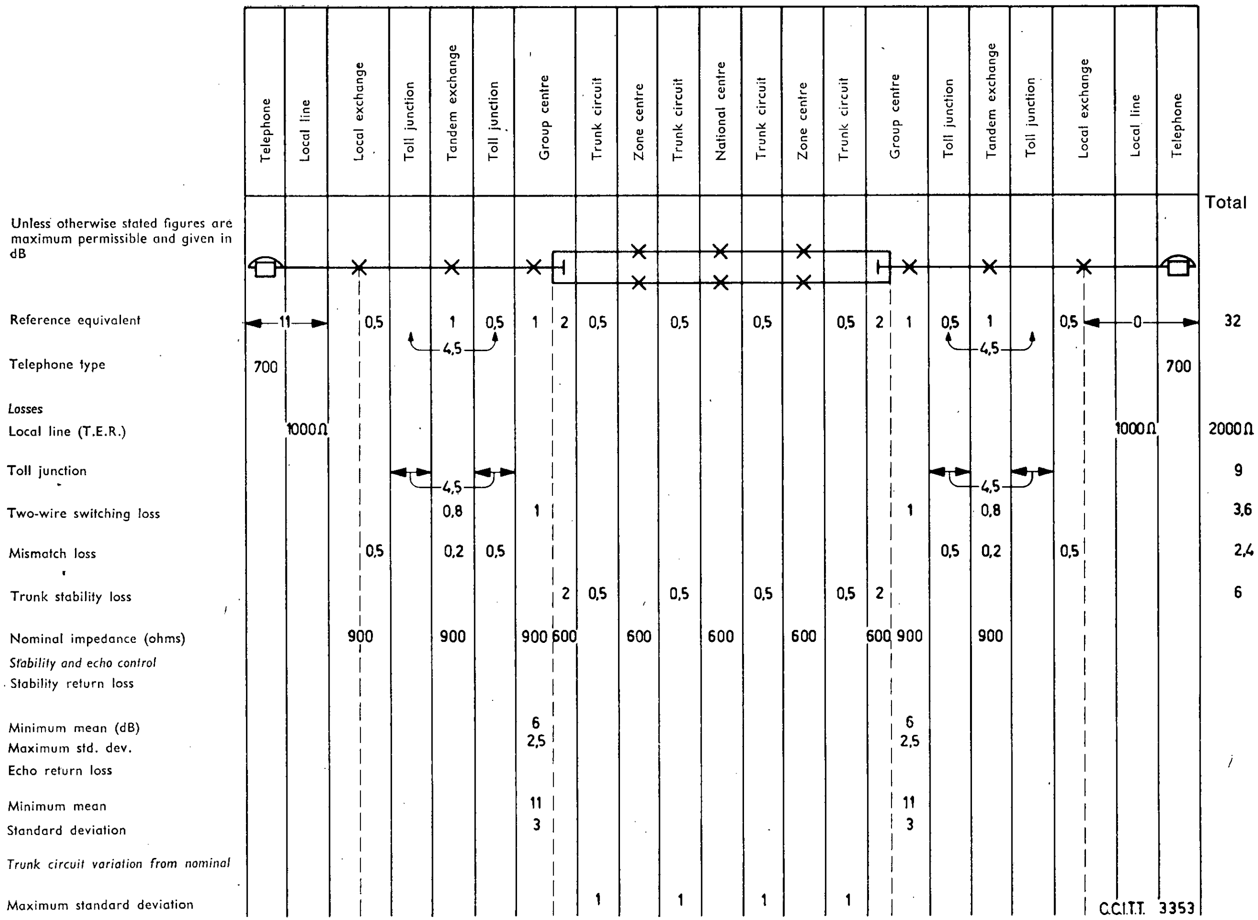
Exchange battery : 50 volts.

In practice, conductor weights from $2\frac{1}{2}$ lbs per mile to 20 lbs per mile are used. Transmission for each conductor weight has been compared subjectively with that for $6\frac{1}{2}$ lb per mile conductors and multiplying factors have been derived to convert the d.c. resistance of the actual cable pair to a resistance of $6\frac{1}{2}$ lb cable having equivalent transmission. The converted figure is known as transmission equivalent resistance (T.E.R.). Typical T.E.R. multiplying factors are given in Table 1.

TABLE 1

Cable conductor (lbs per mile)	mm	*D.c. loop resistance* (ohms per 1000 ft = 305 m)	*T.E.R. multiplying factor*
$2\frac{1}{2}$	0.32	142.8	0.65
4	0.40	83.3	0.8
$6\frac{1}{2}$	0.51	51.1	1.0
10	0.63	33.3	1.2
20	0.90	16.7	1.5
40	1.27	9.4	2.4

When it is necessary to load subscriber cable pairs, 44-mH coils spaced 4500 ft are used. This gives a characteristic impedance of about 740 ohms for unit twin cable which is not too high to impair the side-tone performance of the telephone appreciably. Rules for correcting the T.E.R. for loading are under review at present.



Notes. — (1) Toll junctions direct from local exchange to G.C. to be provided whenever reasonably possible.
(2) Minimum toll junction loss is 2 dB.

FIGURE 1. — **Maximum connection via basic trunk networks (New Zealand)**

In the case of P.A.B.X.s, the maximum permissible T.E.R. of the exchange lines is governed by the T.E.R. of the most distant extension from the P.A.B.X.

T.E.R. corrections are made for multiplied cable pairs, tee-offs and various other factors.

Different T.E.R. limits are set when the telephones, feedbridge or battery voltage are different from standard.

Toll junctions (i.e., junction circuits which carry traffic to or from the trunk network)

Maximum line loss : 4.5 dB

Minimum line loss : 2 dB

Mean line loss for a group : Not greater than 3.5 dB

In most cases local exchanges have direct junctions to the toll centre. In some cases they are switched at an intermediate exchange but it is hoped to reduce such cases progressively.

Direct toll junctions less than 2 dB will be built out to 2 dB and 2-dB pads will be inserted in toll calls to or from subscribers connected to the toll centre exchange.

A maximum average junction loss has been specified in order to avoid always engineering to an upper limit.

Trunk circuits (long-distance circuits)

Two types of trunk circuits are provided, known as basic and auxiliary circuits. Auxiliary circuits are provided between toll centres when there is sufficient traffic to justify a separate group of circuits. They carry only traffic terminating between the toll centres concerned and cannot be through-switched to other toll centres. They are usually provided by the same carrier systems which provide the basic circuits but may be provided by two-wire plant.

The overall loss of carrier circuits used as auxiliary trunks has been fixed at 4.5 dB. Circuits provided by cable pairs may have lower loss.

Basic circuits are provided between toll centres and a higher category of switching centre (termed zone centres) and between all zone centres and the national switching centre, which is at Wellington. Basic circuits are also provided between zone centres when there is sufficient traffic to justify them.

Basic circuits are always four-wire circuits, usually carrier. They terminate on four-wire switches at zone centres and are extended, as required, to build up a connection. The overall two-wire to two-wire loss of basic circuits is $(4 + 0.5n)$ dB, where n is the number of links in tandem.

Calls within an exchange group

Such calls are those which do not extend beyond the group of exchanges which share the same toll centre. A maximum of four junction links is permitted (i.e., three intermediate switching points) and the majority will have fewer links. The maximum junction line loss permitted is 13 dB. In addition switching and mismatch losses of 1 dB are allowed for each of the intermediate switching points, giving a total of 16 dB between terminal exchanges for the maximum linkage.

Impedance matching at the two-wire point

A compromise hybrid balance network of 900 ohms in two series with a 2 μF capacitor is being adopted at the two-wire termination of the trunk network and also for the termination of any junction carrier or four-wire repeated circuits. Junction circuits will consist of cables loaded with 88-mH coils with a half-end section at the toll centre. The impedance at this point is about 1100 ohms at frequencies below about 2000 Hz, rising considerably at higher frequencies. The end section will be built-out with capacity of 0.8 section and then terminated in a 44-mH coil. It will then present an impedance to the toll centre of about 900 ohms. It is expected that this, in conjunction with a minimum toll junction loss of 2 dB, will give adequate control of return loss. It will also minimize mismatch loss at the toll centre.

In the case of subscribers on the toll centre exchange, 2-dB 900-ohm pads will be used to simulate minimum loss junctions. The New Zealand Post Office makes some use of 4½- and 2½- lb. per mile cables which have a fairly high impedance and give a reasonable match to 900 ohms.

Switching losses

The New Zealand transmission plan allows 1 dB switching loss at the toll centres, at each end of a trunk connection, where calls are switched on a two-wire basis. The main loss-producing elements in an exchange are the transmission bridges, a capacitor bridge having a loss at 1000 Hz of about 0.5 dB and a transformer bridge having a loss of about 0.75 dB. The loss tends to rise below 500 Hz due to the effect of the falling shunt inductive reactance and rising series capacitive reactance and these factors also degrade return loss at the lower frequencies.

In order to reduce switching loss and improve return loss it is necessary to minimize the number of transmission bridges in the two-wire circuit at the toll centre. It is proposed to build the trunk circuit hybrid transformer into the signalling relay sets and to use it as a transmission bridge. By this means it is expected that only one bridge will be in circuit at the originating end of operator-dialled calls and none in the incoming end.

Syllabic compandors

Not more than three syllabic compandors are permitted in an overall connection, one in the trunk network and one at each end, either in the junction circuit to the local exchange or in subscriber carrier equipment. In order to prevent compandors being tandem-connected in the trunk network, they will be permitted only in auxiliary circuits.

Variation of trunk circuit losses

In order to control variations of trunk circuit loss from nominal, all line links and supergroups are equipped with pilot-controlled regulators. Groups are regulated only when they are extended on a through-group basis.

Up to the present time, routine measurements of overall trunk circuit loss have been made between two-wire lines, at the hybrids. In future it is proposed to include signalling and switching equipment in the measurement. Automatic routiners and digital measuring instruments are being developed to operate over this path. It is expected that a much closer control of trunk loss will then be possible.

National Transmission Plan of Pakistan

The national transmission plan provides that nominal equivalent between two subscribers in national calls will not exceed 33 dB in 95% of the connections. The reference equivalent between a subscriber and the four-wire terminals of the international circuit shall not exceed 20.8 dB in send and 12.2 dB in receive directions. Loss allocations of the system are as follows :

Inset reference equivalent

Transmitter inset	2.6 to 0.8 dB
Receiver inset	2.6 to 5.2 dB

Reference equivalents in a connection

Maximum sending reference equivalent of subscribers' system	10.8 dB, say 11 dB
Maximum receiving reference equivalent of subscribers' system	3.4 dB, say 3 dB
Sum of send and receive reference equivalents of subscriber instruments (line) and bridge	14 dB
Terminal reference equivalent between two exchanges	19 dB
Total reference equivalent	33 dB

Contribution by the Netherlands Administration

In drawing up a national attenuation plan, the Netherlands Administration took the following principles as a basis :

1) Total subscriber-to-subscriber attenuation (excluding telephone stations) should not exceed 30 dB.
2) This attenuation of 30 dB must be apportioned as economically as possible between the various parts of the network.

Apportionment of attenuation

As far as apportionment of attenuation is concerned, the Netherlands network is subdivided into :

interregional network (between regions);

primary networks (between regional exchange and nodal exchanges);

secondary networks (between nodal exchange and local networks);

local networks.

Traffic efficiency is at its lowest in the local network, because provision has to be made for one pair of wires for each individual subscriber's line. In all circumstances, it will be most economical to tolerate the greatest possible degree of attenuation in the local network. This means that less attenuation will be available for the other networks and, hence, attenuation of the trunk circuits will be very small indeed.

In the present attenuation plan, the maximum tolerable figures for attenuation are :

interregional network	0 dB
primary network	0 dB
secondary network	3 dB
local network	5 dB

The maximum attenuation between two subscribers will then be 27 dB (including the attenuation of the telephone exchanges and feeding current loss).

Transmission in a local network

Because of the low trunk attenuation (see above), an attenuation of 5 dB between a connection exchange and the subscriber's set can be tolerated at present.

This attenuation can be ensured by the use of non-loaded cables 5 km long, with conductors measuring 0.5 mm in diameter, provided there is a telephone set of a standard type at the end of the circuit.

The tolerable resistance of the pair, loop-measured, is 1000 ohms at the most. Hence, in drawing up the plan for local cables, it will suffice to observe the following two rules :

a) a loop resistance of 1000 ohms;

b) a cable length of 5 km.

These standards can be met by the use of cables with a diameter of 0.5 mm.

Formerly trunk circuits took up a considerable part of the total available attenuation, and so considerably less attenuation than at present could be tolerated in the local network. This led to the use of short cables and to the creation of numerous local exchanges, each of which served a relatively small area. Hence, in only a relatively small number of cases the above-mentioned length has been exceeded. In these exceptional circumstances special measures are required (for instance, the use of cable with conductors of greater diameter).

Sometimes it will suffice to create more satellite exchanges. By analogy with the trunk-cable network, the network interconnecting the satellite exchanges and the one linking these exchanges to the trunk exchanges will have to meet the following requirements :

satellite exchange-satellite exchange : 6 dB maximum;

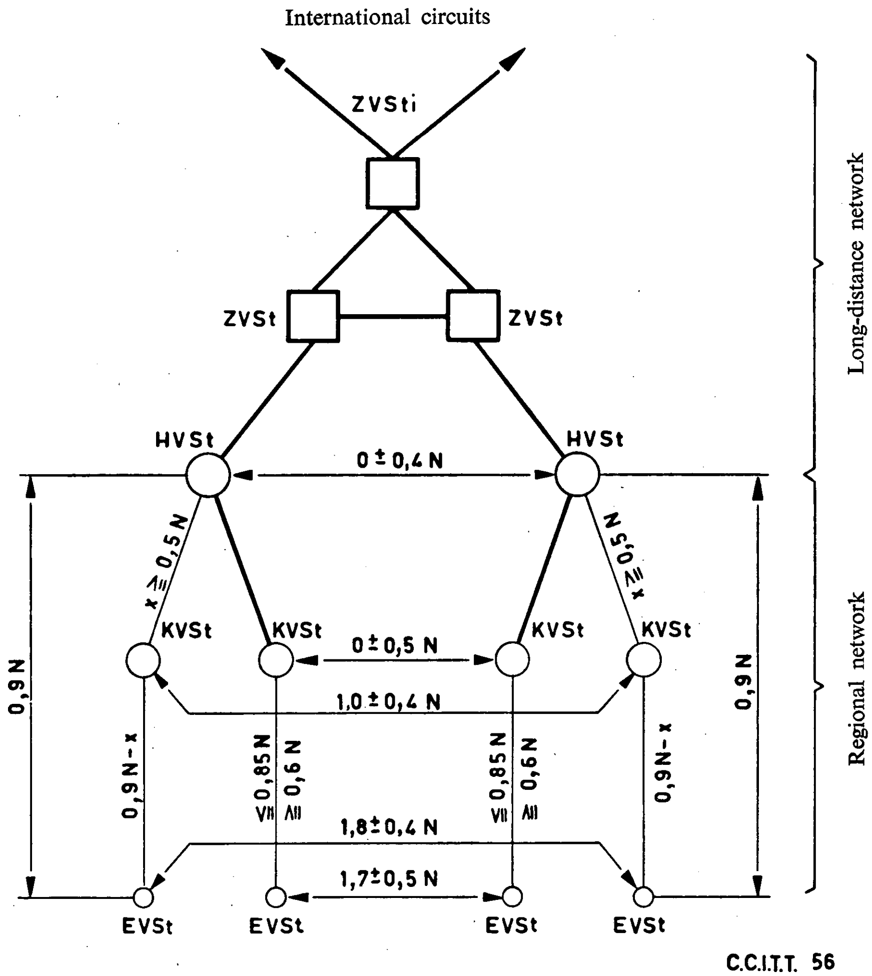
satellite exchange-trunk exchange : 3 dB maximum.

The interconnecting cables are mostly coil-loaded. In general, the diameter of the conductors is 0.8 mm, the self-inductance of the coils is 65 mH, and the coil-spacing 3 km.

Contribution by the Administration of the Federal Republic of Germany

The automatic trunk telephone network of the Federal German Administration has meanwhile been extended in accordance with the attenuation plan published on page 189 of the *Red Book*, Volume V. The new designations of the switching centres are indicated on Figure 1.

The long-distance network now has only four-wire circuits, which are interconnected by selectors for four-wire circuits in the tertiary centres (ZVSt) and the secondary centres (HVSt). The circuits connecting the secondary centres (HVSt) and primary centres (KVSt) are also almost entirely four-wire circuits, but in addition to the primary centres with selectors for four-wire circuits, there are also primary centres (KVSt) with selectors for two-wire circuits. For four-wire interconnection in a primary centre (KVSt), all the two-wire circuits connecting it to a local exchange are equipped with a terminating set. Balancing is adjusted in each terminating



Two-wire circuit — Four-wire circuit

EVSt Local exchange
KVSt Primary centre
HVSt Secondary centre

ZVSt Tertiary centre
ZVSti Tertiary centre with international lines

FIGURE 1. — **Attenuation diagram for the automatic trunk network (Federal Republic of Germany)**

set to the circuit concerned. The attenuation of 4 dNp of the terminating set can be suppressed by putting out of circuit the attenuation lines of 4 dNp at the end of the four-wire circuit that is to be connected (interconnection with attenuation compensation).

All the attenuation values indicated in Figure 1 apply to sections and relate to the interconnection points at the selector contacts. In addition to the nominal attenuation values, the diagram also shows the variations which would be produced allowing for an attenuation variation of ± 2 dNp for each circuit with repeaters and quadratic addition of the attenuation variations at all the circuits of a chain circuit. The maximum equivalent (including these attenuation variations) between two local exchanges (EVSt) is 2.2 Np. Since all the four-wire trunk circuits are connected in series with an insertion loss of 0 Np and since the tertiary centre with international lines (ZVSti) is reached from each primary centre (KVSt) by three circuit sections at the very most, the Federal German attenuation plan also conforms to the new C.C.I.T.T. recommendations relative to an interconnection plan.

The maximum sending and receiving reference equivalents of a subscriber line are respectively 1.2 Np and 0.2 Np with respect to the local exchange (EVSt). These values can be exceeded by 0.3 Np for extensions. Since the attenuation of the section between a local exchange (EVSt) and a four-wire interconnection point of the chain trunk circuits is 0.9 Np, 2.4 Np and 1.4 Np are obtained for the maximum national sending and receiving reference equivalents respectively, which conforms to Recommendation P.11 for national sending and receiving reference equivalents.

The network of circuits indicated on Figure 1 will be completed by a network of direct circuits. Direct circuits between switching centres will be established in all cases where a sufficient volume of telephone traffic justifies the provision of special systems of circuits. The maximum equivalent of 2.2 Np between local exchanges (EVSt) will likewise not be exceeded for telephone calls set up over these direct circuits.

The Federal German Administration is considering the following reference values, in addition to the existing recommendations of the C.C.I.T.T., as regards the operation from the transmission aspect of new equipment in local exchanges and for subscriber lines permanently connected to them :

a) The insertion loss of a local exchange (including feeding-bridge) should not exceed 15 cNp at 800 Hz.

b) The crosstalk attenuation between two circuits should not be less than 8 Np.

c) The balance attenuation of the switching equipment, in the speaking position, should be at least 5.3 Np in the limits 300-3400 Hz of the transmission band.

d) The psophometric voltage of a local exchange should not exceed 0.3 mV (assessed with the C.C.I.T.T. weighting curve for telephony), the measurement being carried out with a connection to the main distribution frame and terminated across 600 ohms.

e) The maximum loop resistance of the subscriber line (excluding the telephone set) is 1600 ohms.

f) The subscriber lines and junction circuits should have an insulation resistance of at least 50 kilo-ohms between the wires and between the wire and the earth.

g) The transmission of switching information on subscriber lines and junction circuits should not be prejudiced by longitudinal voltages of up to 65 $V_{r.m.s.}$

h) The admissible voltage variations for a 60-V supply are +3 V and −0.4 V with respect to the nominal value.

Classification of microphone and receiver insets into groups according to their sensitivity

The reference equivalent of a telephone set depends essentially on the sensitivity of the electro-acoustic transducers used, i.e. the microphone and receiver insets. The sensivitity of carbon microphone insets is so dependent on the supply current that it decreases with it.

In the measurement of inset reference equivalents, it is noted that insets of different makes and those of the same make may present very different sensitivities. Even in the case of flawless manufacture some tolerance has to be allowed for the manufacturer if the production of insets is not to be uneconomical. Hence, it is reasonable to group the insets according to their sensitivity, using the more sensitive insets for the longer subscriber lines and the less sensitive ones for the shorter lines, so as to obtain some attenuation compensation.

For each diameter of the conductors, there exists a relation between the length or the d.c. resistance and the reference equivalent, so that the reference equivalent of a subscriber line consisting of wires of constant diameter can be characterized by the d.c. resistance of the line. For subscriber lines composed of sections with wires varying in diameter, the equivalent measured at 1300 Hz corresponds with sufficient accuracy to the reference equivalent of the line. In the calculation of the reference equivalent, account must be taken of the supply attenuation which occurs in addition at the sending end.

In practice, it has proved sufficient to arrange 0.3 Np steps for receiver insets and 0.4 steps for microphone insets.

A pad is inserted for subscribers who are connected to the exchange by very short lines. This method of providing a homogeneous line causes an increase in the attenuation of the local signal. Furthermore, a group is saved for the microphone and receiver insets respectively.

The following table shows the association of groups of insets with loop resistances in the subscriber line.

Loop resistance (ohms) (excluding telephone set)	0 to 500 [a]	500 to 750	750 to 1000 [b]
Group of microphone insets . . . . . Sending reference equivalent (nepers)	I +0.5 to +0.1	II +0.1 to −0.3	III −0.3 to −0.7
Group of receiver insets . . . . Receiving reference equivalent (nepers)	II −0.3 to −0.6	III −0.6 to −0.9	IV −0.9 to −1.2
Sending reference equivalent of the subscriber system	+0.8 to +1.2	+0.7 to +1.2	+0.6 to +1.2
Receiving reference equivalent of the subscriber system	0 to +0.2	−0.2 to +0.2	−0.3 to +0.2

[a] A pad having a reference equivalent of about 0.3 Np is inserted in subscriber lines with a loop resistance of less than 250 ohms.

[b] By using a supplementary supply, the range can be extended beyond 1000 ohms up to 1600 ohms, provided the limit values for the sending and receiving reference equivalents are not exceeded.

As can be seen from the table, the limit values of +1.2 Np and +0.2 Np fixed in the 1955 attenuation plan of the Federal German Administration, for the sending and receiving reference equivalent of a subscriber system, are respected.

Since the microphone and receiver insets can easily be interchanged, the attenuation compensation mentioned above can be obtained without extra staff or cost. The manufacturer measures the microphone and receiver insets at objective stations for the measurement of reference equivalent and notes their respective sensitivity. During these measurements, other inset characteristics are measured in addition to the reference equivalent.

Staff engaged on fault detection have with them microphone and receiver insets of all the sensitivity groups, so that when insets are replaced they can always insert insets of the required sensitivity group.

CONTRIBUTION BY THE UNITED KINGDOM ADMINISTRATION

Limits applied in national trunk and local networks

As a result of subjective tests made by the United Kingdom Administration some years ago it was decided that the transmission loss at 800 Hz introduced between minor exchanges should not exceed 20 dB [1]. Switching and reflection losses are included in this allowance. The reference equivalent of two limiting subscribers' lines and instruments currently in use in United Kingdom local networks is 13 dB (adding together the separately measured sending and receiving reference equivalents). Thus the total maximum reference equivalent for a national connection may be quoted as 33 dB. (N.B. : The concept of " reference equivalent ", although useful for describing a network, was not used by the United Kingdom Administration for planning the network.)

The way in which the 20-dB loss between minor exchanges is apportioned among the various classes of circuit is shown in Figure 1. It is patently desirable to be able to assign as much loss as possible to the unamplified junction network (minor exchange-group switching centre) where a reduction in allowable loss of even 1 dB means the expenditure of much capital, and to assign as little as possible to the mostly amplified trunk network where amplifier gain must be provided anyway and extra gain costs very little. However, a limit is set by the balance return loss that can be guaranteed at the terminating set of the trunk circuit at the group switching centre which is a two-wire switching point. In order to avoid an expensive scheme of impedance correction at all group switching centres throughout the country and also to avoid having to use echo suppressors, the loss of the trunk network measured between two-wire points was fixed at 7 dB. This loss is independent of the number of trunk circuits, thereby avoiding pad switching. The transmission loss of each of the two junction circuits in a maximum connection is planned to be 4.5 dB which corresponds to about 12 miles (19 km) of unamplified, loaded audio cable (88 mH at 1.83-km spacing, 0.9-mm diameter copper wire).

There thus remain 4 dB for switching and reflection losses of which 3 dB are shown in Figure 1 as two switching-point losses at group switching centres. Reflection losses are significant at the higher audio frequencies on unloaded audio cables because of impedance mismatch at the two-wire switching point. There are no significant switching and reflection losses in the trunk network because this is four-wire switched as indicated and most of the circuits are routed on carrier systems and present a good, designed impedance at the audio switching point.

A useful general reference is :

TOBIN, W. J. E. and STRATTON, J.: A new switching and transmission plan for the inland trunk network; *P.O.E.E.J.*, Volume 53, Part 2, July 1960.

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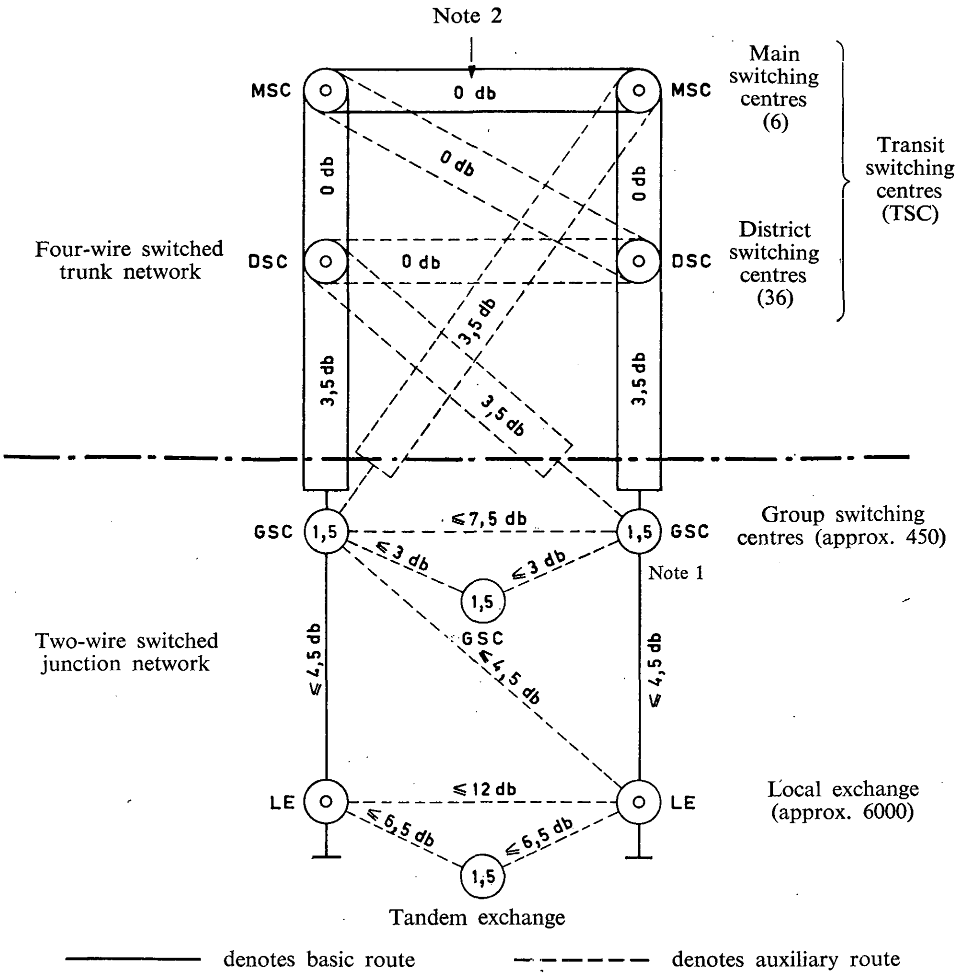
[1] This assumes no significant attenuation/frequency distortion. In the case of unloaded audio cable the attenuation at 1600 Hz is used for planning purposes.



Notes.

1. Nominal exchange loss. This is 0 dB at terminal minor exchanges because the exchange loss is included in the allowance for the subscriber's set and subscriber's line.
2. Nominal loss of the circuit.
3. If one or the other of the circuits leaving the group switching centre is repeatered it is adjusted to a nominal loss of 3 dB.

FIGURE 1. — **New trunk switching and transmission plan (United Kingdom)**

Limits applied in local exchanges

The fundamental principle is that the transmission performance of any subscriber connected by a subscriber's line to a local exchange should be equal to or better than a minimum standard represented by a given combination of a subscriber's station, a subscriber's line and the exchange feedbridge. The design of the local network is such that most subscribers get transmission performance well above the minimum.

The minimum standard and the method of determining the relative transmission performance for sending and receiving are described in detail in Annex 1 of Part II of Volume V of the *Red Book* (reproduced as Supplement No. 8 in this Book).

CONTRIBUTION BY SINGAPORE (SINGAPORE TELEPHONE BOARD)

C.C.I.F. standards; maximum attenuation of junction links 8 dB; maximum sending and receiving for local lines 10 and 5 dB respectively.

TRANSMISSION PLAN OF THE REPUBLIC OF SOUTH AFRICA

General

The present transmission plan aims at an overall target equivalent of 18.5 dB. This stringent requirement will be met by employing four-wire through-switching at all new automatic trunk exchanges, classified as primary, secondary and tertiary in the trunk system and by exploiting the gain of four-wire trunk lines to the maximum practicable extent on both four-wire to four-wire and four-wire to two-wire connections.

However the limit of 18.5 dB will not be fully achieved on all calls in the near future for the following reasons :

a) The nominal equivalent on a small fraction of trunk calls will be permitted to exceed the limits for economic reasons.

b) Many manual trunk exchanges (two-wire switching) have still to be replaced with four-wire automatic exchanges.

c) Junction cables for the larger local networks have in the past been designed around an overall equivalent of 25 dB. A drastic change in the attenuation pattern of these networks and the re-grouping of tandem centres cannot be justified economically on a short-term basis. Endeavours will nevertheless be made in future to reduce the attenuation of junction circuits to values below the existing limits.

Trunk traffic to and from these networks will not be adversely affected as direct low-loss toll circuits are being provided between all terminal exchanges and the area trunk switching centre.

The trunk line network

The basic transmission plan for the trunk system is shown diagrammatically in Figure 1.

The salient features of the plan are as follows :

1) All trunk circuits between primary, secondary and tertiary switching centres are four-wire and lined-up to a four-wire equivalent of —2.5 dB (which yields a corresponding two-wire equivalent of 4.5 dB).

Note.— The majority of trunk circuits in this country are carrier-derived.

2) Trunk junction circuits between trunk exchanges and terminal exchanges may be four-wire or two-wire circuits.

3) When two links are switched together, transit pads are introduced to offset the additional gain of 2.5 dB and to add an extra loss of 0.5 dB at tertiary and secondary exchanges and 1.0 dB at primary exchanges in order to increase the overall two-wire equivalent progressively on multi-link connections. In the most adverse case of an E1 - T1 - S1 - P1 - P2 - S2 - T2 - E2 connection the two-wire equivalent becomes 8.5 dB.

4) Provision has been made for increasing the gain of four-wire links when switched to two-wire links so as to realize overall equivalents approaching those of connections involving four-wire links only.

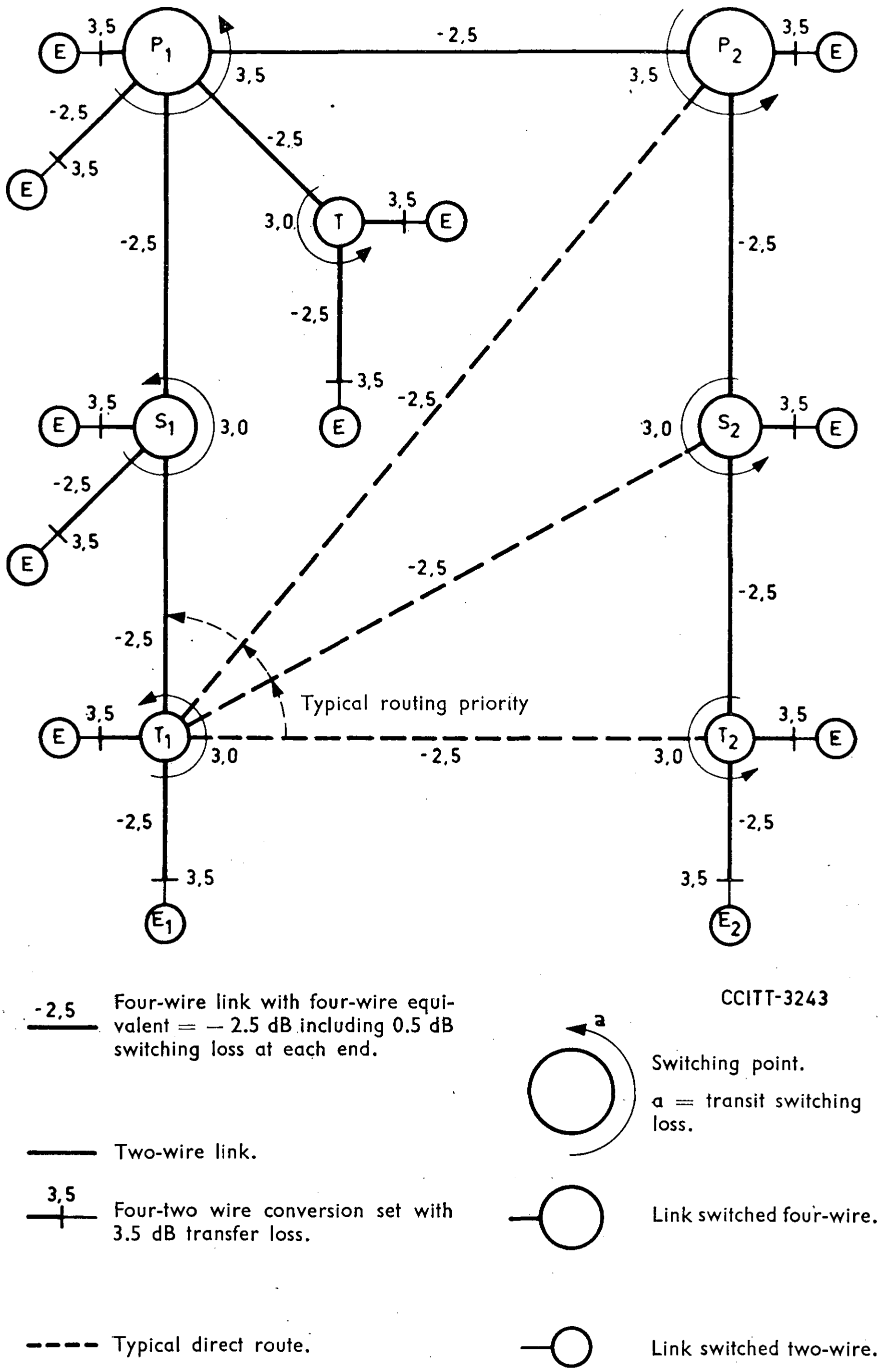


FIGURE 1. — **Basic four-wire transmission and routing plan (Republic of South Africa)**

The losses of all unamplified two-wire trunk and junction circuits terminating at primary exchanges (loss in excess of 2 dB) and other trunk exchanges (loss in excess of 1 dB) are compensated for or de-attenuated. The amount of de-attenuation realizable is governed by the return loss obtainable at the four-two-wire conversion point.

Two-wire circuits requiring de-attenuation are permanently associated with four-two-wire conversion sets and precision balancing networks. Two-wire cable circuits requiring de-attenuation are loaded as a matter of course. The higher impedance of loaded cable circuits (1100 ohms) is matched in the four-two-wire conversion circuit to the impedance of the trunk equipment (600 ohms).

5) Physical trunk circuits terminating at a trunk exchange are amplified irrespective of loss, if de-attenuation facilities are required and in all other cases if the loss exceeds 4.5 dB.

6) The gain of four-wire links is increased to offset switching losses in four-wire paths as well as hybrid losses. A value of 0.5 dB per trunk line termination or 1.0 dB per exchange is catered for.

The junction network

1) The resistance and transmission loss of single continuous junctions are limited to 1500 ohms and 6 dB respectively.

2) Junctions are arranged in such a manner that the cumulative transmission loss of junctions switched in tandem is limited to 12 dB.

3) Junction calls are routed through a maximum of five exchanges—i.e. two terminal and three tandem exchanges. A switching loss of 1 dB per exchange is allowed.

4) Junction circuits are generally unloaded or loaded (88 millihenry coils at 2000-yard intervals) physical circuits. Increasing use is however being made of carrier-derived circuits.

Subscribers' lines

A maximum loop resistance of 1000 ohms is adhered to. The attenuation of the maximum subscriber's loop is of the order of 7 dB at 800 Hz.

Subscribers' instruments

The performance of the majority of telephones in use is as follows when used in conjunction with a 1000-ohm subscribers' loop (6½ lb. per mile cable circuit) :

Send reference equivalent : 8.0 dB [1]

Receive reference equivalent : 0 dB [1]

The latest type of instrument, now being manufactured in South Africa, has send and receive equivalents of 3.0 dB and 2.0 dB, when used on a maximum subscribers' loop of 1000 ohms, i.e. a 3-dB improvement on the performance of the telephones in general use.

Note.— Exchange feedbridge loss has not been included in the above equivalents.

[1] Relative to S.F.E.R.T.

CONTRIBUTION BY THE SWEDISH ADMINISTRATION

The setting-up of our national transmission plan for a fully-automatized telephone network has been governed essentially by the following factors:

1. Reference equivalents for most telephone sets provided in the Swedish network are 0 ± 0.3 Np at sending and -0.7 ± 0.2 Np at receiving.

Note. — Telephone sets now being made have reference equivalents of -0.2 ± 0.3 Np (sending) and -0.9 ± 0.3 Np (receiving).

2. Division of the country into exchange areas, numbering areas and transit routing areas.

3. The existing line network, in particular the long-distance lines and such shorter cables as were already included in automatized areas at the moment of preparation of the plan. In this respect, our initial situation in preparing for long-distance automatization was unfavourable, seeing that the existing plants to be considered in certain decisions were of comparatively great extent. They included mainly coil-loaded cables, but the construction of a carrier frequency cable network had begun and was assumed to prevail in future extensions of the long-distance line system.

4. Existing automatic exchange equipments. Such equipments were already in service, at the moment of planning, in many centres of numbering areas and in many auxiliary exchanges, all of them of the two-wire type. In the building-up of transit centres for the automatic long-distance service, on the other hand, the designers had full liberty as far as technical features of transmission were concerned.

An aim pursued in planning for a national automatic network has, of course, been to fulfil, for all domestic and international calls, the C.C.I.T.T. requirement of a reference equivalent of 4.6 Np and, if possible, to achieve a lower value (see an article by G. Swedenborg : " A survey of the development of telephone apparatus from the speech transmission aspect " appearing in *Tele*, English edition No. 1/1953). To achieve this object at reasonable cost, amplification should be used to the largest possible extent, within the limits set by echo and singing phenomena. The solution arrived at in this respect in described by B. Bjurel, H. O. Björk and E. Waldelius in another article published in the aforesaid issue of *Tele* : " Technical viewpoints respecting automatization of trunk traffic ", pp. 21-24. By way of a summary, the principal items of the transmission plan in question are given below :

a) Four-wire pad switching is used in the switching centres to which the numbering area centres (NAC) are attached.

b) To obtain the simplest possible level conditions, all links of a connection NAC-NAC have the nominal attenuation 0 in transit service. A further 0.3 or 0.5 Np will be added at each terminal point, depending on the type of the component links, so that in the most unfavourable case, the total attenuation NAC-NAC will attain nominally 1.0 Np (in the case of carrier frequency only 0.6 Np).

c) The attenuation over the line from a subscriber to an NAC must not exceed 1.5 Np, including the current feed attenuation (1.2 Np from NAC to subscriber). The attenuation in local and toll networks respectively are thus added; their mutual proportion is decided in each case on the basis of economical considerations. In this way it is possible, for instance, to achieve economies in the local network when the distance between the local exchange and the NAC is small.

d) Taking into account the tolerances applicable to the performance of telephone sets and the variations in time of line attenuations, the maximum reference equivalent for national calls will be about 4 Np. In the majority of cases, however, transmission conditions for interurban calls will be considerably more favourable. Further improvement is however desirable, especially in the international service, where the fulfilment of the C.C.I.T.T. requirements respecting the reference equivalent of the national sending system presents certain difficulties.

Note.— The Swedish Administration is prepared to introduce the overall reference equivalent of 4.14 Np as soon as this is accepted by the C.C.I.T.T., provided that the values chosen for the reference equivalent (sending and receiving) are 2.4 and 1.4 Np respectively.

Transmission performance of local exchanges

1. The highest admissible operating attenuation in a local exchange, measured between the test jacks of the main distribution frame for incoming and outgoing lines is 2 dNp at 300 Hz and 1 dNp in the range 800-3000 Hz.

2. The lowest admissible crosstalk attenuation between any two call connections in a local exchange is shown in Figure 1.

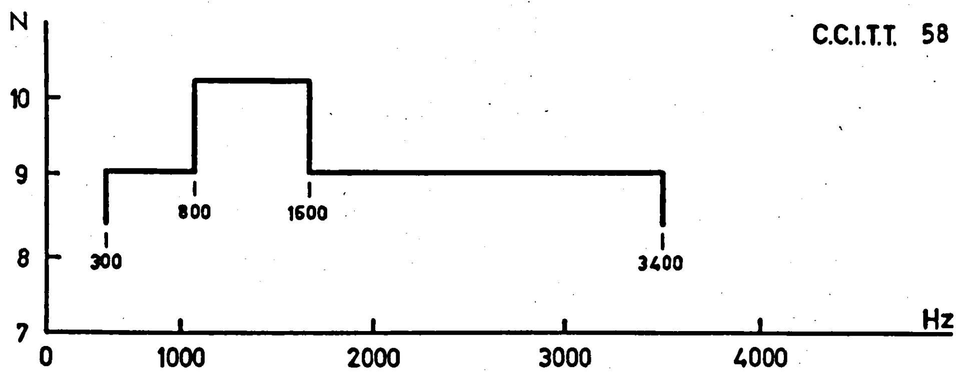


FIGURE 1. — (Sweden)

The crosstalk attenuation is measured at the test jacks of the main distribution frame, the circuits being terminated by a resistance of 600 ohms.

The crosstalk attenuation between different exchange apparatus should be proportioned so that the values indicated above for the whole exchange are obtained. This implies, as a rule, that the crosstalk attenuations to be chosen between different exchange components must be about 1 Np higher than the values shown on the diagram.

THE 1966 TRANSMISSION PLAN OF THE SWISS ADMINISTRATION

Introduction

The 1957 transmission plan [1] was still based to a large extent on two-wire switching in the trunk terminal exchanges. Transmission technique has developed in the meantime, and four-wire switching has been introduced progressively since 1965 in such exchanges. Advantage was taken of the introduction of this new switching method to prepare a new transmission plan.

[1] *Red Book*, Volume V *bis*, Annex 4, section XIV, page 115.

The purposes of the 1966 transmission plan

The transmission plan is a technical basis for the planning and implementation of trunk, rural, and local networks. When this plan was revised, the following requirements had to be taken into account :

— Reduction of possible maximum equivalents in extreme cases,
— Compliance with C.C.I.T.T. Recommendations, e.g. improvement of the signal-to-noise ratio and reduction of attenuation distortion,
— Adequate stability for telephone connections, to avoid a strong echo on passage from a four-wire circuit to a two-wire circuit,
— Negligible crosstalk,
— Sufficiently high quality to justify the additional expenditure.

The attenuation plan

The new transmission plan is characterized by four-wire switching. In principle, the attenuation on a four-wire chain is nil : it is only on passage to a two-wire chain that the nominal equivalent of 0.8 Np is attained, owing to the attenuation of the four-wire—two-wire terminating sets. The audio-frequency circuits between the terminal trunk exchange and the attached local and rural exchanges—the attenuation of which exceeds 0.1 Np—are equipped with a terminal amplifier. These two/four-wire amplifiers compensate the attenuation of the associated circuits so that a value of 0.8 Np is also obtained between the two-wire ends. The carrier circuits of the rural network are four-wire on the trunk terminal exchange side and two-wire on the nodal exchange or terminal exchange side (Figure 1).

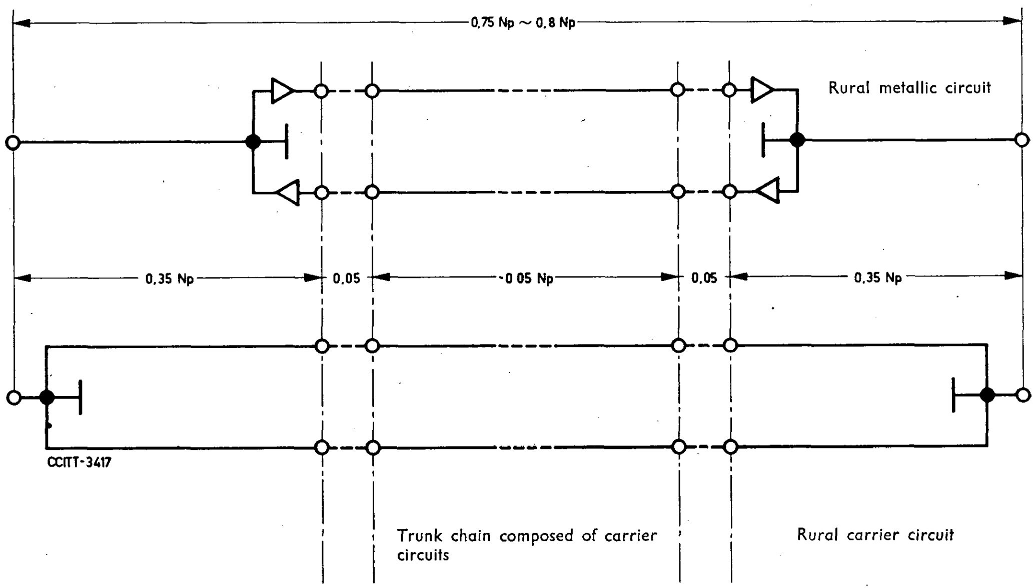


FIGURE 1. — **(Switzerland)**

If the equivalent of a trunk connection is not to be increased by the attenuation of the transit exchanges, each four-wire circuit and each circuit with a terminal amplifier must be lined-up on the average with a gain of 0.05 Np. In this way, a C.C.I.T.T. Recommendation is observed at the same time. Since there are always n circuits and $n - 1$ transit exchanges, the equivalent of 0.8 Np becomes in effect 0.75 Np.

The transmission plan would not be realistic if it were not to allow for the dispersion of the levels of the audio-amplified circuits and the carrier circuits. For each circuit section, a standard deviation 6 of 0.08 Np is to be expected. If account is taken of the distribution of traffic between terminal and tandem traffic, it is easy to show that for scarcely 1% of the trunk traffic the equivalent will differ from the nominal value by more than $\pm$ 0.24 Np = 3 σ.

Figure 2 shows the new transmission plan, which calls for the following comments :

The 1966 attenuation plan differs from the past one chiefly in that four-wire switching in the terminal trunk exchanges enables the equivalent limit of 0.8 Np to be brought much nearer to the subscriber station. The equivalent of 0.8 Np, or 0.75 Np to be more exact, applies generally to all trunk links between local district exchanges (if A>0.1 Np), nodal exchanges, rural exchanges and terminal rural exchanges, provided the latter are attached direct to the terminal trunk exchange.

For links between district exchanges, one restriction should perhaps be mentioned : the junction circuits connecting the trunk exchange with the district exchange, the attenuation of which is 0.1 Np or less, are not equipped with terminal amplifiers but only with hybrid transformers. In this case, the value of 0.75 Np may be increased by 2 $\times$ 0.1 Np for junction circuits and 2 $\times$ 0.05 Np for the uncompensated attenuation of the exchanges, and it may thus reach as much as 1.05 Np.

As regards connections to rural terminal exchanges linked up with a rural nodal exchange, account must be taken of the attenuation of the circuit linking these two exchanges, since it is no longer included in the standard value of the prolonged trunk connection. In these cases, an overall attenuation of 0.9 Np has been fixed for the link comprising the nodal exchange, the junction line from the nodal exchange to the terminal exchange, the terminal exchange and the subscriber line. This value may be freely distributed within certain limits.

The subscriber line will have a maximum equivalent of 0.52 Np at 800 Hz. This value is limited chiefly by the loop ohmic resistance, which must not exceed 700 ohms, and also by the attenuation distortion.

The maximum total equivalent between the subscriber stations may be fixed in accordance with the following :

1.99 Np for general circumstances

2.29 Np for district exchanges

2.55 Np for terminal exchanges.

The reference equivalent

The intelligibility of a telephone call is determined principally by the reference equivalent. When the connecting circuits have only a low attenuation distortion, the reference equivalent of the complete connection is taken to be the sum of the circuit equivalents at 800 Hz and the reference equivalents of the local systems.

Leaving a margin of 0.1 Np for any supplementary equipment, the reference equivalents of the local systems will be :

Subscriber station	*Subscriber cable*	*⌀ 0.6 mm*
	0 km	5.3 km
PTT 1950 model send	0.2 Np	1.2 Np
Projected model send	0.8 Np	1.0 Np
PTT 1950 model receive	0.1 Np	0.8 Np
Projected model receive	−0.15 Np	0.2 Np

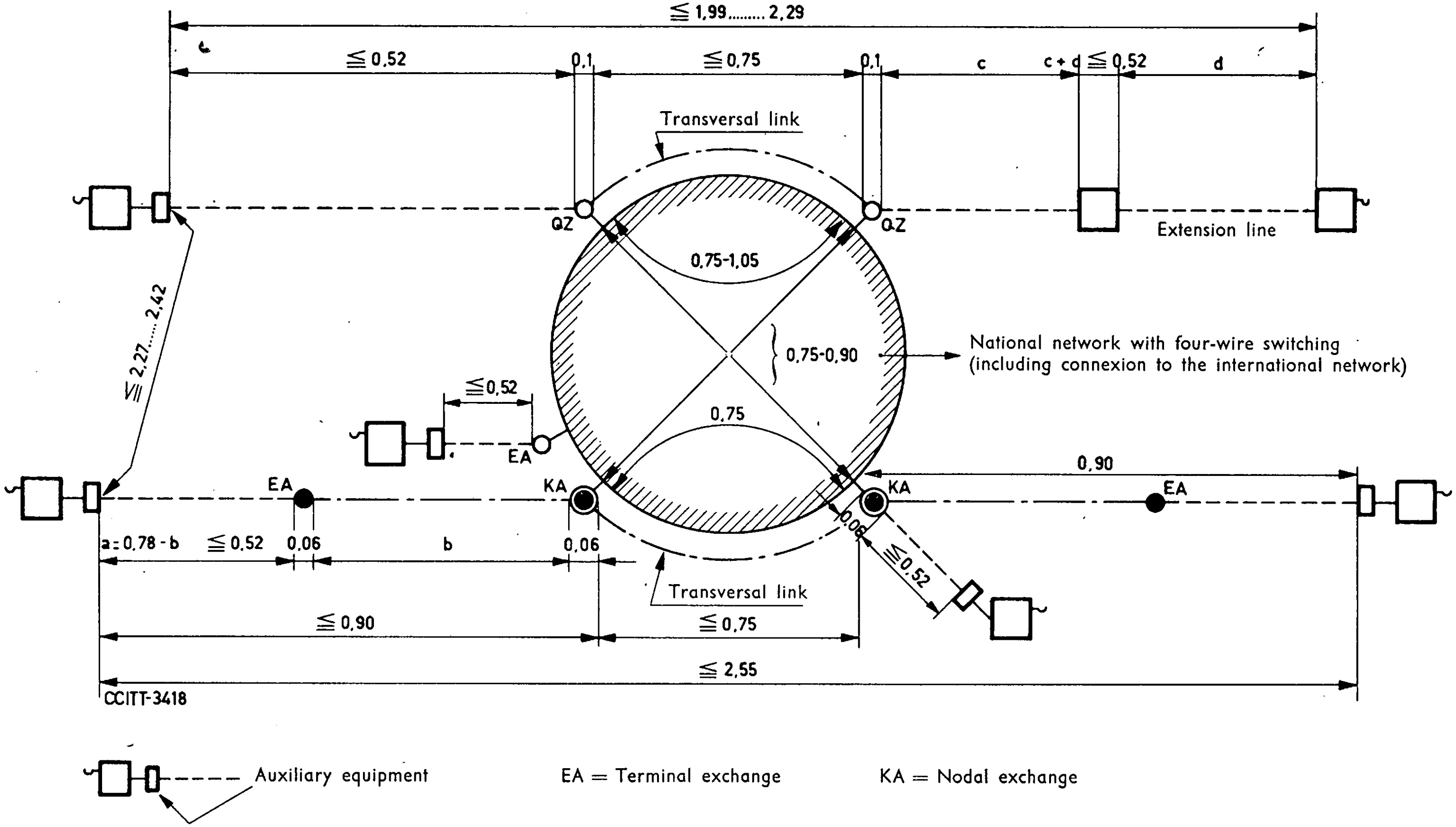


FIGURE 2. — **(Switzerland)**

Thus, for international connections, the following maximum nominal reference equivalents are obtained :

Connection :

National sending system

with 1950 model	2.0 Np
with projected model	1.8 Np

National receiving system

with 1950 model	1.6 Np
with projected model	1.0 Np

The tolerances for the line (± 0.24 Np) and for the subscriber stations (± 0.2 Np) must be added to these figures. It will be noted however that the reference equivalent at reception, with the 1950 model, is always above the value of 1.4 Np recommended by the C.C.I.T.T. But if we consider the number of international calls which will be set up in these extreme cases, we shall see that, even making allowance for possible tolerances, their percentage remains well above the value of 5% admitted by the C.C.I.T.T.

Contribution by the Czechoslovak Administration

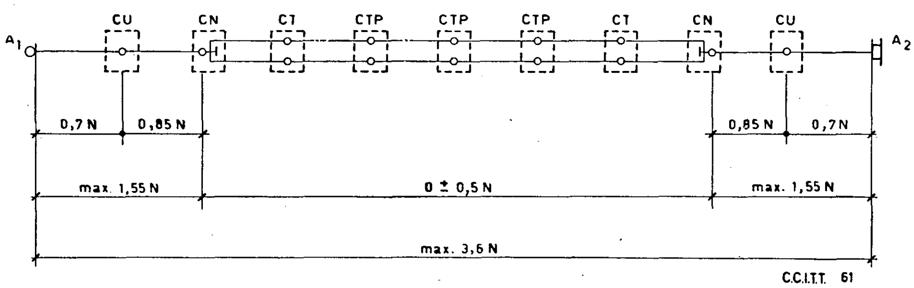
The transmission plan for the national telephone network was devised on the basis of the following principles :

a) the line equivalent for a long-distance call should be not more than 3.6 Np (Figure 1);

b) the relative equivalents of the microphone and the receiver should be 6 and 0 dNp at the present time. Later it is intended to have relative equivalents of 3 dNp and −3 dNp respectively.

c) the call in the nodal exchange is set up as a two-wire call; the call from the nodal exchange to a higher long-distance level is set up on a four-wire connection;

d) four-wire metallic lines will in future (if possible) be connected on a four-wire basis;



CU = trunk exchange CT = transit exchange
CN = " nodal " exchange CTP = main transit exchange

FIGURE 1. — **Reference equivalent between two subscribers (Czechoslovakia)**

The quality of a telephone call is assessed according to the clarity, volume (loudness) and reliability of a connection. The final transmission quality is given by the following conditions :

— the telephone band must be transmitted in the range from 300 to 3400 Hz;
— the psophometric electromotive force of a telephone circuit should not exceed 2 mV at a point of −0.8 Np relative level when measured with a psophometer;
— the feeding-bridges for a local connection should not cause an attenuation of more than 15 cNp.

Notes on the national network transmission plan of Zambia

1. *Introduction*

The Zambian Administration has up to the present been unable to carry out detailed performance tests of subscribers' equipment and lines. In consequence the formulation of the national network transmission plan has been on the basis of information obtained from other administrations and supplied by the manufacturers of equipment. The objective has been to satisfy the requirements of the C.C.I.T.T. Recommendation G.111 (P.11) for a nominal reference equivalent, not exceeding 20.8 dB sending and 12.2 dB receiving, between a subscriber and the four-wire terminals of any international circuits. Also there is a need to provide a terminal loss of at least 3.5 dB between the two-wire point of the national system and the four-wire terminals of the international circuit as up to three national circuits may be transit-switched in such a connection.

2. *Four-wire trunk network*

A new national automatic trunk network, which is currently being installed, will provide four-wire switching at primary and higher order centres. These centres are exclusively interconnected by high velocity multiplex circuits and a nominal transmission loss of 0 dB has been specified for the four-wire connection between any two such centres.

Connections between primary and secondary centres will consist of either high velocity multiplex circuits or within-station cables. The transmission loss between the two-wire point at the primary centre and the four-wire point at the secondary centre has been fixed at 3.5 dB, it being assumed that the loss of within-station cables will be 0 dB. These arrangements will provide :

a) the required 3.5-dB loss between the two-wire points of the national system and the four-wire point of an international circuit;

b) the avoidance of pad switching as the loss will be independent of the number of switching points;

c) 7 dB loss between the two-wire points of the national system which will avoid the necessity for echo suppressors and also, it is hoped, the need for impedance correctors in primary centres.

3. *Primary area network*

With the type of subscribers' instruments in current use for new installations the maximum reference equivalent allowable in the primary area network, to meet the C.C.I.T.T. recommendations for international calls, is determined by the sending equivalent. Due to certain limitations imposed by existing plant and economic considerations it has not been possible to assign a reference equivalent and loss to the local area and the toll circuits independently. However, in order to simplify planning as far as possible primary areas have been classified into three groups, each having a different apportionment of reference equivalent and toll circuit loss.

4. *Overall reference equivalent*

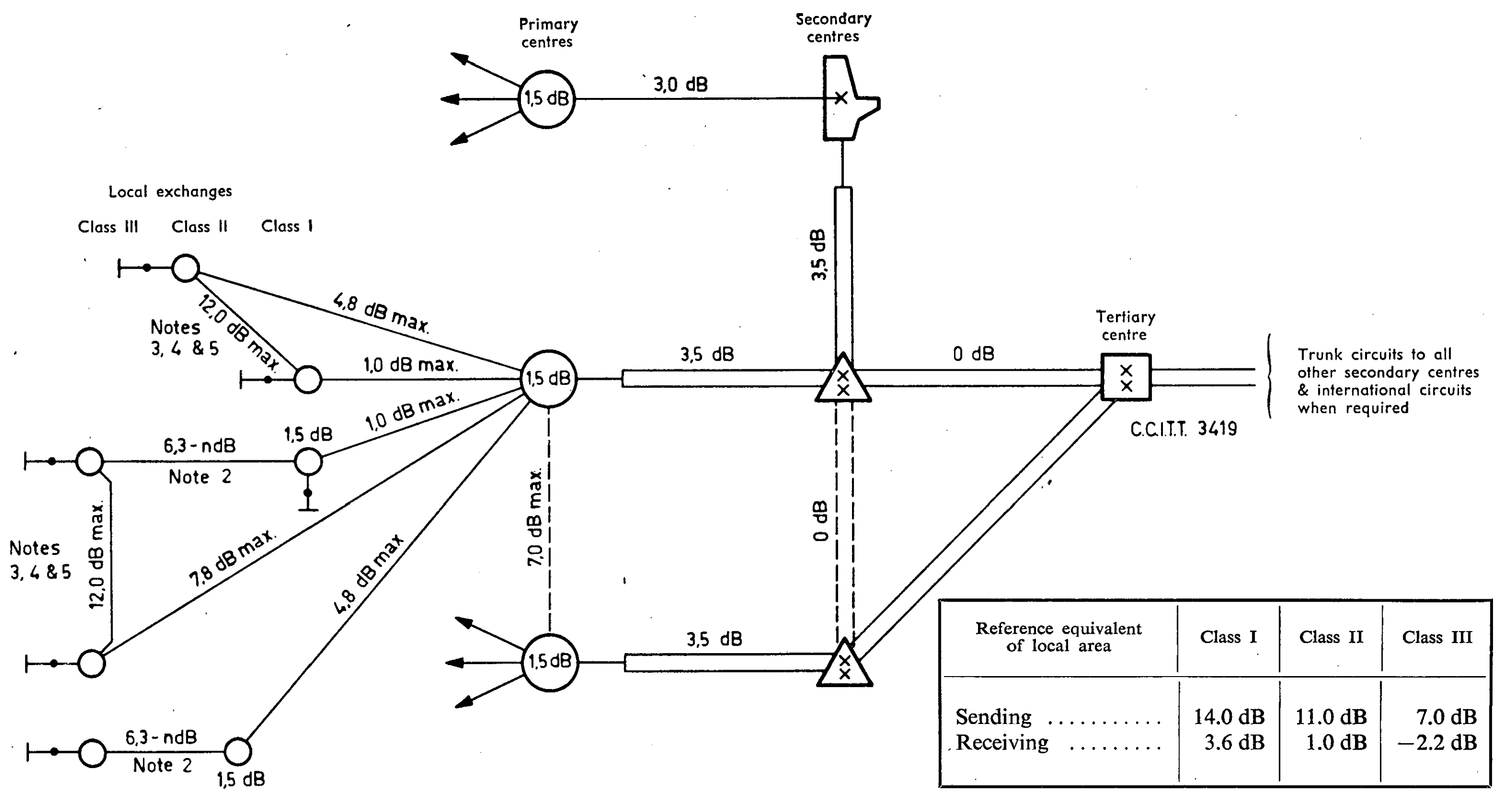
Overall reference equivalents and losses between subscribers in different class primary areas and four-wire switching points are shown in the total of the table below. It follows, therefore, that in the worst case the reference equivalent of a national trunk call will be the sum of the highest totals for sending and receiving, i.e., 31.6 dB.

	CLASS I		CLASS II		CLASS III	
	Send	*Receive*	*Send*	*Receive*	*Send*	*Receive*
Local area reference equivalent	14.0 dB	3.6 dB	11.0 dB	1.0 dB	7.0 dB	−2.2 dB
Toll circuit loss ..	1.0 dB	1.0 dB	4.8 dB	4.8 dB	7.8 dB	7.8 dB
Primary centre switching loss ...	1.5 dB	1.5 dB	1.5 dB	1.5 dB	1.5 dB	1.5 dB
Primary area total	16.5 dB	6.1 dB	17.3 dB	7.3 dB	16.3 dB	7.1 dB
Trunk network loss two wires-four wires	3.5 dB	3.5 dB	3.5 dB	3.5 dB	3.5 dB	3.5 dB
Total ...	20.0 dB	9.6 dB	20.8 dB	10.8 dB	19.8 dB	10.6 dB

5. *Local exchange classification*

5.1 *Class I*

Local exchanges located in the same building as, at least, a primary centre where toll circuits are provided by within-station cables having a loss not exceeding 1 dB. In general these are the larger automatic exchanges where there would be difficulties in meeting a lower reference equivalent for subscribers' equipment and lines in the case of some of the more distant subscribers and particularly in respect of rural party lines. The provision of new subscribers' circuits having a reference equivalent in excess of 11 dB sending and 1 dB receiving (i.e., the limits for class II exchanges) will only be permitted in exceptional circumstances. The provision of pads on very short subscribers' lines may be necessary in these areas.



Reference equivalent of local area	Class I	Class II	Class III
Sending	14.0 dB	11.0 dB	7.0 dB
Receiving	3.6 dB	1.0 dB	−2.2 dB

Notes

1. The reference equivalent of the local area includes limiting subscriber's line instrument and local exchange.
2. n = Actual loss of preceding toll circuit.
3. Loss of unloaded cable circuits measured at 1600 Hz.
4. Toll and junction circuits provided by carrier multiplex circuits, loss adjusted to 3 dB two-wire/two-wire.
5. Junction circuits between any class of local exchange for direct traffic only, i.e. which cannot be used for trunk calls, 12 dB max.
6. $\overset{\times}{\times}$ Four-wire switching point.

National automatic trunk network plan stage I (Zambia)

5.2 *Class II*

Local exchanges remote from primary centres serving either small urban communities some distance from the primary centre or suburban areas in the larger cities and towns.

5.3 *Class III*

Sub-satellite units serving small compact residential areas. At these exchanges the subscribers' line length is limited by signalling considerations and toll circuits are, in the majority of cases, provided by unloaded pairs in normal subscribers' distribution cables. Certain exchanges in remote rural areas will also be included in this category, particularly where unamplified physical lines are utilized for toll circuits.

5.4 *Tandem-connected exchanges*

In certain circumstances small exchanges have access to toll circuits only by tandem switching at a local exchange. This will only be allowed where the local exchange providing the tandem switching is of class I or II and the tandem-connected exchange will be classified as class III. The maximum loss allowable in these circumstances for the tandem circuit will be 6.3 dB minus n, where n is the actual loss of the toll circuit between the local exchange and the primary centre. Thus, with an allowance of 1.5 dB for tandem switching losses the reference equivalent between subscriber and primary centre will be the same as for any other class III exchange area.

6. *Auto-manual trunk switchboards*

It will not be possible, during the initial period of operation of the national automatic trunk network to provide four-wire switching at auto-manual switchboards. These switchboards will be directly connected to secondary centres by trunk circuits having a 3.5-dB loss due to the four-wire-two-wire termination. The maximum reference equivalent therefore for a call on the automatic network established via a trunk switchboard will be increased by 7 dB to 38.6 dB in the worst case.

Additionally, these switchboards will serve as secondary centres for remote primary centres which will not at present have direct access to the national automatic trunk network. Trunk circuits in these cases, which are exclusively high velocity multiplex circuits, will be adjusted to provide 3 dB loss between the switchboard jack and the two-wire point at the primary centre. This additional loss will increase the maximum reference equivalent for calls to or from these centres to 34.6 dB and 37.6 dB between two such centres.

7. *Subscribers' instruments*

The reference equivalents quoted in the foregoing paragraphs are based on the assumed use of modern telephone instruments which are in fact used for all new installations. In cases where transmission complaints result and are found to be due to obsolescent telephone instruments, the Administration's policy is to replace these instruments free of charge.

In calculating the maximum send and receive reference equivalents, a manufacturing tolerance of 1 dB has been allowed in the receiving reference equivalents only. As this would seem to be insufficient and no allowance has been made in respect of send equivalents, it is intended when difficulties are encountered in peripheral areas to remove the automatic regulation devices. This will provide a reduction of 1 dB in the sending equivalent and 0.5 dB in the receiving equivalent for limiting lines at class II exchanges.

SUPPLEMENT No. 8

(formerly Annex 1 of Volume V of the *Red Book*; referred in Recommendations P.20 and P.74)

METHODS USED BY THE BRITISH TELEPHONE ADMINISTRATION FOR RATING TELEPHONE SPEECH LINKS

1. *Introduction*

The British telephone network has been developed with the broad objective of providing most economically a telephone system whose transmission performance is satisfactory to its users at any particular period of time.

However, the degree of satisfaction which a given telephone connection would afford to the population of users cannot readily be determined. Use is therefore made of transmission standards.

2. *Transmission standards*

Combinations of plant (subscribers' sets, lines, etc.) which have been in extensive use and are known to give economical and satisfactory performance are set up in the laboratory; such circuits are termed " transmission standards ", and consist of a specified subscriber's set, subscriber's line and feeding-bridge at each end joined by a 600-ohm non-reactive attenuator to represent the losses and impairments between terminal exchanges.

When new plant, e.g. a subscriber's set, line transmission equipment, etc. is considered, the procedure is to rate typical circuits containing the new plant relative to an appropriate transmission standard. If the assumption is made that circuits which are assessed equal in performance in the laboratory will yield equal satisfaction in service, valid predictions can be made concerning the degree of satisfaction likely to be experienced in the field with the new item.

3. *General assessment method*

For this procedure to be successful it is, clearly, essential that the assessment method used should give equal results for circuits which are equally satisfactory in service. It is very difficult to be sure that this requirement is met and so the British Telephone Administration often makes use of a number of different methods.

The method found most generally applicable, however, is that of collecting opinions expressed by subjects after performing suitable conversational tasks over complete, both-way, telephone circuits. The essential features are :

a) The subjects must be untrained and unconnected, except as subjects, with the speech studies being undertaken.

b) To retain the unbiassed condition of the subjects each person must only be used infrequently, say not more than one experiment every six months.

c) The subjects must base their opinions of the circuit under test on experience in performing a definite task involving conversational co-operation between them. Pictorial puzzles are a suitable medium.

d) The opinions concerning the quality of the call must be expressed on a suitable scale, for example one of five points in which the alternative responses are:

excellent
good
fair
poor
bad

e) It is very desirable that, during an experiment, each subject should experience a wide range of circuit conditions so that practically the whole range of responses is elicited.

f) The observations must be made within the framework of a properly planned experimental design and appropriate statistical analysis applied to the results.

The results of such experiments can be presented in various ways; for example the responses may be scored respectively 4, 3, 2, 1, 0 and mean values calculated. This may be sufficient for many purposes where the rating of one circuit relative to a transmission standard is needed. For other purposes it is instructive to express the percentage of " unsatisfactory " responses, e.g. " poor " or " bad ".

The results of some assessments using this method are shown in Figure 1 expressed both as mean opinion score and percentages " poor or bad " and " bad ". The results relate to typical British telephone sets of type 13.2 P.27 connected by subscribers' artificial lines of average length and two levels of circuit noise and with a room noise of 50 dB are shown. It will be seen that, as a convenient approximation, the percentages within the ranges shown roughly double in value for any 3-dB increase in overall loss.

4. *Simplified assessment methods*

The foregoing assessment method is somewhat lengthy to conduct and several experiments are usually needed before sufficient precision is obtained. A great deal of telephone transmission performance data can be assembled by much simpler methods because the comparisons involved can be confined to those of circuits having similar characteristics. Planning of the local network is carried out by the British Administration on a basis of purely objective tests comparing each circuit with a standard local end containing the same types of microphone and receiver as the local ends under consideration.

The information needed both for the circuit under test and that used as the standard is as follows :

A. Loss :

1) of the electrical circuit of the set,

2) of the subscriber's line and

3) of the feeding-bridge.

The mean value (in dB) of losses measured at 500, 1000, 2000 and 3000 Hz is taken.

A.1 For sending, the loss is measured between the transmitter terminals and a 600-ohm termination on the junction side of the feeding-bridge (see Figure 2). The battery must be connected to the feeding-bridge so that the correct current flows in all the components (including any automatic regulator in bridge or set) but the transmitter capsule should be replaced by a resistor, adjusted at each value of current to be equal to the talking resistance of the transmitter

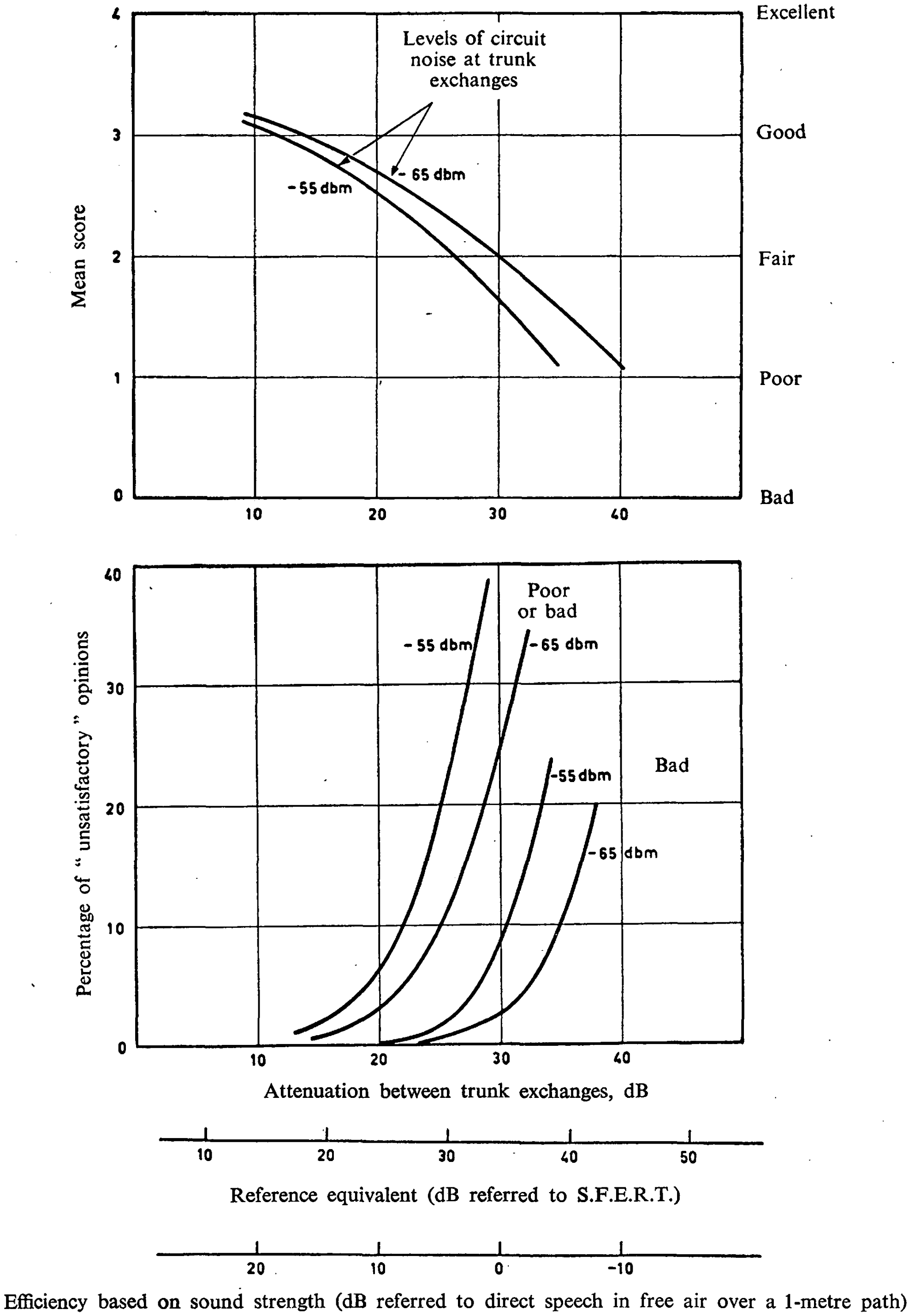


FIGURE 1. — **Mean score given in opinion tests and percentage of " unsatisfactory " opinions in terms of overall circuit loss and noise level**

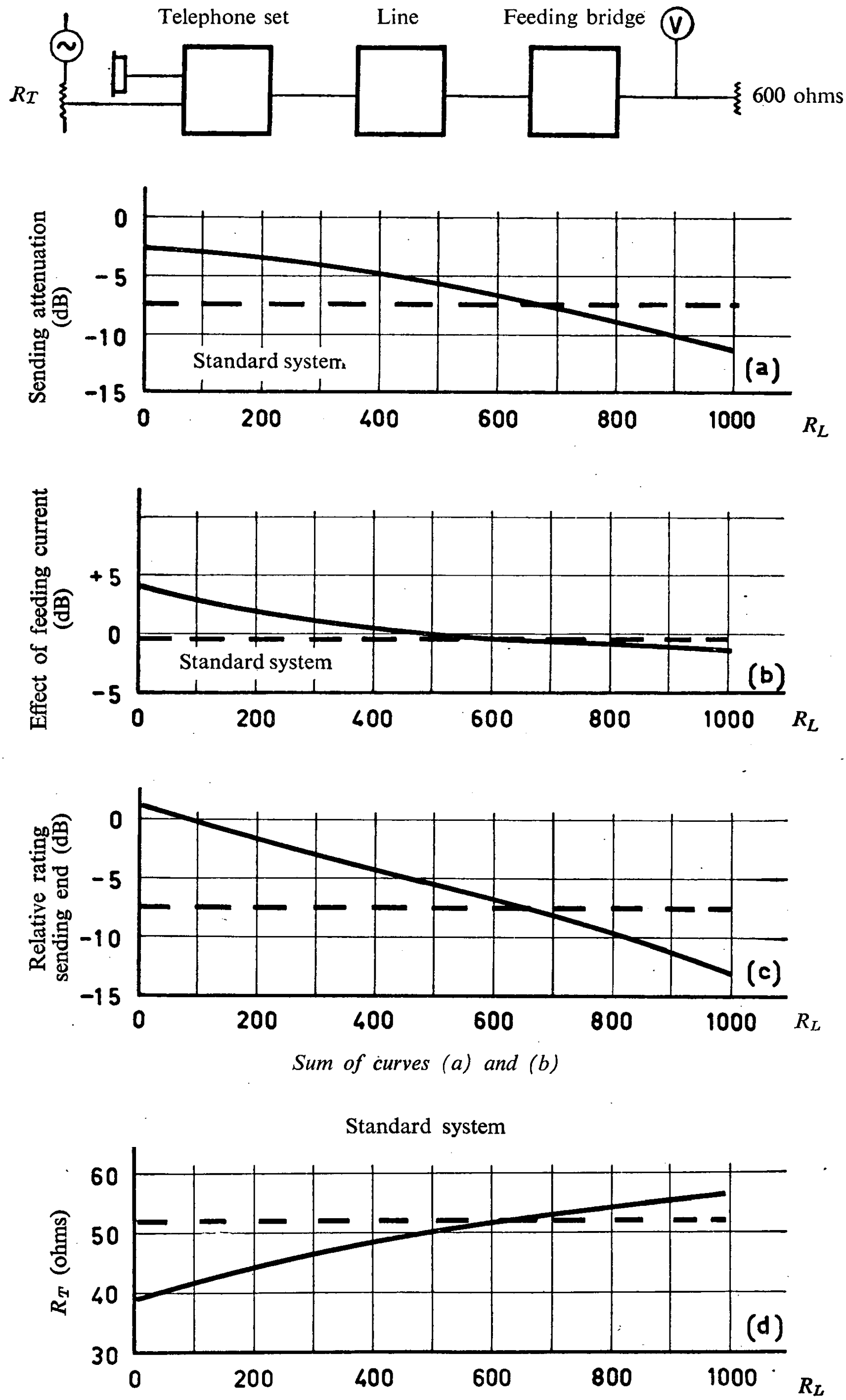


FIGURE 2. — **Calculation of transmission performance rating for local telephone systems by comparison with a standard system using a microphone and a receiver of the same type (sending end)**

capsule at that current, see Figure 2 (d). Arrangements are made to inject from a low impedance source an e.m.f. in series with this resistor to represent the e.m.f. of the transmitter capsule. The potential difference is measured across the function terminals of the feeding-bridge and the loss defined as :

$$20 \log_{10} \frac{\text{potential difference across feeding-bridge 600-ohm termination}}{\text{transmitter e.m.f.}^{1}}$$

A.2 For receiving, the loss is measured between the junction terminals of the feeding-bridge and the receiver (see Figure 3). The battery must, in general, again be connected and the transmitter capsule replaced by a resistor of value appropriate to the current flowing. An e.m.f. in series with 600-ohm resistance is connected to the junction terminals of the feeding-bridge and the potential difference across the receiver terminals measured. To avoid the difficulty of choosing a receiver of average impedance/frequency characteristic and suitably loading it acoustically, it is convenient to replace the receiver by impedance networks designed for each of the four frequencies used. The loss is defined as :

$$20 \log_{10} \frac{\text{receiver potential difference}}{\text{e.m.f./2 in 600-ohm source connected to feeding-bridge}}$$

B. Transmitter feed current effect depending on the supply voltage and the d.c. resistances involved. This is measured as transmitter e.m.f. with speech excitation and conveniently expressed in dB relative to the same measurement at an arbitrary datum feed current.

In Figure 2, curves (a), (b) and (c) show respectively the sending loss, feed current effect and the sum of these as functions of line resistance for British 6½ lb/mile cable (270 ohms/mile loop resistance) associated with a current type telephone set having no automatic regulator.

Figure 3 shows the receiving loss for the same conditions. The dashed horizontal line again represents the same quantity for the standard circuit mentioned above.

Curves of Figures 2 (c) and 3 then give the required transmission performance ratings, sending and receiving, each being relative to an arbitrary datum. The same measurements are also made on a standard circuit giving its performance, sending and receiving, relative to the same datum in either case. The dashed horizontal lines give the respective quantities for a standard circuit used to represent the worst transmission conditions permitted for the local network. The differences between the solid curves 2 (c) and 3 and the respective dashed lines then give the transmission performance ratings relative to the standard chosen.

In the foregoing no account is taken of the effect of sidetone; this is for the following reason :

With modern-type sets under normal conditions with a 600-ohm termination on the junction side of the feeding bridge, sidetone level is sufficiently low to have a negligible effect upon

[1] Microphone e.m.f. relative to the e.m.f. for a feeding current of 50 mA, measured for speech excitation.

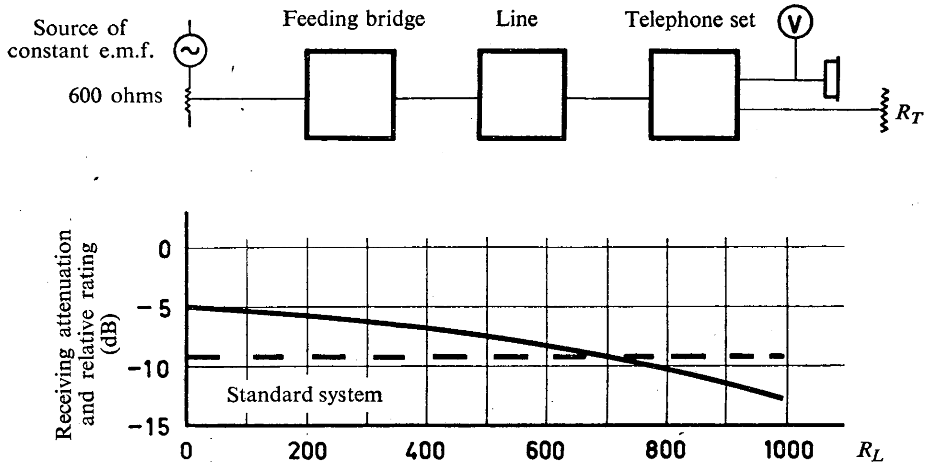


FIGURE 3. — **Calculation of transmission performance rating for local telephone systems by comparison with a standard system using a microphone and a receiver of the same type (receiving end)**

R_T = Microphone resistance during speech

R_L = Resistance of the subscriber's line (ohms)

performance based on subjects' opinions during conversation. A small effect upon speech volume is observed, but for the purpose of local planning this is not taken into account in these simple calculations. Under normal ambient conditions sidetone begins to become troublesome only when the speech sidetone path is more than 25 dB more sensitive than a metre air path.

When the sensitivities of subscribers' sets are increased to allow higher-resistance subscribers' lines to be used, sidetone inevitably becomes worse under short line conditions because the sidetone balance return loss cannot be correspondingly improved. In the British network such sets are provided with automatic regulators so that the high sending and receiving sensitivities are removed under short line conditions. This restores sidetone to about the level for a set of normal sensitivity.

Nevertheless exceptional circumstances can occur where sidetone tends to become excessive, e.g. when particularly unfavourable terminations occur; these cases are, however, treated individually on their merits by taking appropriate remedial measures, and are not taken into account in the general planning of local networks.

It is again emphasized that these simplified methods are only applicable for comparisons between circuits containing the same types of microphone and receiver as the standard. When different types are involved the more elaborate general assessment method is necessary.

5. *Application of the various assessment methods*

Transmission planning data on an extensive scale can be provided by judicious use of (i) the general methods described in section 3 and (ii) the simplified method given in section 4. The former is necessary whenever a new subscriber's set is introduced having a different transmitter, receiver or handset. Only a single line condition (or perhaps two) would be used and very extensive tests would be conducted extending over several months. The line condition used would be that judged from preliminary tests to be near the transmission limit (say, that used as the design objective in developing the new set). These experiments would yield a reliable estimate of the performance of the new set relative to a standard circuit containing an older type set extensively installed in service.

The effects of varying the subscriber's line and feeding bridge, inclusion of P.B.X. circuits and modifications to the electrical properties of the new set would then be treated by the simplified method. Very many such measurements and calculations can be conducted in only a few days.

References

D. L. Richards : Some aspects of the behaviour of telephone users as affected by the physical properties of the circuit (Communication Theory, Willis Jackson, Ed.), p. 442 (Butterworths Scientific Publications, 1953).

J. Swaffield and D. L. Richards : Rating of speech links and performance of telephone networks; *Proc. I.E.E.*, 106 Part B, p. 65, 1959.

D. L. Richards and J. Swaffield : Assessment of speech communication links, ibid., p. 77.

F. E. Williams and F. A. Wilson : Design of an automatic sensitivity control for a new subscriber's telephone set; *Proc. I.E.E.*, 106 Part B, p. 361, 1959.

SUPPLEMENT No. 9

(former Annex 5; referred to in Recommendation P.41)

ABSOLUTE CALIBRATION OF THE A.R.A.E.N. AT THE C.C.I.T.T. LABORATORY

1. *General principles of electro-acoustic calibration*

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

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

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

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

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

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

Note.— Research Report No. 13200 of the British Telephone Administration contains some theoretical notes on these calibrations and the results of the first calibrations made after the installation of the A.R.A.E.N. at the C.C.I.F. Laboratories in Geneva.

2. *Theory of calibration of microphones with a stationary wave tube and a Rayleigh disk* [1]

2.1 *Theory of the Rayleigh disk*

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

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

where ρ is the specific gravity of the fluid,

d the diameter of the disk,

V the velocity of the fluid,

θ the angle between the direction of displacement of the fluid and a normal to the disk.

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

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

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

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

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

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

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

From equation (3)

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

[1] According to W. West, *Acoustical Engineering* (Pitman & Sons, London, 1932).

Observing the displacement δ at a distance l of a light beam reflected by the Rayleigh disk used as a mirror, we have $\delta = 2\,\varphi\,l$

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

The moment of torsion K is determined in advance for the suspension thread by suspending, at the end to which is afterwards fixed the Rayleigh disk, a disk of known moment of inertia J about its vertical axis. The period T and the logarithmic decrement D of the free oscillations of this disk are observed, and we have

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

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

2.2 *Use of a stationary wave resonance tube*

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

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

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

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

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

With the equipment and tube used at the C.C.I.T.T. Laboratory this upper limit occurs at about 6500 to 7000 Hz.

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

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

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

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

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

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

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

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

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

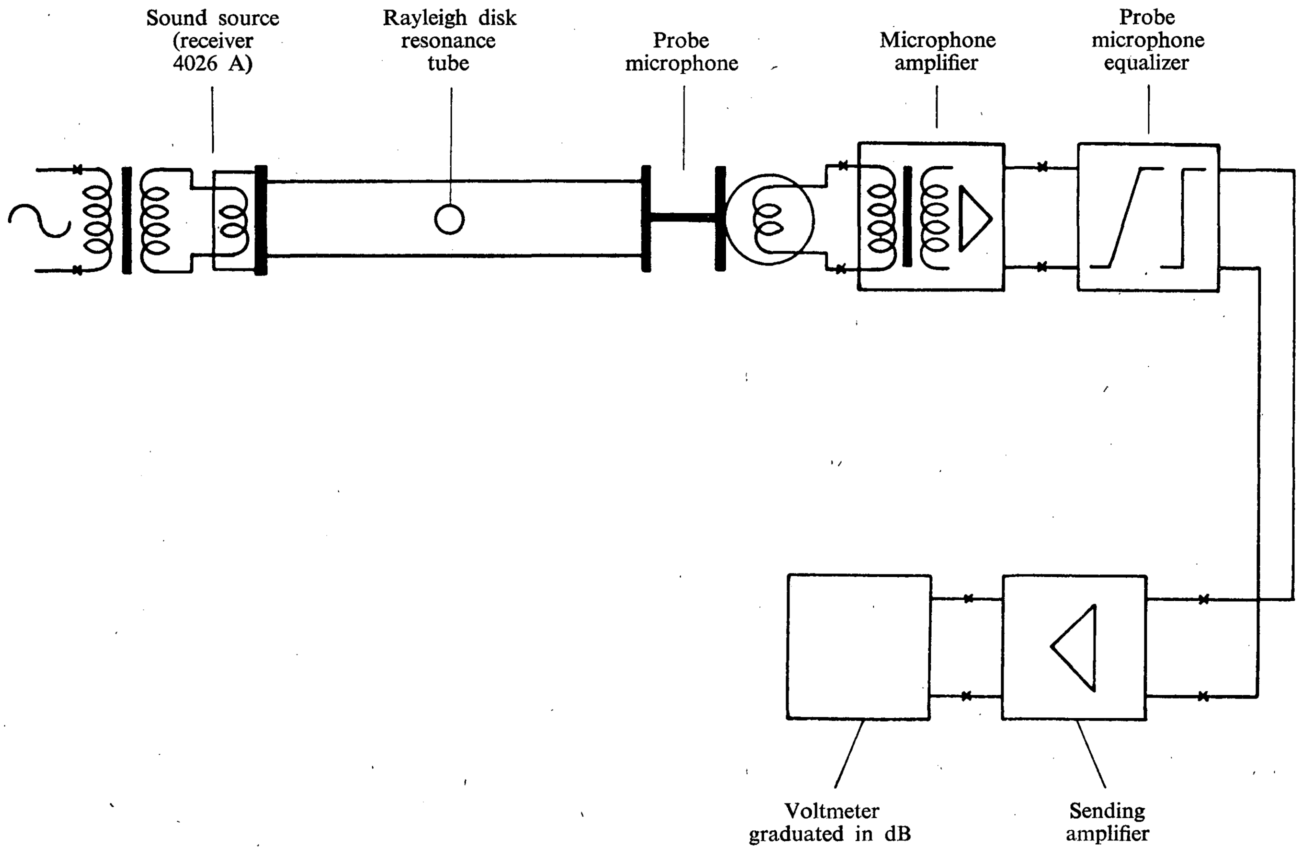
3.1 *Measurement of output voltage of the probe microphone when a pressure P is applied* (see Figure 1)

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

For example if, for a given disk, we have the following relationship between the acoustic pressure and the deflection of the spot : $P = 12.7 \sqrt{\delta}$ dynes/cm^2, taking a value of pressure P of 50 dynes/cm^2, i.e. a value convenient for calculation, the value of 15.5 cm is obtained for δ. This deflection of 15.5 cm for the spot with respect to the zero of the graduated scale is called " normal deflection " and is used for measurements made with this disk.

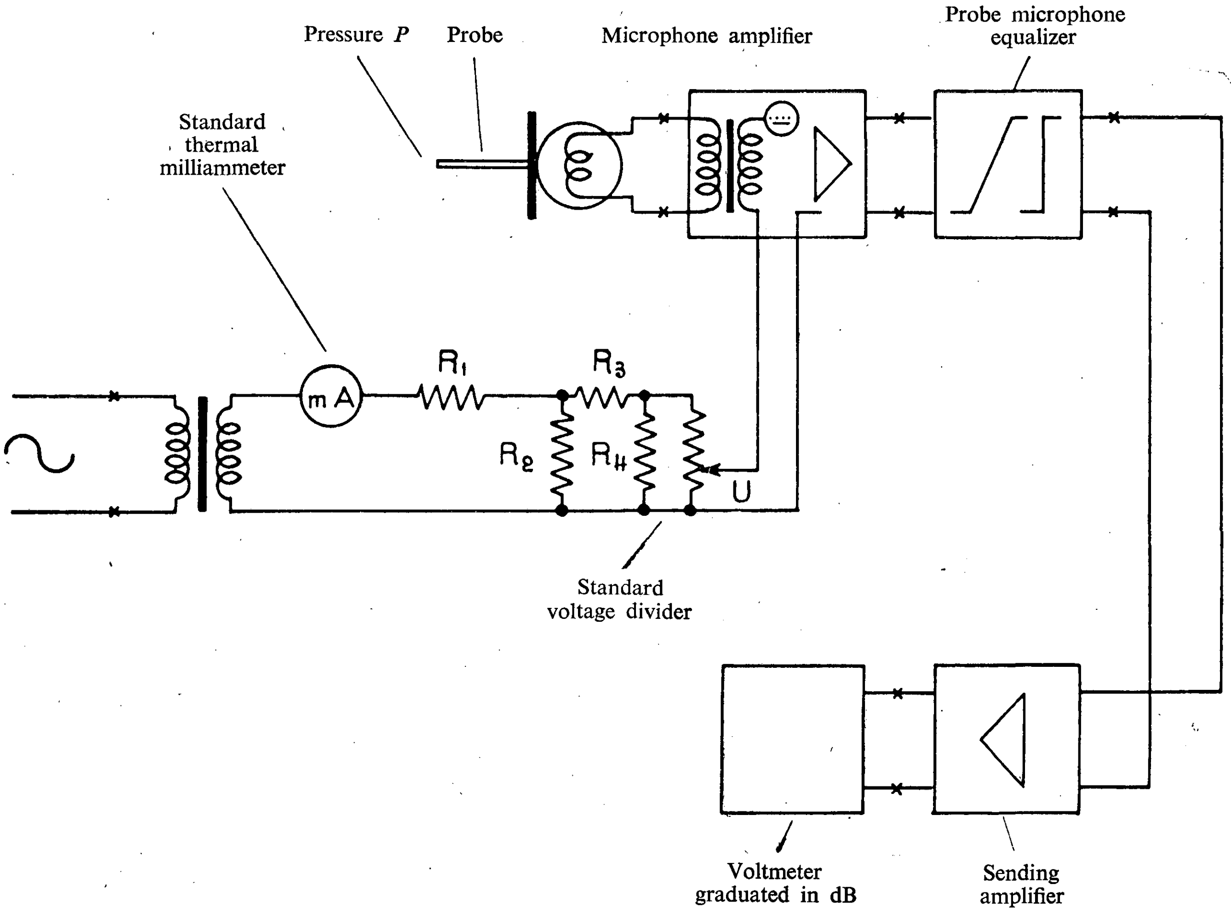
3.2 *Determination of the voltage corresponding to the deflection " x " on the voltmeter* (see Figure 2)

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



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

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



Injection of a known voltage U to obtain the same output voltage as in the previous position (Figure 1) (probe microphone sensitivity = U/P)

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

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

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

The measurements are repeated for odd multiples of the fundamental resonance frequency (for which the length of the tube is equal to one-half wavelength). If the minimum length has been chosen for the stationary wave tube, the fundamental frequency is 154 Hz (the example in the table). The measurement frequencies will be about 450, 750 . . . up to 7000 Hz. To obtain a greater number of results at the low frequencies it is necessary to lengthen the tube to its maximum extent to obtain integral multiples of the fundamental resonance frequency, which in this case is about 80 Hz.

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

4.1 *General*

This measurement is essentially designed to verify the stability of the moving coil microphones of the A.R.A.E.N. Research Report No. 13200 of the British Administration gives a method of calculating the absolute sensitivity of the microphone *under the particular conditions of measurement.*

The microphone to be calibrated is placed on a closed coupler formed by a small cylindrical cavity closed at one end by a moving coil receiver which serves as a sound source. At the other end is placed the microphone to be calibrated. The tip of a probe microphone is introduced into this cavity via an orifice provided and used to measure the sound pressure at a fixed point; the position of this is precisely defined with respect to its proximity to the diaphragm of the microphone to be calibrated. (Figure 3 shows the position of the microphones and receiver in the closed coupler.)

4.2 *Check of the voltage applied to the receiver used as a sound source* (see Figure 4)

In this measuring condition, the voltmeter indicates the voltage at the receiver terminals. It is not strictly necessary with a linear receiver to set up this value, but it allows the measuring conditions to be specified and reproduced. Generally a voltage of -15 dB relative to 1 volt is adopted at a frequency of 1000 Hz.

4.3 *Measurement of the output voltage of the microphone* (see Figure 5)

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

4.4 *Measurement of the output voltage of the A.R.A.E.N. microphone* (see Figure 6)

In this measuring position, the attenuator (which varies between 0 and 100 dB) is adjusted until the voltmeter indicates exactly the same deflection " y ".

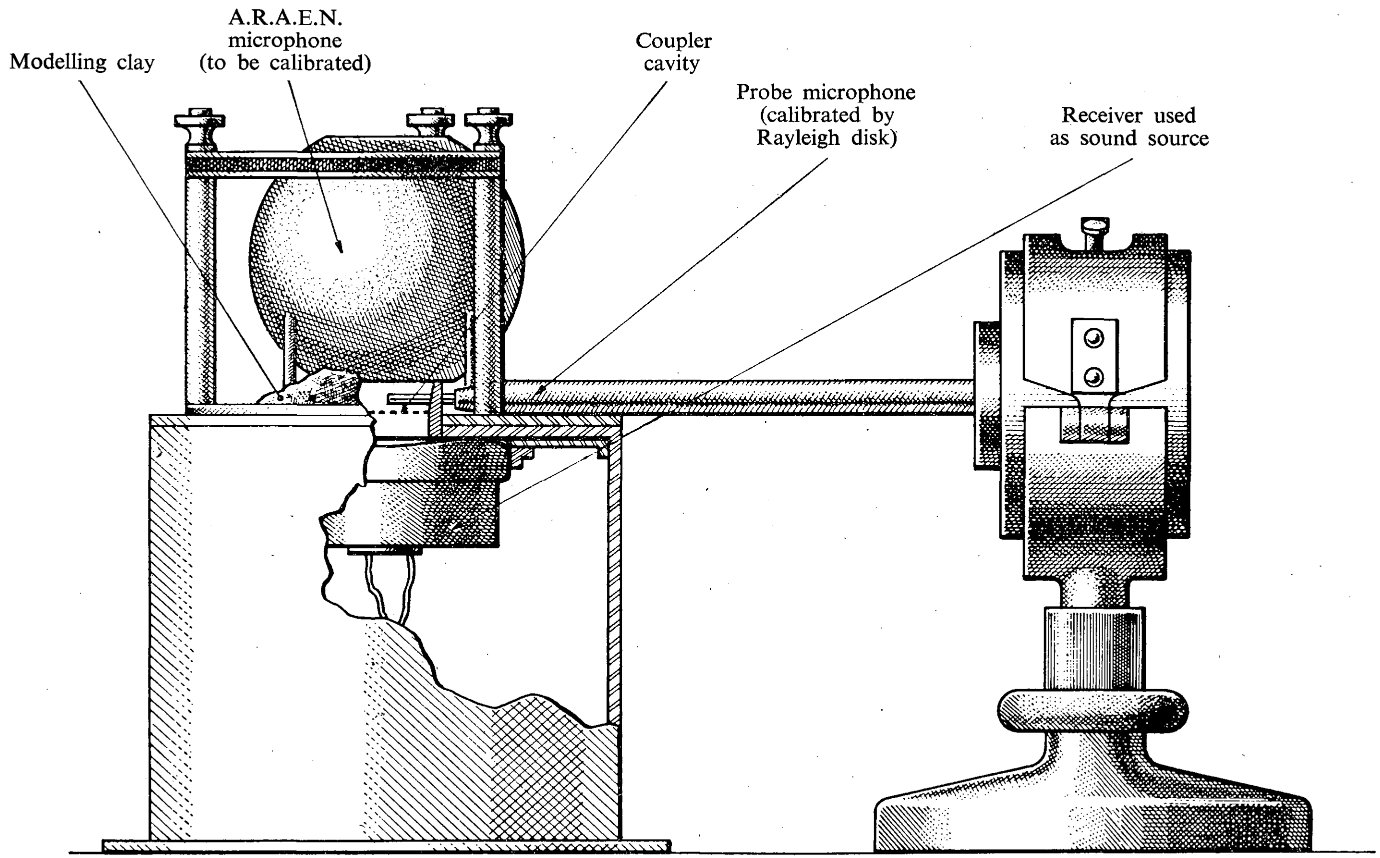
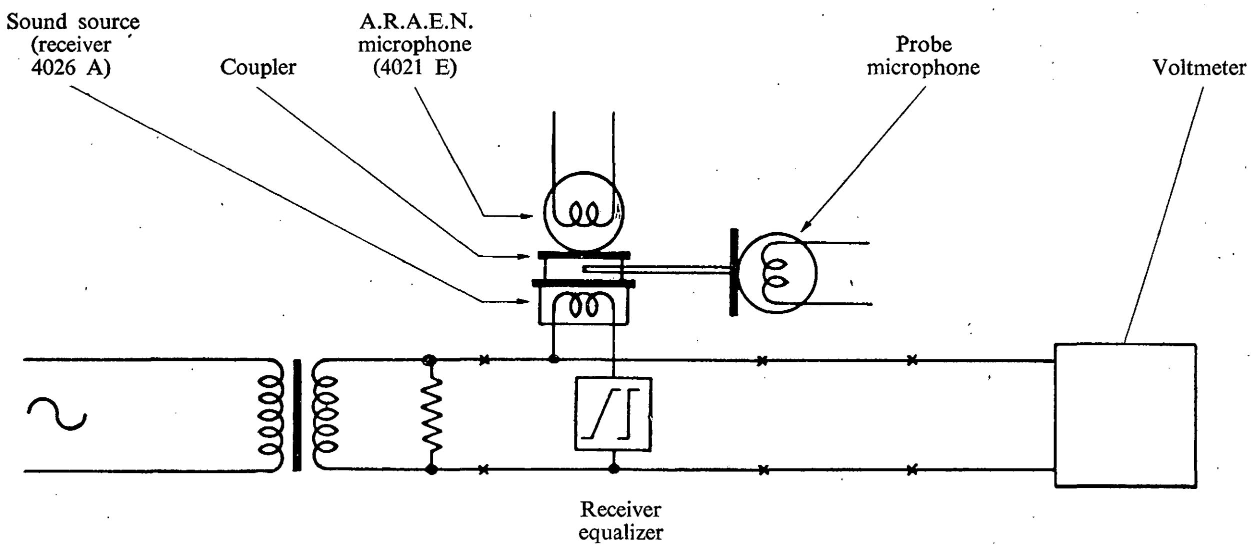
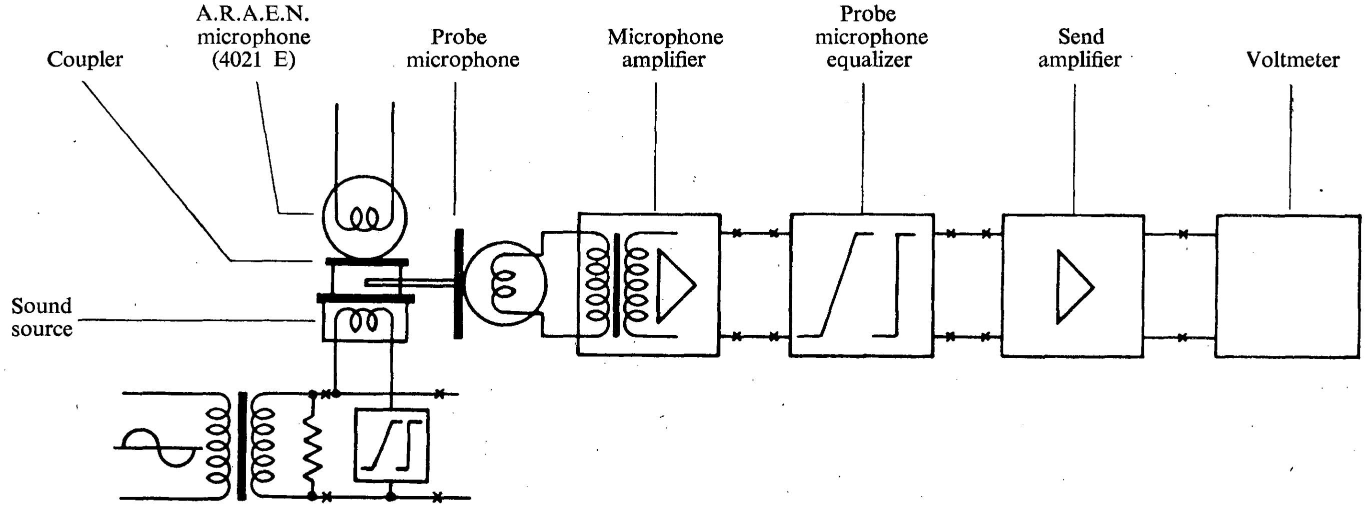


FIGURE 3. — **Closed coupler for calibration of moving coil microphones**



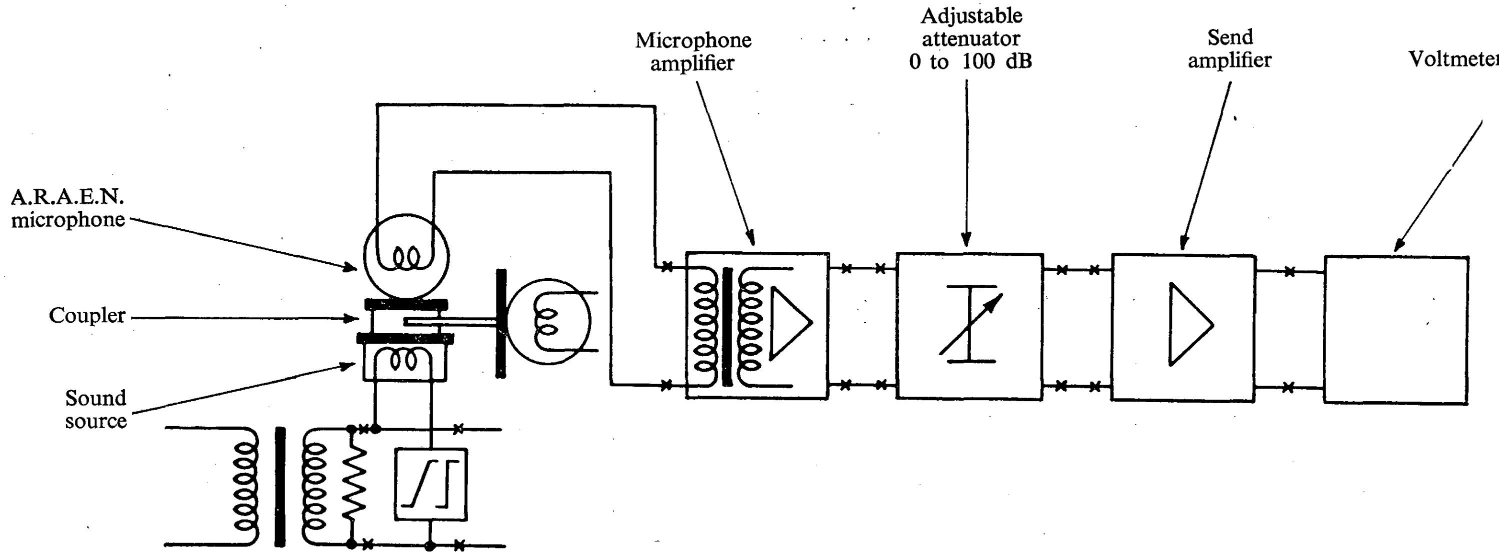
Check of the voltage applied to the receiver used as the sound source

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



Measurement of the output voltage of the probe microphone

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



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

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

If the sensitivity of the probe microphone, measured as indicated in section 3, is added to the value which is read on the attenuator a magnitude is obtained which is a function of the sensitivity of the A.R.A.E.N. microphone *under these particular conditions of measurement*, which must remain constant throughout all the series of measurements and which also serves as a criterion of stability of the A.R.A.E.N. microphone (see also section 6).

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

5.1 *General*

The calibration of an A.R.A.E.N. electrodynamic receiver is obtained by placing the receiver on an artificial ear the acoustical impedance of which represents approximately the mean value of those obtained on human ears. This artificial ear consists essentially of a cylindrical cavity

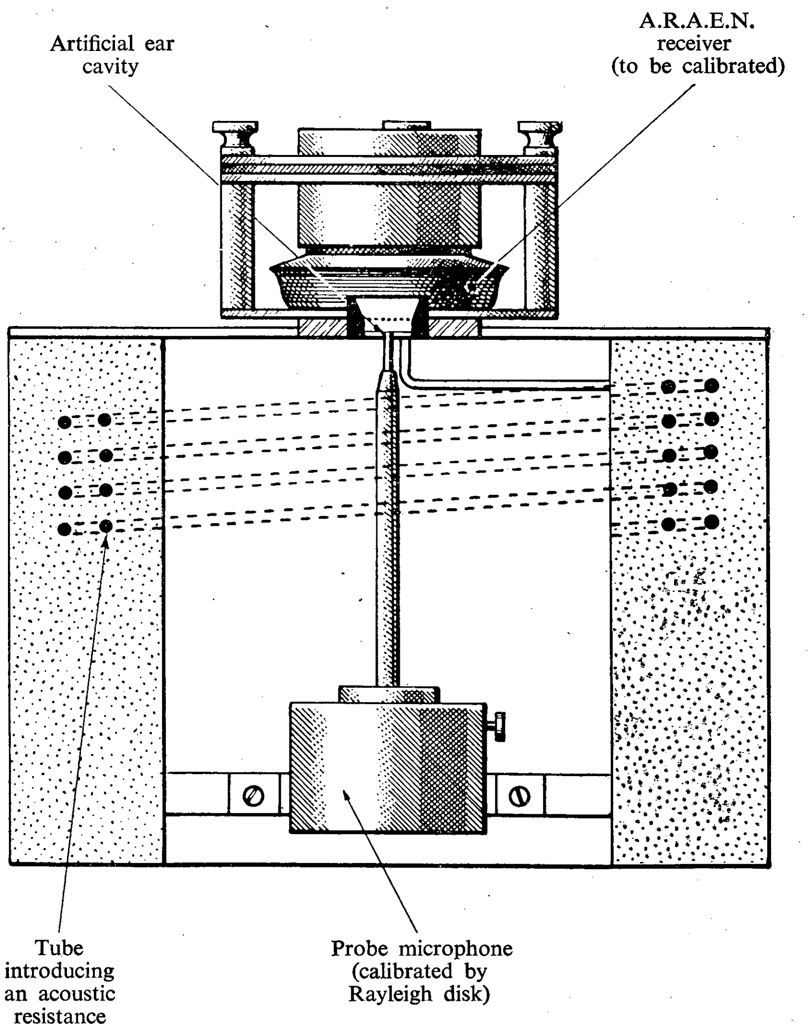


FIGURE 7. — **Artificial ear for the calibration of moving coil receivers**

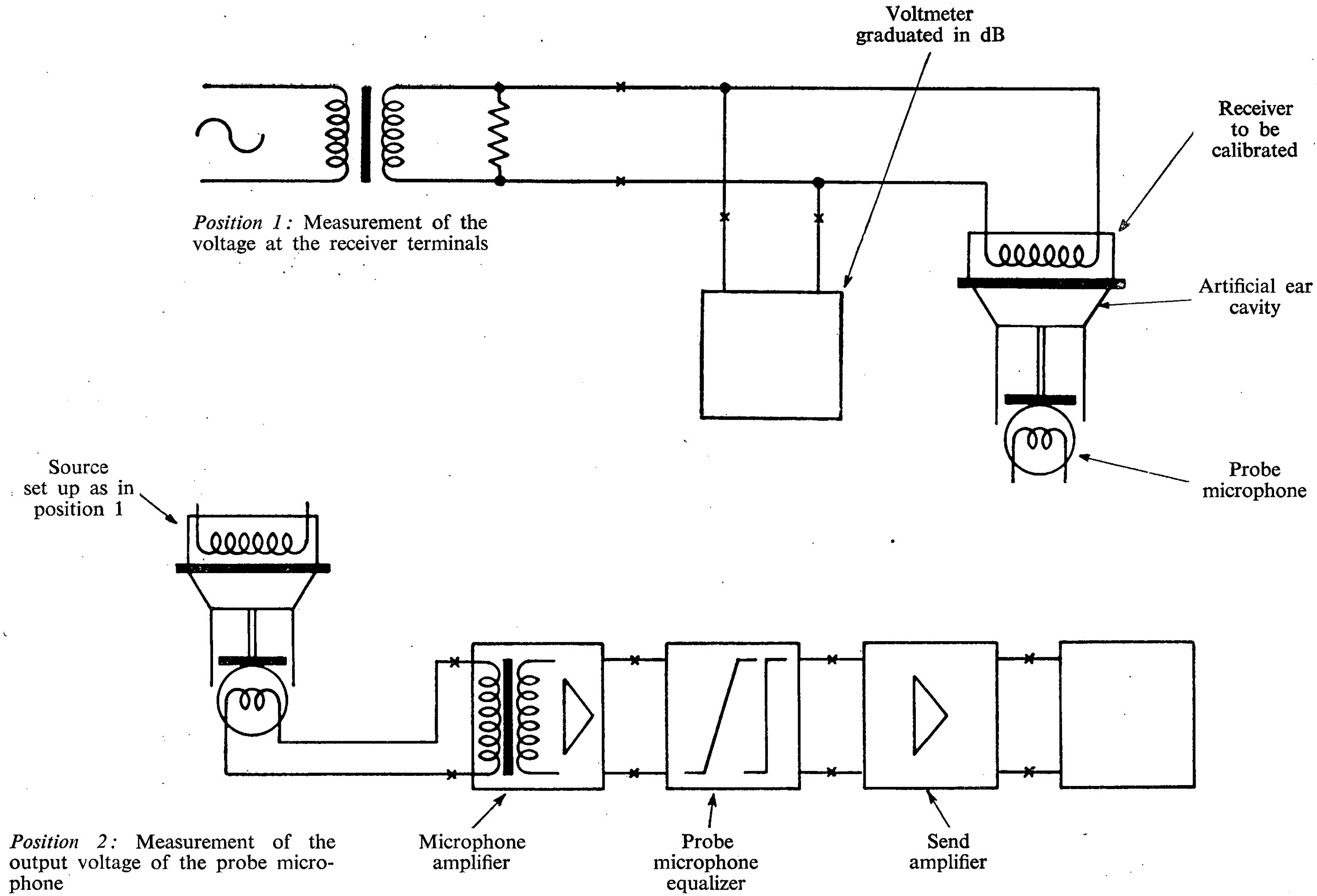


FIGURE 8. — **Calibration of an A.R.A.E.N. receiver**

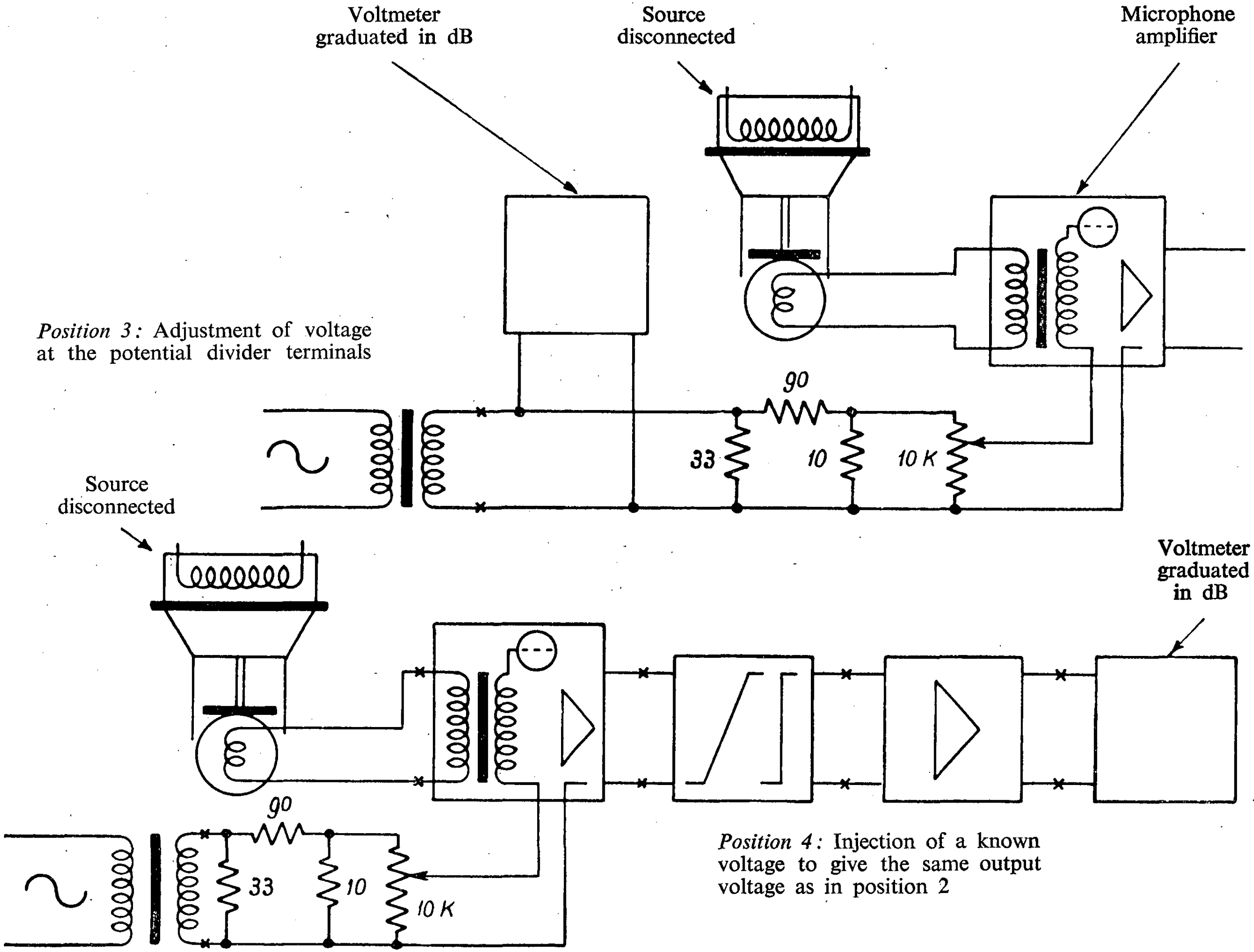


FIGURE 8 (continued). — **Calibration of an A.R.A.E.N. receiver** of 3 cm^3 volume terminated at the lower end by a spiralled tube representing the acoustical impedance (see the description of this artificial ear in Annex 11 of Volume V of the *Red Book*).

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

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

5.2 *Measurement of the voltage applied to the terminals of the receiver to be calibrated* (see Figure 8, position 1)

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

5.3 *Measurement of the voltage at the output of the system formed by the probe microphone and its amplifier* (see Figure 8, position 2).

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

5.4 *Adjustment of the injection voltage which is substituted for the e.m.f. produced by the probe microphone* (see Figure 8, position 3)

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

5.5 *Adjustment of the injection voltage applied to the probe microphone circuit* (see Figure 8, position 4)

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

Measurements are generally made at a series of increasing frequencies starting from 80 Hz and going up to 7000 Hz.

5.6 *Interpretation and discussion of results*

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

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

Note.— In practice, when the input voltage to the potential divider is adjusted, it is advantageous for precision in reading the voltmeter to use a level of −15 dB because this indication is at the centre of the scale whilst the −20 dB division is not very clear on account of the curved scale shape. In this case it is necessary to apply a corresponding correction to the sensitivity value of the receiver.

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

Measurements described in Research Report No. 13180 of the British Administration have shown that a speech pressure of 1 dyne/cm^2 exists at a point 13.25 inches from the lips of a talker speaking at " reference vocal level " for the A.R.A.E.N. (for definition of the term " speech pressure " see the report mentioned). The A.R.A.E.N. is provided with a voltmeter for the purpose of controlling the vocal level of the talking operator. This voltmeter is connected across the input of the A.R.A.E.N. junction and the over-all gain is so adjusted that when an operator speaks at the reference vocal level for the A.R.A.E.N. the speech voltage at the input to the junction is 1 volt. The absolute sensitivity of each of the microphones used with the A.R.A.E.N. varies slightly; consequently when one microphone is changed for another the gain has to be adjusted accordingly.

The sensitivity/frequency of the microphones (measured in a free acoustic field) is approximately flat with a 5-dB peak at 450 Hz but the equalizer network placed after the microphone amplifier largely compensates for this. After allowing for this compensation there is a variation of the order of ± 0.5 dB in the frequency range 100 to 1000 Hz and a slightly larger variation outside of this range. To take full account of these variations, it is necessary to take some mean value for the sensitivity; this value represents the microphone sensitivity over the speech frequency band which has the largest energy content as it is this band which determines the voltmeter reading. A study of the spectrum of the human voice shows that the greatest power is in the range 100 to 900 Hz. Consequently it is in this frequency band that the sensitivity of the microphone should be considered. As the sensitivity/frequency characteristic of the microphone plus equalizer is substantially flat in this region it is reasonable to take the arithmetic mean of the sensitivities of the microphone at 100, 300 and 900 Hz. This mean figure can then be used to calculate the gain required at the sending end of the A.R.A.E.N. to give 1 speech volt at the input of the junction when a speech pressure of 1 dyne/cm^2 is applied to the microphone. The normal gain of the microphone amplifier plus the gain of the send amplifier (the gain being defined as

$$20 \log_{10} \frac{\text{output voltage across a 600-ohm termination}}{\text{input e.m.f. from a 20-ohm generator source}}$$

is equal to 89 decibels, and if this gain is compared with the gain necessary for the specific microphone, the setting of the send amplifier can be obtained. (The controls for this adjustment are calibrated in dB relative to normal gain.)

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

1. at each of the frequencies of 100, 300 and 900 Hz, subtract the value of the insertion loss of the microphone equalizer from the value of the free field sensitivity of the microphone in question;

2. take the mean of the three values of corrected sensitivity so determined;

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

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

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

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

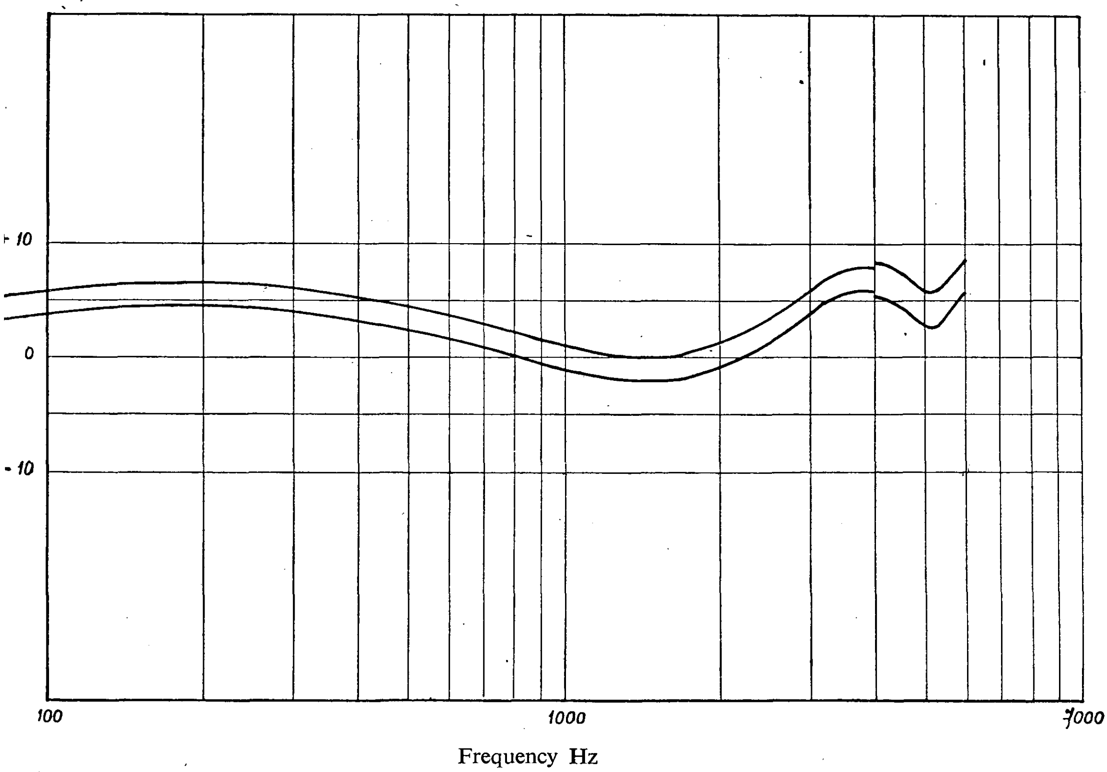


FIGURE 9. — **Curves giving the limits between which the " sensitivity/frequency " of the A.R.A.E.N. receivers, measured under the conditions described in section 5, must lie. Receivers No. 4026 A**

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

The receiving system of the A.R.A.E.N. is terminated by four receivers of the same type (Standard Telephone and Cable, type 4026 A) connected in series. Their relative sensitivity/frequency characteristics and absolute sensitivities must be kept within close limits. Furthermore the *gain of the receive amplifier is adjusted to a fixed value of 23 dB.* The total attenuation of the electrical circuit of the receiving system of the A.R.A.E.N. between the input terminals of the receive attenuation equalizer and the terminals of any one receiver is 19.5 dB at 1000 Hz and takes account of the supplementary attenuation of 12 dB due to the division by 4 of the output p.d. of the receiver impedance matching transformer (the four receivers are in series).

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

If the sensitivity/frequency characteristics are within the fixed limits, the sensitivity of a single receiver only needs to be considered, and the mean of its sensitivities at 100, 300, 1000 and 2000 Hz must be determined. This mean value must be equal to $+43.7$ dB relative to 1 dyne/cm^2 per volt. In order to obtain this value so that all the receivers have the same sensitivity special attenuators, variable in steps of 0.25 dB, are matched to each receiver.

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

The theoretical sensitivity of the A.R.A.E.N. is defined in section D of Recommendation P.41.

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

References

W. Konig : *Annalen der Physik*, Vol. 43, p. 43 (1891).

E. J. Barnes and W. West : *J.I.E.E.*, Vol. 65, p. 871 (1927).

R. A. Scott : *Proc. Roy. Soc. A.*, Vol. 183, p. 296 (1945).

W. West : *Proc. Phys. Soc. B*, Vol. 62, p. 437 (1949).

SUPPLEMENT No. 10

(former Annex 19; referred to in Recommendations P.42, P.45 and P.52)

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

This equipment complies with the S.F.E.R.T. volume indicator (Annex 18 of Volume V of the *Red Book*) except so far as integration time, time to return to zero, and reference point for calibration are concerned. The integration time and time return to zero specified by the C.C.I.T.T. are 200 milliseconds, whilst for the A.R.A.E.N. volume meter these are approximately 100 milliseconds. Experience gained with this volume meter, having a shorter integration time, has shown that it is quite satisfactory for controlling the speech level in articulation tests. If it is necessary to have the longer integration time specified by the C.C.I.T.T. it is possible to change the milliammeters used with this equipment.

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

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

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

Calibration. — The equipment is calibrated to indicate the number of decibels relative to 1 volt at 1000 Hz. The sensitivity controls enable 0 dB on the indicating instrument to be obtained for voltage levels in the range from -50 to $+30$ decibels relative to 1 volt at 1000 Hz.

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

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

c) *To determine the talker's speech power in the reference equivalent measurements.* — The volume meter is connected at the output of the N.O.S.F.E.R. sending system. The sensitivity controls of the volume meter are then adjusted so that the zero mark corresponds to the " normal volume for voice-ear measurements ". The talker then repeats the conventional phrase Paris, Bordeaux, Le Mans, Saint-Leu, Léon, Loudun with a speech level such that the needle reaches the reference mark (0 dB).

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

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

40 to 12 000 Hz	$\pm$ 2 dB
150 to 6 000 Hz	$\pm$ 0.5 dB

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

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

SUPPLEMENT No. 11

(former Annex 20; referred to in Recommendations P.42 and P.45)

VOLUME METER STANDARDIZED IN THE UNITED STATES OF AMERICA, TERMED VU METER

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

Introduction

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

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

General

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

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

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

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

Definitions

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

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

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

" *Volume unit* " (vu, customarily written in small letters). — This expression is used to denote the numerical value of the volume. The volume in " volume units " is numerically equal to the relative magnitude of the waves considered, expressed in decibels above or below the " American reference volume " [1], defined below.

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

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

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

Specification of the American volume meter

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

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

a) A meter,

b) An attenuator.

Dynamic characteristics. — If a sinusoidal voltage between 35 and 10 000 Hz, of such amplitude as to give reference deflection under steady-state conditions is suddenly applied, the meter pointer shall reach 99 % of reference deflection in 0.3 second ± 10 %, and shall then overswing reference deflection by at least 1 % and not more than 1.5 %. The time required for the meter pointer to reach its position of rest on the removal of the sinusoidal voltage shall not be greatly different from the time of response.

Response-versus-frequency characteristic. — The response of the volume meter shall not depart from that at 1 000 Hz by more than 0.2 decibel between 35 and 10 000 Hz nor more than 0.5 decibel between 25 and 16 000 Hz.

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

Reversibility. — The response when measuring unsymmetrical waves must be independent

[1] It will be noted that the " American reference volume " differs from the " reference volume " defined by the C.C.I.T.T.

of the poling of the volume meter. Such a characteristic may be obtained by the use of a direct-current meter in conjunction with a full-wave rectifier.

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

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

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

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

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

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

Correction (to be added algebraically) in vu =

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

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

Good engineering practice

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

Impedance. — The volume meter is normally used as a bridging instrument and when so used its impedance must be sufficiently high so as not to influence unduly the waves in the circuit with which it is used. It is good practice to make the value of impedance not less than 7500 ohms for use on a 600-ohm circuit.

Harmonic distortion. — The root-mean-square value of the harmonic distortion produced when the volume meter is bridged across a resistive circuit impedance through which a sinusoidal wave between 25 and 8000 Hz is being transmitted should not exceed 0,2% of the fundamental.

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

Scale marking. — The point of reference deflection should be located within a sector between $^2/_3$ and $^3/_4$ of full scale. In addition to the vu scale, a 0-to-100 scale proportioned to voltage should be provided, the 100 point coinciding with reference deflection. Samples of the two types of scales in general use are shown in Figures 1 and 2.

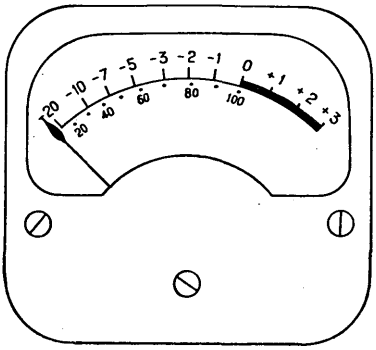


FIGURE 1

This scale emphasizes the graduation in vu.

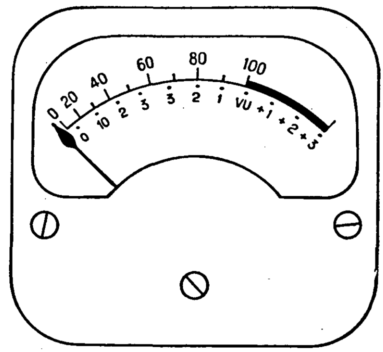


FIGURE 2

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

SUPPLEMENT No. 12

(former Annex 21; referred to in Recommendation P.52)

MODULATION METER USED BY THE BRITISH BROADCASTING CORPORATION

Description

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

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

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

Circuit

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

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

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

Performances and tests

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

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

Use

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

SUPPLEMENT No. 13

(former Annex 22; referred to in Recommendation P.52)

MAXIMUM AMPLITUDE INDICATORS, TYPES U 21 AND U 71 USED IN THE FEDERAL GERMAN REPUBLIC

Application

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

General characteristics

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

Specified values

Frequency range	30 to 15 000 Hz
Input impedance	greater than 30 000 ohms
Dynamic range	1 : 300
corresponding to	0.3 to 100%
(an additional range from 100% to 180% for use as an overload range is marked in red)	
Input voltage for 100% (corresponding to 8 or 32 milliwatts in 600 ohms)	2.2 or 4.4 volts r.m.s.[1] switchable
Variation of reading with frequency, at 100%	less than ±10% relative to the value at 800 Hz
Rectifier characteristic	linear

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

Integration time for sudden application of a sinusoidal voltage at 100%	
to reach 80%	about 5 milliseconds
to reach 90%	about 10 milliseconds
Rise time	
to reach 90%	about 80 milliseconds
Time to return from 100% to 10% on sudden disconnection of a sinusoidal voltage	about 1 or 2 seconds switchable
Overshoot	maximum 10%
Power supply	40 to 60 Hz, 110 or 220 volts, about 55 watts.

SUPPLEMENT No. 14

(former Annex 23; referred to in Recommendations P.42 and P.52)

COMPARISON OF THE READINGS GIVEN ON CONVERSATIONAL SPEECH BY DIFFERENT TYPES OF VOLUME METER

(Tests carried out by the British Administration)

1. *Specification of a conventional volume meter*

A reading taken using a conventional a.c. rectifier-voltmeter type of volume meter can be defined by specifying the following :

1. The reference level or voltage of pure tone (say 1 kHz) which defines the " zero dB " mark.

2. The integration time, defined as the duration of a constant amplitude pulse of pure tone which will cause the needle to rise to a value which is 63% of the angular deflection which would have been obtained if the same amplitude had been applied indefinitely. Alternatively the 99% (or 90%) rise time may be specified; this is the time which must elapse after application of a constant level of pure tone before the needle has risen to 99% (or 90%) of the final angular deflection. The relationship between integration time and rise time is somewhat complicated because it is a function also of overswing.

3. Overswing; unless the meter is critically damped or overdamped it will rise to a maximum value higher than the final value.

4. Law of rectification; the relationship between the instantaneous output from the rectifier and its instantaneous input can be expressed, for a single polarity, in the form :

$$\text{output} = \text{constant} \times (\text{input})^{k}.$$

Full-wave rectification is essential because speech waveforms are frequently unsymmetrical.

5. Method of interpreting the deflections.

2. *Effect of the physical variables*

For a given method of interpreting the readings (actually the C.C.I.T.T. method, see below) the effects of the physical variables (2), (3) and (4), section 1, may be summarized as follows.

For a given integration time (2) and law of rectification (4), the effect of varying the overswing from 0 (critical damping) to 5% (practical range) is very slight and may be neglected.

The effect of varying the integration time, expressing the reading relative to that for an integration time of 100 ms, is as follows :

100 ms	150 ms	200 ms	250 ms	300 ms
0	−0.6 dB	−1.2 dB	−1.9 dB	−2.6 dB

This relationship is practically independent of whether the measurement is made on high-quality or commercial telephone speech signals.

The effect of varying the law of rectification (4) depends somewhat on the type of speech signals. The following results are expressed relative to the reading for $k = 2$.

k	2.0	1.5	1.0
High-quality speech	0	−1.1 dB	−2.8 dB
Speech from British Post Office telephone sets	0	−1.6 dB	−3.6 dB

3. *Comparison of certain conventional volume meters*

Three important examples of conventional volume meters have been studied :

A. The S.F.E.R.T. volume indicator described in Annex 18 of Volume V of the *Red Book*;

B. The British Post Office speech voltmeter which is identical to the A.R.A.E.N. volume meter described in Supplement No. 10 of this volume;

C. The United States vu meter described in Supplement No. 11.

When used for measuring continuously spoken speech (not logatoms or conventional phrases) the deflections of A and B are specified to be interpreted according to the C.C.I.T.T. rule of observing that reading which is exceeded, on the average, once every three seconds. The vu meter, however, is specified to be read by taking the average of the peak deflections about every 10 seconds after excluding the two or three highest.

There is, naturally, an error due to the human element in interpreting any single reading and, for a given observer, this may amount to 5% of readings being as much as 2 dB away from his mean value obtained for a very large number of observations on the same speech sample.

Furthermore, an individual observer is subject to bias which may affect all his readings; even with trained observers a range of 4 dB is encountered. The errors have been found by the British Post Office to be about the same magnitude for both methods of interpretation.

The two methods of interpreting the deflections give results which differ by about 3 dB (the C.C.I.T.T. method yields the higher reading).

The following table shows the relative readings obtained on the three types of volume meter together with the values of their physical parameters. These readings were averaged over a variety of talkers and several experienced observers and the mean values are reliable to within about ± 1 dB. They apply for speech from British Post Office telephone sets.

Meter	Zero reference level	Integration time (ms)	Overswing %	99 % rise time (ms)	Rectifier law (exponent K)	Relative reading	
						C.C.I.T.T. method	Vu meter method
British Post Office speech voltmeter	1 volt	100	2.5	230	2	0	—
S.F.E.R.T. volume indicator	6 mW in 600 ohms	200	0 to 5	(410-650) depending on overswing	2	−6.8	—
Vu meter	1 mW in 600 ohms	(120)	1 to 1.5	300	1.4	—	−3.2

Note.— The figures in brackets are approximate only.

The above relationships are valid for both continuously spoken speech (e.g. passages read from a book) and telephonic conversational speech. (In the latter case, the time occupied by gaps in the speech are, of course, disregarded.) They do not apply, however, for such speech material as single words or syllables used in articulation testing nor for specially chosen phrases such as " can con by dace also " or " Paris, Bordeaux, Le Mans, Saint-Leu, Léon, Loudun," which are used for loudness balancing.

It is difficult with certain other measurements to exclude the effect of gaps in the speech and so the following relationships were obtained using continuously spoken speech.

The B.B.C. peak programme meter (Supplement No. 12) yields readings 1 or 2 dB higher than the 99 percentage level given below. It may be remarked that, in spite of the apparently slower movements of the needle of this instrument its readings are subject to approximately the same order of random error and observer bias as the British Post Office speech voltmeter and vu meter.

Measurement	Circuit	
	High quality	British Post Office telephone sets
Speech voltmeter reading	0	0
Long-time r.m.s. (dB rel. 1 volt)	−6.9	−8.2
Level not exceeded for 99 % of time in a long sample (dB rel. 1 volt)	+5.1	+6.0

4. *Elimination of observer errors*

For many laboratory purposes the observer errors mentioned above are not acceptable and the British Post Office has, for a number of years, been using an electronic equipment which makes an objective interpretation of an electrical function analogous to the movements of a volume meter needle. This is described in the reference below from which the above information also has been taken.

Reference. — J. N. SHEARME and D. L. RICHARDS : "The measurement of speech level"; *P.O.E.E.J. 47* (1954), p.159.

SUPPLEMENT No. 15

(former Annex 6; referred in Recommendation P.42)

EXTRACT FROM A STUDY OF THE DIFFERENCES BETWEEN RESULTS FOR INDIVIDUAL CREW MEMBERS IN LOUDNESS BALANCE TESTS

(Contribution by the British Administration)

1. *Introduction*

Loudness balance tests are usually conducted using a crew of from three to six members, each member performing talking and listening tasks in turn. Thus, with a crew of three, there are six possible different combinations of talker and listener; with five in the crew, there are twenty, and with six, thirty different crew combinations are possible.

Although the crew members may be given considerable training and acquire extensive experience, they retain characteristic individual differences both in the talking role and in that of listening. These individual differences, moreover, tend to change somewhat over periods of some weeks; occasionally, a quite large change has suddenly been encountered and once detected it has usually been possible to trace it to some definite physical cause.

The following methods if used regularly will enable a check to be maintained on the performances of individual crew members and to disclose any unusual changes. The method also discloses differences between results for individual crew members which, though normally present, may lie unsuspected if a particular crew is retained over a long period of time. Such effects as these if undisclosed can result in very misleading interpretations being placed upon results.

2. *Method*

The method of analysis will first be illustrated by a simple example taken from the measurements which contributed to Rapport technique No. 257.

The individual balances for the reference equivalent of the A.R.A.E.N. sending end (test 1) are shown in Table 1 below.

In a table of this form separation of the effects of talkers from those of listeners is complicated by the " missing diagonal ". A method for dealing with the situation was devised by Yates (*Annals of Eugenics*, 1, p. 121, 1936) and is described in several textbooks on statistical methods under the heading of " Balanced Incomplete Blocks "; a convenient reference for the present application is Brownlee, *Industrial Experimentation*, Her Majesty's Stationery Office, fourth edition 1949, p. 147.

TABLE 1

Individual talker/listener balances

Talker	Listener						Sum T	Q_T	V_T dB
	Mk	Lv	El	Ik	Cz	Ct			
Mk		−2	−3	−1	−4	−2	−12	−11	−0.46
Lv	−3		−5	−4	+2	+1	− 9	−13	−0.54
El	+4	−2		−5	−1	0	− 4	+ 7	+0.29
Ik	+3	−3	0		−1	+1	0	+27	+1.13
Cz	+1	−3	−5	−4		+1	−10	−10	−0.42
Ct	0	−2	−4	−3	0		− 9	0	0
Sum L	+5	−12	−17	−17	−4	+1	−44	0	−0.01
Q_L	+57	−25	−45	−41	+14	+40	0		
V_L dB	+2.38	−1.04	−1.88	−1.71	+0.58	+1.67	0		

Grand mean = −44/30 = −1.47 dB

The formation of the last three rows and columns additional to those containing the observed data is as follows :

sum T and sum L are obtained by adding the figures in each row or column ignoring the blank cells.

TABLE 2

Values of V_T and V_L, reference equivalent of A.R.A.E.N. versus S.F.E.R.T.

Circuit	Crew member	Without filter V_T Test 1	V_T Test 2	V_T Test 3	V_T Test 4	V_T Mean	V_L Test 1	V_L Test 2	V_L Test 3	V_L Test 4	V_L Mean
A.R.A.E.N. sending end	Mk	−0.46	−0.38	+0.62	+0.83	+0.16	+2.38	+2.62	+1.46	+0.50	+1.74
	Lv	−0.54	−0.71	+0.42	−0.04	−0.22	−1.04	−0.04	+0.42	−0.88	−0.38
	El	+0.29	+0.71	+0.08	−0.08	+0.25	−1.88	−2.96	−3.25	−2.08	−2.54
	Ik	+1.12	−0.33	−0.17	+0.33	+0.24	−1.71	−2.17	+0.50	+1.00	−0.59
	Cz	−0.42	+0.46	+0.21	+1.38	+0.41	+0.58	+1.79	+1.38	+2.21	+1.49
	Ct	−0.01	+0.25	−1.17	−2.42	−0.83	+1.67	−0.75	−0.50	−0.75	+0.29
	R.E.	−1.47	−2.10	−2.33	−1.93	−1.96	−1.47	−2.10	−2.33	−1.93	−1.96
A.R.A.E.N. receiving end	Mk	−1.42	+0.54	+0.33	−0.50	−0.26	+0.25	+1.71	+0.17	0.0	+0.53
	Lv	+0.25	−0.88	+0.62	+0.21	+0.05	−0.42	−0.38	−0.38	−0.46	−0.41
	El	+0.46	−1.17	−0.42	−0.17	−0.32	+0.62	+0.17	−0.58	−1.33	−0.28
	Ik	−0.42	+0.62	+1.08	+1.29	+0.64	−2.75	−1.88	−1.08	−0.04	−1.44
	Cz	+0.33	+0.08	+0.17	+0.04	+0.16	0.0	−0.58	+0.33	+0.71	+0.11
	Ct	+0.79	+0.79	−1.79	−0.88	−0.27	+2.29	+0.96	+1.54	+1.12	+1.48
	R.E.	+5.47	+5.80	+5.30	+4.90	+5.36	+5.47	+5.80	+5.30	+4.90	+5.36
A.R.A.E.N. complete system	Mk	−1.58	−1.04	+2.83	+0.71	+0.23	+3.75	+2.29	+2.0	+1.04	+2.27
	Lv	0.0	+0.38	+0.58	−0.42	+0.14	−0.33	−0.62	−2.25	−2.58	−1.45
	El	−0.25	−0.04	−0.88	−0.12	−0.32	−1.58	−1.71	−3.54	−3.12	−2.49
	Ik	−0.17	−0.25	0.0	+1.42	+0.25	−4.17	−2.75	+1.83	+0.58	−1.12
	Cz	+1.46	+1.08	−1.0	−0.29	+0.31	+0.96	+0.92	+0.83	+2.04	+1.19
	Ct	+0.54	−0.12	−1.54	−1.29	−0.60	+1.38	+1.38	+1.12	+2.04	+1.60
	R.E.	+3.33	+3.30	+3.37	+2.90	+3.22	+3.33	+3.30	+3.37	+2.90	+3.22

TABLE 2

Values of V_T and V_L, reference equivalent of A.R.A.E.N. versus S.F.E.R.T.

With filter V_T Test 1	V_T Test 2	V_T Test 3	V_T Test 4	V_T Mean	V_L Test 1	V_L Test 2	V_L Test 3	V_L Test 4	V_L Mean	Crew member	Circuit
−0.58	−1.00	+0.92	+0.79	+0.03	+0.92	+1.33	−0.42	−2.04	−0.05	Mk	A.R.A.E.N. sending end
+1.58	+0.25	−0.58	+0.54	+0.45	−0.25	−1.42	−1.92	−2.29	−1.47	Lv	
−1.29	−0.67	−1.00	+0.75	−0.55	−1.62	−2.0	−1.00	−0.25	−1.22	El	
+0.96	−0.12	+0.67	−0.29	+0.30	−2.38	−1.29	+0.08	+0.54	−0.70	Ik	
−0.79	+0.25	−0.08	−0.12	−0.19	+0.88	+0.58	+2.83	+4.38	+2.10	Cz	
+0.12	+1.29	+0.88	−1.67	−0.04	+2.46	+2.79	+0.42	−0.33	+1.33	Ct	
+3.77	+3.47	+3.40	+4.0	+3.66	+3.77	+3.47	+3.40	+4.0	+3.66	R.E.	
−0.33	−1.50	+0.50	+0.21	−0.28	+2.67	+3.0	−1.17	−1.29	+0.80	Mk	A.R.A.E.N. receiving end
−0.62	+1.04	+0.42	−1.58	−0.19	−0.79	−2.29	−0.58	−1.25	−1.23	Lv	
−0.50	−1.08	+1.25	+1.21	+0.22	−1.17	−0.92	−0.42	−2.29	−1.20	El	
+2.42	−0.17	+0.46	+1.08	+0.95	−3.58	−3.33	+0.62	+1.08	−1.30	Ik	
+0.54	+0.62	−0.88	−0.08	+0.05	+1.04	+0.62	−0.04	+2.25	+0.97	Cz	
−1.50	+1.08	−1.75	+0.83	−0.75	+1.83	+2.92	+1.58	+1.50	+1.96	Ct	
+8.87	+8.10	+8.47	+8.13	+8.40	+8.87	+8.10	+8.47	+8.13	+8.40	R.E.	
−0.79	−2.96	+0.21	+1.75	−0.45	+1.38	+3.21	−0.79	−0.92	+0.72	Mk	A.R.A.E.N. complete system
+0.21	+1.29	−0.38	−0.04	+0.27	−0.62	−1.54	−1.71	−1.88	−1.44	Lv	
−0.96	+0.29	+1.17	−0.12	+0.09	−2.46	−0.54	−2.0	−2.39	−2.07	El	
+0.88	−0.58	+0.75	+0.21	+0.31	−4.29	−3.92	+0.92	+2.38	−1.23	Ik	
+1.21	+2.54	+0.38	+0.12	+1.06	+3.38	+1.71	+2.04	+1.96	+2.27	Cz	
−0.54	−0.58	−2.12	−1.92	−1.29	+2.62	+1.08	+1.54	+1.75	+1.75	Ct	
+6.67	+8.60	+6.83	+7.07	+7.29	+6.67	+8.60	+6.83	+7.07	+7.29	R.E.	

Q_T is formed from :

5 (corresponding sum T) + (sum L for column containing the blank cell in the row considered) — grand total.

Thus for the first row :

$$Q_T = 5\ (-12) + (+5) - (-44) = -11$$

Q_L is formed in the corresponding manner, thus for the first column :

$$Q_L = 5\ (+5) + (-12) - (-44) = +57$$

As checks : $\sum Q_T = 0$
$\sum Q_L = 0$

V_T and V_L are equal to 1/24th of the corresponding Q_T or Q_L.

As checks : $\sum V_T = 0$
$\sum V_L = 0$

except that accumulated errors due to rounding the final digits as in the case here for $\sum V_T$ may give a slight residual discrepancy. When the number of talkers and listeners differs from six the values of the coefficients are different; general formulae are given in the Appendix.

The quantities V_T and V_L are the mean departures of the results for each talker and listener from the grand mean after correction for the unbalance in the design due to the " missing diagonal ". Thus for talker Mk the corrected mean result averaged over all listeners including Mk would be :

$$-1.47 - 0.46 = -1.93 \text{ dB}$$

and for listener Mk averaged over all talkers including Mk :

$$-1.47 + 2.38 = +0.91 \text{ dB.}$$

As in this example it is usually found that V_T is less dispersed than V_L.

It is possible, by analysis of variance (described in Brownlee), to separate an error variance and so determine a confidence interval for V_T and V_L. On this basis of a single set of results as in the above Table 1 the interval would be rather wide (perhaps $\pm$ 1.5 dB for 95 % confidence) and so it is rarely profitable to perform the above calculations for only an isolated set of, say, 30 balances. None the less, in the above example it would be found that Mk, El and Ik departed significantly from the mean as listeners.

3. *Application*

3.1 *Reference equivalent of A.R.A.E.N.*

The benefit from applying the method described above is reaped mainly by considering together several sets of results. As an example four replications of reference equivalent measurements of A.R.A.E.N. have been used. (Rapports techniques Nos. 257 and 257 *ter*, Tables 1, 2 and 3 relate to these data.)

The individual talker/listener results of each determination were first assembled in the form of Table 1 above and the values of V_T and V_L determined. The values of V_T and V_L are assembled in Table 2 and scrutiny of these discloses certain prominent features, namely :

a) The range of V_L is much greater than that of V_T.

b) Broadly the listeners here retain their relative performance throughout the values of V_L. Thus Ik is generally some few dB negative whereas Mk and Ct are generally positive by a dB or so. (This feature is not generally true as will be seen later.)

A more informative examination of the results is possible if confidence intervals are estimated and this can be done by analysis of variance of results for several tests suitably grouped together. For simplicity the case of grouping each set of four replications is considered; in general, any number may be taken but it is desirable to group only those which have important features in common, e.g. replications.

Analysis of variance

The computations for each analysis of variance are performed on the results in the form of four replications each arranged as Table 1 together with a summation table similar to Table 1 but containing as each entry the corresponding sum over the four replications (except for V_T and V_L which are mean values). All the usual computational checks should be made.

Details of the quantities computed are given in the Appendix and the results for A.R.A.E.N. sending without filter are as follows :

TABLE 3

Analysis of variance for four replications

Reference equivalent of A.R.A.E.N., sending without filter

Factor	Code	Degrees of freedom	Sum of squares	Mean square	Variance ratio F	Significance
Replication	B	3	(1) 12.09	4.03	1.48	N.S.
" Talkers "	" C "	—	(2) 31.34		—	—
" Listeners "	" D "	—	(3) 247.14	—	—	—
Talkers	C	5	(4) 20.19	4.04	1.48	N.S.
Listeners	D	5	(5) 235.99	47.20	17.35	*
Interaction	C D	19	(6) 77.71	4.09	1.50	N.S.
	B " C "	—	(7) 46.96	—	—	—
	B " D "	—	(8) 87.16	—	—	—
	B C	15	(9) 47.51	3.17	1.16	N.S.
	B D	15	(10) 87.71	5.85	2.15	N.S.
Error	B C D	57	(11) 154.99	2.72	—	—
Total ...		119	(12) 646.79			

* Significant at the 0.01 level.
N.S. = Not significant at the 0.05 level.

From this table the following may be stated :

a) the mean results for the four replications do not differ significantly and their common mean of −1.96 dB (Table 2) may be taken,

b) talkers do not differ significantly, i.e. the dispersion of the V_T s is not significant,

c) listeners do differ highly significantly and so a separate corrected mean ($-1.96 + V_L$) must be taken for each listener,

d) the interaction CD is not significant and so the result for a given talker/listener pair may be estimated as $-1.96 + V_T + V_L$,

e) the interactions BC and BD are not significant so that the dispersion of the V_T s and V_L s does not differ significantly between the four tests,

f) the error mean square (2.72) enables confidence limits to be set to the V_L s (and if desired to the grand mean).

The pattern of factors which are respectively not significant and significant obtained from the results of performing such an analysis of variance on all the tests indicated above is as follows :

TABLE 4

Summary of analysis of variance

Factor	Code	A.R.A.E.N./S.F.E.R.T.					
		sending		receiving		complete system	
		without filter	with filter	without filter	with filter	without filter	with filter
Tests	B	N.S.	N.S.	N.S.	N.S.	N.S.	N.S.
Talkers	C	N.S.	N.S.	N.S.	N.S.	N.S.	N.S.
Listeners	D	***	*	***	N.S.	**	*
Interaction	C D	N.S.	N.S.	N.S.	N.S.	N.S.	***
	B C	N.S.	*	N.S.	N.S.	***	***
	B D	N.S.	***	N.S.	***	***	***
Error mean square		2.72	2.01	2.46	3.10	1.60	0.91

* Significant at the 0.05 level.
** Significant at the 0.01 level.
*** Significant at the 0.001 level.
N.S. = Not significant at the 0.05 level.

Remark.— Main effects have been tested for significance against the largest interaction mean square which is both significant and involves the main effect factor. If no such interaction is significant the error mean square has been used.

From the above certain features emerge as follows :

(i) The factor " tests " is not significant (where this is not so further examination is called for).

(ii) The factor " talkers " is usually not significant but no special action can generally be taken if this is not so.

(iii) The factor " listeners " is almost always significant.

(iv) The CD interaction factor is normally not significant and here its very high significance in the last column would call for examination.

(v) The interaction BC is usually not significant and the very high significance for the overall results calls for investigation.

(vi) The interaction BD is usually not significant when replications separated by only a few days are concerned. Where, as here, the replications are spread over a long period of time differential, changes in listening performance are quite common and will result in the significance of this interaction of the systems compared differ widely.

(vii) Only experience will indicate the normal range of values for the error mean square (here about 1 to 3 decibel units).

Estimation of confidence intervals

If no factor were significant the confidence interval would be given by the expression :

$$\pm t \sqrt{\frac{\text{error mean square}}{\text{number of observations}}}$$

If there are n members of the crew and all combinations are used once per replication, the number of observations per replication is $n\,(n-1)$; the total number of observations would then be $rn\,(n-1)$ where r is the number of replications, t is " Student's " t given in tables as a function of the number of degrees of freedom for error.

If, as is quite common, the factor listeners is significant, the same " internal " confidence interval would apply for comparisons between circuit conditions measured at the same time using the same crew but not for those measured at different times or involving different crews. In the latter case the " external " confidence interval would be inflated by the existence of a component of variance for listeners. A rigorous estimate of such a confidence interval is very complicated but the following approximation is sufficient for the present purposes.

The component of variance for listeners is given approximately by

$$\sigma_L^2 = \frac{(\text{mean square for listeners}) - \text{error mean square}}{r \cdot n\,(n-2)/(n-1)}$$

then the confidence interval becomes

$$\pm t \sqrt{\frac{\text{error mean square}}{\text{total number of observations}} + \frac{\sigma_L^2}{\text{number of listeners}}}$$

where t is " Student's " t for $(n-1)$ degrees of freedom.

Taking values from Table 3,

$$\sigma_L^2 = \frac{47.20-2.72}{4 \cdot 6 \cdot 4/5} = \frac{44.48}{19.2} = 2.32$$

and the 95% confidence interval

$$\pm 2.57 \sqrt{\frac{2.72}{120} + \frac{2.32}{6}} =$$

$$= \pm 2.57 \sqrt{0.0227 + 0.387} = \pm 1.65 \text{ dB}$$

In the absence of a significant listener effect the 95% confidence interval would have been

$$\pm 2.09 \sqrt{\frac{2.72}{120}} = \pm 0.31 \text{ dB}$$

It is obvious from this exercise that the listener component of variance exerts the major controlling influence on the external confidence interval (which is that normally required) and shows clearly that n must be made as large as possible preferably at least 6, and that increasing the number of replications with the same crew is not profitable; indeed a single replication with the CD interaction treated as the error may often suffice.

3.2 *Reference equivalents and relative equivalents of seven commercial local ends*

The values of V_T and V_L for one replication of the measurements of reference equivalent and relative equivalent of seven commercial local ends described in Rapport technique No. 257 are assembled in Table 5 below. It would be appropriate to treat the results of two replications together by analysis of variance in the manner described above in section 3.1 but this is not shown here.

It is, however, also instructive to determine the reference equivalents indirectly by adding the relative equivalents to the appropriate values of reference equivalent of A.R.A.E.N. given in Table 2—i.e. the mean values averaged over the four replications. The results of this addition are given in Table 6 below which includes not only the mean indirect reference equivalents (denoted by R.E.) but the individual departure from this mean for each talker and listener (denoted by V_T and V_L).

Comparison of Table 6 with the corresponding reference equivalents in Table 5 discloses discrepancies which average about 2 dB. The mean values of these for each type of local end are given in Table 7, which also shows (as V_T and V_L) the individual departures from each of these mean discrepancies. It will be seen that many of these, especially V_L for receiving end results, are of comparable magnitude to the mean discrepancies, thus showing that certain subjects give practically no discrepancy while others give some 4 dB. It becomes clear therefore that these discrepancies are very much dependent upon the subjects used and it is futile to expect mean discrepancies less than a few decibels unless a very large number of subjects is used.

4. *Recommended routine method of analysis*

It is recommended that values of V_T and V_L should be calculated for future experiments together with the mean values as in Table 1 and that all replications of each test should be combined and an analysis of variance performed as described in section 3.1. This will serve to detect any irregularities in the results and enable confidence intervals to be determined.

Table 5

TABLE 5

Values of V_T and V_L, reference equivalent (R.E.) and relative equivalent of commercial sets

Circuit	Crew member	V_T						
		Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden
Reference equivalent send	Mk	−1.01	−1.36	−0.38	−0.99	−0.68	−0.19	−1.78
	Lv	+1.33	+0.65	+0.53	+0.24	+1.36	+1.38	−0.09
	El	−0.34	+0.01	−0.32	+0.01	−0.60	+0.41	−0.08
	Ik	−0.78	+0.12	+0.15	+0.57	+0.12	−1.52	+0.18
	Cz	+0.24	+0.93	+0.26	+0.47	+0.12	−0.41	+0.58
	Ct	+0.55	−0.36	−0.24	−0.29	−0.32	−0.34	+1.20
	R.E.	+4.31	+13.26	+8.01	+8.77	+14.19	+12.97	+9.51
Relative equivalent send (Case 1)	Mk	−0.23	−1.11	−0.48	−0.58	−0.70	+0.40	−0.50
	Lv	+0.58	+0.43	+1.12	+0.55	+0.88	+1.05	+0.60
	El	−1.1	−0.98	−0.83	−0.68	−1.93	−0.28	−0.91
	Ik	−0.52	+1.01	+0.90	+0.31	+0.88	−0.43	−0.14
	Cz	+0.64	+0.64	−0.22	+0.17	+0.27	−0.25	+0.37
	Ct	+0.64	+0.01	−0.50	+0.24	+0.61	−0.49	+0.58
	R.E.	+8.23	+17.03	+11.89	+11.53	+18.57	+17.49	+12.01
Relative equivalent send (Case 2)	Mk	−0.45	−1.10	−0.77	−0.49	−0.42	+0.34	−1.58
	Lv	+0.33	+0.18	+0.98	+1.42	+0.97	+0.52	+0.49
	El	−0.42	−0.81	−0.84	+0.33	−0.52	+0.05	−0.62
	Ik	+0.66	+1.02	+2.20	+0.35	+0.59	+0.64	+0.59
	Cz	−0.42	+0.70	−0.57	−0.85	−0.18	−1.30	+0.52
	Ct	+0.31	+0.01	−1.01	−0.58	−0.44	−0.25	+0.59
	R.E.	+5.48	+13.56	+8.67	+8.2	+15.25	+14.43	+9.83
Relative equivalent send (Case 3)	Mk	−0.25	−0.55	−0.55	−0.28	−0.12	+0.46	−1.08
	Lv	+0.41	−0.39	+0.78	+1.15	+1.18	+1.09	−0.13
	El	+0.02	−0.09	−0.26	+0.02	−1.38	+0.32	−0.48
	Ik	−0.43	+0.33	+1.59	+0.42	−0.24	−0.08	−0.03
	Cz	−0.19	−0.18	−0.82	−0.38	+0.64	−0.85	+0.67
	Ct	+0.44	+0.09	−0.73	−0.92	−0.08	−0.31	+1.06
	R.E.	+2.90	+11.41	+6.76	+6.65	+13.21	+12.49	+6.95

TABLE 5

Values of V_T and V_L, reference equivalent (R.E.) and relative equivalent of commercial sets

V_L							Crew member	Circuit
Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden		
+0.69	−0.09	+1.32	+0.11	−0.24	+0.51	+0.25	Mk	Reference equivalent send
+1.80	−0.65	+0.90	−0.72	−0.88	+0.14	+0.11	Lv	
−0.18	−0.66	−1.35	−0.09	−0.67	−0.89	−0.61	El	
−2.34	−0.08	−2.02	−0.30	−0.08	−0.96	−0.96	Ik	
−0.86	−0.63	−1.08	−0.80	−0.44	−0.98	−0.16	Cz	
+0.88	+2.11	+2.22	+1.81	+2.31	+2.18	+1.37	Ct	
+4.31	+13.26	+8.01	+8.77	+14.19	+12.97	+9.51	R.E.	
+0.97	−0.58	+0.89	−0.05	+0.37	−0.93	+0.77	Mk	Relative equivalent send (Case 1)
−0.59	−2.27	−0.31	−2.18	−1.76	−0.28	−0.93	Lv	
+0.23	+0.45	−0.70	+0.45	+0.40	+0.70	−0.88	El	
−0.09	+0.81	−0.23	+0.01	−0.12	+0.29	−0.04	Ik	
−1.26	−0.62	−1.22	−0.30	−1.60	−1.58	−0.10	Cz	
+0.74	+2.21	+1.57	+2.08	+2.71	+1.81	+1.18	Ct	
+8.23	+17.03	+11.89	+11.53	+18.57	+17.49	+12.01	R.E.	
+0.95	−0.10	+0.50	−0.46	−0.25	−0.52	+0.25	Mk	Relative equivalent send (Case 2)
−0.33	−1.52	−0.35	−0.79	−1.13	−0.05	−0.37	Lv	
−0.32	−0.84	−1.28	−0.53	−0.55	−0.78	−0.72	El	
−0.51	−0.12	−0.27	−0.05	−0.01	+0.37	−0.88	Ik	
−1.12	−0.50	−0.70	−0.05	−0.68	−0.93	−0.21	Cz	
+1.34	+2.84	+2.09	+1.88	+2.62	+1.92	+1.92	Ct	
+5.48	+13.56	+8.67	+8.2	+15.25	+14.43	+9.83	R.E.	
+0.65	+0.32	+1.25	+0.85	+0.64	−0.04	+0.92	Mk	Relative equivalent send (Case 3)
+0.74	−0.78	+0.88	−0.38	−0.46	+0.32	−0.13	Lv	
+0.22	−0.79	−1.29	−0.65	−0.81	−0.12	−1.08	El	
−0.47	+0.53	−0.84	+1.15	−0.14	−0.31	−0.59	Ik	
−0.46	−0.21	−0.52	+0.19	−0.92	−0.58	−0.33	Cz	
−0.69	+0.92	+0.53	−0.16	+1.69	+0.72	+1.22	Ct	
+2.90	+11.41	+6.76	+6.65	+13.21	+12.49	+6.95	R.E.	

TABLE 5 (cont.)

Circuit	Crew member	V_T Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden
Receiving reference equivalent, without feeding current	Mk	−0.18	+0.62	+0.20	−0.03	+0.11	−0.75	−0.18
	Lv	−0.08	−0.01	+0.06	+0.32	+0.15	+0.15	+0.19
	El	−0.28	−0.41	−0.72	−0.59	−0.34	−0.07	−0.27
	Ik	+0.08	+0.01	+0.31	−0.07	0.0	+0.63	+0.47
	Cz	+0.41	−0.30	−0.07	+0.34	+0.09	+0.91	−0.10
	Ct	+0.05	+0.09	+0.22	+0.03	−0.01	+0.39	−0.12
	R.E.	+2.28	+5.31	+0.28	+1.33	−1.09	+0.30	−7.86
Receiving reference equivalent; microphone replaced by dummy resistance; feeding current applied	Mk	+0.07	+0.61	−0.02	−0.53	+0.11	−0.32	+0.17
	Lv	+0.39	+0.07	+0.32	+0.52	+0.28	+0.36	+0.23
	El	−0.45	−0.60	−0.40	−0.12	−0.59	−0.28	−0.22
	Ik	−0.35	−0.58	+0.30	+0.12	−0.20	−0.52	−0.28
	Cz	+0.25	+0.14	−0.25	+0.55	+0.01	+0.53	−0.03
	Ct	+0.09	+0.36	+0.04	−0.53	+0.39	+0.23	+0.12
	R.E.	+1.38	+5.78	−0.19	+1.27	−0.68	−0.26	−7.40
Receiving relative equivalent (Case 1)	Mk	+0.74	+1.57	+0.88	+0.47	+1.07	+0.48	+0.28
	Lv	+0.24	+0.44	+0.42	+0.71	+0.18	+0.22	+0.71
	El	−0.36	−0.56	−0.78	−0.59	−0.67	−0.21	−0.54
	Ik	−0.51	−0.66	+0.68	+0.03	+0.34	−0.57	+0.06
	Cz	−0.48	−0.30	−0.83	−0.33	−0.25	−0.31	+0.15
	Ct	+0.36	−0.49	−0.36	−0.28	−0.68	+0.38	−0.65
	R.E.	−1.25	+3.71	−2.21	−0.98	−3.99	−3.58	−10.46
Receiving relative equivalent (Case 2)	Mk	+1.07	+1.63	+1.06	+0.98	+0.99	+0.19	+0.24
	Lv	−0.02	+0.34	+0.22	+0.25	+0.18	+0.34	+0.13
	El	−0.40	−0.49	−0.42	−0.12	−0.37	+0.25	−0.33
	Ik	−0.03	−0.12	+0.15	−0.22	−0.32	−0.32	+0.14
	Ct	−0.32	−0.46	−0.51	−0.11	−0.11	+0.02	−0.08
	Ct	−0.30	−0.91	−0.50	−0.78	−0.38	−0.49	−0.11
	R.E.	−3.33	+1.04	−4.11	+2.97	−5.29	−6.17	−12.19
Receiving relative equivalent (Case 3)	Mk	+0.41	+1.27	+0.37	+0.70	+0.68	+0.04	−0.11
	Lv	+0.92	+0.91	+0.61	+0.77	+0.95	+0.79	+1.32
	El	+0.32	−0.78	−0.13	+0.38	−0.44	−0.02	+0.32
	Ik	−0.48	−0.22	+0.19	−0.38	−0.04	−0.63	+0.28
	Cz	−1.08	−0.99	−0.56	−0.72	−0.68	−0.59	−1.19
	Ct	−0.08	−0.19	−0.48	−0.75	−0.47	+0.41	−0.62
	R.E.	−5.23	−0.26	−6.17	−4.62	−7.99	−8.32	−14.51

TABLE 5 (cont.)

V_L Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden	Crew member	Circuit
+0.52	+1.45	+1.80	+2.30	+1.31	+1.35	+1.82	Mk	Receiving reference equivalent; without feeding current
−0.38	−0.08	−0.31	−1.15	−0.08	−2.35	−0.54	Lv	
−0.42	+0.12	−1.02	−0.29	−0.14	−0.83	+0.17	El	
−1.22	−1.79	+0.74	−1.07	−1.43	+0.13	−0.77	Iz	
+1.04	−0.53	+0.07	+0.38	−0.18	+0.84	−0.20	Cz	
+0.45	+0.82	−1.28	−0.17	+0.52	+0.86	−0.48	Ct	
+2.28	+5.31	+0.28	+1.33	−1.09	+0.30	−7.86	R.E.	
+0.63	+1.94	+0.78	+2.30	+1.34	+0.72	+2.43	Mk	Receiving reference equivalent; microphone replaced by dummy resistance; feeding current applied
−1.14	−0.37	+1.09	−1.01	−0.58	−1.91	−0.43	Lv	
−0.15	+0.10	−1.33	−0.26	−0.36	−0.52	−0.48	El	
−1.25	−1.58	−0.23	−0.85	−1.20	−0.12	−1.58	Ik	
+0.95	−0.99	+0.42	+0.12	+0.04	+1.17	+0.23	Cz	
+0.96	+0.89	−0.72	−0.30	+0.76	+0.67	−0.18	Ct	
+1.38	+5.78	−0.19	+1.27	−0.68	−0.26	−7.40	R.E.	
−0.56	+0.17	+1.74	+1.43	+0.57	+0.28	+0.88	Mk	Receiving relative equivalent (Case 1)
−1.86	−1.86	−0.55	−1.76	−1.85	−3.38	−1.76	Lv	
−0.46	−0.26	−0.31	−0.86	−0.30	−0.14	−0.21	El	
+3.19	+0.04	+2.34	+2.87	+0.94	+3.27	+2.59	Ik	
−0.04	−0.37	−1.00	−1.17	−0.02	−0.04	−0.35	Cz	
−0.28	+2.28	−2.22	−0.52	+0.66	+0.02	−1.15	Ct	
−1.25	+3.71	−2.21	−0.98	−3.99	−3.58	−10.46	R.E.	
+0.30	+0.37	+2.12	+0.84	+1.29	−0.11	+0.64	Mk	Receiving relative equivalent (Case 2)
−2.32	−1.89	−1.28	−1.78	−1.59	−3.16	−1.50	Lv	
−0.43	−0.86	−0.05	−0.46	−0.50	+0.98	−0.03	El	
+2.40	+1.22	+2.38	+3.08	+1.75	+2.15	+2.14	Ik	
+0.18	−0.29	−1.11	−1.38	−1.01	−0.14	−0.74	Cz	
−0.13	+1.46	−2.07	−0.31	+0.06	+0.28	+0.51	Ct	
−3.33	+1.04	−4.11	−2.97	−5.29	−6.17	−12.19	R.E.	
−1.11	+1.43	+1.80	+1.20	+1.21	+0.61	+1.09	Mk	Receiving relative equivalent (Case 3)
−1.55	−1.56	+0.81	−1.07	−0.62	−2.24	−0.98	Lv	
−0.55	−0.18	−0.10	−0.58	−0.38	−0.08	+0.02	El	
+2.05	+1.62	+2.52	+2.82	+1.22	+2.83	+2.25	Ik	
−0.11	−1.06	−1.62	−1.12	−0.54	−0.16	−0.72	Cz	
−0.95	−0.26	−3.41	−1.25	−0.90	−0.96	−1.65	Ct	
−5.23	−0.26	−6.17	−4.62	−7.99	−8.32	−14.51	R.E.	

TABLE 6

Values of V_T and V_L, indirect reference equivalent of commercial sets

Calculated by adding reference equivalent values from Table 2 to reference equivalent values in Table 5

Circuit	Crew member	V_T Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden
Sending (A.R.A.E.N. without filter) (Case 1)	Mk	−0.07	−0.95	−0.32	−0.42	−0.54	+0.56	−0.34
	Lv	+0.36	+0.21	+0.90	+0.33	+0.66	+0.83	+0.38
	El	−0.85	−0.73	−0.58	−0.43	−1.68	−0.03	−0.66
	Ik	−0.28	+1.25	+1.14	+0.55	+1.12	−0.19	+0.10
	Cz	+1.05	+1.05	+0.19	+0.58	+0.68	+0.16	+0.78
	Ct	−0.19	−0.82	−1.33	−0.59	−0.22	−1.32	−0.25
	R.E.	+6.27	+15.07	+9.93	+9.57	+16.61	+15.53	+10.05
Sending (A.R.A.E.N. with filter) (Case 3)	Mk	−0.22	−0.52	−0.52	−0.25	−0.09	+0.49	−1.05
	Lv	+0.86	+0.84	+1.23	+1.60	+1.63	+1.54	+0.32
	El	−0.53	−0.64	−0.81	−0.53	−1.93	−0.87	−1.03
	Ik	−0.13	+0.63	+1.89	+0.72	+0.06	+0.22	+0.27
	Cz	−0.38	−0.37	−1.01	−0.57	+0.45	−1.04	+0.48
	Ct	+0.40	+0.05	−0.77	−0.96	−0.12	−0.35	+1.02
	R.E.	+6.56	+15.07	+10.42	+10.31	+16.87	+16.15	+10.61
Receiving (A.R.A.E.N. without filter) (Case 1)	Mk	+0.48	+1.31	+0.62	+0.21	+0.89	+0.29	+0.02
	Lv	+0.29	+0.49	+0.47	+0.76	+0.23	+0.27	+0.76
	El	−0.68	−0.88	−1.10	−0.91	−0.99	−0.53	−0.86
	Ik	+0.13	−0.02	+1.32	+0.67	+0.98	+0.07	+0.70
	Cz	−0.32	−0.14	−0.67	−0.17	−0.09	−0.15	+0.31
	Ct	+0.09	−0.76	−0.63	−0.55	−0.95	+0.11	−0.92
	R.E.	+4.11	+9.07	+3.15	+4.38	+1.37	+1.78	−5.10
Receiving (A.R.A.E.N. with filter) (Case 3)	Mk	+0.13	+0.99	+0.09	+0.42	+0.40	−0.24	−0.39
	Lv	+0.73	+0.72	+0.42	+0.58	+0.76	+0.60	+1.13
	El	+0.54	−0.56	+0.09	+0.60	−0.22	+0.20	+0.54
	Ik	+0.47	+0.73	+1.14	+0.57	+0.91	+0.32	+1.23
	Cz	−1.03	−0.94	−0.51	−0.67	−0.63	−0.54	−1.14
	Ct	−0.83	−0.94	−1.23	−1.50	−1.22	−0.34	−1.37
	R.E.	+3.17	+8.14	+2.23	+3.78	+0.41	+0.08	−6.11

TABLE 6

Values of V_T and V_L, indirect reference equivalent of commercial sets

Calculated by adding reference equivalent values from Table 2 to reference equivalent values in Table 5

V_L Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden	Crew member	Circuit
+2.71	+1.16	+2.63	+1.69	+2.11	+0.81	+2.51	Mk	Sending (A.R.A.E.N. without filter) (Case 1)
−0.97	−2.65	−0.69	−2.56	−2.14	−0.66	−1.31	Lv	
−2.31	−2.09	−3.24	−2.09	−2.14	−1.84	−3.42	El	
−0.68	+0.22	−0.82	−0.58	−0.71	−0.30	−0.63	Ik	
+0.23	+0.87	+0.27	+1.19	−0.11	−0.09	+1.39	Cz	
+1.03	+2.50	+1.86	+2.37	+3.00	+2.10	+1.47	Ct	
+6.27	+15.07	+9.93	+9.57	+16.61	+15.53	+10.05	R.E.	
+0.60	+0.27	+1.20	+0.80	+0.59	+0.09	+0.87	Mk	Sending (A.R.A.E.N. with filter) (Case 3)
−0.73	−2.25	−0.59	−1.85	−1.93	−1.15	−1.60	Lv	
−1.00	−2.01	−2.51	−1.87	−2.03	−1.34	−2.30	El	
−1.17	−0.17	−1.54	−0.55	−0.84	−1.01	−1.29	Ik	
+1.64	+1.89	+1.58	+2.29	+1.18	+1.52	+1.77	Cz	
+0.64	+2.25	+1.86	+1.17	+3.02	+2.05	+2.55	Ct	
+6.56	+15.07	+10.42	+10.31	+16.87	+16.15	+10.61	R.E.	
−0.03	+0.70	+2.27	+1.96	+1.10	+0.81	+1.41	Mk	Receiving (A.R.A.E.N. without filter) (Case 1)
−2.27	−2.27	−0.96	−2.17	−2.26	−3.79	−2.17	Lv	
−0.74	−0.54	−0.59	−1.14	−0.58	−0.42	−0.49	El	
+1.75	−1.40	+0.90	+1.43	−0.50	+1.83	+1.15	Ik	
+0.07	−0.26	−0.89	−1.06	+0.09	+0.07	−0.24	Cz	
+1.20	+3.76	−0.74	+0.96	+2.14	+1.50	+0.33	Ct	
+4.11	+9.07	+3.15	+4.38	+1.37	+1.78	−5.10	R.E.	
+1.91	+2.23	+2.60	+2.00	+2.01	+1.41	+1.89	Mk	Receiving (A.R.A.E.N. with filter) (Case 3)
−2.78	−2.79	−0.42	−2.30	−1.85	−3.47	−2.21	Lv	
−1.75	−1.38	−1.30	−1.78	−1.58	−1.28	−1.18	El	
+0.75	+0.32	+1.22	+1.52	−0.08	+1.53	+0.95	Ik	
+0.86	−0.09	−0.65	−0.15	+0.43	+0.81	+0.25	Cz	
+1.01	+1.70	−1.45	+0.71	+1.06	+1.00	+0.31	Ct	
+3.17	+8.14	+2.23	+3.78	+0.41	+0.08	−6.11	R.E.	

TABLE 7

Discrepancies between calculated and observed values of reference equivalents

Calculated by subtracting indirect reference equivalent values in Table 6 from reference equivalent values in Table 5
(V_T and V_L represent individual deviations from the common mean)

Circuit	Crew member	V_T						
		Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden
Sending (A.R.A.E.N. without filter) (Case 1)	Mk	+0.94	+0.41	+0.06	+0.57	+0.14	+0.75	+1.44
	Lv	−0.97	−0.44	+0.37	+0.09	−0.70	−0.55	+0.47
	El	−0.51	−0.74	−0.26	−0.44	−1.08	−0.44	−0.58
	Ik	+0.50	+1.13	+0.99	−0.02	+1.00	+1.33	−0.08
	Cz	+0.81	+0.12	−0.07	+0.11	+0.56	+0.57	+0.20
	Ct	−0.74	−0.46	−1.09	−0.30	+0.10	−1.66	−1.45
Mean discrepancy ..		+1.96	+1.81	+1.92	+0.80	+2.42	+2.56	+0.54
Sending (A.R.A.E.N. with filter) (Case 3)	Mk	+0.70	+0.84	−0.14	+0.74	+0.59	+0.68	+0.73
	Lv	−0.47	+0.19	+0.70	+1.36	+0.27	+0.16	+0.41
	El	−0.19	−0.65	−0.49	−0.54	−1.33	−1.28	−0.95
	Ik	+0.65	+0.51	+1.74	+0.15	−0.06	+1.74	+0.09
	Cz	−0.62	−1.30	−1.27	−1.04	+0.33	−0.63	−0.10
	Ct	−0.15	+0.41	−0.53	−0.67	+0.20	−0.69	−0.18
Mean discrepancy ..		+2.25	+1.81	+2.41	+1.54	+2.68	+3.18	+1.10
Receiving (A.R.A.E.N. without filter) (Case 1)	Mk	+0.41	+0.70	+0.64	+0.74	+0.70	+0.54	−0.15
	Lv	−0.10	+0.42	+0.15	+0.24	−0.05	−0.09	+0.53
	El	−0.23	−0.28	−0.70	−0.79	−0.40	−0.25	−0.64
	Ik	+0.48	+0.56	+1.02	+0.55	+1.18	+0.59	+0.98
	Cz	−0.57	−0.28	−0.42	−0.72	−0.10	−0.70	+0.34
	Ct	0.00	−1.12	−0.67	−0.02	−1.34	−0.12	−1.04
Mean discrepancy ..		+2.73	+3.29	+2.96	+3.11	+2.05	+2.04	+2.30
Receiving (A.R.A.E.N. with filter) (Case 3)	Mk	+0.06	+0.38	+0.11	+0.95	+0.29	+0.08	−0.56
	Lv	+0.34	+0.65	+0.10	+0.06	+0.48	+0.24	+0.90
	El	+0.99	+0.04	+0.49	+0.72	+0.37	+0.48	+0.74
	Ik	+0.82	+1.31	+0.84	+0.45	+1.11	+0.84	+1.51
	Cz	−1.28	−1.08	−0.26	−1.22	−0.64	−1.07	−1.11
	Ct	−0.92	−1.30	−1.27	−0.97	−1.61	−0.57	−1.49
Mean discrepancy ..		+1.79	+2.36	+2.42	+2.51	+1.09	+0.34	+1.29

TABLE 7

Discrepancies between calculated and observed values of reference equivalents

Calculated by subtracting indirect reference equivalent values in Table 6 from reference equivalent values in Table 5
(V_T and V_L represent individual deviations from the common mean)

V_L							Crew member	Circuit
Switzerland	Belgium	Italy	Mexico	Netherlands	F. R. of Germany	Sweden		
+2.02	+1.25	+1.31	+1.58	+2.35	+0.30	+2.26	Mk	Sending (A.R.A.E.N. without filter) (Case 1)
−2.77	−2.00	−1.59	−1.94	−1.26	−0.80	−1.42	Lv	
−2.13	−1.43	−1.89	−2.00	−1.47	−0.95	−2.81	El	
+1.66	+0.30	+1.20	−0.28	−0.63	+0.66	+0.33	Ik	
+1.09	+1.50	+1.35	+1.99	+0.33	+0.89	+1.55	Cz	
+0.15	+0.39	−0.36	+0.56	+0.69	−0.08	+0.10	Ct	
+1.96	+1.81	+1.92	+0.80	+2.42	+2.56	+0.54	Mean discrepancy	
−0.09	+0.36	−0.12	+0.69	+0.83	−0.60	+0.62	Mk	Sending (A.R.A.E.N. with filter) (Case 3)
−2.53	−1.60	−1.49	−1.13	−1.05	−1.29	−1.71	Lv	
−0.82	−1.35	−1.16	−1.78	−1.36	−0.45	−1.69	El	
+1.17	−0.09	+0.48	−0.25	−0.76	−0.05	−0.35	Ik	
+2.50	+2.52	+2.66	+3.09	+1.62	+2.50	+1.93	Cz	
−0.24	+0.14	−0.36	−0.64	+0.71	−0.13	+1.18	Ct	
+2.25	+1.81	+2.41	+1.54	+2.68	+3.18	+1.10	Mean discrepancy	
−0.66	−1.24	+1.49	−0.34	−0.24	+0.09	−1.02	Mk	Receiving (A.R.A.E.N. without filter) (Case 1)
−1.13	−1.90	−2.05	−1.16	−1.68	−1.88	−1.74	Lv	
−0.59	−0.64	+0.74	−0.88	−0.22	+0.10	−0.01	El	
+3.00	+0.18	+1.13	+2.28	+0.70	+1.95	+2.73	Ik	
−0.88	+0.73	−1.31	−1.18	+0.05	−1.10	−0.47	Cz	
+0.24	+2.87	−0.02	+1.26	+1.38	+0.83	+0.51	Ct	
+2.73	+3.29	+2.96	+3.11	+2.05	+2.04	+2.30	Mean discrepancy	
+1.28	+0.29	+1.82	−0.30	+0.67	+0.69	−0.54	Mk	Receiving (A.R.A.E.N. with filter) (Case 3)
−1.64	−2.42	−1.51	−1.29	−1.27	−1.56	−1.78	Lv	
−1.60	−1.48	+0.03	−1.52	−1.22	−0.76	−0.70	El	
+2.00	+1.90	+1.45	+2.37	+1.12	+1.65	+2.53	Ik	
−0.09	+0.90	−1.07	−0.27	+0.39	−0.36	−0.02	Cz	
+0.05	+0.81	−0.73	+1.01	+0.30	+0.33	+0.49	Ct	
+1.79	+2.36	+2.42	+2.51	+1.09	+0.34	+1.29	Mean discrepancy	

Appendix

Analysis of balanced incomplete blocks. Subjective, talker-listener tests

1. *Values of Q_T, Q_L, V_T and V_L*

In the example shown in section 2, values of Q_T, Q_L, V_T and V_L are given for a speech test crew of six. General formula (derived from Brownlee) is given below.

Let sum T_1, and sum L_1 respectively correspond to the additions of the figures in the row and column assigned to crew member 1.

Let G_1 = grand total for 1st replication
n = number of crew members.

Then the required values for crew member 1 for the 1st replication would be as shown and would be similarly calculated for other members.

$$Q_{T1} = (n-1)\,T_1 + L_1 - G_1$$

$$Q_{L1} = (n-1)\,L_1 + T_1 - G_1$$

$$V_{T1} = \frac{Q_{T1}}{n\,(n-2)}$$

$$V_{L1} = \frac{Q_{L1}}{n\,(n-2)}$$

2. *Analysis of variance*

General values for the analysis of variance are shown in the following table for an experiment where

r = number of replications
$G = G_1 + G_2 \ldots\ldots G_r$

Factor	Code	Degrees of freedom	Sum of squares	Mean square	Variance ratio F	Significance
Replications	B	$(r-1)$	(1)	∅	—	—
" Talkers "	" C "	—	(2)	—	—	—
" Listeners "	" D "	—	(3)	—	—	—
Talkers	C	$(n-1)$	(4)	∅		
Listeners	D	$(n-1)$	(5)	∅		
Interactions	C D	$(n-1)\,(n-2)-1$	(6)	∅		
	B " C "	—	(7)	—	—	—
	B " D "	—	(8)	—	—	—
	B C	$(r-1)\,(n-1)$	(9)	∅		
	B D	$(r-1)\,(n-1)$	(10)	∅		
Error	B C D	$(r-1)\,[(n-1)\,(n-2)-1]$	(11)	∅		
Total ...		$r \cdot n\,(n-1)-1$	(12)			

Remark. — The mean squares ∅ are obtained by dividing the sum of squares by the corresponding number of degrees of freedom.

Typical results relating to a test with four replications and six crew members are shown in Table 3.

The sums of squares (1), (2), etc. are computed as follows from values given in individual replication tables of the form of Table 1 and from a combined replication table. The latter contains for each entry the corresponding sums over r replications (apart from V_T and V_L which are mean values).

$$(1) = \frac{G_1^2 + G_2^2 + \ldots\ldots G_r^2}{n\,(n-1)} - \frac{G^2}{r \cdot n \cdot (n-1)}$$

$$(2) = \frac{\sum \text{(grand total for each talker from combined replication table)}^2}{r \cdot (n-1)} - \frac{G^2}{r \cdot n \cdot (n-1)}$$

$$(3) = \frac{\sum \text{(grand total for each listener from combined replication table)}^2}{r \cdot (n-1)} - \frac{G^2}{r \cdot n \cdot (n-1)}$$

$$(4) = \frac{\sum Q_T^2}{r \cdot n \cdot (n-1)\,(n-2)} \quad \text{using } Q_T\text{s from combined table}$$

$$(5) = \frac{\sum Q_L^2}{r \cdot n \cdot (n-1)\,(n-2)} \quad \text{using } Q_L\text{s from combined table}$$

$$(6) = \frac{\sum (y_1 + y_2 + \ldots\ldots y_r)^2}{r} - \frac{G^2}{r \cdot n \cdot (n-1)} - [\{(2) + (5)\} \text{ or } \{(3) + (4)\}]$$

where $y_1, y_2 \ldots\ldots y_r$ are sum of individual entries from single replication as shown by combined table.

$$(7) = \frac{\sum \text{(total for each talker from individual replication tables)}^2}{(n-1)} - \frac{G^2}{r \cdot n \cdot (n-1)} - (1) - (2)$$

$$(8) = \frac{\sum \text{(total for each listener from individual replication tables)}^2}{(n-1)} - \frac{G^2}{r \cdot n \cdot (n-1)} - (1) - (3)$$

$$(9) = \frac{\sum Q_T^2}{n \cdot (n-1)\,(n-2)} - (4) \quad \text{using } Q_T\text{s from individual replication}$$

$$(10) = \frac{\sum Q_L^2}{n \cdot (n-1)\,(n-2)} - (5) \quad \text{using } Q_L\text{s from individual replication}$$

$$(11) = (12) - (1) - [\{(2) + (5)\} \text{ or } \{(3) + (4)\}] - [\{(7) + (10)\} \text{ or } \{(8) + (9)\}] - (6)$$

$$(12) = \sum (y_r)^2 - \frac{G^2}{r \cdot n \cdot (n-1)} \quad \text{where } y_r = \text{ entries from individual replication tables}$$

The following checks should be made :

$$(2) + (5) = (3) + (4)$$
$$(7) + (10) = (8) + (9)$$

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