THE INTERNATIONAL TELEGRAPH AND TELEPHONE CONSULTATIVE COMMITTEE

CCITT

SIXTH PLENARY ASSEMBLY

GENEVA, 27 SEPTEMBER - 8 OCTOBER 1976

ORANGE BOOK

VOLUME V

TELEPHONE TRANSMISSION QUALITY

Published by the
INTERNATIONAL TELECOMMUNICATION UNION
GENEVA, 1977

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CONTENTS OF THE CCITT BOOK
APPLICABLE AFTER THE SIXTH PLENARY ASSEMBLY (1976)

ORANGE BOOK

Each volume also contains, for its field and where appropriate:

- — definitions of specific terms used;
- — supplements for information and documentary purposes.

TABLE OF CONTENTS OF VOLUME V OF THE ORANGE BOOK

Part I – Series P Recommendations [1)]

Quality of telephone transmission; local networks and telephone installations

1) For the text of former Recommendations P.11 to P.21 see Recommendations G.111 to G.121 in Volume III.

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Part III – Questions on transmission performance and local networks entrusted to Study Group XII for the period 1977-1980

PRELIMINARY NOTES

1. This Volume fully supersedes Volume V of the CCITT *Green Book* (Geneva, 1972).

It has been indicated (immediately after the titles of Recommendations or Supplements) whether the texts are new ones approved by the Plenary Assembly of Geneva, 1976 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.

2. The units used in this Volume are in conformity with CCITT Recommendations B.3 and B.4 (Volume I of the *Orange Book*).

The indication "amended Geneva, 1976" has not been affixed to those Recommendations in which the only amendment has been an editorial change concerning units.

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

dBm the absolute power level in decibels;

dBm0 the absolute power level in decibels referred to a point of zero relative level;

dBr the relative power level in decibels;

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

The units for air pressure are related as follows:

$1 \text{Pa (Pascal)} = 1 \text{N (Newton)/m}^2 = 10 \text{ dyne/cm}^2 = 10 \text{ barye} = 10\ \mu\text{bar}$

3. In this Volume, the expression "Administration" is used for shortness to indicate both a telecommunication Administration and a recognized private operating agency.

PART I

Series P Recommendations

QUALITY OF TELEPHONE TRANSMISSION; LOCAL NETWORKS AND TELEPHONE INSTALLATIONS

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SECTION 1

GENERAL CHARACTERISTICS OF NATIONAL SYSTEMS FORMING PART OF INTERNATIONAL CONNECTIONS

Recommendation P.22

MANUAL TRUNK EXCHANGES

A. OPERATORS' POSITIONS

The CCITT,

considering,

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

unanimously recommends

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

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

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

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

B. SUPERVISORS' DESKS

The CCITT

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 CCITT,

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 simultaneously; that, consequently, it is appropriate, from the point of view of insertion loss caused by observation, to be more severe in the case of observation on the part of the supervisor's desk than in the case of supervision by an operator,

unanimously recommends

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

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

C. ARRANGEMENTS FOR CONFERENCE CALLS

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

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

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

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

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

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

The connecting equipment for conference calls should permit interconnection of 2-wire or 4-wire circuits without any change in setting up the circuits; the connecting equipment should equally permit 2-wire or 4-wire subscriber lines to be connected to the international circuits.

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

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

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

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

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

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

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

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SECTION 2

SUBSCRIBERS' LINES AND SETS

Recommendation P.31

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

The CCITT is at present studying the conditions imposed generally on the sensitivity of local sending and receiving telephone circuits [1]. Until the results of this study become available, Administrations should refer to this Recommendation, which lays down the conditions which should be satisfied by subscribers' stations used with international circuits leased for private purposes.

The CCITT,

considering

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

unanimously recommends

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

considering, too,

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

unanimously recommends

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

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

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

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

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[1] See Question 10/XII.

Recommendation P.32

DEVICES FOR RECORDING MESSAGES OR TELEPHONE CONVERSATIONS

The CCITT,

considering

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

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

unanimously recommends

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

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

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

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

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

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

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

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

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

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

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

5. In order to preserve the secrecy of telephone conversations, a conversation recorded with the maximum possible gain should be quite unintelligible if the speech volume is at least 55 dB 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 which is at most equal to that given by the receiver of the subscriber's telephone equipment connected to the line.

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

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

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

Recommendation P.33

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

(Mar del Plata, 1968; amended at Geneva, 1972)

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

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

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

the CCITT *makes the following provisional recommendation*:

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

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

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SECTION 3

TRANSMISSION STANDARDS

Recommendation P.41

DESCRIPTION OF THE ARAEN

A set of equipment which is kept in the CCITT Laboratory is known, for historical reasons, as the ARAEN (Reference apparatus for the determination of transmission performance ratings). Actually, ARAEN is used in the constitution:

- of NOSFER, for the determination of reference equivalents (see Recommendation P.42),
- or of SRAEN, for the determination of the AEN (see Recommendation P.44).

The ARAEN 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 ARAEN or in the test telephone circuit. The complete transmission path, when the filter is switched out of circuit, is designed to reproduce the transmission characteristics of a free field air path, one metre long, the air path being assumed to be used with monaural listening. Normal settings of the send and receive amplifiers are such that these characteristics are reproduced with 30 dB non-reactive attenuation in the junction.

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

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

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

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

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

Supplement No. 9 to the *White Book*, Volume V, describes the method for the absolute calibration of the ARAEN in the CCITT 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 CCITT Secretariat and the CCITT Laboratory; furthermore the mimeographed document entitled *Draft summary of instructions for the use and maintenance of the CCIF 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/P.41 and which are interconnected according to the arrangement in Figure 1/P.41 by means of the junction switching panel.

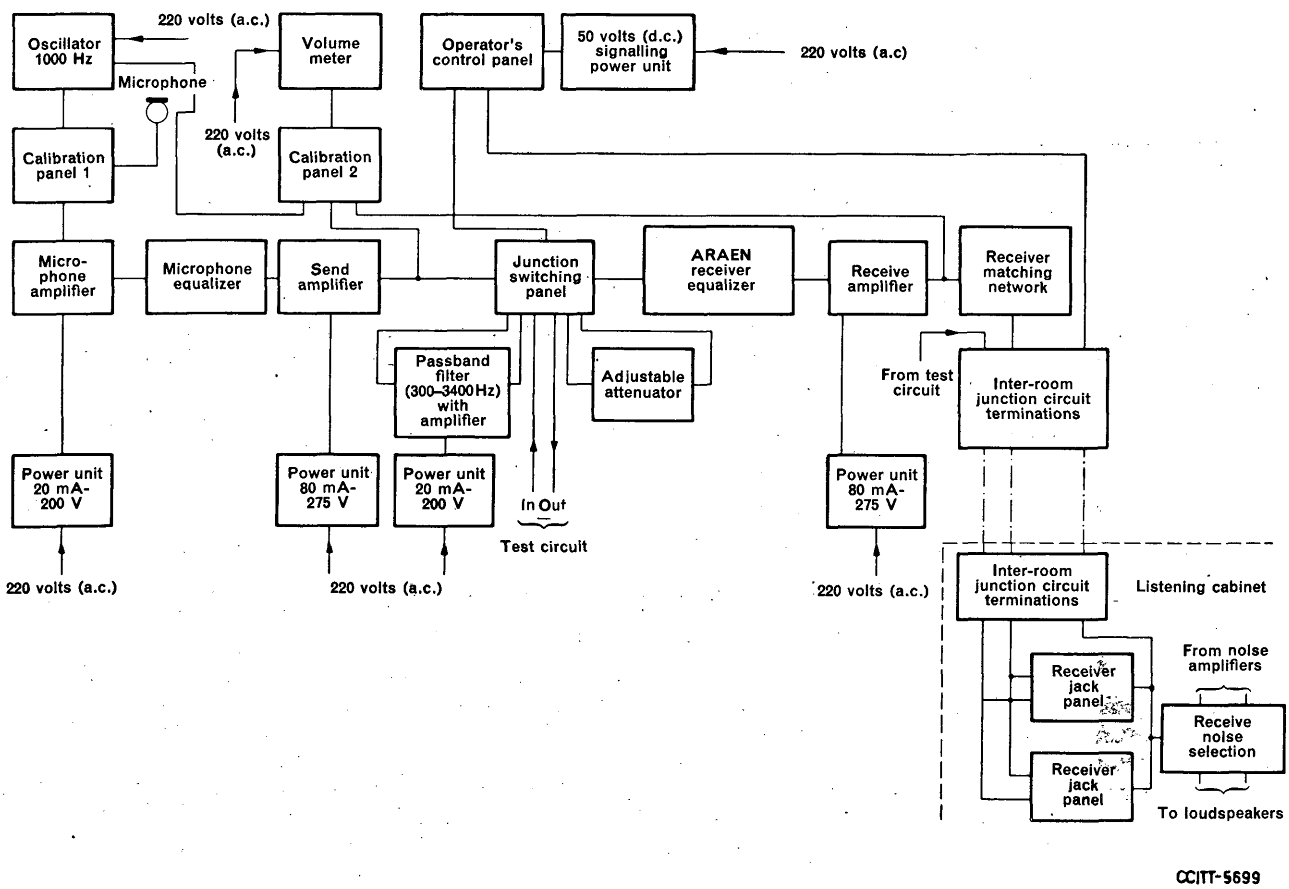


FIGURE 1/P.41 – Schematic diagram of the reference equipment for the determination of AEN

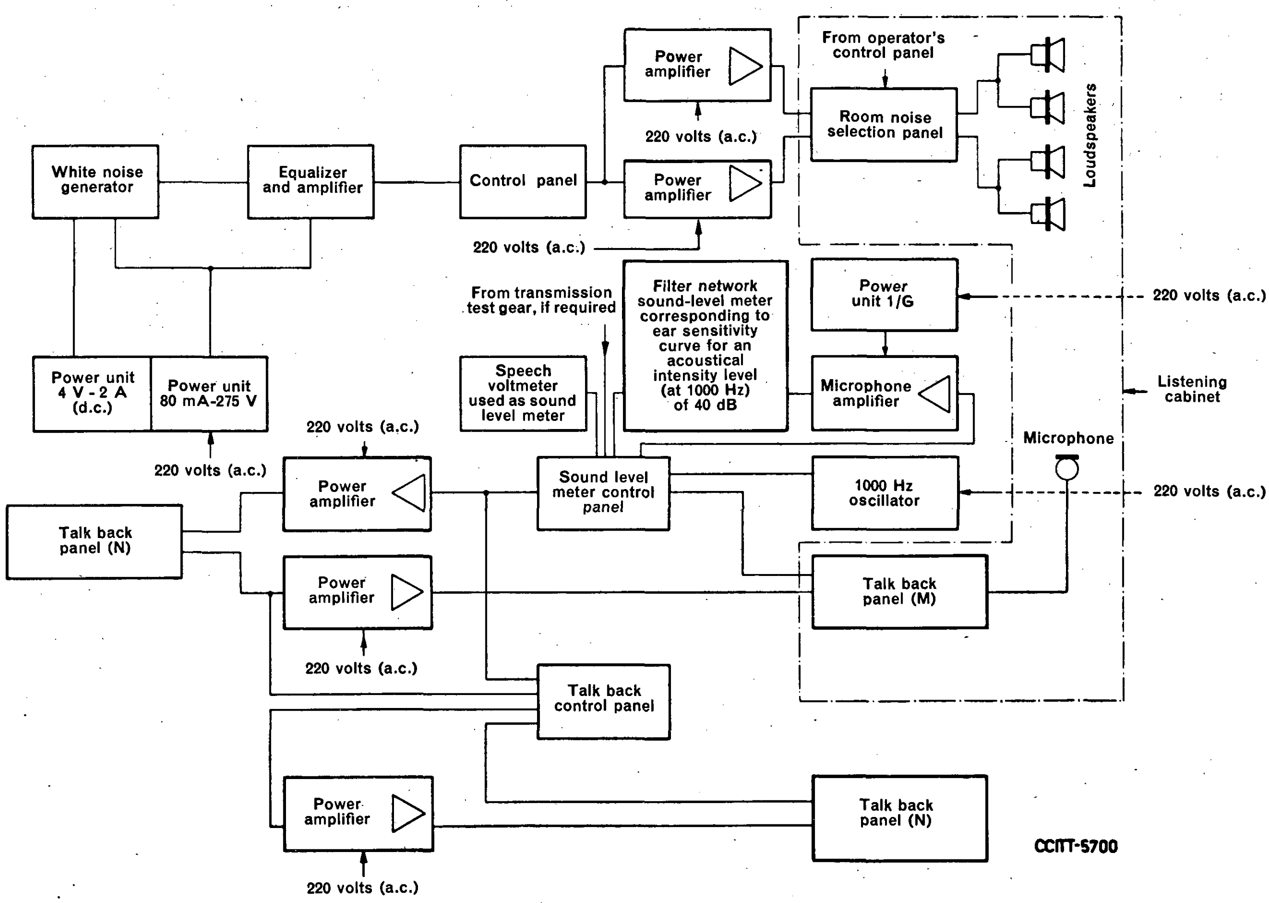
B. EQUIPMENT FOR SUPPLY OF ROOM NOISE AND INTERCOMMUNICATION CIRCUIT

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

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

TABLE 1/P.41

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 80-6000 Hz A fixed value of gain is provided Gain without feedback: 68 dB Gain with feedback: 47 ± 0.2 dB over range 80-6000 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 impedance: 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)



Note. – The type M talk back panel is similar to Type N except that it enables a microphone type 4021A to be switched in place of the small loudspeaker mounted on the panel normally used as the microphone.

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

C. CALIBRATION EQUIPMENT

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

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

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

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

Probe-tube microphone – For calibration of the ARAEN, two microphones and one amplifier and equalizer are provided; the equalizer frequency characteristic of the probe-tube microphone and amplifier is substantially flat from about 80 to 600 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.

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 ARAEN exists and has been sent to CCITT Laboratory. This equipment should not be regarded as forming a specific part of the reference system.

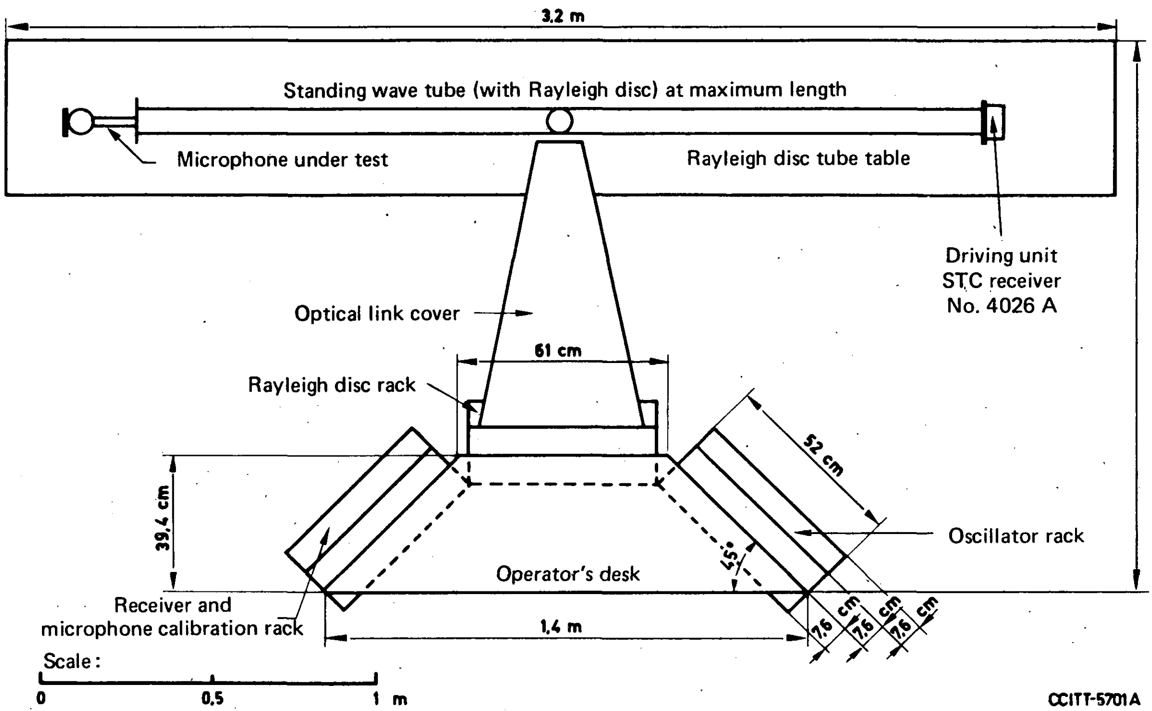


FIGURE 3/P.41 – **Plan of microphone and receiver calibration equipment ARAEN**

D. THEORETICAL EFFICIENCY OF THE COMPLETE ARAEN

The construction of the ARAEN 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 ARAEN represents that portion included between a point situated at 33.5 cm from the talker's lips (the position of the centre of the microphone) [1] and the head of the listener, the latter being situated at a point one metre from the talker's lips and facing the talker.

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

$$20 \log_{10} \frac{100}{33{,}5} = 9{,}5 \text{ dB}$$

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

Sensitivity of the ARAEN sending end – The sensitivity of the sending end of the ARAEN 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 "ARAEN reference vocal level" (see Recommendation P.45). Under these conditions the acoustic pressure applied to the diaphragm of the microphone is 0.1 Pascal (0.1 Pa).

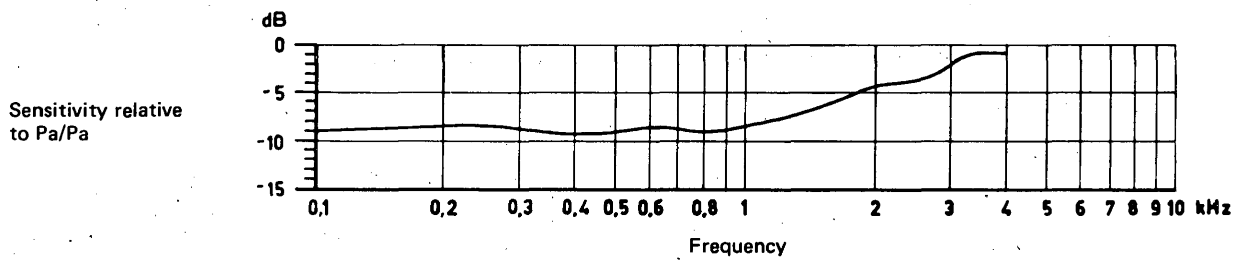
Sensitivity of the ARAEN receiving end – The sensitivity of the receiving end has been determined conventionally such that the condition indicated above (for the "air-to-air" efficiency of the ARAEN) is complied with for a junction attenuation equal to 30 dB.

---

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

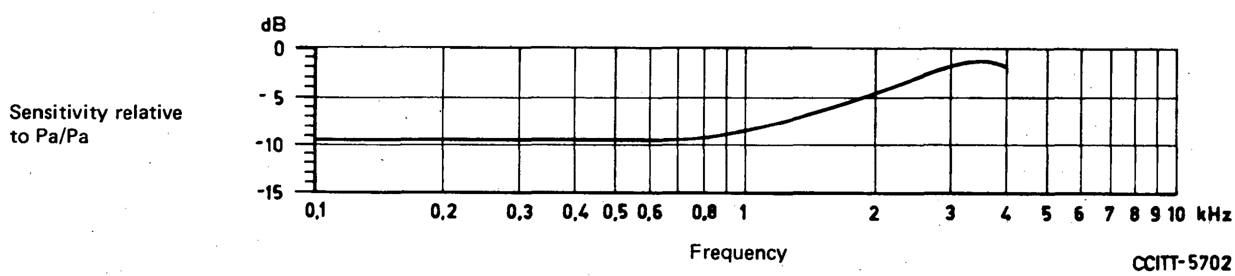
TABLE 2/P.41

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



a) Overall working characteristic of the ARAEN taken with microphone No. 1284 (type 4021 E) and a typical receiver (type 4026 A), the bandpass 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 ARAEN set up in accordance with the adjustments shown above).

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

FIGURE 4/P.41 – ARAEN

Table 3/P.41 gives the values of acoustic pressure (in decibels relative to 0.1 Pa) produced by a receiver when a level of −30 dB relative to 1 volt is applied to the input of the receiving system – i.e. when an acoustic pressure of 0.1 Pa is applied to the microphone.

TABLE 3/P.41

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 μbar/volt	dB relative to 1 μbar
100	−30	25.8	−55.8	46.0	−9.8
300	−30	25.2	−55.2	46.1	−9.1
1000	−30	19.5	−49.5	41.2	−8.3
2000	−30	15.4	−45.4	41.4	−4.0

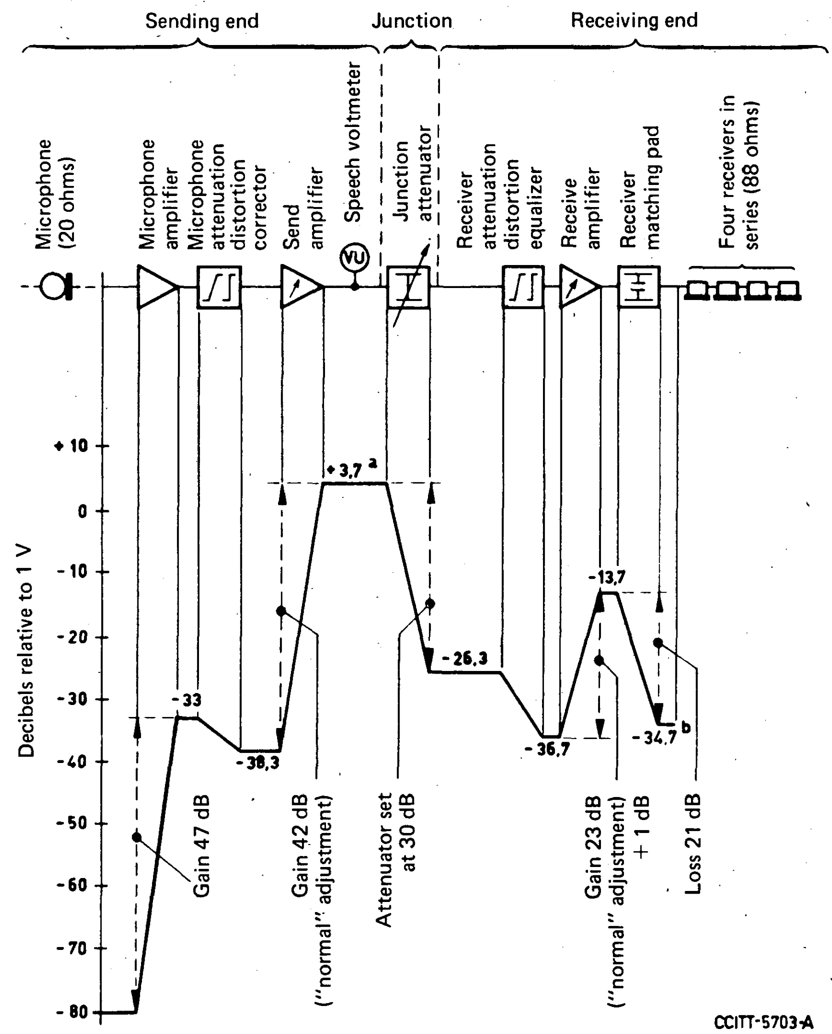
Table 4/P.41 below shows the comparison of the theoretical and actual values of the overall attenuation of the ARAEN.

TABLE 4/P.41

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

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

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



[a] 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 ARAEN.

[b] With tolerance of ± 1.0 dB without the bandpass filter in circuit.

Note. – The conditions of adjustment used are: send amplifier: "normal", receive amplifier: "normal" + 1 dB, junction attenuator: 30 dB.

FIGURE 5/P.41 – **Diagram showing the levels at various points in the ARAEN when a pure tone of 1000 Hz at a level of −80 dB relative to 1 V is applied to the microphone sockets**

Recommendation P.42

SYSTEMS FOR THE DETERMINATION OF REFERENCE EQUIVALENTS

(amended at Mar Del Plata, 1968)

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 (NOSFER);
2. primary systems for the determination of reference equivalents;
3. working standard systems.

The new fundamental system for the determination of reference equivalents (NOSFER) is the system used in the CCITT Laboratory. Formerly, reference equivalents were determined by comparison with the European master reference system for telephone transmission (SFERT), defined on pages 27 to 43 of Volume IV of the CCIF *Green Book.* [2)]

Values of reference equivalents determined by comparison, directly or indirectly, with the SFERT 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 CCIF *Green Book.*

A. THE NEW FUNDAMENTAL SYSTEM FOR THE DETERMINATION OF REFERENCE EQUIVALENTS (NOSFER)

This system consists of the ARAEN (described in Recommendation P.41) with the following modifications:

1. *Sending end*

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

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

The ARAEN volume-measuring set having the characteristics given in Supplement No. 10, *White Book*, Volume V, is bridged across the output terminals of the NOSFER sending system.

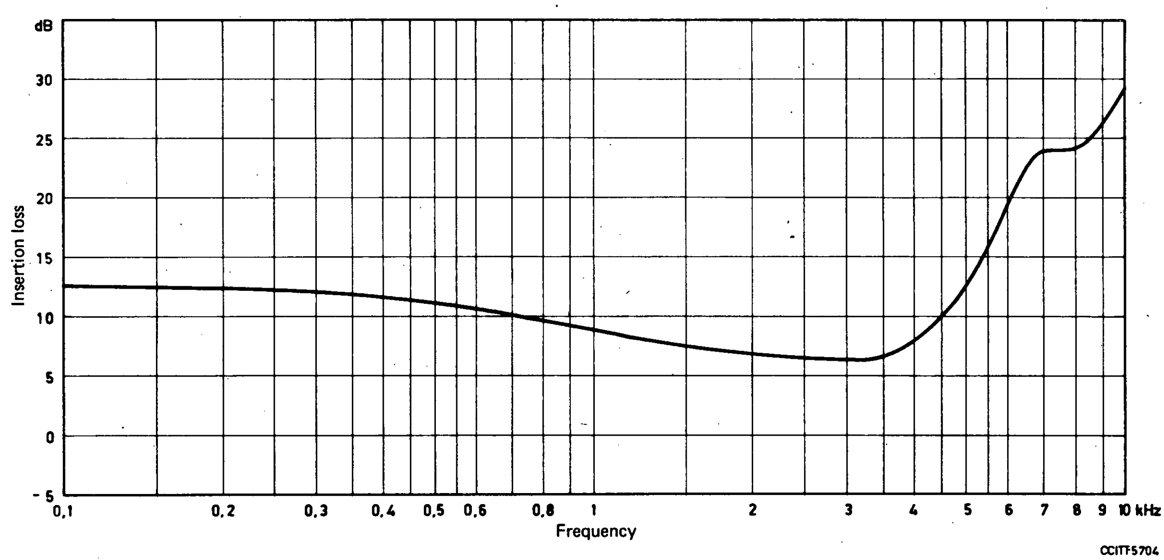


FIGURE 1/P.42 – **Insertion loss characteristic of the NOSFER sending end equalizer (measured between 600-ohm terminations)**

[2)] Since the CCITT Laboratory has high-quality transmission apparatus (ARAEN), it appeared reasonable to keep only one reference system at the CCITT Laboratory which, after appropriate modification, could replace the SFERT; tests have shown this to be possible.

The SFERT 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.

[3)] In the present constitution of the ARAEN, this network fulfils two functions:

a) it corrects the distortion of the ARAEN receivers, and

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

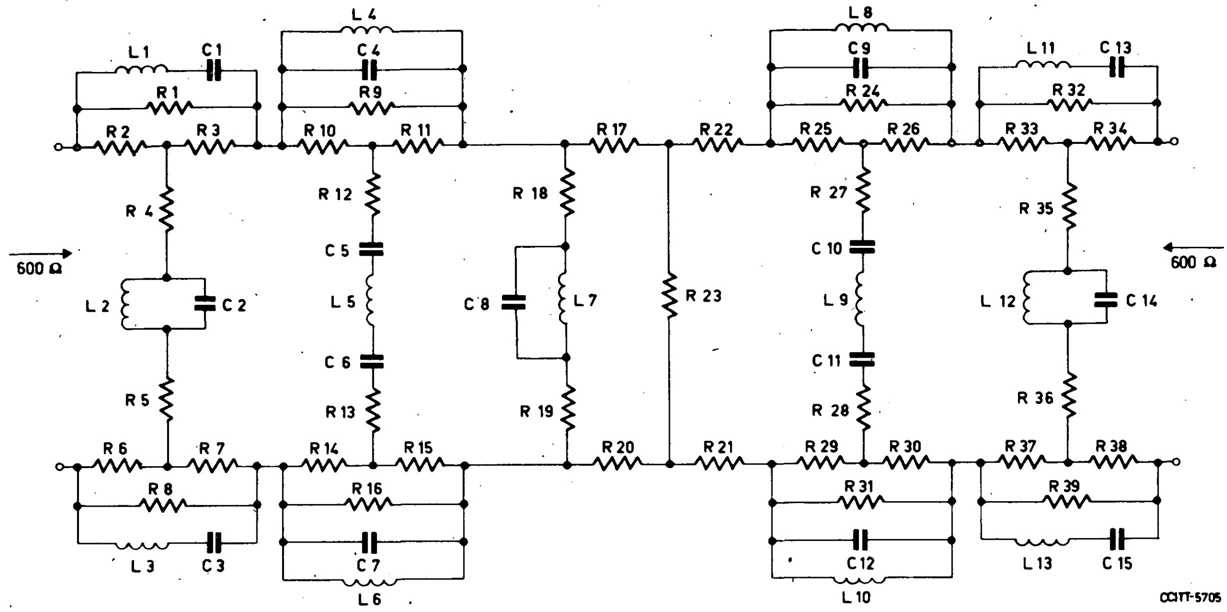


FIGURE 2/P.42 – **Circuit of the NOSFER sending end equalizer**

TABLE 1/P.42 – **Insertion loss of the NOSFER sending end equalizer**
(measured in the CCITT Laboratory between two non-reactive resistances of 600 ohms)

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

TABLE 2/P.42 – **Values of the components used in the NOSFER sending end equalizer** (Figure 2/P.42)

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

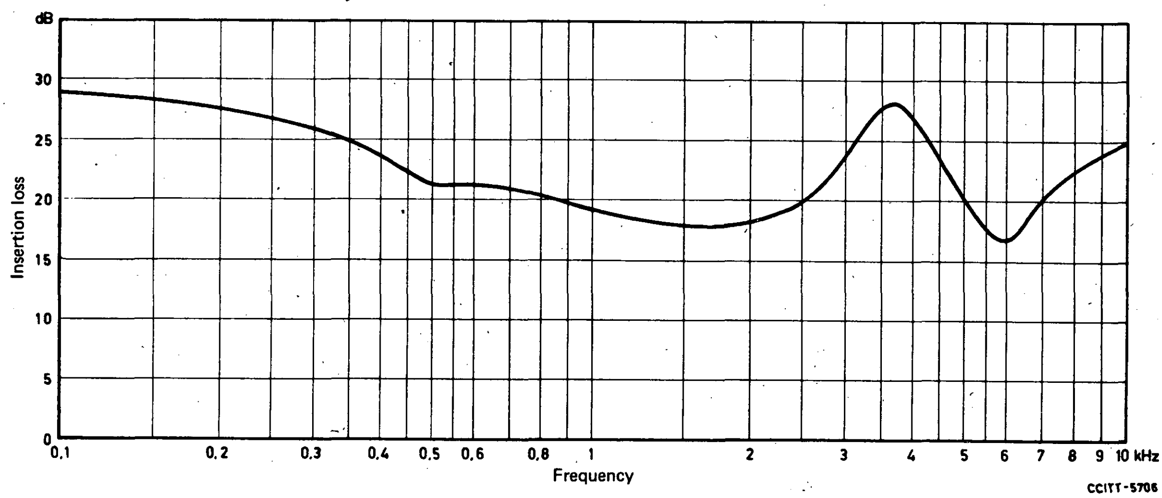


FIGURE 3/P.42 – **Insertion loss characteristic of the NOSFER receiving end equalizer (measured between 600-ohm terminations)**

2. *Receiving end*

An equalizer as defined by Figures 3/P/42 and 4/P.42 and Tables 3/P.42 and 4/P.42 is inserted at the input of the receiving amplifier in place of the ARAEN receiver equalizer 3) (see Figure 1/P.41). Unlike the ARAEN system, only one receiver is used in NOSFER; the other three receivers as used in the ARAEN system are replaced in this case by a resister of 66 ohms.

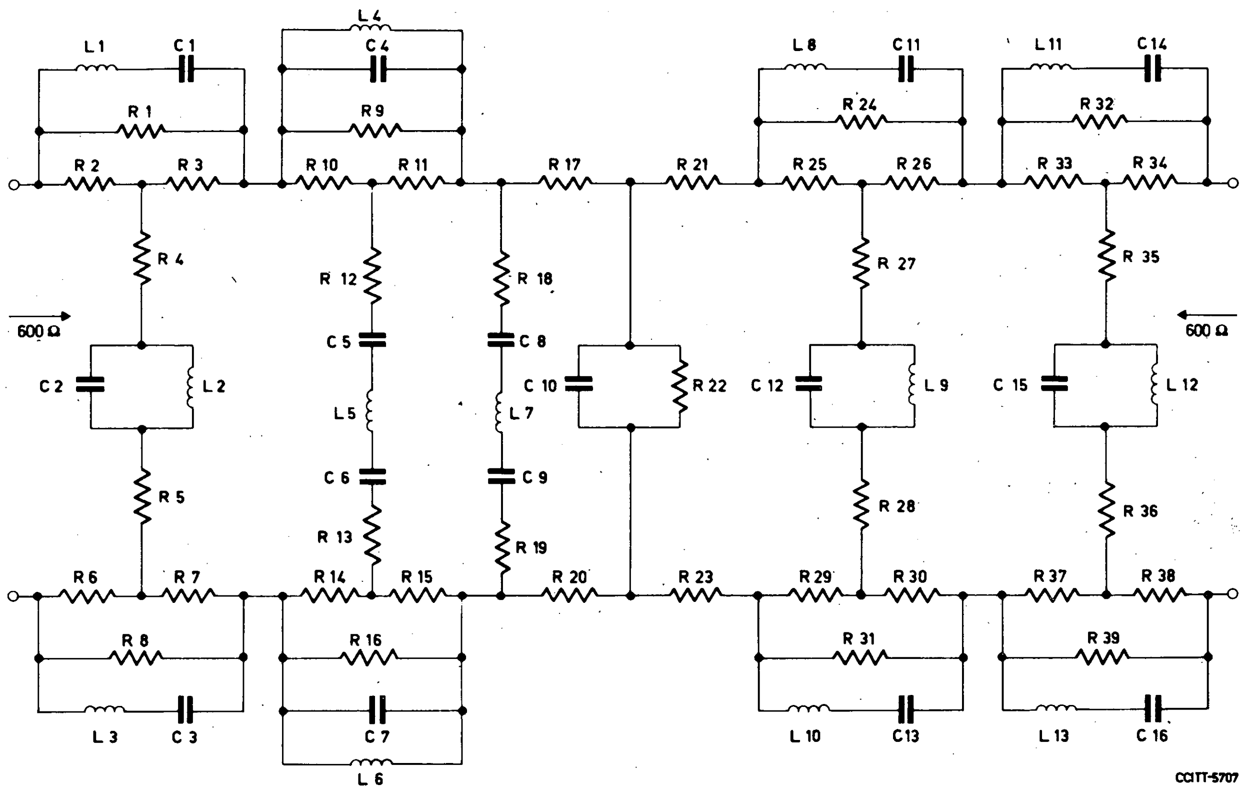


FIGURE 4/P.42 – **Circuit of the NOSFER receiving end equalizer**

TABLE 3/P.42 – **Insertion loss of the NOSFER receiving end equalizer**
(measured in the CCITT Laboratory between two non-reactive resistances of 600 ohms)

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

TABLE 4/P.42 – **Values of the components used in the NOSFER receiving end equalizer**
(Figure 4/P.42)

R (non-inductive)	ohm	L	mH	d.c. resistance in ohms	Q (at f_r)	f_r Hz	C	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 NOSFER

The ARAEN having been adjusted to take into account the characteristics of the microphone used, the equalizers described in 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 ARAEN (normal + 1 dB); the gain of the sending amplifier is not to be changed.

1. *Sensitivity of the NOSFER transmitting system*

As indicated above, the adjustment of the sending amplifier gain is not changed when passing from the ARAEN sending system to the NOSFER sending system.

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

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

The amplifier gain is adjusted according to the result of the following operations, described in Table 5/P.42:

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

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

TABLE 5/P.42

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/μbar) 2	Microphone equalizer attenuation (dB) 3	(2-3) 4	Correction to be made to sending amplifier adjustment (with microphone No. 1292) 5
100 300 900	89.0 89.0 89.0	−85.2 −81.1 −83.0	4.5 8.0 5.8	−89.7 −89.1 −88.8	−(−89.2)−89
Average	89.0	−83.1	6.1	−89.2	+0.2

TABLE 6/P.42 – **Characteristic values defining the variation, as a function of the frequency, of the sensitivity of the NOSFER transmitting system, calculated from the mean sensitivity values of a certain number of microphones measured in a free field**

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

[a] Values extracted from Research Report No. 13 200 of the United Kingdom Post Office (April 1950).

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

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

Table 6/P.42 gives the characteristic value defining variations, as a function of frequency, of the sensitivity of the NOSFER 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 7/P.42 gives the characteristic values defining the variation, as a function of the frequency, of the NOSFER sending system sensitivity of microphone No. 1292, measured in a free field (figures supplied by the United Kingdom Post Office) and also on a closed coupler. Transmitter amplifier gain is adjusted to the value corresponding to this microphone ("normal" +0.2).

TABLE 7/P.42 – **Characteristic values defining the variation, as a function of the frequency, of the sensitivity of the NOSFER 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)	Sensitivity of microphone No. 1292 measured in free speech field (dB relative to 1 volt/μbar)	Sensitivity of the sending system in the free speech field for the associated microphone No. 1292 (1 + 2)	Sensitivity of microphone No. 1292 measured on the closed coupler (dB relative to 1 volt/μbar)	Sensitivity of the sending system with microphone No. 1292 measured on the closed coupler (1 + 4)
	1	2	3	4	5
80	+73.0	−86.8	−13.8	−89.9	−16.9
100	+72.7	−85.2	−12.5	−87.7	−15.0
120	+72.2	−83.9	−11.7	−86.2	−14.0
200	+70.6	−81.6	−11.0	−83.3	−12.7
300	+69.3	−81.1	−11.8	−82.6	−13.3
400	+69.4	−81.5	−12.1	−82.6	−13.2
500	+70.2	−81.1	−10.9	−82.6	−12.4
600	+71.3	−81.0	−9.7	−82.6	−11.3
700	+72.4	−81.7	−9.3	−82.7	−10.3
800	+73.5	−82.6	−9.1	−82.8	−9.3
900	+74.3	−83.0	−8.7	−83.0	−8.7
1000	+75.2	−83.2	−8.0	−83.2	−8.0
1500	+77.7	−85.6	−7.9	−84.6	−6.9
2000	+79.0	−86.7	−7.7	−85.8	−6.8
2500	+79.7	−87.8	−8.1	−86.2	−6.5
3000	+80.0	−86.6	−6.6	−85.9	−5.9
3500	+80.0	−85.3	−5.3	−85.3	−5.3
4000	+78.9	−85.0	−6.1	−85.0	−6.1
4500	+77.0	−84.9	−7.9	−84.6	−7.6
5000	+74.3	−84.7	−10.4	−84.1	−9.8
5500	+71.2	−86.0	−14.8	−83.0	−11.8
6000	+67.3	−84.8	−17.5	−79.2	−11.9
6500	+64.8	−83.2	−18.4	−76.6	−11.8
7000	+62.7	−84.7	−22.0		

Table 8/P.42 gives, for information, the sensitivity of the sending system determined from measurements made in the anechoic chamber, and with the artificial mouth used by the Swiss Administration, the microphone being placed at 14 cm from the mouth with its protective grille placed horizontally. The acoustic pressure was measured before the microphone was put into position.

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

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

TABLE 8/P.42

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

Figure 5/P.42 gives the sensitivity/frequency characteristics of the NOSFER sending end, calculated from the sensitivity values of the microphone as measured under various calibration conditions.

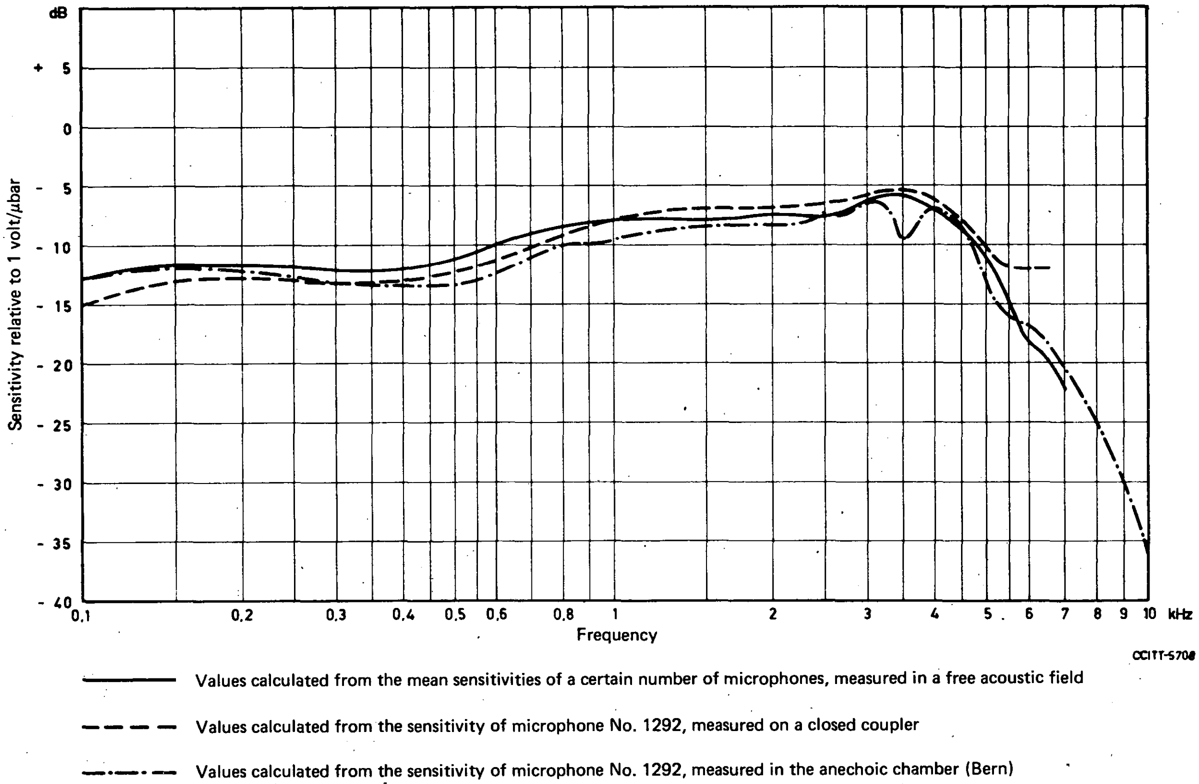


FIGURE 5/P.42 – **Response curve of the NOSFER sending end**

2. *Sensitivity of the NOSFER 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 NOSFER receiving system is adjusted to the fixed value of 37 dB.

Table 9/P.42 (column 5) gives the characteristic values of the sensitivity of the NOSFER receiving system. The sensitivity values of the receiver taken into account in the calculation are extracted from Research Report No. 13200 (April 1950) of the United Kingdom Post Office.

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

In practice the receivers 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 in relation to 1 µbar per volt.

Table 10/P.42 gives the characteristic values defining the sensitivity of the NOSFER receiving system, with the particular set of four different receivers which are in the possession of the CCITT Laboratory.

TABLE 9/P.42

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

Note 1. – In the same conditions, a vu measuring set (see Supplement No. 11, *White Book*, Volume V), connected at the output of the sending system of NOSFER 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, when connected at the output of the sending system of SFERT, should give a reading of −16 dB.

Note 3. – The relationships between the readings of the ARAEN 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 obtained from the indications of the various types of volume measuring sets, during a telephone conversation, are given in Supplement No. 14, *White Book*, Volume V.

TABLE 10/P.42

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. 936	946	1039	1140	Sensitivity of the receivers (dB relative to 1 μbar/volt) Receivers Nos. 936	946	1039	1140	Sensitivity of the receiving system with each of the 4 receivers (dB relative to 1 μbar/volt) Receivers Nos. 936	946	1039	1140	Mean sensitivity of the receiving system with the 4 receivers (dB relative to 1 μbar/volt)
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	+26.8	+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=1.5 dB	b=1.5 dB	b=2.0 dB	b=2.0 dB					

NOSFER

Figure 6/P.42 gives the sensitivity/frequency characteristics of the NOSFER receiver system.

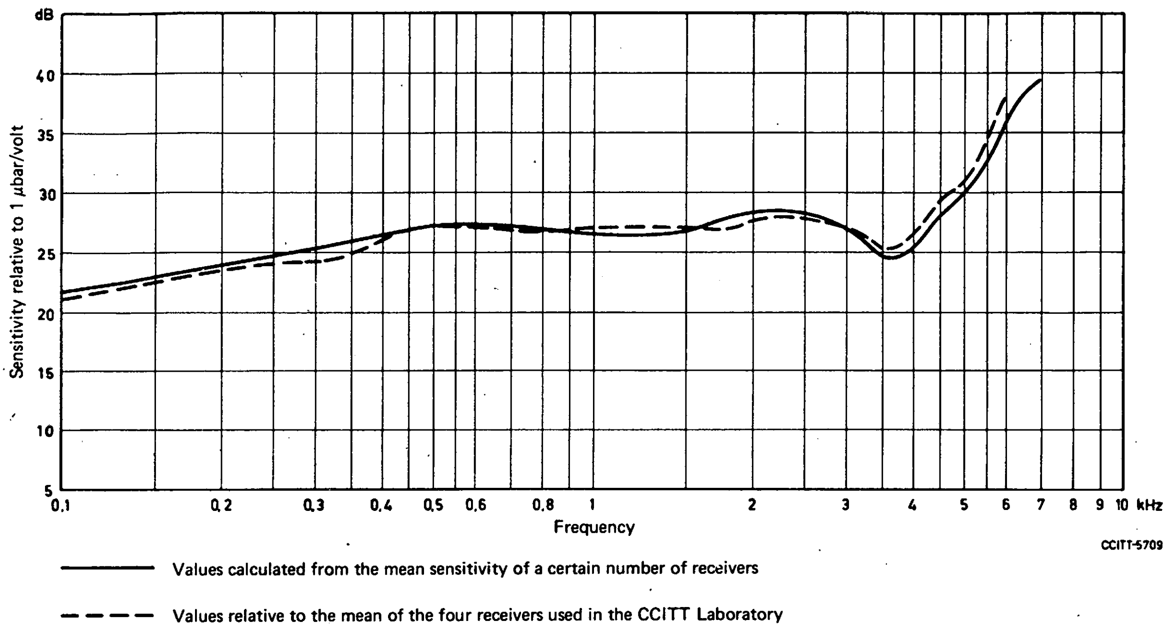


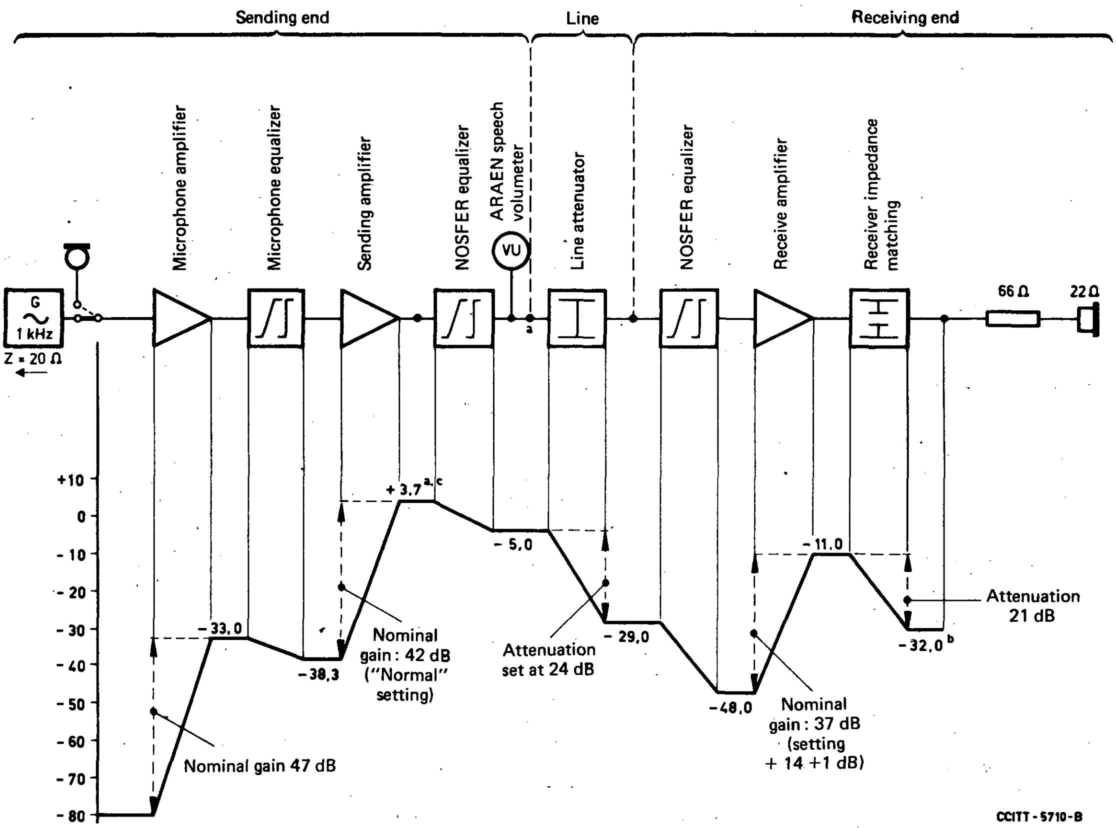
FIGURE 6/P.42 – Frequency response curve of the NOSFER receiving system (values calculated from the calibration of the receivers on the ARAEN artificial ear)

3. *Level diagram of NOSFER*

Figure 7/P.42 gives the theoretical level diagram of NOSFER.

C. NORMAL SPEECH POWER FOR VOICE-EAR MEASUREMENTS

The volume measuring set of ARAEN is connected at the output of the sending system of NOSFER. The sensitivity controls of the volume measuring set must be adjusted to -10 dB and the operator speaks at the microphone of the sending system of NOSFER 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".



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

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

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

Note. – The conditions of adjustment used are: send amplifier: "normal", receive amplifier: "14 dB + 1 dB", line attenuator: 24 dB.

FIGURE 7/P.42 – **Level diagram for the NOSFER 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**

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 NOSFER,

b) a system conforming to the description given in 3.1.1.II of the CCIF *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 NOSFER.

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 NOSFER.

E. WORKING STANDARD SYSTEMS

It is admitted for the purposes of the application of 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 SFERT should be compared with the NOSFER 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 NOSFER or a primary system for the determination of reference equivalents. It indicates in decibels the amount by which the respective sending or receiving system of the working standard is worse or better relative to the sending or receiving system of NOSFER (or a primary system for the determination of reference equivalents).

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

The tests are carried out by telephonometric comparison (voice and ear tests), substituting the component to be compared (sending or receiving system) for the corresponding component of NOSFER. 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 SFERT, are shown in Figures 8/P.42 and 9/P.42 respectively.

The method of comparison employed in the CCITT 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 variable from 0 to 34 dB in 1-dB steps. The second attenuator S which may vary as well between 0 and 34 dB in 1-dB steps, introduces the hidden-loss; this loss as well as the value set for the attenuator A are not known to the listening operator and the total value of $A + S$ may vary between 24 dB and 34 dB of attenuation. The third attenuator E, called a "balancing attenuator", is adjusted by the listening operator and is to enable equality of loudness to be obtained.

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

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

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

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

Each individual balance is carried out between two operators. The talker repeats a predetermined sentence [4] in front of each microphone alternately; the hidden loss is set at a particular value.

The total attenuation inserted between the sending system to be measured and the NOSFER receiving system varies between 24 dB and 34 dB (according to the attenuation of the hidden loss). This method is used in the CCITT Laboratory so as to leave a greater margin of variation in the balancing attenuator, which appeared necessary with apparatus having a reference equivalent in the neighbourhood of that of the standard system: $S + A$ of the attenuations of the lines S and A varies between 24 dB and 34 dB; thus, the hidden loss can vary from 0 to 34 dB.

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

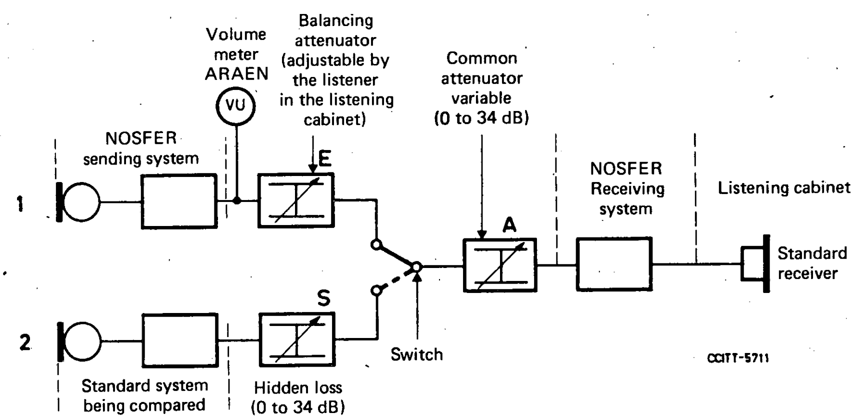


FIGURE 8/P.42 – **Comparison of a standard sending system with a NOSFER sending system (method termed "two-operator-hidden-loss method")**

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

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

[4] In the CCITT Laboratory the conventional sentence is as follows: Paris, Bordeaux, Le Mans, Saint-Leu, Léon, Loudun.

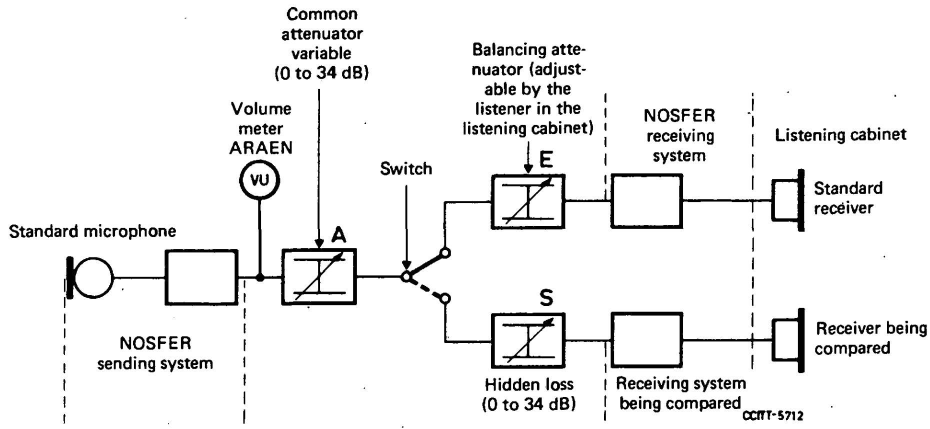


FIGURE 9/P.42 – Comparison of a standard receiving system with a NOSFER receiving system (method termed "two-operator-hidden-loss method")

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

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

In each replication, the results are entered in appropriate forms, on which the hidden loss values and balance attenuations are shown respectively for each elementary balance. The value of the reference equivalent for a replication is the arithmetical mean of the values obtained for all the elementary balances of the replication concerned. When a single replication does not suffice to determine the reference equivalent, two replications are carried out in periods with a spacing of one week between the two. The test results are then submitted to statistical analysis. The test results and the statistical analysis are sent to Administrations in the form of a technical report by the CCITT 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 (SETAB or SETAC 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) *Periodic calibration of working standard systems*

Working standard systems must be periodically compared against the international telephonometric standard consisting of NOSFER 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 (SETAB)

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

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

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

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

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

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

ANNEX 2

(to Recommendation P.42)

Description of a working standard having an electro-dynamic microphone and receiver (SETED)

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

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

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

A vu meter connected to the output of the sending part of SETED allows the talking level to be monitored and controlled.

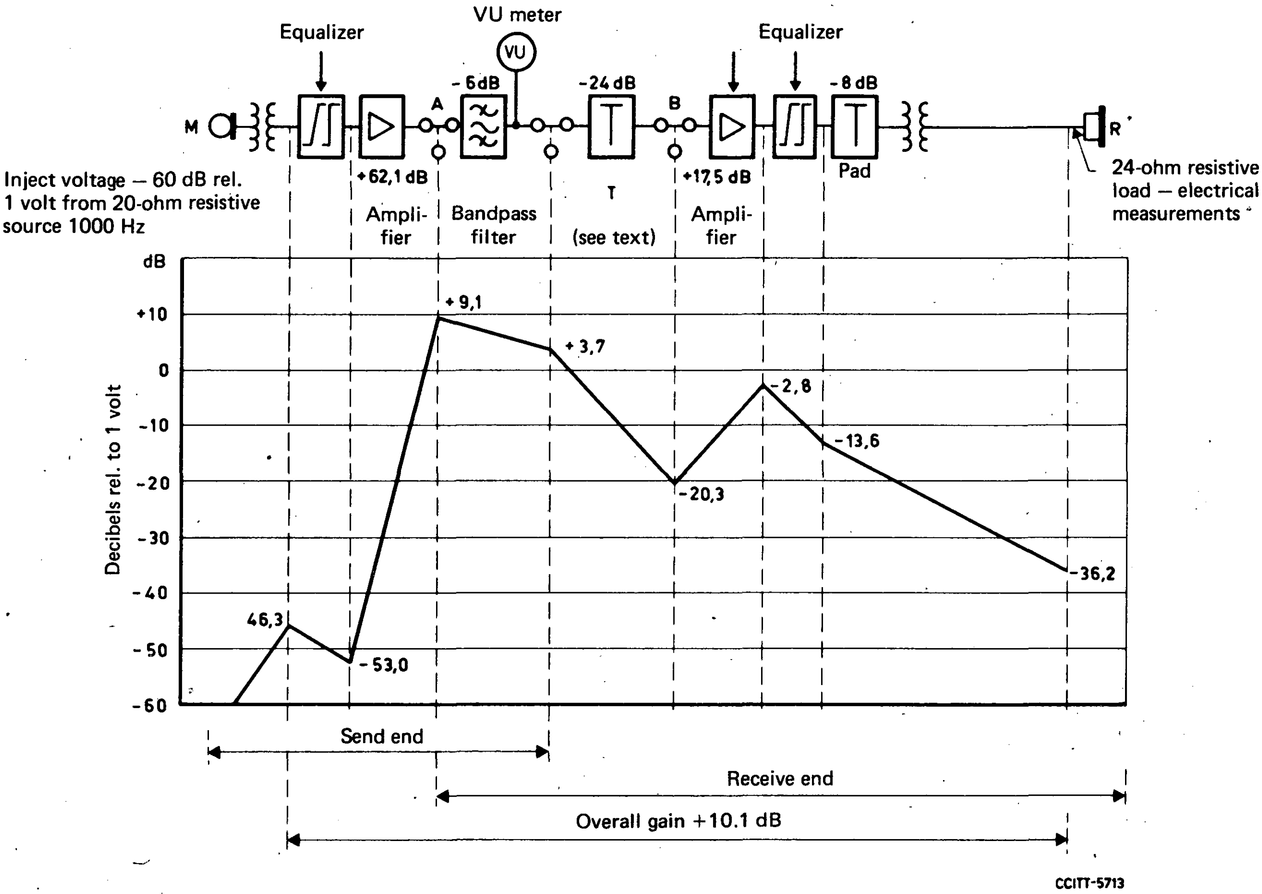


FIGURE 1 – **SETED, distribution of gains and levels**

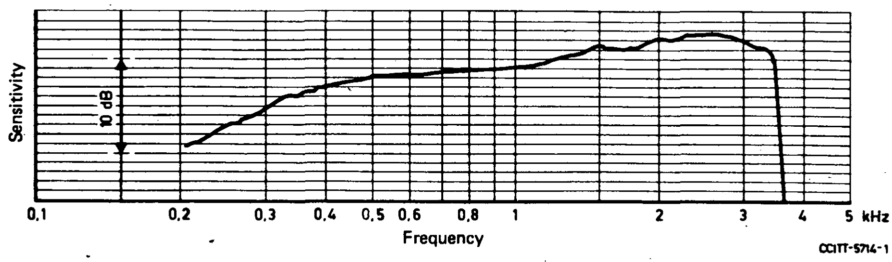
The distribution of gains and losses throughout the SETED working standard are given in Figure 1, and the model at the CCITT Laboratory is within 0.1 dB of these design figures. Sensitivity/frequency characteristics have been determined for the send and receive ends and the overall connection of the SETED and are shown in Figures 2, 3 and 4 respectively. For these measurements the sensitivities quoted under each curve refer to the condition with the bandpass filter and its inband loss of 6 dB included, and the trunk attenuator (*T*) set to zero.

Reference equivalents determined for the SETED by direct comparison against the NOSFER in the CCITT Laboratory in 1973 were as follows referring to Figure 1:

Sending reference equivalent:	M → B =	+7.8 dB, quieter than NOSFER
Receiving reference equivalent:	A → R =	+4.5 dB, quieter than NOSFER
Overall reference equivalent:	M → R =	+0.8 dB, quieter than NOSFER

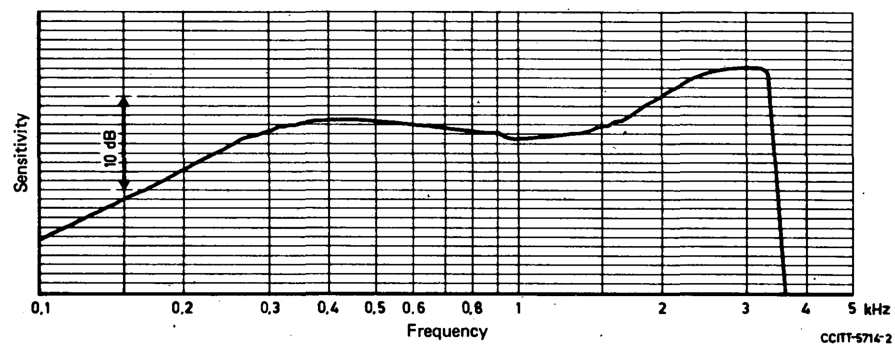
The determinations were carried out with loudness balancing at constant listening level employing a trunk attenuation (*T*) of 24 dB in SETED and varying the loss in the appropriate NOSFER path to obtain a loudness balance.

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



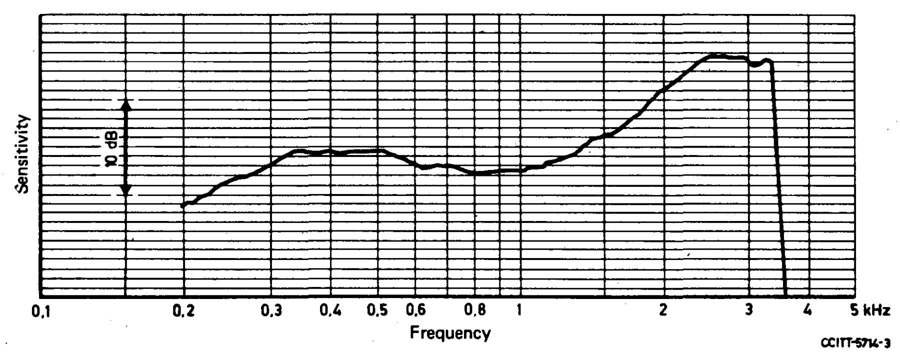
Sensitivity at 1000 Hz = −6.0 dB rel. 1 Volt/Pa referred to 1 Pa free field sound pressure 25 mm on axis in front of artificial mouth (Bruel & Kjaer 4219). At 1000 Hz reference sound pressure level caused reading of −2.8 dB on SETED.

FIGURE 2 – **Sensitivity/frequency characteristic of the send end (M → B) of the SETED (T = 0 dB)**



Sensitivity at 1000 Hz = +7.7 dB rel. 1 Pa/volt with receiver loaded in IEC artificial car (no leak). At 1000 Hz, 1 volt input at A caused a reading of −3.6 dB on SETED vu meter.

FIGURE 3 – **Sensitivity/frequency characteristic of the receive end (A → R) of the SETED (T = 0 dB)**



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

FIGURE 4 – **Sensitivity/frequency characteristic of the overall connection (M → R) of the SETED (T = 0 dB)**

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 ny 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 CCITT 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 CCITT Laboratory but with an indefinite variety of teams, each of n operators, drawn from the same population as the sample of operators used in the test analyzed.

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

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

Recommendation P.43

INSTRUCTIONS FOR FORWARDING STANDARD SYSTEMS AND COMMERCIAL TELEPHONE APPARATUS TO THE CCITT LABORATORY TO HAVE THE REFERENCE EQUIVALENTS OF THESE SYSTEMS DETERMINED

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

A. PRIMARY SYSTEMS FOR THE DETERMINATION OF REFERENCE EQUIVALENTS

If an Administration wishes to have the reference equivalent of its primary system determined, assuming that the system concerned can be transported without risk of deterioration, it may supply the CCITT 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 ARAEN (Supplement No. 10, *White Book*, Volume V), the volume meter must be sent to the CCITT 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 microphone 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 NOSFER (or a primary system for the determination of reference equivalents).

Administrations wishing to have their systems calibrated (or recalibrated) by the CCITT Laboratory should follow the instructions given under A above.

2. *Working standard systems using carbon microphones*

2.1 *Working standard systems using subscribers' sets (SETAB)*

When a SETAB system is set up, the Administration should first make preliminary checks to see whether the microphones and receivers are stable, whether they are subject to frying, and whether the transmission quality is acceptable. These tests should be spread over a fairly long period (six months).

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

- six subscribers' handsets equipped with a microphone and receiver (each bearing a number);
- six feeding bridges (with an indication of their characteristics);
- where necessary, six artificial subscribers' lines if the system concerned comprises such lines;
- a ring-guard for the reference equivalents;
- a guard-ring for the AENs, should the Administration want the CCITT Laboratory to carry out articulation tests using the method specified for AEN;
- the associated volume meter.

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

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

- three systems as a sending standard,
- three systems as a receiving standard, when reference equivalents are measured, or
- one system as a sending standard,
- four systems as a receiving standard, in the case of AEN measurements.

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

- three subscriber's 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 (SETAC)*

The CCITT does not recommend the use of such systems as working standard systems; however, Administrations which still use them and which wish to have their microphones and receivers recalibrated should send only the microphones and receivers to the CCITT Laboratory, as the latter already has some SETAC systems (see Volume IV of the CCIF *Yellow Book*, Paris, July 1949).

General comment on Divisions A and B

The object of the general recommendations given above is to guide Administrations. When an Administration wishes to have a system for the determination of reference equivalents calibrated (or recalibrated) it should get into touch with the CCITT 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 (NOSFER). This being so, it is desirable for the technical conditions to be defined by agreement between the Administration and the CCITT Laboratory.

The cost of determining reference equivalents in the CCITT 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 AEN (SRAEN)

The reference system for the determination of AEN (SRAEN) is a system consisting of the following elements:

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

The schematic diagram of the ARAEN is given in Figure 1/P.44.

a) *ARAEN*

The ARAEN is described in detail in Recommendation P.41; this also contains a definition of the normal adjustment of the ARAEN.

b) *300 to 3400-Hz bandpass filter*

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

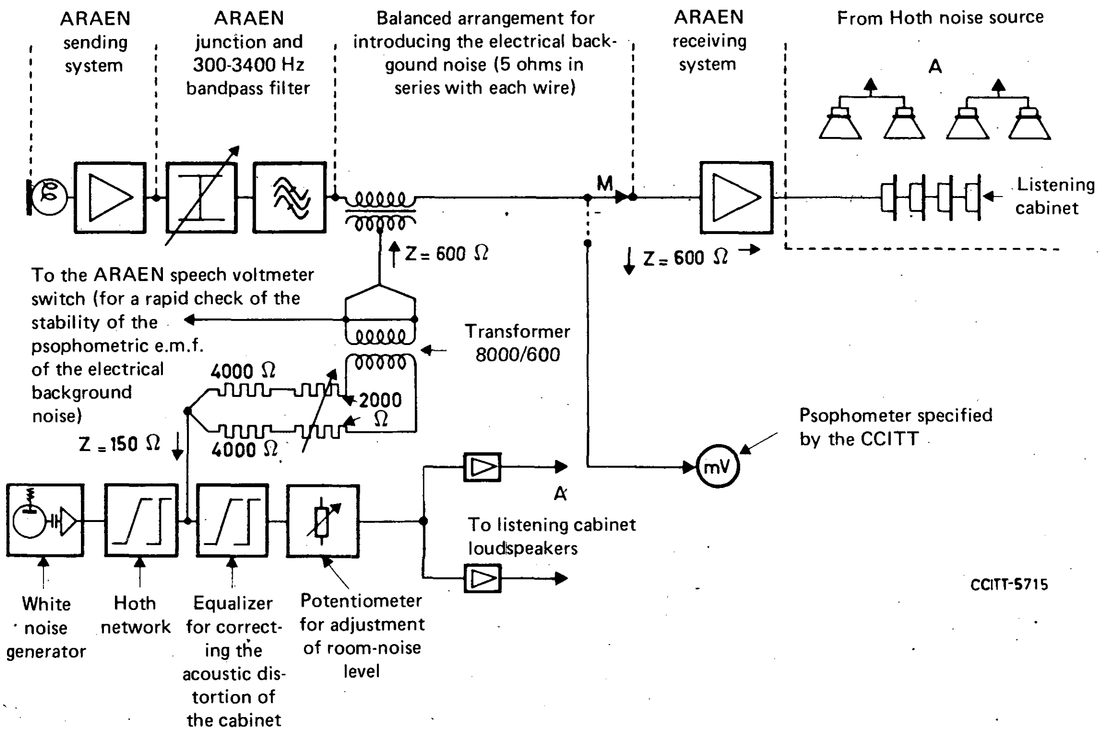


FIGURE 1/P.44 – **Schematic diagram of the SRAEN including the arrangement for injecting electrical background noise into the ARAEN, and the measurement of the psophometric voltage of this noise**

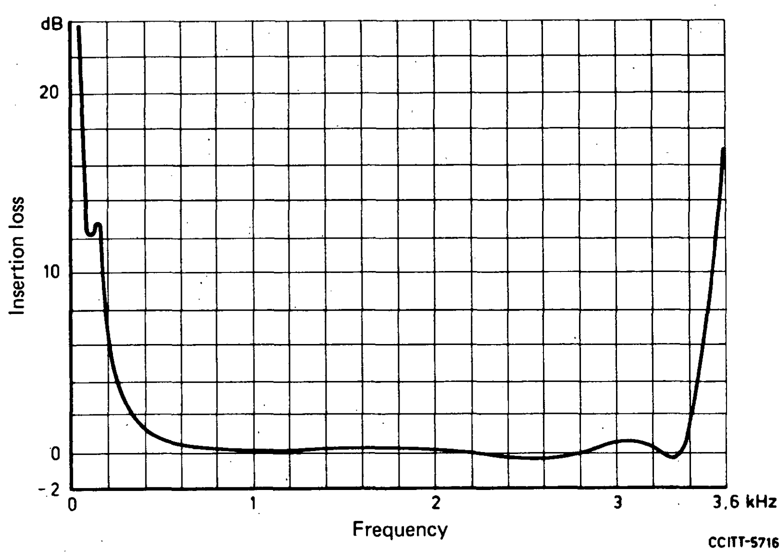


FIGURE 2/P.44 – **Insertion loss (between 600-ohm terminations) of the 300 to 3400-Hz bandpass filter**

c) *Electrical background noise*

At the input of the ARAEN 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 CCITT for commercial telephone circuits (see Recommendation P.53). Figure 3/P.44 gives the mean power density spectrum observed at subscribers' telephone stations (Hoth spectrum) (curve *a*) together with typical graphs *b* and *c* obtained at the CCITT Laboratory with two sets of half-octave filters.

Note. – Administrations may consider the use of other working standards for the determination of AEN values, these systems being capable of being calibrated by comparison with the SRAEN.

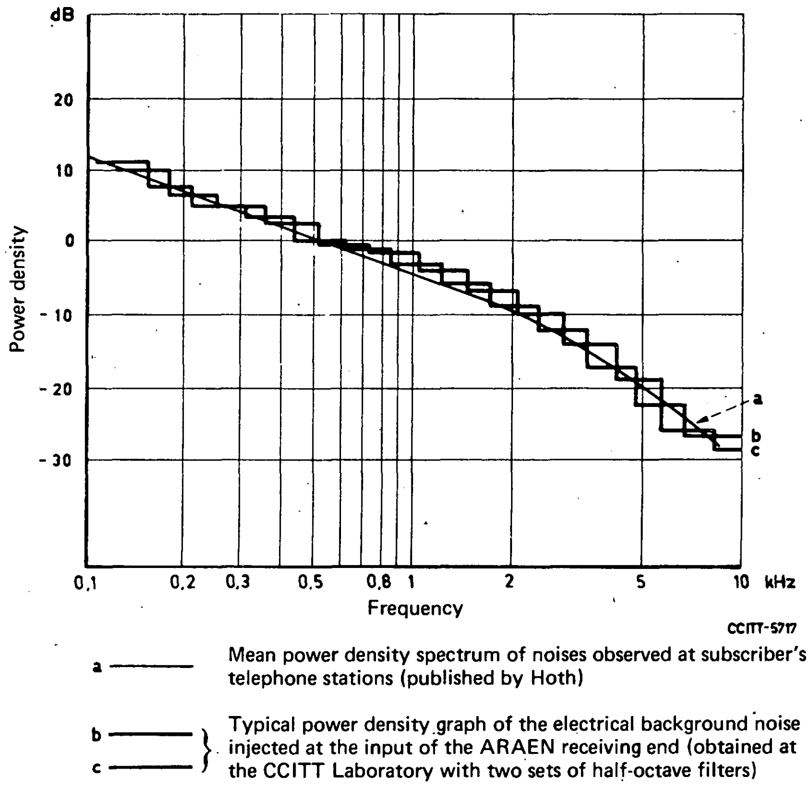


FIGURE 3/P.44 – **Power density spectrum of the electrical background noise injected at the input of the ARAEN receiving end**

Recommendation P.45

MEASUREMENT OF THE AEN VALUE OF A COMMERCIAL TELEPHONE SYSTEM (SENDING AND RECEIVING) BY COMPARISON WITH THE SRAEN

(amended at Mar del Plata, 1968)

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 AEN value is determined by the mean values of the following parameters (defined in Recommendation P.72):

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

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

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

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

The speech power used will be the reference vocal level for ARAEN. – The reference vocal level for ARAEN 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 sets), a deflection of the needle of the indicating instrument of the specified speech voltmeter (see Supplement No. 10, *White Book*, Volume V) connected to a specified microphone and amplifier system equal to that obtained when an acoustic pressure of 1 μbar at 1000 Hz is continuously applied at this same point.

c) *Mounting of the telephone handsets*

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

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

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

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

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

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

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

Note. – The position of the guard-ring with respect to the handset is determined uniquely for AEN 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:

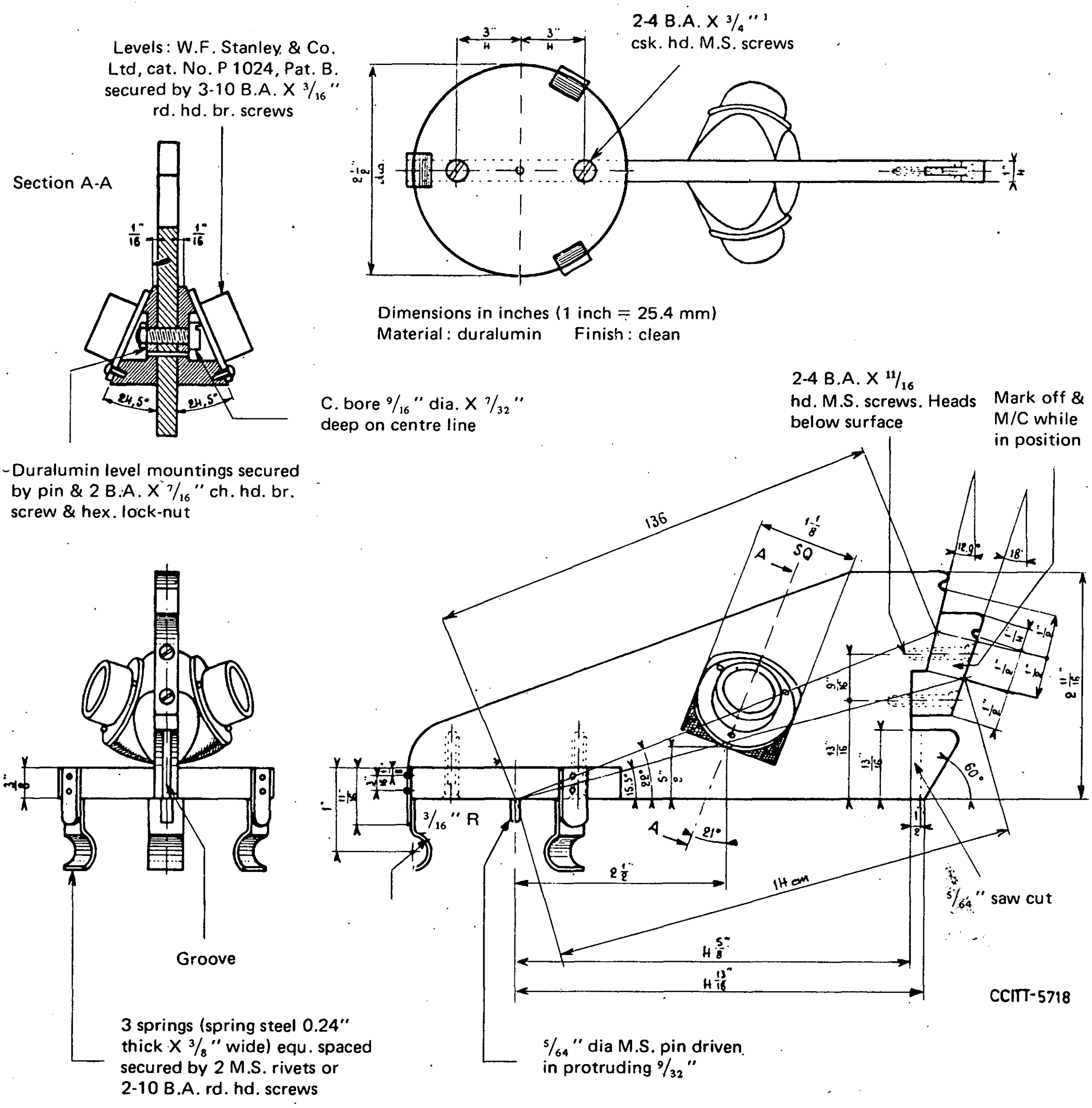
1) the plane of the guard-ring is vertical;

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.

[e) and f) of the former Recommendation have not been reproduced.]



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

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

g) *Noise at the receiving end*

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

For measurements using a *commercial receiving circuit*, a room noise is used at the receiving end only. This room noise should have a power density spectrum corresponding to that published by Hoth; this is reproduced in Figure 2/P.45 which also shows the spectral distribution, curve a, of a typical room noise measured in the listening cabinet of the CCITT 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} μbar 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 CCIF (Geneva, October 1954), the CCIF Laboratory determined AEN values in all cases (sending and receiving) with room noise at the receiving end; the present method introduces, with respect to the values previously measured, a difference of −2 dB in the receiving transmission performance rating of a commercial telephone circuit.

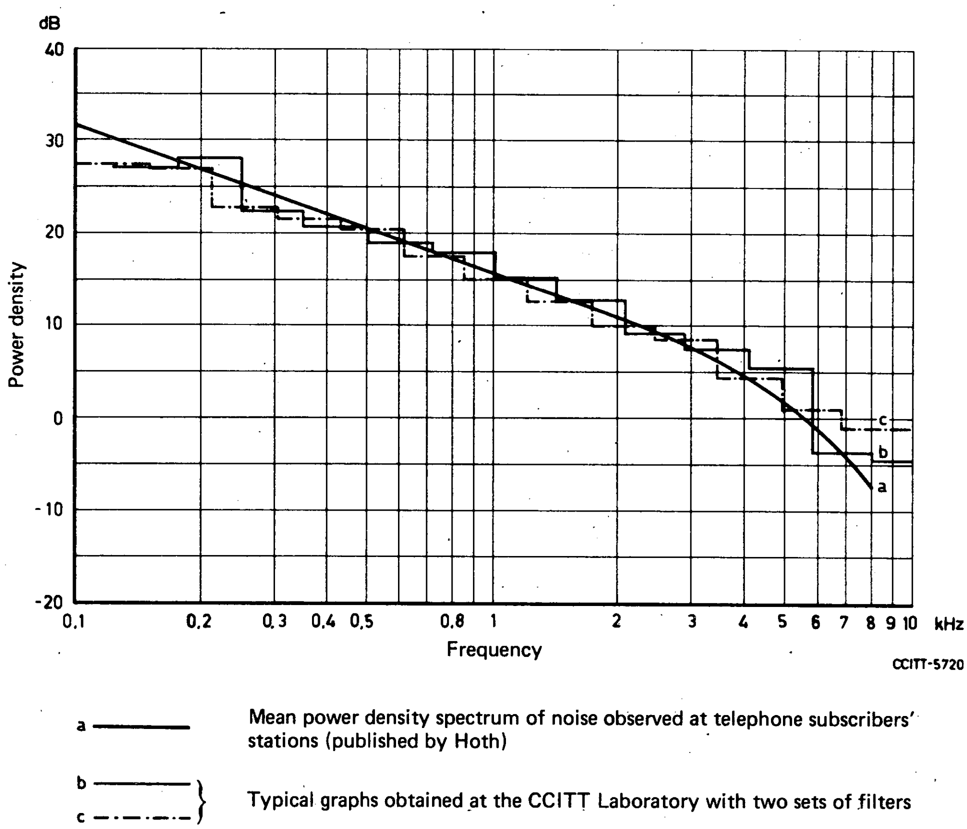


FIGURE 2/P.45 – **Power density spectrum of room noise produced in the listening room of the CCITT Laboratory**

h) *Junction*

The junction used throughout the tests comprises a bandpass filter 300-3400 Hz and a variable distortionless attenuator (the junction of the ARAEN). This junction has always the same composition whatever the system, SRAEN or commercial, under test.

Recommendation P.47

CHARGES FOR THE DETERMINATION OF REFERENCE EQUIVALENTS AND AENS (SENDING AND RECEIVING) OF WORKING STANDARD SYSTEMS AND COMMERCIAL TELEPHONE CIRCUITS IN THE CCITT LABORATORY

These costs are assessed on the basis of the number of hours of work carried out in the CCITT Laboratory; cost per hour of work of the CCITT Laboratory team (of five technical operators) is assessed periodically in Swiss francs (general running costs of the CCITT 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 recalibrating equipment.

a) *Calibration of systems using carbon microphones (SETAB or SETAC)*

- calibration test (sending): 5 hours;
- calibration test (receiving): 5 hours.

b) *Recalibration of systems using carbon microphones (SETAB or SETAC)*

- recalibration test (sending): 3 hours;
- recalibration test (receiving): 3 hours.

c) As regards the calibration or recalibration 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 measurement, in agreement with the Administration concerned.

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

a) measurement of AEN (sending): 28 hours;

b) measurement of AEN (receiving): 28 hours;

c) measurement of AEN for a complete telephone system: 35 hours.

Recommendation P.48

SPECIFICATION FOR AN INTERMEDIATE REFERENCE SYSTEM

(Geneva, 1976)

Summary

This Recommendation intends to specify the intermediate reference system (IRS) to be used for defining loudness ratings. The description should be sufficient to enable equipment having the required characteristics to be reproduced in different laboratories and maintained to standardized performance. The sensitivity/frequency characteristics of the sending and receiving parts conform with those agreed upon provisionally by Study Group XII at Munich (1974).

CONTENTS

1. *Design objectives*

The chief requirements to be satisfied for an intermediate reference system to be used for tests carried out on handset telephones [5] are as follows:

a) the circuit must be stable and specifiable in its electrical and electro-acoustic performance. The calibration of the equipment should be traceable to national standards;

b) the circuit components that are seen and touched by the subjects should be similar in appearance and "feel" to normal types of subscribers' equipment;

c) the sending and receiving parts should have frequency bandwidths and response shapes standardized to represent commercial telephone circuits;

d) the system should include a junction which should provide facilities for the insertion of loss, and other circuit elements such as filters or equalizers;

e) the system should be capable of being set up and maintained with relatively simple test equipment.

Since the detailed design of an IRS may vary between different Administrations, the following specification defines only those essential characteristics required to ensure standardization of the performance of the IRS.

The principles of the IRS are described and its nominal sensitivities are given in 2., 3., 4. and 5. below; requirements concerning stability, tolerances, noise limits, crosstalk and distortion are dealt with in 6. to 9. below. Some information concerning secondary characteristics is given in 10. below.

Certain information concerning installation and maintenance are given in Supplement No. 1.

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[5] For other types of telephone, e.g. headset or loudspeaking telephone, a different IRS will be required.

2. *Use of the IRS*

The basic elements of the IRS comprise:

a) the sending part,
b) the receiving part,
c) the junction.

When one example each of a), b) and c) are assembled, calibrated and interconnected, a reference (unidirectional) speech path is formed, as shown in Figure 1/P.48. For performing loudness rating determinations, suitable switching facilities are also required to allow the reference sending and receiving parts to be interchanged with their commercial counterparts.

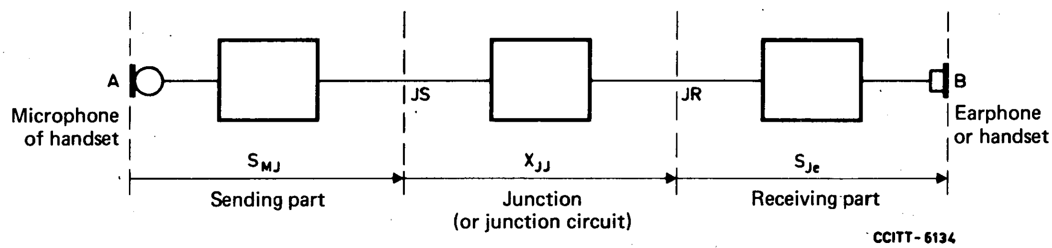
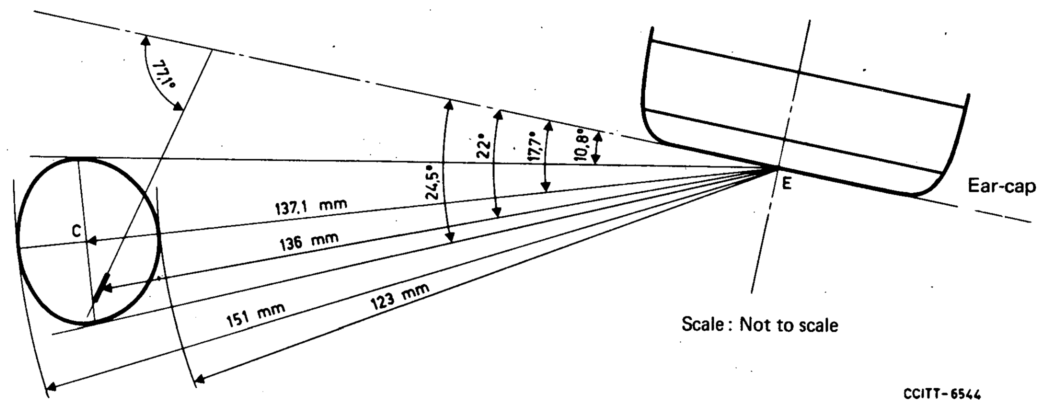


FIGURE 1/P.48 – Composition of the complete intermediate reference system

3. *Physical characteristics of handsets*

The sending and receiving parts of an IRS shall each include a handset symmetrical about its longitudinal place and the profile produced by a section through this plane should, for the sake of standardization, conform to the dimensions indicated in Figure 2/P.48. In practice, any convenient form may be considered use being made, for example, of handsets of the same type as those used by an Administration in its own network. The general shape of the complete handset shall be such that, in normal use, the position of the ear-cap on the ear shall be as definite as possible, and not subject to excessive variation.



Notes

1. The guard ring is shown in the 'special guard ring' position.
2. The boundary of the ellipse shown encloses 80% of observations on a sample of heads.
3. The centre C of the ellipse is located as shown above.
4. The minor axis of the ellipse is 28 mm and is colinear with the line joining the centre C of the ellipse with the ear reference point E.
5. The major axis of the ellipse is 33 mm.
6. The contour of the mouthpiece shall preferably just touch the ellipse. In any case, it should not overlap, or be separated from the ellipse by more than 5 mm.

FIGURE 2/P.48 – Location of ellipse defining certain preferred dimensions for the IRS handset

The microphone capsule, when placed in the handset, shall be capable of calibration in accordance with the method described in Recommendation P.64. The ear-cap shall be such that it can be sealed on the circular knife-edge of the IEC/CCITT artificial ear for calibration in accordance with Recommendation P.64, and the contour of the ear-cap shall be suitable for defining the ear reference point as described in the Appendix to Recommendation P.64.

Transducers shall be stable and linear, and their physical design shall be such that they can be fitted in the handset chosen. A handset shall always contain both microphone and earphone capsules, irrespective of whether either is inactive during tests. The weight of a handset, so equipped, shall not exceed 350 g.

4. *Subdivision of the complete IRS and impedances at the interfaces*

Figure 1/P.48 shows the composition of the complete IRS, subdivided as specified in 2. above. The principal features of the separate parts are considered below.

4.1 *Sending part*

The sending part of the IRS is defined as the portion A-JS extending from the handset microphone A to the interface with the junction at JS. The sending part shall include such amplification and equalization as necessary to ensure that the requirements of 5.1 and 7. below are satisfied.

The return loss of the impedance at JS, towards A, against $600\underline{/0^\circ}$ ohms, when the sending part is correctly set up and calibrated, shall be not less than 20 dB over a frequency range 200-4000 Hz, and not less than 15 dB over a frequency range 125-6300 Hz.

4.2 *Receiving part*

The receiving part of the IRS is defined as the portion JR-B extending from the interface with the junction at JR to the handset earphone at B. The receiving part shall include such amplification and equalization as necessary to ensure that the requirements of 5.2 and 7. below are satisfied.

The return loss of the impedance at JR, towards B, against $600\underline{/0^\circ}$ ohms, when the receiving part is correctly set up and calibrated, shall be not less than 20 dB over a frequency range 200-4000 Hz, and not less than 15 dB over a frequency range 125-6300 Hz.

4.3 *Junction*

For loudness balance and sidetone tests, the junction of the IRS shall comprise means of introducing known values of attenuation between the sending and receiving parts, and shall consist of a calibrated 600 ohm attenuator having a maximum value of not less than 100 dB

(e.g. 10×10 dB + 10×1 dB + 10×0.1 dB)

and having a tolerance, when permanently fitted and wired in position in the equipment, of not more than $\pm$ 1% of the dial reading or 0.1 dB, whichever is numerically greater. Provision shall be made for the inclusion of additional circuit elements (e.g. attenuation/frequency distortion) in the junction. The circuit configuration of such additional elements shall be compatible both with that of the attenuator and the junction interfaces. The return loss of the junction against $600\underline{/0^\circ}$ ohms, both with and wuthout any additional circuit elements, shall be not less than 20 dB over a frequency range 200-4000 Hz, and not less than 15 dB over a frequency range 125-6300 Hz. For these tests, the port other than that being measured shall be closed with $600\underline{/0^\circ}$ ohms.

5. *Nominal sensitivities of sending and receiving parts*

The absolute values given below are provisional and may require changes to some extent as a result of the study of new Question 19/XII.

5.1 *Sending part*

The sending sensitivity, S_{MJ} is given in Table 1/P.48, column (2) (see 9.2 of Supplement No. 1).

5.2 *Receiving part*

The receiving sensitivity, S_{Je}, on a CCITT/IEC measured artificial ear (see Recommendation P.64) is given in Table 1/P.48, column (3) (see 9.2 of Supplement No. 1).

TABLE 1/P.48 – **Nominal sending sensitivities and receiving sensitivities of the IRS**

(These values were adopted provisionally in Munich, October 1974)

Frequency (Hz)	S_{MJ}	S_{Je}
	dB V/Pa	dB Pa/V
(1)	(2)	(3)
100	−45.8	−27.5
125	−36.1	−18.8
160	−25.6	−10.8
200	−19.2	−2.7
250	−14.3	2.7
300	−11.3	6.4
315	−10.8	7.2
400	−8.4	9.9
500	−6.9	11.3
600	−6.3	11.8
630	−6.1	11.9
800	−4.9	12.3
1000	−3.7	12.6
1250	−2.3	12.5
1600	−0.6	13.0
2000	0.3	13.1
2500	1.8	13.1
3000	1.5	12.5
3150	1.8	12.6
3500	−7.3	3.9
4000	−37.2	−31.6
5000	−52.2	−54.9

6. *Stability*

The stability should be maintained, under reasonable ranges of ambient temperature and humidity, at least during the period between routine re-calibrations. (See also Supplement No. 1.)

7. *Tolerances on sensitivities of sending and receiving parts*

This section specifies tolerances on:

a) the shape of the sensitivity/frequency characteristics of the send and receive parts of the IRS, and

b) the loudness weighted mean sensitivities.

7.1 *Shapes of sensitivity/frequency characteristics*

The shape of the sensitivity/frequency characteristics of the sending and receiving parts of the IRS shall be such that the limits specified in Table 2/P.48 are satisfied. In checking the shape, the mean values of sensitivity may be adjusted to best advantage.

TABLE 2/P.48 – **Tolerances on shapes of sending and receiving sensitivities**

Frequency (Hz)	Relative sensitivity (dB)	
	Sending part	Receiving part
180- 225	± 2.0	−13.0, +2.0
225- 280	± 2.0	−7.5, +2.0
280-2800	± 2.0	± 2.0
2800-3550	± 2.5	± 3.0
3550-4500	± 6.7	± 8.2

7.2 *Tolerances on mean values of sensitivity*

The gain setting in the send and receive parts of the IRS shall be such that the loudness weighted mean sensitivities shall be within ± 0.2 dB of the loudness weighted mean of the sensitivities given in Table 1/P.48. The loudness weighted means should be determined in accordance with the principles laid down in draft Recommendation P.XXE (found in Annex 2 to Question 15/XII in Contribution COM XII-No. 1 for the Study Period 1977-1980).

8. *Noise limits*

It is important that the noise level in the system be well controlled. See 5. of Supplement No. 1.

9. *Non-linear distortion*

In order to ensure that non-linear distortion will be negligible with the vocal levels normally used for loudness rating, requirements in respect of distortion shall be met.

10. *Complete specifications*

Certain secondary characteristics of an IRS may be included in Administrations' specifications. Particularly, special care must be given to adjustable components, stability and tolerances, crosstalk, installation and maintenance operations, etc. Supplement No. 1 gives some guidance on these points.

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SECTION 4

OBJECTIVE MEASURING APPARATUS

Recommendation P.51

ARTIFICIAL VOICES, ARTIFICIAL MOUTHS, ARTIFICIAL EARS

(amended at Mar del Plata, 1968, and at Geneva, 1972 and 1976)

A. GENERAL

The CCITT,

considering

a) 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,

b) 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 CCITT,

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

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

provisionally recommends

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

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

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

Note 3. – The Plenary Assembly at Copenhagen in 1936 considered that it would be of interest to deal separately with the design, on the one hand, of an artificial speech source and, on the other hand, 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 later.

B. ARTIFICIAL EAR PROVISIONALLY RECOMMENDED BY THE CCITT

1. *Introduction*

For many years the CCIF 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 CCIF 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 "CCITT reference coupler" [1)].

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

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

1 – simple conventional type,

2 – simple type used for telephonometric applications,

3 – wideband type for audiometric measurements,

4 – special type for calibrating insert earphones,

5 – a type which faithfully reproduces the characteristics of the average human ear, for use in laboratory.

Artificial ear type 1 (or reference coupler) is the subject of IEC Publication 303; this coupler is different from the "CCITT reference coupler".

The IEC Working Group then concentrated on a study of specifications relating to an ear of type 3. Agreement was reached on the acoustic impedance of the average human ear, after which the Working Group defined an electrical network equivalent to the average human ear and prepared specifications for constructing the type 3 artificial ear. The IVth Plenary Assembly of the CCITT (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 IEC Publication 318, with some minor amendments, are reproduced below.

The study of type 2 artificial ear and the study of acoustic leaks have therefore been deleted from the programme of work of the IEC and are carried on by the CCITT.

2. *Scope, purpose and definition*

2.1 *Scope and purpose*

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

2.2 *Definition*

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

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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.

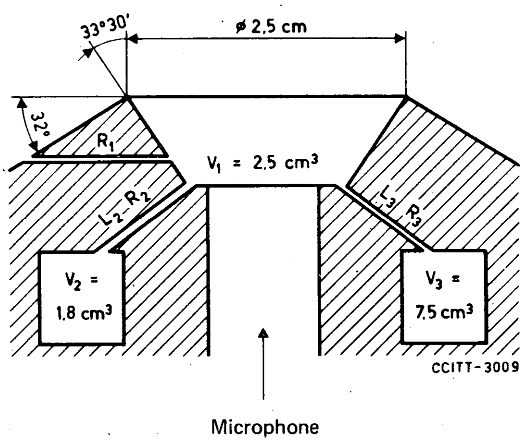
3. *Description of the artificial ear for audiometric measurements*

3.1 *Basic design*

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

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

These values relate to normal atmospheric conditions.



Note 1. – For tolerances see 3.2 of the text.

Note 2. – The volume V_1 includes the equivalent volume of the microphone capsule; a corresponding correction for the presence of a protective grid also being taken into account.

FIGURE 1/P.51

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 CCITT 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 of Question 12/XII in Volume V of the *Green Book.*

3.3 *Pressure equalizing leak*

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

3.4 *Microphone*

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

3.5 *Material*

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

3.6 *Example of design*

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

4. *Method of use*

See also Annex 1 to Question 12/XII in Volume V of the *Green Book.*

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

5. *Calibration*

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

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

6. *Use of ARAEN earphones with the IEC/CCITT artificial ear*

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

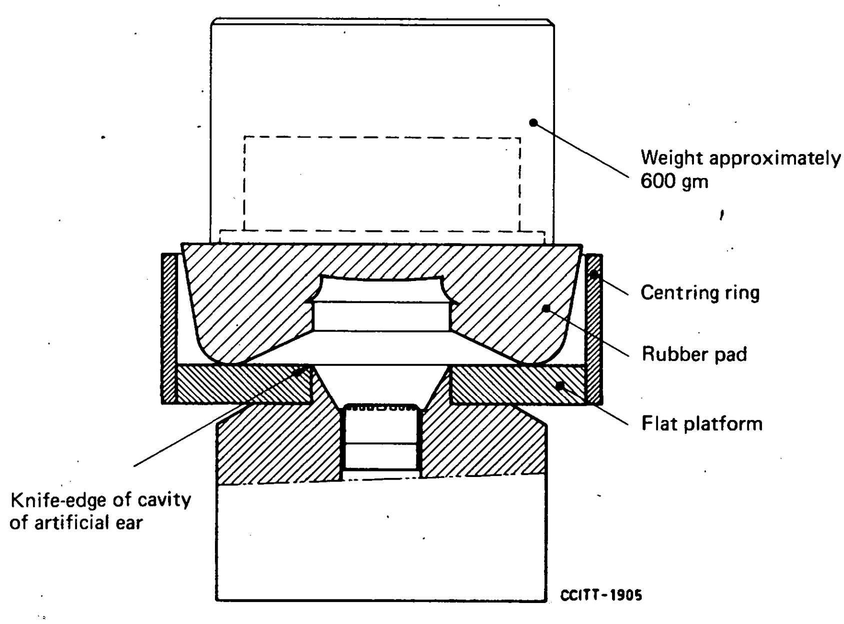


FIGURE 2/P.51 – **ARAEN receiver applied to a platform mounted flush with the top of the input cavity of the IEC/CCITT artificial ear**

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

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

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

Note 2. – This recommendation applies not only to the ARAEN receiving end but also to that of NOSFER, for tests of the type described above. It implies no change in the absolute calibration of ARAEN described in Recommendation P.41 and Supplement No. 9 to Volume V of the *White Book*.

C. SOUND SOURCE PROVISIONALLY RECOMMENDED BY THE CCITT

1. *Introduction*

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

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

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

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

2. *Acoustic characteristics of the sound source*

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

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

Table 1/P.51 shows the sound pressures measured by some Administrations at points along the axis and expressed in relation to the sound pressure at 25 mm from the lip plane. The sound pressures should be measured with a very small (say about 6 mm diameter) microphone or a probe microphone.

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

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

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

2.6 The source should be stable and reproducible.

TABLE 1/P.51

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

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

3. *Choice of a model*

The result of measurements made with the modified B & K 4216 source and the United Kingdom Post Office artificial mouth have shown good agreement between the two models. These results are not very different from the values measured for the human mouth as far as the distribution of acoustic pressure in a free field along the axis is concerned. These two models also meet the other specifications of 2. above.

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

Note 2. – The B & K 4219 model is at present manufactured, meeting the specifications in this Recommendation in the frequency range 200 Hz to 4000 Hz.

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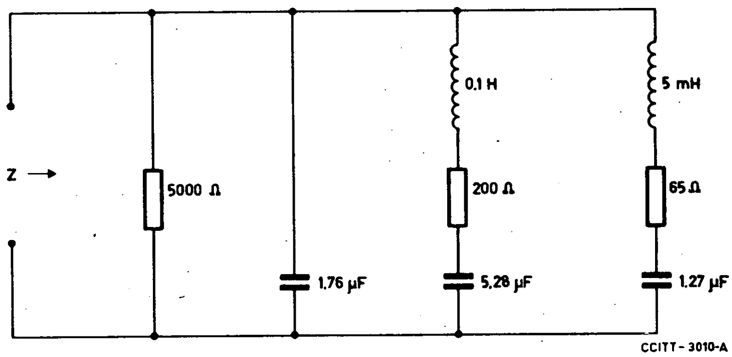
ANNEX 1

(to Recommendation P.51)

Lumped-parameter electrical network analogue of the average human ear

Three independent determinations of the acoustical impedance of the average human ear under no-leak conditions were available (see Bibliography above) covering various ear-cap contours used on audiometric earphones. In each case an analogue network of the type shown in Figure 1 was devised with values of the elements adjusted to produce an optimum fit to the experimental impedance data. The values of the lumped parameters shown in Figure 1 are average values corresponding to a plane ear-cap.



Note. – One electrical ohm corresponds to 10^5 Ns m^{-5}.

FIGURE 1 – **Lumped-parameter electrical network analogue of the average human ear. The real and imaginary components of the impedance Z are shown as functions of frequency in Figures 2 and 3**

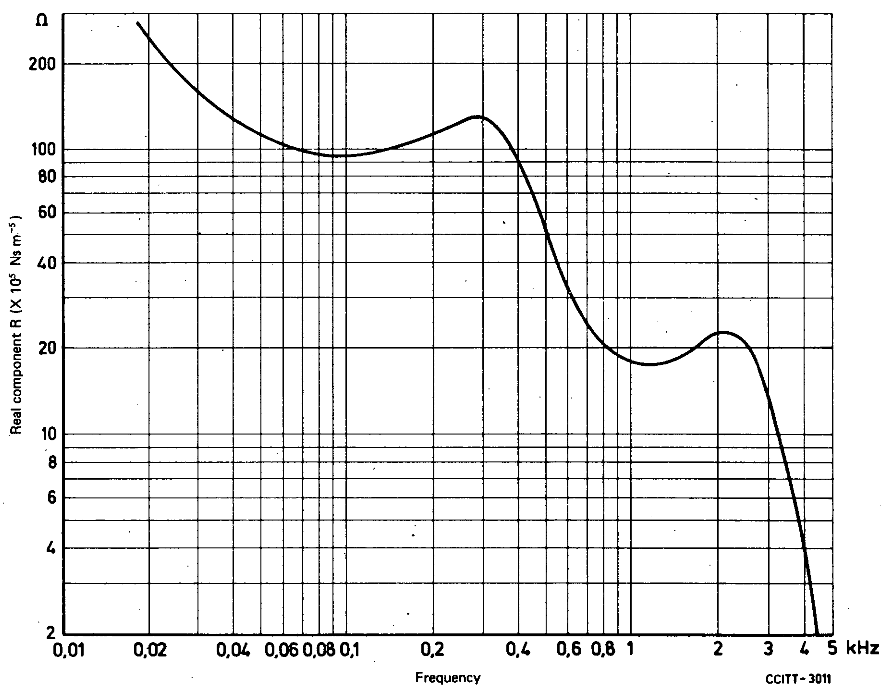


FIGURE 2 – **Real component of the impedance of the electrical analogue network**

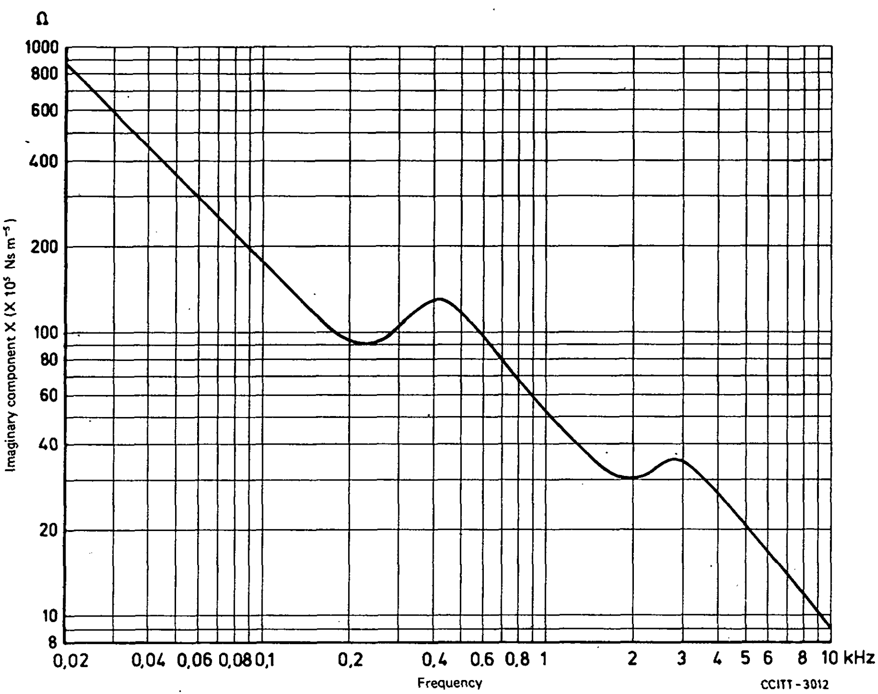
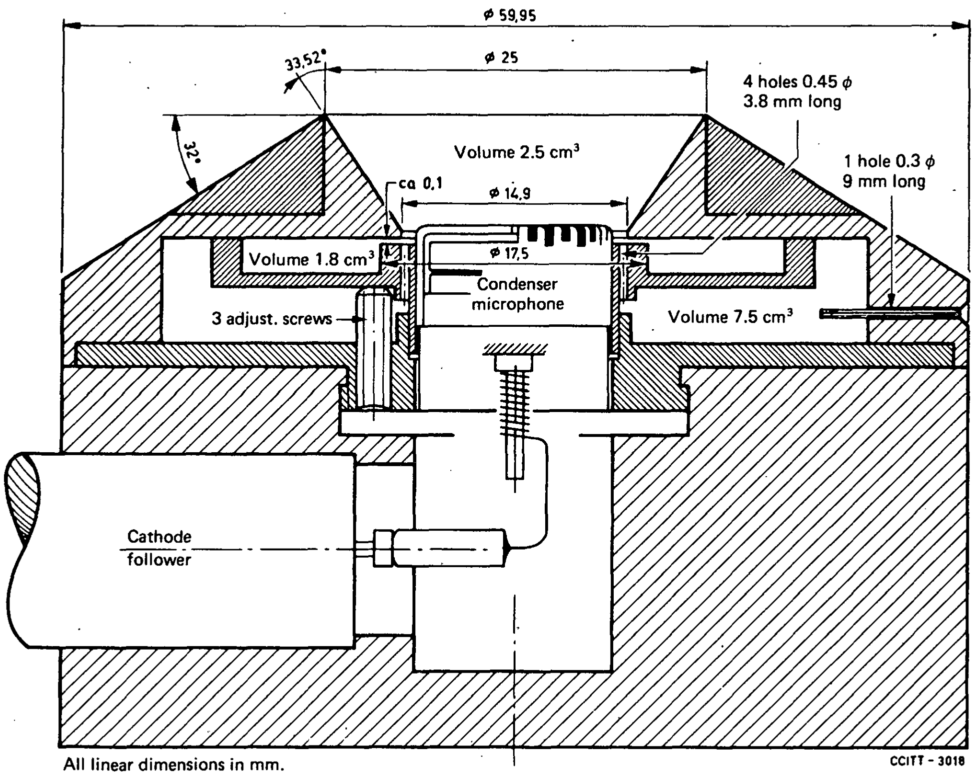


FIGURE 3 – **Imaginary component of the impedance of the electrical analogue network**

ANNEX 2

(to Recommendation P.51)

Example of one specific design of the artificial ear



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

Recommendation P.52

VOLUME METERS

The CCITT considers that, in order to ensure continuity with previous practice, it is not desirable to modify the specification of the volume meter of the ARAEN employed at the CCITT Laboratory.

Table 1/P.52 gives the principal characteristics of various measuring devices used for monitoring the volume or peak values during telephone conversations or sound-programme transmissions.

Note. – Descriptions of the following devices are contained in the Supplements to *White Book*, Volume V:

- ARAEN 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 U71 used in the Federal Republic of Germany: Supplement No. 13.

The Volume Indicator – SFERT Volume Indicator which used to be used in the CCITT Laboratory is described in Annex 18, 2nd Part of Volume V, *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 CCIF (Budapest, 1934) gives some information on the results of preliminary tests conducted at the SFERT Laboratory to compare the volume indicator with different impulse indicators.

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

TABLE 1/P.52 – **Principal characteristics of the various instruments used for monitoring the volume or peaks during telephone conversations or sound-programme transmissions**

Type of instrument	Rectifier characteristic (see Note 3)	Time to reach 90% of final reading (milliseconds)	Integration time (milliseconds) (see Note 4)	Time to return to zero (value and definition)
(1) "Speech voltmeter" United Kingdom Post Office Type 3 (S.V.3) identical to the speech power meter of the ARAEN	2	230	100 (approx.)	equal to the integration time
(2) Vu meter (United States of America) (see Note 1)	1.0 to 1.4	300	165 (approx.)	equal to the integration time
(3) Speech power meter of the "SFERT volume indicator"	2	around 400 to 650	200	equal to the integration time
(4) Peak indicator for sound-programme transmissions used by the British Broadcasting Corporation (BBC Peak Programme Meter) (see Note 2)	1		10 (see Note 5)	3 seconds for the pointer to fall to 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: type A sound meter type B sound meter		for both types: less than 300 ms for meters with pointer indication and less than 150 ms for meters with light indication	 10 ± 5 60 ± 10	for both types: 1.5 to 2 seconds from the 0 dB point which is at 30% of the length of the operational section of the scale

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 NRU-ON301) similar to the one defined in line (4) of the table has been standardized.

Note 3. – 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 4. – The "integration time" was defined by the CCIF 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 5. – 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.

Note 6. – In Italy a sound-programme meter with the following characteristics is in use:
Rectifier characteristic: 1 (see Note 3).
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.

Recommendation P.53

PSOPHOMETERS (APPARATUS FOR THE OBJECTIVE MEASUREMENT OF CIRCUIT NOISE)

(amended at Geneva, 1976)

The CCITT,

considering

a) 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 as far as the smoothness of the sensitivity/frequency characteristic is concerned;

b) that the "Joint Sub-committee 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;

c) that numerous tests and measurements made in the course of the past 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 necessary to repeat in Europe similar tests to those described by the Joint Sub-committee.

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/P.53 (see also the curve given in Figure 1/P.53); only the values in bold type in the table should be considered as specifying the psophometer filter network and should be taken into consideration for check tests of the apparatus; the other values, obtained by interpolation, are given to facilitate any calculations.

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

Permissible tolerances

The following permissible tolerances are:

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

Note. – During the XVIth Plenary Assembly (Florence, 1951), the CCIF 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.

TABLE 1/P.53 – **Table of commercial telephone circuit psophometer weighting coefficients**

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

TABLE 1/P.53 *(concluded)* – **Table of commercial telephone circuit psophometer weighting coefficients**

Frequency Hz	Weight		
	Numerical value	Numerical value squared	Value in decibels
2 850	553	305 809	–5.15
2 900	543	294 849	–5.30
2 950	534	285 156	–5.45
3 000	525	275 625	–5.60
3 100	501	251 001	–6.00
3 200	473	223 729	–6.50
3 300	444	197 136	–7.05
3 400	412	169 744	–7.70
3 500	376	141 376	–8.5
3 600	335	112 225	–9.5
3 700	292	85 264	–10.7
3 800	251	63 001	–12.0
3 900	214	45 796	–13.4
4 000	178	31 684	–15.0
4 100	144.5	20 880.25	–16.8
4 200	116.0	13 456	–18.7
4 300	92.3	8 519.29	–20.7
4 400	72.4	5 241.76	–22.8
4 500	56.2	3 158.44	–25.0
4 600	43.7	1 909.69	–27.2
4 700	33.9	1 149.21	–29.4
4 800	26.3	691.69	–31.6
4 900	20.4	416.16	–33.8
5 000	15.9	252.81	–36.0
> 5 000	< 15.9	< 252.81	< –36.0

Note. – If, for the planning of certain telephone transmission systems, calculations are made on a basis of the psophometric weighting values and if it appears useful to adopt, for frequencies above 5000 Hz, more precise values than those given in the above table, the following values may be used:

5 000 to 6 000	< 15.9	< 252.81	< –36.0
> 6 000	< 7.1	< 50.41	< –43.0

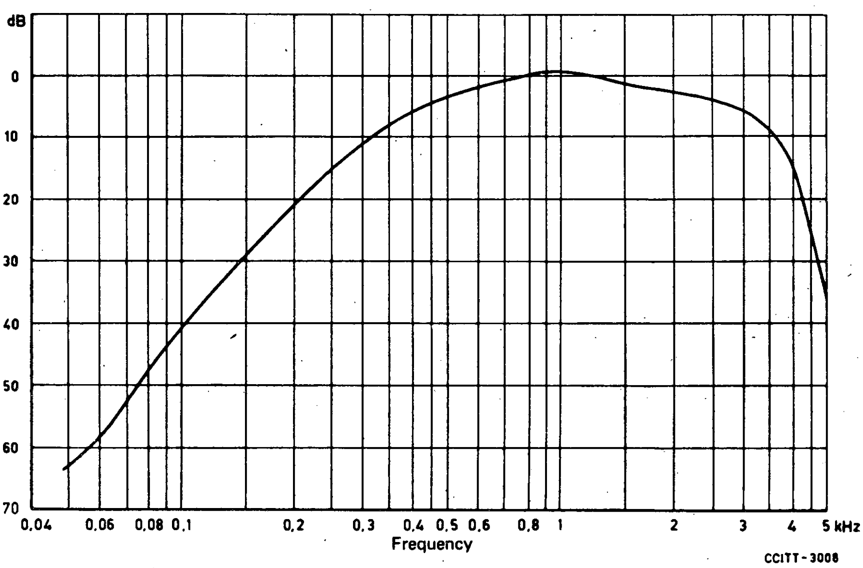


FIGURE 1/P.53 – **Characteristic curve of the psophometer filter network used for measurements at the terminals of a commercial trunk telephone circuit**

Measurements at the terminals of a subscriber's telephone receiver

The psophometer, which was standardized by the XVIth Plenary Assembly of the CCIF 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 the filter network specified by the XVIth Plenary Assembly of the CCIF 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 CCIF can very well be used, without modification, as a voltmeter of which the characteristics have been arbitrarily fixed, to make measurements at the terminals of the subscriber's telephone receiver.

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

Correspondence with the readings of American psophometer

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 CCITT 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
CCITT 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:

CCITT 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 CCITT psophometer, 1000 Hz in the American noise meters).

Detailed information concerning the noise meters used in the United States is given 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.)

Recommendation P.54

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

(amended at Mar del Plata, 1968, and Geneva, 1972)

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

Recommendation P.55

APPARATUS FOR THE MEASUREMENT OF IMPULSIVE NOISE [2)]

(Mar del Plata, 1968)

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 CCITT recommends that Administrations use the impulsive noise counter defined in Recommendation H.13 in Volume III for measuring the occurence of series of pulses on circuits for both speech and data transmission.

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

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2) Former Recommendation P.55 (*Red Book*, Volume V, p. 134) has been deleted.

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SECTION 5

OBJECTIVE ELECTRO-ACOUSTICAL MEASUREMENTS

Recommendation P.61

METHODS OF ABSOLUTE CALIBRATION OF MEASURING MICROPHONES

(amended at Geneva, 1976)

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

a) *Rayleigh disc method*

Application of this method at the CCITT Laboratory for the absolute calibration of the ARAEN is described in Supplement No. 9 to the *White Book*, Volume V.

b) *The reciprocity method for the calibration of condenser microphones*

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

Publication 327 (1971): Precision method for pressure calibration of one-inch standard condenser microphones by the reciprocity technique

Publication 402 (1972): Simplified method for pressure calibration of one-inch condenser microphones by the reciprocity technique

Publication 486 (1974): Precision method for free-field calibration of one-inch standard condenser microphones by the reciprocity technique

Recommendation P.62

MEASUREMENTS ON SUBSCRIBERS' TELEPHONE EQUIPMENT

A. MEASUREMENT OF THE ATTENUATION DISTORTION OF A TELEPHONE SET

The curve of the variation of the absolute sensitivity of an item of telephone equipment (sending or receiving system) as a function of frequency does not supply complete information on the manner in which this equipment reproduces the human voice or music, although such a curve may often be called the frequency characteristic.

However, the curve of variation of the absolute sensitivity of telephone equipment as a function of frequency gives useful indications from the point of view of the transmission of speech. On the other hand, for the transmission of music, in the absence of a precise criterion of the quality of transmission (corresponding to articulation, or repetition rate, in commercial telephony) such curves should be sufficient to enable the quality of the terminal equipment used (microphone or loudspeakers) to be appreciated.

For tracing sensitivity/frequency characteristics several modern commercial instruments are available which fall into two categories:

1. Recording devices which trace the frequency characteristics of telephone equipment automatically.
2. Devices which employ a cathode-ray tube and which allow rapid determination of the frequency characteristics of equipment.

Information on the measurement methods used by various Administrations for the maintenance of telephone apparatus and factory acceptance testing is given in Recommendations P.81 and P.82 below.

B. MEASUREMENT OF THE NON-LINEAR DISORTION OF A TELEPHONE SET AND OF MICROPHONE NOISE

Whilst the non-linear distortion of telephone receivers is in general negligible, microphones (and particularly carbon microphones of the type generally used in commercial telephone equipment) show considerable non-linearity: the relationship between the variation of microphone resistance and the acoustic pressure on the diaphragm is not linear. This non-linearity becomes more important as the variation of resistance in relation to the total resistance of the microphone increases, i.e. when the microphone is more sensitive. Furthermore, there are two supplementary effects:

1. The microphone is insensitive to acoustic pressure lower than a certain value (threshold of excitation).
2. As a consequence of the mechanical inertia of the carbon granules (delay in establishing electrical contact between the granules), the various states of agitation of the carbon under the influence of acoustic waves are not the same for all frequencies (for example, slow beats between two sounds are in general enhanced in reproduction by a carbon microphone).

Microphone noise is directly related to non-linearity. When non-linear distortion is measured, harmonic distortion as well as the variation of sensitivity with amplitude can be measured. As an example of such measurements, reference can be made to a contribution of the Federal Republic of Germany described in Annex 26, Part II of Volume V of the *Red Book.*

C. OBJECTIVE MEASUREMENT OF THE REFERENCE EQUIVALENT (SENDING AND RECEIVING) AND OF THE SIDETONE REFERENCE EQUIVALENT

1. As far as the objective measurement of reference equivalent (sending and receiving) of subscribers' telephone equipment is concerned, attention may be drawn to the equipment, described in Annexes 27 to 29, Part II of Volume V of the *Red Book* and in Annex G (Part II of Volume V *bis* of the *Red Book*), used by the Administrations of France, the Federal Republic of Germany, Switzerland and Sweden.

2. As far as the objective measurement of the sidetone reference equivalent of subscribers' telephone equipment is concerned, no objective method is recommended, this whole question being studied by the CCITT (see Recommendation P.63).

Recommendation P.63

METHOD FOR EVALUATING TRANSMISSION QUALITY ON THE BASIS OF OBJECTIVE MEASUREMENTS

These measuring methods are being studied by the CCITT.

Methods which have been used by the Swiss Administration and the U.S.S.R. Administration are described in Annexes 30 and 31, Part II of Volume V of the *Red Book.*

A new method for evaluating relative transmission performance ratings of complete connections by means of objective measurements has been studied by the American Telephone and Telegraph Company. The basic measuring equipment is described in Annex 1 to Question 15/XII studied in 1968-1972 (*White Book*, Volume V) [1].

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[1] See also SULLIVAN (J. L.) – A laboratory system for measuring loudness loss of telephone connections; *B.S.T.J. 50*, No. 8, October 1971, pp. 2663-2739.

Recommendation P.64

DETERMINATION OF SENSITIVITY/FREQUENCY CHARACTERISTICS OF LOCAL TELEPHONE SYSTEMS TO PERMIT CALCULATION OF THEIR LOUDNESS RATINGS

(Geneva, 1976)

See Recommendation P.76 for general principles concerning the determination of loudness ratings.

1. *Introduction*

The sending or receiving sensitivity/frequency characteristic of a complete local telephone system can be measured directly, or the information can be obtained by measuring three parts separately, namely *a)* the transducers, *b)* the electrical part of the subscriber's set and *c)* the subscriber's line and feeding bridge. Provided that each measurement is made in an appropriate manner and due regard paid to impedance mismatches at the interfaces, the respective quantities can be combined to yield the required sensitivities for the local telephone system. Measurement of complete local telephone systems is considered here but the same principles apply to measurements of microphones or earphones separately.

Since electro-acoustical measurements of the type being considered may be required for different purposes, it is important to distinguish the following:

a) supplying the designer of a transducer with information concerning the success he has achieved in aiming at a given sensitivity/frequency response;

b) checking that the manufactured product meets the specified requirements;

c) supplying sensitivity/frequency characteristics suitable for use in estimating loudness ratings, reference equivalents or other subjectivity-determined quantities.

The present Recommendation is concerned only with *c)* and, for this purpose, measurements under real conditions must form the basis. Artificial mouths and artificial ears must be used with due regard to obtaining good agreement between these measurements and those from real mouth and ear determinations. Measurements under real conditions are complicated, time-consuming and not reproducible with great precision, especially when carbon microphones are involved.

The present Recommendation describes measurement methods using recommended forms of artificial mouths and artificial ears (see Recommendation P.51).

2. *Sending sensitivities and calibration of microphones*

For the present purposes, the sending sensitivity of a local telephone system and the sensitivity of a microphone are specified in terms of the free-field sound pressure at a reference point in front of the mouth [2)], and the electrical output from the local telephone system or the microphone as the case may be. The input sound pressure cannot therefore be measured simultaneously with the electrical output and therefore the measurement must be made in an indirect manner. The sound pressure at the reference point is measured in the absence of the handset and, with the artificial mouth source unchanged, the handset is placed in the defined position in front of the mouth and the output measured. When a human mouth and voice are used, the source cannot be relied upon to maintain its output constant between the measurement of free-field sound pressure and that of the electrical output from the microphone. Artificial mouths suffer from imperfect representation of the source impedance and field distribution that applies to real mouths.

2) The mouth reference point used in the present Recommendation is defined in the Annex.

In addition to providing the proper source conditions, it is necessary to ensure that the mouthpiece is located for every design of telephone handset at the position that would be used in the real situation. This can be achieved by locating the mouthpiece properly with respect to an ear reference point; this ensures that longer handsets are measured with a greater mouth-to-microphone distance than is the case for shorter handsets. The success of using a given handset measuring position for measurement of sensitivity/frequency characteristics can be judged only by making comparisons, for handsets of different lengths, between real conversation test results using the artificial mouth and real mouths under suitably controlled measuring conditions. For the present Recommendation, the telephone handset shall be located as defined in Annex 1 of Recommendation P.76.

Special problems are encountered when making measurements with real mouths and real voices, even under controlled talking conditions. Under such circumstances the sound pressure cannot be measured directly at the required mouth reference point and therefore it has to be measured at some other point and referred indirectly to the mouth reference point. Some previous determinations have made use of a measuring microphone 1 metre from the mouth but this requires anechoic surroundings and is affected by obstruction from the handset under test.

When the sound pressure input to a carbon microphone is increased, the corresponding increase in output voltage does not bear a linear relationship to the increase in sound pressure. This non-linearity is a very complicated function of applied sound pressure, frequency, feeding current, conditioning and granule-chamber orientation. Reproducible results are obtained with an artificial mouth only if proper attention is paid to all these factors. One method used in the study of loudness ratings is described in 6. below; this is the "upper envelope" method whereby the sensitivity is measured at three different sound pressures and the highest value at each frequency taken as the equivalent sensitivity for speech under real talking conditions.

3. *Receiving sensitivities and calibration of earphones*

The CCITT/IEC artificial ear (see Recommendation P.51) provides means for precise measurements of the sensitivities of earphones. However, the sound pressures measured with it do not always agree well with those existing at the ear reference point in real ears under the test conditions used when subjective determinations of loudness ratings are being made. This can be attributed partly to the presence of appreciable acoustical leakage between the earphone and the real ear (such leakage is not represented in available recommended forms of the artificial ear) and partly to an increase in enclosed volume between the forms of earphone and the forms of real ear. Therefore, to use the results of measurements made according to the present Recommendation, it is necessary to make a correction (see 7. below).

Clearly, it would be very desirable if the artificial ear could be modified so as to avoid the need for the correction. Some further work has been done on this matter but it is not yet clear whether a single modification to the artificial ear would suffice for all types of telephone earphone. Further evidence is required, preferably from several laboratories so that a much wider variety of types of earphone can be examined.

4. *Artificial mouth*

The following properties are required:

a) the distribution in sound pressure around the orifice must be a good approximation to that around a human mouth;

b) the acoustical impedance looking into to the mouth must simulate that for human mouths, so that the pressure increase caused by the obstruction effect of telephone microphones will be representative;

c) it must be possible to establish definite sound pressures at the mouth reference point as a function of frequency. A convenient feature to embody in a practical artificial mouth is the linearity, over a suitable range of sound pressures, of the ratio of sound pressure at the mouth reference point to the voltage input to the artificial mouth. The ratio must be independent of frequency at least over the range 200 to 4000 Hz.

For the present purposes the mouth reference point is defined by the point on the axis of the artificial mouth located 25 mm in front of the equivalent lip position (see the Annex).

Recommendation P.51 defines the requirements for artificial mouths suitable for the present purposes.

5. *Artificial ear*

The following properties are required:

a) the acoustical impedance presented to telephone earphones must simulate that presented by real ears under practical conditions of use of telephone handsets;

b) the sensitivity of the artificial ear, namely the ratio of voltage output to sound pressure in the coupler of the artificial ear, shall be independent of frequency at least from 200 to 4000 Hz.

For a human ear, the ear reference point is defined as the centre O of the circular trace made by the circular concave ear-cap on the plane of the ear-cap (see Figure 1 of Annex 1 of Recommendation P.76), when it is placed comfortably against the ear. The corresponding point when the ear-cap is fitted to an artificial ear will usually differ from the place at which the sound pressure is measured and for this and other reasons certain corrections are necessary when the results are used for calculating loudness ratings (see 3. above).

6. *Definition of sending sensitivity of an LTS*

The sending sensitivity of a local telephone system (LTS) depends upon the location of the handset relative to the equivalent lip position of the artificial mouth. For the present purposes the speaking position defined in Annex 1 to Recommendation P.76 shall be used.

The sending sensitivity of a local telephone circuit is expressed in dB as folows:

$$S_{MJ} = 20 \log_{10} \frac{V_J}{p_M}$$

where V_J is the voltage across a 600 ohms termination and p_M is the sound pressure at the mouth reference point. Note that p_M must be measured in the absence of the "unknown" microphone of the test item. The units of S_{MJ} are dB relative to 1 V per Pa.

6.1 *Measurement of carbon telephone microphones ("upper envelope method")*

When measuring LTSs that contain no non-linear items (e.g. without carbon microphones), it does not matter at which sound pressure the measurements are made as long as it is known; however, when carbon microphones are present, different sensitivities will be obtained depending upon the sound pressure used. For calculation of sending loudness rating, these must be reduced to single values at each frequency and the method of reduction must take account of the characteristics of human speech. Various methods have been suggested, but the following method is being used in the CCITT Laboratory in studies concerned with Questions 8/XII, 12/XII and 15/XII:

a) determine the sensitivity as a function of frequency at the sound pressure level of −4.7 dB relative to 1 Pa. This is somewhat higher than the mean power of active speech of a talker, emitting speech at the vocal level used to determine reference equivalents in accordance with Recommendation P.72 and loudness ratings in accordance with the subjective test method described in draft Recommendation P.XXC (see Annex 1 to Question 15/XII, in Contribution COM XII-No. 1 for the study period 1977-1980;

b) repeat *a)* but with the sound pressure level increased by 10 dB;

c) repeat *a)* but with the sound pressure level decreased by 10 dB;

d) select from *a), b)* and *c)* the highest sensitivity at each frequency.

Carbon microphones must be given appropriate conditioning treatment at suitable intervals during the measurements (see Recommendation P.75).

6.2 *Measurement of carbon telephone microphones (other methods)*

The foregoing description relates to the "upper envelope" method. Other methods may be used and the following are some examples:

a) The method used in currently available types of objective instrumentation for measuring loudness ratings (OREM), see draft Recommendation P.XXF (Annex 1 to Question 15/XII, in Contribution COM XII-No. 1 for the study period 1977-1980). Such equipment uses a sweeping frequency covering the range from 200-4000-200 Hz at a periodicity of 1 sweep per second; the instantaneous level within any narrow frequency band varies as a function of frequency approximately in accordance with the spectrum of speech emitted from the human mouth.

b) Noise-burst method. This is similar to the upper envelope method in using a fairly slow rate of sweep, but here only one level of sound pressure is used. The sweep is interrupted at intervals to apply a short burst of noise at a fairly high level during which the level recorder may be disconnected. This method is described in the *Red Book*, Volume V, page 76.

c) Real-voice calibration. This may be performed by measuring speech spectra emitted alternately from the carbon microphone under test and a calibrated linear microphone. A very small linear microphone can be mounted on the telephone being tested. Naturally the most appropriate results will be obtained when the talkers are conducting telephone conversations but it is then difficult to have reliable knowledge of the sensitivity/frequency characteristic of the linear microphone. It is usually necessary to rely upon a suitable artificial mouth to provide the calibration of the linear microphone.

d) The application of a wideband signal generated by a pseudo-random binary sequence [3]. The output from the carbon microphone is then processed by a digital computer using the Fourier transform. This method, like the conversation method, requires calibration by a linear microphone of known sensitivity/frequency characteristic. This method has the advantage that the frequency characteristic may be obtained with a very short duration (e.g. 50 ms) sample of test signal.

7. *Definition of receiving sensitivity of an LTS*

The receiving sensitivity of a local telephone system, as measured directly with an artificial ear complying with Recommendation P.51, is expressed as follows:

$$S_{Je} = 20 \log_{10} \frac{p_E}{\frac{1}{2} E_J}$$

where p_E is the sound pressure in the artificial ear and $1/2\ E_J$ is half the emf in the 600 ohms source. The units of S_{Je} are dB relative to 1 Pa/V.

Note. – The receiving sensitivity suitable for use in calculation of loudness is given by $S_{JE} = S_{Je} - L_E$, where L_E is a correction explained in 3. above. Further information on this topic is given in draft Recommendation P.XXE (see Annex 1 to Question 15/XII, in Contribution COM XII-No. 1 for the study period 1977-1980).

8. *Methods for determining S_{MJ} and S_{Je}*

When the sending and receiving sensitivities of an actual local telephone system are required, the measurements according to the definitions given in 6. and 7. above can be made as illustrated in Figures 1/P.64, 2/P.64 and 3/P.64. These methods have been used by the CCITT Laboratory.

Figure 1/P.64 shows the method of setting up the artificial mouth so that the sound pressure p_M at the mouth reference point is known at each test frequency.

[3] U.K. Patent Application No. 32370/73.

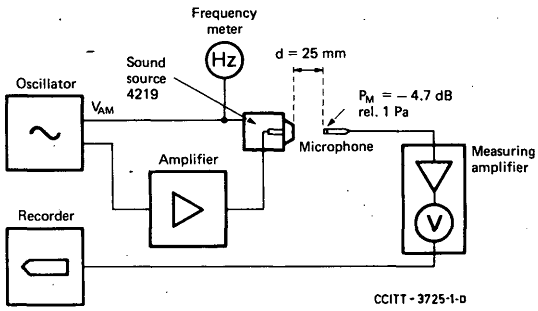


FIGURE 1/P.64 – Measurement of acoustic pressure p_M at the mouth reference point 25 mm from the artificial lip plane of the B and K 4219 sound source

Figure 2/P.64 shows the measurement of output V_J from the local telephone circuit when the microphone is placed at the appropriate position in front of the artificial mouth and the artificial mouth is energized in the same manner as when the sound pressure p_M was set up in the absence of the test microphone (Figure 1/P.64).

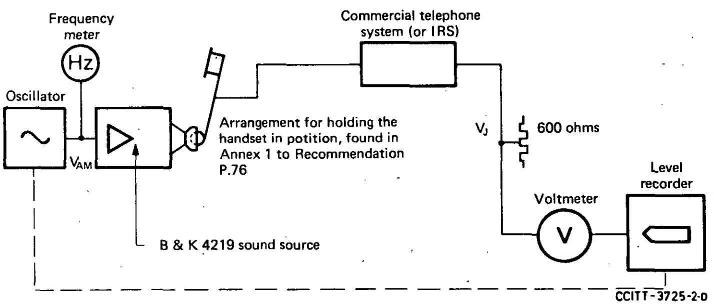
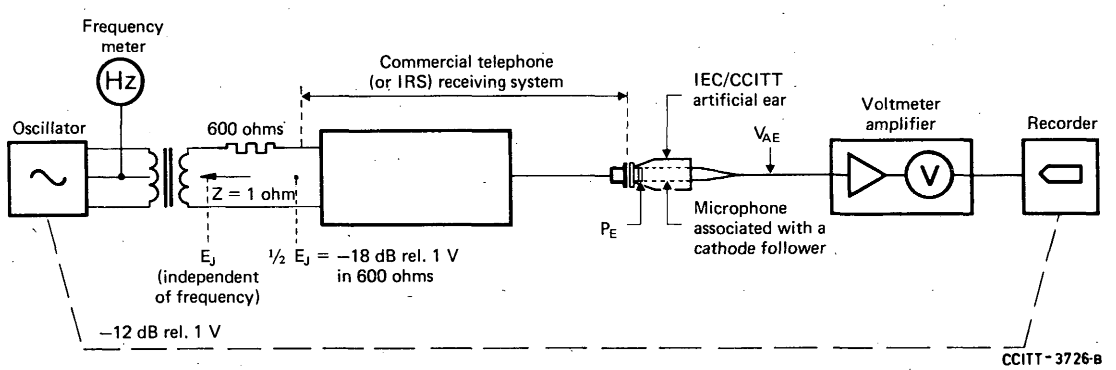


FIGURE 2/P.64 – Voltages V_J, measured across the terminals of a 600 ohms pure resistance connected to the output of a commercial sending system or of the sending part of the intermediate reference system

Figure 3/P.64 shows the measurement of the sound pressure p_E in the artificial ear when the local telephone system is connected to a 600-ohm source of internal emf E_J. Note that the definition of S_{Je} is in terms of 1/2 E_J and not the potential difference across the input terminals of the local telephone system; this potential difference will, of course, differ from 1/2 E_J, if the input impedance of the local telephone system is not 600 ohms.



The following items of apparatus have been used in the CCITT Laboratory for these measurements:

Oscillator	Bruel & Kjaer Type 1022
Level Recorder	Bruel & Kjaer Type 2305
Amplifier	Bruel & Kjaer Type 2606
Measuring Amplifier	Bruel & Kjaer Type 2603
Microphone	Bruel & Kjaer Types 4135 and 4134
Sound source	Bruel & Kjaer Type 4219
Voltmeter	Bruel & Kjaer Type 2107
Cathode Follower	Bruel & Kjaer Type 2615

FIGURE 3/P.64 – Calibration of a commercial telephone receiving system or the receiving part of the intermediate reference system

If the complete local telephone system is not actually available, it will be necessary to estimate the sensitivity, sending and receiving, by combination of the sensitivities and transmission losses of the component parts. For example, S_{MJ} can be made up of the following components:

S_M = sensitivity of a telephone microphone referred to an MRP;

L_S = electrical transmission loss from the terminals of a microphone to the line terminals of a telephone set;

L_{INS} (SL + FB) = transmission loss of the combination of subscriber's line and feeding bridge.

Similarly, S_{Je} is shown as composed of the following:

S_E = sensitivity of a telephone receiver referred to an ERP;

L_R = electrical transmission loss from the terminals of a receiver to the line terminals of a telephone set;

L_{INS} (SL + FB) = see above.

Suitably defined, the sensitivities and losses of the separate components can be combined algebraically to yield the sending and receiving sensitivities, S_{MJ} and S_{Je} that were defined in 6. and 7. above; proper allowances must be made for any impedance mismatches.

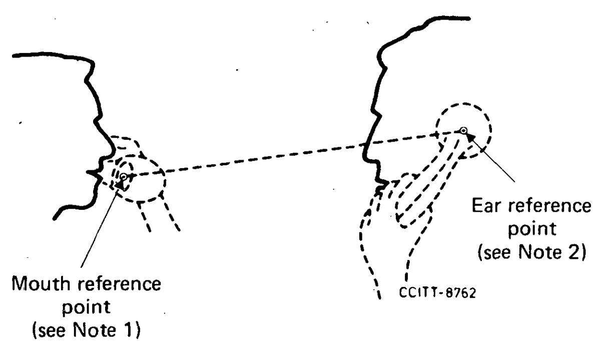
The decomposition described above is convenient for the treatment of most current types of telephone sets containing a transformer induction coil and transducers having relatively low electrical impedances, which are approximately matched to the circuit and are without any amplification (except that provided by the operation of the carbon microphone).

New types of telephone sets containing, for example, amplification in the electrical paths must be treated differently. Each case will have to be treated on its merits to ensure that the overall sensitivities, S_{MJ} and S_{Je} of the local telephone systems conform to the definitions given in 6. and 7. above.

ANNEX

(to Recommendation P.64)

Definitions of mouth reference point and ear reference point



Note 1. – The mouth reference point is located at a distance of 25 mm in front of the lips on the horizontal axis through the centre of the opening of the mouth. It is defined in the absence of any obstruction.

Note 2. – The ear reference point is located at the entrance to the ear canal of the listener's ear. It is defined as lying at the centre of the plane of a circular concave ear-cap.

FIGURE 1 – **Definitions of mouth and ear reference points**

SECTION 6

SUBJECTIVE VOICE-EAR MEASUREMENTS

Recommendation P.71

MEASUREMENT OF SPEECH VOLUME

(amended at Mar del Plata, 1968)

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 comparison by voice and ear with the new master system for the determination of reference equivalents (NOSFER); 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 NOSFER 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 (SETAB) or of the electrodynamic microphone and receiver type (SETED), 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:

[1)] This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the CCITT Laboratory by using reference systems are described in Section 3 of this volume.

a) *Use of a working standard system of the SETAB 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 (SETAB) can be made in one of the two following methods:

a.1 *Method termed "two-operator with hidden-loss method"*

The method is based on the simultaneous use of two adjustable attenuators; one of these (the balancing attenuator) serves the purpose of equalizing the sound intensities at the receiving end; the second attenuator (the hidden-loss attenuator) can be adjusted arbitrarily, before the test and unknown to the listening operator, in order to modify the apparent sensitivity of one of the instruments compared.

The results must be expressed as: x dB "better" (M) or "worse" (P) than the NOSFER taking account of the reference equivalent of the SETAB.

The particulars given below refer to setting-up details, and are given only as examples.

a.1.1 *Comparison of a sending system with a standard sending system*

The schematic diagram together with the necessary switching arrangements for this comparison are shown in Figure 1/P.72.

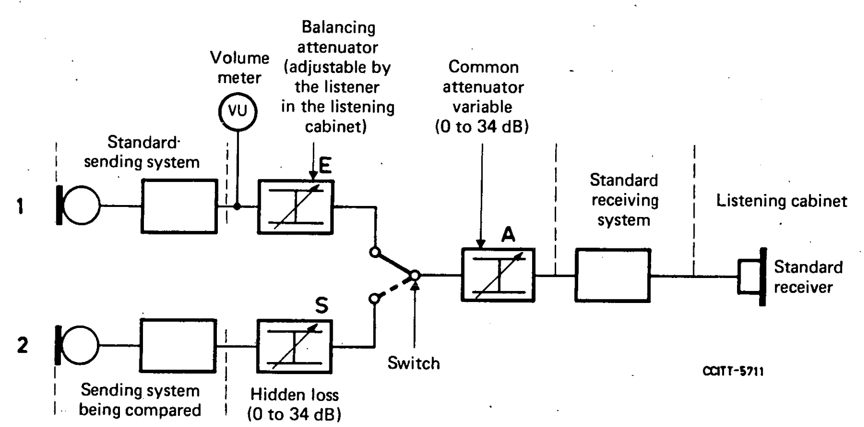


FIGURE 1/P.72 – **Comparison of a given sending system with a standard sending system (two-operator hidden-loss method)**

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, Sait-Leu, Léon, Loudon (used in France and in the CCITT Laboratory).

He maintains, when talking, the normal volume for telephonometric measurements defined in Recommendation P.42, C., and places his lips so that they are approximately tangential to the plane of the circle which bounds the guard-ring [2]. At the same time he operates the switch in the appropriate manner for controlling the switching system.

[2] The position of the guard-ring is defined under C. of the present Recommendation.

A second operator B receives, in a single receiver, the signals from the two microphones compared. He compares them by ear and adjusts the balancing attenuator so as to obtain the same sound intensity.

To enable the listening operator to follow the respective positions of the key, it is advisable to use a lamp, the lighting circuit of which is controlled synchronously by the key. When glowing, it indicates that the balancing attenuation is inserted in the listening circuit. When the balance is thus obtained the test is completed and it is sufficient to record the readings of the two attenuators, and to interpret them according to the example given below.

a.1.2 *Comparison of a receiving system with a standard receiving system*

The schematic diagram together with the switching arrangements necessary for this comparison are shown in Figure 2/P.72.

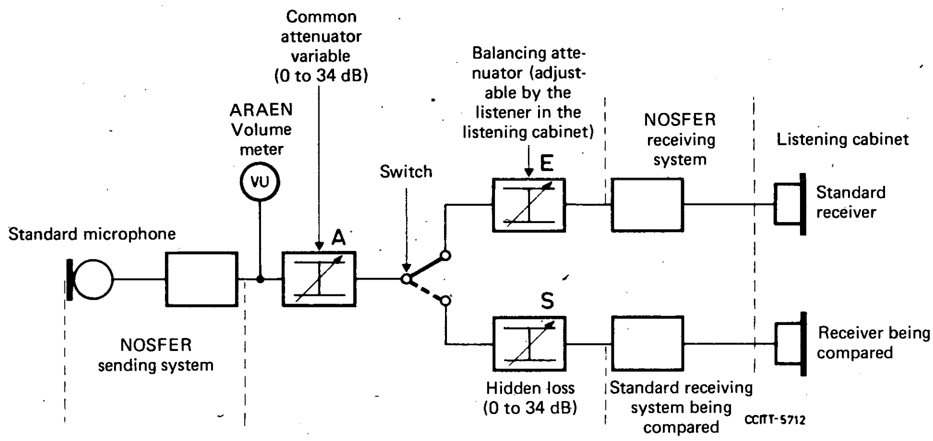


FIGURE 2/P.72 – **Comparison of a given receiving system with a standard receiving system (two-operator hidden-loss method)**

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. Table 1/P.72 gives an example of the recording of a telephonometric measurement conducted at the Laboratory with a crew of five.

TABLE 1/P.72 – **Example of the recording of a telephonometric measurement**

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 CCITT Laboratory), it is necessary to take account of the value of reference equivalent of this sending (or receiving) standard system. The reference equivalent of a sending (or receiving) system is then determined from the test results in the following manner, e.g.:

Uncorrected mean result	−5.0 (5 dB better)
Reference equivalent of the working standard system	+1.3 (1.3 dB worse)
Reference equivalent of the system under test	(−5.0) + (+1.3) = −3.7 dB or (3.7 dB better)

a.2 *Method termed "Three-operator without hidden-loss method"*

This method requires positions for three operators:

a) sending position;

b) receiving position (where the telephonometric comparisons are made);

c) balancing position.

The sending and receiving positions are identical with those already described, the only difference between the two methods being in the number and positions of the attenuators. The comparison method employing three operators requires, in effect, only one adjustable attenuator in addition to the fixed attenuator. This is adjusted by operator C, who occupies the balancing position and receives signals from operator B at the receiving end. The hidden-loss attenuator is replaced by direct metallic connections.

The method of operations is as follows:

a.2.1 *Comparison of a sending system with a standard sending system* (Figure 3/P.72)

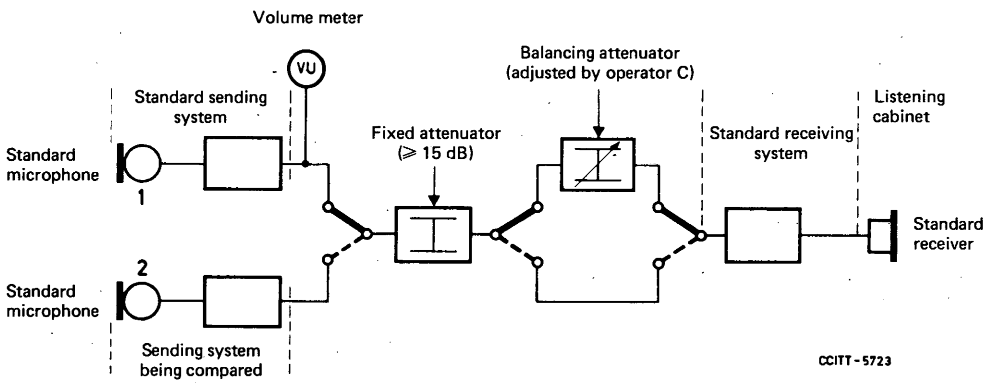


FIGURE 3/P.72 – **Comparison of a sending system with a standard sending system (three-operator without hidden-loss method)**

Operator C adjusts the balancing attenuator to a preliminary value a_1, he then signals by lamp, by buzzer, or orally to operator A that he may begin talking. The latter repeats into the two microphones alternately the conventional phrase adopted once and for all, maintaining the normal volume for telephonometric measurement defined above in Recommendation P.42, C. Operator B receives, in a standard receiver, the signals produced successively by the two microphones. A luminous indicator, controlled by the general switching system, indicates to him the microphone being spoken into at any instant (No. 1 or No. 2). If the sound intensity corresponding to microphone 2 is less than the sound intensity corresponding to microphone 1 (standard), B presses the signalling button marked P (worse). A luminous signal (lighting of a lamp on the cap of which is marked the letter P), together with, if necessary, a buzzer signal, indicates to operator C the first decision. A signal of the same type is also used to inform operator A that he may stop talking. Operator C records immediately the test result in a table in the form a_1 P.

The number a_1 can be entered in either of two columns. In the first, it indicates that the attenuation was introduced into the circuit at the same time as the standard, with the effect of attenuating the standard; inserted in the second column, it indicates that the attenuation was introduced into the circuit at the same time as the test apparatus, with the effect of increasing the attenuation of the latter.

In the opposite case, if the sound intensity corresponding to microphone 2 is greater than the sound intensity corresponding to microphone 1 (standard), operator B presses the signalling button marked M (better). A luminous signal (lighting of a lamp, on the cap of which is marked the letter M), accompanied by a buzzer signal if necessary, then appears in front of operator C. If the test result corresponds to an exact balance, operator B presses a third button controlling the circuit of a third lamp, which is used for signalling exact balance.

The balancing operator C then sets the balancing attenuator at a second value a_2. He then signals to operator A that he may resume talking. The result of this measurement will be a second decision, for instance M, signifying that the microphone compared appears to be better than the standard, when the latter is in series with an attenuation of a_2 dB; operator C records the corresponding information in the form a_2 M.

He then adjusts the attenuation, at his description, to new values in order to diminish the interval between the two values for which the balancing result changes its sign. When successive intervals (forming a convergent series) have determined, if not the number corresponding to an exact equality of the sound impressions, at least two values a and a' differing at the most by 1 or 2 decibels, and for which one of the two instruments appears better or worse than the other, the test is considered as finished. Operator C at the control position signals the end of the test to the other two operators A and B and a new balance can then begin.

A single determination of equality cannot be considered sufficient to denote balance, and must be confirmed by at least two decisions (M and P) enclosing it.

TABLE 2/P.72 – **Example of the recording of a (three-operator without hidden-loss method) measurement**

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

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.

Table 2/P.72 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 +.

a.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.

b) *Use of the SETED type working standard*

The SETED 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 CCITT recommended use of working standards either with a carbon microphone (SETAC) or with an electromagnetic microphone (SETEM). Administrations which still use these working standards will find information concerning them in Volume IV of the *Yellow Book* (Paris, 1949), pp. 254 to 266.

C. PRECAUTIONS TO BE TAKEN DURING TELEPHONOMETRIC MEASUREMENTS

Volume to be maintained. − The speech volume produced during telephonometric measurements is of great importance in the conduct of such measurements as it influences the absolute and relative sensitivities of the equipment (especially in the case of carbon microphones). This volume must correspond to the normal power for telephonometric measurements employed in the CCITT Laboratory and determined as shown above (see Recommendation P.42, C.).

It is necessary to adjust this volume by means of a volume indicator whose needle is in view of the talker and which is connected at the input of the fixed junction attenuator (which has an input impedance of 600 ohms). This volume indicator must have been compared with the SFERT 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 SFERT 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. First, from measurements made on the heads of a large number of individuals, the characteristic head dimensions of an average subscriber have been determined together with the position in which he holds the handset to his ear during a telephone conversation. Such measurements have been made in various countries by means of an instrument referred to as a "Device for measuring the dimensions of the head".

This device is shown in Figure 4/P.72. It consists of a telephone receiver to which is applied a complex voice frequency tone and to which is fixed a system of graduated scales. The device is held in the plane passing through the centres of the ears and of the mouth, the individual placing the receiver to his ear as he would normally do. The distance d_1 between the centre of the ear and the line of the lips and the distance d_2 of the displacement of the centre of the mouth are read on the scales. By means of the abac (Figure 5/P.72) the following data are deduced:

1. the distance δ between the centre of the ear and the centre of the mouth;
2. the angle α between the plane of the earpiece of the telephone receiver and the straight line from the centre of this earpiece to the centre of the mouth.

The distance l between the midpoints of two telephone receiver ear-caps placed one against each ear is also measured (distance between the centres of the ears). The angle β is computed; the intersection of the plane of the telephone ear-cap placed against the ear and the plane through the centres of the ears and the centre of mouth defines one straight line; β is the angle between this line and the "direction of speech". The "direction of speech" is the straight line formed by the intersection of the median plane of the head with a plane through the centres of the ears and the centre of the mouth.

The value of β is obtained from the formula:

$$\beta = \arcsin \frac{l}{2\delta} - \alpha$$

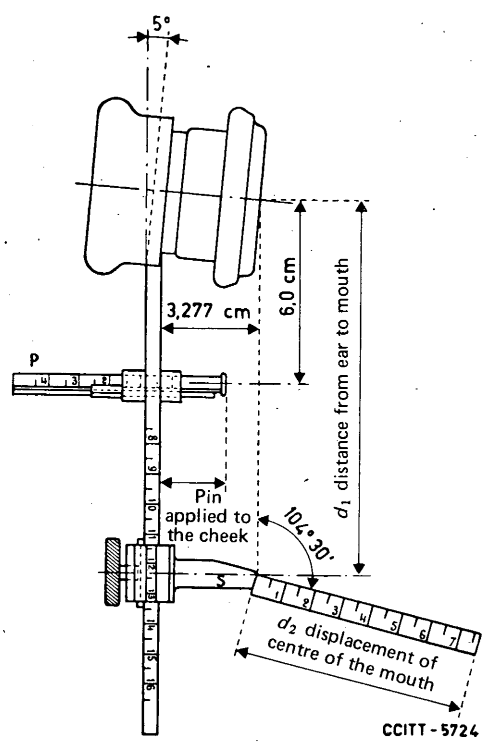
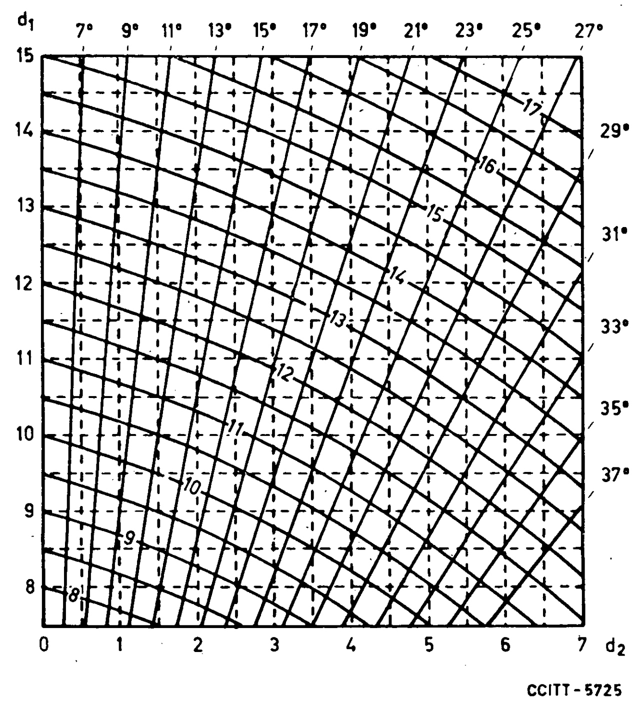


FIGURE 4/P.72 – **Device for measuring the dimensions of the head**



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

FIGURE 5/P.72 – **Abac used with the device for measuring the dimensions of the head**

The CCITT recommends the following values for α, β and δ in the case of reference equivalent measurements:

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

These figures are the most probable values observed in the United States. Although other measurements of the dimensions of heads of subscribers have given slightly different values, it is desirable to keep the above values for the sake of worldwide standardization and also because, on the basis of these values, much information concerning the reference equivalents of commerical 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 midpoint of the ear-cap 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 midpoint 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 defined above, 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.

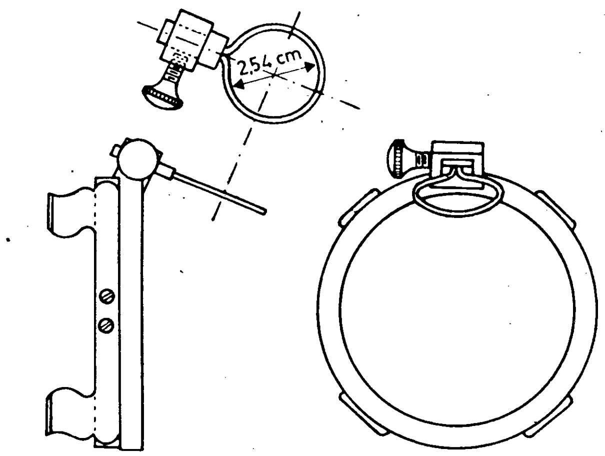
It should be noted that the position of the guard-ring, thus defined, has been fixed without reference to the inclination of the diaphragm of the microphone and does not necessarily correspond to the best operating conditions of the latter.

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.

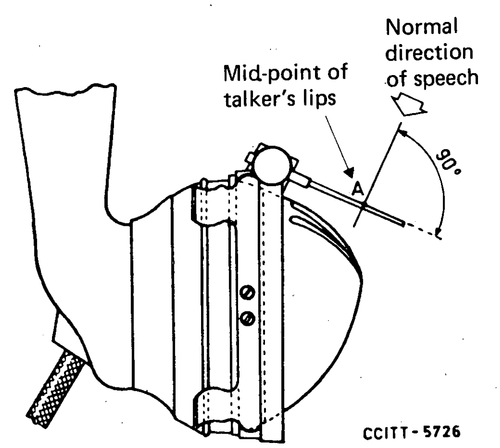
In order to avoid this trouble the receiver of the handset should be disconnected and is not to be applied to the operator's ear; in addition, in the test arrangement a similar receiver should be inserted in place of the disconnected receiver which should be placed face downwards on the table so as to 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/P.72 is recommended.

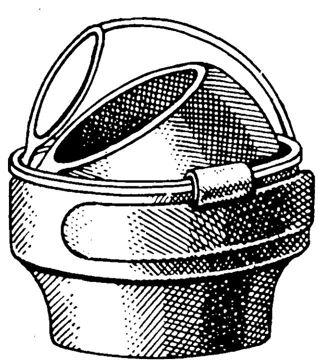


1. Example of guard-ring for tests of handsets

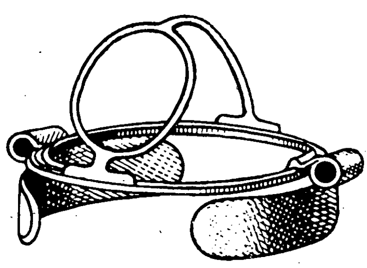


2. Attachment of guard-ring to a handset

FIGURE 6/P.72



Attachment of guard-ring to a handset



Perspective sketch of the guard-ring

FIGURE 7/P.72 – **Guard-ring used by the American Telephone and Telegraph Company for tests of handsets**

ANNEX

(to Recommendation P.72)

Remark on measurements of reference equivalent

It is necessary to draw a very clear distinction between, on the one hand, measurements required in the design and development of commercial telephone equipment to satisfy service conditions as well as possible and, on the other hand, the exchange between Administrations of numerical data which enable different types of equipment to be compared, from the standpoint of reference equivalent 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 CCITT 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 to 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 are 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 as 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 AEN determinations in the CCITT 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 [3)]

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.

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3) This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the CCITT Laboratory by using reference systems are described in Section 3 of this Volume.

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 NOSFER to be varied at will before the measurement and by an amount unknown to the listener. The value of sidetone reference equivalent of the telephone system is equal to the sum $S + Q$ of the values of the hidden-loss and balancing attenuators.

b) For measurement of the room-noise sidetone reference equivalent of a telephone set by aural comparison between the Master Reference System (or a calibrated working standard) and the sidetone path from microphone to receiver of the telephone set considered, one should, strictly, employ "normal room noise" produced by a loudspeaker situated at a specified distance from the microphone.

The noise source could consist, for example, of a gramophone pickup reproducing from a disc on which typical room noises had been recorded. The CCITT, having adopted a reference room noise for AEN determinations (see Recommendation P.45), advises the use of such a noise.

The measurement technique used in the CCITT Laboratory is given in Figure 1/P.73, where the real voice is replaced by a noise source giving the reference room noise at the positions of the two microphones (1 and 2).

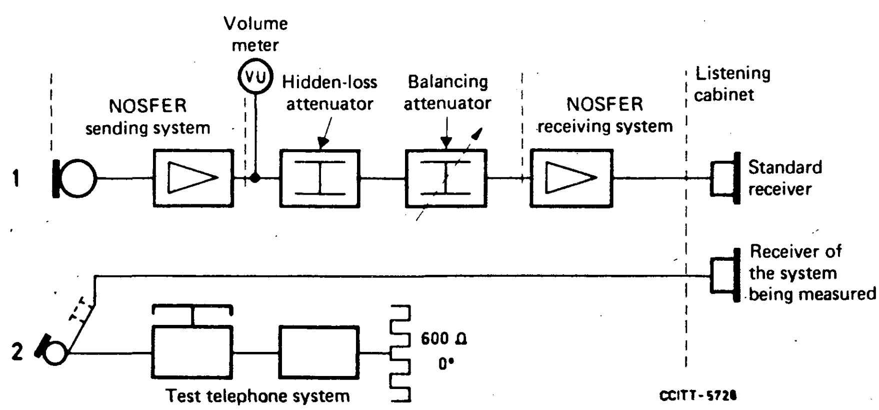

FIGURE 1/P.73 – **Measurement of the sidetone reference equivalent of a commercial telephone system**

The value of the reference equivalent of sidetone for room noise is equal to $S + Q - 17$ dB, where S is the hidden-loss and Q is the loss of the balancing attenuator. The correction of 17 dB takes account of the fact that, under these conditions of measurements, the NOSFER is more efficient than the SFERT and it is with respect to the latter that sidetone reference equivalents have been defined.

Note. – The CCITT 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 [4)]

A. REPETITION OBSERVATION TESTS

One of the criteria used for assessing the quality of transmission in service is based on the observation of repetions in the course of telephone conversations conducted under commercial service conditions.

No direct measurement of effective transmission losses exists having international acceptance.

As far as trunk telephone circuits are concerned, attention is confined to the individual measurement of various transmission impairments due respectively to circuit noises, distortions, etc., without even being certain that close agreement to the effective transmission can always be obtained by calculation, for example by adding the reference equivalent of the circuit (which is approximately equal to the loss at 800 or 1000 Hz) to the transmission impairments due to the circuit distortions (attenuation distortion, phase distortion, non-linear distortion) and to the transmission impairments due to various noises (induced noise, repeater noise, crosstalk noise, etc.). These transmission impairments have been defined in the Recommendations under 1.1 of Part I in Volume III and are 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 the 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 noise, expressed in decibels.

The test circuit used for measurements by this method should reproduce the average conditions of a typical commercial trunk telephone call. Each Administration should set it up using the sending and receiving systems of its own working standard (with typical room noise) and connecting these sending and receiving systems together by means of a circuit (or better still an artificial line) of adjustable attenuation and similar in all respects to the trunk circuit considered (particularly from the point of view of the various distortions), except that this circuit (or artificial line) used for the measurements is noise-free (no circuit noise).

The method of repetition observations indicated above can be adapted for measurement of the transmission impairment due to a certain distortion (for example: limitation of the band of frequencies effectively transmitted, or attenuation distortion) of the trunk circuit considered, on condition that, instead of the circuit noise, the quantity is taken which characterises 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.

[4)] This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the CCITT Laboratory by using reference systems are described in Section 3 of this Volume.

B. IMMEDIATE APPRECIATION TESTS

The method of immediate appreciation tests is described in Annex 32, Part II, of Volume V of the *Red Book*.

C. OTHER METHODS

For example, Annex 2 to Chapter I of the CCITT Handbook on *Transmission Planning of Switched Telephone Networks* describes a method used by the United Kingdom Post Office for subjectively determining telephone transmission quality (see also Recommendation P.77 and Supplement No. 2).

Recommendation P.75

STANDARD CONDITIONING METHOD FOR HANDSETS WITH CARBON MICROPHONES

(Geneva, 1972)

1. The CCITT considers that, since the characteristics of carbon microphones are strongly dependent on conditioning techniques, it is necessary to follow a consistent procedure prior to measuring sensitivity/frequency characteristics in order to obtain reproducible results. The following steps are specified for the *standard conditioning method*:

a) Place the handset in a holding fixture with the handset clamped in a position corresponding to that in which the microphone is going to be measured.

b) Connect the microphone or telephone set terminals as required to the d.c. feed circuit and appropriate terminating load.

c) Turn the feed current on. After 5 seconds, condition the microphone by rotating it through an arc slowly and smoothly. The microphone face should reach a vertical plane during the initial part of the rotation. In this vertical plane, a reference vector should be visualized which passes through the centre of the microphone and points straight up. Rotation should then be continued until this reference vector points straight down (that is, a movement of the reference vector through 180°). The direction of rotation should then be reversed and the microphone returned to its starting position.

Without interrupting the d.c. current or jarring the microphone, repeat this process two more times. The speed of rotation is not critical, but should be slow enough to ensure that the effect of centrifugal force on the carbon granules is negligible. Finally, return the handset to the measuring position.

Note. – Depending on the axis of rotation in c), e.g. about a diameter or about the axis, the carbon granules can flow smoothly in the granule chamber in a number of ways. Any of these is allowable.

2. For any type of microphone which does not give repeatable results by the standard method, the following alternative method may be used. In this case, the artificial mouth is fed from alternate sources of measuring tone or noise of Hoth spectrum [1].

Following "standard conditioning" during the frequency characteristic measurement the normal tone stimulus is interrupted at approximately 1.5-second intervals by a short burst of noise at a sound level of the order of 98 dB (linear weighting of sound level meter). This level is referred to the same point as the tone used for measurement.

Note. – The timing of the noise bursts quoted here is based on the assumption that the total measurement time is about 60 seconds. Precise timing of the two signals is not thought to be essential.

3. For measurements other than sensitivity/frequency characteristics, e.g. subjective and objective loudness ratings, it may not be possible to use the above methods. It is, however, desirable to simulate the movements of the standard method as far as possible, even for situations where the handset is held in the hand for subsequent measurements.

Reference

[1] CCITT *Red Book*, Volume V, p. 76.

Recommendation P.76

DETERMINATION OF LOUDNESS RATINGS; FUNDAMENTAL PRINCIPLES

(Geneva, 1976)

Preface

This Recommendation is one of a set of closely related Recommendations concerned with determination of loudness ratings. The present one deals with the fundamental principles and the others, as follows, deal with certain additional matters.

P.48	"Specification for an intermediate reference system"
P.XXC (draft)	"Subjective testing method for determination of loudness ratings in accordance with Recommendation P.76" [5)]
P.64	"Determination of sensitivity/frequency characteristics of local telephone circuits to permit calculation of their loudness ratings"
P.XXE (draft)	"Calculation of loudness ratings based on sensitivity/frequency characteristics" [5)]
P.XXF (draft)	"Objective instrumentation for measuring loudness ratings" [5)]

1. *Introduction*

A speech path is, broadly, a transmission path that exists between a talker's mouth and the ear of a listener or, in the case of sidetone, between the mouth and ear of a talker. In typical face-to-face conversation, the speech is transmitted by means of the air path connecting the mouth and ear. Depending on environmental conditions, transmission may be:

a) more or less direct, as in the case of two persons conversing in an open, unobstructed location, such as a golf course;

b) largely indirect, as in the case of two persons conversing in a small, hard surfaced room where a large proportion of the energy reaching the ear may be due to reflections from the walls, ceilings and floor; or

c) something between the two extremes of *a)* and *b)*.

In the case of telephony, the air path is replaced by a system comprising:

a) an air path from the mouth to the telephone microphone;

b) an air path between the telephone earphone and the ear; and

c) a telephone connection consisting of the microphone, earphone and interconnecting circuitry together with a similar system for the reverse direction of transmission. The two situations – face-to-face and using the telephone – differ appreciably in detail but, for speech transmission purposes, they are alike as far as their function is to provide a means of both-way speech communication.

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5) These Recommendations are not yet complete. Partial texts are to be found in Annex 1 to Question 15/XII in Contribution COM XII-No. 1 for the study period 1977-1980.

Telephone engineering is concerned with providing telephone connections which, while not identical to the face-to-face situation, are comparable in effectiveness for providing a means of exchanging information by speech; such telephone connections should also optimize customer satisfaction within technical and economic constraints.

Various tools are used by transmission engineers in planning, design and assessment of the performance of telephone networks. Reference equivalent, based on the criterion of loudness of speech emitted by the talker and perceived by the listener, has been one of the most important of these tools; it provides a measure of the transmission loss, from mouth to ear, of a speech path.

The *reference equivalent method* is defined in Recommendations P.42 and P.72 and the fundamental principles are briefly explained in Annex 1 to Chapter I of the CCITT Handbook: *Transmission Planning of Switched Telephone Networks.* The method for determining *loudness ratings* of local telephone circuits is in accordance with rather similar fundamental principles to those of the reference equivalent method but embodies modifications which render the method much more flexible and should greatly simplify transmission planning.

A desire to depart from use of reference equivalents as defined by Recommendation P.72 arises for the following reasons:

1. Reference equivalents cannot be added algebraically; discrepancies of at least ± 3 dB are found.

2. Replication accuracy of reference equivalents is not good; changes in crew can cause changes of as much as 5 dB.

3. Increments of real (distortionless) transmission loss are not reflected by equal increments of reference equivalent; 10 dB increase in loss results in an increase in reference equivalent of only about 8 dB.

Use of loudness ratings defined in accordance with the principles given below should largely obviate these troubles.

In addition to these advantages, the same values of loudness ratings should be obtained whether the determination is by subjective tests, by calculation based on sensitivity/frequency characteristics or by objective instrumentation. The fundamental principles of the method are described below and these differ from those applicable to reference equivalents by the least possible extent to achieve the desirable flexibility.

The loudness rating (which has the dimensions and sign of "loss") is, in principle, like the reference equivalent, defined by the amount of loss inserted in a reference system to secure equality of perceived loudness to that obtained over the speech path being measured. Practical telephone connections are composed of several parts connected together. To enable the transmission engineer to deal with these parts in different combinations, loudness ratings must be defined in a suitable manner so that "overall", "sending", "receiving" and "junction" ratings can be used.

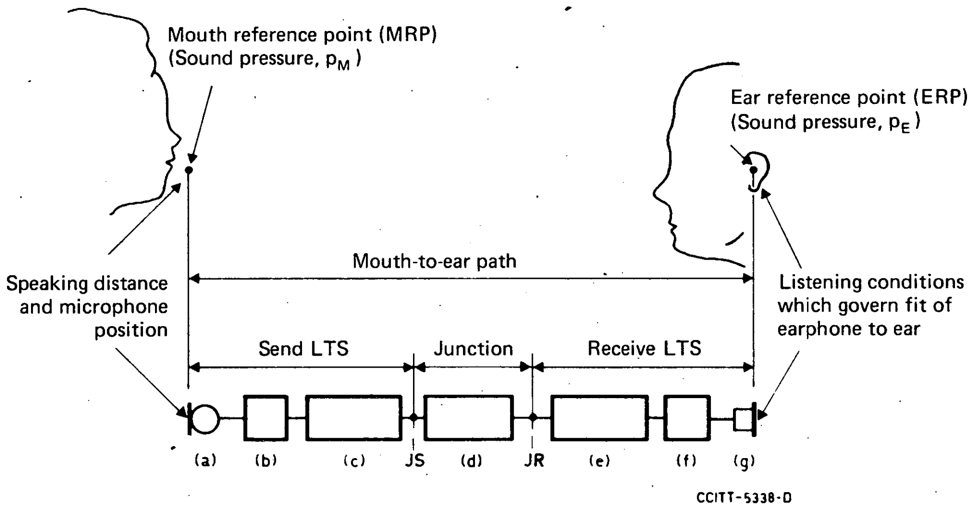
"Sidetone" loudness ratings can also be determined in an analogous manner. Sidetone reference equivalent is defined in Recommendation P.73 and sidetone loudness ratings are defined in 3. below.

2. *Definitions of loudness ratings for principal speech paths*

2.1 *General*

This section deals with principal speech paths, namely from a talker at one end of a connection to a listener at the other. Sidetone paths are treated in 3. below.

In general, loudness ratings are not expressed directly in terms of actual perceived loudness but are expressed in terms of the amounts of transmission loss, independent of frequency, that must be introduced into an *intermediate* reference speech path and the *unknown* speech path to secure the same loudness of received speech as that defined by a fixed setting of NOSFER. This implies that some interface exists or could, by some arrangement, be found in the unknown speech path into which the transmission loss can be introduced. In practice the unknown speech path is composed of a sending local telephone circuit coupled to a receiving local telephone circuit through a chain of circuits interconnecting the two local systems [6]. Figure 1/P.76 shows this subdivision of one principal speech path of a telephone connection. The interfaces JS and JR separate the three parts of the connection to which loudness ratings are assigned, namely: *sending loudness rating*, from the mouth reference point to JS; *receiving loudness rating* from JR to the ear reference point; and *junction loudness rating* from JS to JR. The *overall loudness rating* is assigned to the whole speech path from mouth reference point to ear reference point.



Note. –
(a) represents the microphone of the sending local telephone system;
(b) represents the electrical circuit of the telephone set of the sending local telephone system;
(c) represents the subscriber's line and feeding/transmission bridge of the sending local telephone system;
(d) represents the chain of circuits interconnecting the two local systems;
(e) represents the subscriber's line and feeding/transmission bridge of the receiving local telephone system;
(f) represents the electrical circuit of the telephone set of the receiving local telephone system;
(g) represents the earphone of the receiving local telephone system.

FIGURE 1/P.76 – **Subdivision of a telephone connection**

Note that in practical telephone connections:

a) the transmission loss of the junction may be frequency dependent;

b) the image impedances of the "junction" may not be constant with frequency and may not be resistive;

c) the impedances of the local telephone systems presented to the junction at JS and JR may not be constant with frequency and may not be resistive;

d) impedance mismatches may be present at JS or JR or both.

Overall, sending, receiving and junction loudness ratings (OLR, SLR, RLR and JLR respectively) are defined so that the following equality is achieved with sufficient accuracy for practical telephone connections.

$$\mathrm{OLR} = \mathrm{SLR} + \mathrm{RLR} + \mathrm{JLR}$$

2.2 *Definitions of overall, sending, receiving and junction loudness ratings*

Figure 2/P.76 shows the principles used to define the overall, sending, receiving and junction loudness ratings.

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[6] See Annex 2 for explanation of certain terms.

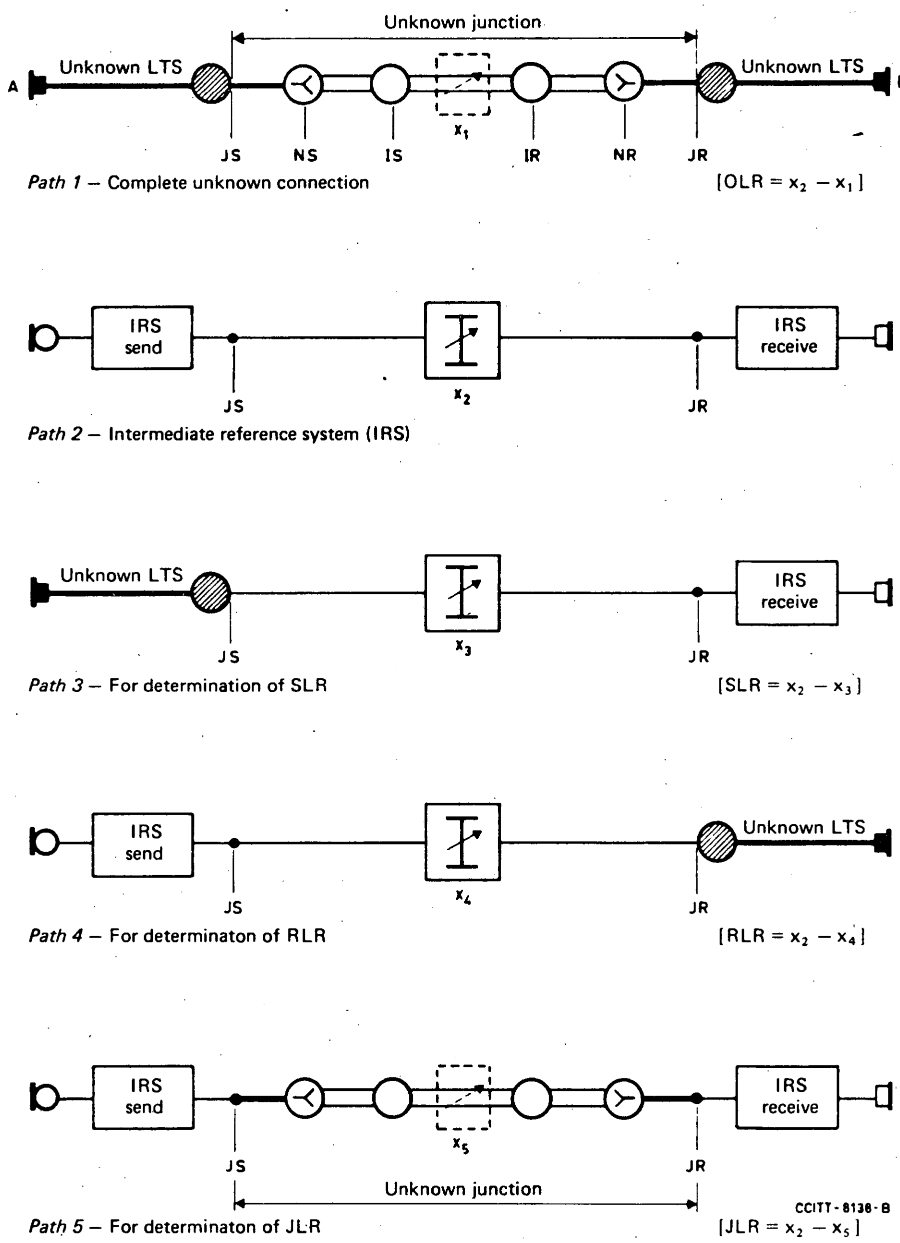

FIGURE 2/P.76 – **Principles used for defining OLR, SLR, RLR and JLR**

2.2.1 *Overall loudness rating*

Path 1 in Figure 2/P.76 shows the complete unknown speech path subdivided into local telephone systems and junction. In this example the junction comprises a chain of circuits represented by trunk junctions (JS-NS and NR-JR) and trunk circuits (NS-IS, IS-IR and IR-NR). A suitable arrangement for inserting transmission loss independent of frequency must be provided at some point such as in IS-IR.

Path 2 shows the complete IRS with its adjustable, non-reactive, 600 ohms junction between JS and JR.

The level of received speech sounds to which the additional loss x_1 in Path 1 and the junction attenuator setting x_2 of Path 2 are both adjusted is defined by using the fundamental reference system NOSFER with its attenuator set at 25 dB. When these adjustments have been made, the overall loudness rating (OLR) of the complete unknown connection is given by $(x_2 - x_1)$ dB.

2.2.2 *Sending loudness rating*

Path 3 in Figure 2/P.76 shows the IRS with its sending part replaced by the local telephone system of the unknown. The junction is adjusted to produce, via Path 3, the same loudness of received speech sounds as the NOSFER with its attenuator set at 25 dB. If x_3 is the required setting in Path 3, the sending loudness rating (SLR) is given by $(x_2 - x_3)$ dB.

2.2.3 *Receiving loudness rating*

Path 4 in Figure 2/P.76 shows the IRS with its receiving part replaced by the local telephone system of the unknown.

The junction is adjusted to produce via Path 4 the same loudness of received speech sounds as the NOSFER with its attenuator set at 25 dB. If x_4 is the required setting in Path 4, the receiving loudness rating (RLR) is given by $(x_2 - x_4)$ dB.

2.2.4 *Junction loudness rating*

Path 5 in Figure 2/P.76 shows the IRS with its junction replaced by the unknown chain of circuits as located in Path 1 of Figure 2/P.76 between JS and JR. The arrangement for introducing transmission loss, independent of frequency, must be provided as was required in Path 1. The additional loss is adjusted to produce, via Path 5, the same loudness of received speech as the NOSFER with its attenuator set at 25 dB. If x_5 is the required additional loss in Path 5, the junction loudness rating is given by $(x_2 - x_5)$ dB.

2.3 *Conditions under which loudness ratings are determined*

2.3.1 *General*

The loudness of received speech sounds depends upon certain factors that are not well defined under practical conditions of use, but must be defined as precisely as possible to obtain accurately reproducible loudness ratings. Clearly, as shown in Figure 1/P.76, the loudness rating is largely governed by the characteristics of the mouth-to-ear path. This path can be made precise by defining a *mouth reference point* at which the sound pressure p_M of speech emitted by the talker is measured or referred, and an ear reference point at which to measure or to which to refer the sound pressure p_E of speech reproduced by the earphone. These points can be chosen in a fairly arbitrary manner and this becomes important when loudness ratings are to be determined objectively; suitable definitions for such purposes are given in Recommendation P.64 which deals with measurement of sending and receiving sensitivity/frequency characteristics.

It is essential, however, to define vocal level, speaking distance, microphone position and listening conditions which govern the fit of the earphone to the ear. These are indicated in Figure 1/P.76. The essential features that define the conditions under which loudness ratings are determined are indicated in Table 1/P.76.

Some remarks on the items listed in Table 1/P.76 are given below.

2.3.2 *Intermediate reference system*

The intermediate reference system is defined in Recommendation P.48. It has been chosen with the following in mind:

a) It shall correspond approximately, as far as the shapes of sending and receiving frequency characteristics are concerned, with those of national sending and receiving systems in use at present and likely to be used in the near future. For this reason the frequency bandwidths for sending and receiving parts are confined to the nominal range 300-3400 Hz [7].

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[7] The IRS is specified for the range 100-5000 Hz (see Recommendation P.48). The nominal range 300-3400 Hz specified is intended to be consistent with the nominal 4 kHz spacing of FDM systems, and should not be interpreted as restricting improvements in transmission quality which might be obtained by extending the transmitted frequency bandwidth.

b) The absolute sensitivity has been chosen to reduce as much as possible changes in values from reference equivalents to loudness ratings.

c) In external form its handsets are similar to conventional handsets used in actual telephone connections.

TABLE 1/P.76 – **Conditions under which loudness ratings are determined**

No.	Item specified	Specification
1	Intermediate reference system	Recommendation P.48
2	Vocal level of speaker	As Recommendation P.72
3	Level of received speech sounds at which loudness is judged constant	NOSFER set at 25 dB
4	Handset position relative to talker's mouth	See Annex 1
5	Direction of speech	Head erect
6	Handset arrangement for listening	See 2.3.7 of the text
7	Conditioning of carbon microphones	Recommendation P.75

2.3.3 *Vocal level of speaker*

The vocal level at which speech is emitted from the speaker's mouth conforms to that in use for determining reference equivalents and is defined in Recommendation P.72. This approximates the level actually used by customers under good transmission conditions. It is defined in terms of the speech level at the output of the NOSFER sending system.

2.3.4 *Listening level*

The level of received speech sounds at which loudness is judged constant is defined by the vocal level (see 2.3.3 above) and the setting (25 dB) of NOSFER against which all the speech paths shown in Figure 2/P.76 are adjusted. This corresponds to a fairly comfortable listening level of the same order as that commonly experienced by telephone users.

2.3.5 *Handset position*

The position of the telephone handset relative to the talker's mouth is defined in Annex 1 to this Recommendation. It is intended to approximate fairly well the position used by customers under real telephone connections. The definition covers not only the distance between lips and mouthpiece but also the attitude of the microphone relative to the horizontal axis through the centre of the mouth opening. It is defined in such a way that the lips-to-mouthpiece distance becomes greater as the length of a handset is increased.

2.3.6 *Direction of speech*

The speaker shall hold his head erect and it will be assumed that speech is emitted horizontally from his mouth.

2.3.7 *Handset arrangement for listening*

The listener shall hold the handset in his hand with the earphone placed comfortably against his ear.

2.3.8 *Conditioning of carbon microphones*

Telephone handsets with carbon microphones usually require to be conditioned. This shall be done in accordance with Recommendation P.75.

3. *Sidetone loudness rating*

Loudness ratings of sidetone paths are determined in a manner identical to that used for overall loudness ratings with the following differences:

a) The microphone and earphone of the telephone to be tested may have to be separated physically from each other. Thus, for example, the earphone may be replaced by an identical item in a different handset. This will be the case when the measurements are made by subjective determinations [see draft Recommendation P.XXC [8)]].

b) In principle, arrangements must be made for introducing additional transmission loss, accurately adjustable, into the sidetone path. This additional loss (x_6) must be independent of frequency and must be introduced in such a way that the proper functioning of the telephone set under test is not affected. For measurements by objective determinations, the additional loss can be provided in the measuring equipment [see draft Recommendation P.XXF [8)]]. For subjective determinations, special arrangements are necessary [see draft Recommendation P.XXC [8)]].

c) The telephone set under test must be connected to the desired combination of subscriber's line, feeding/transmission bridge and impedance termination on the junction side of the feeding/transmission bridge; this impedance termination must be specified and need not be resistive.

d) The sidetone loudness rating is determined by adjusting the additional loss x_6 [see *b)* above] and the setting x_2 of the intermediate reference system, until both produce loudness of received speech sounds equal to that of speech sounds received via the NOSFER with its attenuator set at 25 dB. The sidetone loudness rating (STLR) is then given by $(x_2 - x_6)$ dB.

ANNEX 1

(to Recommendation P.76)

Definition of the speaking position for measuring loudness ratings of handset telephones

A method is described in Recommendation P.64 for measuring the sensitivities of commercial telephone sets; this method has been used to determine the sensitivities referred to in CCITT Laboratory Reports relating to loudness ratings. This Annex describes in more detail the speaking position which is referred to in the CCITT Laboratory Reports as the special guard-ring position.

The definition of a speaking position falls into two parts; description of the relative positions of mouth opening and ear-canal opening on a *modal* human head; and description of the angles that define the attitude in space of telephone handsets held to a modal head. For any given telephone handset, these descriptions together describe the relative special disposition of the microphone opening and the talker's lips, and hence the direction in which speech sound waves arrive at the mouthpiece and the distance they have travelled from a *virtual point source.*

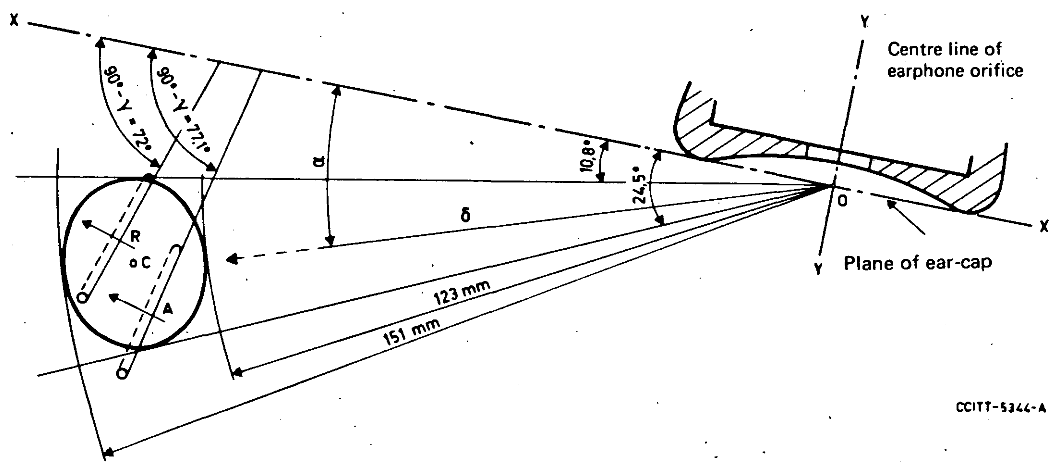
[8)] These draft Recommendation are to be found in Annex 1 to Question 15/XII, in Contribution COM XII-No. 1 for the study period 1977-1980.

The relative positions of the centre of the opening of the mouth and that of the ear canal can be described in terms of a distance δ and an angle α as shown in Figure 1. Point R in that figure represents the centre of a lip-ring located at the reference equivalent speaking position in accordance with Recommendation P.72. Position A is that used to determine ratings by the articulation method defined in Recommendation P.45. The approximately elliptical area encloses about 80% of the lip positions found in a sample of 3889 heads in the United States of America before 1930; averages of more recent, but less extensive and more diversified, results cluster round the point A.

A second angle is required to define the direction in which speech is emitted from the mouth into the mouthpiece of the microphone. It is difficult to measure this precisely and it may be defined in various ways. In Recommendations P.45 and P.72, reference is made to an angle β which lies in the plane passing through both ear openings and the centre of the lips. The angle β is defined in Recommendation P.72:

$$\beta = \arcsin \frac{\varepsilon}{\delta} - \alpha$$

where ε is the *semi-interaural* distance, i.e. half the length of a line joining the centres of the two ears. This line is, of course, perpendicular to the medial plane of the head and therefore ε is the length of a perpendicular from the centre of the plane of the ear-cap (Point 0 in Figure 1) to the medial plane of the head.



Note 1. – Points R and A are located as follows: A) δ = 136 mm α = 22° γ = 12.9°
R) δ = 140 mm α = 15.5° γ = 18°

Note 2. – Area shown includes about 80 % of the sample of 3889 lip positions.

Note 3. – A straight line tangential to the boundary of the area shown excludes 5 % of lip positions.

Note 4. – Solid line shows plane of lips; broken line shows lip-ring 1.6 mm thick.

FIGURE 1 – **Location of lip position relative to opening of ear canal**

The plane referred to in defining β does not coincide with the plane of symmetry of the handset [9] which must be used in practice for defining the position of a lip-ring for attachment to the mouthpiece of the microphone. It is therefore more convenient to use an angle γ lying in the plane of symmetry of the handset as shown in Figure 1. The direction of speech may be taken to lie in the medial plane of the head, although the precise direction in this plane is difficult to determine; for many test purposes it can be assumed to be horizontal. The line, in the plane of symmetry of the handset, formed by the intersection of this plane with the (vertical) medial plane of the head can be specified by its angle with the line XX in Figure 1 and this angle is denoted by γ. This line that defines γ can therefore be considered as being the vertical projection of the direction of speech on the plane of symmetry of the handset.

[9] These two planes are inclined to each other at the angle Θ referred to hereafter.

Although β has been defined in this manner and appears in the descriptions of speaking positions given in Recommendations P.45 and P.72, actual setting-up procedures usually adopted treat the values quoted for β as though they applied for γ. If we treat the matter strictly and take $\beta = 12.9°$ (Recommendation P.45), and $\varepsilon = 77.8$ mm, then γ would be about 15°. If $\gamma = 12.9°$, as has been customary, when $\Theta = 19°$, and $\varepsilon = 73.8$ mm, then $\beta = 10.9°$.

Fortunately this inconsistency is of no practical importance because such small differences affect the sensitivity values of telephone handset microphones to a negligible extent. In order to avoid continuing this confusion, the symbol γ is employed here and is defined as shown in Figure 1, i.e. as lying in the plane of symmetry of the handset and not in a plane that is perpendicular to the medial plane of the head.

The position of the centre of the lips as defined by A in Figure 1 is used also to define the new speaking position, but two additional angles must also be defined, namely: the earphone rotational angle $\o$ and the handset rotational angle ϑ. Earphone rotation is considered about an axis through the centre of the ear-cap (YY in Figure 1); handset rotation is taken about a longitudinal axis of the handset (XX in Figure 1); both angles are zero when the plane of symmetry of the handset is horizontal. Naturally, the earphone rotational angle is positive when the handle is pointed downwards away from the earphone and the handset rotational angle is positive in the sense that the upper part of the earphone is moved farther from the medial plane of the head.

The new speaking position is described by the following values for the distance and angles defined above:

$$\alpha = 22°,\ \gamma = 12.9°,\ \delta = 136 \text{ mm},\ \o = 37° \text{ and } \Theta = 19°$$

The angle γ cannot be determined very precisely and is not convenient for use when setting up a handset for test in front of an artificial mouth. The semi-interaural distance ε may be used in its place, and for the new speaking position $\varepsilon = 73.8$ mm.

The foregoing description of the speaking position has shown the complexities of expressing the relative location of the ear reference point and the lip ring centre, and the relative orientation of the earphone axis and the lip-ring axis. It is often more convenient, particularly in terms of constructing and setting up handset jigs, to express the position of the ear reference point [10] and the direction of the earphone axis with respect to the lip-ring. This is easier since the axis of the lip-ring is horizontal as would be the axis of an associated artificial mouth.

Use has been made of a method of vector analysis to determine the orthogonal coordinates of the handset ear-cap relative to the lip position when the handset is mounted in the special guard ring position. It is necessary to define a set of cartesian axes with origin at the centre of the lips (or equivalent lip position of an artificial voice) as follows:

x – axis: horizontal axis of the mouth, with positive direction into the mouth.

y – axis: horizontal, perpendicular to the x – axis, with positive direction towards the side of the mouth on which the handset is held;

z – axis: vertical, with positive direction upwards.

The ear reference point is defined by the vector:

(93.1, 76.9, 62.6) (dimensions in millimetres)

The handset is mounted so that the ear reference point lies at the intersection of the axis of the ear-cap with a plane in space on which the ear-cap can be considered to be resting. With some shapes of handset, this definition is not adequate; in such cases the position of the ear reference point relative to the handset should be clearly stated.

The orientation of the handset is defined by vectors normal to the plane of the ear-cap and the plane of symmetry of the handset:

Unit vector normal to plane of the ear-cap:

$$\pm\ (0.0728,\ -0.963,\ 0.260)$$

Unit vector normal to plane of symmetry of the handset:

$$\pm\ (0.637,\ -0.156,\ -0.755)$$

When using an artificial voice, the equivalent lip position must be used as the datum; this is not normally the same as the plane of the orifice of the artificial mouth.

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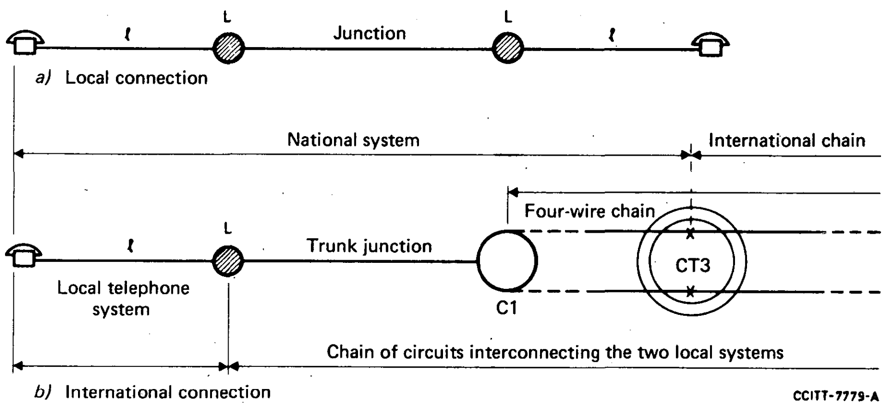
[10] See Recommendation P.64 for definition of ear reference point.

ANNEX 2

(to Recommendation P.76)

Explanations of certain terminology



l = subscriber's line
L = local exchange
C1 = primary centre

Terminology applying to parts of a telephone connection according to Recommendations G.101, G.111, G.121 and CCITT handbooks.

Note. – In the present Recommendation the word "junction" is used in a special sense to denote "chain of circuits interconnecting the two local systems" and the "junction attenuator" used in laboratory tests for determination of loudness ratings.

Recommendation P.77

METHOD FOR EVALUATION OF SERVICE FROM THE STANDPOINT OF SPEECH TRANSMISSION QUALITY

(Geneva, 1976)

a) *General*

The CCITT recommends that Administrations make use of telephone users' surveys in the manner of Recommendation E.425 as a means of measuring speech transmission quality on international calls.

Such surveys being call-related (in this instance to the last international call made) can be conducted either by the full use of the Recommendation E.425 questionnaires (when other valuable information is obtained on users' difficulties, e.g. knowing how to make the call, difficulties in dialling or understanding tones, etc.) or by making use of those questions solely related to transmission quality which appear in the Annex.

b) *Conduct of surveys*

In order to make valid comparisons between data collected in different countries, Recommendation E.425 should be strictly adhered to. Specifically the preamble to the Recommendation, the notes of intended use of the questionnaires and the precise order and wording of the questions should be rigidly followed.

c) *Treatment of results*

To provide quantative information suitable for comparisons, the subjective assessments (e.g. those obtained from Question 9.0 of the Annex) of excellent, good, fair or poor should be accorded scores of 4, 3, 2 and 1, respectively and a mean opinion score (MOS) calculated for all associated responses. Similarly for all those experiencing difficulty (under Question 10 of the Annex) a percentage of the total responses should be calculated. These two criteria of MOS and percentage difficulty are now internationally recognized and have been measured under many different laboratory simulated connections and practical situations.

The results can be classified in a number of ways, e.g. in terms of the call-destination countries or by nature/composition of the connection i.e. cable/satellite circuits, presence or otherwise of echo suppressors etc. Typical methods of presentation of the results are shown in Annex 2 to Question 2/XII (found in Contribution COM XII-No. 1 for the study period 1977-1980), in this case for several countries. It should be noted that in all presentations it is essential to show the number of responses.

Note. – Among the reasons which lead to the limitation of users' opinions of transmission quality to four classes, i.e. excellent, good, fair and poor, is the following. The experience gained in human factor investigations has shown that when a question which requires a selection from several different classifications is posed in aural form, e.g. by face-to-face interview or by telephone as with Recommendation E.425, the respondent is frequently unable to carry a clear mental separation of more than four categories. As a consequence, he is unable to draw on his short-term memory and judgement ability in a sufficiently precise manner to avoid confusion and gives an unreliable response. This restriction does not apply to other situations where a written presentation of the choices is used, in which case frequently five or more classes may be appropriate and shown to yield reliable responses.

ANNEX

(to Recommendation P.77)

Extract from the questionnaire annexed to Recommendation E.425

Reproduced below are the questions relating to transmission quality which appear in the questionnaire annexed to Recommendation E.425.

Which of these four words comes closest to describing the quality of the connection during conversation?

9.1	– *excellent*	1 [ ]	48
9.2	– *good*	2 [ ]	
9.3	– *fair*	3 [ ]	
9.4	– *poor*	4 [ ]	

10.0 *Did you or the person you were talking to have any difficulty in talking or hearing over that connection?* YES 1 [ ] NO 2 [ ] 49

(If answer is "yes") probe for nature of difficulty, but without suggesting possible types of difficulty, and copy down answers verbatim: e.g. *"Could you describe the difficulty a little more?"*

....

....

At end of interview, categorize the answers in terms of the items below:

10.1	– low volume	1 [ ]	50
10.2	– noise or hum	1 [ ]	51
10.3	– distortion	1 [ ]	52
10.4	– variations in level, cutting on and off	1 [ ]	53
10.5	– crosstalk	1 [ ]	54
10.6	– echo	1 [ ]	55
10.7	– complete cut off	1 [ ]	56
10.8	– other (specify) .... ←	1 [ ]	57

SECTION 7

MEASUREMENTS FOR MAINTENANCE OF SUBSCRIBERS' TELEPHONE EQUIPMENT AND FOR FACTORY ACCEPTANCE TESTING

Recommendation P.81

MAINTENANCE OF SUBSCRIBERS' EQUIPMENT

(modified at Geneva, 1964, and at Mar del Plata, 1968)

To ensure good transmission on international connections, the CCITT recommends periodic testing of each subscriber's equipment.

Different procedures exist for making check tests from the exchange of subscribers' stations under working conditions by means of subjective or objective measurements.

The most important of these are the following:

A. *Subjective measurements*

a) Quick conversation test;

b) Complete telephonometric test.

B. *Objective measurements*

It is possible to envisage maintenance also based on the procedures used for factory acceptance testing. (This form of maintenance does not involve the exchange.)

A. SUBJECTIVE MEASUREMENTS

a) *Quick conversation test*

This method is used mainly in the United States (by the American Telephone and Telegraph Company) and in Switzerland. Furthermore, in the United Kingdom Post Office, the transmission quality of telephone equipment in public call-boxes and subscribers' stations having extension sets is assessed by means of a conversation with the exchange test desk clerk [1)].

b) *Complete telephonometric test*

This method seems to be no longer used.

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

B. OBJECTIVE MEASUREMENTS

a) *Measurements made from the test desk*

In Switzerland the checking of transmission quality is made subjectively by an exchange of conversation with the test desk which deals with fault control for each terminal trunk exchange. From this test desk, measurements and checks of subscribers' lines can also be made from the point of view of insulation, loop resistance, transmission of dialling impulses, etc.

b) *Electrical measurements of a general nature*

To check the transmission quality of subscribers' telephone equipment in service, the Dutch Administration uses the same measuring equipment and methods as for factory testing; nevertheless it must be 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 sine-wave swept linearly with respect to time over the range 300-3000-300 Hz, 25 times per second. In the tester a moving coil microphone unit with suitable frequency characteristics is made to serve as either an artificial voice or an artificial ear, as required. A closed coupler is used for each measurement and the telephone microphone is subjected to a sound pressure of approximately 95 dB rel. 20 microbar during the transmitting efficiency test.

A number of types of subscribers' instruments are in use in Australia and have a variety of mouthpieces and receiver insets. Each of these will give slightly different performance figures. To avoid the need for a multipoint switch to adjust the instrument calibration for each type, a common calibration adjustment is used and the figure of merit for the measurement is read off a meter scale calibrated in decibels and is compared against the appropriate limit value listed on a card attached to the tester. On the same meter a 0-100 scale is used to indicate the line current directly in milliamperes. The decibel scale is used also in measuring the subscriber's line loss.

The tester is calibrated *in situ* so that its readings approximate to those which the telephone would give under limiting line conditions, which are the conditions under which new telephones are acceptance tested. Consideration was originally given to measuring the volume efficiency of the telephone plus its local line, on the basis that instruments which had fallen below their initial performance might still give adequate performance on shorter lines. This possibility was rejected because only the volume efficiency is measured and if this is very low other impairments may be present (e.g. the receiver diaphragm may be deformed), and because telephone instruments possessing faults of a marginal nature may have unsatisfactory service life.

A prototype instrument received a very favourable reception in a field trial. Both the maintenance technicians and the subscribers expressed satisfaction that faults could now be more readily diagnosed and demonstrably remedied. Further field testing on a wider scale with about 20 testers distributed over a number of local exchange areas is being arranged.

United States of America. – The American Telephone and Telegraph Company assesses subscriber sets with an electro-acoustic rating system (EARS) [2]. 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 Administration of the Federal Republic of Germany uses the following methods.

The testing of telephone equipment to check the transmission quality of subscribers' telephone equipment in service is applied mainly to the measurement of microphone and receiver capsules, because their transmission quality depends very much on the material used and the quality of manufacture. Specifications have been fixed for microphone and receiver capsules against which they are checked by means of the equipment for objective measurement of reference equivalents described in Annex 28, Part II of Volume V of the *Red Book*.

The equipment for the objective measurement of reference equivalents enables the reference equivalents of microphone and receiver capsules to be measured. For microphone capsules, non-linear distortion and microphone noise are measured at the same time as reference equivalent by means of the modulation products. Furthermore, it is possible to check the sensitivity/frequency characteristic by means of a visual display.

The microphone capsules are divided according to their sensitivity into groups in steps of 3.5 dB and the receiver capsules in steps of 2.5 dB. These groups correspond for microphone capsules to values of sending reference equivalent 8 to 4.5 dB, 4.5 to 1 dB and 1 to −2.5 dB and, for receiver capsules, to values of receiving reference equivalent 0 to −2.5 dB, −2.5 to −5 dB and −5 to −8 dB. This allocation into groups is then used to associate the capsules with corresponding groups of subscribers' lines (loop resistance 0 to 250 ohms, 250 to 500 ohms and 500 to 750 ohms (see Annex 3 to Chapter II of the CCITT Handbook: *Transmission Planning of Switched Telephone Networks*, under "Federal Republic of Germany").

For this allocation the capsules are stamped with the figures I, II, or III. Thus it is possible not only to compensate for too high values of reference equivalent of subscribers' lines but also, on replacement of capsules when the telephone set is repaired, to make sure that the capsules have not been changed after being put into service. For this reason the lineman who is dealing with the location of faults must always have with him some capsules of the various groups; capsules which are removed from subscribers' sets are checked at the headquarters stores depot by the equipment for the objective measurement of reference equivalents so as to determine whether they are still serviceable.

The measurement and grouping of microphone and receiver capsules with the aid of the equipment for the objective measurement of reference equivalents were introduced several years ago in the Federal German Posts and Telecommunications Administration. Each headquarters has at its telecommunications stores depot one such measuring equipment operated by non-specialist female staff. The measuring precision is so high that when the same capsule is measured with a different measuring equipment the differences are less than 1 dB. The grouping of capsules and their correct allocation to the telephone sets can, as 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.

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[2] See Annex 1 to Question 15/XII in Part II of Volume V of the *White Book*, and the following article: SULLIVAN (J. L.) – A laboratory system for measuring loudness loss of telephone connections; *B.S.T.J. 50*, No. 8, October 1971, pp. 2663-2739.

Recommendation P.82

FACTORY ACCEPTANCE TESTING OF SUBSCRIBERS' EQUIPMENT

(modified at Geneva, 1964)

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 then measurements are made of:
      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 breakdown 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 PTT 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 a spectrum restricted to the band 300-3400 Hz is applied to the microphone in an acoustic chamber. The microphone is connected to an electrical circuit which, for both a.c. and d.c., is equivalent to the average conditions obtained when the microphone is connected in the telephone network. The d.c. resistance is also measured in this condition at the current which would apply in practice. The noise voltage produced by the microphone is measured by means of a d.c. voltmeter connected in a Graetz circuit. The voltmeter indicates approximately the r.m.s. value.

For measuring the telephone receiver, the reciprocity principle is used by applying the white noise to the receiver, acoustically, and measuring the voltage across the receiver.

In this case too, the receiver is connected in a circuit which has the same nominal impedance as that of normal telephone equipment.

The levels measured in this way yield a statistical distribution and the Administration requires that no microphone or receiver capsule may be accepted which departs more than ± 3 dB from the mean. The absolute level of the mean is also fixed by the Administration.

As far as the sensitivity/frequency characteristic is concerned the manufacturers are required to guarantee, for each capsule, that this complies with the tolerances specified in the Administration's standard. Experience has shown that the Dutch Administration can confine itself to checking from time to time by sampling whether the relevant clauses concerning the sensitivity/frequency characteristic are being observed. In general, the Administration uses the same measuring equipment for checking as is used in the factory. The measuring equipment used by the manufacturer for final checking in the factory must have been approved by the Administration. Furthermore, the Administration has reserved itself the right to make measurements on the microphones and receivers in the factory.

The transmission characteristics of each induction coil must be guaranteed by the manufacturer. He can conduct his checking during manufacture in a manner approved by the Dutch Administration.

FEDERAL REPUBLIC OF GERMANY

For tests made from the transmission point of view, the Federal German Posts and Telecommunications Administration uses, for the acceptance of subscribers' telephone equipment by its telecommunications stores depots, the equipment for objectively measuring reference equivalent described in Annex 28, Part II of Volume V of the *Red Book*. It has been possible to prove that, in the case of good manufacture, there are scarcely any faults in assembling telephone equipment. It is therefore sufficient to make random tests at the time of acceptance. Nevertheless, on delivery all microphone and receiver capsules are again measured and grouped as described in Recommendation P.81. Furthermore, all reconditioned telephone equipment must be tested, but this is an easy matter because only a small number of items is generally involved.

When testing telephone equipment the mean sending and receiving loss is measured between the frequency limits of 200 and 400 Hz. The resistance of line, receiver and microphone are each replaced by a 600-ohm resistor.

UNITED KINGDOM POST OFFICE

General. – The processes of manufacture and the measurements made by the manufacturer are liable for inspection at any time by the Inspection Branch. Acceptance measurements are made on every piece of equipment manufactured or on samples chosen at random at the discretion of the Inspection Branch. The nature of the acceptance measurements is determined by agreement between the purchasing authority and the manufacturer before the contract is placed.

Electro-acoustical measurements on telephone microphones and receivers. – Each manufacturer is required by the Post Office to equip himself with measuring equipment of an approved design. This employs specified bandwidths of continuous-spectrum noise. To ensure that each manufacturer uses 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 1 V, 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 overall sensitivity of the microphone is measured by applying wideband noise (300-3400 Hz); for the microphone Transmitter Inset No. 16, the permitted tolerance is ± 2 dB on the specified value. Secondly, a measurement is made with each of three narrow bands, to check that the frequency response is within tolerance.

For testing telephone receivers, the noise source feeds the receiver being measured which is placed on an artificial ear (see Annex 11 in Part II of Volume V of the *Red Book*); the output voltage of the artificial ear is measured across a standard circuit by means of a voltmeter. For the type of subscriber's telephone receiver normally manufactured, acceptance measurements are specified with three narrow bands of noise.

Complete telephone sets are not measured for performance. As all the component parts have been separately tested before assembly, only a simple check to ensure that the telephone does work is considered necessary.

SWITZERLAND

Subscribers' equipment and spares purchased by the Swiss Telephone Administration are acceptance-tested. This work is entrusted to the stores testing section of the Research and Testing Division. Bulk tests are made with appropriate measuring apparatus. Some components are furnished to the manufacturers of telephone apparatus after being tested by the PTT, for example, the capacitance and insulation of condensers, the handset and various cords. The return speed and impulse ratio of the dial contacts and the short-circuit contacts are tested in a few seconds with SC12/SC14 (Sodeco) equipment.

The subscribers' equipment (without handset) is acceptance-tested with TLPG3 (Zellweger) measuring apparatus in a minute or so. This test covers insulation, loudness of the bell, impedance for the circuit conditions applying when calls are being received, composite attenuations for sending, receiving and sidetone, at 400 and 1600 Hz if necessary for two different feeding currents; the check may also be extended to cover high-frequency interference suppression, the busy impulse for party lines and any auxiliary circuits in the subscriber's equipment.

Microphones and earphones are checked rapidly with the KP51/MPG12 (Autophon/Zellweger) measuring apparatus described in Annex 29, Part II of Volume V of the *Red Book*. Checks are made of the reference equivalent and the frequency curve as well as the resistance and noise level of the microphones and the centring of the transmission system in respect of the earphones.

Manufacturers of telephone equipment and components use similar measuring apparatus.

Apparatus returned as faulty by the operating services is tested in the same way and by the same testing section as that delivered by the suppliers.

PART II

SUPPLEMENTS TO THE SERIES P RECOMMENDATIONS

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SUPPLEMENTS

Supplement No. 1

PRECAUTIONS TO BE TAKEN FOR CORRECT INSTALLATION AND MAINTENANCE OF AN IRS

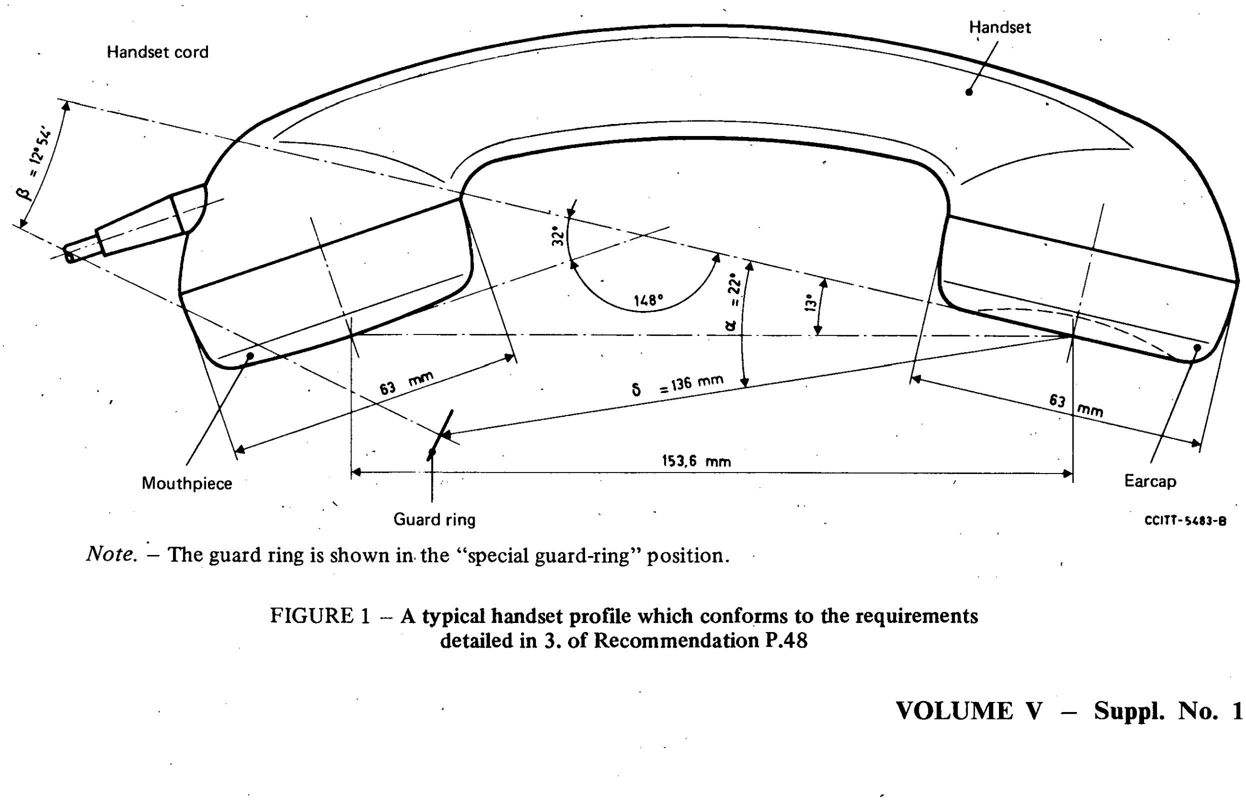
(Geneva, 1976)

(quoted in Recommendation P.48)

1. *General*

The comments below are for the purpose of information and advice and are aimed at ensuring the correct installation and maintenance for satisfactory working conditions of an IRS conforming to Recommendation P.48.

The requirements of 3. of Recommendation P.48 are illustrated in Figure 1 by a typical handset profile.



Note. – The guard ring is shown in the "special guard-ring" position.

FIGURE 1 – **A typical handset profile which conforms to the requirements detailed in 3. of Recommendation P.48**

2. *Gain adjustments*

2.1 *Sending part*

Facilities may be provided for adjusting the gain by approximately ± 5 dB in steps of not more than 0.1 dB from its nominal value by means of calibrated attenuators [1]. This is to allow a record to be maintained of any adjustments made to the nominal gain necessitated, for example, by replacement of microphone capsules. To reduce the possibility of faulty operation the adjustment controls shall not be readily accessible, e.g. should be adjustable only by screwdriver or special tool.

2.2 *Receiving part*

Facilities may be provided for adjusting the gain by approximately ± 5 dB in steps of not more than 0.1 dB from its nominal value by means of calibrated attenuators [1]. This is to allow a record to be maintained of any adjustments made to the nominal gain necessitated, for example, by replacement of earphone capsules. To reduce the possibility of faulty operation, the adjustment controls shall not be readily accessible, e.g. should be adjustable only by screwdriver or special tool.

3. *Special cases for the speech path*

If "unknown"/"unknown" tests are required to be conducted, and the loss in the junction in these circumstances is likely to be less than 10 dB, it is recommended that the junction described above is replaced by a "junction circuit" detailed in the following.

3.1 *Junction circuit*

The junction circuit will provide all the facilities described in 4.3 of Recommendation P.48, but will be designed such that for all values of attenuation, the return loss of both input and output impedances against $600\underline{/0^\circ}$ ohms shall be not less than 20 dB over a frequency range 200-4000 Hz, and not less than 15 dB over a frequency range 125-6300 Hz.

The gain of any active devices (e.g. buffer amplifiers) used to meet the above requirements shall be constant to within ± 0.05 dB over the frequency range 200-4000 Hz.

3.2 *Interfacing with commercial local telephone circuits, other IRSs, test equipment, etc.*

Compatibility of circuit configurations at interfaces JS and JR (Figure 1/P.48) is essential. In general, interfaces may be designed basically to provide either:

a) balanced and floating circuit configurations, or

b) unbalanced circuit configurations.

Whichever is adopted, precautions (such as the inclusion of transformers) need to be taken when elements having dissimilar configurations require to be interconnected.

4. *Stability*

Before the IRS is calibrated or used for subjective tests sufficient time shall be allowed, after connection of power supplies, for stable operating conditions to be reached. After this initial period the sending and receiving sensitivities, and the gain of active elements such as the junction circuit or interface units (if provided), shall not deviate with time at any frequency within the range 200-4000 Hz by more than ± 0.1 dB from their nominal values at that frequency.

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[1] Continuously adjustable control is acceptable provided the calibration markings permit resetting with an error of not more than 0.1 dB.

5. *Noise limits*

5.1 *General*

The dominant source of inherent noise in the IRS will generally lie in the sending part, due to the high gain required in this part of the speech path when relatively insensitive linear microphones are used. For the purpose of specifying noise limits therefore, the send part, the receive part, and the junction will be considered independently.

Noise limits will be specified in terms of:

a) readings of a psophometer having telephone weighting, and

b) measurements using 1/3-octave filters.

The purpose of *b)* is to ensure that no single frequency (in particular harmonics of mains frequency) occurring within the frequency range of the IRS, and which may not be revealed by *a)*, is above a specified level.

Measurement procedure shall be as follows. It is important that the physical locations of the handset and amplifier are approximately as they will be in actual use of the IRS.

5.2 *Sending part*

The microphone shall be replaced by a resistor equal in value to the modulus of the impedance of the microphone at 1000 Hz. The level across a $600\underline{/0^\circ}$ ohm impedance terminating the sending part, when correctly set up, shall be measured via 1/3-octave filters covering the frequency range 40-5000 Hz. The noise spectral density shall not exceed -95 dB V^2/Hz.

The wideband noise level measured across the terminating impedance using a psophometer with telephone weighting shall not exceed -70 dBm0p.

5.3 *Receiving part*

The permissible noise levels in the receive part of the IRS cannot be specified, as for the sending part, in terms of diret psophometric measurements. The practical realization of an IRS may involve earphones of different sensitivities and different earphone feeding arrangements. Strictly, it is necessary to specify noise levels in terms of the sound pressure level produced, for example, in an artificial ear. However, due to its low level, it is difficult to measure the noise objectively and therefore limits cannot be specified. Steps must be taken to minimize noise in the receive part by careful attention to design. For example, since negative gain is present, active elements should be associated as far as possible with that part of the circuit near its input, provided the conditions of 7.2 below are not infringed. It may be necessary to apply other methods (possibly subjective) to ensure that the noise level present at the earphone of the receive part, when the input is closed with $600\underline{/0^\circ}$ ohms, is not such that the standardized performance of the IRS is impaired. As a guide, the level of noise generated at the receiving end should sound less loud than a noise of -70 dBVp applied to the input of a 30 dB attenuator with the receiving end connected to its output. This could be checked by noting whether the noise heard at the output of the receiving end decreases appreciably when the 30 dB attenuator is increased in value to, say, 40 dB.

5.4 *Junction and junction circuit*

If the junction contains only an attenuator, no appreciable thermal noise will be present. If active elements are included the noise level, measured across a $600\underline{/0^\circ}$ ohm load at the output of the junction circuit when its input is closed with $600\underline{/0^\circ}$ ohms, will be too low to be measured with conventional test equipment. Alternative methods must be used to ensure that the noise level is not such that the standardized performance of the IRS is impaired. An approximate subjective check could be made in the manner indicated in 5.3 above.

6. *Crosstalk*

The general design of the IRS shall ensure that electrostatic and electromagnetic coupling between wiring, components and complete circuit elements is such that crosstalk between send and receive parts of the IRS when each is independently terminated in $600\underline{/0^\circ}$ ohms shall be negligible.

7. *Non-linear distortions*

Measurements shall be made at frequencies from 160 to 4000 Hz in 1/3-octave steps, using, for example, a total harmonic distortion measuring set.

7.1 *Sending part*

The microphone shall be replaced by a sinusoidal signal source having an impedance equal to the modulus of the microphone impedance at 1000 Hz, and the input signal level shall be adjusted until the level across a $600\underline{/0^\circ}$ ohm impedance terminating the sending part is +15 dBV. Under these conditions, the total harmonic distortion content at the load resistor shall not exceed 0.2%.

7.2 *Receiving part*

The earphone shall be replaced by a load having a value equal to the modulus of the earphone impedance at 1000 Hz. A sinusoidal signal source having an impedance of $600\underline{/0^\circ}$ ohm shall be applied to the 600 ohm input of the receiving part and its level set to +15 dBV. Under these conditions, the total harmonic distortion content at the load impedance shall not exceed 0.2%.

7.3 *Junction and junction circuit*

A sinusoidal signal source having an impedance of $600\underline{/0^\circ}$ ohm shall be applied to the input of the junction circuit (set at 0 dB loss) and its level set to +15 dBV. The total harmonic distortion content, measured across a $600\underline{/0^\circ}$ ohm terminating impedance, shall not exceed 0.2%.

7.4 *Transducers*

The electro-acoustic transducers used should have a harmonic distortion of less than 2% when measured with a sound pressure of 10 dB above 1 Pa at the mouth and ear reference points.

8. *Frequency of routine tests and procedure when a fault is recognized*

8.1 *Routine tests*

Check of the performance of the IRS shall be made on the following occasions:

a) on the initial setting-up of the IRS;

b) periodically, not less than twice per year;

c) on the commencement and conclusion of a major series of tests;

d) during a major series of tests; the system should be checked daily against 7.2 of Recommendation P.48.

The necessity for *c)* will depend, to some extent, on the time interval between *b)* and *c)*.

The results of all measurements obtained under the above headings shall be recorded, with the dates on which they were made. This running record shall be scrutinized for incipient faults.

8.2 *Procedure when a fault or maladjustment is discovered*

A strict watch should be kept on the performance of the IRS for any abnormalities, both at the time of the routine objective measurements, and during the conduct of subjective tests. In the latter case judicious scrutiny of results as the test proceeds may reveal abnormal values, for example, of attenuation not attributable to the test conditions or normal experimental variation.

A fault condition revealed, for example, by a change of gain of send or receive parts which exceeds the normal and expected day-to-day variation should not be cleared by readjusting the gain controls; the cause of the change should be investigated and rectified.

The above provision excludes gain readjustment necessitated by a change of transducer capsule.

When a fault has been cleared, such tests shall be applied as necessary to ensure that the performance of the IRS is within the specified limits in all respects.

9. *Calibration methods*

9.1 *General*

Calibration, and check of performance is essential when an IRS is initially brought into service, and periodic checks are required to ensure that the performance characteristics remain within their specified tolerances. The calibration procedure considered below is based on the required overall performance of the IRS as detailed in Recommendation P.48. Characteristics of individual components of the IRS, such as transducers, equalizers, etc., although useful as an aid to maintenance, will depend upon individual practical realizations of an IRS and cannot therefore be considered here.

9.2 *Methods of measuring sensitivities of sending and receiving parts*

The principles underlying these measurements are described in Recommendation P.64. The actual measuring technique adopted may take one or more of the following forms:

a) measurements at specific frequencies, using an oscillator and level measuring set;

b) use of sweep-frequency oscillator and graphic level recorder;

c) use of OREM equipment;

The technique under *a)* is suitable for accurate initial calibration, or recalibration of the IRS when necessary.

The technique under *b)* provides a permanent record of the shape of the sensitivity/frequency characteristics.

The technique under *c)* is useful for rapid day-to-day visual checks of the sensitivity/frequency characteristics. It will also provide a single reading which is related to the reference equivalent of the sending or receiving parts. The latter facility is not a sufficient criterion of the performance, as differently-shaped sensitivity/frequency characteristics may produce the same reading.

The technique under *b)* and *c)* should be supplemented by an accurate measurement of electrical gain, say at 1000 Hz.

Interpretation of the traces produced by these two methods would be facilitated by the use of suitable masks indicating permitted tolerances. The values to be ascribed to tolerances of sending and receiving sensitivities are given in 7. of Recommendation P.48.

In determining sensitivities it must be ensured that the sound pressure level used for sending sensitivity measurements, and the electrical signal level applied to the receiving part for receive sensitivity measurements, are not so high as to cause overloading at any frequency or so low that the signal-to-noise ratio is unsatisfactory.

Supplement No. 2

METHODS USED FOR ASSESSING TELEPHONY TRANSMISSION PERFORMANCE

(Geneva, 1976)

(quoted in C. of Recommendation P.74)

1. *Introduction*

This Supplement gives brief descriptions of the methods for assessing telephony transmission performance that are recommended by the CCITT or have been employed over the last two study periods (1968-1972 and 1973-1976) in studying Questions assigned to Study Group XII. Some of the methods are already fully described in Recommendations and these will merely be listed here with reference to the appropriate Recommendation. Other methods are also described in detail elsewhere; the essential features of these are given here with a brief description of how they are conducted, with references to descriptions published elsewhere.

2. *List of methods*

a) loudness comparison for speech (reference equivalents and loudness ratings);

b) articulation (AEN) ratings;

c) listening opinion tests;

d) conversation opinion tests.

3. *Brief descriptions and references to more complete descriptions*

3.1 Loudness comparisons for speech are intended to quantify the relative level at which speech, transmitted over a given telephone connection, reaches the ears of customers while they are listening to a person talking at the other end. In order to standardize the measuring procedure, the talking and listening conditions are each controlled in a specified manner. Circuit and room noise are excluded from the determination and so the results are governed by the overall mouth-to-ear transmission loss of the speech path being considered. The present recommended method is given in Reference 35 and proposals for new methods are to be found in Reference 42. More general information can be found in Reference 26.

3.2 Articulation measurements are based on measurement of the fraction of speech sounds recognized correctly when transmitted and reproduced over the speech path in question. Circuit noise and room noise at specified levels should be present and the result is affected by their levels. Just as for 3.1 above, talking and listening conditions are controlled. The method recommended by the CCITT is described in Reference 34. Other information will be found in Reference 26.

3.3 Listening opinion tests are conducted using speech material in the form of sentences and the listeners judge the speech received over the path according to a given criterion. Two such criteria are loudness preference and listening effort and the scales used for these are as follows.

– Loudness preference scales:

Opinion scale No. 4	*Opinion scale No. 4A*
A Much too loud	A Much louder than preferred
B Too loud	B Louder than preferred
C OK	C Preferred
D Too quiet	D Quieter than preferred
E Much too quiet	E Much quieter than preferred

Note. – The working of Opinion scale No. 4A is preferable to that of No. 4.

– Listening effort scale:

Opinion scale No. 7

A Complete relaxation possible; no effort required

B Attention necessary; no appreciable effort required

C Moderate effort required

D Considerable effort required

E No meaning understood with any feasible effort

Listening tests using sentence material can also be conducted as pair-comparisons but these should be undertaken with due consideration to ensure that subjects become suitably adapted to each test condition.

The votes using the above scales are scored 4, 3, 2, 1 or 0 and mean values calculated; the result is called "mean opinion score". The method has been widely used and further details can be found in Reference 26. The speech is usually recorded so that it can be reproduced at a given level. Circuit and room noise may be present and their effects are taken into account.

3.4 Conversation tests may be conducted either as interviews after real customers have made actual calls or as laboratory tests. Further information regarding methods recommended by the CCITT for the former are given in Reference 37. Laboratory conversation tests are intended as far as possible to reproduce under laboratory conditions the actual service conditions experienced by telephone customers. To this end it is necessary to choose the subjects suitably and to administer the tests in an appropriate manner. A method intermediate between field observations and laboratory tests is that used by the AT&T Co. and called SIBYL (see also Reference 16). Particulars of the method used by the United Kingdom Post Office are given below.

3.4.1 *Method for conducting conversation tests*

3.4.1.1 *Choice of subjects*

Subjects taking part in the conversation tests were chosen at random (actually in alphabetical order of surname initial) from the Research Station personnel, with the provisos that:

a) they were not directly involved in work connected with assessment of the performance of telephone circuits; and

b) they had not participated in any objective test whatever for at least the previous six months. The numbers of male and female subjects were generally found to be roughly equal. Subjects were arbitrarily paired in the experimental design prior to the test and remained thus paired for its duration, but steps were taken, in the pairing process, to avoid gross inequalities in rank.

3.4.1.2 *Environment*

Subjects were seated in separate sound-proof cabinets near the point from which the experiment was controlled. Room noise (50 dB Hoth spectrum [2)]) was introduced for all circuit conditions in each test.

3.4.1.3 *Method of establishing the connection*

The telephone sets used by subjects were, in appearance and feel, identical to the standard United Kingdom Post Office Telephone No. 706. The means of establishing telephone contact between subjects were made as realistic as possible. The calling subject, on lifting the handset, obtained dialling tone; the dialling of a predetermined number of digit trains established the connection, and suitable fixed delays were provided between completion of dialling and commencement of ringing tone, and between the commencement of ringing tone and the application of ringing current to the called-subject's bell.

[2)] For this measurement, Dawe Instruments Sound Level Meter Type 1400F, weighting characteristic "A" meter "slow" was used.

3.4.1.4 *Conversation task*

The means adopted to ensure that the conversations were purposeful, and to ensure that subjects fully exploited the transmission capabilities of the text circuit were as follows:

Each subject was provided, for each conversation, with a set of six postcard-size reproductions of paintings in the Tate Gallery. The pictures embraced diverse schools of art (i.e. impressionist, abstract as well as pictorial views, etc.) and the sets used by each pair were identical. Cards were identified by a code, comprising pairs of letters on the rear only (all other descriptive matter being obscured): identical pictures in each set bore the same codes.

The subjects were asked to imagine that a choice was to be made from the set for hanging in the Post Office Research Station restaurant. Immediately before each conversation, each subject was given a minute or so to arrange the cards, independently, in his own order of preference, and to note the identifying letters shown on the backs of the cards. The probability of the orders of preference of the two subjects being identical was extremely remote, and on this basis, when connection had been established over the test circuit, they were to negotiate so as to arrive at an order of the six pictures which satisfied both. The final or compromise order was also noted.

The time occupied for each conversation was about five or six minutes.

3.4.1.5 *Instruction of subjects*

In order to standardize the instructions given to subjects, and to ensure that each subject was fully apprised of what was required of him, the following procedure was adopted:

a) After selection of prospective subjects as detailed in 3.4.1.1 above initial approach was made by telephone to ascertain their willingness to take part in the tests. A confirmatory note was sent to those agreeing, with a brief description of the conversational task.

b) Prior to the first condition of the first entry, the subjects were invited to listen to a recording in the presence of the controlling officer, which gave more detailed instructions on the conversational task. Any queries raised by the subjects were dealt with at this stage. A text of the recording is given in Annex 1.

3.4.1.6 *Procedure and subjects' assessment of the connection*

After listening to the recording, the subjects were taken to their respective cabinets and were handed their picture card sets and an opinion form (see Annex 2). After a short period during which they arranged and noted their orders of preference of the pictures, a lamp indication was given by the controlling officer to the subject nominated to be the caller, who then set up the connection by dialling, and the conversation proceeded normally. At the conclusion of the conversation, the subjects noted the agreed picture set order and replaced the handsets. They were then called, separately, by the controlling officer over the test connection and reminded to complete Section 5 of the opinion form. They were then asked if they had any difficulty (the actual question being: "Did you, or the person who spoke to you, have any difficulty in talking or hearing over that connection?"). If the answer was "Yes" the controlling officer asked the subject to explain the nature of the difficulty which he (the controlling officer) noted.

3.4.1.7 *Experimental design*

Each experiment is usually based on a 12 × 12 graeco-latin square, with twelve circuit conditions, twelve pairs of subjects and twelve sets of picture cards. As stated in 3.4.1.1 above subjects were paired in the experimental design prior to each experiment and remained thus paired for its duration. Also, subjects occupy respectively the same cabinets throughout each test. For subject pairs 1 to 6, subject A of the pair initiates the call to subject B for odd-numbered conditions: for subject pairs 7 to 12, subject A initiates the call for even-numbered conditions.

Although the experimental design is based on twelve conditions, subjects sometimes take part in thirteen conversations. The first conversation of the first entry is regarded as a "conditioning" conversation, and the circuit parameters are set at arbitraty fixed values. The results from the "conditioning" conversation are not taken into account in the analysis of the experiment.

4. *References to Recommendations and other CCITT publications where use of each method [methods in a)-d) under 2. above] is mentioned*

a) Many Recommendations include requirements based on reference equivalents of which References 28-33 are examples. See also Reference 42.

b) Recommendation G.112 (see Reference 31) requires certain articulation values to be satisfied but the method is now mainly used for diagnostic purposes. See Reference 34.

c) Study of various Questions, for example see References 38 and 41.

d) Study of various Questions, for example see References 38 and 41.

5. *General comments on subjective methods used in the laboratory*

More detailed information on the conduct of subjective tests and the interpretation of their results are given in References 26 and 36. A rather broad survey of the relationship between various methods is given in Annex 1 to Reference 39.

When used to provide information to assist in transmission planning of telephone networks, subjective methods should be employed with the following considerations in mind:

a) A clear description must be available of the type of telephone connections to which the results are to be applied. This is provided by formulating appropriate hypothetical reference connections (HRCs) (see Reference 29).

b) The levels, transmission losses, sending and receiving reference equivalents, etc., of the HRCs must guide the establishment of laboratory arrangements and the conduct of the tests. Speech spectra and levels must be properly chosen to correspond to those at the various points in the HRC.

c) Subjects must be drawn from an appropriate population.

d) Subjects must be treated within the experiments so that the results obtained are valid for the desired applications.

e) Suitable experimental designs must be used so that the results can be properly analyzed and confidence intervals estimated.

f) Even with proper precautions under *c), d)* and *e)* reliance should not be placed on absolute values of scores unless "control" conditions (e.g. a set of reference conditions) is included within the experiment.

g) A set of reference conditions will enable results to be expressed as ratings in terms of equivalent settings of some reference device [e.g. attenuator noise source, modulated noise reference unit (see Reference 43) etc.]. This enables much more reliable comparisons to be made with information from other sources.

6. *Objective methods*

Clearly the ultimate aim must be to be able to assess telephony transmission performance in terms purely of the objective characteristics of the telephone connections concerned. This aim is partly satisfied by use of tabulated information based on previous laboratory and other tests and an example of such usage appears in Annex 4 to Reference 43. Considerable progress has now been made towards the prediction of assessment scores, speech levels, etc. by use of subjective modelling as described in Reference 27 and the UKPO is now updating its tabulated information using such modelling. This enables many other important features like sidetone and attenuation/frequency distortion to be treated in a much more general manner.

The results obtained so far have shown that the effects of certain factors were underestimated according to the older tabulated information. One example is the effect of higher levels of circuit noise (> −60 dBmp at 0 dB RRE) when the overall transmission loss is low (NORE from 6-18 dB). This particular discrepancy can be ascribed to insufficient attention in earlier work to *d)* in 5. above.

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[32] CCITT Recommendation G.120 "Transmission characteristics of national networks", *Orange Book*, Volume III, 1977.

[33] CCITT Recommendation G.121 "Reference equivalents of national systems", *Orange Book*, Volume III, 1977.

[34] CCITT Recommendation P.45 "Measurement of the A.E.N. value of a commercial telephone system (sending and receiving) by comparison with the SRAEN" *Orange Book*, Volume V (an abbreviated version of the 1962 text), 1977.

[35] CCITT Recommendation P.72 "Measurement of reference equivalents and relative equivalents", *Orange Book*, Volume V, 1977.

[36] CCITT Recommendation P.74 "Methods for subjective determination of transmission quality", *Orange Book*, Volume V, 1977.

The following Questions under study are relevant and their texts are to be found in this Volume and together with their Annexes in Contribution COM XII-No. 1 for the study period 1977-1980.

[37] Question 2/XII, Assessment of service transmission quality.

[38] Question 4/XII, Effect of circuit noise on transmission performance.

[39] Question 7/XII, Models for predicting transmission qualities from objective measurements.

[40] Question 9/XII, Sidetone.

[41] Question 14/XII, Effect of attenuation distortion on transmission performance.

[42] Question 15/XII, Measurement of loudness ratings.

[43] Question 18/XII, Transmission performance of digital systems.

ANNEX 1

Text of recording replayed to subjects immediately prior to participation in a conversation test

Tests over simulated telephone connections

A series of tests is being carried out to study the performance of telephone connections, for which your assistance is requested. The tests involve conversing with another person, who has also agreed to take part, over a telephone connection which includes certain test equipment. You will be given a task which, it is hoped, will result in vigorous conversation devoted to argument and negotiation. The arrangement will be as follows. In the cabinet from which you make each test call, you will be provided with a set of six cards, showing reproductions of paintings. You will be asked to imagine that a choice is to be made from these for hanging, say, in the Refreshment Club at Martlesham, and during a period of about two minutes for each test call you should arrange all six cards in your own order of preference, and write the six identifying numbers, in this order, on the form provided. Your partner will also have an identical set of pictures, and his order will almost certainly differ somewhat from yours. The purpose of the conversation is to *negotiate* with your partner to arrive at an agreed compromise order of the pictures that satisfies both of you. It is important that the agreed list should again include all six pictures; you should then enter this list on the form. To make the situation realistic, one person will be nominated as the caller and will dial the number given on the form. At the conclusion of the conversation you will replace the handset and enter your opinion of the connection on the form. You will then be called individually by the operator to answer a few additional questions about the connection.

Subsequent conversations will be similar, but using different sets of pictures.

The complete test will require four visits, during each of which you will make three calls.

ANNEX 2

Opinion form

1. Before you commence your call, would you please insert below your order of preference for the picture cards. For identification, please use the serial numbers on the backs.

	1st	2nd	3rd	4th	5th	6th
Your order of preference						

2. The green light indicates that it is your turn to originate the call.

If the green light is on, please call your companion as soon as you are ready by dialling . . .

If the green light is off, it is your turn to be called.

3. Please insert below the order of preference arrived at after discussion with your companion.

	1st	2nd	3rd	4th	5th	6th
Agreed order of preference						

4. When you have completed the conversation, please replace the handset.

5. Please mark by a cross, your opinion of the telephone call you have just had.

N.B. – Please do not discuss your opinion with your companion.

Excellent	Good	Fair	Poor	Bad

6. You will then be called by the operator who will ask you to complete the following:

Did you, or the person who spoke to you, have any difficulty in talking or hearing over the connection?

YES	
NO	

If YES, please explain the nature of your difficulty to the operator.

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PART III

QUESTIONS ON TELEPHONE TRANSMISSION QUALITY AND LOCAL NETWORKS ENTRUSTED TO STUDY GROUP XII FOR THE PERIOD 1977-1980

(For the annexes to these Questions, reference should be made to Contribution No. 1 of the period 1977-1980 of Study Group XII)

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QUESTIONS ON TELEPHONE TRANSMISSION QUALITY AND LOCAL NETWORKS ENTRUSTED TO STUDY GROUP XII FOR THE PERIOD 1977-1980 [1)]

Important note

Notes to the effect that a particular Question is of interest to various Study Groups when no joint study group has been set up to study it are intended primarily for the information of the members of the Study Group dealing with the Question, to enable them to arrange for the necessary coordination within their national Administrations, in accordance with the decision taken by the Plenary Assembly.

List of Questions

Number of the Question	Short title	Remarks
1/XII	Reference equivalents of national systems in the international transmission plan	
2/XII	Assessment of service transmission quality	Coordination with Questions 4/II and 15/II
3/XII	Loudness ratings of operators' telephone systems and headsets	
4/XII	Effect of circuit noise on transmission performance	
5/XII	Noise clauses for telephone	Coordination with Questions 4/CMBD and 8/XVI
6/XII	Subscribers' tolerance of echo and propagation time	Coordination with e) of Question 10/XV
7/XII	Models for predicting transmission qualities from objective measurements	
8/XII	Measuring the efficiency of a microphone or a receiver	
9/XII	Sidetone	
10/XII	Increase in the sensitivity of local systems	Documentary Question; coordination with a) of Question 5/XVI
11/XII	Limits of intelligible crosstalk	Coordination of 5), 6) and 7) with Question 3/XVI; see also Question 6/XV
12/XII	Artificial voices, mouths and ears	
13/XII	Nonlinear distortion of telephone apparatus	

[1)] There is a strong interest of Study Group XVI in Study Group XII's work; Study Group XII is therefore requested to keep Study Group XVI continuously informed of the progress made.

Number of the Question	Short title	Remarks
14/XII	Effect of attenuation distortion on transmission performance	Coordination with ii) of Question 1/XVI
15/XII	Measurement of loudness ratings	
16/XII	Return loss variations in subscriber lines and telephone sets	Documentary Question
17/XII	Loudspeaker telephones	
18/XII	Transmission performance of digital systems	Coordination with Question 10/XVI; see also Questions 9/XVIII and 10/XVIII
19/XII	Recommended values of loudness ratings	New Question; coordination with Question 11/XVI
20/XII	Devices for protection against acoustic shocks	New Question; coordination with Question 5/V
21/XII	Eficiency of telephone kiosks and booths	New Question

Question 1/XII – Reference equivalents of national systems in the international transmission plan

(continuation of Question 1/XII studied in 1973-1976)

Provisionally-agreed values for reference equivalents are recommended in Recommendations G.111 and G.121.

Can these recommendations be confirmed? If not, what values could be?

Note. – Annex 4 in Volume V of the *Green Book* gives instructions concerning how contributions to the Question should be made (see also Supplement No. 2 to Volume V of the *Orange Book*).

Question 2/XII – Assessment of service transmission quality

(continuation of Question 2/XII studied in 1973-1976)

(coordination with Questions 4/II and 15/II)

Considering

a) that with the advent of worldwide automatic and semi-automatic networks operating personnel will be less able to detect the onset of unsatisfactory service conditions;

b) that connections in such worldwide networks will be more complex and include more elements as potential sources of transmission difficulty, and

c) that customers will expect higher quality of service as their use of the worldwide networks increases,

the following questions shall be studied:

1. What methods are suitable for the evaluation of service from the standpoint of speech transmission quality?

2. Is it desirable to standardize methods to be used as part of overall appraisals of worldwide subscriber-to-subscriber connections?

3. Can the data arising from these methods be used to recommend preferred magnitudes or limits of speech transmission quality criteria, such as mean opinion score and percentage of users experiencing difficulty, for different classes of worldwide subscriber-to-subscriber connections?

Note 1. – General approaches which might be considered for this purpose include service observations by third party observers and subscriber interrogation by interview or questionnaire. Annex 1 to Question 2/XII, *White Book*, Volume V, describes this kind of method and Annex 2 to Question 2/XII, *White Book*, Volume V, gives an example of test programmes. See also the annexes to this Question.

Note 2. – It has been noted by certain Administrations that some measures of service quality vary during the course of a conversation. For example, it has been observed that repetition events are more frequent at the beginning of a conversation (see Annexes 3 and 4 to Question 2/XII, *White Book*, Volume V). This factor should be considered in studying methods of evaluating transmission quality.

Note 3. – Attention is drawn to the importance of using suitable methods for preparing test programmes and for the analysis of results.

Note 4. – Study Groups II and IV study questions relating to service quality by means of service observations and questionnaires which include elements on speech transmission quality which are also relevant to the study of Question 2/XII. Coordination is particularly desirable between these Study Groups on transmission data arising from the use of Recommendations E.422, E.423, E.424 and E.425 and study of Questions 8/II, 15/II and 23/IV.

Note 5. – Attention is drawn to Recommendation P.77 which outlines one method of assessing speech transmission quality by use of the questionnaires of Recommendation E.425.

Question 3/XII – Loudness ratings of operators' telephone systems and headsets

(continuation of Question 3/XII studied in 1973-1976)

What methods of measurement should be recommended for the determination of loudness ratings and corresponding sensitivity/frequency charactertistics of operators' headsets for sending and receiving?

Note 1. – Due consideration should also be given to the distribution of the speech power which operators' sets may produce.

Note 2. – Maximum and minimum values of loudness ratings of operators' telephone systems cannot be recommended until a method of determining these quantities has been agreed. The Annex to this Question in the *Green Book*, Volume V, indicates the criteria guiding the choice made by the United Kingdom Post Office concerning the sensitivity of operators' sets.

Note 3. – When the study has reached a suitable stage it will be possible to make precise recommendations (in the context of Question 1/XII) for values of sending and receiving loudness ratings (or reference equivalents) to be associated with Recommendation G.111, E.

Note 4. – Question 12/XII is concerned with artificial mouths and ears suitable for measuring operators' headsets (see Note 3 to Question 12/XII).

Note 5. – Question 15/XII is concerned with measurement of loudness ratings for local telephone systems and should be read in conjunction with Question 3/XII. The text of the latter draws freely on the principles and text established under Question 15/XII.

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

(continuation of Question 4/XII, studied in 1973-1976)

1. What is the family of subjective opinion curves which characterizes the effect of white circuit noise as a function of the indications given by a psophometer standardized by the CCITT and for different values of overall reference equivalent?

2. What is the relative effect of other types of noise such as hum, impulses, and single frequency tones?

3. How do factors such as frequency response of the connection, telephone instrument sidetone, and room noise influence the effect of circuit noise?

Note 1. – Annex 1 provides an outline of suggestions and desired information which will facilitate comparison of results of various Administrations.

Note 2. – Supplement No. 2 to this Volume outlines a method for conducting onversational tests as used by the United Kingdom Post Office.

Note 3. – Annex 2 summarizes results submitted by several Administrations along the lines outlined in Annex 1.

Note 4. – Annex 1 to Question 7/XII describes a transmission rating model for reference equivalent and circuit noise submitted by AT&T.

Note 5. – Annex 3 provides a comparison between the transmission rating model described in Annex 1 to Question 7/XII and the test results summarized in Annex 2.

Note 6. – Annex 4 provides a preliminary model for the effect of white circuit noise and overall reference equivalent based on four laboratory conversational tests.

Note 7. – Annex 5 is a proposal by the United Kingdom Post Office on the factors which need to be taken into account in devising a limit for single frequency interference.

Question 5/XII (8/XVI) – Noise clauses for telephony

(continuation of Question 5/XII, studied in 1973-1976)

(coordination with Question 4/CMBD and 8/XVI)

If the hourly-mean clause in CCITT Recommendation G.222 (CCIR Recommendation 393-2) were deleted:

1. would sufficient control over the noise of the systems concerned in the recommendations to protect against high noise on long or consecutive calls be exercised by the remaining noise clauses, or

2. would the distribution of 1-minute means for any month need to be more completely specified, or

3. is some other form of additional clause preferable, and if so what?

Note. – Annex 1 supports the view that the existing 1-minute mean clauses and 5-ms mean clause are sufficient by themselves, whereas Annex 2 suggests that additional 1-minute mean clauses are necessary.

Annex 3 is the version of CCIR Recommendation 393-2, as approved by the CCIR Plenary Assembly in 1974. Finally, Annex 4 reproduces the reply prepared in the study period 1973-1976.

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

(continuation of Question 6/XII, studied in 1973-1976)

[coordination with e) of Question 10/XV]

1. What criteria and techniques can be recommended for the prediction of subjective performance of echo control devices?

Note. – These tests should be performed under controlled laboratory conditions. This part of the Question is related to e) of Question 10/XV.

2. Can the method of calculating (and measuring) loss in an echo path, provisionally recommended in Recommendation G.122, B. b), Volume III be endorsed as providing an indication of loss which sufficiently corresponds to the subjective effect produced by speech echo in that path?

Note 1. – Annex to Recommendation G.122 contains explanations on path a-t-b. For further information related to this part of the Question, see also Parts II and III of Annex 4 to Question 6/XII in Volume V of the *Green Book.*

3. Does Recommendation G.161, Volume III sufficiently control the quality of telephone transmission on connections fitted with echo-suppressors complying with the Recommendation?

4. What is the effect on transmission performance of the following factors in telephone connections having mean one-way propagation times (MOPTs) of 150 ms and upwards?

i) 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;

ii) interaction between end-delay and return loss. This should be established by cooperation between Study Groups XII and XV for a range of values inluding 36-ms round trip end-delay;

iii) the effects of asymmetry at the echo suppressors of speech levels in the two directions of transmission;

iv) higher return losses that might be achieved by special techniques;

v) echo control devices of different types at the two ends of the international circuit. Both types should, of course, comply with the specification prepared by Study Group XV.

Note 2. – Annex 1 gives information on some points to be studied within the framework of 4. above.

5. What characteristics of echo-cancelling devices need to be specified to ensure satisfactory telephone transmission quality on connections to which they are fitted at one or both ends?

i) for connections with MOPTs less than 400 ms;

ii) for connections with MOPTs greater than 400 ms.

Note 3. – Annex 5 to Question 6/XII in Volume V of the *Green Book* contains contributions from two Administrations to this part of the Question.

6. How should the implementation of echo suppression, when integrated into call-concentrating systems, be taken into account when considering the performance requirements of such call-concentrating systems and of the echo suppressors incorporated in such systems?

Note 4. – Annex 2 is the reply of Study Group XII made at the end of the 1973-1976 study period.

Question 7/XII – Models for predicting transmission qualities from objective measurements

(continuation of Question 7/XII, studied in 1973-1976)

Considering that

a) there is evidence of increasing use being made of opinion scores (or percentages of subjects experiencing difficulty) while engaged in conversation, as a criterion for expressing the performance of complete telephone connections. (See replies to Questions 1/XII, 2/XII, 4/XII, 6/XII, 9/XII, 10/XII, 14/XII and 18/XII);

b) such a criterion can be used to provide information for planning and other purposes by:

b.1 conducting suitable field observations or laboratory tests on individual real or simulated telephone connections;

b.2 making use of some model for smoothing, averaging and combining results obtained by application of b.1 to prepare tabulated or graphical information suitable for use in planning. Such a model may include arrangements for combining the effects of different factors, such as transmission loss, noise, distortions, echo;

b.3 making use of a theoretical model based on fundamental considerations in order to relate the desired form of opinion score or "percentage difficult" to objective data descriptive of the transmission characteristics of each telephone connection to be assessed. Such a model will require frequency-dependent quantities as inputs,

the following questions should be answered:

1. What forms of model are being used for the purpose indicated in b.2 above and what are their respective advantages and restrictions?

2. What ought to be the structure of a model according to b.3 above to ensure that the following factors [i) to v)] at least, are taken properly into account?

i) transmission loss,

ii) circuit noise,

iii) room noise,

iv) attenuation/frequency distortion,

v) sidetone.

It would be desirable also to include the effects of:

vi) nonlinear distortions,

vii) echo.

Although desirable to do so, it scarcely seems possible at present to include the effects of:

viii) propagation time,

ix) delay distortion,

x) mutilation by voice-switching.

3. What values should be assigned to the parameters of a model according to b.3 above so that recommendations can be obtained from these models? The model must provide adequate agreement between predicted values of opinion scores and "percentage difficult" and those obtained from actual observations.

Question 8/XII – Measuring the efficiency of a microphone or a receiver

(continuation of Question 8/XII, studied in 1973-1976)

The shape of the sensitivity/frequency characteristic of a sending system depends very much on the artificial mouth used and also upon the method of measurement employed.

Similarly, the shape of the sensitivity/frequency characteristic of a receiving system depends upon the artificial ear used and also upon the method of measurement employed.

What methods of measurement should be recommended for tracing these curves and what accuracy should be recommended for these measurements?

Note. – In studying this question, account should be taken of Recommendation P.75.

The attention of Administrations is also drawn to the following documentation:

1. For the 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. 75 (Helsinki Telephone Co.)
2. For the period 1968-1972, COM XII-No. 20 (SIP and STET, Italy).
3. IEEE Standard 269-1971: "IEEE standard method for measuring transmission performance of telephone sets", IEEE, Stockholm.
4. Annexes 27 to 31 (*Red Book*, Volume V, Part II).
5. Annex to Question 8/XII (*White Book*, Volume V).
6. Annexes 1 and 2 to Question 12/XII (*Green Book*, Volume V).

Question 9/XII – Sidetone

(continuation of Question 9/XII, studied in 1973-1976)

Considering that

a) reference equivalents determined in accordance with Recommendation P.73 and used hitherto to rate sidetone levels of subscribers' telephone sets are rather unsatisfactory because the frequencies most contributing to loudness (the low frequencies) are those which are most subject to masking by the talker's own speech via bone conduction and other paths;

b) the new loudness rating proposals, if applied to sidetone paths may also be subject to the same objections, but in addition could introduce other problems associated with the threshold of detection of sidetone while speech is being uttered, and the response of the IRS used to determine the loudness rating;

c) Recommendation P.73 takes no account of the mechanical and acoustical transmission of sound along the handset to the receiver, *the following questions should be studied:*

1. Is it appropriate to rate the equivalent loss of the sidetone path of a subscriber's set in terms of:

i) some form of loudness rating? (see Question 15/XII);

ii) preferred and limiting frequency characteristics? or

iii) some other means?

Whichever method is used, what numerical values should be recommended to replace those contained in Recommendation G.121?

2. In view of the effects of listener sidetone in which local room noise heard in the telephone receiver via the sidetone path may mask speech signals received from the line, and which may demand a different set of rating criteria and calculation procedures from those applying when a subscriber speaks, what recommendations, additional to those in reply to 1. above should be followed to ensure that listener sidetone does not adversely affect listener opinion?

Note. – Annexes 1, 2 and 3 give information which are useful for the study of the Question. See also Annex 4 on pages 106 to 119 of Contribution COM XII-No. 134 (study period 1973-1976).

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

(continuation of Question 10/XII, studied in 1973-1976)

[concerns also a) of Question 5/XVI]

(documentary Question)

Considering

that modern developments have enabled considerable improvements to be made in the sensitivity of telephone sets and that even further increases in sensitivity can readily be achieved, it is desirable to examine the consequences of such increases in sensitivity and the manner in which they may be turned to advantage.

In extracting advantages from these possibilities care is necessary to ensure that certain disadvantages do not follow (such as excessive levels on circuits, excessive loudness of received speech, excessive sidetone, excessive echo, excessive crosstalk, etc.).

Clearly, compromises are necessary to balance the advantages and disadvantages.

What transmission characteristics are desirable for subscribers' sets so that the best compromise will be obtained under conditions likely to apply in future telephone networks?

Note 1. – The fundamental characteristics to be considered in the specification of such subscribers' sets include the following, which must be expressed as functions of line current:

a) sensitivity/frequency characteristic, sending;

b) sensitivity/frequency characteristic, receiving;

c) line terminal impedance;

d) sidetone balance impedance.

The other transmission characteristics of importance can be derived from these.

Note 2. – The above-mentioned characteristics relate to conventional 2-wire handset telephone sets. Additional characteristics would need to be considered for other types of subscribers' telephone equipment, e.g. loudspeaker telephones (Question 17/XII).

Note 3. – The Annex contains some considerations relating to characteristics for subscriber sets.

Question 11/XII – Limits of intelligible crosstalk

(continuation of Question 11/XII, studied in 1973-1976)

[coordination of 5., 6. and 7. with Question 3/XVI; see also Question 6/XV]

What should be the planning rules governing the use of the noise/crosstalk threshold curves given in Recommendation G.116? The study should include different types from local to international communications. The following points need special clarifications:

1. Which threshold criterion should be used, audibility or intelligibility?

2. What levels of room noise and line noise should be assumed?

3. What mean speech level and standard deviation should be used?

4. What standard deviation of the listeners' threshold about the median value should be used?

5. What transmission plan arrangements should be used (e.g. distribution or reference equivalents, circuit losses and crosstalk attenuations)?

6. What are the probability limits appropriate to various classes of connections?

7. What allowances should be made for the joint probability of calls being made simultaneously and the incidence of mutual silence between the two correspondents on the disturbed connections?

8. How should the effects due to the various conditions be combined to arrive at the traffic-weighted probability of intelligible crosstalk for a given type of local system or international circuit?

Note 1. – The work of Study Group XVI, in the framework of Question 3/XVII, will have a bearing on 5., 6. and 7. of this Question. See also Question 6/XV.

Note 2. – The Annex is the reply made by Study Group XII at the end of the study period 1973-1976.

Question 12/XII – Artificial voices, mouths and ears

(continuation of Question 12/XII studied in 1973-1976)

1. How should Recommendation P.51 (A. and B.) concerning the artificial ear provisionally advocated by the CCITT be amplified?

2. What general characteristics should be fixed for artificial voices and mouths?

3. Pending a reply to 2. above, how should C. of Recommendation P.51 be amplified?

Note 1. – The Annex to the Question outlines the status of the Question at the end of the 1973-1976 study period (see also Annex 2 in the *Green Book*, Volume V).

Note 2. – Details on the use of the artificial ear and information for the study of 1. of this Question are given in Annex 1 to Question 12/XII in the *Green Book*, Volume V. Contribution COM XII-No. 53 (1964-1968) gives the results of tests on the effect of acoustic leaks.

Note 3. – A considerable amount of documentation concerning artificial mouths and ears is to be found in Annexes 8 to 16 in Volume V of the *Red Book* (pages 241-415), in Annex G in Volume V *bis* of the *Red Book* (pages 119-131) and in Annexes 1 to 5 to former Question 12/XII (*Red Book*, Volume V *bis*, pages 202-244).

Note 4. – In studying this Question, attention should be paid not only to artificial mouths and ears used for measuring subscriber sets but also those used for measuring operators' microtelephone sets (which may have internal headphones). See Question 3/XII.

Question 13/XII – Nonlinear distortion of telephone apparatus

(continuation of Question 13/XII, studied in 1973-1976)

Nonlinear distortion in subscribers' telephone apparatus is at present produced mainly by carbon microphones. The introduction of linear microphones will probably reduce the amount of distortion in telephone connections, but the nonlinear distortion in amplifiers contained in future standard telephones or loudspeaker telephones may still be high enough to affect the speech transmission quality as perceived by the subscriber.

In order to arrive at a threshold value for the perceptible distortion, measured in a suitable manner, the following items should be studied:

1. the effects which the nonlinear distortion of a subscriber's telephone apparatus has on the quality of telephone transmission;
2. the methods of measuring the nonlinear distortion of a subscriber's telephone apparatus;
3. the effects of carbon microphone noise in a subscriber's telephone apparatus on the quality of telephone transmission;
4. the difference between carbon microphones, and linear microphones, caused by the difference in nonlinear distortion, as concerns:
   i) speech transmission quality;
   ii) speech voltage at a given talking level.

The study of 1., 2. and 3. above should lead to a revision of Recommendation P.62, B. Administrations are invited to submit Contributions.

Note. – The Annex represents the reply made by Study Group XII at the end of the 1973-1976 study period. For further information, see also the Annex to this Question in Volume V of the *Green Book*.

Question 14/XII – Effect of attenuation distortion on transmission performance

(continuation of Question 14/XII, studied in 1973-1976)

[coordination with ii) of Question 1/XVI]

Attenuation/frequency distortion likely to be encountered in practical telephone networks could, under certain conditions, introduce a significant effect on transmission performance in connection with other types of impairment factors, such as loudness loss, circuit noise and quantizing distortion. Whereas networks make gradual progress towards better transmission quality, following some development stages.

1. What will be the characteristics of the effect of attenuation distortion upon telephone transmission performance as compared with other impairment factors, under various conditions of practical networks in the present and future?

2. What is the interaction between the sensitivity/frequency characteristics of a local telephone system, i.e. telephone set, subscriber's line and feeding bridge, and the attenuation/frequency distortion in a chain of circuits connecting two local exchanges?

3. Can the significance of the effect of attenuation distortion on telephone transmission performance be confirmed in view of the results of the studies under 1. and 2.?

4. If so, what Recommendations will be necessary or useful in characterizing the effect of attenuation distortion, for transmission planning purpose and how should Recommendation G.113, A. be improved? Can the basic effect of phase distortion on transmission performance be confirmed? (See Recommendation G.115.)

Note 1. – When studying 1. and 2. above, the existence of 3-kHz spaced channels and the effect of extending lower frequency band should be taken into account.

Note 2. – The subject under 2. above could be studied by determining the effect of a given attenuation distortion characteristic when the sensitivity/frequency characteristics of the local telephone systems are changed, e.g. by changing the subscriber lines.

Note 3. – A useful and simple way of quantitatively expressing attenuation distortion shapes must be defined for studying the Question, also bearing in mind the variation of the contribution of distortion according to frequency.

Note 4. – This Question should be studied in close coordination with studies of Questions 1/XII, 4/XII, 18/XII, 1/XVI, iv) and, if needed, others.

Note 5. – The Annex represents the reply made by Study Group XII for the study period 1973-1976.

Question 15/XII – Measurement of loudness ratings

(continuation of Question 15/XII, studied in 1973/1976)

Considering

a) that rather elaborate facilities, including the use of a highly-trained team of operators, are required to determine reference equivalents according to Recommendation P.72;

b) that many Administrations use other methods (subjective tests, calculations and objective measurements) to determine loudness ratings and that it is very difficult to establish a relationship between the different results;

c) that Study Group XII has been studying the problems and difficulties of determining reference equivalents and loudness ratings for a very long time and that various suggestions have been put forward to improve methods;

d) that it is desirable that the method ultimately chosen as the result of the study of this Question should yield measurements as precise as possible and readily reproduced in different laboratories. To this end, attention should be given to defining practical procedures that will ensure the greatest degree of precision. In the case of subjective measurements, for example, the best procedures for ensuring the stability of the testing crew should be studied. Similarly, in any method, the stability of commercial telephone sets, especially those with carbon microphones, will affect the precision obtained and attention should be given to the method of preconditioning the handsets before use (see Recommendation P.75) and to the fact that a sufficiently large number of sets should be tested so as to ensure reasonably narrow confidence limits. For the same reason, attention is drawn to the importance of using appropriate methods for planning the experiments and for analyzing the results obtained;

e) that the following new Recommendations have been made by the VIth Plenary Assembly:

Recommendation P.48	"Specification for an intermediate reference system."
Recommendation P.64	"Determination of sensitivity/frequency characteristics of local telephone systems to permit calculation of their loudness ratings."
Recommendation P.76	"Determination of loudness ratings: fundamental principles."

f) that Annex 2 contains partial draft texts for the following three proposed Recommendations which are intended to complete the description of how loudness ratings in accordance with Recommendation P.76 can be determined:

Draft Recommendation P.XXC	"Subjective testing method for determination of loudness ratings in accordance with Recommendation P.76."
Draft Recommendation P.XXE	"Calculation of loudness ratings."
Draft Recommendation P.XXF	"Objective instrumentation for determination of loudness ratings."

g) that Annex 1 gives a report of the study of this Question during the last study period.

How can the partial draft texts given in Annex 2 to this Question be amended and completed so that Recommendation P.76 and the associated Recommendations can be used for replacing reference equivalents in Recommendations by loudness ratings? (see Note 1).

Note 1. – A new Question has been set entitled "Recommended values of loudness ratings". This Question is to be studied jointly by Study Group XII and Study Group XVI.

Note 2. – The study of this Question should be undertaken as follows:

i) Measurement of as many different local telephone systems as possible according to Recommendation P.76 and the associated Recommendations; these measurements should be conducted in the CCITT Laboratory and other laboratories.

ii) The results of these measurements should be analysed and conclusions drawn.

iii) From the conclusions, proposals should be made for amending and completing the partial draft texts given in Annex 2.

Question 16/XII – Return loss variations in subscriber lines and telephone sets

(continuation of Question 16/XII, studied in 1973-1976)

(documentary Question)

1. What is the range of variation of the return loss measure of subscriber lines, telephone sets and supply bridges 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?

2. What method can be used in the design of telephone sets, of subscribers' lines, supply bridges and other local exchange and PBX equipment to reduce this range of variations?

Note 1. – In addition to the return loss/frequency characteristic of local telephone circuits (i.e. subscriber's instrument, local line and feeding bridge) versus the appropriate balance network, it would be useful if the return loss from the points of view of echo and of stability could also be presented for each of a number of significant configurations (e.g. maximum, median, and zero-length line).

A definition of echo return loss may be based upon the provisional definition of the transmission loss of the path a-t-b from the point of view of echo which is to be found in Recommendation G.122, B. b).

The definition of stability return loss is given in Recommendation G.122, A. a) and extends over the whole range 0-4 kHz. In practice the somewhat more restricted band 200-3500 Hz is adequate. The fact that the minimum value of the return loss at the local exchange may not coincide with minimum value of transmission losses in the 4-wire portion of the national system provides some planning margin, and in any case cannot be relied upon.

Note 2. – Opportunity should also be taken to measure the various return losses presented by PBX exchange lines when PBX extensions are in the speaking condition.

Note 3. – The information collected together during 1968-1972 and 1973-1976 has been incorporated in the CCITT Handbook, *Transmission Planning of Switched Telephone Networks.*

Question 17/XII – Loudspeaker telephones

(continuation of Question 17/XII, studied in 1973-1976)

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 satsify from the point of view of transmission performance. Annex 2 is a report of studies carried out in 1968-1972.

Note 2. – Further information can be found in annexes to earlier versions of Question 17/XII as follows:

Federal Republic of Germany:	*Red Book*, Volume V, pp. 705-706
Swedish Administration:	*Red Book*, Volume V, pp. 706-708
Polish Administration:	*Red Book*, Volume V *bis*, pp. 284-286
United Kingdom Post Office:	*White Book*, Volume V, Question 17/XII pp. 2-3
American Telephone & Telegraph Co.:	*White Book*, Volume V, Question 17/XII pp. 4-5

Note 3. – The study of this Question should also take note of loudspeaking telephones in connection with videophone systems, being studied under Question 4/XV.

Question 18/XII – Transmission performance of digital systems

(continuation of Question 18/XII, studied in 1973-1976)

(coordination with Question 10/XVI; see also Questions 9/XVIII and 10/XVIII)

1. What recommendation might be made by the CCITT concerning the standard of transmission performance assessment that ought to be achieved for a single audio-link provided by a practical engineered digital system bearing in mind the conditions under which such a link may form part of an international connection?

2. 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 digital system, e.g. peak clipping and centre clipping, should be assessed in the same units as used for 1. above.

3. Since any economically engineered digital system will be liable to other forms of degradation due, for example, to transmission errors, quantization inaccuracies, synchronization difficulties, jitter, etc., how should such systems be measured in ordinary working conditions, to ensure that they comply with the requirements set forth under 1. above?

Note. – Annex 1 provides detailed guidelines for continuing study of the Question. Annex 2 describes the Modulated-Noise Reference Unit (MNRU) which some Administrations have found to be very useful in the study of the Question. Annex 3 provides information on the relation between equivalent speech-to-idle-circuit-noise ratio and speech-to-speech-correlated-noise ratio obtained using the MNRU to provide speech-correlated noise. Annex 4 provides further information on speech-correlated noise and describes a method for estimating the transmission performance of telephone connections containing digital systems. Quantization distortion and its effect on telephone transmission performance is the subject of Annex 6 to Question 18/XII, Volume V, *Green Book*, pages 282-297.

Question 19/XII – Recommended values of loudness ratings

(new Question to be studied jointly by Study Groups XII and XVI, Study Group XII coordinating)

In Recommendations P.48, P.64 and P.76 a new method of characterizing telephone instruments and local lines in terms of "loudness rating decibels" is introduced with the intention that this measure will eventually replace reference equivalent decibels.

The desire to depart from use of reference equivalents as at present defined by the CCITT (Recommendation P.72) arises for the following reasons:

- reference equivalents cannot be added algebraically; discrepancies of at least 3 dB may be found;
- replication accuracy of reference equivalents is not good; changes in crew can give rise to a spread of values for one item extending over a range as large as 5 dB;
- increments of real (distortionless) transmission loss are not reflected by equal increments of reference equivalent. As an example, statistical analysis has shown that a 10 dB increase in loss results in an increase in reference equivalent of only about 8 dB.

The use of loudness ratings defined broadly in accordance with the principles being studied would largely obviate these troubles.

In the expectation that for many planning purposes a simple relationship can be used to derive sending and receiving loudness ratings from the corresponding traditional reference equivalents, an appropriate relationship (see Note 3) would be:

$$\text{Loudness rating} = \text{M (Reference equivalent)} - \text{K},$$

in which M is of the order of 5/4 and K is constant, the value of which is to be determined taking account of:

- the implications on planning procedures of such a change of units (see Annex 1);
- measurement results of the loudness ratings of subscriber local systems.

How shall values currently recommended in terms of reference equivalent be re-expressed in terms of loudness ratings?

Note 1. – Conversion rules are required for sending, receiving, overall, sidetone, echo, and crosstalk reference equivalents. These relationships may not all be linear ones.

Note 2. – Annex 1 to this Question gives further explanations and a discussion of some of the factors to be considered.

Note 3. – Annex 2 gives a more precise idea of the form of the relationship.

Question 20/XII – Devices for protection against acoustic shocks

(new Question; coordination with Question 5/V)

1. To ensure greater efficiency of the devices for protection against acoustic shocks, which constitute the topic of Recommendation K.7, how should this Recommendation be clarified, especially in 4. of the Recommendation? More precise information would be required on the following points:

i) the physical definition of the pressure amplitude, or any other parameter associated with the pressure produced, in order to evaluate the hazards;

ii) measuring equipments and test conditions.

Note. – In order to undertake this study, information is required on the amplitude-time characteristics of the surge voltages forecast or recommended for testing, which are dealt with in Question 5/V.

2. Is the table under 5. of Recommendation K.7 necessary and sufficient to ensure the desired result, or can it be modified and/or supplemented in order to take into account:

i) the increased sensitivity of modern telephone receivers?

ii) the possibility of making protective devices with various semi-conducting materials (especially silicon)?

3. Should this Recommendation be supplemented by information on nonlinear distortion, the conditions which should govern its measurement and its maximum acceptable value? (See Question 13/XII.)

Question 21/XII – Efficiency of telephone kiosks and booths

(new Question)

Considering

a) that international telephone calls may be made from telephone sets situated in noisy surroundings and protected by telephone kiosks or booths;

b) that there is no satisfactory method for assessing the improvement in transmission quality made by these telephone kiosks and booths;

c) that measuring methods based exclusively on acoustic attenuation do not appear to be satisfactory in evaluating their efficiency;

d) that there are no recommendations on criteria governing the use of these devices,

the following questions should be studied:

1. Which method should be standardized to assess the improvement in transmission quality attributable to the use of telephone kiosks or booths?

2. Which method should be standardized for measuring the efficiency of these devices?

3. Which efficiency standards should be recommended taking account of external noise levels being experienced and sidetone.

Note 1. – The Annex describes a method for assessing open telephone kiosks or booths which may be useful for these studies.

Note 2. – Replies to 3. of this Question may provide valuable data applicable to study of sidetone under Question 9/XII. Administrations providing contribution to this part of the study are asked to provide full details of the characteristics of sidetone paths included, in particular of sidetone sensitivity/frequency characteristics, reference equivalent, etc.

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