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TELEPHONE TRANSMISSION QUALITY

OBJECTIVE ELECTRO-ACOUSTICAL MEASUREMENTS

DETERMINATION OF SENSITIVITY/FREQUENCY CHARACTERISTICS OF LOCAL TELEPHONE SYSTEMS

ITU-T Recommendation P.64

(Previously “CCITT Recommendation”)

FOREWORD

The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the International Telecommunication Union. The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

ITU-T Recommendation P.64 was revised by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

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NOTES

1 As a consequence of a reform process within the International Telecommunication Union (ITU), the CCITT ceased to exist as of 28 February 1993. In its place, the ITU Telecommunication Standardization Sector (ITU-T) was created as of 1 March 1993. Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

In order not to delay publication of this Recommendation, no change has been made in the text to references containing the acronyms "CCITT, CCIR or IFRB" or their associated entities such as Plenary Assembly, Secretariat, etc. Future editions of this Recommendation will contain the proper terminology related to the new ITU structure.

2 In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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DETERMINATION OF SENSITIVITY/FREQUENCY CHARACTERISTICS OF LOCAL TELEPHONE SYSTEMS

(Geneva, 1976; amended at Malaga-Torremolinos, 1984; Melbourne, 1988; and Helsinki, 1993)

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

1 Introduction

The sending, receiving or sidetone sensitivity/frequency characteristic of a local telephone system (LTS) is usually measured directly.

NOTES

1 The sending, receiving or sidetone sensitivity/frequency characteristic can also be calculated provided the relevant information of the telephone line and feeding bridge is known. Some of the information required for sidetone is outside the scope of the existing Recommendations.

2 The same principles also apply to the measurement of microphones and earphones.

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 calculating loudness ratings, or estimating other subjectivity-determined quantities.

The present Recommendation is mainly concerned with c), but the principal is also applicable to a) and b). For these purposes, especially for c), 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 Recommendations P.51 and P.57).

This Recommendation applies mainly to LTSs with handset telephones. However, the principles also apply to other types of telephones. Specific considerations for headsets are described in Recommendation P.38, and for loudspeaker telephones in Recommendation P.34.

2 Sending sensitivities of the LTS

For the present purposes, the sending sensitivity of a local telephone system is specified in terms of the free-field sound pressure at a reference point in front of the mouth[1)], and the electrical output from the local telephone system or the microphone as the case may be. The input sound pressure cannot 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.

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

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 C.

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. Other methods have been also tried and none seems satisfactory so far.

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 nonlinearity 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.

3 Receiving sensitivities of the LTS

The IEC-318 model artificial ear (see Recommendation P.57) provides means for precise measurements of the receiving sensitivities of the LTS. 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 (L_E) 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 clause 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 and voice

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 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 but preferably 100 to 8000 Hz.

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

Recommendation P.51 defines the requirements for artificial mouths.

NOTE – However, the send loudness ratings calculated from the sending sensitivities measured when using an artificial mouth do not always agree well with the loudness ratings determined subjectively using real mouths. The subject is still under study.

In principle, the artificial voice defined in Recommendation P.50 should be used as the acoustic test signal. However, sine waves at defined frequencies have been used satisfactorily so far as stable sets are concerned. Some other signals with continuous spectra, for example pink noise and Gaussian noise having the same long-term spectrum as speech, can also be used as the acoustic test signal. Sine waves can also be used for the measurement of some types of carbon microphones if appropriate techniques are used (see 3.6.3 of the *Handbook on Telephonometry*, ITU, Geneva, 1993).

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 is defined as the pressure sensitivity of the measuring microphone. It should be constant within ± 0.5 dB over the frequency range 100-8000 Hz.

For a human ear, the ear reference point (ERP) is defined in Annex A. 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 clause 3 above).

6 Definition of sending sensitivity of an LTS

The sending sensitivity of an 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 C shall be used. Usually, the sending sensitivity is a function of frequency.

The sending sensitivity of a local telephone system at a specified frequency or in a narrow frequency band is expressed as follows:

$$S_{mJ} = 20 \log_{10} \frac{V_J}{P_m} \text{ dB rel 1 V/Pa}$$

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" handset of the test item.

Administrations who wish to use complex impedance termination for the measurement, are referred to the method given in Annex B. The use of complex impedance terminations may change the SLR and RLR values slightly, in comparison to the values using 600 ohms termination. Furthermore, the break points for sending and receiving local telephone systems could change. The effect will be under study.

6.1 Measurement of telephone sets containing carbon microphones

It is intended that the Recommendation should apply for measuring systems containing carbon microphones as well as those having noncarbon microphones. When measuring LTSs that contain linear items, it does not matter at which sound pressure the measurements are made as long as it is known and does not cause overloading. However, when carbon microphones are present, different sensitivities will be obtained depending upon the sound pressure and characteristics of the acoustic signal 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. At present, there is no single method that can be recommended for universal use. The problem is being under study. Until a suitable method can be defined, Administrations may take note of the various methods that have been suggested and are undergoing appraisal; they are indicated in 3.6.3 of *Handbook on Telephonometry*.

7 Definition of receiving sensitivity of an LTS

Usually, the receiving sensitivity is a function of frequency. The receiving sensitivity of a local telephone system at a specified frequency or in a narrow frequency band, as measured directly with an artificial ear complying with Recommendation P.57, is expressed as follows:

$$S_{Je} = 20 \log_{10} \frac{p_e}{½ E_J} \text{ dB rel 1 Pa/V}$$

where p_e is the sound pressure at the ERP and ½ E_J is half the emf in the 600 ohm source. See also clause 6 for complex impedance termination.

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 above in clause 3 and S_{JE} is the receiving sensitivity determined using a large number of real ears.

Further information on this topic is given in Recommendation P.79.

8 Definitions of talker and listener sidetone sensitivities of an LTS

The talker sidetone sensitivity of an LTS is a function of the sending and receiving sensitivities of the telephone set, but also depends on a number of factors including the local subscriber's line conditions, the effective terminating impedance at the local exchange and the sidetone balance circuit within the telephone set.

The sidetone sensitivity as measured from an artificial mouth to the telephone earphone is expressed as

$$S_{meST} = 20 \log_{10} \left(\frac{p_e}{p_m} \right) \text{ dB}$$

where p_m is defined in clause 6 and p_e is the sound pressure developed at the ERP with the handset in the loudness rating guard ring position (LRGP).

The listener sidetone sensitivity as measured in a diffuse room noise field is expressed as

$$S_{RNST} = 20 \log_{10} \left(\frac{p_e}{p_{RN}} \right) \text{ dB}$$

where p_e is the sound pressure developed at the ERP with the handset held at LRGP in front of an unenergised artificial mouth, for a diffuse room noise sound pressure p_{RN} measured at the MRP, but in the absence of all obstacles (e.g. test head, handset, etc.).

9 Methods for determining S_{mJ}, S_{Je}, S_{meST}, S_{RNST} and Δ_{SM}

When the sending, receiving and sidetone sensitivities of an actual local telephone system are required, the measurements according to the definitions given in clauses 6, 7 and 8 above can be made as illustrated in Figures 1, 2, 3, 4 and 5. These methods have been used by CCITT Laboratory and elsewhere successfully.

When using fast Fourier transform (FFT) techniques for measuring the characteristics of non-linear LTS, the measurement principle used, i.e. ratio of r.m.s. variables, or crosspectrum (coherent) method, should be specified.

More detail may be found in Section 3 of the *Handbook on Telephonometry*.

Figure 1 shows the method of setting up the artificial mouth so that the sound pressure p_m at the mouth reference point is known at each test frequency or frequency band. It is recommended to provide equalization in the artificial mouth drive circuit to maintain the free-field sound pressure constant at the MRP to within ± 1 dB over the frequency range 100 to 8000 Hz. In no case should the deviation exceed ± 2 dB over the frequency range 200 to 4000 Hz and +2/–5 dB over the frequency range 100 to 8000 Hz. It is recommended that any deviations from the desired sound pressure level be taken into account when determining the sending or sidetone sensitivity of a local telephone system. This is particularly true if the deviation exceeds ± 1 dB.

For any test signal, p_m of –4.7 dBPa is recommended (see *Handbook on Telephonometry*, 3.6.3).

When sine waves are used as the test signal, the test frequency points should be those of ISO 1/3 octave frequencies from 200 Hz to 4000 Hz for the purpose of calculating sending and receiving loudness ratings. However, for the purpose of frequency response measurement, the interval of 1/3 octave only gives a rough estimation of the system under test. In this case more frequency points are recommended. It is considered that the interval should be 1/12 octave or smaller. Also when various noise signals, or artificial voice such as Recommendation P.50, are used as test signal, the 1/3 octave filters will be used for both test signal and the output from the system under test for the purpose of calculating loudness rating. If more accurate frequency response curve is desired, the use of narrower bandwidth filters are recommended.

Supplement No. 20 to Series P Recommendations gives information on the effect of earphones coupling loss to the frequency response in real use condition.

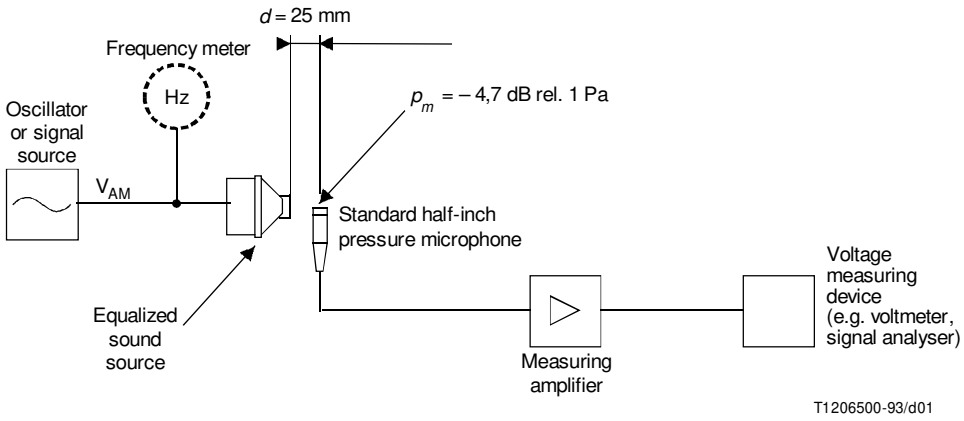


FIGURE 1/P.64

Measurement of acoustic pressure p_m at the mouth reference point 25 mm from the artificial lip plane of the sound source

Figure 2 shows the measurement of output V_J from the local telephone system when the handset 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 handset under test (see Figure 1).

Figure 3 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 ½ 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. Care must be taken to ensure that there is no coupling loss (acoustic leakage) between the ear-piece of the receiving system under test and the artificial ear. Usually E_J = –12 dBV is recommended.

NOTE – Some receiving systems incorporate electronic circuits to provide special features, for example, compression to limit the level of the received sound signal. Particular care must be exercised during the measurement of such systems to ensure that the resulting sensitivity is correct and relevant. In some cases it may be necessary to determine the receiving sensitivity over a range of input levels.

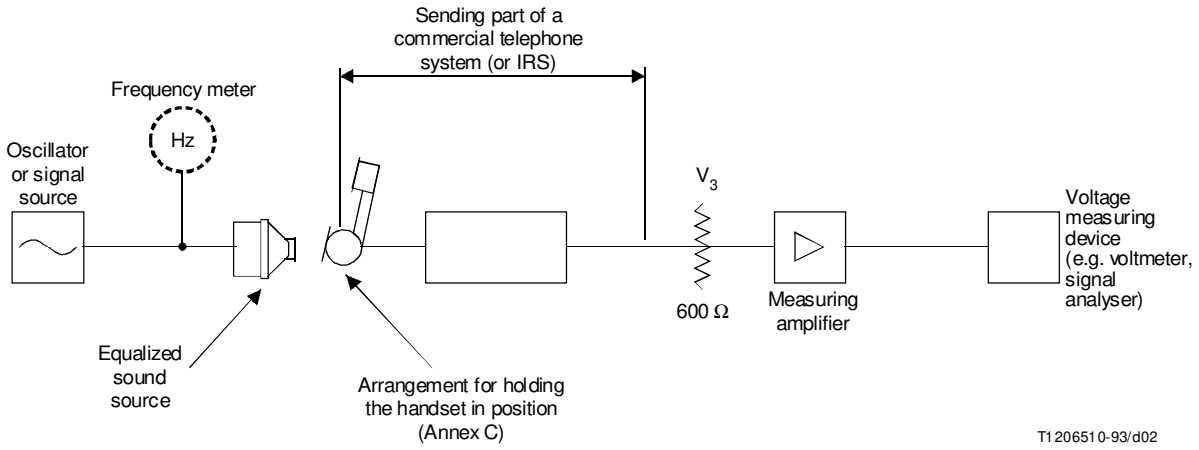


FIGURE 2/P.64

Voltage V_J measured across the terminals of a 600 ohms pure resistance connected to the output of a sending system under test

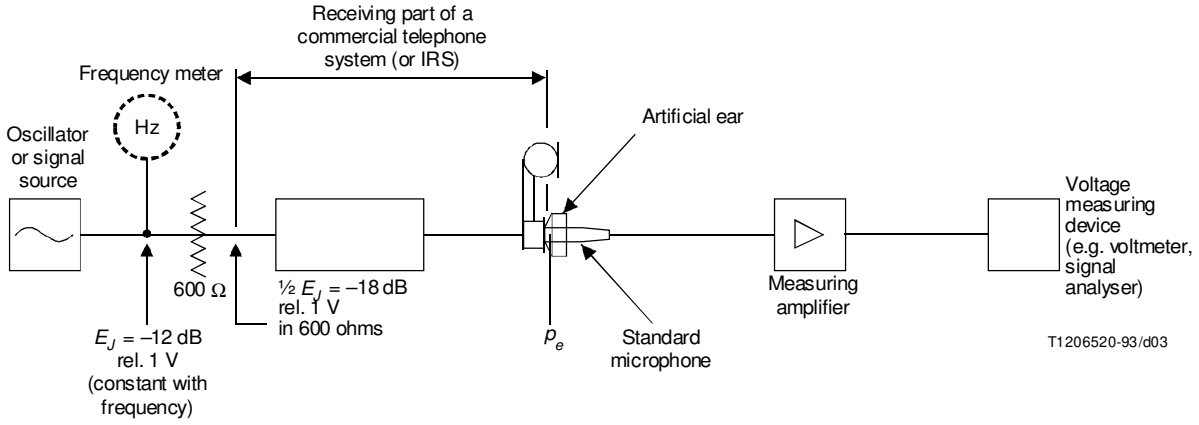


FIGURE 3/P.64

Measurement of acoustic pressure p_e at the ERP developed by the receiving system under test

Figure 4 shows the measurement of sidetone sensitivity. The resulting value of S_{meST} is highly dependent on the impedance connected to the telephone set terminals and therefore, under short line conditions, on the exchange termination. As this impedance often deviates considerably from 600 ohms, particularly when there is a complete connection present, 600 ohms is given only as an example.

The determination of the room noise sidetone sensitivity S_{RNST} is illustrated in Figure 5. For this measurement, sine wave signals are unsuitable and it is necessary to make use of continuous spectrum sound having, for example, a Hoth or pink noise spectrum (see B.3). First, the magnitude of the diffuse field p_{RN} is determined and then the sound pressure at the artificial ear is measured.

Using the above method, the sound pressure developed at the artificial ear usually is very low. An alternative way to determine S_{RNST} is to measure the sending sensitivity S_{mJ} using an artificial mouth and one of the methods in 3.6.3, *Handbook on Telephonometry*, using a continuous spectrum signal and then to measure the room noise sending sensitivity $S_{mJ/RN}$ using a diffuse field method such as described for room noise sidetone sensitivity above. (A detailed description of the method is given in the *Handbook on Telephonometry*).

The definition of Δ_{SM} is

$$\Delta_{SM} = S_{MJ/RN} - S_{MJ}$$

where S_{MJ} is the real voice sensitivity.

However, for all practical purposes, when using the artificial mouth, we may consider that Δ_{SM} is equal to Δ_{Sm}:

$$\Delta_{Sm} = S_{mJ/RN} - S_{mJ}$$

so that S_{RNST} can be determined by the approximation:

$$S_{RNST} \cong S_{meST} + \Delta_{Sm}$$

NOTES

1 For an explanation of how Δ_{Sm} may be used in the determination of Listener Sidetone Rating (LSTR) from Sidetone Masking Rating (STMR), see Recommendations P.76, P.79 and G.111.

2 In many cases, especially for carbon microphones, Δ_{Sm} , and hence also S_{RNST} is a function of the level of P_{RN}. It is recommended that in these cases the level of P_{RN} should be mentioned together with Δ_{Sm}. Typical value of P_{RN} should lie within 40-65 dBA (see *Handbook on Telephonometry*, 3.3).

3 Both S_{mJ} and $S_{mJ/RN}$ should use the same techniques, e.g. wideband signals measured in 1/3 octave bands.

4 The approximate formulae for S_{RNST} can be deemed to be equal for linear systems.

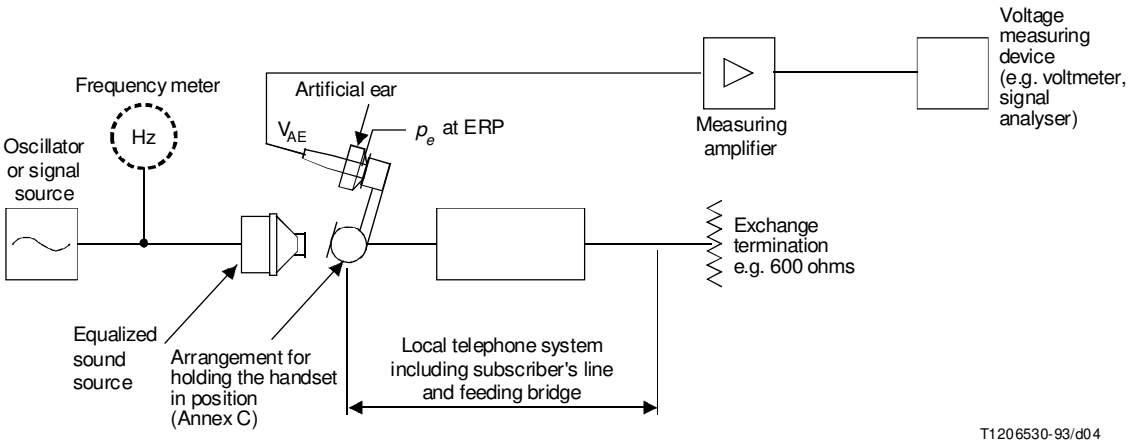
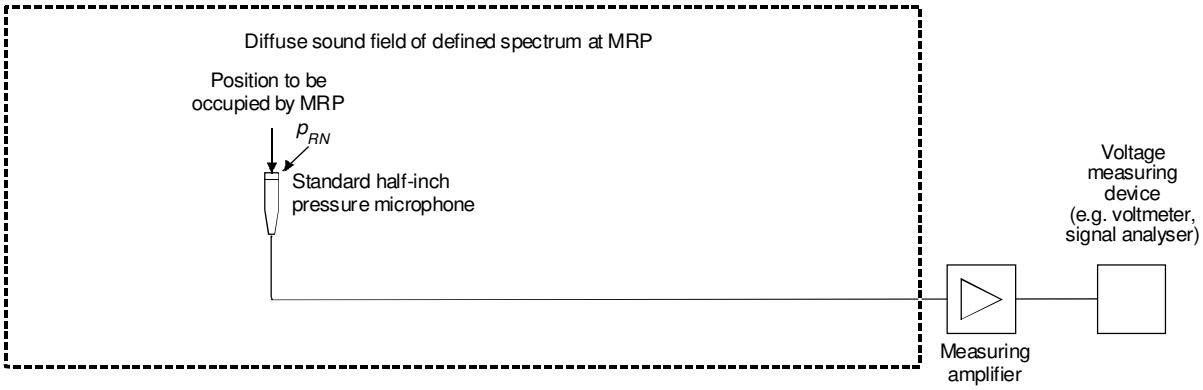
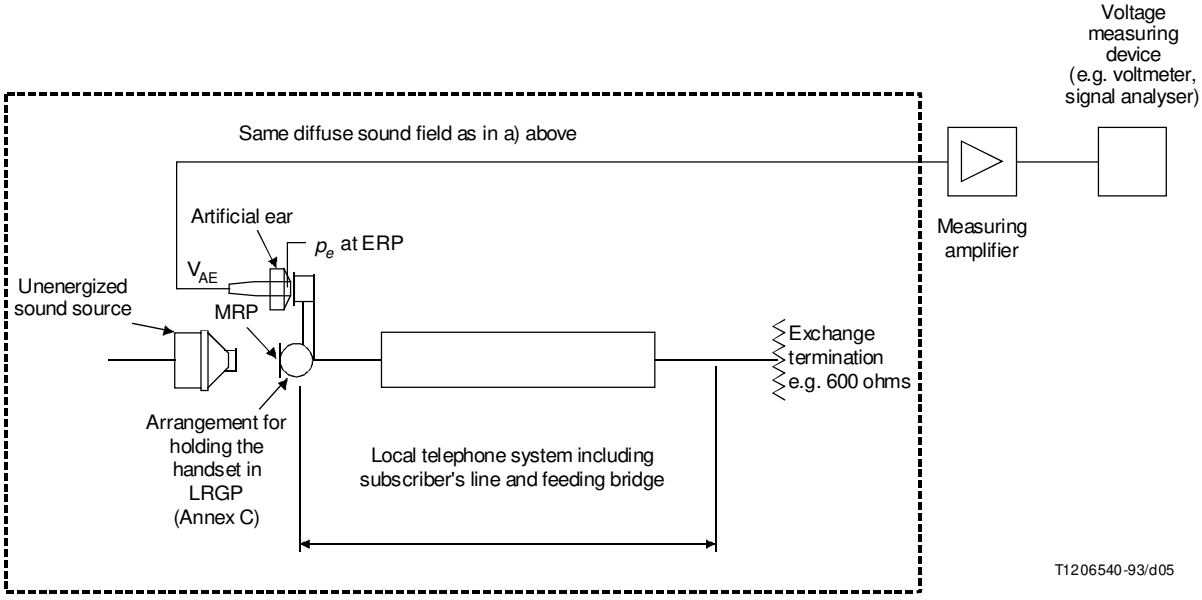


FIGURE 4/P.64

Measurement of the sidetone sensitivity of a commercial telephone system by determination of the sound pressure p_e developed at the ERP for a given sound signal at the MRP



a) Measurement of diffuse sound field at MRP



b) Measurement of p_e at the artificial ear due to diffuse sound field, p_{RN} heard via the telephone sidetone path

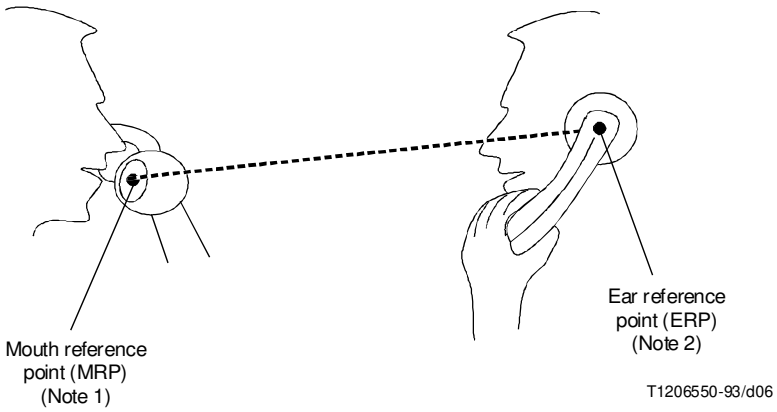
FIGURE 5/P.64

Annex A

Definitions of mouth reference point and ear reference point

(This annex forms an integral part of this Recommendation)

The definitions of mouth reference point (MRP) and ear reference point (ERP) are illustrated in Figure A.1.



NOTES

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.

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 front plane of a circular concave earcap when sealed to the ear.

3 The ERP should not be confused with the earcap reference point (ECRP) which is a point in the earphone reference plane used as a handset reference parameter (see Recommendation P.10).

FIGURE A.1/P.64

Definitions of mouth and ear reference points

Annex B

The application of complex terminating impedances when measuring SLR and RLR

(This annex forms an integral part of this Recommendation)

B.1 Complex impedance terminations

Administrations have found that the use of a complex impedance Z(f) as the nominal (2-wire) terminating impedance instead of 600 ohms improves echo and sidetone performance in the network.

The nominal impedance is implemented in such a way that it gives a reasonable approximation of the characteristic impedances of the unloaded subscriber cables most commonly used by the Administration. In this way, the impedance variations with cable lengths are very much diminished. In particular, it is a great advantage if an equipment connected to the 2-wire interface of a digital exchange (directly or via a cable) is designed to have an input impedance closely resembling this nominal impedance Z.

In general, Z is defined as a resistance R_s in series with a parallel combination of a resistance R_p and a capacitance C_p. Because the subscriber networks may differ between Administrations, the values of R_s, R_p and C_p also differ somewhat between Administrations. (See for instance Recommendation Q.552 for examples.)

When a particular complex nominal impedance Z is specified by an Administration, all 2-wire equipment measurements are usually made with that impedance, and not with 600 ohms. This rule also applies for telephone sets.

In transmission planning, the SLR and RLR of a telephone set are considered in conjunction with the relative levels at the point where the set is connected. For a certain relative level, the voltage of the (sinusoidal) 1020 Hz reference signal depends on the nominal impedance. Thus, to relate the send and receive sensitivities to the relative level, a correction factor K must be applied when the nominal impedance no longer is 600 ohms.

$$K = 10 \cdot \lg \left| \frac{Z\,(1020\ \text{Hz})}{600} \right| \ \text{dB};\ |\,Z\,| \text{ in ohms}$$

K is added to the send sensitivity and subtracted from the receive sensitivity.

The correction factor K can be included in the measuring set-up by use of an impedance converter which is specified in B.2. (In this case, the factor K need not be included in the calculations. Another advantage is that normal 600 ohms interfaces can be used in the rest of the measuring set-up.)

B.2 Impedance converter

This device converts the defined 600 Ω-points on the device under test to the appropriate nominal impedance Z_N and takes account of the corresponding correction factor. A further advantage is that all measurements are done with the specified level taking into account all influences even in non-linear systems. Thus, the value of the loudness rating is indicated correctly.

For measuring sending (SLR) and receiving (RLR), this converter must be switched in direction of amplification and correction factor. For measuring side-tone, no converter is needed because network-specific terminations are used anyway.

Figures B.1 and B.2 show how the device is inserted in Figures 2 and 3, respectively.

Specification of the impedance converters, the block diagram of which are shown in Figure B.3:

- open-loop gain at the sending end: (6 – k) dB ± 0.1 dB
- open-loop gain at the receiving end: (6 + k) dB ± 0.1 dB

where

$$K = 10 \lg \left| \frac{Z_N\ (1020\ \text{Hz})}{600} \right| ;$$

- return loss against the nominal value (600 Ω or ZN): $a_R \geq 40$ dB
- dynamic range: At input levels from –70 dB (775 mV) up to +15 dB (775 mV), deviation of linearity is less than 0.1 dB
- unweighted signal-to-noise ratio > 75 dB at 600 Ω; frequency range: 20 Hz - 20 kHz
- common mode rejection: > 50 dB.

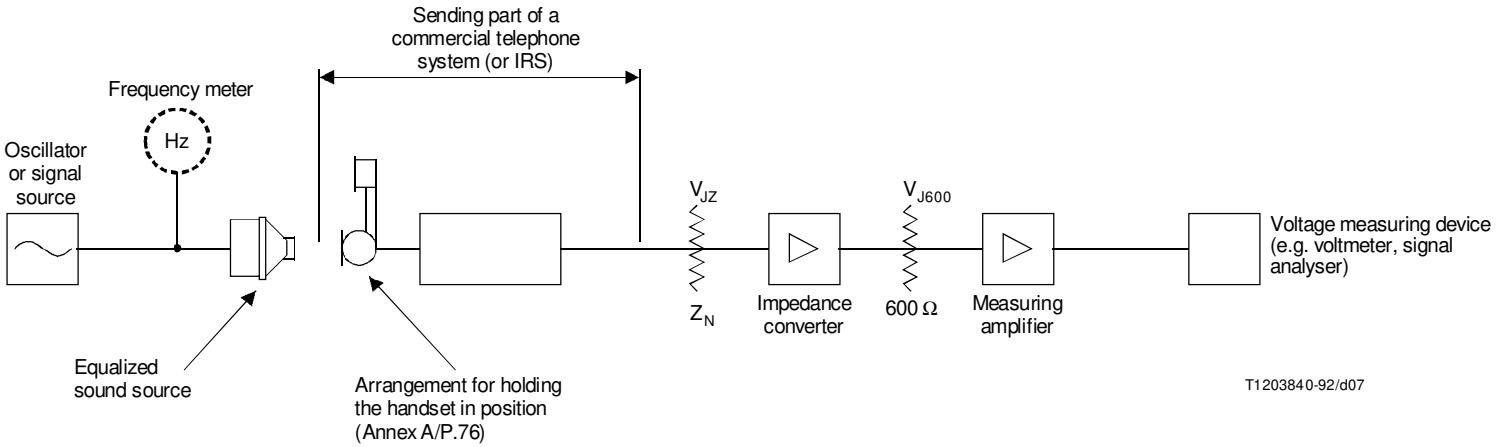


FIGURE B.1/P.64

Voltage V_{jc} measured across the terminals an impedance converter connected to the output of a sending system under test

(Addition to Figure 2/P.64)

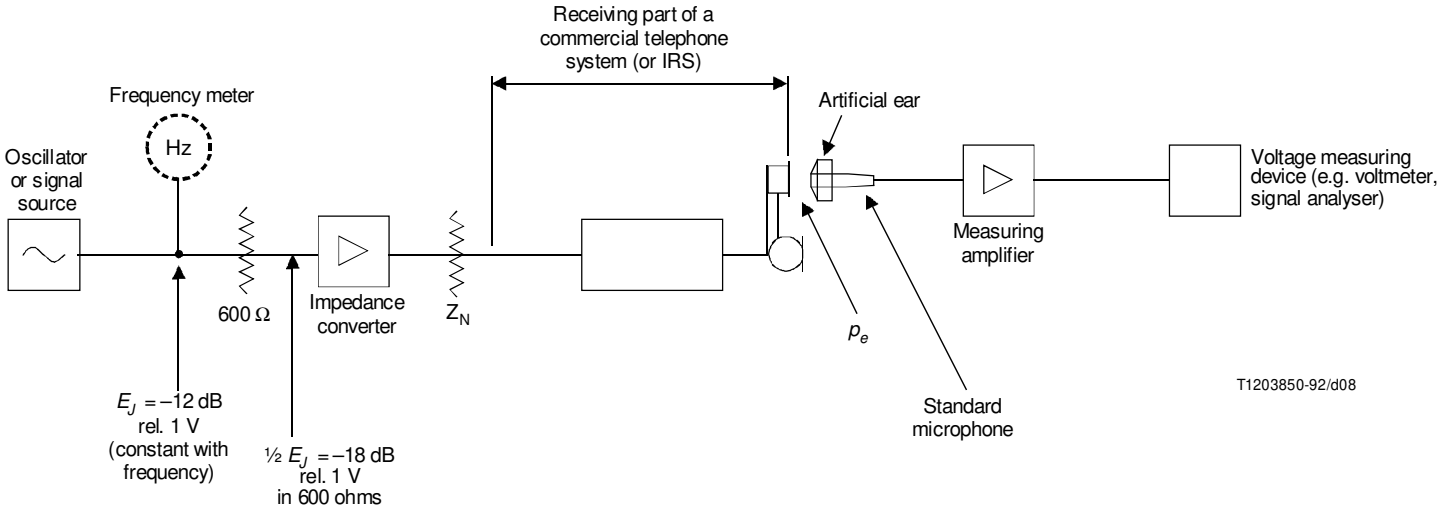


FIGURE B.2/P.64

Measurement of acoustic pressure p_e in the artificial ear developed by the receiving system under test connected to an impedance converter
(Addition to Figure 3/P.64)

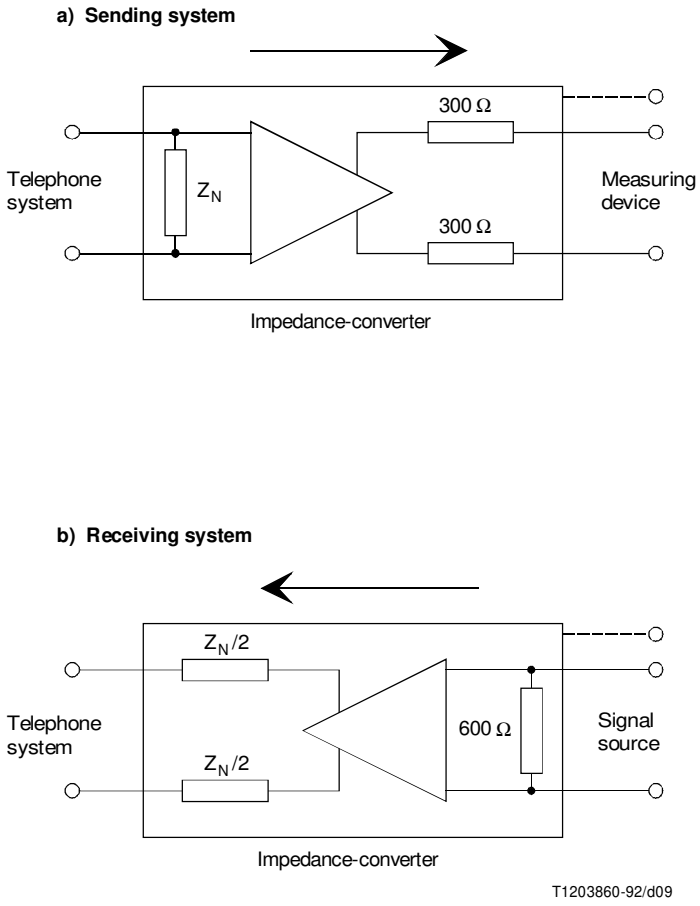


FIGURE B.3/P.64

Impedance converter

B.3 Differences when measuring a telephone set with a complex impedance using 600 ohms terminations

Figure B.4 shows the equivalent electric circuits when the send and receive sensitivities of a telephone set with the impedance Z are measured with 600 ohms terminations. Figure B.5 depicts a typical configuration of Z, a nominal complex impedance. (For the sake of simplification, it is assumed that the set impedance is exactly equal to this nominal impedance.)

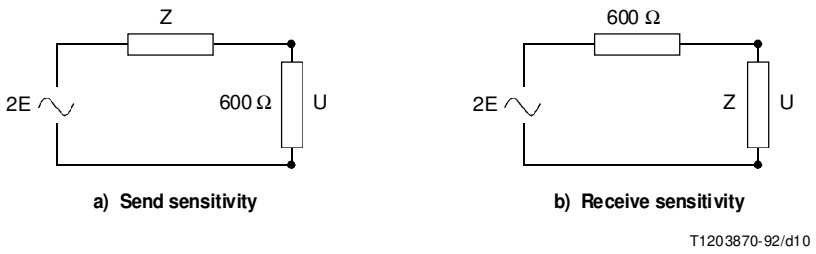


FIGURE B.4/P.64

Equivalent electric circuits when measuring with 600 ohms terminations

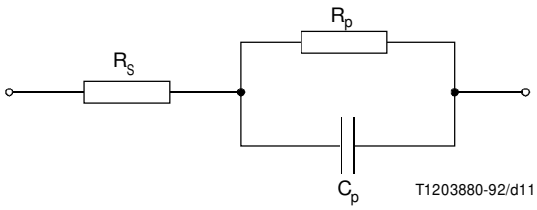


FIGURE B.5/P.64

Configuration of complex nominal impedance Z

Figure B.6 shows the equivalent circuit when the correct termination Z is used.

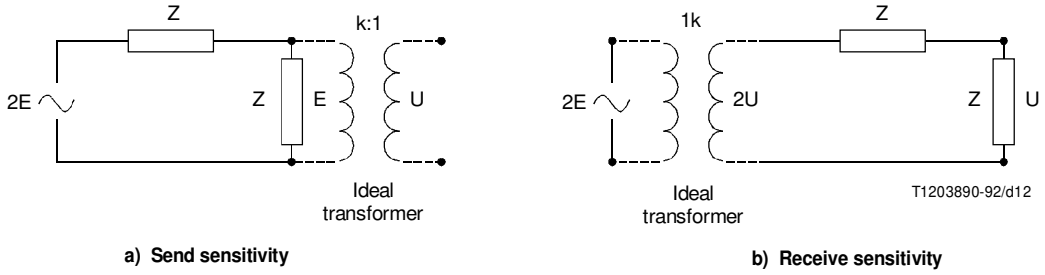


FIGURE B.6/P.64

Equivalent electric circuits when measuring with «Z» terminations

The (notional) ideal transformers in Figure B.6 are used to bring the (absolute) impedance value at the reference frequency 1020 Hz to 600 ohms at the measuring interface. The winding ratio is k:1, where

$$k = \sqrt{\left| \frac{Z\,(1020\ \text{Hz})}{600} \right|}\ ; \qquad \left| Z\,(1020\ \text{Hz}) \right| \text{ in ohms}$$

This corresponds to a dB correction of

$$K = 20 \cdot \lg k\ \text{dB}$$

The voltage gain in Figure B.6 a) is

$$20 \cdot \lg \frac{U}{E} = -20 \lg k = -K\ \text{dB}$$

and in Figure B.6 b)

$$20 \cdot \lg \frac{U}{E} = 20 \lg k = K\ \text{dB}$$

If 600 ohms terminations are used as in Figure B.4 instead of Z as in Figure B.6, certain differences will occur in the measured sensitivities.

For sending, the difference between the measured and the real sensitivity will be

$$Dsend = 20 \lg \left| \frac{600 + Z(f)}{2 \cdot 600} \right| - K \text{ [dB]}$$

and, correspondingly, for receiving

$$Drec = 20 \lg \left| \frac{600 + Z(f)}{2 \cdot Z(f)} \right| + K \text{ [dB]}$$

A numerical example

For Europe, the following compromise nominal complex impedance Z has been found feasible:

$$R_s = 275 \text{ ohms}, R_p = 850 \text{ ohms}, C_p = 150 \text{ nF}$$

Figure B.7 depicts the resulting differences.

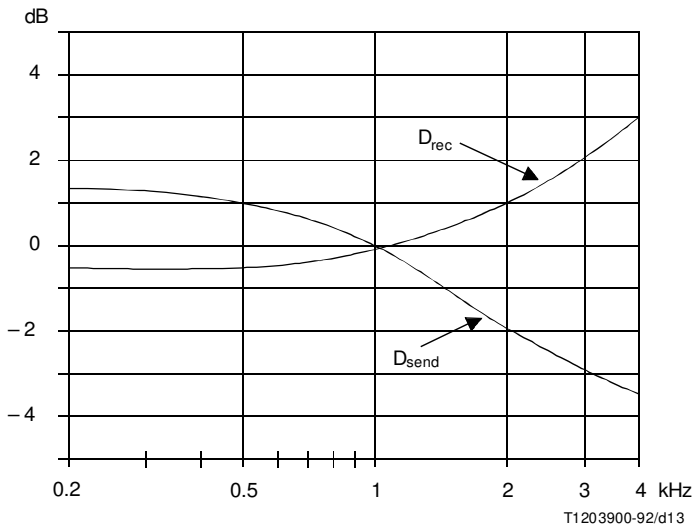


FIGURE B.7/P.64

Differences between measured and real sensitivities when 600 ohms terminations are used instead of the complex nominal impedance Z = 275 ohms + 850 ohms || 150 nF

Annex C

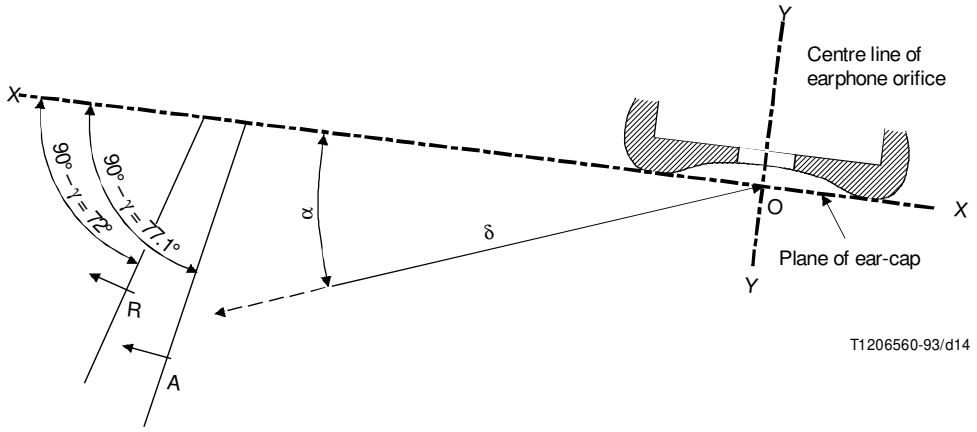
Definition of the speaking position for measuring loudness ratings of handset telephones

(This annex forms an integral part of this Recommendation)

This annex describes the speaking position which should be used to measure the sensitivities of commercial telephone sets (by the method described in Recommendation P.64) for the determination of loudness ratings.

C.1 The definition of a speaking position falls into two parts: description of the relative positions of mouth opening and ear-canal opening on an *average* human head; and description of the angles that define the attitude in space of telephone handsets held to such a head. For any given telephone handset, these descriptions together describe the relative special disposition of the microphone opening and the talker's lips, and hence the direction in which speech sound waves arrive at the mouthpiece and the distance they have travelled from a *virtual point source*.

The relative positions of the centre of the lips and that of the ear canal can be described in terms of a distance δ and an angle α as shown in Figure C.1. Point R in that figure represents the centre of a guard ring located at the reference equivalent speaking position in accordance with Recommendation P.72, *Red Book*. Position A is that used to determine ratings by the articulation method defined in Recommendation P.45, *Orange Book*. Averages of lip positions of 4012 subjects in the People's Republic of China cluster round the point A (see Recommendation P.35).



NOTES

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°.

2 Solid lines through A and R show plane of lips.

FIGURE C.1/P.64

Location of lip position relative to opening of ear canal

A second angle is required to define the direction in which speech is emitted from the mouth into the mouthpiece of the microphone. In former Recommendation P.45 and P.72 reference is made to an angle β, but this does not lie in the plane of symmetry of the handset, so it is more convenient to use an angle γ, which describes the vertical projection of the direction of speech on this plane.

C.2 The position of the centre of the lips as defined by A in Figure C.1 is used also to define the new speaking position, but two additional angles must also be defined, namely: the earphone rotational ange Φ and the handset rotational angle Θ. Earphone rotation is considered about an axis through the centre of the ear-cap (YY in Figure C.1); handset rotation is taken about a longitudinal axis of the handset (XX in Figure C.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:

$$\alpha = 22°, \gamma = 12.9°, \delta = 136 \text{ mm}, \Phi = 39° \text{ et } \Theta = 13°.$$

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 $\varepsilon = 77.8$ mm.

For any test jig, the manufacture tolerance should be within $\pm 0.5°$ for the angles defined above.

C.3 The foregoing description of the speaking position has shown the complexities of expressing the relative location of the ear reference point and the guard-ring centre, and the relative orientation of the earphone axis and the guard-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[2)] and the direction of the earphone axis with respect to the lip-ring. This is easier since the axis of the guard-ring is horizontal as would be the axis of an associated artificial mouth.

C.4 Use has been made of a vector analysis method to determine the orthogonal coordinates of the handset ear-cap relative to the lip position when the handset is mounted in the LR 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: verticla, with positive direction upwards.

The ear reference point is defined by the vector

$$(86.5, 77.8, 70.5) \text{ mm}.$$

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:

$$\pm (0.1441, -0.974, 0.1748)$$

Unit vector normal to plane of symmetry of the handset:

$$\pm (0.6519, -0.0394, -0.7572)$$

---

2) See Annex A for definition of ear reference point.

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.

Alternatively, it can be convenient to define the speaking position in terms of axes with the origin at the ear reference point. These are defined as follows:

x-axis: axis of ear-cap with positive direction away from earphone;

y-axis: line of intersection of the plane of symmetry of the handset with the ear-cap plane, with positive direction towards the microphone;

z-axis: normal to the plane of symmetry of the handset with positive direction obliquely upwards.

The lip-ring centre is defined by the vector

(50.95, 126.10, 0) mm.

The orientation of the lip-ring is defined by a unit vector along its axis

± (0.1441, –0.7444, –0.6250)

and the orientation of the handset is defined by specifying the vertical by the unit vector

± (0.1748, –0.6293, +0.7572).

NOTES

1 The speaking position defined above differs from the special guard-ring position in the values of Φ (= 37°) and Θ (= 19°). It has been found that altering the handset position from the special guard-ring position to the loudness rating guard-ring position described above affects sensitivity measurements to a negligible extent.

2 The term Loudness Rating GuardRing Position (LRGP) is to be used only when the definition described in this annex is strictly followed. It should be stressed that not only the relative position between the handset and the mouth should be followed, but also the mouth should be horizontal.