SECTION 5

OBJECTIVE ELECTRO-ACOUSTICAL MEASUREMENTS

Recommendation P.61

METHODS FOR THE CALIBRATION OF CONDENSER MICROPHONES

(amended at Malaga-Torremolinos, 1984)

Primary and secondary calibrations of condenser microphones can be carried out using the methods described below.

1 Primary calibration by the reciprocity method

The recommended procedure for primary calibration of condenser microphones is the reciprocity calibration technique method for reciprocity pressure calibration is described in [1]. A simplified method, suitable for calibration over the frequency range of interest for telephonometric measurements, is given in [2]. Although the methods described are specifically for one-inch microphones , similar methods are applicable to half-inch microphones microphones are under study by IEC.

A precision method for free-field reciprocity calibration is given in [3]. Alternatively, the free-field correction curves given in [4] may be applied to the pressure calibration of one-inch condenser microphones to determine their free-field responses. The reciprocity free-field calibration method may in principle be extended to half-inch microphones. Free-field correction curves have not been standardized for half-inch microphones.

2 Secondary calibration by the comparison method

The secondary calibration of a condenser microphone may be achieved by direct comparison with a physically identical microphone having a known calibration. The procedure used is a modification of the ‘‘two microphones and auxiliary sound source’’ method described in [1] to [3]. The output of the calibrated microphone is first determined for a given drive level applied to the auxiliary sound source. The calibrated microphone is then replaced by the microphone to be calibrated, and its output is determined for the same drive level applied to the auxiliary sound source. The difference in level (in dB) between the outputs of the two microphones is then applied to the known calibration of the first microphone to determine the calibration of the second. The procedure is repeated at each frequency of interest.

3 Secondary calibration using pistonphones and other sound level calibrators

Secondary calibrations can also be made using pistonphones and other sound level calibrators which produce a known sound level. Such

devices are often used to check the calibration of a microphone at a single frequency Care should be taken to follow the manufacturer's instructions when using such devices; in particular, it may be necessary to apply corrections for barometric pressure, coupler volume, microphone type, etc. Standardization of these calibrators is currently under study by the IEC.

Calibrations with an accuracy of ± | .3 dB are possible.

References

[1] International Electrotechnical Commission *Precision method for pressure calibration of one-inch standard condenser microphones by the reciprocity technique* , IEC publication 327, Geneva, 1971.

[2] International Electrotechnical Commission *Simplified method for pressure calibration of one-inch condenser microphones by the reciprocity technique* , IEC publication 402, Geneva, 1972.

[3] International Electrotechnical Commission *Precision method for free-field calibration of one-inch standard condenser microphones by the reciprocity technique* , IEC publication 486, Geneva, 1974.

[4] International Electrotechnical Commission *Values for the difference between free-field and pressure sensitivity levels for one-inch standard condenser microphones* , IEC Publication 655, Geneva, 1979.

Recommendation P.62

MEASUREMENTS ON SUBSCRIBERS' TELEPHONE EQUIPMENT

(amended at Malaga-Torremolinos, 1984 and Melbourne, 1988)

1 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 the methods described in Recommendation P.64 and its associated Annex B may be used.

2 Measurement of the nonlinear distortion of a telephone set and of microphone noise

While the nonlinear distortion of telephone receivers is in general negligible, microphones (and particularly carbon microphones of the type generally used in commercial telephone equipment) show considerable nonlinearity: the relationship between the variation of microphone resistance and the acoustic pressure on the diaphragm is not linear. This nonlinearity

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 may be two supplementary effects:

1) The microphone is less sensitive 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).

Existing information on the general effect of harmonic distortion on telephone speech quality indicates that the effect of second order distortion is considerably less than that of third order distortion. Absolute detection thresholds obtained in different test are, however, difficult to compare because of differences in definition and measurement of the distortion.

Note 1 — Summaries of information available in this area are given in [1] and [2]. It is clear that measurements with sinusoidal signals can predict the speech transmission performance of nonlinear systems only to a limited extent, particularly if the peak value of the test signal is much smaller than the transmitted speech signal. A complex signal having the same spectral density at the same amplitude density function as real speech, see Recommendation P.50, is therefore expected to be a more useful test signal.

Note 2 — The application of complex test signals or actual speech signals for the measurement of nonlinearity in telephone circuits is studied under Question 13/XII [3].

Certain types of carbon microphones may produce an audible stationary noise, often depending on the size of feeding current. The measurement of this kind of noise and its effect on transmission quality is the same as for other kinds of additive circuit noise.

3 Objective measurement of loudness rating (LR)

Examples of apparatus that objectively measure LRs conforming to Recommendation P.65 are ‘‘CERF’’ of the French Administration [4], ‘‘AURAL’’ of NTT [5], ‘‘TIGGER’’ [6] of British Telecom and ‘‘Loudness Rating Meter’’ [7] of STL. Short descriptions of the apparatus named above can be found in Chapter 5 of the CCITT *Handbook on Telephonometry* [8].

References

[1] CCITT — Question 13/XII, Annex 1, Contribution COM XII-No.1, Study Period 1981-1984, Geneva, 1981.

[2] CCITT — Question 13/XII, Annex, Green Book, Vol. V, ITU, Geneva, 1973.

[3] CCITT — Question 13/XII, Contribution COM XII-No. 1, Study Period 1985-1988, Geneva, 1985.

[4] CCITT — Contribution COM XII-No. 184, *Equipment for the objective measurement of equivalent R25 and of the sidetone — used by the French Administration* (France), Study Period 1981-1984.

[5] CCITT — Contribution COM XII-No. 79, *Objective loudness rating measurement system* , (NTT), Study Period 1981-1984.

[6] WARD (H. | .) and CROSS (R. | .) TIGGER: An Automatic Test System for measuring the Transmission Performance of Telephones, *British Telecommunications Engineering* , Volume 2, July 1983.

[7] CCITT — Question 15/XII, Annex 6, Contribution COM-No. 1, Study Period 1985-1988.

[8] CCITT *Handbook on Telephonometry* , UIT, Geneva, 1987.

Recommendation P.63

METHODS FOR THE EVALUATION OF TRANSMISSION QUALITY ON THE BASIS OF OBJECTIVE MEASUREMENTS

These measuring methods are being studied by the CCITT under Question 7/XII [1]. Annexes A and B to Recommendation P.11 and Supplements No. 2 and 3, at the end of this Fascicle, describe methods used respectively by British Telecom and AT&T. Attention is also drawn to methods for calculating loudness ratings given in Recommendation P.79.

Reference

[1] CCITT — Question 7/XII, Contribution COM XII-No. 1, Study Period 1985-1988, Geneva, 1985.

Recommendation P.64

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

OF THEIR LOUDNESS RATINGS

(Geneva, 1976; amended at Malaga-Torremolinos, 1984

and Melbourne, 1988)

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.

Note 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 side-tone is outside the scope of the existing Recommendations.

Note 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 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).

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 , 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.

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 A 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. 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.51) 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 § 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 Questions 8/XII and 12/XII.

In principle, the artificial voice defined in Recommendation P.50 should be used as the acoustic test signal 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 Annex B).

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 ± | .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 § 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 A to Recommendation P.76 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:

$$S_{mJ} = 20 \log_{10} \left(\frac{V_J}{p_m} \right) \text{ dB rel 1 V/Pa}$$

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.

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 studied under Question 8/XII [1]. 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 Annex B.

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.51, is expressed as follows:

$$S_{Je} = 20 \log_{10} \frac{p_e}{\tfrac{1}{2} E_J} \quad \text{dB rel 1 Pa/V}$$

where p_e is the sound pressure in the artificial ear and ½ E_J is half the emf in the 600 ohm source.

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 § 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 § 6 and p_e is the sound pressure developed in the artificial ear 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 in the artificial ear 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 ΔS_M

When the sending, receiving and sidetone sensitivities of an actual local telephone system are required, the measurements according to the definitions given in §§ 6, 7 and 8 above can be made as illustrated in Figures 1/P.64, 2/P.64, 3/P.64, 4/P.64 and 5/P.64. 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 of Telephonometry* [2].

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 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 ± | dB over the frequency range 100 to 8000 Hz. In no case should the deviation exceed ± | 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 ± | dB.

For any test signal, p_m of —4.7 dBPa is recommended (see Note 2 to Annex B for information).

FIGURE 1/P.64, p.

Figure 2/P.64 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/P.64).

FIGURE 2/P.64 p.

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 $S_{J\backslash de}$ 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. 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 3/P.64, p.

Figure 4/P.64 shows the measurement of sidetone sensitivity. The resulting value of $S_{m\backslash de\backslash dS\backslash dT}$ 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.

FIGURE 4/P.64, p.

The determination of the room noise sidetone sensitivity S_{RNST} is illustrated in Figure 5/P.64. 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 $p_{R\backslash dN}$ is determined and then the sound pressure at the artificial ear is measured.

Figure 5/P.64, p.

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 $S_{m\backslash dJ}$ using an artificial mouth and one of the methods in Annex B, using a continuous spectrum signal (e.g. §§ B.3, B.4) 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

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

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}:

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

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

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

Note 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.

Note 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).

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

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

ANNEX A

(to Recommendation P.64)

Definitions of mouth reference point and ear reference point

The definitions of mouth reference point (MRP) and ear reference point (ERP) are illustrated in Figure A-1/P.64.

FIGURE A-1/P.64, p.

ANNEX B

(to Recommendation P.64)

Measurement of local telephone systems containing

carbon microphones

For the measurement of local telephone systems containing carbon microphones, various methods have been suggested and tried. The following gives, as examples, some of these methods. These same methods can also apply to telephones using linear microphones.

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

Further information can be found in [3].

Note 1 — The efficiency of the artificial mouth used is not generally constant with frequency, so it is necessary, for most of the methods described below, to insert appropriate equalization networks between the electrical signal generator and the loudspeaker of the artificial mouth. It is the free field acoustical signal which shall conform to the complex signal or the artificial voice specified.

Note 2 — It has been found that for specific applications it may be advantageous to use speech levels other than the —4.7 dBPa recommended below. This should only be done with care having due regard to the particular application. Studies carried out during the Study Period 1985-1988 have shown that better agreement with subjective test results are obtained with somewhat lower levels, e.g. over the range —4.7 to —7.0 dBPa.

B.1 The *upper envelope method* has been used in the CCITT Laboratory with success for some types of carbon microphone but has been less successful with others. The upper envelope method is as follows:

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 loudness ratings in accordance with the subjective test method described in Recommendation P.78;

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.

B.2 *Sweeping frequency method*

Some available types of objective instrumentation for measuring loudness-related ratings use 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.

This method should not be used for determining $\Delta_{S\backslash dM}$.

B.3 *Pink-noise method*

The handset containing the carbon microphone is placed in front of an artificial mouth producing at the MRP pink noise (power spectrum density diminishing by 3 dB/octave) over 1/3rd octave frequency bands centred on the preferred frequencies specified in ISO Standard 266-1975 at 1/3rd octave intervals in the range 100 to 8000 Hz with the band edges conforming to the filters described in IEC 225.

The total level of the signal, measured over the same bandwidth, should be —4.7 dBPa with a tolerance of ± |.0 dB.

Note — This may not be practical with all artificial mouths, and a narrower bandwidth of 200 to 8000 Hz may have to be used for some types of artificial mouths.

The sensitivity/frequency characteristic is obtained by finding the ratio of the spectrum density of the signal delivered by the telephone system to the spectrum density of the signal obtained using a small linear microphone placed at the MRP under free-field conditions (after removing the handset).

B.4 *Shaped gaussian noise method*

The method uses shaped Gaussian noise at the MRP whose long-term average spectrum density is the same as shown in Table 1/P.50. The total level of the signal should be —4.7 dBPa ± | dB.

The sensitivity/frequency characteristic is obtained as in § B.3.

B.5 *Real-voice calibration*

This may be performed by measuring speech spectra emitted alternately or simultaneously 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.

B.6 *Application of a wideband signal*

The wideband signal is generated by a pseudo-random binary sequence and is then equalized to have a long-term average spectrum density flat or as defined in Recommendation P.50. The output from the carbon microphone is then processed by fast Fourier transform (FFT) techniques. This method, like the previous method, requires calibration by a linear microphone of known sensitivity/frequency characteristic to determine the value of p_m. This method has the advantage that the frequency characteristic may be obtained with a sample of test signal of very short duration (e.g. 50 ms).

B.7 *Method using the artificial voice*

The method uses the artificial voice, having spectral and time characteristics similar to those of speech.

The sensitivity/frequency characteristic is obtained as in § B.3 above, but with the artificial mouth supplying the acoustic artificial voice, defined in Recommendation P.50.

References

[1] CCITT — Question 8/XII, Contribution COM XII-No. 1, Study Period 1989-1992.

[2] CCITT — *Handbook of Telephonometry* ; ITU, Geneva, 1987.

[3] CCITT — Contribution COM XII-R 27, § B.2, Study Period 1985-1988.

Recommendation P.65

OBJECTIVE INSTRUMENTATION FOR THE DETERMINATION OF LOUDNESS RATINGS

(Malaga-Torremolinos, 1984, amended at Melbourne, 1988)

1 Introduction

This Recommendation describes the essential features of objective instrumentation suitable for the determination of loudness ratings. These features are drawn from current Recommendations relating to loudness ratings , the principles of which are defined in Recommendation P.76.

It is possible to realize objective instrumentation for loudness rating purposes in a number of ways, for example by the assembly of a number of separate instruments, each having its own defined function, and possibly under some central control, or by means of a dedicated piece of apparatus specially designed for the purpose. However, in order to ensure that loudness rating measurements made in different laboratories have an acceptable level of agreement, say ± 1 dB, it is essential that the Recommendations relating to the measurement of the electro-acoustic performance of telephone systems should be followed.

The relevant Recommendations are:

P.48 Specification for an intermediate reference system

P.51 Atificial ear and artificial mouth

P.64 Determination of sensitivity/frequency characteristics of local telephone systems to permit calculation of their loudness ratings

P.75 Standard conditioning method for handsets with carbon microphones

P.76 Determination of loudness ratings; fundamental principles

P.79 Calculation of loudness ratings

2 Instrumentation

The four electro-acoustic sections that are required to be included in equipment intended for use in determining loudness ratings are described below. In each case appropriate calibration is required as a function of frequency, and calibration values recorded in the fifth section where the particular sensitivity/frequency charac teristic is derived and the loudness rating calculated. If the instrumentation is to include the measurement of listener sidetone rating (LSTR), a sixth section must be provided, namely a diffuse room noise source together with appropriate facilities for calibration, measurement and analysis in one-third octave bands.

It is necessary to provide certain auxiliary apparatus, such as feeding circuits, artificial subscriber cable and exchange terminations, as required by the particular Recommendation(s) being followed for any given measurement.

2.1 *Artificial ear*

See *a)* of Figure 1/P.65.

The artificial ear in the system should be in accordance with Recommendation P.51 and contain within it a measuring amplifier so that the pressure p_e occurring at the artificial ear cavity can be measured as a function of frequency, or in frequency bands within the recording and measurement system, *e)* of Figure 1/P.65. Means must also be available to calibrate the standard microphone used in the artificial ear employing, for example, an acoustic calibrator or piston-phone

2.2 *Artificial voice*

See *b)* of Figure 1/P.65.

An artificial mouth complying with Recommendation P.51 must be part of the system and be able to produce a prescribed sound field at the MRP 25 mm in front of the lip plane. A signal source will be part of the artificial voice and this source may be sine waves (swept or discrete

frequencies) or a wideband signal (e.g. the artificial voice defined in Recommendation P.50, or shaped Gaussian noise as defined in Recommendation P.64, § B.4). Equalization and gain control should be part of the drive system to the artificial mouth such that the sound pressure at the MRP can be controlled in accordance with the requirements of Recommendation P.64, §§ B.1 and B.4, or as appropriate.

Calibration of the sound pressure and/or spectrum at the MRP may be carried out using the standard microphone used in the artificial ear of § 2.1 above, making use of the recording and measurement system of § 2.5 below to

Other algorithms are being studied by Study Group XII, under Question 15/XII.

determine p_m as a function of frequency, or in frequency bands.

Mechanical means must be provided to hold the test handset in the LRGP (loudness rating guard-ring position), in accordance with the requirements of Recommendation P.76, Annex A. If handsets having carbon microphone are being tested, conditioning in accordance with Recommendation P.75 must be provided.

2.3 *Electrical termination*

See *c)* | f Figure 1/P.65.

The system should contain a 600 ohm balanced terminating impedance with means for measuring the terminating voltage, V_J (see Recommendation P.64, §§ 6 and 9, as a function of frequency, or in frequency bands, using the recording and measurement system of § 2.5 below. Calibration of this section may be carried out using a calibrated voltage source.

2.4 *Electrical signal source*

See *d)* | f Figure 1/P.65.

An electrical signal source must be provided having a 600 ohm balanced impedance. The electrical source need not be the same as that used for the artificial voice but should either be sine waves or a wideband signal. There should be means for calibrating and adjusting the generator voltage, E_J, to the requirements of Recommendation P.64, §§ 7 and 9 over the frequency range 100-8000 Hz. This may be carried out using calibration of the electrical termination of § 2.3 above.

2.5 *Recording and measurement system*

See *e)* | f Figure 1/P.65.

In order to determine the sound pressure p_e at the artificial ear or the voltage V_J at the electrical termination it will be necessary to provide a recording and measurement system. This measurement system may, using hardware or software, contain filters in order to improve signal-to-noise ratio or for analysing the output of the telephone set in 1/3rd octave frequency bands. Where a bank of 1/3rd octave filters is used these should be centred on the preferred frequencies in accordance with ISO 266 and have the characteristics in accordance with IEC Publication 225.

Within this part of the system there should be recording or storage facilities so that calibration and measurement data may be used to derive the necessary sensitivity/frequency characteristics in accordance with Recommendation P.64. The various loudness ratings are then computed in accordance with Recommendation P.79 from the sensitivity/frequency characteristics, taking into account any recognized adjustments, for example L_E or L_M. Values for L_E and L_M may be fed into the calculation using default values (e.g. those listed for L_E in Table 4/P.79) or from other more appropriate data sources when available.

2.6 *Diffuse room noise source*

See *f)* | of Figure 1/P.65.

If LSTR is to be measured, a diffuse room noise source must be available, calibrated to provide a prescribed sound field at the position to be occupied by the MRP in the absence of the test head and all other obstacles, and as described in Recommendation P.64, § 9. Calibration of the diffuse sound pressure $p_{R\backslash dN}$ may be carried out using the standard microphone used in the artificial ear of § 2.1, making use of the recording and measurement system of § 2.5 to determine $p_{R\backslash dN}$ as a function of frequency in the frequency bands.

Because of the nature of room noise sidetone, it will normally be appropriate to use a diffuse sound pressure $p_{R\backslash dN}$ that is much lower than the value of —4.7 dBPa used for p_m in determining STMR and SLR. Typical values for $p_{R\backslash dN}$ would lie in the range 40-65 dB SPL (—54 to —29 dBPa, A weighted), and it should have a frequency spectrum appropriate for the application, for example as given in Supplement No. 13, § 2. The actual level and type of noise should always be stated in quoting test results.

3 Measurements

Facilities should be provided to enable the various sections of the instrumentation to be connected allowing the measurement of the necessary sensitivity/frequency characteristics and calculation of the loudness ratings.

A summary of these interconnections, together with the sensitivity/frequency characteristics (SFC) measured for particular loudness rating determinations, are given below.

3.1 *Send loudness rating (SLR)*

Source: *b)* | f Figure 1/P.65

Load: *c)* | f Figure 1/P.65

Send SFC given by:

$$S_{mJ} = 20 \log_{10} \frac{V_J}{p_m} \text{ dB}$$

Figura 1/P.65, p.

3.2 *Receive loudness rating (RLR)*

Source: *d)* | f Figure 1/P.65

Load: *a)* | f Figure 1/P.65

Receive SFC given by:

$$S_{Je} = 20 \log_{10} \frac{p_e(f)}{\frac{1}{2} E_J(f)} \text{ dB}$$

3.3 *Sidetone masking rating (STMR)* (Talker Sidetone)

Source: *b)* | f Figure 1/P.65

Load: *a)* | f Figure 1/P.65

Sidetone SFC is given by:

$$S_{meST} = 20 \log_{10}$$

$$\frac{f\,Ip \quad e\,f\,R}{f\,Ip \quad m\,f\,R} \text{ dB}$$

Note — The quantity L_{meST} used in the calculation of STMR is given by:

$$L_{meST} = S_{meST} \text{ dB}$$

3.4 *Listener sidetone rating (LSTR)*

Source: *f)* | f Figure 1/P.65

Load: *a)* | f Figure 1/P.65

Room noise sidetone SFC is given by:

$$S_{RNST} = 20 \log_{10} \frac{f\,Ip \quad e\,f\,R}{f\,Ip \quad RN\,f\,R} \text{ dB}$$

3.5 *Overall loudness rating (OLR) (Overall Send + Receive (OSR))*

Source: *b)* | f Figure 1/P.65

Load: *a)* | f Figure 1/P.65

Overall SFC given by:

$$S_{me} = 20 \log_{10} \frac{f\,Ip \quad e_{fR}}{f\,Ip \quad m_{fR}} \text{ dB}$$

3.6 *JLR Junction loudness rating*

Source: *d)* | f Figure 1/P.65

Load: *c)* | f Figure 1/P.65

Junction loss/frequency characteristics given by:

$$X_{U} = 20 \log_{10} \frac{(12\,E \quad J_{fR}}{f\,IV \quad J_{fR}} \text{ dB}$$

Note — Impedance terminations of 600 ohms are assumed.

Recommendation P.66

METHODS FOR EVALUATING THE TRANSMISSION PERFORMANCE OF DIGITAL TELEPHONE SETS

(Melbourne, 1988)

1 Introduction

The specifications in this Recommendation are subject to future enhancement and therefore should be regarded as provisional.

The CCITT recommends the following method to evaluate the voice transmission performance of a digital telephone set using encoding conforming to Recommendation G.711 (see also Recommendation P.31). A digital telephone

set is one in which the A/D and D/A converters are built in and the connection to the network is via a digital bit-stream. This poses a fundamental problem in applying existing performance and measurement techniques, such as Recommendations P.64, P.34 and P.38, since these are generally given in terms of a transfer function of analogue input and output quantities, e.g. a frequency response. The principles involved in this Recommendation are applicable to handset, headset and hands-free operation; however, at present only procedures applicable to handset operation have been developed.

2 Approaches for testing digital telephones

There are two methods for evaluating the transmission performance of a digital telephone, the codec approach and the direct approach.

In the short term, use of the codec approach is advocated since many Administrations already have some experience with this methodology (see Recommendation O.133).

Figure 1/P.66, p.

2.1 *Codec approach*

In this approach, shown in Figure 1/P.66, a codec is used to convert the companded digital input/output bit-stream of the telephone set to the equivalent analogue values, so that existing test procedures and equipment can be used. This codec should be a high-quality codec whose characteristics are as close as possible to ideal (see § 5).

2.2 *Direct digital processing approach*

In this approach, shown in Figure 2/P.66, the companded digital input/output bit-stream of the telephone set is operated upon directly.

Note — This approach is still under study in Question 38/XII.

Figure 2/P.66, p.

3 Definition of 0 dB reference point

To preserve compatibility with existing codecs already in use in local digital switches, which are defined as a 0 dBr point, the codec (A- or μ-law) sould be defined as follows:

— D/A converter: a digital test sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose r.m.s. value is 3.14 dB (A-law) or 3.17 dB (μ-law) below the maximum full-loaded capacity of the codec will generate 0 dBm across a 600 ohm load;

— A/D converter: a 0 dBm signal generated from a 600 ohm source will give the digital test sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose r.m.s. value is 3.14 dB (A-law) or 3.17 dB (μ-law) below the maximum full load capacity of the codec;

where DTS is defined as a periodic sequence of character signals as given in Tables 5/G.711 and 6/G.711.

4 Definition of interfaces

The digital telephone test equipment will, in general, be connected to the telephone under test through an interface.

Such an interface should be able to provide all the signalling and supervisory sequences necessary for the telephone set to be working in all test modes. The interface must be capable of converting the digital output stream from the tested set (which may be in various formats, depending on the specific type of telephone set, e.g. conforming to Recommendation I.412 for ISDN sets), to a form compatible with the test equipment. Interfaces can be applied for sending and receiving separately, taking into account telephone sets which are connected to various types of exchanges.

5 Codec specification

5.1 *Ideal codec*

This characteristic can be realized, for example, using oversampling techniques and digital filters.
The ideal codec consists of an independent encoder and decoder whose characteristics are hypothetical and comply with Recommendation G.711. The ideal encoder is a perfect analogue-to-digital converter preceded by an ideal low-pass filter (assumed to have no attenuation/frequency distortion and no envelope-delay distortion), and may be simulated by a digital processor. The ideal decoder is a perfect digital-to-analogue converter followed by an ideal low-pass filter (assumed to have no attenuation/frequency distortion and no envelope-delay distortion), and which may be simulated by a digital processor

For the measurement of the sending side of a telephone set, the output digital signal is converted by the decoder to an analogue signal. The electrical characteristics of this output signal are measured using conventional analogue instruments. For the measurement of the receiving side of a telephone set, the analogue output from a signal source is converted to a digital signal by the ideal encoder and fed to the receiving input of the digital telephone set.

5.2 *Reference codec*

A practical implementation of an ideal codec may be called a reference codec (see Recommendation O.133, § 4).

For the reference codec, characteristics such as attenuation/frequency distortion, idle channel noise, quantizing distortion, etc. should be better than the requirements specified in Recommendation G.714, so as not to mask the corresponding parameters of the set under test. A suitable reference codec may be realized by using:

1) at least 14 bit linear A/D and D/A converters of high quality, and transcoding the output signal to the A- or μ-law PCM format;

2) a filter response that meets the requirements of Figure 3/P.66.

5.2.1 *Analogue interface*

The output and input impedances return loss and longitudinal conversion losses of the analogue interface of the reference codec should be in accordance with Recommendation O.133, § 3.1.1.

5.2.2 *Digital interface*

The fundamental requirements for the reference codec digital interface are given in the appropriate Recommendations (e.g. I.430-Series Recommendations for ISDN telephone sets).

6 Measurement of digital telephone transmission characteristics

Use of the codec test approach means that test procedures for digital telephone sets in general follow those for analogue sets (see Recommendation P.64). The reference codec should meet the requirements of § 5. An important difference, however, concerns the test circuits themselves, see Figures 4/P.66 to 10/P.66.

The set is connected to the interface and is placed in the active call state.

Note — When measuring digital telephone sets, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance on the frequencies of ± | % which may be used to avoid this problem, except for 4 kHz where only the —2% tolerance may be used.

Figure 3/P.66, p.

6.1 *Sending*

6.1.1 *Sending frequency characteristic*

The sending frequency characteristic is measured according to Recommendation P.64 using the measurement set-up shown in Figures 4/P.66 or 5/P.66, depending on the excitation signal used.

Figure 4/P.66, p.

Figure 5/P.66, p.

6.1.2 *Send loudness rating*

This should be calculated from the sensitivity/frequency characteristic determined in § 6.1.1 by means of Recommendation P.79.

Note — Other methods for calculating loudness ratings used by some Administrations for their own internal planning purposes can be found in Supplement No. 19.

6.1.3 *Distortion*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.1.4 *Noise*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.1.5 *Linearity*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.1.6 *Discrimination against out-of-band input signal*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.2 *Receiving*

6.2.1 *Receiving frequency characteristic*

The receiving frequency characteristic is measured according to Recommendation P.64 using the measurement set-up shown in Figures 6/P.66 or 7/P.66, depending on the excitation signal used.

Figure 6/P.66, p.

Figure 7/P.66, p.

6.2.2 *Receiving loudness rating*

This should be calculated from the sensitivity/frequency characteristic determined in § 6.2.1 by means of Recommendation P.79.

Note — Other methods for calculating loudness rating used by some Administrations for their own internal planning purposes can be found in Supplement No. 19.

6.2.3 *Distortion*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.2.4 *Noise*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.2.5 *Linearity*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.2.6 *Supurious out-of-band signals*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.3 *Sidetone*

Provision should be made for driving the microphone of the telephone set under test as described in § 6.1 and measuring the receiver output as described in § 6.2. The recommended method of measuring sidetone is with the microphone and receiver mounted in the same handset, and using a test fixture which includes the artificial mouth and the artificial ear located relative to each other in accordance with Recommendation P.64.

Note — Care should be taken to avoid mechanical coupling between the artificial mouth and the artificial ear.

6.3.1 *Sidetone frequency characteristic*

6.3.1.1 *Talker sidetone frequency characteristic*

The talker sidetone frequency characteristic is measured according to Recommendation P.64 using the measurement set-up of Figures 8/P.66 or 9/P.66 depending on the excitation signal used.

Figure 8/P.66, p.

Figure 9/P.66, p.

6.3.1.2 *Listener sidetone frequency characteristic*

The listener sidetone frequency characteristic is measured according to Recommendation P.64 using the measurement set-up of Figure 10/P.66.

Figure 10/P.66, p.

6.3.2 *Sidetone masking rating*

This should be calculated from the sensitivity/frequency characteristic determined in § 6.3.1.2 by means of Recommendation P.79.

6.3.3 *Listener sidetone rating*

This should be calculated from the sensitivity/frequency characteristic determined in § 6.3.1.2 by means of Recommendation P.79.

6.4 *Echo return loss*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

6.5 *Delay*

Under study in Question 38/XII, but a method used by several European Administrations can be found in [1].

References

[1] CCITT — Contribution COM XII-No.179, *Transmission aspects for digital telephony* (Norway), Study Period 1985-1988.

Blanc