SECTION 2

SUBSCRIBERS’ LINES AND SETS

Recommendation P.30

TRANSMISSION PERFORMANCE OF GROUP AUDIO TERMINALS (GATs)

(Melbourne, 1988)

1 Introduction

Group Audio Terminals (GATs) are terminals which have been specifically designed to be used by several users.

GATs cover a wide range of products ranging from the hands-free telephone when it is used by several users, to the sophisticated teleconference studio.

The CCITT recommends that GATs satisfy the specifications in this Recommendation.

GATs must also comply with Recommendation P.34 as far as loudness is concerned, when they are connected to the telephone network. If they use voice-activated circuits, Recommendation P.34 may also be applied. Such terminals are sensitive to the acoustics of the location where they are utilized and they may resort to sophisticated acoustical echo processing devices.

The first generation of GATs will operate mainly on 4-wire digital networks and will make use of the wideband (WB) speech coding algorithm specified in Recommendation G.722. Such terminals urgently need specifications that can be based on the present Recommendation.

A typical GAT configuration is represented in Figure 1/P.30.

Such a terminal includes one or several microphones, one or several loudspeakers, sending and receiving amplification. Optionally, it includes a sound-managing and mixing device to the loudspeakers and from the microphones, a coder-decoder for digital networks, a voice-activated gain processing device and an echo processing device.

The location where the GAT is to be used is very important. Several measurements defined in this Recommendation have to be made at the location where the GAT is to be used. These are referred to as ‘‘in situ’’ measurements. They are to be made with the full complement of equipment in the conference room, but with no conferees present.

The present Recommendation is devided into three parts:

— interconnection specifications,

— transmit specifications,

— near-end specifications.

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

Figure 1/P.30, p.

Two test signals are used in this Recommendation:

— an acoustic test signal as defined in Recommendation P.50 (see Note): i.e. an artificial voice as defined in Recommendation P.50 produced by a sound source (an artificial mouth) as described in § 2 of Recommendation P.51 and,

— an electric test signal whose long-term spectrum is identical to the acoustic signal; when applied by a source with a matched internal resistive impedance, it provides a level of —22 dBV.

Both test signals are filtered in the transmission system bandwidth.

Note — The preferred acoustic signal to be used in the measurements for the audio alignment is defined in Recommendation P.50. However, other signals such as speech-shaped noise or pink noise may be used in some applications.

2 Interconnection specifications

These specifications are the basic requirements for a GAT to be connected to a network and to allow communication between several locations.

2.1 *Sending sensitivity*

2.1.1 *Wideband GATs*

For wideband applications, the transmission characteristics of the audio-channel shall be in accordance with Recommendation G.722.

2.1.1.1 *Send side alignment*

The sound source is positioned over the edge of the conference table on the centre line of each conferee's position, as defined in Recommendation P.34 (see Figure 3/P.34), and delivers a signal which complies with Recommendation P.64 [i.e. —4.7 dBPa at the mouth reference point (MRP)].

During the send side alignment the microphones of the GAT shall be positioned on the table as in real use.

The microphone gain controls must be adjusted to achieve, for each position of the source, an output line level of — | 2 (± |) dBV at point X (see Figure 1/P.30), assuming the signal recommended in Recommendation P.50 is used. This value takes account of an 18 dB peak factor of the speech signal and 6 dB for the variations between speakers and the variations due to conferees' movements.

2.1.2 *GATs connected to the public switched telephone network*

Such terminals must comply with Recommendation P.34.

2.2 *Stability test*

The GAT shall have a minimum stability margin of 3 dB when the microphone and loudspeaker paths are looped at reference point X in Figure 1/P.30 and the sound source is activated as described in § 2.1

During the measurement, the volume control shall be in maximum position.

3 Transmit quality specifications

These specifications limit the degradations induced on the network by a GAT.

3.1 *Electro-acoustical specifications*

3.1.1 *Microphone*

The electro-acoustical characteristics of the microhpones should conform to IEC Publication 581-5.

3.1.2 *Octave band measurements*

In situ measurement of the overall transmission frequency response characteristic is recommended. It is defined as the difference between the octave spectra of the electrical signal at the X interface and the acoustic excitation at the MRP. The artificial mouth is positioned as in § 2.1.1.

In order to prevent excessive fluctuations of the frequency response of the system, and since the measurements are performed on site, octave band measurements are recommended in the range 125 Hz to 4 kHz.

The sum of the absolute differences between the measured values and their average should be as low as possible. A practical target of 10 dB is achievable.

3.2 *Echo performance*

3.2.1 *Acoustic echo control*

To get satisfactory suppression of acoustic echoes it is necessary to provide the audio processor with either an echo canceller or an echo suppressor. The echo cancellation technology is recommended if highest possible speech quality performance is aimed at. However, it is recommended always to complement echo cancellation with a mild echo suppression, in order to prevent the undue transmission of room background noises when no talkers are active in the room. This condition should particularly be met in multi-conference environments.

3.2.2 *Echo return loss*

The echo return loss of the audio system shall be measured at reference point X of Figure 2/P.30, with the volume control in maximum position. When the electric test signal, as specified in § 1, is applied to the input port (receive in), the level measured at the output port (send out) shall not be higher than —62 dBV.

An acoustic echo loss of 40 dB includes a margin of 5 dB in order to provide an echo return loss of 35 dB when several GATs are used in a conference situation. This value of 35 dB should be understood as a minimum value. The long-term target value for the acoustic echo loss must be considered as being 45 dB (especially, to take into account the case where a handset is connected to a hands-free terminal). This value is known to prevent any subjective degradations due to delayed acoustic echo [1, 2]. The level measured at reference point X will then be —72 dBV.

Note — The echo canceller shall permit double-talk with negligible speech quality degradation (under study with Question 2/XII).

3.3 *Electrical noise*

The electrical noise emitted by the GAT at the reference point X should be less than —55 dBm, within the transmission bandwidth. No component outside the band should exceed 20 dB above the noise level in the band.

The measurement must be done with no conferees in the room and without incoming signals on the receiving side of the equipment in order not to activate the microphone circuits.

The noise emitted by the GAT at the reference point X when the microphones are active should be no more than —50 dBm. It must be measured by forcing the system into the emission mode as if one speaker were active in the room.

3.4 *Reverberated field picked up by the microphone*

For this measurement, the sound source is positioned in order that the distances between the sound source and all the microphones greater than three times the distance between the microphone and the position defined for the send side alignment. It is also recommended that the source be, at least, one meter from the walls. Then the signal measured at point X shall be not more than —29 dBV (this accounts for a direct-field over reverberated-field ratio of 6 dB [3]). It must be measured by forcing the system into the emission mode as if one speaker were active in the room. The test must be performed for each microphone in the room.

Basic requirements for the choice of the conference room, for its acoustical treatment and for the positioning of microphones and loudspeakers can be found in Supplement No. 16.

4 **Near-end quality specifications**

This part of the Recommendation tests the minimum specifications intended for the local users.

4.1 *Electro-acoustical specifications*

4.1.1 *Louspeakers*

The electro-acoustical characteristics of the loudspeakers should conform to IEC Publication 581-7.

4.1.2 *Octave band mesurements*

In-situ measurement of the overall reception frequency response characteristics is recommended. It is defined as the difference between the octave spectra of the acoustic signal delivered by the loudspeaker(s) at the listening positions and the input electric signal at the X interface.

The sum of the absolute differences between the measured values and their average should be as low as possible. A practical value of 12 dB is achievable.

4.2 *Receiving sensitivity*

4.2.1 *Volume control*

The audio conference terminal shall be provided with a volume control. The gain at maximum position should conform to § 4.2.2. The volume control should ideally be linked to the echo control mechanism.

4.2.2 *Receiving side alignment*

4.2.2.1 *Wideband GATs*

The electrical test signal is connected to the input port of the system. The receiving gain shall be adjusted in order to reach a sound pressure level of at least 65 dB and 20 dB above the acoustical noise level at the MRP. The alignment procedure should be performed with the volume control in the maximum position.

4.2.2.2 *GATs connected to the analogue public switched telephone network*

Such terminals must commply with Recommendation P.34.

References

[1] CCITT — Contribution COM XII-No. 170, Study Period 1985-1988

[2] CCITT — Contribution COM XII-No. 171, Study Period 1985-1988

[3] CCITT — Contribution COM XII-No. 172, Study Period 1985-1988

TRANSMISSION CHARACTERISTICS FOR DIGITAL TELEPHONES

(Melbourne, 1988)

This Recommendation deals with sending and receiving loudness ratings, sidetone masking rating, listener sidetone rating, and sending and receiving sensitivity/frequency characteristics. Other important characteristics are still under study.

1 Sending loudness rating (SLR) and receiving loudness ratings (RLR)

In view of Recommendation G.111, § 3.2, the following values are recommended:

— as a short-term objective, nominal values of SLR in the range 5 to 11 dB and nominal values of RLR in the range —1 to 5 dB;

— as a long-term objective, the following nominal values for SLR, 8 dB and for RLR, 2 dB.

Note 1 — The recommended values for SLR and RLR do not imply that echo control in the network can always be avoided.

Note 2 — The acoustic loss in the telephone set is an important factor in the echo path and will need careful consideration. A volume control in the telephone set will decrease the echo loss by the same amount as the gain is raised.

Note 3 — For digital telephones connected to a digital PABX (to which analogue telephones may also be connected), values at the lower end of the ranges above might be needed. The reason is to give customers the same receiving level as they are used to having with the analogue telephones. A receiving volume control might be considered.

2 Sidetone masking rating (STMR) and listener sidetone rating (LSTR)

In view of the following considerations:

— the optimum STMR for conditions free from echo;

— the sidetone masking effect on talker echo at short delays;

— the difficulties of high ambient noise conditions;

— what subscribers are used to having with present analogue sets,

the following values are recommended :

The specifications given here are subject to future enhancement and therefore should be regarded as provisional

— nominal values of STMR in the range 10 to 15 dB;

— nominal values of LSTR >15 dB,

(No maximum values for LSTR need to be imposed.)

Note — These values may be modified when information becomes available on the effects of short delay echo (e.g. 10 ms).

3 Sending and receiving sensitivity frequency characteristics for digital telephones

In view of the following considerations:

— the compatibility with analogue telephones in a mixed analogue digital network;

— the absence of line-length-dependent frequency distortion to be compensated for as with analogue telephones;

— the aim to achieve the best possible overall quality with the digital telephone, sending and receiving sensitivity/frequency characteristics as specified below are recommended:

— a substantially flat receiving frequency response $S_{J\backslash \mathrm{d}E}$ between 300 Hz and 3 | 00 Hz should be chosen;

— a nominal sending frequency response $S_{M\backslash \mathrm{d}J}$ rising with a slope within the area indicated in Figure 1/P.31 should be striven for;

— below 200 Hz, the send slope should fall by at least 6 dB/octave.

Note 1 — $S_{J\backslash \mathrm{d}E}$ and $S_{M\backslash \mathrm{d}J}$ are normally estimated from measurements of $S_{J\backslash \mathrm{d}e}$ and $S_{m\backslash \mathrm{d}J}$ according to Recommendation P.66.

Note 2 — An expansion of the lower frequency range to 200 Hz will increase the naturalness of the speech.

Note 3 — The normal considerations for anti-aliasing filters must be applied to the frequency responses.

Note 4 — Marked peaks in the responses might cause stability problems and should therefore be avoided.

Note 5 — The preferred curves for $S_{J\backslash \mathrm{d}E}$ and $S_{M\backslash \mathrm{d}J}$ defined in this way should be considered as a design objective. Individual microphone and receiver curves will, for several reasons, deviate more or less from the ‘‘ideal’’ curves. However, it is hardly possible to specify in a Recommendation concerning desirable frequency characteristics how much and in which way individual response curves may deviate from the target curve, without becoming unacceptable. For type approval of telephone sets, it is generally necessary to specify limits for the shape of sending and receiving frequency curves nationally, in the same way as tolerance limits for loudness ratings are usually specified. These limits are based on technical considerations as well as on cost of implementation, manufacturing tolerances and other economic factors.

Figure 1/P.31, p.

Recommendation P.32

EVALUATION OF THE EFFICIENCY OF TELEPHONE BOOTHS AND ACOUSTING HOODS

(Melbourne, 1988)

The purpose of this Recommendation is to define the methods of measurement to evaluate the efficiency of either acoustic hoods or telephone booths intended to improve the quality of telephone transmission in noisy environments. In addition to the improvement of the transmission quality during a conversation between two users, this Recommendation takes into consideration the need to guarantee speech privacy for the user speaking from the acoustic hood or the booth with respect to a listener situated on the outside of the telephone booth.

1 Evaluation methods

The efficiency of a telephone booth or an acoustic hood can be evaluated using either subjective or objective measurements.

The objective measurements suitable for that purpose are those based on the acoustic insulation (for example in a reverberation chamber) resulting from the difference between the sound levels registered inside and outside the telephone booth or vice-versa. As the acoustic characteristics vary outside and inside the telephone booth, the acoustic insulation obtained in each case (noise source outside or inside the telephone booth) is not the same. In addition, if we consider open telephone booths or acoustic hoods, the measurement of the acoustic insulation gives results not correlatable to the ones obtained by means of a subjective evaluation of the booth performance.

A subjective measurement of the efficiency of booths or acoustic hoods consists in determining the intelligibility index inside the booth, in conditions of external noise (room noise, road noise, etc.). This measurement can also be objectively obtained by calculating the articulation index, by using, for example, the Kryter method as indicated in Annex A.

Another method used to measure subjectively the efficiency of booths and acoustic hoods consists in evaluating the intelligibility threshold variation observed between the intelligibility inside and outside the booth placed in a noisy ambient.

The performance of booths and acoustic hoods, related to the user's privacy while speaking inside the booth, can be subjectively evaluated by measuring the intelligibility of the coversation from the inside to the outside of the booth or by using an objective measurement such as calculating the articulation index (according to Kryter's method for example) outside the booth under specific noise conditions.

Since the intelligibility inside the booth is also a function of the sidetone of the telephone set used, a simple measurement of acoustic insulation which does not take into consideration the intelligibility reduction caused by the sidetone cannot furnish correct evaluations on the improvement of transmission quality due to telephone booths, or acoustic hoods.

Bearing in mind the following observations:

1) international telephone communication can be originated from telephone sets installed in noisy ambients and protected by booths or acoustic hoods;

2) there are no measurement methods recommended for evaluating the transmission quality improvement resulting from the use of the telephone booth;

3) an evaluation of the booth efficiency, based only on the acoustic insulation obtained by traditional methods (acoustic attenuation of the panels of the booth) is not always correlated to the subjective evaluation of the booth performance;

4) subjective measurements either of the intelligibility or of the intelligibility threshold variation give the possibility of evaluating the efficienty of a booth, but are time-consuming and expensive and also require a qualified and well-trained operator team;

5) there are no recommendations giving criteria relating the employment of the booths to the ambient noise level, in order to determine an acceptable quality of transmission,

methods of measurement as specified below are recommended :

a) evaluating the efficiency of telephone booths and acoustic hoods taking into consideration the intelligibility index , obtained from a listener inside the booth with the external ambient noise having a certain acoustic spectrum;

b) calculating the intelligibility index inside the telephone booth or the acoustic hood by means of the objective method defined in § 3, taking into consideration the acoustic attenuation of the booth and the sidetone of the telephone set used. This objective method allows a rapid evaluation of the booth performance, sufficiently precise for practical purposes;

c) considering the logatom intelligbility as an evaluation criterion related to the booth performance, calculated by means of the articulation index (AI) intelligibility is language-dependent and it shall be performed with the appropriate relation;

d) evaluating the booth and the acoustic hood at the conditions of utilization, that is, when a user is speaking from the inside using a telephone set with a determined sidetone and with an external ambient noise having an average intensity level and a certain acoustic spectrum, both already known.

2 Definition and descriptions of parameters of calculation

Telephone conversations taking place in conditions of ambient noise are affected by ambient noise through three different paths:

1) acoustic noise (N_a) at the ear which is not engaged in the telephone call;

2) acoustic noise (N_b) at the ear which is engaged in the telephone call, determined by the acoustic leak between ear and handset;

3) noise picked up by the microphone and directed by sidetone (N_s) to the ear which is engaged in the conversation.

The acoustic noise flowing through the acoustic leak between ear and handset has a spectrum which changes as a function of the pressure of the handset against the ear. To evaluate the performance of booths, the acoustic attenuation ($L_{R\backslash dN\backslash dE}$) of this path can be taken into consideration.

The noise N_s is due to sidetone changes according to the telephone set used and it generally has a spectrum which is different from that of N_b. In spite of their mutual correlation, the power summation of the respective spectra seems the best estimate of the global noise (N_g) which affects the ear engaged in the conversation.

In addition, the noises at the two ears (N_a, N_b) are generally different, both in level and in spectrum; experimental intelligibility measurements [1] [2] have demonstrated that this disturbing effect can be evaluated by subtracting 10 dB from the noise level (N_a) at the free ear.

The aforesaid experiment measurements have also shown that the *total* equivalent noise N_T to be used in intelligibility calculations is given by the amplitude sum of noise spectra at the two ears. Consequently, the total equivalent noise N_T is given from the relation:

The sidetone noise N_s is a function of the mouth-to-ear sidetone loss $L_{M\backslash dE\backslash dS\backslash dT}$ and it should be measured at the actual noise level, typically 65 dB SPL, under diffuse field conditions. This is particularly important in the case of telephone sets with carbon microphones or of electronic telephone sets with automatic gain control or provided with noise cancelling microphones.

Documentation about the specifications in this Recommendation is not yet sufficient to confirm their validity, thus they are subject to future enhancement and should be regarded as provisional.

3 Calculation of the booth or acoustic hood efficiency

Given a particular telephone booth or an acoustic hood, the following procedure shall be followed for determining the articulation index in actual operating conditions.

Calculate:

a) the noise spectrum N_a inside the booth by subtracting the acoustic attenuation of the booth (L_a) from the external noise spectrum (N_e). The attenuation should be measured in third octave bands, with a person inside the booth (or a baffle providing an equivalent acoustic absorption) and in a diffuse field condition;

b) the spectrum of the noise N_b by subtracting the leakage attenuation of the handset ($L_{R\backslash \mathrm{d}N\backslash \mathrm{d}E}$) from the noise spectrum inside the booth N_a;

c) the sidetone noise spectrum N_s by subtracting the acoustic sidetone attenuation L_{RNST} from the noise spectrum inside the booth N_a;

d) the spectrum of global noise N_g at the ear pressed against the handset as the power sum of N_s and N_b;

e) the spectrum of total equivalent noise N_T as the amplitude sum of noises at both ears, after having subtracted 10 dB from the noise spectrum at the ear not engaged;

f) the articulation index, AI by Kryter’s method [3], assuming a listening speech level of 70 dBA, a value corresponding to the limit of the attenuation of the line loss distribution.

An example of application of the calculation method is shown in Appendix I.

4 Efficiency limits of booths and acoustic hoods

Efficiency of booths or acoustic hoods can be considered satisfactory if an AI equal to 0.6 is guaranteed.

This value corresponds for most languages to a logatom intelligibility of 80% inside the booth, according to the results of French and Steinberg [4], in Figure 1/P.32. It can be assumed as the minimum acceptable limit of performance, corresponding to the maximum external noise level that the booth can withstand in order to guarantee a good quality of telephone transmission inside the booth.

Therefore, each booth can simply be classified by specifying a *maximum external noise level* (MENL), which is the level that gives AI = 0.6.

The MENL that classifies the telephone booth shall be determined by repeating the calculation of the AI, as is indicated in § 3, with different levels of external noise. By means of the curve representing the values of the AI as a function of the outside noise level, the MENL corresponding to an AI = 0.6 can be determined. This MENL depends not only on the acoustic attenuation of the booth or acoustic hood, but also on the received speech level which is assumed to have a reference value of 70 dBA, and on the sidetone performances of the telephone set which should be measured at a proper sound pressure level, (about 65 dB SPL) and in free field conditions.

Figure 1/P.32, p.

It is important to determine the room noise sidetone sensitivity L_{RNST} which makes use of a diffuse room noise source within the booth. It may also be necessary to include within the booth a manikin to simulate the presence of a subscriber.

5 Speech privacy of telephone communications

The booth can also guarantee speech privacy of conversation by reducing the vocal signals radiated towards outside in order to make them unintelligible. Applying Kryter's calculation method of the articulation index of the speech signals transmitted through the booth to the external ambient at a predetermined noise level, the distance at which the logatom intelligibility or AI falls to a pre-determined value (for example, AI = 0.3) can be estimated. This method can be used to determine the curves of equal intelligibility (isophenes) in any direction, increasing distance from the booth.

Note — The quality improvement of the conversation for the subscriber at the other end of the telephone connection, during a call with a telephone in a booth or acoustic hood has not yet been studied. The evaluation of this aspect is required in any case to consider a number of other factors such as the natural increase of speech loudness in noisy environments and the effective signal-to-noise ratio of transmitted signals.

ANNEX A

(to Recommendation P.32)

Example of efficiency calculation of a telephone booth

The articulation index (AI) is calculated according to Kryter's method.

The acoustic attenuation of a telephone booth measured in an echo chamber at each one-third octave band is reported in Table A-1/P.32, column 2. The total noise level outside the booth is 80 dBA and the sound level of the noise at each centre frequency band is indicated in column 3. The sidetone response characteristics ($L_{R\backslash \mathrm{d}N\backslash \mathrm{d}S\backslash \mathrm{d}T}$) of the telephone set used inside the booth is given in column 4.

The noise level inside the booth at each centre frequency band (N_a) is obtained by subtracting column 2 from column 3 (column 5). It is supposed that the handset of the telephone instrument used in the booth has the acoustic attenuation indicated in Figure A-1/P.32 and reported in column 6.

Figure A-1/P.32, p.

The values of the noise (N_b) due to acoustic leakage between ear and handset obtained by subtracting column 6 from column 5 are reported in column 7.

The values, at each frequency band, of the sidetone noise (N_s) obtained by subtracting column 4 from column 5 are reported in column 8. The global noise at engaged ear (N_g) is reported in column 9 as the power sum of the levels indicated in columns 8 and 7. The total equivalent noise is obtained by adding the levels of column 9 to the values of column 5 reduced by 10 db (column 10). The speech spectrum (β') is reported in column 11 and the signal-to-noise ratio corrected by 12 dB (considering the peaks of the speech signal) is indicated, at each one-third octave band, in

column 12. Kryter's coefficients are indicated for each one-third octave band in column 13.

The articulation index (AI) is obtained by multiplying the values of column 12 by those of column 13 and adding the results. By repeating the calculation with other external noise levels, it is possible to draw the diagram of the AI as a function of external noise levels for the considered booth, as shown in Figure A-2/P.32. It can be seen that this booth is designed for withstanding a maximum external noise of about 77 dBA which is the MENL value that classifies the booth.

Figure A-2/P.32, p.

H.T. [T1.32]

TABLE A-1/P.32

{
Central frequency one-third octave band
}
Acoustic attenuation of the booth, *L*
}
Acoustic sidetone attenuation, L
}
Noise inside the booth, N
}
Acoustic attenuation of handset, L
}
Noise due to acoustic leakage, N
}
Global noise at engaged ear, N
}
Total equivalent noise, N
}
$\frac{ignal+12\ dB}{oise}$
}
(Hz)
	{
	External noise, N
	{
	{
	{
	Sidetone noise, N
	{
	Speech spectrum, β‘
	Kryter's coefficient
	(dB)
		{
		{
		{
		Products (13)×(12)
		(dB SPL)
			(dB)	*(dB SPL)*	*(dB)*	*(dB SPL)*	*(dB SPL)*

(1)	*(2)*	*(3)*	*(4)*	*(5) =(3)-(2)*	*(6)*	*(7) =(5)-(6)*	*(8) =(5)-(4*
200	*10*	*77.5*	*12*	*67.5*	*3*.**0**	64.5	55.5
250	13	76.5	12	63.5	4.**0**	59.5	51.5
315	13	73.5	11	60.5	5.**0**	55.5	49.5
400	15	74.0	9	59.0	6.**0**	53.5	50.0
500	14	72.5	9	58.5	7.**0**	51.5	49.5
630	14	72.0	10	58.0	8.5	49.5	48.0
800	16	72.0	12	56.0	10.0	46.0	44.0
1000	15	71.0	12	56.0	11.5	44.5	44.0
1250	15	69.5	9	54.5	13.0	41.5	45.5
1600	15	68.0	9	53.0	14.5	38.5	44.0
2000	11	66.0	8	55.0	16.0	39.0	47.0
2500	11	64.0	**.5** 10.5	53.0	17.5	35.5	42.5
3150	12	62.0	14	50.0	19.0	31.0	36.0
4000	12	61.5	14	49.5	20.5	29.0	35.5
TOTAL (dBA) AI = 0.52 SPL Sound pressure level }		80.0		66.3			

Tableau A-1/P.32 [T1.32], p.

References

[1] CCITT Contribution — COM XII-No. 122, (France), Study period 1981-1984.

[2] CCITT Document — Annex 2, AP VII-No. 115.

[3] KRYTER, (K.): Methods for the calculation and use of Articulation Index, *J.A.S.A.* Vol. 34, 1962.

[4] FRENCH, (N. | .) and STEINBERG (J. | .): Factors governing the intelligibility of speech sounds, *J.A.S.A.* Vol. 19, 1947.

Bibliography

CCITT Contribution — COM XII-No. 139, (Italy), Study period 1973-1976.

CCITT Contribution — COM XII-No. 130, (Norway), Study period 1977-1980.

CCITT Contribution TD 26, (Sweden), WP Laboratory (Geneva, 17-19 January 1984)

KRYTER (K.): The effects of noise on man, *Academic Press* , pp. 70-77, 1970.

Recommendation P.33

SUBSCRIBER TELEPHONE SETS CONTAINING EITHER LOUDSPEAKING RECEIVERS OR

MICROPHONES ASSOCIATED WITH AMPLIFIERS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1980)

The CCITT,

considering

(a) that an increasing number of loudspeaker sets is being used in the telephone network,

(b) the complex nature of factors introduced by this equipment and affecting telephone transmission performance,

(c) the need to help Administrations to determine the conditions in which the use of such equipment may be authorized in telephone networks,

makes the following recommendation:

(1) In order to avoid overload of carrier systems, the mean long-term power of speech currents should not exceed the mean absolute power level assumed for system design. In Recommendation G.223 [1] the value adopted for this mean power level is — 15 dBm0 (mean power = 31.6 microwatts). Loudspeaker telephones having a sending sensitivity that complies with Recommendation P.34 can be assumed to fulfil this Recommendation. Furthermore, in order to avoid excessive crosstalk from high-level speech currents and/or inadequate received volume from low-level speech currents, care should be taken to ensure that the variation of speech currents is not substantially greater than that from modern handset telephones.

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

Reference

[1] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony* , Vol. III, Rec. G.223.

Recommendation P.34

TRANSMISSION CHARACTERISTICS

OF HANDS-FREE TELEPHONES

(Melbourne, 1988)

1 Introduction

The sending and receiving sensitivities of handset telephones, normally expressed as Loudness Rating (LR) values, are used in most countries in connection with their national transmission plan for the design of the national network.

However, since it is possible to fulfil Recommendations such as G.121 by distributing LR values between the telephone sets and the network in different ways, it is not possible to issue an international Recommendation stating LR values of telephone sets alone — whether these are handset or hands-free telephones.

On the other hand, it is possible to recommend sensitivity values for hands-free telephones (HFTs) relative to the standard handset telephone used nationally. The object of such Recommendations should be to obtain equivalent performance with both types of telephones, at least with respect to send and receive loudness. This means that the average user's behaviour and preferences while talking and listening must be taken into account. The relative sensitivities defined in §§ 2 and 3 are derived from performance tests aimed at fulfilling this requirement.

Other important features contributing to the quality of telephone calls made from hands-free telephones cannot presently be dealt with by existing Recommendations and are studied within Question 2/XII [1].

For loudspeaking telephones (see Recommendation P.10) which do not provide full hands-free operation, the relevant parts of this Recommendation may be referred to.

2 Sending sensitivity

The sending LR (SLR) of an HFT should be about 5 dB worse

(i.e. higher) than the SLR of the corresponding handset telephone (the actual value will depend on the type of handset used).

Note — Conversation tests in several countries have shown that comparable speech voltages are obtained on the line when the sending loudness rating of the HFT is 5 dB higher than that of the handset telephone used.

The difference of 5 dB has several components:

a) the average talking level for HFTs, which is about 3 dB higher than for handsets;

b) the output level from a handset telephone in conversational use, which is about 1-2 dB lower than what is obtained in the speaking position specified for loudness ratings measurements;

c) other minor differences such as different frequency response curves.

If the sending sensitivity is controlled by the room noise level, this control should be designed to compensate the expected rise of the talking level with room noise.

It should not be possible for the user to adjust the sending sensitivity.

3 Receiving sensitivity

The receiving sensitivity of a hands-free telephone without automatic gain control should be adjustable within a range of 15 to 30 dB. This range should span the value of the receiving loudness rating (RLR) which is equal to that of the corresponding handset telephone, as well as a RLR value about 10 dB better.

Note 1 — Every precaution should be taken to ensure that the increase in gain due to the volume control does not allow the overhearing of other telephone conversations due to crosstalk.

Note 2 — In principle, the RLR of the HFT should be equal to the RLR of the corresponding handset telephone in a quiet room. The range of room noise levels met in normal office use necessitates, however, an additional gain of at least 10 dB.

For hands-free telephones equipped with an automatic gain control for the receive level (the gain being controlled by the incoming speech voltage), loudness ratings may not be applicable. In this case, the HFT should be designed so that the listening level at the maximum line length for which the HFT is intended to be used can be preset to a value that may be considered as the best compromise between the levels required for listening in quiet and noisy rooms.

Note 3 — The preferred listening level depends on the room noise level as well as on other external conditions. There is, furthermore, a great variance between individual listeners.

The average preferred level for listening only appears to be a sound pressure level of about 65 dB for 45 dBA room noise, or 70 dB for 55 dBA room noise. However, to obtain maximum Mean Opinion Scores in conversation tests, listening levels of about 5 to 10 dB higher may be required.

4 Frequency response curves

4.1 *Sending*

Available information indicates that the optimum slope of the sending response curve when measured with the HFT on a table lies between 0 and +3 dB/octave, if the receiving response curve is flat.

Only under highly reverberant conditions may a somewhat higher preemphasis increase the intelligibility. Therefore, if a frequency compensation for the probable cable attenuation is included, the sending curve should not rise with frequency by more than 2-3 dB/octave.

Below 300 Hz there should be a gradual roll-off. The slope may be steeper below 200 Hz.

Note — The interval 200-300 Hz makes a significant contribution to the naturalness of the transmitted speech and should therefore be included in the transmission band of the HFT.

Above 4000 Hz, a roll-off by at least —6 dB/octave (preferably —12 dB/octave) is appropriate in order to avoid interference by crosstalk to adjacent channels in certain types of long-distance circuits.

4.2 *Receiving*

The receiving response curve should be substantially flat in the frequency range of 200-4000 Hz.

The requirement refers to the sound pressure in the undisturbed field at the listener's position with a set-up including the table as described in § 6.

5 Switching characteristics

Most loudspeaker and hands-free telephones contain voice-switched circuits whose main function is to avoid singing through acoustic feedback. Such circuits insert a loss in either the sending or receiving direction in various ways. Switching from one direction to the other occurs when a signal above a given threshold is applied from the opposite direction, or when the control circuit, taking into account the relative levels and the nature of the signals in both directions, allows the switching.

The fundamental voice-switching parameters of the switching function are defined as follows (see Figures 1/P.34 and 2/P.34):

— threshold level $V_{T\backslash dH}$: minimum necessary signal level for removing insertion loss,

— build-up time T_R: time from the input signal going above the threshold level up to 50% of the complete removal of the insertion loss,

— hang-over time T_H: time from the input signal going below the threshold level up to the insertion of 50% of the switched loss,

— switching time T_S: time from one transmission direction to the other (see Figure 2/P.34).

Figure 1/P.34, p.

Figure 2/P.34, p.

By a suitable choice of parameter values, the degradation of speech quality that is introduced by voice switching can be made negligible, while an inadequate choice of parameter values, switching times in particular, may lead to serious clipping effects and loss of initial or final consonants in the transmitted speech.

Measurements of voice switching characteristics may be divided into those dealing with:

a) characteristics for alternate conversation, in which two parties communicate by alternating speech spurts without interrupting each other. In this case, it may be assumed that the voice switch circuit returns to an idle state before being activated by an input signal in either direction;

b) characteristics for simultaneous conversation, in which both parties may interrupt each other by simultaneous talk, or where speech at one end of a connection is present simultaneously with noise at the other end.

The first case is of fundamental importance, as its characteristics also affect simultaneous conversation characteristics, and hands-free telephones should therefore always be checked in that respect.

A suitable signal for measuring the characteristics for case a) above consists of a periodic tone burst signal (see Figure 1/P.34). The on/off times T1/T2 and the amplitudes H and L should be adjustable. For case b), in order to switch alternately the hands-free telephone from sending to receiving state, the use of two out-of-phase tone burst sequences is recommended (for instance acoustical 1 kHz, electrical 400 Hz). Switching characteristics measured this way will probably be more readily used in the analysis of subjective conversation tests results.

There are three types of hands-free sets under consideration:

5.1 *Type 1* — Hands-free telephone sets for which switching occurs when an absolute level $V_{T\backslash dH}$ is reached

In general, it is desirable to keep the threshold value low, the build-up time short and the hang-over time long. On the other hand, in practical applications extremely short build-up times (about a few milliseconds) may cause the voice-switching circuit to be operated by impulsive noises, while very long hang-over times are likely to impede the natural switch-over in conversation. Furthermore, if the threshold level is more than 25 dB below the active speech level, the voice-switching circuit will be activated too easily by ambient noise.

The following switching characteristics are recommended:

a) The build-up time T_R should be less than 15 ms, preferably below 10 ms.

b) The hang-over time T_H should be greater than 100 ms. If the threshold level is in the preferred range, values of T_H between 150 and 250 ms are recommended. Hang-over times greater than 400 ms do not improve the performance noticeably.

c) The threshold level $V_{T\backslash dH}$ should be at least 20 dB below the active speech level. Levels between —20 and —15 dB may be used if the hang-over time is greater than 300 ms. Levels above —15 dB should not be used.

In order to measure $V_{T\backslash dH}$, the amplitude is gradually increased from a low level until switching occurs. By doing this, an absolute threshold value is obtained. Generally, the threshold is expressed as the difference between this value and the average r.m.s. speech voltage present in the active state.

5.2 *Type 2* — Hands-free telephone sets for which switching depends on the relative levels in both transmission directions, and also in some cases on noise levels (acoustical and electrical), amplifiers gains, automatic gain controls, previous transmission direction, etc.

The following values are recommended:

a) T_R should be less than 15 ms, preferably below 10 ms,

b) T_H can be less than 50 ms,

c) T_S is recommended to be approximately 100 ms and is measured by using 2 excitation signals (see Input 1 and Input 2 in Figure 2/P.34).

Note — Under highly reverberant conditions, some hands-free sets with such a T_S may operate in an unsatisfactory way.

More information about measuring levels and methods can be found in the Handbook on Telephonometry in § 3.5.

5.3 *Type 3* — Hands-free sets using echo cancellation techniques

Some indications about the evaluation of sets using echo cancellation are given in Recommendation P.30.

Note — For loudspeaking telephone sets, an insertion loss may be introduced in the receiving side to avoid the acoustical coupling with the handset microphone. This insertion loss may be introduced when the received level on the loudspeaker is too high, or when the signal from the handset microphone is sent onto the loudspeaker at too high a level.

It is recommended that the delay of application and withdrawal of this insertion loss be limited to 20 ms and its value limited to avoid any clipping effect on the received speech.

6 Conditions of measurement

For both subjective and objective measurements, physical test arrangements as described in this section should be used.

6.1 *Test table*

During the measurements, the HFT is placed on a table defined as follows:

The surface of the table should be hard (e.g. polished marine plywood or suitable hardwood), flat, rigid and horizontal to provide a sound-reflecting surface on which the HFT being tested rests. The dimensions of the table should be such that the surface area is about 1 m^2 but not less than 0.96 m^2 and the width not less than 800 mm [2].

Note — This arrangement should be used for all measurements, including the recording of frequency responses, although diffraction effects due to the table are likely to cause severe dips or peaks in the response curve (see § 6.5.2).

6.2 *Test arrangements*

The physical test arrangements of one- and two-piece HFTs [3] for subjective and objective measurements is shown in Figure 3/P.34.

If the projections of the housing are not rectangular, the point B is positioned at the crossing of the centre line through the housing and the outline of the vertical projection of the housing.

The edge of the front of the box should be perpendicular to the line A-B.

FIGURE 3/P.34, p.

6.3 *Test environment*

When performing tests, the room acoustics must not have a dominating influence. It is recommended for objective measurements that the test environment be practically free-field (anechoic) down to a lowest frequency of 175 Hz, and be such that the test arrangement lies totally within the free-field volume.

Note — Satisfactory free-field conditions may be considered to exist where errors due to the departure from ideal conditions do not exceed ± | dB.

The tests should be made in an environment where the ambient noise level is negligible. For objective measurements this is achieved if the Noise Rating (NR) of the Noise Criterion (NC) is lower than 15 [4, 5]. For subjective tests, it may be sufficient to keep the sound level of ambient noise below 35 dBA.

6.4 *Subjective determinations*

Loudness rating should be determined in accordance with Recommendation P.78.

Note — Some information about reference equivalents can be found in the CCITT *Red Book* | (Vol. V 1985), or in the Handbook on Telephonometry.

6.4.1 *Sending*

The talking level for the measurement of sending loudness rating (SLR) of an HFT should normally be the same as specified for measurements on handset telephones.

It is not necessary for the talker during the test to shift between the reference microphone guard-ring and the guard-ring positioned relative to the HFT if the obstacle effect of the reference microphone can be assumed to be negligible.

Normally the specified talking level and the use of a conventional test phrase or sentence should be sufficient to ensure that a voice-switched HFT will be in the sending condition during the determination of SLR. If this is not the case the talking level may be increased by up to 5 dB, which may be compensated in the reference system to preserve the same listening level.

If the sending sensitivity is controlled by the room noise level the subjective measurement should be done in a quiet environment (< 35 dBA). Further information about the HFT performance may then be estimated by repeating the sending measurements with increasing levels of room noise, up to a maximum of 60 dBA.

6.4.2 *Receiving*

The talking level at the reference microphone for the measurement of RR25E or RLR should normally be the same as specified for the measurement of handset telephones. This should normally ensure that when loudness balance is achieved between the reference system and the test system path, a signal of sufficient magnitude is present at the HFT to switch it into the receive condition.

Problems can sometimes occur when approaching the balance condition from the condition of high attenuation in the balance attenuators, when the low level input signal may fail to switch the HFT into the receiving condition. If this does occur the talking level may be increased by up to 5 dB in order to minimize the difference in loudness.

Note — The listening level will thus also increase at balance, but in this case it will not be possible to correct it by changing the reference system attenuator.

Obtaining the loudness balance for the receiving condition may be facilitated by use of a loudspeaking intermediate reference system. The specification of such a system is, however, outside the scope of this Recommendation.

6.5 *Objective evaluations*

Objective evaluations of loudspeaker and hands-free telephones concern:

— the sending and receiving frequency sensitivity curves measurements,

— the objective determination of loudness ratings according to the method described in 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.

6.5.1 *Sensitivity measurements*

6.5.1.1 *Sending sensitivity measurements*

The sending response curves of a looudspeaker and/or hands-free telephone is recorded at the output terminals of the thetelephone with the same electrical connections as for the handset telephones. The acoustical input to the telephone microphone is supplied from an artificial mouth in the position shown in Figure 3/P.34.

In such a case, the sending sensitivity of the local telephone system is expressed as follows:

$$S_{mJ} = 20 \log_{10} \frac{V_s(f)}{p_m(f)} \quad \text{dB rel } 1 \text{ V/Pa}$$

where V_s is the voltage across a 600 ohm termination and P_m is the sound pressure at the MRP.

The measuring level proposed in Recommendation P.64 may be used: —4.7 dBPa at the MRP (Figure 3/P.34), which corresponds to —29 dBPa at 50 cm from the lip when there is no table nor set.

Note 1 — Some HFTs use ‘‘noise guard’’ circuits and therefore the source signal has to be modified. A suitable method is to pulse the source signal at an adequate rate, e.g. 250 ms ‘‘ON’’ and 150 ms ‘‘OFF’’. Sending sensitivities determined for HFTs in this way are not suitable for use in calculating send loudness ratings (SLR). For this purpose, the reference sound pressure should have a level at the MRP which is (on average, over the frequency range of interest) 24.2 dB higher.

6.5.1.2 *Receiving frequency sensitivity response measurements*

The receiving sensitivity of a loudspeaker and/or hands-free telephone is expressed as follows:

$$S_{Je} = 20 \log_{10} \frac{p_R(f)}{\frac{1}{2} E_J(f)} \quad \text{dB rel } 1 \text{ Pa/V}$$

where p_R is the sound pressure at point C in Figure 3/P.34 and E_J is the e.m.f in the 600 ohms source.

6.5.2 *Measure and computation of loudness ratings*

6.5.2.1 *Sending*

The computation of the sending loudness rating may be performed according to Recommendation P.79 by using the frequency sensitivity response measured between the electrical output of the set and the acoustical sound pressure at the MRP (Figure 3/P.34).

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

However, some care must be taken in the test design and the interpretation of the results. Results available up to now concern only a limited number of sets and the measuring signal is of some importance. Under some conditions, an artificial speech-like signal may activate the noise-guard circuits (by inserting some loss at the sending side). Better results are expected by using an artificial voice satisfying Recommendation P.50 (temporal characteristics of the signal closer to those of real speech).

6.5.2.2 *Receiving*

Objective measurements described in § 6.5.1.2 are made with a free-field microphone at point C (Figure 3/P.34).

Loudness Ratings are computed following Recommendation P.79, provided the following phenomena are taken into account:

— the diffraction effect of the listener head,

— an appropriate correction for the difference between one-ear and two-ears listening.

These subjects are still under study under Question 2/XII.

Provisionally, a correction term of 14 dB should be subtracted in the computed loudness ratings.

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

References

[1] CCITT — Question 2/XII Contribution COM XII-No. 1, Study Period 1989-1992.

[2] European Committee for Standardization (CEN) *Office chair/desk working position — dimensions and design requirements* , CEN: prEN91/August 1981.

[3] CCITT *A method for measuring the sensitivity of a loudspeaking telephone set* , Annex 2 to Question 17/XII, White Book, Vol. V, ITU, Geneva, 1969.

[4] ISO *Assessment of noise with respect to community response* , ISO Recommendation 1996, 1971.

[5] BERANEK (L. |): Noise and Vibration Control, *McGraw Hill* , pp. 564-566, New York, 1971.

Recommendation P.35

HANDSET TELEPHONES

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

1 Transmission characteristics

The transmission plan for international telephone connections is given in Recommendation G.101.

Recommendations G.111 and G.121 deal with the transmission quality, i.e. loudness ratings for international and national telephone connections, respectively.

These Recommendations permit administrations to split the requirements between analogue telephone sets and the other parts of the network as long as the overall specifications are fulfilled.

Therefore no precise specifications can be given for analogue telephone sets, although some design considerations can be provided. The latter are contained in Supplement No. 10.

Recommendations for digital telephones are found in Recommendation P.31.

2 Handset dimensions

The shape and the dimension of the handset have an important influence on both send and receive levels. The earpiece must be capable of forming a good seal to the ear and the handgrip of the handset must be such that it will encourage the user to hold it to the head in the optimum position.

Reference [1] is an ergonomic study which presents data on the distribution of the relevant finger and head dimensions.

A later head dimension study carried out in the People’s Republic of China is reported in [2]. A subsequent investigation [3] shows that, for convenience in use, the mouthpiece of the handset should be somewhat outside (e.g. 10-12 mm) a circle enclosing the centre of the lip of 80% of the subjects tested (over 4000). A handset conforming to these dimensions (see Figure 1/P.35) will then be acceptable to more than 90% of users. When a longer lip-to-mouthpiece distance is chosen, the signal-to-ambient-noise ratio will be worse and recommended LSTR values will be more difficult to meet (see Recommendations G.121, P.11, P.76, P.79 and Supplement No. 11). Therefore both signal-to-ambient-noise ratio and mouthpiece position for convenient use must be considered and probably a compromise must be made.

3 Recommendation on handset

Based on the information given above, the CCITT recommends that handset telephones conform to the dimensions outlined in Figure 1/P.35, with respect to mouthpiece positions and cheek-to-handset clearance.

Note — An earpiece with a design that forms a good seal to the IEC 318 ear (Recommendation P.51) will facilitate testing both in laboratories and during manufacturing. Experience has shown that earpieces with a good seal to the IEC 318 artificial ear also give in most cases a good seal to the human ear.

Figure 1/P.35, p.

References

[1] CCITT — Contribution COM XII-No. 49 (ITT), Study Period 1973-1976.

[2] CCITT — Contribution COM XII-No. 21 (People's Republic of China), Study Period 1977-1980.

[3] CCITT — Contribution COM XII-No. 112 (People's Republic of China), Study Period 1977-1980.

Bibliography

CCITT — Contribution COM XII-No. 32 (U.K), Study Period 1973-1976.

EFFICIENCY OF DEVICES FOR PREVENTING THE OCCURRENCE OF EXCESSIVE ACOUSTIC PRESSURE

BY TELEPHONE RECEIVERS

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

The use of devices for preventing the occurrence of excessive acoustic pressure by telephone receivers is recommended in Recommendation K.7. Methods for checking the efficiency of such devices in response to short duration impulses and for longer duration disturbances, such as tones, are given in this Recommendation. A method is also given for checking that such devices do not have adverse effects on normal speech signals.

Preliminary Note — On the basis of the findings of scientific studies, several authors or organizations have proposed ear-damage risk criteria based on variations in acoustic pressure, under impulse conditions for which, parenthetically, there is no single definition. Likewise, ear-damage risk criteria have also been proposed for longer duration acoustic disturbances, such as tones. However, these criteria cannot be directly transposed to the test conditions and measurements described below. Nor could the results be cross-checked without introducing certain hypotheses that are not specified in this Recommendation, the purpose of which is merely to describe a method simple both in its application and in the analysis of the results obtained. The criteria recommended are based on experience gained in several countries about the telephone receiver quality necessary to ensure the safety of users and operators. Administrations may wish to adopt lower limiting levels to reduce user annoyance caused by acoustic disturbances, but the limiting levels should not be so low as to have adverse effects on normal speech levels.

1 Efficiency of protection against short duration impulses

In order to check whether a telephone set affords satisfactory protection against the risk of acoustic shocks due to short duration impulses, it is recommended that its characteristics be examined as follows:

a) the entire telephone set, including the protective device, is placed in normal operating conditions as regards current supply and its position for the exchange of a call (e.g. with the handset raised);

b) the earpiece of the handset earphone is applied in the normal way to an artificial ear conforming to Recommendation P.51 (which corresponds to IEC Publication 318);

c) the artificial ear is electrically connected to a precision sound level meter conforming to IEC Publication 651, correctly calibrated and having the necessary circuits for measuring peak acoustic pressure levels. This equipment must be of class 2 for prototype testing, and may be of class 3 for checking mass-produced sets;

d) electrical impulses are applied to the telephone set by a suitable assembly which enables these impulses to be superimposed on the d.c. supply without the latter short-circuiting them. These impulses are produced by a generator which conforms with Figure 1/K.17, and whose components are those described for symmetric-pair repeater tests (R_3 = 25 ohms, C_2 = 0.2 μF, see Table 1/K.17). The test voltage is between 0 and 1.5 kV;

e) the telephone set is also checked for self-generated acoustic impulses such as those produced by operation of the hook switch or by dialing;

f) for both cases d) and e) above, the peak acoustic pressure level observed (maximum instantaneous value) should be below 140 dB relative to 20 μPa. In the long term, Administrations are recommended to limit this value to 135 dB for sets in common use.

Note — Administrations may deem it appropriate to use different limits for specific cases, for instance for the headsets used by operators.

2 Efficiency of protection against longer duration disturbances

In order to check whether a telephone set affords satisfactory protection against the risk of acoustic hazards due to longer duration disturbances, such as tones, it is recommended that its characteristics be examined as follows:

a) the entire telephone set, including the protective device, is placed in normal operating conditions as regards current supply and its position for the exchange of a call (e.g. with the handset raised);

b) the earpiece of the handset earphone is applied in the normal way to an artificial ear conforming to Recommendation P.51 (wich corresponds to IEC Publication 318);

c) the artificial ear is electrically connected to a precision sound level meter conforming to IEC Publication 651, correctly calibrated to measure A-weighted sound pressure levels. This equipment must be of class 2 for prototype testing, and may be of class 3 for checking mass-produced sets;

d) a 1000 ± 20 Hz sinewave signal is applied to the telephone set and its amplitude is increased until it reaches 10 $V_{r\backslash dm\backslash ds}$ across the set's terminals or until the steady-state acoustic output from the telephone receiver reaches its limiting value, whichever occurs first;

e) the telephone set is also checked for self-generated acoustic disturbances, such as tone dialing signals fed back to the receiver;

f) for both cases d) and e) above, the steady-state A-weighted sound pressure level should be below 125 dBA (''slow'' response).

Note 1 — Tones or other disturbances which are inherently limited to less than 0.5 s duration should be evaluated as short duration impulses under § 1. Repetitive disturbances, such as those which might be produced during automatic tone-type dialing, should be evaluated under § 2 using the sound level meter set for ''slow'' response averaging.

Note 2 — Administrations may deem it appropriate to use different levels for specific cases, for instance, for the headsets used by operators.

3 Effect on normal speech signals

It is recommended to check whether the strong-signal attenuation obtained by protective devices does not cause deterioration of the normal speech signals, e.g. by nonlinear distortion. This may be done by conducting a series of measurements using steady-state sine wave signals at a frequency of 1000 ± 20 Hz and relating to the following magnitudes:

N is an electric voltage level at the terminals of the set. *N* | s determined by the relation:

$$N = 20 \log_{10} \frac{fIV_{rms} fR}{.775} \quad \text{(dB)}$$

where V_{rms} represents the r.m.s. value of the voltage across the terminals. The value of $V_{r\backslash dm\backslash ds}$ = 0.775 volts (—2.2 dBV) gives $N = 0$ and corresponds to a power level of 0 dBm into 600 ohms.

P (*N*) is an acoustic pressure produced by the telephone receiver under given conditions, (this may be the pressure measured on an artificial ear in accordance with Recommendation P.51), corresponding to the application of voltage level *N* across the terminals of the set.

The ISO list of preferred frequencies includes 1000 Hz. It is a commonly used reference frequency in acoustic testing. Recommendation O.6 suggests 1020 Hz be used when testing PCM systems to avoid being at a submultiple of the 8000 Hz sampling rate. This Recommendation may need to be considered when testing digital telephones.

$A(N)$ is an attenuation of electroacoustic efficiency in relation to its reference value determined for $N = -20$ dB. $A(N)$ is determined by the relation:

$$A(N) = 20 \log_{10} \frac{f\,IP(-20)}{f\,IP(N)} + N + 20 \quad \text{(dB)}$$

[$A(N) = 0$ when $N = -20$ dB].

The values obtained for $A(N)$ must match those in Table 1/P.36 which have been obtained from measurements carried out on several types of set fitted with various protective devices.

Note 1 — It may be useful to make a few additional measurements to ensure that, at frequencies between 200 Hz and 4000 Hz, the values observed for A (N) are of the same order.

Note 2 — Some sets of recent design have special features, such as electroacoustic sensitivity which depends on the conditions of d.c. current supply or on the level of the speech signals received, quite apart from the effect of the protective devices. In that case, Administrations intending to use such sets may have to adapt the above conditions, taking care nevertheless to comply with their principles.

TABLE 1/P.36 [T1.36], p.

Recommendation P.37

MAGNETIC FIELD STRENGTH AROUND THE EARCAP OF TELEPHONE HANDSETS WHICH PROVIDE FOR COUPLING

TO HEARING AIDS

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

1 Introduction

Magnetic induction systems incorporated in telephone handsets generate an alternating magnetic field with spatial characteristics which make the field detectable by hearing aids equipped with induction pick-up coils

Reception of an audio-frequency signal via an induction pick-up coil can often allow an acceptable signal-to-noise ratio to be achieved in cases where the acoustical reception would otherwise be degraded by reverberation and background noise.

The magnetic field strength which enables induction pick-up coils in hearing aids to function effectively must be high enough to produce an acceptable signal-to-noise ratio but not so high as to cause overloading of the hearing aid.

The value of magnetic field strength recommended in this standard has been chosen so that these requirements are met as far as possible.

2 Scope

This Recommendation applies to telephone handsets which provide a magnetic field for coupling to hearing aids. It specifies the level linearity and frequency dependence of the magnetic field strength produced by the handset and characteristics for a calibrated probe coil.

3 Explanation of terms

3.1 *Level of magnetic field strength*

The maximum value of the magnetic field strength is measured in accordance with § 4.2.

3.2 *Plane of measurement*

A plane parallel to the earcap plane at a distance of 10 mm.

4 Magnetic field strength measurements and recommended values

4.1 *Calibration of acoustic receive level*

Using the measurement configuration shown in Figure 3/P.64, the drive level of the oscillator shall be adjusted to produce a sound pressure level of 80 dB at 1000 Hz. This drive level shall be used for measuring the level and frequency characteristics of the magnetic field strength.

4.2 *Magnetic field strength level*

Place (per § 5) the centre of the calibrated probe coil in the plane of measurement and circuit orientate it for maximum coupling. Determine the magnetic field strength at 1000 Hz using the drive level as per § 4.1.

Recommended range of values for the magnetic field strength is:

—17 to —30 dB relative to 1 A/m.

Note — Hearing aids with magnetic pick-up coils primarily intended for coupling to magnetic loops in auditoria in accordance with IEC Publication 118-4 are likely to have a sensitivity that corresponds to a field strength in the upper end of the range recommended for coupling to telephones.

4.3 *Linearity of the magnetic field strength*

With the probe coil positioned as in § 4.2, increase the 1000 Hz drive level specified in § 4.1 by 20 dB and measure the resulting magnetic field strength.

The field strength should increase by 20 dB ± 1 dB, or if the telephone set has a higher linearity the linearity of the magnetic field shall be equally as good.

4.4 *Measurement of frequency characteristics*

With the probe coil positioned as in § 4.2 and the drive level as in § 4.1, vary the frequency from 300 Hz to 3400 Hz and measure the resulting field strength. The magnetic field strength frequency characteristics shall fit within the template shown in Figure 1/P.37.

FIGURE 1/P.37, p.

5 Probe coil

5.1 *Dimensions*

For measuring the magnetic field strength, a calibrated probe coil having the following dimensions is recommended:

Core: length (12.5 ± 1 mm)

cross section (1 mm ± 0.5 mm) × (2 mm ± 0.5 mm)

Winding: length (10 mm ± 1 mm)

cross section (2 mm ± 0.5 mm) × (3 mm ± 0.5 mm)

The winding shall be shorter than the core.

Note 1 — The magnetic field may be inhomogeneous within distances comparable to the length of the probe coil. The introduction of a magnetic core material may also redirect the magnetic field contours, therefore the magnetic material of the core may be of importance.

Note 2 — The probe coil may be combined with frequency correcting elements to obtain a flat frequency response in the range of 300 Hz to 3400 Hz.

5.2 *Calibration of the probe coil*

The sensitivity as a function of frequency of the probe coil shall be measured with an accuracy of ± | .5 dB.

A method of producing a homogeneous magnetic field of known intensity is given in IEC Publication 118-1. The harmonic distortion of the magnetic field used for the calibration shall be less than 1%.

5.3 *Distortion*

The distortion of the probe coil shall be less than 2%, when measuring field strength up to +2 dB relative to 1 A/m.

ANNEX A

(to Recommendation P.37)

Measurement of an acousto-magnetic adapter generating a magnetic field

A.1 *Scope*

This annex specifies the measuring method for an acousto-magnetic adapter that converts the acoustic output of an associated telephone receiver to a magnetic field, in accordance with §§ 4.1 and 4.2, that can be received by the magnetic pick-up coil in a hearing aid.

A.2 *Definition of the adapter plane*

The adapter plane is defined as the plane formed by the contacting points of a flat surface against the surface of the acousto-magnetic adapter opposite the earcap connection.

A.3 *Definition of the plane of measurement*

The plane of measurement is defined as a plane parallel to the adapter plane at a distance of 10 mm.

A.4 *Measurement procedures*

Measurements are made in accordance with this Recommendation.

The output sound pressure level of the telephone receiver is measured against the artificial ear without the acousto-magnetic adapter being mounted.

The characteristics of the magnetic field of the acousto-magnetic adapter are measured when mounted on the actual telephone receiver.

Note — In reporting results, the type of telephone set used should be specified.

A.5 *Magnetic field requirements*

The magnetic field produced by the adapter when fitted to a handset should meet the level and frequency characteristic requirements given in §§ 4.2 and 4.4.

A.6 *Physical properties*

Desirable physical properties of the acousto-magnetic adapter are:

— easy to place on the earcap and remove again;

— a firm contact to the earcap so that the acousto-magnetic adapter and the telephone handset can be used as an integrated unit;

— forming a good and well-defined acoustic coupling to the earcap (see Note);

— the surface of the acousto-magnetic adapter defining the adapter plane should be flat or should have a shape easily defining the adapter plane;

— the adapter plane should be approximately parallel to the earcap plane;

— the magnetic field produced by the adapter should be orientated so that the magnetic coupling to the hearing aid is only to a small extent dependent on the position of the hearing aid.

Note — The inner diameter of an acoustic seal is recommended to be equal to the edge diameter of the IEC 318 artificial ear.

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Recommendation P.38

TRANSMISSION CHARACTERISTICS OF OPERATOR TELEPHONE SYSTEMS (OTS)

The measurement methods adopted for measuring on operator telephone systems (OTS) which comprise a headset, feeding circuit and subscriber's line (the same principles can be applied to any system that uses a headset), conform to the methods described in Recommendation P.64 with the following exceptions:

1 Sending sensitivities of OTS

In principle, the OTS is similar to the Local Telephone System (LTS) of Recommendation P.64 with the exception that in a headset the earphone and microphone may not have a fixed relationship as has a conventional telephone handset. Those headsets which are not adjustable in distance from the receiver to the microphone should be positioned per Annex A of Recommendations P.76. For those which are adjustable, a modal position of the input port of the mouthpiece must be specified by the manufacturer in 3-dimensional coordinates relative to the lip, horizontal and vertical reference planes of the mouth as defined in Recommendation P.51.

This modal position is defined by the manufacturer to be representative of the position of normal usage.

Note — The term ‘‘corner of the mouth’’ used by some manufacturers in defining the normal use position is assumed to be 21 mm from the centre 9 mm behind the lip plane. The sound field of the artificial mouth is not defined behind the lip plane and therefore measurement points behind the lip plane are not recommended.

The sending sensitivity is then determined as per §§ 2, 4 and 6 of Recommendation P.64

The sending loudness rating (SLR) is computed as described in Recommendation P.79.

2 Receiving sensitivities of OTS

2.1 For headsets using supra-aural earphones , the IEC 318 artificial ear is used.

The receiving sensitivity is determined as per §§ 3, 5 and 7 of Recommendation P.64.

The receiving loudness rating (RLR) is computed as described in Recommendation P.79 using the L_E values of Table 4/P.79.

2.2 For headsets using insert type receivers , the IEC 711 ear simulator is used.

The receiving sensitivity is determined as per §§ 3, 5 and 7 of Recommendation P.64.

2.3 The receiving sensitivity suitable for use in the calculation of loudness requires:

a) a transfer function (S_{DE}) for the eardrum to the ear reference point (ERP) and is given in Table 1/P.51, and

b) the $L_{E(I)}$ values of Table 1/P.38 appropriate for insert type receivers.

The sensitivity is defined as:

$$S_{JE} \quad S_{Jd} \quad S_{DE} \quad L_{E(I)}$$

where

S_{JE} is the sensitivity from the junction to the real ear.

S_{Jd} is the sensitivity from the junction to the 711 IEC ear simulator (eardrum).

$L_{E(I)}$ is the ear coupling loss of insert type receivers (Table 1/P.38).

S_{DE} is the transfer function from the eardrum to the ERP (Table 1/P.51).

The receiving loudness rating (RLR) is computed as described in Recommendation P.79 using S_{JE} from the above formula. (*Note* — The L_E values of Table 4/P.79 have been replaced by the values Table 1/P.38.)

Note 1 — Study is still continuing under Question 8/XII, to evaluate intra-concha, circum-aural and non-contact types of ear-pieces.

Note 2 — Further information on the measurement of OTS can be found in § 3.4 of the Handbook on Telephonometry.

H.T. [T1.38]

TABLE 1/P.38

Values of $L_{E(I)}$ for insert type receivers

Frequency (Hz)	*L ( ) (dB)*
200	*23.0*
250	*19.0*
315	*18.0*
400	*17.4*
500	*12.8*
630	*9.0*
800	*6.8*
1000	*3.2*
1250	*1.5*
1600	*1.4*
2000	*0.4*
2500	*-1.5*
3150	*3.0*
4000	*0.0*

Tableau 1/P.38 [T1.38], p.13

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