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**TELEPHONE TRANSMISSION QUALITY
SUBSCRIBERS' LINES AND SETS**

**TRANSMISSION CHARACTERISTICS
OF HANDS-FREE TELEPHONES**

ITU-T Recommendation P.34

(Previously "CCITT Recommendation")

FOREWORD

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

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

ITU-T Recommendation P.34 was revised by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

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 Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

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

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

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TRANSMISSION CHARACTERISTICS OF HANDS-FREE TELEPHONES

(Melbourne, 1988; amended in Helsinki, 1993)

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 clauses 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 still under study.

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.

NOTES

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.

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.

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 clause 6.

5 Switching characteristics

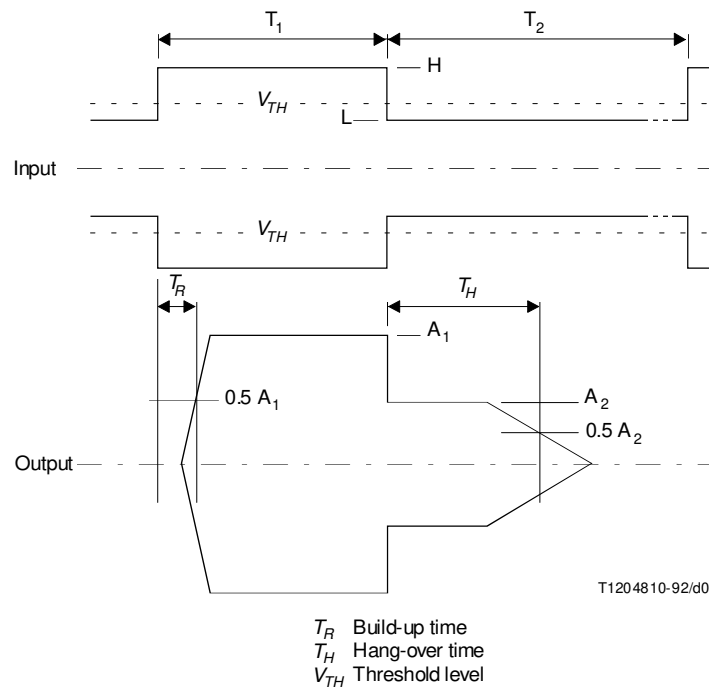
Most loudspeaker and hands-free telephones contain voice-switched circuits whose main function is to avoid singing through acoustic feedback. Such circuits insert a loss in either the sending or receiving direction in various ways. Switching from one direction to the other occurs when a signal above 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 and 2):

- Threshold level V_{TH} – 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).

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.



NOTE – In the case shown here, the switching time T_S is larger than $T_R + T_H$.

FIGURE 1/P.34

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

- 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.
- 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.

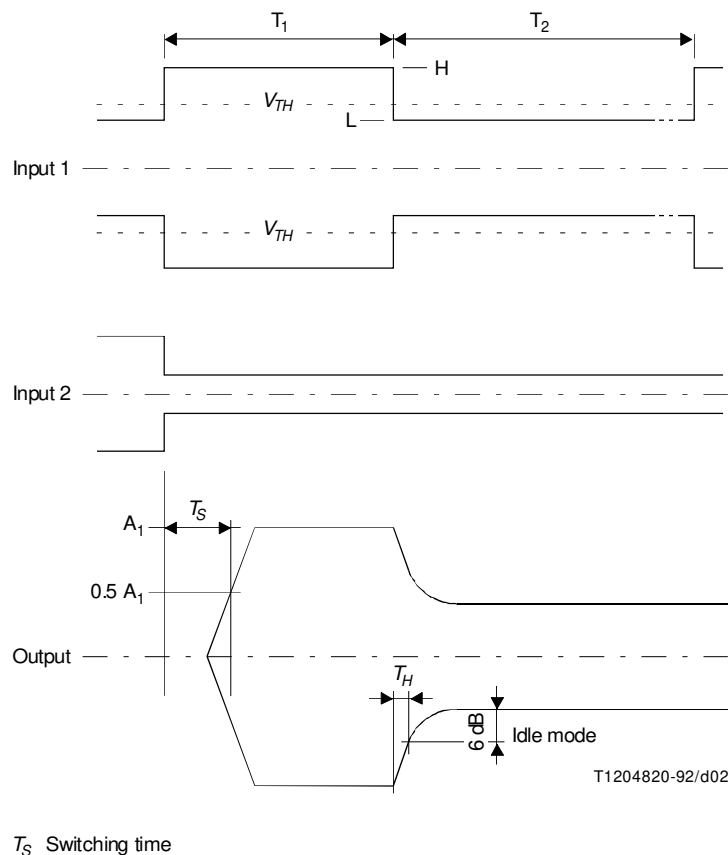


FIGURE 2/P.34

A suitable signal for measuring the characteristics for case a) above consists of a periodic tone burst signal (see Figure 1). The on/off times T_1/T_2 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_{TH} 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_{TH} 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_{TH} , 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).

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 clause 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 m² but not less than 0.96 m² and the width not less than 800 mm [1].

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 [2] for subjective and objective measurements is shown in Figure 3.

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.

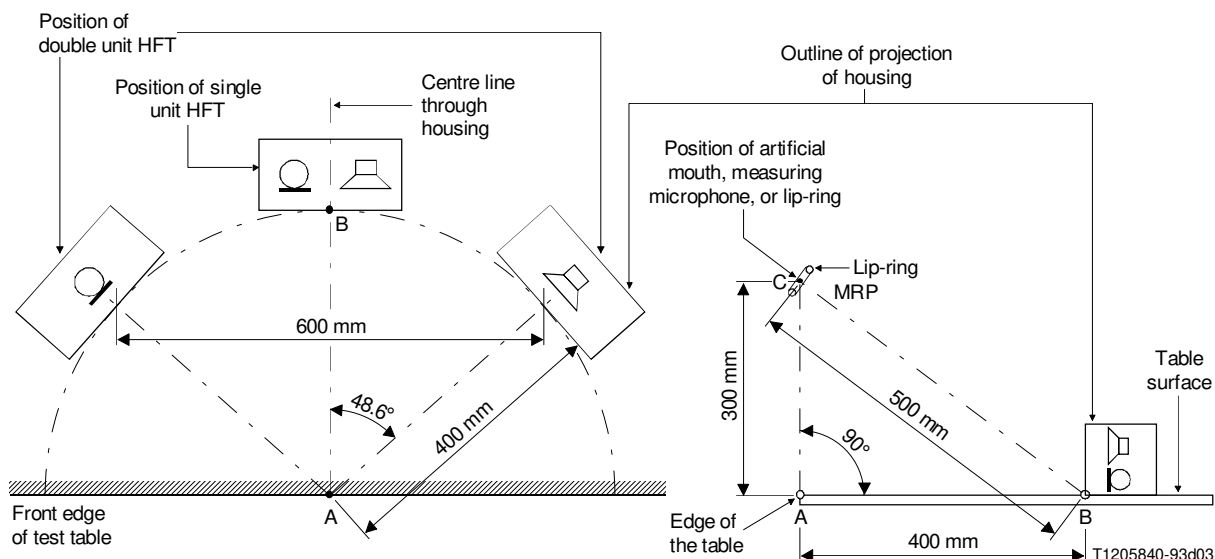


FIGURE 3/P.34

Physical test arrangements for subjective and objective measurements

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 ± 1 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 [3], [4]. 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 Telephony*.

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 hands-free telephone is recorded at the output terminals of the telephone 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.

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

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

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 (see Figure 3), which corresponds to -28.7 dBPa at 50 cm from the lip when there is no table nor set.

NOTE – 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”. The test signal level of such a signal is referred to the continuous signal before the ON/OFF modulation. The sensitivity can be computed from the transfer function measured by the pulse signal.

6.5.1.2 Receiving frequency sensitivity response measurements

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

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

where p_R is the sound pressure at point C in Figure 3 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 (see Figure 3).

NOTES

1 Other methods for calculating loudness ratings used by some Administrations for their own internal planning purposes can be found in Supplement No. 19 to Series P Recommendations.

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).

2 Better results are expected by using an artificial voice satisfying Recommendation P.50 (temporal characteristics of the signal closer to those of real speech).

3 The principles of a composite source signal as an example of a test signal to determine the transfer characteristics of terminal equipment are given in Supplement No. 21 (to this Recommendation).

6.5.2.2 Receiving

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

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.

A correction factor of 14 dB shall be subtracted from the computed loudness ratings.

NOTES

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

2 Better results are expected by using an artificial voice satisfying Recommendation P.50 (temporal characteristics of the signal closer to those of real speech).

3 The principles of a composite source signal as an example of a test signal to determine the transfer characteristics of terminal equipment are given in Supplement No. 21 (to this Recommendation).

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