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**P.340**

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SERIES P: TELEPHONE TRANSMISSION QUALITY  
Subscribers' lines and sets

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**Transmission characteristics  
of hands-free telephones**

ITU-T Recommendation P.340

(Previously "CCITT Recommendation")

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**ITU-T P-SERIES RECOMMENDATIONS**  
**TELEPHONE TRANSMISSION QUALITY**

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*For further details, please refer to ITU-T List of Recommendations.*

## **FOREWORD**

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). 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, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation P.340 was prepared by ITU-T Study Group 12 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 30th of August 1996.

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### **NOTE**

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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## **SUMMARY**

This Recommendation updates and replaces Recommendation P.34 (1993). It recommends the transmission and quality characteristics of hands-free and loudspeaking terminals, whatever the type of terminal implementation.

However, some paragraphs apply specifically to analogue terminals.

The first part of this Recommendation recommends the transmission characteristics of hands-free terminals, as shapes for the response curves, sensitivity values, the test principles and methods, and the switching parameters.

The second part recommends the test conditions, in particular the test table, the hands-free terminal positioning according to the type of application, the test room environment, the test signals and a calibration method for the artificial mouth.

The third part proposes a set of parameters to qualify the conversation quality given by hands-free terminals. A test method is recommended for each parameter. Finally, the recommended values for some of these parameters are defined, according to a classification, based on the capability to work or not in full duplex.



## **TRANSMISSION CHARACTERISTICS OF HANDS-FREE TELEPHONES**

*(Geneva, 1996)*

### **1 Introduction**

The sending and receiving sensitivities of handset telephones, 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 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 4.1 and 4.2 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.

This Recommendation covers generic requirements that are applicable to both analogue and digital hands-free terminals. Additional requirements that are applicable strictly to digital terminals can be found in Recommendation P.342 – Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free terminals, and Recommendation P.341 – Transmission characteristics of wideband hands-free telephones.

### **2 Normative references**

The following Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision: all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

- [1] ITU-T Recommendation G.121 (1993), *Loudness Ratings (LRs) of national systems*.
- [2] CCITT Recommendation G.223 (1984), *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*.
- [3] ITU-T Recommendation P.342 (1996), *Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free telephony terminals*.
- [4] ITU-T Recommendation P.341 (1995), *Transmission characteristics of wideband handsfree telephones*.
- [5] ITU-T Recommendation P.78 (1996), *Subjective testing method for determination of loudness ratings in accordance with Recommendation P.76*.

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<sup>1)</sup> Previously Recommendation P.34.

- [6] ITU-T Recommendation P.79 (1993), *Calculation of loudness ratings for telephone sets*.
- [7] ITU-T Recommendation P.50 (1993), *Artificial voices*.
- [8] ITU-T Recommendation P.51 (1996), *Artificial mouth*.
- [9] ITU-T Recommendation P.501 (1996), *Test signals for use in telephonometry*.
- [10] ITU-T Recommendation G.167 (1993), *Acoustic echo controllers*.
- [11] ITU-T Handbook on Telephonometry, 1993.

### 3 Definitions and abbreviations

The relevant definitions given in Recommendation P.10 apply along with the following:

**3.1 hands-free reference point (HFRP):** A point located on the axis of the artificial mouth, at 50 cm from the lip ring, where the level calibration is made, in free field. It corresponds to the measurement point 11, as defined in Recommendation P.51.

**3.2 hands-free (telephone) set (HFT):** A telephone set using a loudspeaker associated with an amplifier as a telephone receiver and which may be used without a handset.

**3.3 loudspeaking (telephone) set (LST):** A handset telephone using a loudspeaker associated with an amplifier as a telephone receiver.

For the purposes of this Recommendation, the following abbreviations are used.

$a_H$	Insertion attenuation range, sum of maximum attenuations possible in the sending and receiving path
$a_{H,S}$	Attenuation in the sending path at idle mode
$a_{H,R}$	Attenuation in the receiving path at idle mode
$a_{k\min}$	Minimum compressor-gain of dynamic compressor
$a_{k\max}$	Maximum compressor-gain of dynamic compressor
$a_{Kf'}$	Attack-slope: speed of the compressor-controlled attenuation at a positive jump
$a_{Kr'}$	Decay-slope: speed of the compressor-controlled attenuation at a negative jump
CSS	Composite Source Signal
EEB	Early Energy Balance
HFRP	Hands-Free Reference Point
MRP	Mouth Reference Point
OLR	Overall Loudness Rating
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
$T_R$	Build-up Time
$T_H$	Hang-over Time
$T_S$	Switching Time
TUT	Terminal Under Test
$V_{TH}$	Threshold Level

### 4 Transmission characteristics

#### 4.1 Sending sensitivity

The Sending LR (SLR) of an HFT should be about 5 dB higher than the SLR of the corresponding handset telephone (the actual value will depend on the type of handset used).

NOTE 1 – 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.

NOTE 2 – Hands-free telephones having a sending sensitivity that complies with this Recommendation can be assumed to fulfil Recommendation G.223.

Furthermore, in order to avoid excessive crosstalk from the 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 handset telephones.

NOTE 3 – Necessary precautions should be taken so that the terminal user may be able to break the sending circuit if oscillations occur, or to provide suitable methods so that a device controlled by the voice may prevent oscillations.

## 4.2 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 an RLR value about 10 dB lower.

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 of the Overall Loudness Rating (OLR) of the connection 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 -29 dBPa for -49 dBPa(A) room noise, or -24 dBPa for -39 dBPa(A) room noise. However, to obtain maximum mean opinion scores in conversation tests, listening levels of about 5 to 10 dB higher may be required.

## 4.3 Frequency response curves

### 4.3.1 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 pre-emphasis increase the intelligibility. Therefore, if a frequency equalisation is used to compensate the cable attenuation of the analogue part of a connection, 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 for analogue terminals in order to avoid interference by crosstalk to adjacent channels in certain types of long-distance circuits.

If the analogue terminals are intended to be connected through short lines to a digital connection, it is appropriate that the sensitivity above 4 kHz be as low as possible, to avoid spurious out-of-band signals.

For (300-3400 Hz) digital terminals, information is available in Recommendation P.342.

For wideband digital terminals, information is available in Recommendation P.341.

#### **4.3.2 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 5.

### **4.4 Subjective determinations**

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

NOTE – Some information about reference equivalents can be found in the Handbook on Telephonometry.

#### **4.4.1 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 [ $< -59 \text{ dBPa(A)}$ ]. 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  $-34 \text{ dBPa(A)}$ .

#### **4.4.2 Receiving**

The talking level at the reference microphone for the measurement of 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 minimise 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.

## 4.5 Objective evaluations

### 4.5.1 Sensitivity measurement

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 to P-Series Recommendations.

#### 4.5.1.1 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  $S_{mJ}$  is expressed as dB relative to 1 V (electrical interface)/Pa (MRP) as follows:

$$S_{mJ} = 20 \log V_s - 20 \log P_{MRP} + \text{Corr} - 24$$

where:

$V_s$  is the measured voltage across the appropriate termination (unless stated otherwise, a 600 ohms termination).

$P_{MRP}$  is the applied sound pressure at the MRP.

$\text{Corr}$  is  $20 \log (P_{MRP}/P_{HFRP})$  of the used artificial mouth.

NOTE – The value of Corr is the value given in the calibration chart of the artificial mouth. (24.0 dB is the ideal value).

#### 4.5.1.2 Receiving sensitivity measurements

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

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

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.

### 4.5.2 Measure and computation of loudness ratings

#### 4.5.2.1 Sending loudness rating

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.

NOTE 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 P-Series 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).

NOTE 2 – Appropriate types of signals can be found in Recommendation P.501.

#### 4.5.2.2 Receiving loudness rating

Objective measurements described in 4.5.1.2 are made with a free field microphone at point C (see Figure 3).

Loudness ratings are computed following Recommendation P.79.

When calculating the RLR of a hands-free or loudspeaking telephone, a factor  $HFL_E$  could replace the  $L_e$  factor used in the algorithm of Recommendation P.79

where:

$$HFL_E = E_B + E_D$$

and where:

$E_B$ : is the binaural effect of listening with two ears rather than one.

$E_D$ : is the diffraction effect of the human head at ERP.

For the purposes of calculating the RLR for hands-free telephones, sufficient accuracy will be obtained if the following values are assumed:

$$E_B = -12 \text{ dB}$$

$$E_D = -2 \text{ dB}.$$

### 4.6 Switching parameters

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.

The following values are recommended:

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

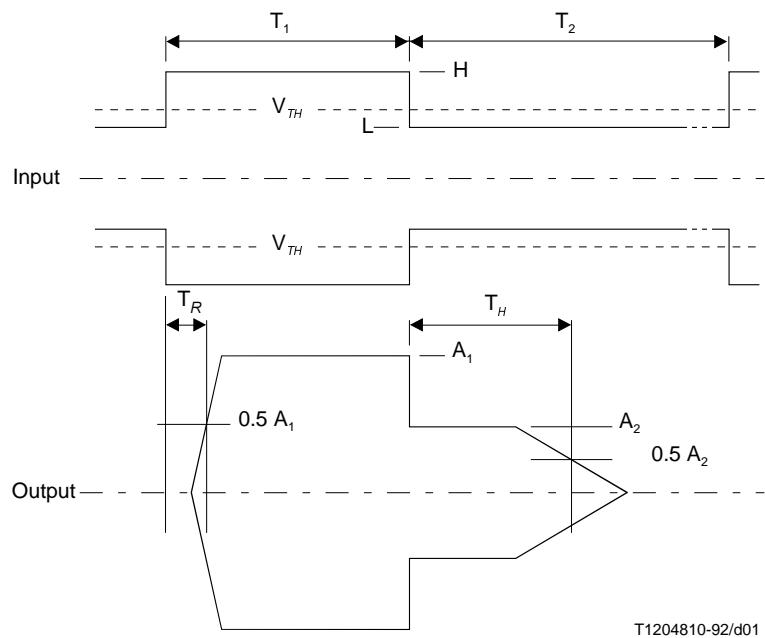
The hang-over time  $T_H$  is defined in Table 4.

The switching time  $T_S$  should be approximately 100 ms.

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.

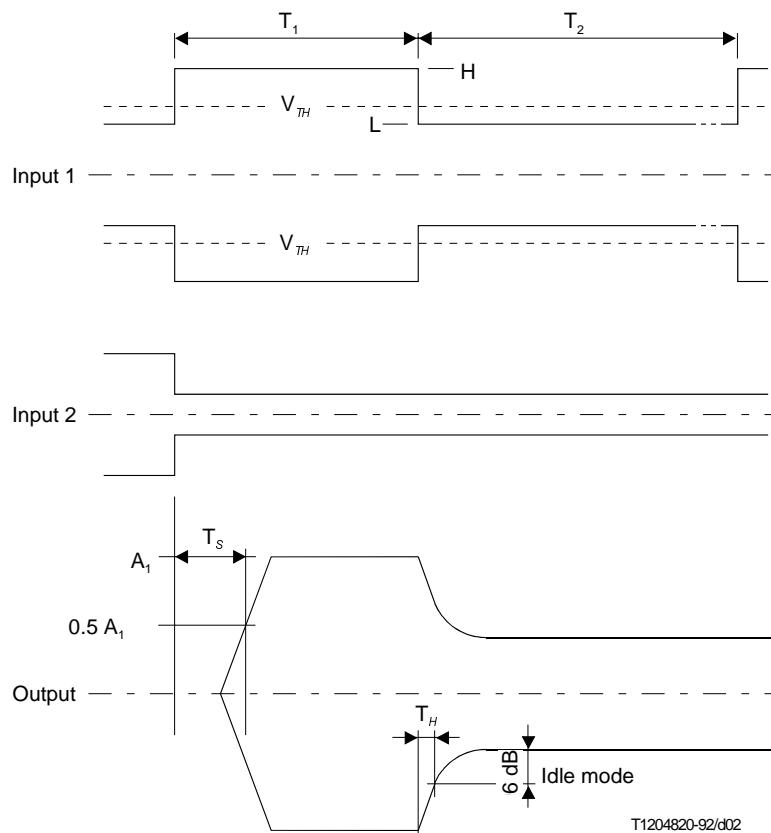
The first case is of fundamental importance, as its characteristics also affect simultaneous conversation characteristics, and hands-free telephones should therefore always be checked in that respect.



$T_R$  Build-up Time  
 $T_H$  Hang-over Time  
 $V_{TH}$  Threshold Level

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

FIGURE 1/P.340



$T_S$  Switching time

FIGURE 2/P.340

## 5 Testing conditions

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

### 5.1 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<sup>2</sup> but not less than 0.96 m<sup>2</sup> 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.

### 5.2 Test arrangement

The physical test arrangements of one piece (single unit) HFTs for subjective and objective measurements are 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.

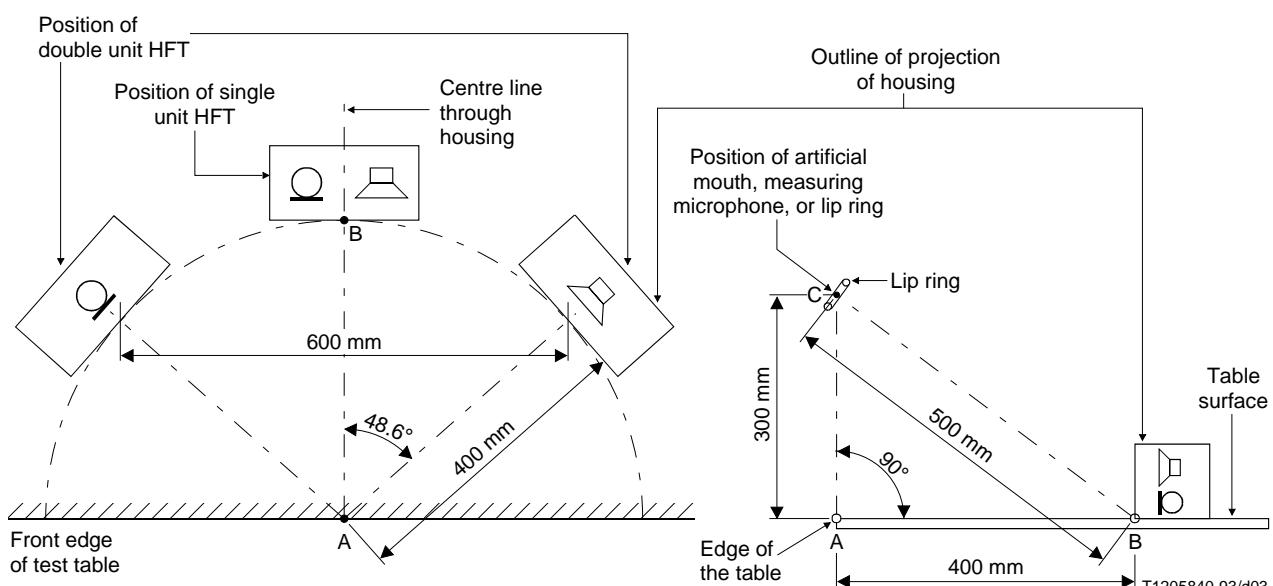


FIGURE 3/P.340  
Physical test arrangements for subjective and objective measurements

For more-than-one-piece HFTs, videophony and multimedia terminals, the test arrangement shall be modified to what it is stated in the instruction manual of the Terminal Under Test (TUT). The positioning of the terminal shall be referred to the point C defined and located as in Figure 3. Sending and receiving loudness ratings, sending and receiving sensitivity responses shall be tested and adjusted for this test arrangement. The axis of the artificial mouth shall be defined by manufacturer's declaration.

If there is no manufacturer's declaration, the test arrangement of Figure 3 shall be implemented.

### 5.3 Calibration of the artificial mouth

Recommendation P.51 defines the tolerances that apply to the acoustic radiation of the mouth. To reduce the possible errors that could be due to these tolerances, the following procedure shall be used to calibrate the artificial mouth.

The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP.

The level of the acoustic signal is adjusted to  $-4.7 \text{ dBPa}$  at the MRP.

The spectrum at MRP is then recorded and used as a reference.

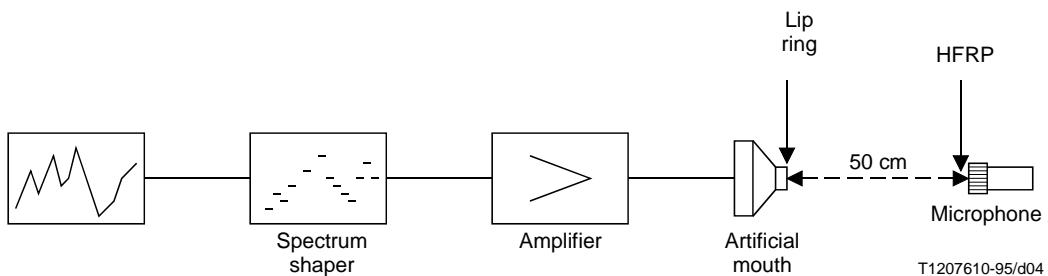


FIGURE 4/P.340  
Physical test arrangement for calibration of the artificial mouth

The broadband signal level then is adjusted to  $-28.7 \text{ dBPa}$  at the HFRP and the spectrum is not altered.

The spectrum at the MRP and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity  $S_{mJ}$ .

See Figure 4.

### 5.4 Test environment

**1** For the repeatability of the tests, the environment for most of the measurements shall be free field (anechoic) down to the lowest frequency of the 1/3 octave band centred on 200 Hz.

Satisfactory free field conditions exist where errors, due to the departure from ideal conditions, do not exceed the values defined in Table 1, inside a sphere centred on point B (see Figure 5), with one metre radius, in absence of the table.

TABLE 1/P.340

1/3 octave band centre frequency (Hz)	Allowable departure (dB)
< 630	$\pm 1.5$
800 to 5000	$\pm 1$
> 6300	$\pm 1.5$

The test signal level for verification of the free field is  $-20 \text{ dBPa}$ .

Measurements are made along the seven axes which are numbered 1) to 7) in Figure 5, with the sound source placed at positions equivalent to B or C, as appropriate. Measurement points along each axis, taken from the front plan of the artificial mouth lip-ring are at the distances of 315 mm, 400 mm, 500 mm, 630 mm, 800 mm and 1000 mm.

**2** The broadband noise level shall not exceed  $-70 \text{ dBPa(A)}$ . The octave band noise level shall not exceed the values specified in Table 2:

TABLE 2/P.340

**Noise level**

Centre frequency (Hz)	Octave band pressure level (dBPa)
63	-45
125	-60
250	-65
500	-65
1 k	-65
2 k	-65
4 k	-65
8 k	-65

**5.5 Test signals**

The test signals shall comply with the contents of Recommendation P.501.

In clause 7, the appropriate type of test signal is a complex signal as defined in Recommendation P.501 (e.g. CSS).

**5.6 Test signal levels**

Measurement level, unless otherwise specified, corresponds to the standard level. This is -28.7 dBPa at HFRP for send, -30 dBm0 for receive.

**6 Conversational quality**

In the case of many of the hands-free telephones which have been hitherto available, the advantages over handsets have usually meant a sacrifice in terms of a reduction in speech transmission quality. The parameters relevant to a quality description were obtained from auditory tests. The instrument-based parameters to be correlated were extracted. Hands-free telephones are not linear, time-invariant systems, and newly developed measurement engineering techniques for measuring transmission response must be additionally applied. A comprehensive definition of quality for hands-free telephones requires the adaptation or extension of current test measurements.

Table 3 outlines the auditory parameters for defining quality for both transmission directions and lists these parameters in relation to their importance.

The parameters "room noise transmission" and "duplex capability" are the most important auditive characteristics and are crucial in deciding the subjective impression of quality of hands-free equipment. The most important value measurable in terms of instrumentation, correlating to these auditive values, is, in the case of duplex capability, the attenuation range, and for room noise transmission the attenuation distribution at idle. These parameters provide a first indication of the product quality of hands-free equipment and allows a classification of the equipment. Three types can be included in such a classification:

- 1) Fully duplex operative:

In duplex operation, attenuation of the conversation partner either does not occur or is unnoticeable. Room noise is transmitted in the send path (with or without room noise reduction).

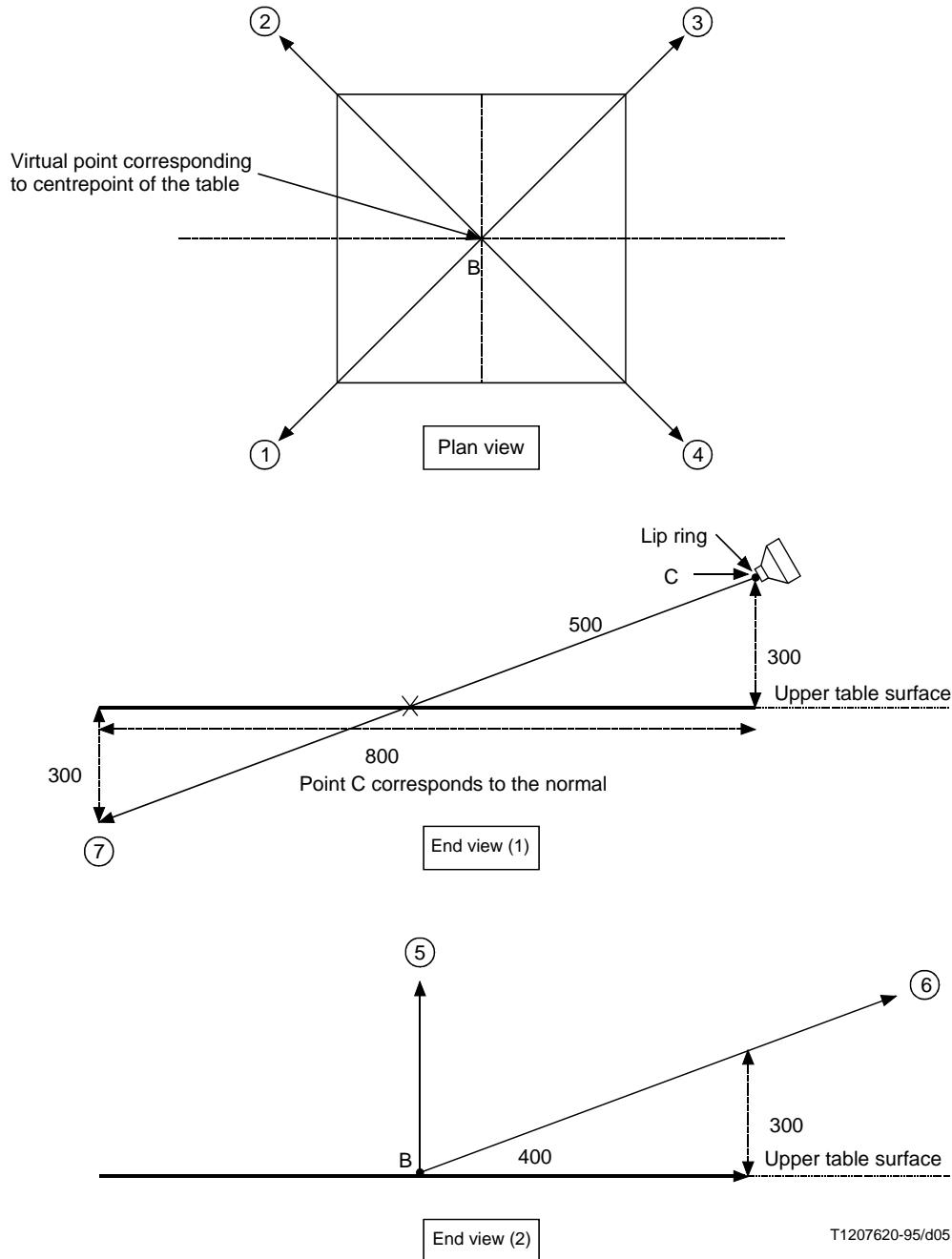
- 2) Equipment with limited duplex operation:

In duplex operation, one or both of the speech paths is attenuated. The conversation partner and room noise in the send path are still audible.

3) Equipment where no duplex operation is possible:

When one conversation partner is talking, the other is fully attenuated. Room noise in the send direction is not transmitted.

Every telephone must meet the requirements within its particular class in order to achieve a minimum quality requirement.



Dimensions in millimetres.

Points 1, 2, 3 and 4 are in the horizontal plane normally occupied by the table surface.

Measurements of free field sound pressure are made in absence of the table.

Axes used in the determination of free field conditions for 1 m radius sphere.

FIGURE 5/P.340  
Determination of the free field conditions

TABLE 3/P.340

Sending	Receiving
Quality of room noise transmission (at idle)	
Duplex capability	Duplex capability
Completeness of speech transmission	
Sound quality	Sound quality
Clipping	Clipping
Perception of slight control operations	Perception of slight control operations
Reverberance	Volume
Noise	Noise

## 7 Parameters measured instrumentally

In this clause, based on [9] are defined technical characteristics and additional and/or adapted test procedures:

- attenuation range,  $a_H$ ;
- attenuation distribution at idle mode;
- hang-over time (switch-off time);
- dynamic compression;
- reverberance: impulse response, EEB;
- time-dependent frequency response;
- duplex behaviour;
- control response in echo cancellation;
- quality of speech recognition.

The measurement techniques involved in these additional measurements are outlined in more detail below.

Unless specified otherwise, the test conditions shall conform to clause 5.

### 7.1 Attenuation range $a_H$

Attenuation range is determined by the difference in sensitivity response which results when one speech path is activated and when the duplex branch is activated. An activating signal (e.g. voiced sound of CSS) is used as the receive measurement signal, immediately followed by an activating signal in the send direction (see Figure 6). In the analysis the level of the measurement output signal is represented versus time (time constant = 5 ms). The attenuation range is obtained from the difference between the maximum level at full activation and the minimum level obtaining immediately after switch-over.

### 7.2 Attenuation distribution at idle mode

The attenuation range is derived from the sum of the maximum attenuation in the send direction  $a_{H,S}$  and the maximum attenuation in the receive direction  $a_{H,R}$ . In the case of a system with no compressor,  $a_H = a_{H,S} + a_{H,R} = \text{constant}$  applies. (In the case of a system including a compressor, the attenuation at maximum and minimum level due to the compressor must be obtained, see 7.4).

For measuring attenuation distribution a signal can be used which consists of a train of activating signals (e.g. voiced sound of CSS) and pauses in both speech directions, whereby extremely large variation in transition combinations are possible (see Figure 7). The attenuation value can thus be obtained at the appropriate transition points from idle to activation for each of the speech paths. Again, as in 7.1, measurement signal level is represented versus time.

Attenuation in the send direction  $a_{H,S}$  can be read off from idle to send at the transition point, and is derived from the difference between the maximum level at full activation in the send direction and the minimum level obtaining immediately after the send activation signal is present. The attenuation  $a_{H,R}$  can be correspondingly read off at the transition idle/receive activation signal.

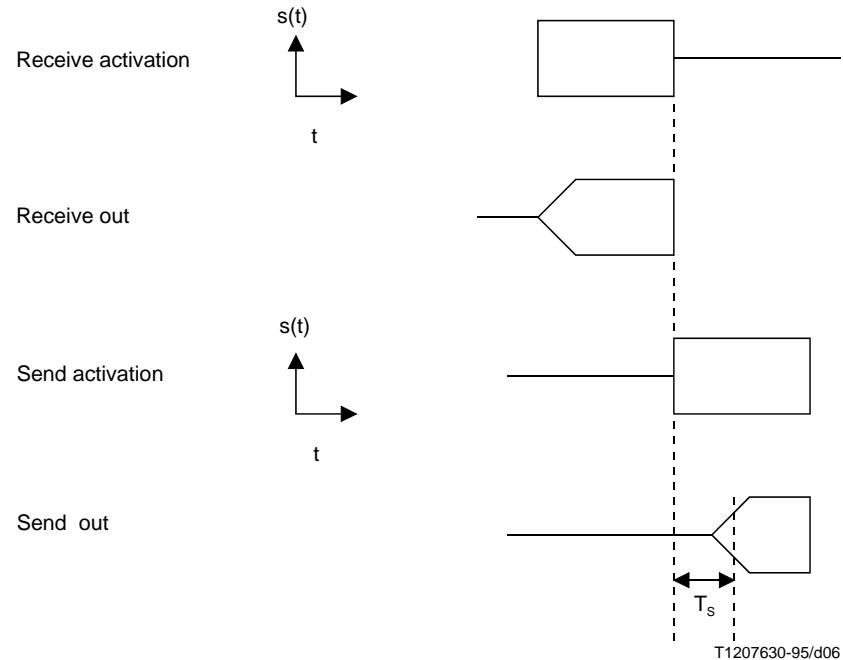


FIGURE 6/P.340  
**Attenuation range measurement**

If the switch-on time of the equipment is very fast, thus making reading off difficult, the measurement technique described in 7.3 (Hang-over time) can be used. The attenuation  $a_{H,S}$  or  $a_{H,R}$  of each speech path can be derived by calculating the difference in level at maximum and minimum attenuation in the second part of the signal (noise signal).

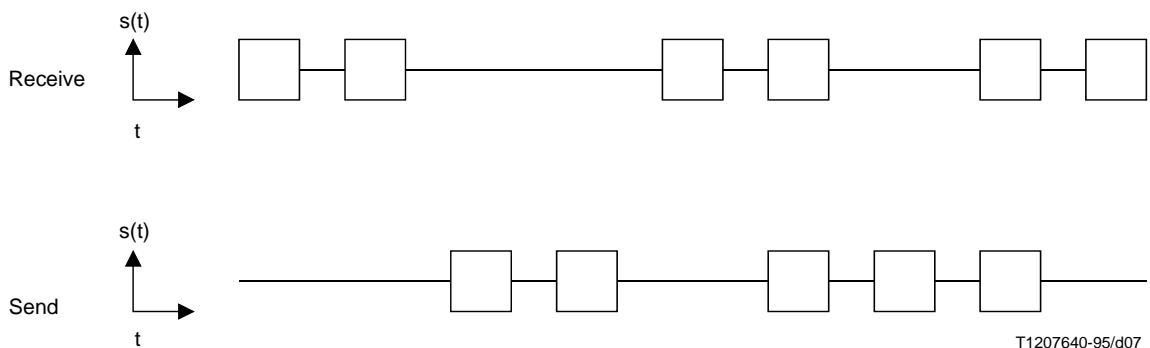


FIGURE 7/P.340  
**Attenuation distribution measurement**

### 7.3 Hang-over time

The transition from activation to idle can be represented by feeding in an activation signal (e.g. voiced sound of CSS) in one direction, followed by a second signal in the same direction but of lower level, which does not activate the hands-free telephone (noise signal) (see Figure 8). The second part of the signal measured thus indicates the attenuation, from which the Hang-over Time (switch-off time) can be determined.

The duration of the voiced sound is 0.5 s in order to reach a final stable system condition. The level corresponds to standard level, as defined in 5.6. The second part of the signal (noise signal) has a duration of 1 s. The level must be selected low enough so as not to activate the equipment. The suggested levels to be applied are:  $-58.7 \text{ dBPa}$  at HFRP for send, and  $-60 \text{ dBm0}$  for receive.

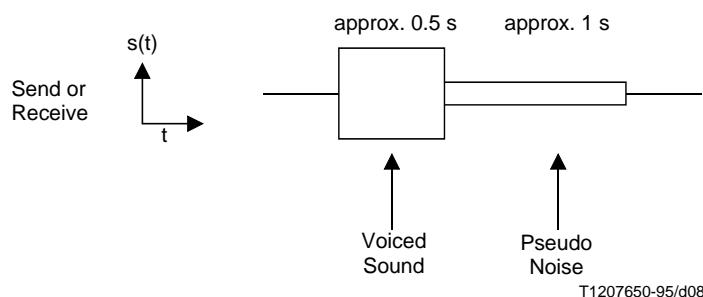


FIGURE 8/P.340  
Switch-off response measurement

### 7.4 Dynamic compression

A compressor response can be measured using an activation signal (e.g. voiced sound of CSS) with steeply monotone rising or falling level response (see Figure 9). The measurement signal starts at a maximum level (send  $-18.7 \text{ dBPa}$ , receive  $-20 \text{ dBm0}$ ), and is constantly reduced in level down to 0 and then again increased up to the maximum level. Both signal halves have a duration of 20 s. The compressor is active during the period where the output signal has a constant level. The level limits  $a_{k\max}$  and  $a_{k\min}$ , within which the compressor operates can be obtained from the original signal. The compression range is obtained from the difference  $a_{k\max} - a_{k\min}$ .

Using another test signal containing jumps, it is possible to represent the control response of a compressor, from which the attack- and decay-slope  $a_{K'r}$  and  $a_{K'r'}$  respectively can be obtained. The test signal consists of individual blocks with the activation signal (e.g. voiced sound of CSS) of 1 s duration, the level of which is first decreased and then increased in steps of 5 dB. For send the starting level is once again  $-18.7 \text{ dBPa}$  at HFRP, while for receive the starting level is  $-20 \text{ dBm0}$  (see Figure 10).

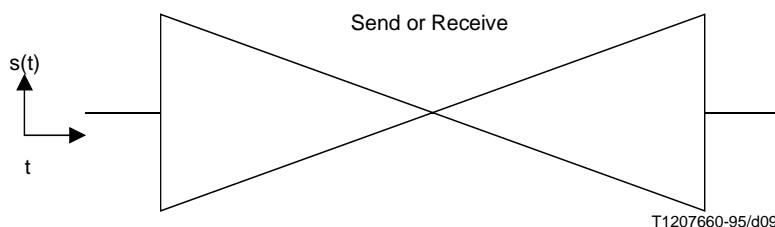


FIGURE 9/P.340  
Dynamic compressor measurement 1

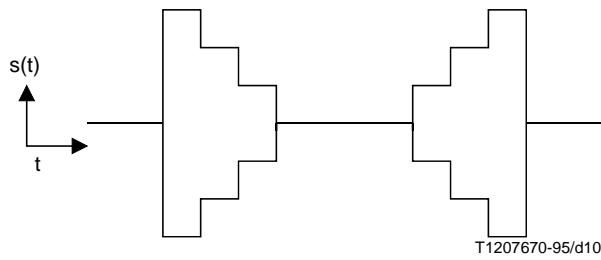


FIGURE 10/P.340  
**Dynamic compressor measurement 2**

## 7.5 Reverberance

The impulse response in the send direction is measured using maximum-length sequences, as described in [5]. The test signal should be composed of an activation signal (e.g. voiced sound of the CSS) and a maximal-length sequence of several periods (one segment is shown in Figure 11). The period length must be longer than the length at the impulse response of the system under test. The impulse response is calculated from the average of several periods, beginning with the second period. From this the Early Energy Balance (EEB) is calculated. The EEB represents fairly well the subjective impression of reverberance. It is given by [6]:

$$EEB = 10 \cdot \log \left\{ \frac{\int_0^{35 \text{ ms}} h^2(t) dt}{\int_0^{5 \text{ ms}} h^2(t) dt} \right\} \text{ dB}$$

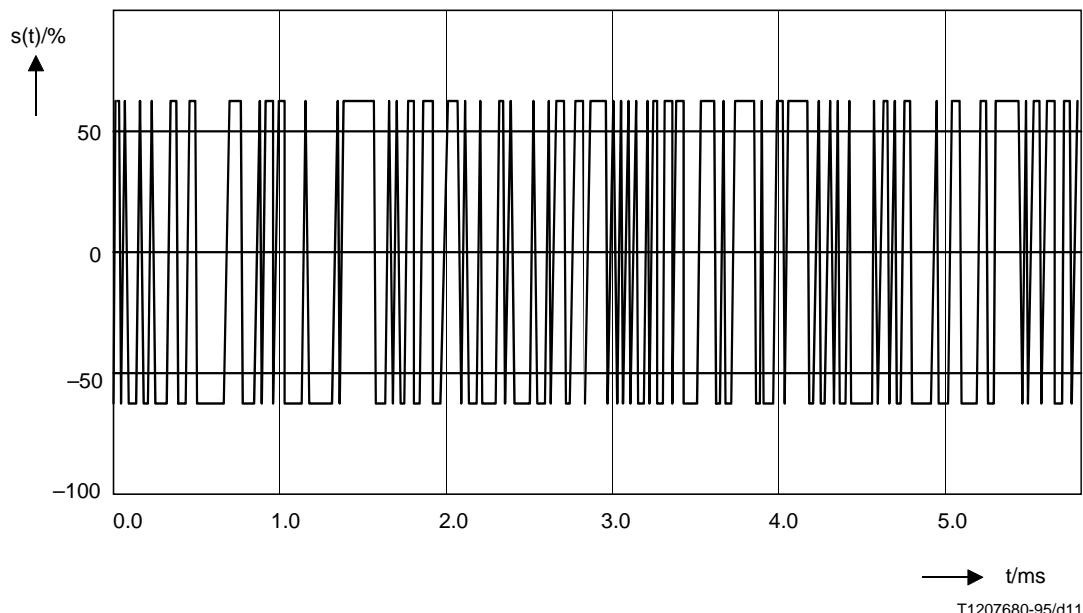


FIGURE 11/P.340  
**Segment of maximum-length sequence**

## **7.6 Time-dependent transfer function**

In analysis, the output signals are referred to the corresponding original signals. For display of the time and frequency dependent structure of the output signal referred to the input signal, a spectrographic display may be selected where the x-axis represents time, the y-axis represents the frequency and the level is represented in different colours. In this way, any possible time or frequency variant structures can be revealed.

## **7.7 Duplex behaviour**

Special duplex speech signals can be fed in both directions (send and receive) for investigating the duplex response. Appropriate signals are those of differently varying level, (e.g. speech sequence of fixed definition or sequences from CSS of fixed definition) fed in after each other with varying periodicity in the send and receive directions.

A double talk sequence as described in Recommendation P.501 should be used.

The output signals are referred to the corresponding original signals and spectrographically displayed as described in 7.6. This allows to represent how often gaps or signal interrupting occur during transmission. Conclusions can thus be reached regarding the duplex capability of the equipment.

## **7.8 Control characteristics of echo cancellation**

Subjective quality is determined by the following parameters:

- time-variable echo attenuation;
- echo attenuation over frequency;
- time response of NLP, centre clipper;
- response in duplex operation;
- attenuation response with background noise.

Until today no instrumental measurement procedure fully representing the subjective impression exists.

Thus subjective investigation is required.

## **7.9 Quality of speech detection**

Speech detection is measured in the send direction. From subjective listening and conversation tests [8], sending path appears as the more critical of the two transmission paths.

For telephones incorporating level detection, the level measured is the minimum at which the equipment switches through.

A sequence of a complex signal should be used as a test signal, whereby the level of the active part (e.g. voiced sound and PN signal of the CSS) is increased in steps of 1 dB. The signal sequence starts with a level of -58.7 dBPa. The pause between the active parts is 1 s.

The peak level (integration time  $\tau = 5$  ms) of the signal (e.g. voiced sound of the CSS) of the input signal at HFRP, at which switching through can first be detected in the output signal, must be measured. This level then corresponds to the minimum switch-on level. The signal must then be reliably switched through by all the following sequence of the signal of higher level.

## **7.10 Quality of speech detection with ambient noise**

Measurement is done in the send direction. A 10 s sequence of a complex signal should be used.

A minimum of four loudspeakers, fed with non-coherent noise signals, are simulating a room noise. They are positioned according to Figure 12. The resulting signal is measured at the point located in the middle of the test table (in absence of the table), the spectrum is a Hoth noise [7], and the level is set to -44 dBPa(A).

## 8 Classification

Hands-free terminal should be classified into three types on the basis of the important characteristic "duplex capability" and room noise transmission at idle (send). From [9], it appears that these characteristics are the most important on the perceived quality of the HFT, and that they may be characterised by the following parameters: "attenuation range"  $a_H$  and attenuation distribution at idle in the send direction  $a_{H,S}$ .

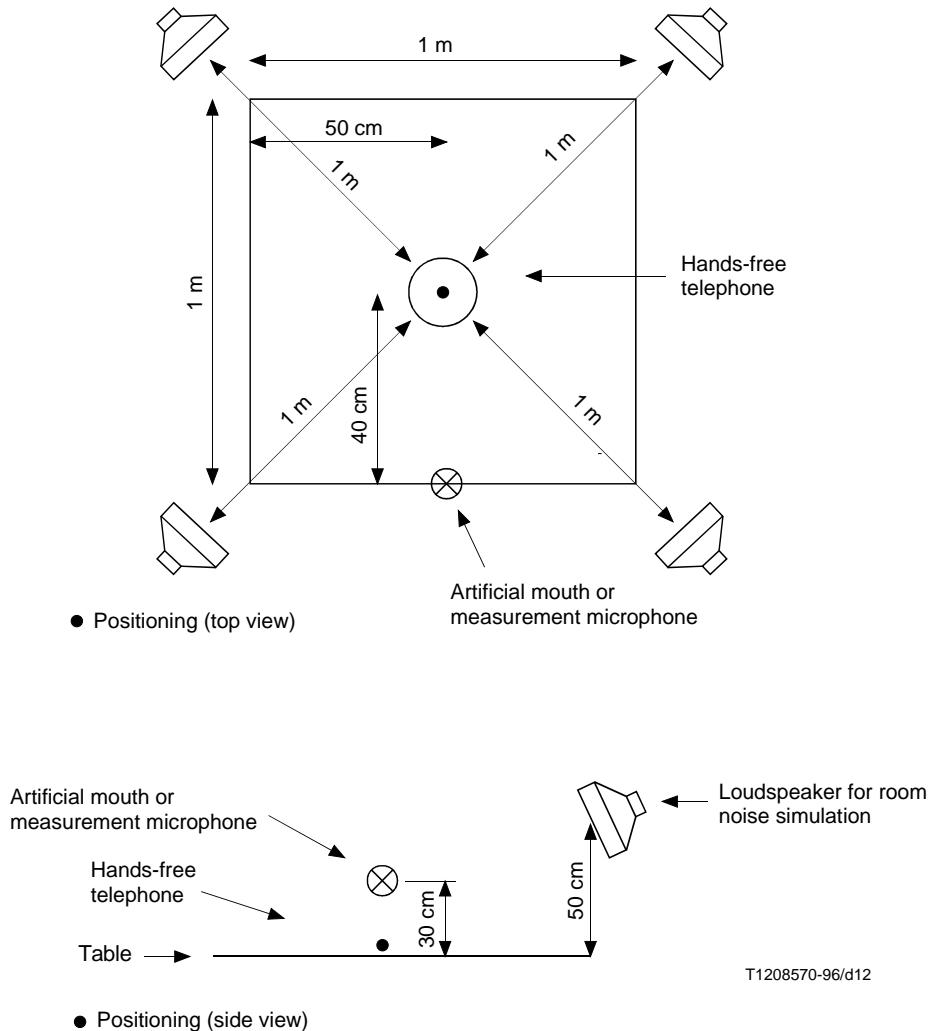


FIGURE 12/P.340  
Positioning of the equipment

In particular,  $a_H$  is directly associated to duplex capability.

The following values should be met for the different types of HFT:

$$\text{Type 1: } a_H < 3 \text{ dB} \quad a_{H,S} < 3 \text{ dB}$$

$$\text{Type 2: } 3 \text{ dB} \leq a_H \leq 20 \text{ dB} \quad a_{H,S} \leq a_H/2$$

$$\text{Type 3: } a_H > 20 \text{ dB} \quad a_{H,S} \leq a_H/2$$

## 9 Parameters to be evaluated for each type of HFT

The requirement of good duplex capability, the quality of room noise transmission in the send direction and other auditory quality factors result in specific requirements which should be met for each group. Different parameters are given priority in each type of equipment, according to the different quality level attaching to each type, each of which

can be measured to different measurement regulations. The measurements include parameters already outlined, and defined in appropriate Recommendations, as P.341, P.342, plus additional and/or adapted measurements. See Table 4.

TABLE 4/P.340

Type 1 full duplex capability	Type 2 partial duplex capability	Type 3 no duplex capability
SLR/RLR  Values and tests are available in the appropriate Recommendation	SLR/RLR  Values and tests are available in the appropriate Recommendation	SLR/RLR  Values and tests are available in the appropriate Recommendation
Distortion  Values and tests are available in the appropriate Recommendation	Distortion  Values and tests are available in the appropriate Recommendation	Distortion  Values and tests are available in the appropriate Recommendation
Noise  Measurement technique and limits are available in the appropriate Recommendation  The noise should be investigated with respect to time and frequency variable response. There should be a constant noise level showing neither peaks more than 6 dB over the average noise level in the frequency nor time variant structures of the noise. In case of time and frequency variant noise, auditory evaluation is required	Noise  Measurement technique and limits are available in the appropriate Recommendation  The noise should be investigated with respect to time and frequency variable response. There should be a constant noise level showing neither peaks more than 6 dB over the average noise level in the frequency nor time variant structures of the noise. In case of time and frequency variant noise, auditory evaluation is required	Noise  Measurement technique and limits are available in the appropriate Recommendation  The noise should be investigated with respect to time and frequency variable response. There should be a constant noise level showing neither peaks more than 6 dB over the average noise level in the frequency nor time variant structures of the noise. In case of time and frequency variant noise, auditory evaluation is required
	Quality of speech detection in the send direction  The minimum switch-on level must be < -30 dBPa at HFRP. A principle of measurement can be found in 7.9	Quality of speech detection in the send direction  The minimum switch-on level must be < -30 dBPa at HFRP. A principle of measurement can be found in 7.9
	Hang-over time (switch-off time)  The switch-off is measured according to 7.3  Requirement: $T_H$ should be greater than 50 ms, preferably more than 100 ms	Hang-over time (switch-off time)  The switch-off is measured according to 7.3  Requirement: $T_H$ should be greater than 250 ms
Echo cancellation  Auditory quality investigations required	Echo cancellation  Auditory quality investigations required	Echo cancellation  Auditory quality investigations required
		Switch-over time $T_S$ Limit: $T_S < 150$ ms
	Dynamic compression  Control through the use of a compressor should be undetectable. The compressor control range $a_{kmax} - a_{kmin}$ (see 7.4) must be small. Compressor speed $a_{Kr'}$ or $a_{Kf'}$ must be low  Exact compressor values are pending discussion. In general a requirement like the graph in Figure 13 is needed	Dynamic compression  In the case of hands-free telephones without reliable noise detection, noise control through a compressor should not be apparent. The compressor control range $a_{kmax} - a_{kmin}$ (see 7.4) must be small. The compressor control speed $a_{Kr'}$ or $a_{Kf'}$ must be low  Exact compressor values are pending discussion. In general a requirement like the graph in Figure 13 is needed

TABLE 4/P.340 (*concluded*)

Type 1 full duplex capability	Type 2 partial duplex capability	Type 3 no duplex capability
	Adaptive gain control For further study	Adaptive gain control For further study
	Build-up Time See in 4.6 and in the Handbook on Telephonometry	Build-up Time See in 4.6 and in the Handbook on Telephonometry
Reverberance Impulse response measurement according to 7.5 The EEB value should be as small as possible. A limit value is pending	Reverberance Impulse response measurement according to 7.5 The EEB value should be as small as possible. A limit value is pending	Reverberance Impulse response measurement according to 7.5 The EEB value should be as small as possible. A limit value is pending
Delay Measurement techniques are defined in the appropriate Recommendation Delay must be < (16) ms (preliminary value from Recommendation G.167)	Delay Measurement techniques are defined in the appropriate Recommendation	Delay Measurement techniques are defined in the appropriate Recommendation
Time and frequency variable transfer function  The transfer function is measured according to 7.6 and spectrographically displayed. The transfer function should show a constant frequency dependent response over time within $\pm 3$ dB, measured in 1/12 octaves  In the case of time and frequency variable response, auditory evaluation of the system is required		
		Room noise detection  Room noise detection must be reliable. Hoth noise [7] with a level $L_N$ at the hands-free equipment (HFRP) is generated. No accidental switch through of the hands-free equipment should be registered at a level of $L_N = -50$ dBPa(A) over a period of 10 s
	Quality of speech detection with ambient noise  Hoth noise with a level $L_N$ at the HFRP is generated. No accidental switch through of the hands-free equipment should be registered at a level of $L_N = -50$ dBPa(A) over a period of 10 s  The measurement setup is described in 7.10	Quality of speech detection with ambient noise  Hoth noise with a level $L_N$ at the HFRP is generated. No accidental switch through of the hands-free equipment should be registered at a level of $L_N = -50$ dBPa(A) over a period of 10 s  The measurement setup is described in 7.10

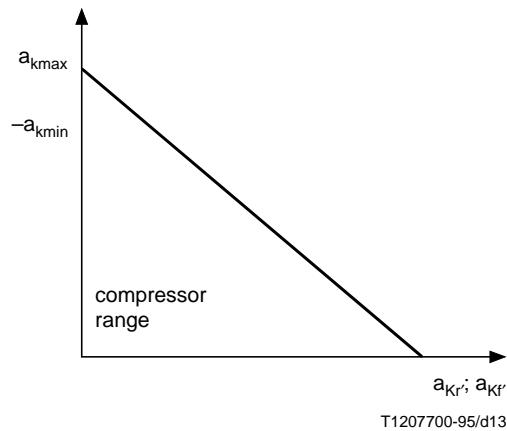


FIGURE 13/P.340  
**Dynamic compression**

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- [9] ITU-T COM 12-61 (1995), Supplementary Measurement Proposal for Certification Measurements of Hands-free Equipment, Federal Republic of Germany.

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