INTERNATIONAL TELECOMMUNICATION UNION

# ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

# P.313

(09/99)

SERIES P: TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Subscribers' lines and sets

---

## Transmission characteristics for cordless and mobile digital terminals

ITU-T Recommendation P.313

(Previously CCITT Recommendation)

---

ITU-T P-SERIES RECOMMENDATIONS

**TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS**

| | |
|---|---|
| Vocabulary and effects of transmission parameters on customer opinion of transmission quality | Series P.10 |
| **Subscribers' lines and sets** | **Series P.30** |
| | **P.300** |
| Transmission standards | P.40 |
| Objective measuring apparatus | P.50 |
| | P.500 |
| Objective electro-acoustical measurements | P.60 |
| Measurements related to speech loudness | P.70 |
| Methods for objective and subjective assessment of quality | P.80 |
| | P.800 |
| Audiovisual quality in multimedia services | P.900 |

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

**ITU-T RECOMMENDATION P.313**

# TRANSMISSION CHARACTERISTICS FOR CORDLESS AND MOBILE DIGITAL TERMINALS


## Summary

This Recommendation provides audio performance requirements for portable digital cordless and mobile handsets. The requirements apply to narrowband systems (3.1 kHz) regardless of a coding algorithm used in the terminal. Associated test methods are also given.

Requirements are specified for the major electro-acoustic performance parameters affecting audio quality, including sending and receiving levels, frequency responses, noise, sidetone, stability, echo path and delay. The requirements given in the Recommendation should ensure satisfactory service quality in a high percentage of installations under normal conditions.



## Source

ITU-T Recommendation P.313 was prepared by ITU-T Study Group 12 (1997-2000) and was approved under the WTSC Resolution No. 1 procedure on 30 September 1999.

## Keywords

Electro-acoustics measurements, mobile, performance specification, terminals, wireless.

## FOREWORD

ITU (International Telecommunication Union) is the United Nations Specialized Agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the 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.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

## NOTE

In this Recommendation the term *recognized operating agency (ROA)* includes any individual, company, corporation or governmental organization that operates a public correspondence service. The terms *Administration, ROA* and *public correspondence* are defined in the *Constitution of the ITU (Geneva, 1992)*.

## INTELLECTUAL PROPERTY RIGHTS

The ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. The ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, the ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

## CONTENTS

Page

**Recommendation P.313**

# TRANSMISSION CHARACTERISTICS FOR CORDLESS AND MOBILE DIGITAL TERMINALS

*(Geneva, 1999)*

## 1 General

### 1.1 Scope

This Recommendation deals with electro-acoustic performance parameters of portable digital cordless and mobile terminals. The requirements given below should ensure satisfactory voice service in a high percentage of installations under normal conditions, but other factors impacting the performance such as radio link are not included.

The Recommendation does not include specifications for hands-free mode. The requirements contained in this Recommendation apply only to narrowband systems (3.1 kHz) regardless of a coding algorithm used in the handset.

Requirements are given for handsets in conjunction with a 0 dBr 4-wire reference (see Figure 1) base station having an appropriate air interface and are specified irrespective of the particular technology and air interface.

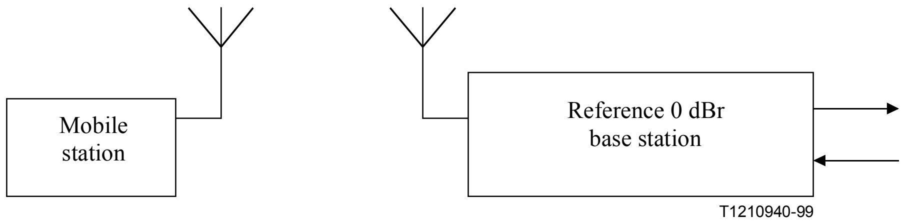


**Figure 1/P.313 – Reference configuration**

### 1.2 References

The following ITU-T 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 P.79 (1999), *Calculation of loudness ratings for telephone sets*.

[2] ITU-T Recommendation P.64 (1999), *Determination of sensitivity/frequency characteristics of local telephone systems to permit calculation of their loudness ratings*.

[3] ITU-T Recommendation P.50 (1999), *Artificial voices*.

[4] ITU-T Recommendation O.41 (1994), *Psophometer for use on telephone-type circuits*.

[5] ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections*.

[6] ITU-T Recommendation G.131 (1996), *Control of talker echo*.

[7] ITU-T Recommendation G.114 (1996), *One-way transmission time*.

[8] CCITT Recommendation G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

[9] ITU-T Recommendation P.501 (1996), *Test signals for use in telephonometry*.

[10] ITU-T Recommendation P.360 (1998), *Efficiency of devices for preventing the occurrence of excessive acoustic pressure by telephone receivers*.

[11] ITU-T Recommendation G.174 (1994), *Transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN*.

[12] ITU-T Recommendation P.57 (1996), *Artificial ears*.

[13] ISO 3:1973, *Preferred numbers – Series of preferred numbers*


### 1.3 Terms and definitions

This Recommendation defines the following term:

**1.3.1 base station**: For the purpose of this Recommendation, refers to a fixed part of any wireless system.

### 1.4 Abbreviations

This Recommendation uses the following abbreviations:

ARL Acoustic Reference Level

DRP Eardrum reference point

ERP Ear reference point

HATS Head and torso simulator

LR Loudness rating

LRGP Loudness rating guard-ring position

LSTR Listener sidetone rating

MRP Mouth reference point

RLR Receiving loudness rating

SLR Sending loudness rating

STMR Sidetone masking rating

$TCL_W$ Terminal coupling loss weighted

## 2 Sending characteristics

### 2.1 Sending loudness rating (SLR)

In view of the following considerations:

– wireless terminals provide connectivity to the existing wireline-based international telecommunication networks;

– digital wireless access networks should provide the same signal levels as the digital access wireline networks, the following loudness rating value is recommended as a long term objective:

– a nominalvalue of SLR = 8 dB.

As a short-term objective, nominal value of SLR in the range 5 to 11 dB is recommended.

NOTE – Manufacturers may want to reduce nominal transmit loudness to improve performance of the terminals in noisy conditions. This reduction may be correlated with the adjustments of the receiving loudness (as discussed in clause 3).

The SLR shall be calculated, based on the measurements described in 2.2, according to Recommendation P.79 [1], using equation 2-1,

$$LR = \frac{10}{m} \cdot \log\left\{ \sum_{i=N_1}^{N_2} 10^{0.1m(S_i - W_i)} \right\} \qquad (2\text{-}1)$$

with $m = 0.175$, over bands 4 to 17 and the send weighting factors from Table 1.

**Table 1/P.313 – Weighting factors $W_i$ for SLR and RLR**

| Band No. | Mid-frequency (Hz) | Send $W_{si}$ | Receive $W_{ri}$ |
|---|---|---|---|
| 4 | 200 | 76.9 | 85.0 |
| 5 | 250 | 62.6 | 74.7 |
| 6 | 315 | 62.0 | 79.0 |
| 7 | 400 | 44.7 | 63.7 |
| 8 | 500 | 53.1 | 73.5 |
| 9 | 630 | 48.5 | 69.1 |
| 10 | 800 | 47.6 | 68.0 |
| 11 | 1000 | 50.1 | 68.7 |
| 12 | 1250 | 59.1 | 75.1 |
| 13 | 1600 | 56.7 | 70.4 |
| 14 | 2000 | 72.2 | 81.4 |
| 15 | 2500 | 72.6 | 76.5 |
| 16 | 3150 | 89.2 | 93.3 |
| 17 | 4000 | 117.0 | 113.8 |

## 2.2 Sending frequency responses

In view of the following considerations:

– the compatibility with wireline digital telephones and the mixed analog/digital telephone network;

– the compatibility with most existing wireless systems;

– the aim to achieve the best possible overall quality with the cordless and mobile terminals,

sending nominal sensitivity/frequency response within the limits given in Table 2 is recommended.

**Table 2/P.313 – Sending**

| Frequency (Hz) | Upper Limit (dB) | Lower Limit (dB) |
|---|---|---|
| 100 | –12 | |
| 200 | 0 | –∞ |
| 300 | 0 | –14 |
| 1000 | 0 | –8 |
| 2000 | 4 | –8 |
| 3000 | 4 | –8 |
| 3400 | 4 | –11 |
| 3400 | 4 | –∞ |
| 4000 | 0 | |

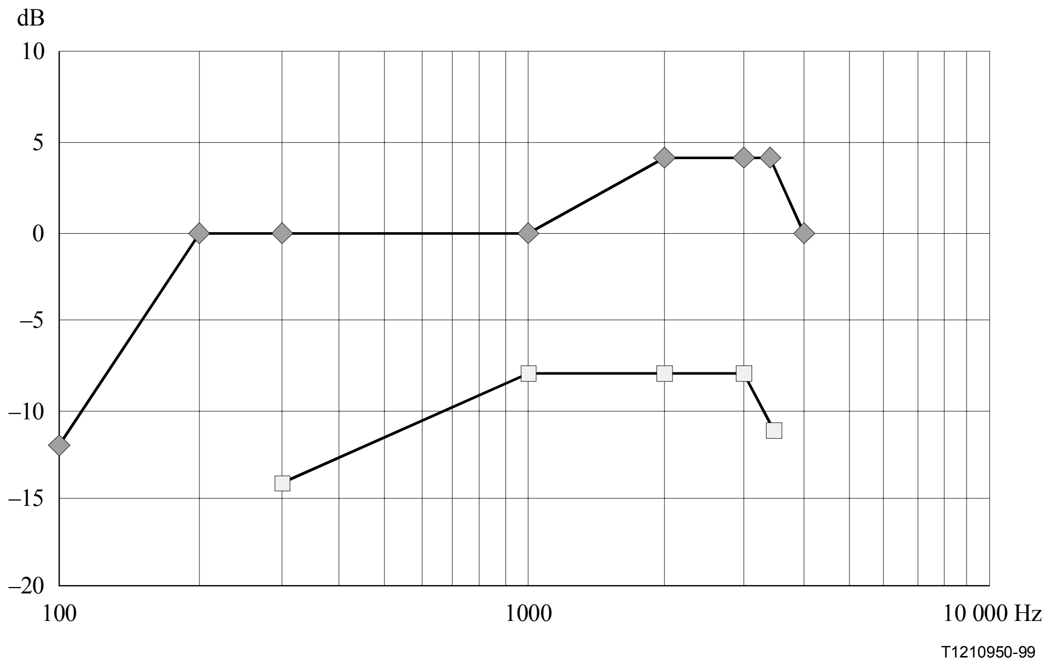


**Figure 2/P.313 – Sending mask**

### 2.2.1 Measurement method

The send frequency response is measured according to Recommendation P.64 [2] using the measurement set-up shown in Figure 3. The test signal level shall be –4.7 dBPa at the MRP.

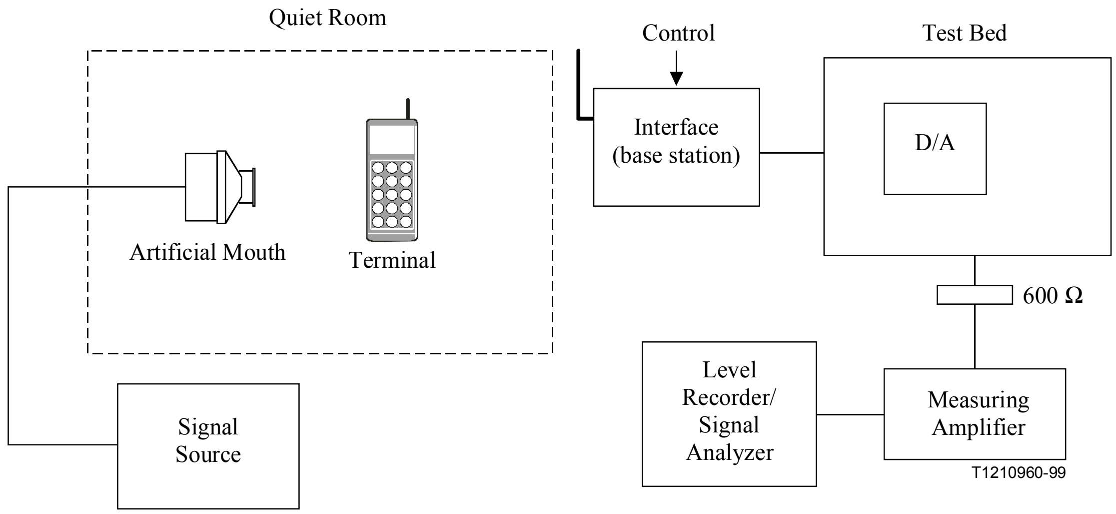


**Figure 3/P.313 – Send frequency response measurement method –
Swept sinewave technique**

If the swept sinewave techniques can not be used, other appropriate technique should be applied. For example an artificial speech generator (e.g. as specified in Recommendation P.50 [3]) and a spectrum analyzer can be applied.

## 2.3 Noise

### 2.3.1 Idle channel noise

The following limit is recommended:

– send noise level maximum −64 dBm0p.

#### 2.3.1.1 Measurement method

With the handset mounted at LRGP and the earpiece sealed to the knife-edge of the artificial ear in a quiet environment (ambient noise less than 30 dBA), the send noise level at the digital output is measured with apparatus including psophometric weighting according to Recommendation O.41 [4].

## 2.4 Non-linear distortion

(For further study.)

## 2.5 Variation of gain with input level

If the system is intended to operate in a linear fashion the following is recommended:

– the gain variation relative to the gain for Acoustic Reference Level (ARL) should remain within the limits given in Table 3.

**Table 3/P.313 – Variation of gain with input level, sending**

| Sending dB relative to ARL | Upper limit (dB) | Lower limit (dB) |
|---|---|---|
| +13 | 1 | –11 |
| +4 | 1 | –2 |
| –10 | 1 | –2 |
| –20 | 1 | –5 |
| –25 | 1 | –8 |
| –30 | 1 | –12 |
| <–30 | 6 | –∞ |

The limits for intermediate levels can be found by drawing straight lines between the break points in the table on a linear (dB signal level) – linear (dB gain) scale.

### 2.5.1 Measurement method

The handset should be mounted at the LRGP and the earpiece should be sealed to the knife-edge of the artificial ear. If the artificial ear type 3.3 or 3.4 coupler is used, the handset should be placed in the HATS position as described in Recommendation P.64 [2].

A sinewave signal with a frequency in the range 1004 Hz to 1025 Hz should be applied at the MRP. The level of this signal is adjusted until the output of the terminal is –10 dBm0. The level of the signal at the MRP is then the ARL.

NOTE 1 – In general, care must be taken in case of the timevariant and/or non linear terminals. In such cases a sine wave may not be appropriate as the test signal; a speech like test signal should be chosen as described in Recommendations P.501 [9] and P.50 [3].

The test signal should be applied at the following levels:

–30, –25, –20, –15, –10, –5, 0, 4, 10, 13 dB relative to ARL.

The variation of gain relative to the gain for the ARL should be measured.

NOTE 2 – Selective measurement may be used to avoid the effects of ambient noise.

## 3 Receiving characteristics

### 3.1 Receiving loudness rating (RLR)

In view of the following considerations:

– wireless terminals provide connectivity to the existing wireline based international telecommunication networks;

– digital wireless terminals should be compatible with the digital access wireline networks,

the following loudness rating value is recommended as a long-term objective:

– a nominal value of RLR = 2 dB.

As a short-term objective, nominal value of RLR in the range –3 to 7 dB is recommended.

The RLR shall be calculated, based on the measurements described in 3.3, according to Recommendation P.79 [1], using equation 2-1, with $m = 0.175$, over bands 4 to 17 and the receive weighting factors from Table 1. The artificial ear sensitivity shall be corrected using the leakage correction from Table 4 according to Recommendation P.79 [1].

**Table 4/P.313 – Leakage correction $L_E$ used for sealed measurements on an IRS-type receiver**

| Frequency (Hz) | $L_E$ (dB) | Frequency (Hz) | $L_E$ (dB) |
|---|---|---|---|
| 200 | 8.4 | 1000 | –2.3 |
| 250 | 4.9 | 1250 | –1.2 |
| 315 | 1.0 | 1600 | –0.1 |
| 400 | –0.7 | 2000 | 3.6 |
| 500 | –2.2 | 2500 | 7.4 |
| 630 | –2.6 | 3150 | 6.7 |
| 800 | –3.2 | 4000 | 8.8 |

### 3.2 Volume control

In view of the following considerations:

– the mobile terminals are used often in noisy environment;

– the need to provide service for people with hearing impairments,

manufacturers might implement volume control by which the receiving loudness level can be increased. It is suggested that it should allow at least 12 dB volume increase relative to the nominal value of RLR = 2 dB.

NOTE – In order to improve performance of the terminals in noisy conditions it might be of advantage to increase the SLR (reduce loudness) relative to the nominal level when the RLR is decreased (increased loudness) using volume control adjustment. This will help to reduce the sidetone level, improve listener sidetone and echo performance and lower the noise level sent to the line.

### 3.3 Receiving frequency responses

In view of the following considerations:

– the compatibility with wireline digital telephones and the mixed analog/digital telephone network;

– the compatibility with most existing wireless systems;

– the aim to achieve the best possible overall quality with the cordless and mobile terminals,

receiving nominal sensitivity/frequency response within the limits given in Table 5 is recommended.

**Table 5/P.313 – Receiving**

| Frequency (Hz) | Upper limit (dB) | Lower limit (dB) |
|---|---|---|
| 100 | –10 | |
| 200 | 2 | –∞ |
| 300 | 2 | –9 |
| 2000 | 2 | –7 |
| 3400 | 2 | –12 |
| 4000 | 2 | –∞ |
| 8000 | –18 | |

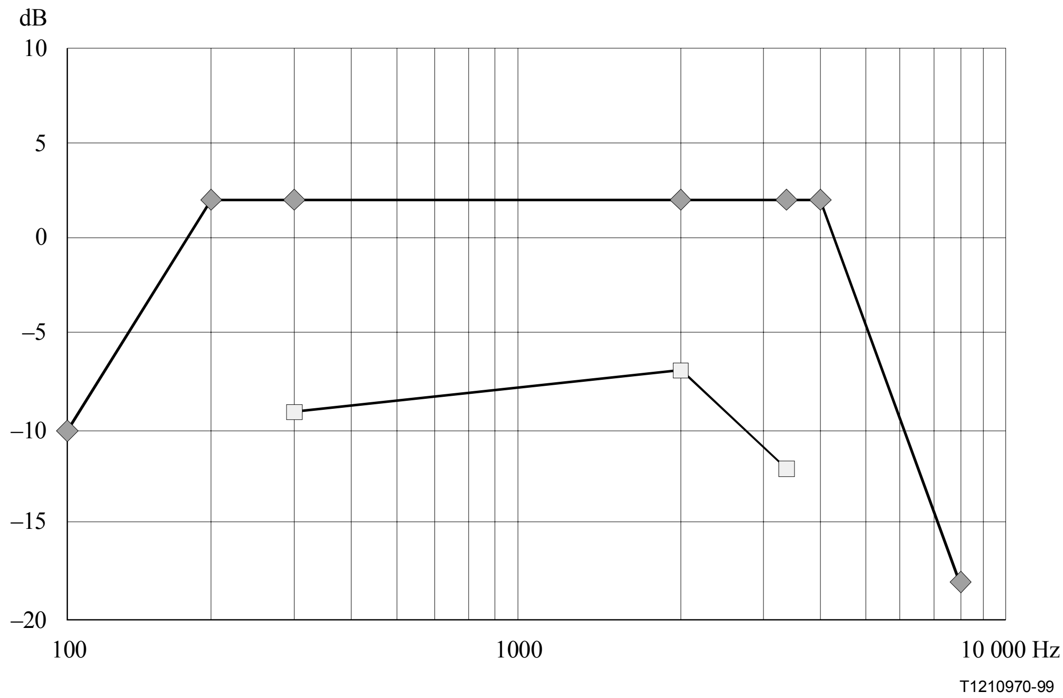


**Figure 4/P.313 – Receiving mask**

NOTE – The following rules may be additionally applied:

In general the frequency response – regardless what coupler is used for measurements – should not introduce a strong roll off at lower frequencies. Signal levels at frequencies down to 300 Hz should not be attenuated more than 5 dB as compared to the level measured at 1 kHz. Too much emphasis for high frequencies should be avoided as well. Compared to the level measured at 1 kHz, the emphasis introduced up to 3.4 kHz should not be more than 5 dB.

### 3.3.1 Measurement method

The receive frequency response is measured according to Recommendation P.64 [2] using the measurement set-up shown in Figure 5 where the artificial ear is type 1 (Recommendation P.57) unless specified differently in Recommendation P.57 [12]. The test signal level shall be –16.0 dBm0. Terminals with adjustable receive levels shall be adjusted so that their RLR is as close as possible to the nominal value of 3.1 for this test.

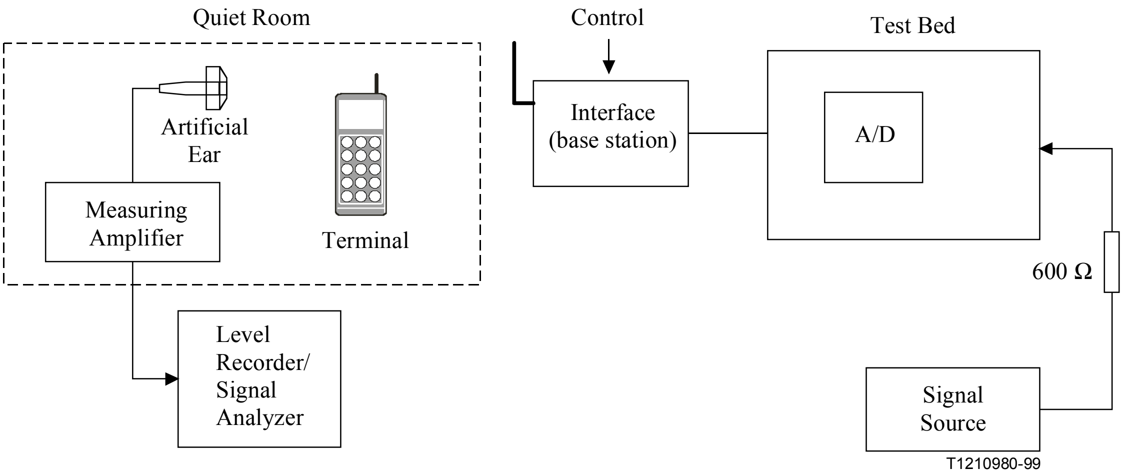


**Figure 5/P.313 – Receive frequency response measurement method – Swept sinewave technique**

If the swept sinewave techniques can not be used, other appropriate technique should be applied. For example an artificial speech generator (e.g. as specified in Recommendation P.50 [3]) and a spectrum analyzer can be applied.

## 3.4 Noise

### 3.4.1 Idle channel noise

The following limit is recommended:

– receive noise level of maximum –56 dBPa(A) at the nominal RLR value.

#### 3.4.1.1 Measurement method

To measure receive noise, a G.711 [8] PCM signal of the lowest quantized value of segment number 1 is applied at the digital interface. The A-weighted noise level is measured in the artificial ear. The ambient noise for this measurement shall not exceed 30 dBA.

Telephone sets with adjustable receive levels shall be close as possible to the nominal RLR value when driven by a G.711 [8] PCM signal of the lowest quantized value of segment number 1.

## 3.5 Non-linear distortion

(For further study.)

## 3.6 Variation of gain with input level

If the system is intended to operate in a linear fashion the following is recommended:

– the gain variation relative to the gain at an input level of –10 dBm0, should be within the limits given in Table 6.

**Table 6/P.313 – Variation of gain with input level, receiving**

| Receiving level at the digital interface | Upper limit (dB) | Lower limit (dB) |
|---|---|---|
| +3 dBm0 | 1 | –11 |
| –6 dBm0 | 1 | –2 |
| –50 dBm0 | 1 | –2 |
| –50 dBm0 | 1 | –∞ |

The limits for intermediate levels can be found by drawing straight lines between the break points in the table on a linear (dB signal level) – linear (dB gain) scale.

### 3.6.1 Measuring method

The handset should be mounted at the LRGP and the earpiece should be sealed to the knife-edge of the artificial ear. If the artificial ear type 3.3 or 3.4 coupler is used, the handset should be placed in the HATS position as described in Recommendation P.64 [2].

A digitally simulated sinewave signal with a frequency in the range 1004 Hz to 1025 Hz should be applied at the digital interface at the following levels:

–50, –45, –40, –35, –30, –25, –20, –15, –10, –6, 0, 3 dBm0.

NOTE 1 – In general, care must be taken in case of the timevariant and/or nonlinear terminals. In such cases a sine wave may not be the appropriate test signal; a more speech like test signal should be chosen as described in Recommendations P.501 [9] and P.50 [3].

The variation of gain relative to the gain at an input level of –10 dBm0 should be measured using the artificial ear.

NOTE 2 – Selective measurement may be used to avoid the effects of ambient noise.

## 4 Sidetone characteristics

### 4.1 Sidetone masking rating (STMR)

In view of the following considerations:

– the optimum STMR for conditions free from echo;

– the difficulties of high ambient noise conditions,

the following is recommended:

– the value of the STMR shall be in the range of 10 dB to 18 dB, when corrected to eliminate SLR and RLR manufacturing tolerances.

Where a user-controlled receiving volume adjustment is provided, the STMR shall meet the requirement given above at the setting where the RLR is equal to the nominal value (2 dB).

### 4.1.1 Measurement method

The handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A test signal level of –4.7 dBPa shall be applied at the MRP. For each frequency given in Table 7, bands 1 to 20, the sound pressure (at the ERP) shall be measured.

**Table 7/P.313 – Weighting factors $W_{MSi}$ for STMR**

| Band No. | Mid-frequency (Hz) | $W_{MSi}$ |
|---|---|---|
| (1) | | (2) |
| 1 | 100 | 110.4 |
| 2 | 125 | 107.7 |
| 3 | 160 | 104.6 |
| 4 | 200 | 98.4 |
| 5 | 250 | 94.0 |
| 6 | 315 | 89.8 |
| 7 | 400 | 84.8 |
| 8 | 500 | 75.5 |
| 9 | 630 | 66.0 |
| 10 | 800 | 57.1 |
| 11 | 1000 | 49.1 |
| 12 | 1250 | 50.6 |
| 13 | 1600 | 51.0 |
| 14 | 2000 | 51.9 |
| 15 | 2500 | 51.3 |
| 16 | 3150 | 50.6 |
| 17 | 4000 | 51.0 |
| 18 | 5000 | 49.7 |
| 19 | 6300 | 50.0 |
| 20 | 8000 | 52.8 |

The test set-up shown in Figure 6 is used to measure the sidetone frequency response. The sidetone path loss $L_{meST}$ and the STMR shall be calculated according to Recommendation P.79 [1], using equation 2-1 ($m = 0.225$) and the weighting factors in Table 7.

Other than the swept sinewave techniques can be used. For example an artificial speech generator (e.g. as specified in Recommendations P.50 [3] and P.501 [9]) and a spectrum analyzer can be applied.

## 4.2 Listener sidetone rating (LSTR) and D-factor

In view of the following considerations:

– mobile sets are being used often in noisy environment;

– the difficulties of high ambient noise conditions,

the following is recommended:

– when the ambient noise level is –34 dBPa(A) or higher, the value of the LSTR shall not be less than 15 dB when corrected to eliminate SLR and RLR manufacturing tolerances.

The provisional recommendation for the D-factor is as follows:

– the value of the weighted average D ("D-factor") of the difference of the send sensitivities between direct and diffuse sound shall not be less then 0 dB. As the long-term objective the value of +3 dB is recommended.

NOTE 1 – The key parameter for the handset performance in noisy conditions is the LSTR limit. However, the D-factor may be important in cases when LSTR can not be measured. This is the case when couplers type 3.2, 3.3 or 3.4 are used.

NOTE 2 – Terminals designed for quiet environments (e.g. some indoor applications) may have lower LSTR and D-factor limits, but the LSTR value should not be less than 10 dB and the D value should not be less than –3dB.

### 4.2.1 Measurement method

#### 4.2.1.1 Listener sidetone (LSTR)

The listener sidetone frequency characteristic is measured using the set-up shown in Figure 6 except no signal is generated by the artificial mouth and the measurement is performed by a spectrum analyzer. The diffuse sound field should be calibrated in the absence of any local obstacles. When measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20), the averaged field shall be uniform to within +4 dB/–2 dB within a radius of 0.15 m of the MRP.

**Error! Objects cannot be created from editing field codes.**

**Figure 6/P.313 – Sidetone frequency response measurement method – Swept sinewave technique**

A calibrated half-inch microphone is mounted at MRP. The sound field is measured in one-third octave bands. The spectrum shall be "Pink noise" to within ±1 dB and the level shall be adjusted to 70 dBA (–24 dBPa(A)). Tolerance: ±1 dB.

The artificial mouth and ear are placed in the correct position relative to MRP, the handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

Measurements are made in one-third octave bands for the 20 bands centred at 100 Hz to 8 kHz (bands 1 to 20). For each band the sound pressure at the ERP shall be measured.

The listener sidetone path loss and the LSTR shall be calculated according to equation 2-1, and the weighting factors in Table 7 ($m = 0.225$).

If the terminal supports user-controlled receiving volume control, the LSTR shall meet the required value at the setting where the RLR is equal to the nominal value (2 dB).

#### 4.2.1.2 D-factor

For the weighted average D ("D-factor") of the difference of the send sensitivities between direct and diffuse sound the diffuse sound sensitivities shall be used for the calculation as $S_{si}$(diff) at 20 bands from 100 Hz to 8 kHz. The sending sensitivities for the direct sound $S_{si}$(direct) shall be measured according to the measurements of the sending frequency response, but at one-third octave bands for 20 bands centred at 100 Hz to 8 kHz with the test signal "pink noise". The D-factor is computed with $S_{si}$(diff) and $S_{si}$(direct) from formulas E-3/P.79 [1] and E-2/P.79 [1] and the coefficients $K_i$ in Table E.1/P.79.

## 5 Noise contrast and comfort noise

In some circumstances, such as application of voice-operated devices, the continuous background noise present regardless of whether the users are talking or not may be interrupted. This switching on and off is annoying to the users and may in fact degrade speech intelligibility. To reduce this effect, noise contrast should be minimized by increasing signal-to-noise ratio.

Comfort noise may be injected during silent periods to reduce the impairments created by the noise contrast. This may create undesirable performance degradation by itself if not done properly, due to the level or spectrum contents differences between the injected and the transmitted noise. Effort should be made to match the characteristics of the injected comfort noise to the transmitted noise to reduce any perceptible contrast between them.

## 6 Weighted terminal coupling loss ($TCL_w$)

In the view of the following consideration:

– the aim to achieve as high coupling loss as possible to minimize degradation caused by echo,

the following is recommended:

– weighted terminal coupling loss $TCL_w$ should be greater than 45 dB when measured under the free field conditions and normalized to the nominal LR values of SLR = 8 dB and RLR = 2 dB.

### 6.1 Measurement method

$TCL_w$ is measured in free air in such a way that the inherent mechanical coupling of the handset in not affected.

Noise and reflections in the test space must not influence the measurement. The test should be performed in an anechoic chamber (with free-field condition at least down to 275 Hz) with the handset positioned at least 50 cm away from the nearest part of the test chamber (a handset may be suspended as shown in Figure 7). The ambient noise level shall be less than 30 dBA.

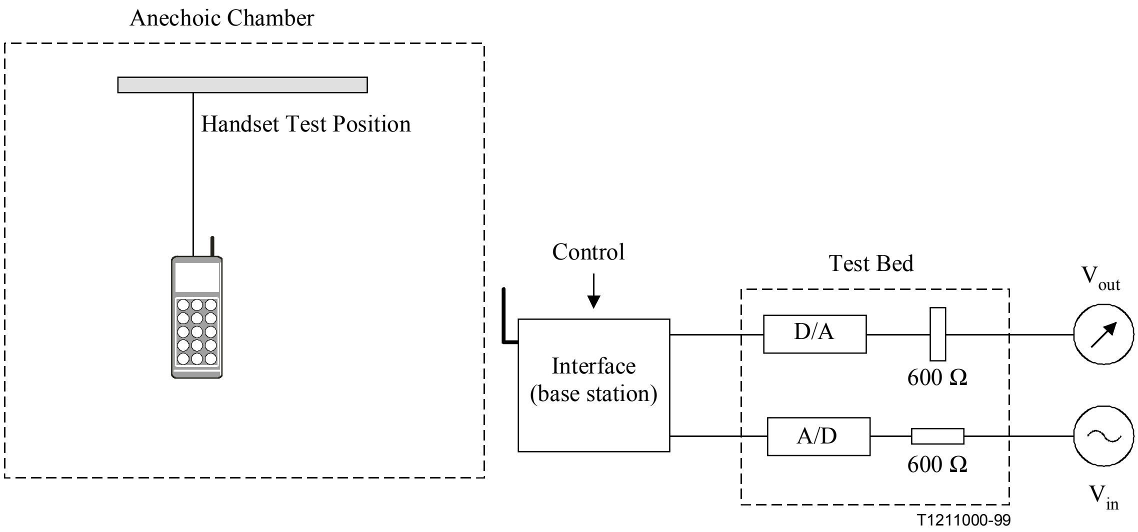


**Figure 7/P.313 – Terminal coupling loss measurement method**

The attenuation from digital input to digital output is measured at one-twelfth octave frequencies as given by the R.40-series of preferred numbers in ISO 3 [13] for frequencies from 300 to 3350 Hz, using the measurement arrangement shown in Figure 7.

The weighted terminal coupling loss is calculated according to B.4/G.122 [5] (trapezoidal rule).

Terminals with adjustable receive levels shall be tested at the nominal setting. For the nominal setting, adjust the level so that the RLR is as close as possible to the nominal RLR value.

NOTE – There might be problems to measure 45 dB TCL in case sophisticated coding with limited dynamic range is used. In those cases, typically speech or speech-like test signals need to be used that themselves have crest factors in the range of 15 dB which reduces the measurement dynamic by the same amount. In those cases, the signal measured in sending direction should be evaluated more carefully in order to find whether an echo signal is present or whether this signal is completely masked by the noise signal introduced by the codec. If the signal measured in the sending direction is completely masked by the noise, the requirement can be considered to be fulfilled. If this is not the case, more sophisticated measurement procedures such as time averaging (in order to improve the signal to noise ratio) need to be applied in order to achieve reliable measurement results.

## 7 Stability loss

The following limit is recommended:

– with the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 10 dB at all frequencies in the range of 200 Hz to 4 kHz with LRs normalized to nominal values;

– the minimum stability loss at any volume control setting should be at least 6 dB.

### 7.1 Measurement method

The stability measurement is made at an input signal of –10.0 dBm0, at one-twelfth octave intervals for frequencies from 200 Hz to 4 kHz. With the handset and transmission circuit fully active, measure the attenuation from the digital input to the digital output using Method 1 and Method 2.

#### 7.1.1 Method 1

Place the handset in the reference corner, as shown in Figure 8, with the earcap and mouthpiece facing a hard, smooth surface. The handset shall be placed along the diagonal from the apex of the reference corner to the outside corner, with the earcap end of the handset 250 mm from the apex. The telephone set shall be fully active.

The reference corner consists of three perpendicular plane, smooth, hard surfaces extending 0.5 m from the apex of the corner.

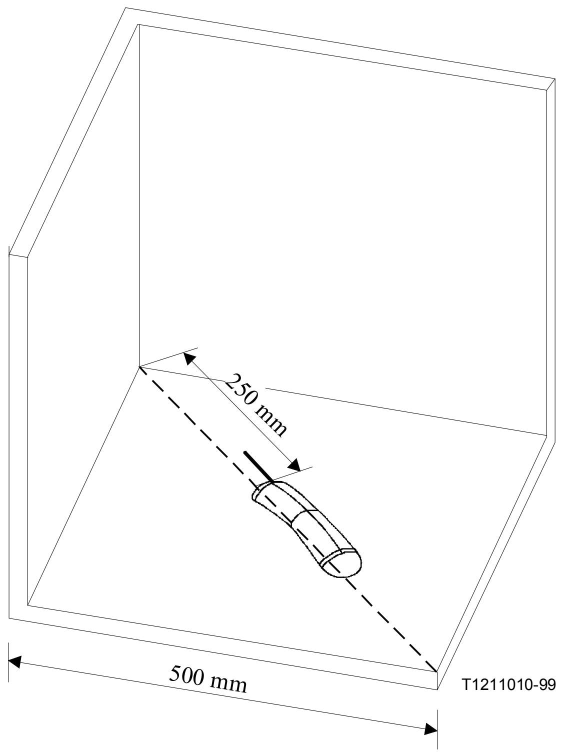


**Figure 8/P.313 – Reference corner**

### 7.1.2 Method 2

Place the handset with the earcap and mouthpiece facing a hard, smooth surface free of any other object for 0.5 m.

## 8 Delay

In view of the following considerations:

– that delay has impact on echo performance and the dynamics of voice conversation;

– the amount of delay introduced by wireless systems depends on specific technology and may be inherent to the adopted coding technique,

the following is recommended:

– delay added by the terminal equipment should be minimized in accordance to the guidelines provided in Recommendation G.114 [7] even with the use of echo control;

– the sum group delays, from the mouth reference point to the digital interface and from the digital interface to the ear reference point, of less than 20 ms delay is desirable (Recommendation G.174 [11]);

NOTE – It is recognized that some existing systems will not meet the above limit.

– terminal manufacturers must ensure that appropriate echo control measures are in place according to the guidelines provided in Recommendation G.131 [6]. This may include, for example, meeting limits specified in clause 6.

## 9 Out-of-band signals

(For further study.)

## 10 Crosstalk ratio

(For further study.)

## 11 Speech clipping

In view of the following considerations:

– that wireless systems may employ a variety of speech interpolation techniques as well as are susceptible to bursts of errors in the radio channel;

– excessive loss of speech signal may affect quality of a connection;

– subjective impact of clipping depends upon duration of clipping, percentage of speech clipped, frequency of clipping and overall speech activity,

the following is recommended for speech clipping, i.e. loss of speech:

– no speech loss occurrences longer than 64 ms should be present;

– speech loss periods shorter than 64 ms should be kept below 0.2 percent of active speech.

NOTE – Percent of clipped speech is 100 times the product of the frequency of speech clipping times clipping duration, divided by the speech activity factor.

## 12 Maximum steady state acoustic pressure

In order to ensure safety and to minimize annoyance to the user, the terminal shall meet limits specified in Recommendation P.360 [10]. If the terminal can operate in other than handset (private) modes, such as handsfree or monitoring, a safety mechanism should be implemented to ensure that the P.360 limits are never exceeded when the operation reverts to the handset mode.

This Recommendation applies also to all tones and audio signals generated by the terminal.

### 12.1 Measurement method

The maximum steady state acoustic pressure is measured by applying maximum positive digital code to the receive input defined for the handset under test. The test procedure is in accordance with 3.3.1 for the receive characteristics except that the acoustic pressure in the artificial ear is measured with a sound level meter.

Terminals with adjustable receive levels shall be tested at the maximum setting.

# ANNEX A

## Test equipment requirements

### A.1 Electro-acoustic equipment

It is recognized that for most handset designs the applicable artificial ear is type 1 (Recommendation P.57). However, when type 1 artificial ear is not applicable it is advisable to check the terminal performance with other types of artificial ears specified in Recommendation P.57 [12] such as types 3.2, 3.3 or 3.4.

When using a type 1 or 3.2 artificial ear, the handset is mounted in the LRGP position, as described in Recommendation P.64 [2].

When type 3.3 or 3.4 artificial ear is used, the handset is mounted on the HATS as described in Annex D/P.64 or E/P.64.

The sound pressure measurements shall be referred to the Ear Reference Point (ERP) by the correction characteristic specified in Recommendation P.57.

When type 3.2, type 3.3 or type 3.4 artificial ear is used, no leakage correction shall be made in the calculations of RLR and STMR (i.e. $L_E = 0$).

### A.2 Test signals

In general, the test signals as described in this Recommendation should be applied. The use of the proposed test signals requires a linear and time invariant operation of the equipment under test. This cannot be ensured in all cases. For devices where the transmission properties are level and signal-dependent, alternative test signals should be chosen. In this case a more speech-like test signal such as described in Recommendations P.50 [3] and P.501 [9] should be applied. The use of the alternative test signals should be stated in the test report. Test house and manufacturer should ensure that the appropriate type of test signal is chosen.

### A.3 Accuracy of test equipment

Unless specified otherwise, the accuracy of the measurements made by the test equipment shall be better than given in Table A.1.

**Table A.1/P.313**

| Item | Accuracy |
|---|---|
| Electrical Signal Power | ±0.2 dB for levels ≥ –50dBm |
| Electrical Signal Power | ±0.4 dB for levels < –50dBm |
| Sound pressure | ±0.7 dB |
| Time | ±5% |
| Frequency | ±0.2% |
| **Quantity** | **Accuracy** |
| Sound pressure level at MRP | ±3 dB for 100 Hz to 200 kHz<br>±1 dB for 200 Hz to 4 Hz<br>±3 dB for 4 kHz to 8 kHz |
| Electrical excitation levels | ±0.4 dB (see Note 1) |
| Frequency generation | ±0.2% (see Note 2) |
| NOTE 1 – Across the whole frequency range.<br>NOTE 2 – When measuring sampled systems, it is advisable to avoid measuring at sub-multiples frequency. There is a tolerance of ±2% on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where the only –2% tolerance may be used. | |

## ITU-T RECOMMENDATIONS SERIES

| | |
|---|---|
| Series A | Organisation of the work of the ITU-T |
| Series B | Means of expression: definitions, symbols, classification |
| Series C | General telecommunication statistics |
| Series D | General tariff principles |
| Series E | Overall network operation, telephone service, service operation and human factors |
| Series F | Non-telephone telecommunication services |
| Series G | Transmission systems and media, digital systems and networks |
| Series H | Audiovisual and multimedia systems |
| Series I | Integrated services digital network |
| Series J | Transmission of television, sound programme and other multimedia signals |
| Series K | Protection against interference |
| Series L | Construction, installation and protection of cables and other elements of outside plant |
| Series M | Tmn and network maintenance: international transmission systems, telephone circuits, telegraphy, facsimile and leased circuits |
| Series N | Maintenance: international sound programme and television transmission circuits |
| Series O | Specifications of measuring equipment |
| **Series P** | **Telephone transmission quality, telephone installations, local line networks** |
| Series Q | Switching and signalling |
| Series R | Telegraph transmission |
| Series S | Telegraph services terminal equipment |
| Series T | Terminals for telematic services |
| Series U | Telegraph switching |
| Series V | Data communication over the telephone network |
| Series X | Data networks and open system communications |
| Series Y | Global information infrastructure and internet protocol aspects |
| Series Z | Languages and general software aspects for telecommunication systems |