INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

P.313

(05/2004)

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

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	Series	P.40
Objective measuring apparatus	Series	P.50 P.500
Objective electro-acoustical measurements	Series	P.60
Measurements related to speech loudness	Series	P.70
Methods for objective and subjective assessment of quality	Series	P.80 P.800
Audiovisual quality in multimedia services	Series	P.900

For further details, please refer to the list of ITU-T 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 and hands-free terminals. The requirements apply to narrow-band systems (3.1 kHz) regardless of the 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.

Changes over the previous ITU-T Rec. P.313 (1999) are as follows:

- Specifications for hands-free terminals have been added. These requirements are intended for mobile terminals in the car environment.
- For handsets fitted with a volume control, the maximum weighted Terminal Coupling Loss (TCLw) requirement of handsets with a volume control has been established at not less than 40 dB for the higher gain settings above the nominal setting of the volume control.
- The receiving mask for handsets has been slightly modified to align with other mobile standards.
- Out-of-band singles requirement for handsets has been added.
- The Sidetone Masking Rating (STMR) requirement is modified. The STMR range is increased from 10 dB to 18 dB, to a range of 10 dB to 20 dB.

Testing methods for handsets have been updated.

This revision provides, for the first time, an ITU standard for transmission characteristics of hands-free mobile terminals. Changes in the handset requirements should improve quality and increase usage with updated testing methods.

Source

ITU-T Recommendation P.313 was approved on 14 May 2004 by ITU-T Study Group 12 (2001-2004) under the ITU-T Recommendation A.8 procedure.

Keywords

Cordless, electro-acoustic measurements, hands-free, mobile, performance specification, terminals, wireless.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. 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 Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 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 expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. 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, 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

ITU-T Recommendation P.313

Transmission characteristics for cordless and mobile digital terminals

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 the radio link, are not included.

The Recommendation does include specifications for hands-free mode of mobile handsets. These specifications may be applicable to digital cordless hands-free terminals. The requirements contained in this Recommendation apply only to narrow-band systems (3.1 kHz) regardless of the 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.

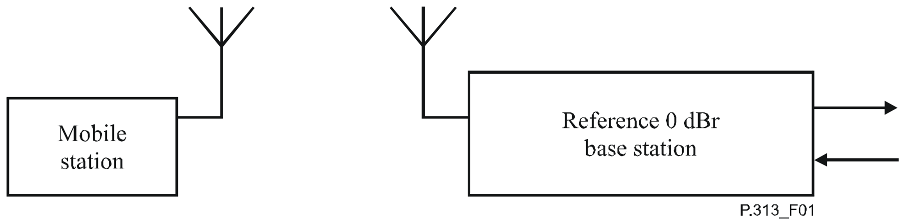

Figure 1/P.313 – Reference configuration

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; 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. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[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*.

[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 (2003), *Talker echo and its control*.

[7] ITU-T Recommendation G.114 (2003), *One-way transmission time*.

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

[9] ITU-T Recommendation P.501 (2000), *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 (2002), *Artificial ears*.

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

[14] ITU-T Recommendation P.330 (2003), *Speech processing devices for acoustic enhancement.*

[15] ITU-T Recommendation P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals.*

[16] ITU-T Recommendation P.581 (2000), *Use of head and torso simulator (HATS) for hands-free terminal testing.*

[17] ITU-T Recommendation P.832 (2000), *Subjective performance evaluation of hands-free terminals.*

[18] ITU-T Recommendation P.502 (2000), *Objective test methods for speech communication systems using complex test signals.*

[19] ITU-T Recommendation P.342 (2000), *Transmission characteristics for telephone band (300-3400 Hz) digital loudspeaking and hands-free telephony terminals*.

[20] ITU-T Recommendation G.168 (2004), *Digital network echo cancellers*.

1.3 Abbreviations

This Recommendation uses the following abbreviations:

ARL	Acoustic Reference Level
Ardt	Attenuation range in receiving direction during double talk
Asdt	Attenuation range in sending direction during double talk
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
POI	Point of Interconnection
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
STMR	SideTone Masking Rating
TCLw	Terminal Coupling Loss weighted
TCLwdt	Weighted terminal coupling loss during double talk

TCLwst	Weighted terminal coupling loss during single talk
TELR	Talker Echo Loudness Rating
TELRDT	Talker Echo Loudness Rating during double talk
TRst-r	Build-up time, single talk, receive signal
TRst-s	Build-up time, single talk, send signal
VAD	Voice Activity Detector

2 Handset technical requirements

2.1 Sending characteristics

2.1.1 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:

– a nominal value of SLR = 8 dB.

NOTE – Several reports have indicated that sending speech levels in mobile networks are 5 dB higher on average than speech levels from fixed networks. The most likely cause is that mobile terminals are frequently used in noisy environments where users tend to talk louder. Manufacturers of mobile handsets should be aware that this may lead to saturation, clipping, and distortion of the signal and reduced speech quality. Changing SLR at this time is not recommended due to concerns of reduced speech quality when used at normal talker levels. However, the manufacturers of devices intended for use in noisy conditions can consider adopting a higher SLR.

The SLR shall be calculated, based on the measurements described in 2.1.2, according to ITU-T Rec. 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/P.79.

2.1.2 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 1 is recommended.

Table 1/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	–∞

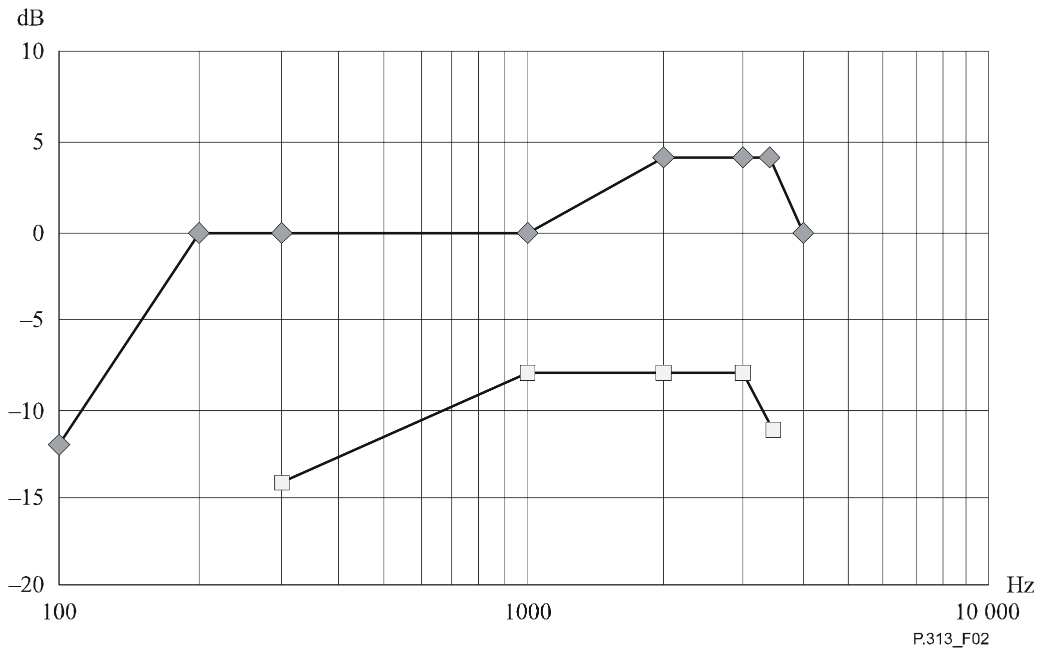

Figure 2/P.313 – Sending mask

2.1.2.1 Measurement method

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

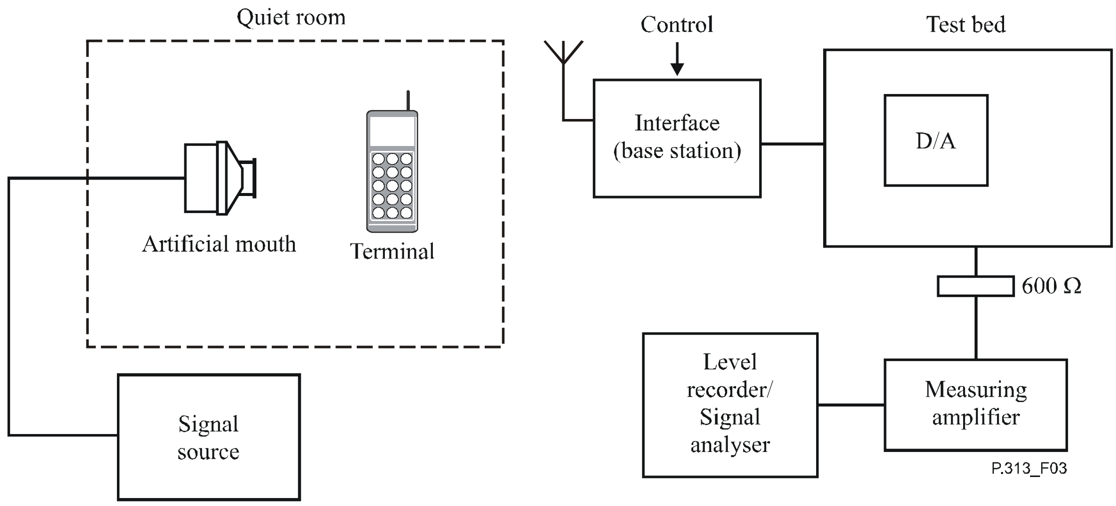

**Figure 3/P.313 – Send frequency response measurement method –
Swept sinewave technique**

If the swept sinewave techniques cannot be used (as is likely to be the case if the terminal has a noise reduction device), other appropriate techniques should be applied. For example, an artificial speech generator (e.g. as specified in ITU-T Recs P.50 [3] and P.501 [9]) and a spectrum analyser can be applied. Additional test methods may be found in ITU-T Rec. P.502 [18]. The test signal used shall be specified in the test report.

2.1.3 Noise

2.1.3.1 Idle channel noise

The following limit is recommended:

– send noise level maximum −64 dBm0p.

2.1.3.1.1 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 ITU-T Rec. O.41 [4].

2.1.4 Non-linear distortion

(For further study.)

2.1.5 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 2.

Table 2/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.

2.1.5.1 Measurement method

The handset should be mounted at test position specified in ITU-T Rec. P.64 using a suitable ear simulator as recommended in ITU-T Rec. P.57.

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 time variant 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 ITU-T Recs P.501 [9] and P.50 [3]. If the swept sinewave techniques cannot be used (such is likely the case if the terminal has a noise reduction device), other appropriate technique should be applied. For example, an artificial speech generator (e.g. as specified in ITU-T Recs P.50 [3] and P.501 [9]) and a spectrum analyser can be applied. Additional test methods may be found in ITU-T Rec. P.502 [18]. The test signal used shall be specified in the test report.

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.

2.2 Receiving characteristics

2.2.1 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:

– a nominal value of RLR = 2 dB.

The RLR shall be calculated, based on the measurements described in 2.2.3, according to ITU-T Rec. P.79 [1], using Equation 2-1, with $m = 0.175$, over bands 4 to 17 and the receive weighting factors from Table 1/P.79. The artificial ear sensitivity shall be corrected using the leakage correction from Table 2/P.79.

2.2.2 Volume control

In view of the following considerations of mobile handsets:

– 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.

2.2.3 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 3 is recommended.

Table 3/P.313 – Receiving

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	–10	–∞
200	2	–∞
300	2	–9
1000	2	–7
3400	2	–12
4000	2	–∞
8000	–18	–∞

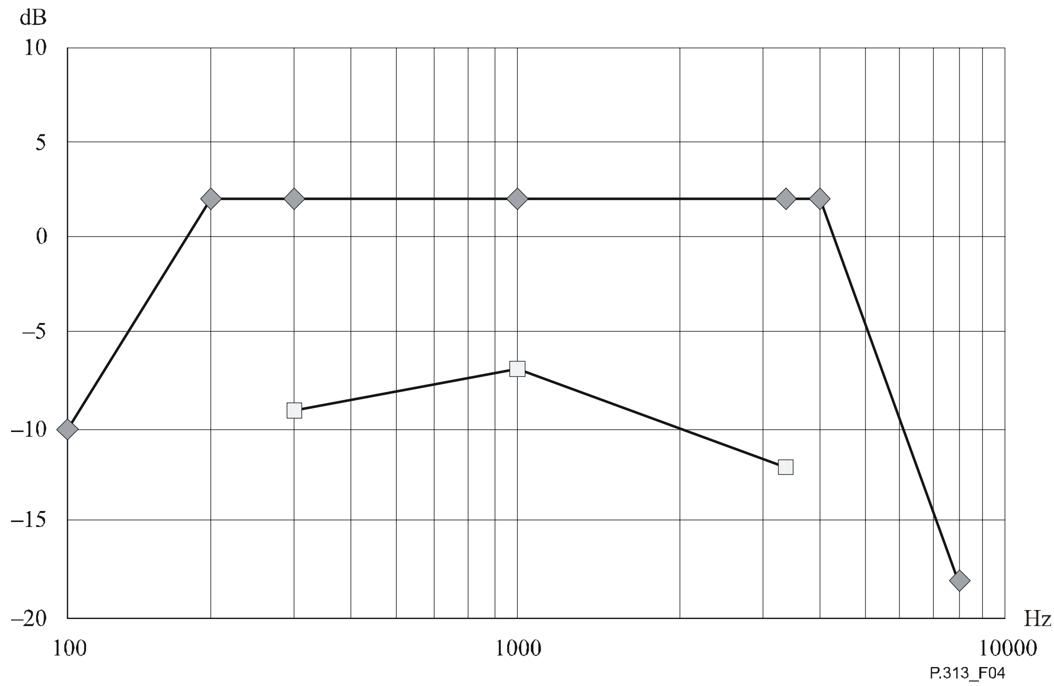

Figure 4/P.313 – Receiving mask

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.

2.2.3.1 Measurement method

The receive frequency response is measured according to ITU-T Rec. P.64 [2] using the measurement set-up shown in Figure 5 using a suitable ear simulator as recommended in ITU-T Rec. P.57. 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 2.2.1 for this test.

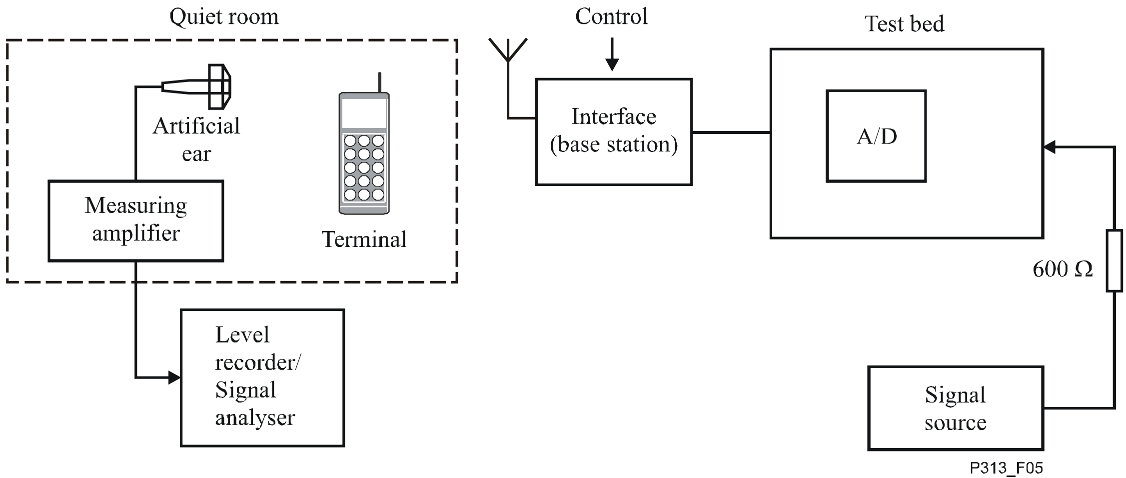

Figure 5/P.313 – Receive frequency response measurement method – Swept sinewave technique

If the swept sinewave techniques cannot be used, other appropriate techniques should be applied. For example, an artificial speech generator (e.g., as specified in ITU-T Recs P.50 [3] and P.501 [9]) and a spectrum analyser can be applied. The test signal used shall be specified in the test report.

2.2.4 Noise

2.2.4.1 Idle channel noise

The following limit is recommended:

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

2.2.4.1.1 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 during this measurement shall not exceed 30 dBA.

Telephone sets with adjustable receive levels shall be set as close as possible to the nominal RLR value.

2.2.5 Non-linear distortion

(For further study.)

2.2.6 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 4.

Table 4/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.

2.2.6.1 Measuring method

The handset should be mounted at test position specified in ITU-T Rec. P.64 using a suitable ear simulator as recommended in ITU-T Rec. P.57.

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 time variant and/or non-linear 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 ITU-T Recs P.501 [9] and P.50 [3]. The test signal used shall be specified in the test report.

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.

2.3 Sidetone characteristics

2.3.1 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 20 dB.

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).

2.3.1.1 Measurement methods

The Type 3.2 low leak artificial ear is recommended. A test signal level of –4.7 dBPa shall be applied at the MRP. For each frequency given in Table 3/P.79, bands 1 to 20, the sound pressure (at the ERP) shall be measured.

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 ITU-T Rec. P.79 [1], using Equation 2-1 ($m = 0.225$) and the weighting factors in Table 3/P.79.

Other than the swept sinewave techniques can be used. For example, an artificial speech generator (e.g., as specified in ITU-T Recs P.50 [3] and P.501 [9]) and a spectrum analyser can be applied. Additional test methods may be found in ITU-T Rec. P.502 [18]. The test signal used shall be specified in the test report.

NOTE – Use of HATS is under further study.

2.3.2 Listener sidetone rating (LSTR) and D-factor

In view of the following considerations:

– mobile sets are being used often in noisy environments;

– 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 value of the weighted average D ("D-factor") of the difference of the send sensitivities between direct and diffuse sound shall not be less than 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 cannot be measured. This is the case when artificial ears 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 –3 dB.

2.3.2.1 Measurement method

2.3.2.1.1 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 analyser. 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.

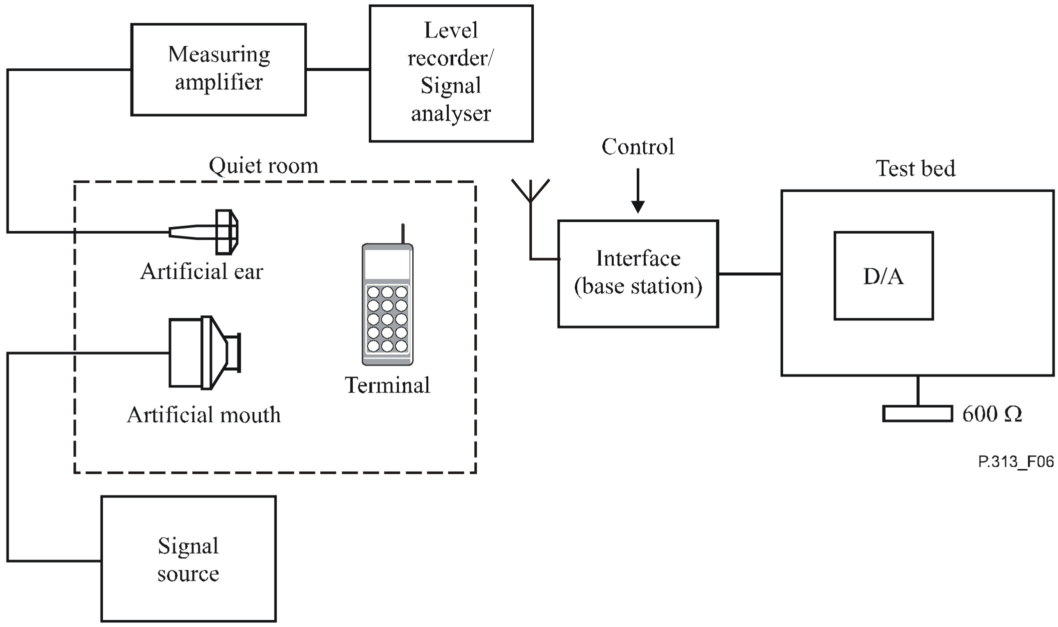

Figure 6/P.313 – Sidetone frequency response measurement method – Swept sinewave technique

The Type 3.2 low leak artificial ear is recommended. 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 3/P.79 ($m = 0.225$).

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).

NOTE – Use of HATS is under further study. Adjustments to the LSTR may have to be made when using HATS.

2.3.2.1.2 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.

2.4 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.

2.5 Weighted terminal coupling loss (TCLw)

In view of the following considerations:

– the aim to achieve as high an acoustic coupling loss as possible to minimize degradation caused by echo;

– that the far end talker echo should be controlled under all volume control settings, and for the range of transducers sensitivities as long as the handset is properly used;

– the far end terminal may be connected via a mobile or an IP network which introduces long talker echo path delay;

– what is practically obtainable in real use where the customer himself chooses the way to hold his handset,

the following limit is recommended:

In order to meet the G.131 [6] talker echo objective requirements, the weighted Terminal Coupling Loss (TCLw) should be greater than 45 dB when measured under free field conditions and with LR normalized to SLR = +8 dB and RLR = +2 dB. For example, if the measured TCLw is 48 dB, the measured SLR is +9 dB and the measured RLR is +3 dB, then the normalized value of TCLw = 48 dB + (8 – 9) dB + (2 – 3) dB = 46 dB.

NOTE – In consideration of the increasing delays introduced by modern networks, a higher TCLw value than that here specified may be necessary for proper operation with these networks.

For handsets fitted with a volume control, the TCLw shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control. This TCLw value shall be normalized by considering the SLR and RLR values measured with the volume control at its nominal position.

2.5.1 Measurement method

TCL_w is measured in free air in such a way that the inherent mechanical coupling of the handset is 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.

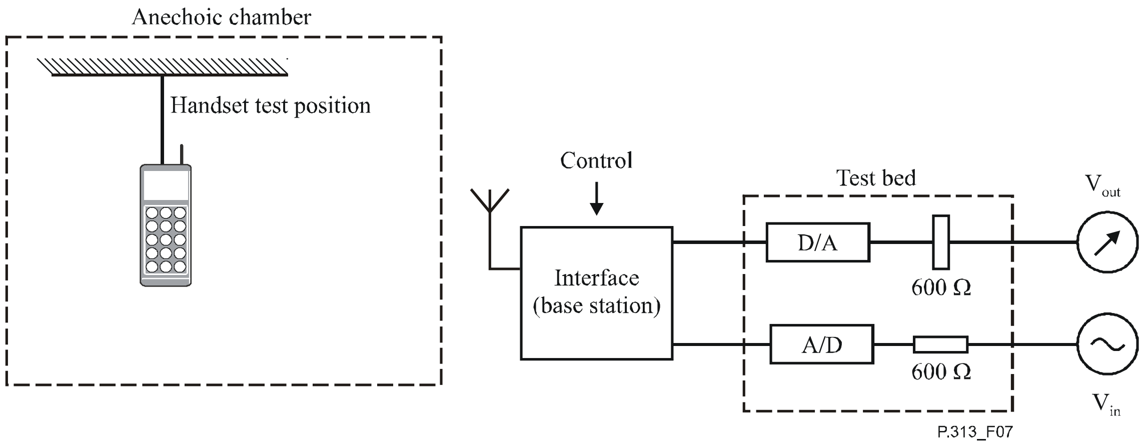

Figure 7/P.313 – Terminal coupling loss measurement method

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 measuring 45 dB TCL in the case where sophisticated coding with limited dynamic range is used. In such 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 such cases, the signal measured in the 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.

2.6 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.

2.6.1 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.

2.6.1.1 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, smooth, hard surfaces of perpendicular planes extending 0.5 m from the apex of the corner.

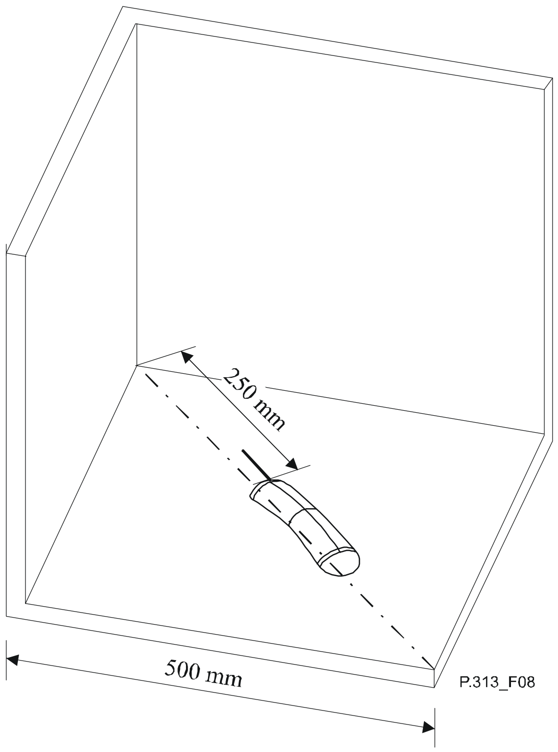

Figure 8/P.313 – Reference corner

2.6.1.2 Method 2

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

2.7 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 with the guidelines provided in ITU-T Rec. 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 (ITU-T Rec. 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 ITU-T Rec. G.131 [6]. This may include, for example, meeting limits specified in 2.5.

2.8 Out-of-band signals

Any spurious out-of-band image signals in the frequency range from 4.6 to 8 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in Table 5.

Table 5/P.313 – Out-of-band signal limit, receiving

Frequency (kHz)	Signal limit (dB)
4.6	35
8	45
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) – logarithmic (kHz) scale.	

2.9 Crosstalk ratio

(For further study.)

2.10 Speech clipping

In view of the following considerations:

– that wireless systems may employ a variety of speech interpolation techniques as well as being 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.

2.11 Maximum steady state acoustic pressure

In order to ensure safety and to minimize annoyance to the user, the terminal shall meet limits specified in ITU-T Rec. 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.

2.11.1 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 2.2.3.1 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.

3 Hands-free technical requirements

The most common type of mobile hands-free terminal is the type found in cars. These are either designed and built into the vehicle by the automaker or purchased and installed later. These types of hands-free terminals cannot easily be removed and transported to another vehicle. (However, many are designed so that a mobile handset device can connect to the hands-free hardware in the vehicle by means of a cable or wireless connection.) These requirements are, therefore, intended for the car environment. Test setup in the car for the following transmission characteristics are described in Annex A.

Refer to ITU-T Recs P.340 and P.342 for transmission characteristics and test methods of mobile and cordless terminals intended for home or office use.

Hands-free terminals use speech processing devices for acoustic enhancement (SPDA) to control acoustic echo, reduce background noise transmission, etc. These parameters are defined in ITU-T Rec. P.330.

Appropriate test signals are described in ITU-T Rec. P.501. Test methods appropriate for parameters defined in this Recommendation may be found in ITU-T Rec. P.502. Proper use of HATS testing may be found in ITU-T Rec. P.581. Measurement methods for the transmission characteristics defined below are not specified in this Recommendation.

3.1 Hands-free sending characteristics

3.1.1 Sending loudness rating (SLR)

According to ITU-T Rec. P.340 [15], the SLR of a hands-free telephone should be about 5 dB higher than the SLR of the corresponding handset telephone.

Therefore, the nominal value of SLR shall be +13 dB.

NOTE – It is recognized that some portable hands-free systems may have some variation in SLR from car-to-car due to variation of microphone position and vehicle acoustics.

3.1.2 Sending frequency response

In view of the following considerations:

– ideally, the frequency response should be flat for optimal voice quality;

– the acoustics of the particular automobile play an important role;

– the effect of the talker position inside the vehicle;

– the high levels of noise in an automobile, particularly at frequencies below 500 Hz;

– many manufacturers use microphones and analog filters to reduce such noise;

the sending tolerance mask for the nominal sensitivity/frequency response mask shown in Table 6 is recommended. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) – linear (dB sensitivity) scale.

Table 6/P.313 – Hands-free sending frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
200	0	–∞
250	0	–∞
315	0	–14
400	0	–13
500	0	–12
630	0	–11
800	0	–10
1000	0	–8
1300	2	–8
1600	3	–8
2000	4	–8
2500	4	–8
3100	4	–8
4000	0	–∞
NOTE – As stated in ITU-T Rec. P.340, the interval between 200-300 Hz makes a significant contribution to the naturalness of the transmitted speech. However, the noise energy inside a car in that frequency range might be significantly higher than the speech energy. Therefore, the manufacturers should consider this trade-off between naturalness of the transmitted speech and background noise transmitted when determining the optimal frequency response in the sending direction.		

3.1.3 Idle channel noise

The following limit is recommended:

– send noise level maximum –64 dBm0p.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

3.1.4 Non-linear distortion

When used in an automobile, the microphone and front-end circuitry of mobile hands-free terminals with AEC should be linear up to 100 dB_{SPL} with less than 1% harmonic distortion. A test signal of 1 kHz and evaluation of the first 3 harmonics, using narrow-band FFT (32 Hz resolution or less) is recommended. Requirements for non-linear distortion of the entire terminal in the sending direction are for further study.

3.1.5 Out-of-band signals in sending direction

With any signal above 4.6 kHz and up to 8 kHz, the level of any image frequency shall be below the level obtained for the reference signal by at least the amount (in dB) specified in Table 7.

Table 7/P.313 – Out-of-band signal limit, sending

Frequency (kHz)	Signal limit (dB)
4.6	30
8	40
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) – logarithmic (kHz) scale.	

3.2 Hands-free receiving characteristics

3.2.1 Receiving loudness rating (RLR)

The nominal value of RLR shall be +2 dB.

Due to the varying levels of background noise in an automobile, it is recommended that a user-specific volume control is provided. The RLR value shall be met for at least one setting of the volume control.

This value is derived from ITU-T Rec. P.310. According to ITU-T Rec. P.340 [15], the volume control range should span the value of the receiving loudness rating which is equal to that of the corresponding handset telephone, as well as an RLR value about 10 dB lower.

For car hands-free implementations the received level is typically amplified by the car's audio system in a mobile hands-free terminal. It is preferable that the car's rear loudspeakers are disabled in hands-free mode, so that the received signal is only emitted from one or two loudspeakers in the front half of the car. To improve the control of acoustic echo as well as to ensure safety, a maximum acoustic output level, corresponding to a minimum RLR is recommended for car applications. As a short-term objective, the minimum RLR shall be –18 dB. This corresponds to a 20 dB gain over nominal levels.

3.2.2 Receiving frequency response

In view of the following considerations:

– ideally, the frequency response should be flat for optimal voice quality;

– the acoustics and loudspeakers of the particular automobile play an important role;

– the effect of the listener position inside the vehicle;

– the high levels of noise in an automobile, particularly at frequencies below 500 Hz, may necessitate an emphasis of higher frequencies to improve intelligibility;

the receiving tolerance mask for the nominal frequency response mask shown in Table 8 is recommended. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) – linear (dB sensitivity) scale.

Table 8/P.313 – Hands-free receiving frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
200	0	–∞
250	0	–∞
315	0	–15
400	0	–12
500	0	–12
630	0	–12
800	0	–12
1000	0	–12
1300	2	–12
1600	3	–12
2000	4	–12
2500	4	–12
3100	4	–12
4000	0	–∞
NOTE – Due to the fact that the acoustics of the car and its loudspeakers are beyond the control of the hands-free terminal manufacturers, it may not be possible to meet the above tolerance mask without the use of some type of equalizer. The trade-off of adding an equalizer is increased delay in the receiving direction.		

3.2.3 Idle channel noise

The following limit is recommended at the nominal RLR setting:

– receive noise level maximum –53 dBPa(A).

Spectral peaks shall be less than 10 dB higher than the average noise spectrum. A narrow-band FFT analysis with a frequency resolution of 32 Hz or less is recommended.

The noise produced by the hands-free terminal providing volume control shall be less than –45 dBPa(A) with the volume control set to its maximum level (minimum RLR setting).

3.2.4 Non-linear distortion

Non-linear distortion of the received level should be as low as possible. It is recommended that the distortion limit for the audio components (including loudspeaker(s)) of the hands-free terminal should be less than 1% at the maximum volume setting (minimum RLR). A test signal of 1 kHz and evaluation of the first 3 harmonics, using narrow-band FFT (32 Hz resolution or less) is recommended. This is not only for optimal voice quality, but for optimal performance of the AEC within the hands-free terminal. Requirements for non-linear distortion in the receiving direction for the entire terminal are for further study.

3.2.5 Variation of gain with input level

The audio components (including loudspeaker(s)) of the hands-free terminal should operate in a linear fashion. This is not only for optimal voice quality, but for optimal performance of the AEC within the hands-free terminal. Specific requirements are for further study.

3.2.6 Out-of-band signals in receiving direction

Any spurious out-of-band image signals in the frequency range from 4.6 to 8 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in Table 9.

Table 9/P.313 – Out-of-band signal limit, receiving

Frequency (kHz)	Signal limit (dB)
4.6	35
8	45
NOTE – The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) – logarithmic (kHz) scale.	

3.3 Weighted terminal coupling loss

In order to meet the G.131 [6] talker echo objective requirements, the weighted terminal coupling loss during single talk (TCLwst) should be greater than 40 dB when measured under free field conditions and with LR normalized to SLR = +13 dB and RLR = +2 dB. For example, if the measured TCLw is 48 dB, the measured SLR is +14 dB and the measured RLR is +3 dB, then the normalized value of TCLw = 48 dB + (13 – 14) dB + (2 – 3) dB = 46 dB.

For terminals fitted with a volume control the TCLwst shall be not less than 35 dB for the higher gain settings above the nominal setting of the volume control. This TCLw value shall be normalized by considering the SLR and RLR values measured with the volume control at its nominal position. Permanently increasing SLR as the RLR is decreased when increasing the receive volume level is not recommended.

The limits for weighted terminal coupling loss during double talk (TCLwdt) are defined in 3.5.

3.4 Switching characteristics

Many mobile hands-free terminals use a Voice Activity Detector (VAD) and a loss controller to control acoustic echo. The VAD distinguishes silent periods (no active speech signal), single talk periods (near-end speech periods or far-end speech periods) and double-talk periods (near-end and far-end speech signals active at the same time). According to the state determined by the VAD, the loss controller reduces the acoustic echo level by inserting variable losses on the received and/or transmitted audio signals. (The VAD and loss controller may also be used to control electrical echo from the network if the mobile terminal operates on an analog wireless network without echo control.) An AEC and/or NLP may also be used to control acoustic echo. Yet most devices with an AEC and/or NLP also require a loss controller to control residual acoustic echo. Although no VAD is perfect and instantaneous, errors by the VAD may lead to speech clipping and acoustic echo.

The switching parameters and switching functions are described in detail in ITU-T Recs P.330 and P.340.

The minimum activation level in the sending direction shall be ≤ –20 dBPa. The build-up time for activation in the sending direction shall be:

$$TRst\text{-}s \leq 50 \text{ ms}$$

The minimum activation level in the receiving direction shall be ≤ –35.7 dBm0. The build-up time for activation in the receiving direction shall be:

$$TRst\text{-}r \leq 50 \text{ ms}$$

Requirements for the attenuation and build-up time for receive-to-send states (and send-to-receive states) are for further study.

3.5 Double talk performance

The speech quality during double talk is mainly determined by two parameters: Talker Echo Loudness Rating (impairment caused by echo during double talk, and related to TCLwdt) and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions the TELR should be high and the attenuation inserted should be as low as possible. Terminals that do not allow double talk should provide a good echo attenuation realized by high attenuation.

The most important parameters to determine the quality during double talk are as follows:

– Attenuation range in sending direction during double talk, Asdt.

– Attenuation range in receiving direction during double talk, Ardt.

– Echo attenuation during double talk.

The requirements for Asdt and Ardt at nominal RLR for each category of hands-free terminals are provided in Table 4/P.340. Higher values for Asdt and Ardt are permitted at lower RLR (due to increasing the received volume level) without affecting the Behaviour categorization. However, the TELRDT must maintain the requirements in ITU-T Rec. P.340 for all received volume settings.

Subjective evaluation methods for double talk quality, as well as single talk quality, are described in ITU-T Rec. P.832.

NOTE – Acoustic echo control is more easily and effectively done in terminals. However, network equipment, as described in ITU-T Rec. G.168 [20], may also include acoustic echo control processing. This leads to tandeming issues of acoustic echo control functionalities. As specified in Rec. G.168 [20], the added network component shall prevent any degradation of the overall perceived quality. In practice, however, double-talk capability may be reduced due to network equipment.

3.6 Background noise transmission and comfort noise injection

Mobile hands-free terminals are typically used in noisy environments. When used in a car, background noise levels may be simulated as described in Annex A.

Most hands-free terminals have some type of noise reduction capability. The main purpose of a noise reduction (NR) system in a device is to reduce the annoying and fatiguing effects of the transmitted background noise. The techniques used to reduce background noise may be classified as analog only, digital only, and combined analog and digital techniques. These techniques are described in ITU-T Rec. P.330.

Some parameters describing NR and some requirements are also given in ITU-T Rec. P.330. There is usually a trade-off between the level of digital noise reduction and the voice quality. Too much noise reduction may reduce the transmitted voice level and distort the speech signal.

Without comfort noise, the acoustic echo control system (loss controller, VAD, NLP, and AEC) will reduce and/or distort the background noise transmitted during receive and double talk states. This can be annoying to the far-end talker. Some hands-free terminals inject comfort noise in order to maintain a constant level of background noise transmitted. Some parameters for comfort noise are described in ITU-T Rec. P.330.

The level of comfort noise should be within a range of +2 and –5 dB from the original (transmitted) background noise.

3.7 Delay

Additional processing delay is usually required in a hands-free terminal in order to control acoustic echo and noise. Other acoustic enhancement features, such as a receive-end equalizer, may further increase delay.

It is recommended that the delay in the sending direction due to the terminal is ≤ 30 ms. This is measured from the MRP to the POI (electrical reference point), while subtracting the system delay of the network simulator.

It is recommended that the delay in the receiving direction due to the terminal is ≤ 30 ms. This is measured from the POI to the DRP (drum reference point) while subtracting the system delay of the network simulator.

Annex A

Information on testing a hands-free mobile terminal in a car

A.1 Electro-acoustic equipment

The test equipment includes a network simulator, Head and Torso Simulator (HATS), and audio measurement system. The artificial mouth of the HATS should be calibrated and equalized at the mouth reference point (MRP).

All settings of the network simulator have to ensure that the audio signal is not disturbed by any processing, and that the transmission of the HF signal is error-free.

A.2 Test arrangement in a car

The transmission performance of car hands-free terminals is measured in a car cabin. The acoustical interface for hands-free terminals is realized by using a HATS according to ITU-T Rec. P.581 [16]. The HATS should be positioned in the driver's seat at the same head position as an average size adult. The position of the HATS (mouth/ears) should be recorded in detail so that it can be repositioned in the same place for later testing. The hands-free terminal (including loudspeaker and microphone) should be installed according to the requirements of the manufacturer.

For the background noise transmission testing, real binaural car noise recordings can be made and played back to simulate real car noise in a controlled environment. The simulated noise should match closely the real car noise recorded by the HATS and should be equalized so that the third octave levels in bands between 100 Hz and 10 kHz shall not deviate by more than ±3 dB from the original spectrum. The maximum deviation of the A-weighted sound pressure level shall be ±1 dB. (This may be difficult to achieve in small car cabins.) At a minimum, background noise simulation for maximum highway speed in the concerned region should be used. Binaural recordings and playback are preferable to monaural recordings. In addition, the simulated noise at the hands-free microphone should match in level and in spectrum to the original car noise. The background noise tests may also be performed under vehicle operating conditions.

A.3 Test signals and methods

Because the transmission properties of most hands-free terminals are level and signal-dependent, complex and speech-like test signals should be applied when relevant. These test signals can be found in ITU-T Recs P.50 [3] and P.501 [9]. Associated test methods may be found in ITU-T Rec. P.502 [18]. The use of the test signals and the specific method applied for each test needs to be stated in the test report.

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of 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	Cable networks and 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, Internet protocol aspects and Next Generation Networks
Series Z	Languages and general software aspects for telecommunication systems

Printed in Switzerland
Geneva, 2005