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TELEPHONE TRANSMISSION QUALITY
SUBSCRIBERS' LINES AND SETS

TRANSMISSION CHARACTERISTICS FOR DIGITAL TELEPHONES

ITU-T Recommendation P.31

(Previously "CCITT Recommendation")

FOREWORD

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

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

ITU-T Recommendation P.31 was revised by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

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NOTES

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 Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

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

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

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CONTENTS

INTRODUCTION

This Recommendation deals with sending and receiving loudness ratings, sidetone masking rating, listener sidetone rating, sending and receiving sensitivity/frequency characteristics, noise and distortion characteristics, out-of-band signals, TCLw stability loss and delay of telephony 3.1 kHz terminals. Supplement No. 22 of P-Series Recommendations provides preliminary audio performance requirements for telephony 7 kHz terminals while the annex to the Supplement provides objective measuring methods for telephony 7 kHz terminals. Other important characteristics are still under study.

TRANSMISSION CHARACTERISTICS FOR DIGITAL TELEPHONES

(Melbourne, 1988; amended in Helsinki 1993)

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

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

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

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

NOTES

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

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

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

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

In view of the following considerations:

– the optimum STMR for conditions free from echo;

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

– the difficulties of high ambient noise conditions;

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

the following values are recommended[1)]:

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

– nominal values of LSTR >15 dB,

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

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

3 Sending and receiving sensitivity frequency characteristics for digital telephones

In view of the following considerations:

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

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

– the aim to achieve the best possible overall quality with the digital telephone,

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

sending and receiving sensitivity/frequency characteristics as specified below are recommended:

– a substantially flat receiving frequency response S_{JE} between 300 Hz and 3400 Hz should be chosen;

– a nominal sending frequency response S_{MJ} rising with a slope within the area indicated in Figure 1 should be striven for;

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

NOTES

1 S_{JE} and S_{MJ} are normally estimated from measurements of S_{Je} and S_{mJ} according to Recommendation P.66.

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

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

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

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

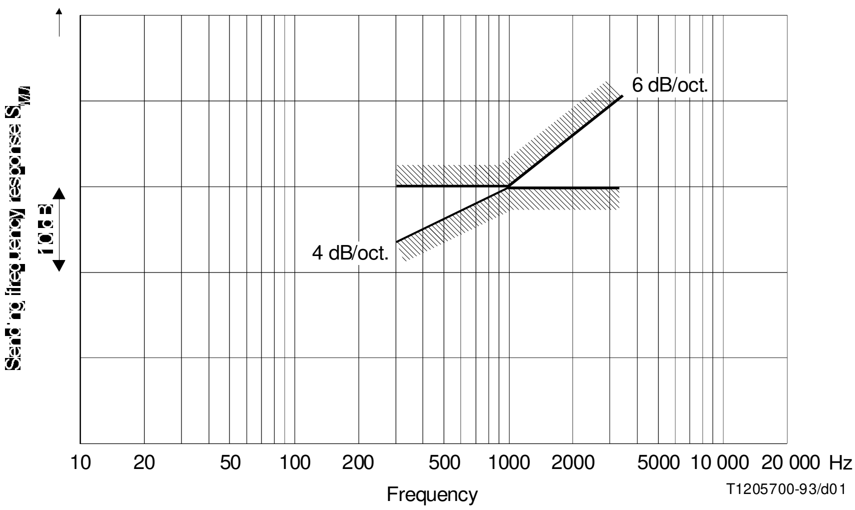

FIGURE 1/P.31

4 Noise characteristics at sending and receiving

In view of the following considerations:

– the compatibility with coder and decoder requirement according to Recommendation G.714;

– certain additions of noise must be allowed for in the electrical and acoustical parts;

– the compatibility with existing analogue telephones,

the following limits are recommended:

– sending noise level maximum –64 dBm0p;

– receiving noise level maximum 38 dB (A) if no user-controlled volume control is provided or when the volume control is set to nominal RLR value when driven by a PCM signal corresponding to the decoder output value No. 1 for A-law and 0 for µ-law.

NOTE – The noise levels are related to the long-term objective for SLR and RLR.

Compliance should be checked according to Recommendation P.66.

5 Distortion characteristics at sending and receiving

In view of the following considerations:

- compatibility with coder and decoder requirement according to Recommendation G.714;
- certain additions of distortion must be allowed for in the electrical and acoustical parts;
- compatibility with existing analogue telephones,

the following limits are recommended:

Two different sets of values are recommended relating to two different measuring methods (see 14/G.714). Either is acceptable.

Method 1 sending (Noise method)

The ratio of signal-to-total distortion (harmonic and quantizing) power of the digitally encoded signal output by the terminal equipment shall be above the limits given in Table 1, unless the sound pressure at the MRP exceeds +5 dBPa.

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) – linear (dB ratio) scale.

Compliance shall be checked by the test described in Recommendation P.66.

TABLE 1/P.31

Limits for signal-to-total distortion ratio for method 1

Sending level dB relative to ARL	Receiving level at the digital interface	Sending ratio (dB) A-law µ-law	Receiving ratio (dB) A-law µ-law
–45	–55 dBm0	5.0	5.0
–30	–40 dBm0	20.0	20.0
–24	–34 dBm0	25.5	25.0
–17	–27 dBm0	30.2 [a)]	30.6 [a)]
–10	–20 dBm0	32.4	33.0
0	–10 dBm0	33.0	33.7
+4	–6 dBm0	33.0	33.8
+7	–3 dBm0	23.5	24.0

a) Means that values for µ-law need to be added.

Method 1 receiving

The ratio of signal-to-total distortion (harmonic and quantizing) power of the signal in the artificial ear shall be above the limits given in Table 1 above unless the signal in the artificial ear exceeds +5 dBPa or is less than –50 dBPa.

Compliance shall be checked by the test described in Recommendation P.66.

Method 2 sending

The ratio of signal-to-total distortion power measured with the proper noise weighting (see Table 4/G.223) shall be above the limits given in Table 2 unless the sound pressure at MRP exceeds +10 dBPa.

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) – linear (dB ratio) scale.

Compliance shall be checked by the test described in Recommendation P.66.

TABLE 2/P.31

Limits for signal-to-total distortion ratio for method 2

Sending level dB relative to ARL A-law	µ-law	Receiving level at the digital interface		Sending ratio (dB) A-law	µ-law	Receiving ratio (dB) A-law	µ-law
–35	–37.5	–45 dBm0	–47.5	17.5	20	17.5	20
–30	–32.5	–40 dBm0	–42.5	22.5	24	22.5	24
–20	–22.5	–30 dBm0	–32.5	30.7	30	30.5	30
–10		–20 dBm0		33.3	30	33.0	30
0		–10 dBm0		33.7	30	33.5	
+7		–3 dBm0		31.7		31.2	
+10	+7.5	0 dBm0	–2.5	25.5	30	25.5	30

Method 2 receiving

The ratio of signal-to-total distortion power measured in the artificial ear with the proper noise weighting (see Table 4/G.223) shall be above the limits given in Table 1 unless the signal in the artificial ear exceeds +10 dBPa or is less than –50 dBPa.

Compliance shall be checked by the test described in Recommendation P.66.

6 Out-of-band signals

In view of the following considerations:

– the compatibility with coder and decoder requirements according to Recommendation G.714;

– compatibility with existing practice in the mixed analogue – digital network in use today, the following limits are recommended:

Sending

With any sine wave signal above 4.6 kHz and up to 8 kHz applied at the MRP at a level of –4.7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (–4.7 dBPa at MRP) by at least the amount (in dB) specified in Table 3.

Compliance shall be checked by the sets described in Recommendation P.66.

TABLE 3/P.31

Discrimination levels – Sending

Applied sine wave frequency	Limit (minimum) [a)]
4.6 kHz	30 dB
8.0 kHz	40 dB

[a)] The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) – linear (dB) scale.

Receiving

With a digitally-simulated sine wave signal in the frequency range of 300 Hz to 3400 Hz and at a level of 0 dBm applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4.6 kHz to 8 kHz measured selectively in the artificial ear shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in Table 4.

Compliance shall be checked by the test described in Recommendation P.66.

TABLE 4/P.31

Discrimination levels – Receiving

Image signal frequency	Equivalent input signal level [a)]
4.6 kHz	–35 dBm0
8.0 kHz	–50 dBm0

[a)] The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) – linear (dB) scale.

7 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;
- what is practically obtainable in real use where the customer himself chooses the way to hold his handset,

the following limit is provisionally recommended:

The weighted terminal coupling loss (TCLw) should be greater than 40 dB when measured under free field conditions and with SLR + RLR normalized to OLR = +10 dB.

However, in order to meet the G.131 talker echo objective requirements a weighted terminal coupling loss greater than 45 dB is desirable and should be striven for.

Compliance shall be checked by the test described in Recommendation P.66.

8 Stability loss

In view of the following considerations:

- the aim to achieve a good stability;
- what is practically obtainable with normal type of handsets and transducers,

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 SLR + RLR normalized to OLR = +10 dB.

Compliance shall be checked by the test described in Recommendation P.66.

9 Delay

In view of the following considerations:

- the delay introduced by the coding and decoding according to Recommendation G.714;
- the delay introduced by the airpaths involved,

the following is recommended:

The sum of the delays from the mouth reference point to the digital interface and from the digital interface to the ear reference point shall not exceed 2.0 ms.

Compliance shall be checked by the test described in Recommendation P.66.

10 Input versus output (amplitude) characteristics

Switched gain amplifiers are commonly used in loudspeaking (hands-free) telephones and in handsets. This technique may also be advantageous for handset telephone applications.

Other non-linear techniques which could be used are automatic volume control or compressor/expander techniques. These devices may deliberately be non-linear over the input level range specified and may have dynamic characteristics (e.g. attack and hang over time).

At present there are no CCITT recommended characteristics or verification test methods for such devices in digital telephones. Unless a digital telephone has specifically designed non-linear characteristics, it is desirable to meet the variation of gain characteristics given in Annex A.

Annex A

Variation of gain with input level

(This annex forms an integral part of this Recommendation)

Sending direction

For digital telephones that are intended to have linear input versus output characteristics the gain variation relative to the gain for ARL should remain within the limits given in Table A.1. For intermediate levels, the same limits for gain variation apply.

Compliance shall be checked by the test described in Recommendation P.66.

TABLE A.1/P.31

Variation of gain with input level, sending

Sending dB relative to ARL	Upper limit (dB)	Lower limit (dB)
13	0.5	–0.5
0	0.5	–0.5
–30	0.5	–0.5
–30	1	
–40	1	
–40	2	
–45	2	

Receiving direction

For digital telephones that are intended to have linear input versus output characteristics the gain variation relative to the gain at an input level of –10 dBm0, should be within the limits given in Table A.2. For intermediate levels, the same limits for gain variation apply.

Compliance should be checked by the test described in Recommendation P.66.

TABLE A.2/P.31

Variation of gain with input level, receiving

Receiving level at the digital interface	Upper limit (dB)	Lower limit (dB)
+3 dBm0	0.5	–0.5
–10 dBm0	0.5	–0.5
–40 dBm0	0.5	–0.5
–40 dBm0	1	–1
–50 dBm0	1	–1
–50 dBm0	2	–2