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New presentation - see History box

Contents

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Foreword

Part three of this Interim European Telecommunication Standard (I-ETS) was produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI).

An ETSI standard may be given I-ETS status either because it is regarded as a provisional solution ahead of a more advanced standard, or because it is immature and requires a "trial period". The life of an I-ETS is limited to three years after which it can be converted into an ETS, have its life extended for a further two years, be replaced by a new version or, be withdrawn.

This is the third Part of an I-ETS which is currently intended to comprise eight Parts:

Part 1: "General (I-ETS 300 245-1 [1])".

Part 2: "PCM A-Law, handset telephony (I-ETS 300 245-2 [2])".

Part 3: "Pulse Code Modulation (PCM) A-Law, Loudspeaking and handsfree telephony".

Part 4: Interface for additional equipment.

Part 5: Wideband (7 kHz) telephony.

Part 6: Wideband (7 kHz) handsfree telephony.

Part 7: Locally generated information tones.

Part 8: Terminal application of 16 kbit/s speech coding algorithms (T/TE 10-07H).

Proposed announcement date	
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1 Scope

Part 3 of this I-ETS specifies technical characteristics for Pulse Code Modulation (PCM) A-law, 3,1 kHz loudspeaking and handsfree telephony terminals to be used at the basic access for the coincident S and T reference point of the Integrated Services Digital Network (ISDN).

Such terminals are intended to be used by a single person.

This Part of I-ETS 300 245 applies in conjunction with I-ETS 300 245-1 [1] and the characteristics specified in this Part are additional to those of I-ETS 300 245-1 [1].

2 Normative references

Part 3 of this I-ETS incorporates, by dated or undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to Part 3 of this I-ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to apply.

[1] I-ETS 300 245-1: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals, Part 1: General".

[2] I-ETS 300 245-2: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals, Part 2: PCM A-law handset telephony".

[3] ITU-T Recommendation P.10 (1993): "Vocabulary of terms on telephone transmission quality and telephone sets".

[4] CCITT Recommendation G.701 (1988): "Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms".

[5] ETS 300 111 (1992): "Integrated Services Digital Networks (ISDN); Telephony 3,1 kHz teleservice, Service description".

[6] CCITT Recommendation G.122 (1988): "Influence of national systems on stability, talker echo, and listener echo in international connections".

[7] ITU-T Recommendation P.51 (1993): "Artificial mouth".

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

[9] ITU-T Recommendation P.34 (1993): "Transmission characteristics of hands-free telephones".

[10] CCITT Recommendation O.131 (1988): "Quantizing distortion measuring equipment using a pseudo-random noise test signal".

[11] ISO 266 (1975): "Acoustics - Preferred frequencies for measurements".

[12] ITU-T Recommendation P.79 (1993): "Calculation of loudness ratings for telephone sets".

[13] CCITT Recommendation G.223 (1988): "Assumptions for the calculation of noise on hypothetical reference circuits for telephony".

[14] CCITT Recommendation O.41 (1988): "Psophometer for use on telephone-type circuits".

[15] IEC 651 (1979): "Sound level meters".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this Part of the I-ETS, the relevant definitions given in ITU-T Recommendations P.10 [3] and G.701 [4] apply along with the following:

Acoustic Reference Level (ARL): The acoustic level which gives - 10 dBm0 at the digital interface.

telephony 3,1 kHz teleservice: A description of telephony 3,1 kHz teleservice is given in ETS 300 111 [5], clause 5.

loudspeaking function: The handset is used in the normal position. The incoming signal is simultaneously presented to the user(s) from loudspeaker(s).

hands-free function: For free handling no handset or any other equipment with transducers is held to the ear of the user. If a handset is implemented then it is placed at a distance from the user. Normally, the handset is not active. The numbers, the implementation and the use of microphone(s) and loudspeaker(s) are not limited.

Call Progress Monitoring (CPM): The loudspeaker is used to monitor the received signals while the voice transmission in the sending direction is disconnected.

Terminal Coupling Loss (TCL): The frequency dependent coupling loss between the receiving port and sending port of a terminal due to:

- acoustical coupling at the user interface;
- electrical coupling due to crosstalk in the handset cord or within the electrical circuits;
- seismic coupling through the mechanical parts of the terminal.

NOTE 1: The receiving port and the sending port of a digital voice terminal is a 0 dBr point.

NOTE 2: The coupling at the user interface depends on the conditions of use.

weighted Terminal Coupling Loss (TCLw): The Terminal Coupling Loss (TCL) calculated using the weighting of CCITT Recommendation G.122 [6].

Hands-Free Reference Point (HFRP): A point located on the axis of the artificial mouth, at 50 cm from the lip ring, where the level calibration is made, in free field. It corresponds to the measurement point n° 11, as defined in ITU-T Recommendation P.51 [7].

idle mode: Idle mode is when the terminal is not activated by an input signal (e.g. input signal level below implemented threshold level).

active mode: Active mode is when the terminal is activated by an input signal (e.g. input signal level above implemented threshold level).

single talk: An operation mode where only one user is speaking.

double talk: An operation mode where two users are speaking simultaneously.

3.2 Abbreviations

For the purposes of this Part of the I-ETS, the following abbreviations apply:

AEC	Acoustic Echo Controller
AGC	Automatic Gain Control
Ardt	Received speech attenuation during double talk
ARL	Acoustic Reference Level
Asdt	Sent speech attenuation during double talk
CPM	Call Progress Monitoring
CSS	Composite Source Signal
ETS	European Telecommunication Standard
ETSI	European Telecommunications Standards Institute
I-ETS	Interim European Telecommunication Standard
ISDN	Integrated Services Digital Network
FFT	Fast Fourier Transformation
HATS	Head And Torso Simulator
HFRP	Hands-Free Reference Point
HFT	Hands-Free Telephony Terminal
LRGP	Loudness Rating Guard-ring Position
LST	Loudspeaking Telephony Terminal
MRP	Mouth Reference Point
PCM	Pulse Code Modulation
PN	Pseudo Noise
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
TCLwdt	Weighted Terminal Coupling Loss - double talk
TCLwst	Weighted Terminal Coupling Loss - single talk
Tondt	Break in time - double talk
Tonst	Break in time - single talk
TR	Built up time
TS	Switching time

4 Call control functions

The requirements of I-ETS 300 245-1 [1] shall be met.

5 Transmission aspects

5.1 General

Requirements for PCM A-law terminals are given in this Part of the I-ETS.

When using other coding algorithms other Parts of this I-ETS may apply.

5.1.1 Encoding

The default speech encoding algorithm for all speech terminals shall be the A-law encoding at 64 kbit/s, as defined in CCITT Recommendation G.711 [8].

Any other possible encoding algorithm are additional. For some encoding algorithms, requirements are given in other Parts of this I-ETS.

5.1.2 Relative level

The digital interface is a 0 dBr point according to CCITT Recommendation G.101.

5.1.3 Volume control

Unless stated otherwise, the requirements apply for all positions of the user-controlled receiving volume control, if the volume control is manually operated.

5.1.4 Requirements applicable to Hands-Free Telephone, Loudspeaking Telephony and Call Progress Monitoring

Table 1 presents the requirements that shall be applicable to Hands-Free Telephony, Loudspeaking Telephony and Call Progress Monitoring.

The applicability is indicated by an "X" in the corresponding column. The characteristics indicated by "(X)" are for further study.

For Loudspeaking Telephony Terminals (LSTs), the requirements for handset telephony can be found in I-ETS 300 245-2 [2].

Table 1: Applicability of requirements

	HFT	LST	CPM
Sensitivity frequency response (5.2)			
Sending (5.2.1)	X		
Receiving (5.2.2)	X	X	
Loudness ratings (5.3)			
Sending (5.3.1)	X		
Receiving (5.3.2.1) Maximum sensitivity	X	X	
Receiving (5.3.2.2) Volume control range	X	X	
Receiving (5.3.3) Adaptive gain control (optional)	X		
Terminal coupling loss (5.4)			
Handsfree Function (5.4.1)	X		
Loudspeaking Function (5.4.2)		X	
Call Progress Monitoring (5.4.3)			X
Stability loss(5.5.1) Handsfree function	X		
Stability loss(5.5.2) Loudspeaking function		X	
Distortion (5.6)			
Harmonic distortion (5.6.1.1) Sending	X		
Harmonic distortion (5.6.1.2) Receiving	X	X	
Total distortion (5.6.2.1) Sending	X		
Total distortion (5.6.2.2) Receiving	X	X	
Out-of-band signals (5.7)			
Sending (5.7.1)	X		
Receiving (5.7.2)	X	X	
Noise (5.8)			
Sending (5.8.1)	X		
Receiving (5.8.2)	X	X	
Delay (5.9)	X		
Switching characteristics (5.10)			
Build-up time	X		
Switching time	X	X	
Acoustic Echo Controllers characteristics (5.11)	(X)	(X)	
Articulation (5.12)	(X)	(X)	

5.2 Sensitivity frequency response

5.2.1 Sending

The sending sensitivity frequency response, from Mouth Reference Point (MRP) to digital interface, shall be within the mask drawn in figure 1.

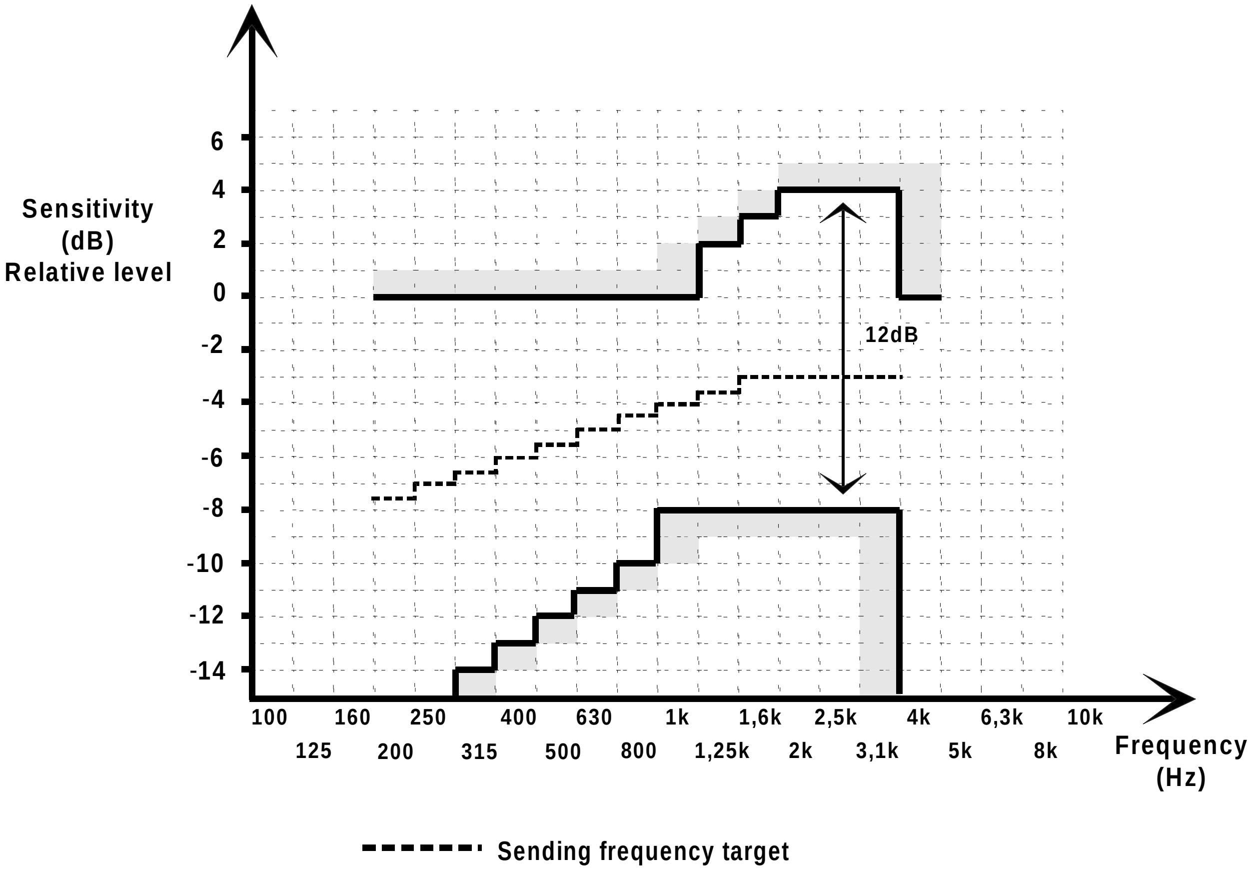


Figure 1: Sending sensitivity frequency mask for HFT

All sensitivity values are dB on an arbitrary scale.

Compliance shall be checked by the test described in annex A, subclause A.2.1.1.

NOTE: A sending sensitivity frequency response target is included in figure 1.

5.2.2 Receiving

The receiving sensitivity frequency responses, from the digital interface to the measuring point C, shall be within the masks drawn on figures 2 and 3 respectively for Hands-Free Telephony Terminals (HFTs) and Loudspeaking Telephony Terminals (LSTs).

All sensitivities are dB on an arbitrary scale.

Compliance shall be checked by the test described in annex A, subclause A.2.1.2.

NOTE: A receiving sensitivity frequency response target is included in figure 2.

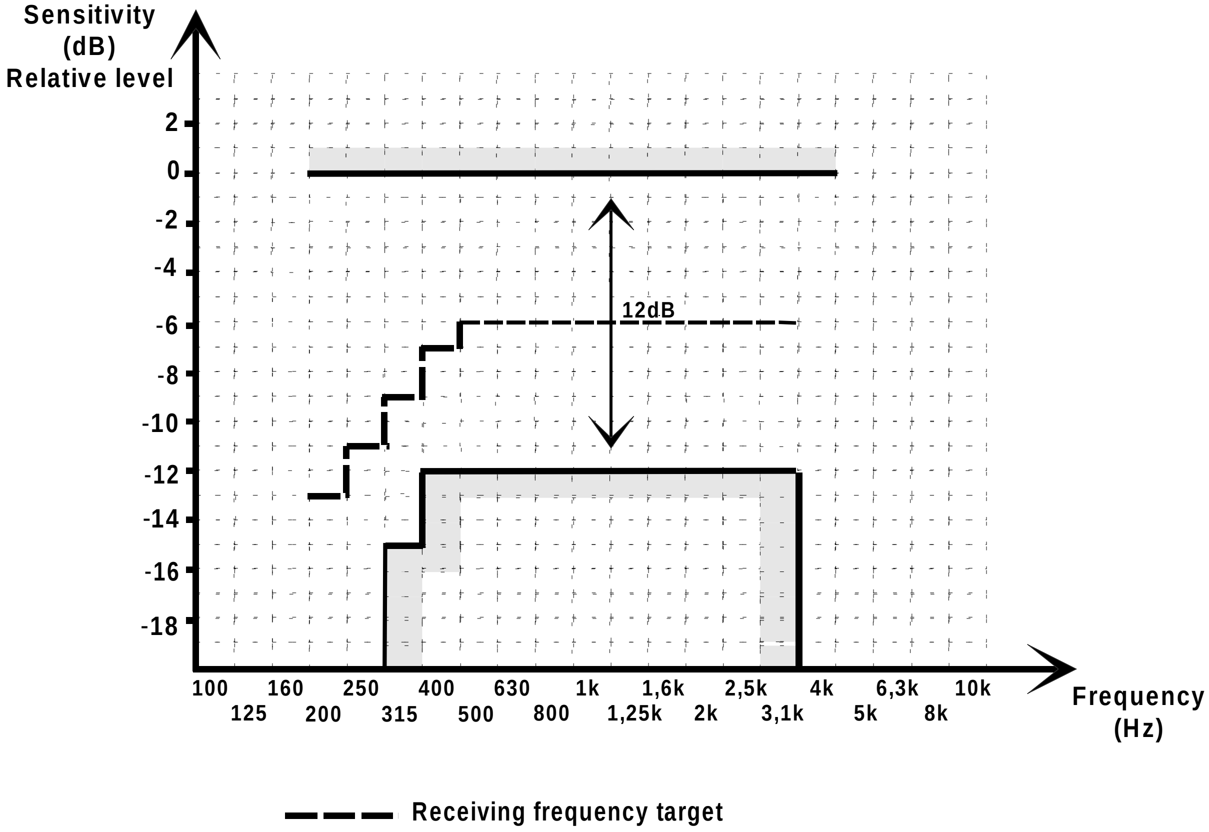


Figure 2: Receiving sensitivity frequency mask for HFT

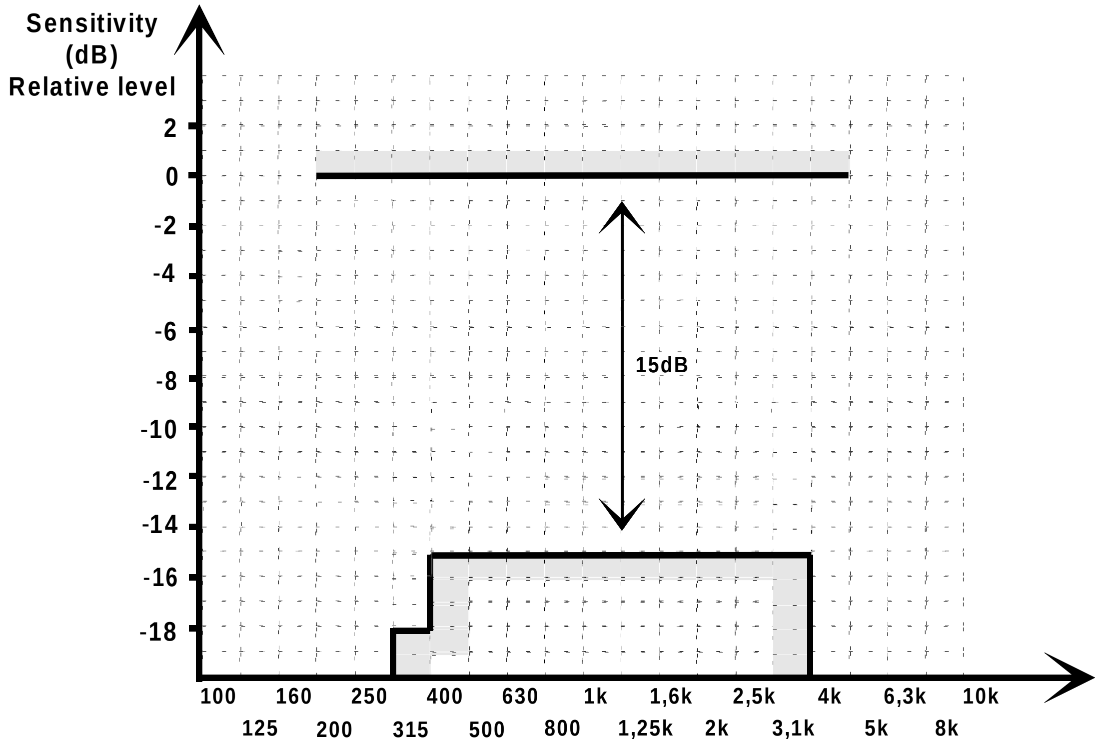


Figure 3: Receiving sensitivity frequency mask for LST

5.3 Loudness ratings

The values given in this subclause correspond to those for handset telephones as specified in I-ETS 300 245-2 [2] and will need to be adjusted if the referred values are adjusted.

5.3.1 Sending

Nominal value Sending Loudness Rating (SLR) = 12 dB.

There is a manufacturing tolerance of ± 4 dB.

Compliance shall be checked by the test described in annex A, subclause A.2.2.1.

5.3.2 Receiving

5.3.2.1 Maximum sensitivity

If a manually operated volume control is provided, the Receiving Loudness Rating (RLR) value, measured for the volume control set at its maximum, shall be:

RLR = - 6 dB ± 4 dB (including manufacturing tolerances).

Compliance shall be checked by the test described in annex A, subclause A.2.2.2.

5.3.2.2 Volume control range

With a line level of -15 dBm0 it shall be possible to obtain an RLR value which is at least 15 dB greater (quieter) than the RLR at - 30 dBm0 with manual and automatic gain control (if provided).

The acoustic output level shall be user controllable with a minimum range of 15 dB.

When a manual gain control is not used and if an automatic gain control is provided, the RLR value obtained with a line level of - 15 dBm0 shall not exceed that RLR value which is obtained with a line level of - 30 dBm0 by more than 15 dB. This avoids parts of negative law at the input/output characteristic.

The range of the acoustic output level shall be measured as a change in RLR.

NOTE: The volume control, if it operates manually, controls only the receiving path.

5.3.3 Adaptive gain control (optional)

An adaptive gain control, depending on the level of environmental noise, may be implemented into the set. The gain variation in the set corresponds to a gain in the receiving path and to a symmetrical attenuation in the sending path for increased ambient noise level.

Table 2 presents, for guidance and illustration only, three examples of gain variation characteristics.

Table 2: Gain variation characteristics

	Ambient noise level	Relative gain variation
Single threshold	between - 44 dBPa(A) and -39 dBPa(A)	8 dB
Double threshold	- 44 dBPa(A)	5 dB
	- 34 dBPa(A)	10 dB
Continuous variation	from - 64 dBPa(A) to - 44 dBPa(A)	3 dB
	from - 44 dBPa(A) to - 24 dBPa(A)	8 dB

5.4 Terminal Coupling Loss (TCL)

5.4.1 Hands-free function

The required values for TCL and TCLw correspond to single talk and one way transmission time, greater than 25 ms, as defined in table 3.

If information on one way transmission time is available in the terminal from the network, then all values in table 3 are applicable.

The TCL and TCLw shall be tested with the volume control in its maximum setting. The results shall be corrected to RLR + SLR = 15 dB.

NOTE: The correction is given by the nominal values of RLR = 3 dB and SLR = 12 dB. These values are derived from ITU-T Recommendation P.34 [9] for low room noise. Further study is needed for noisy situations.

Table 3: TCL values HFT

	One-way transmission time	**TCL (1/3 octave band)**	**TCLw**
Single talk	> 25 ms	> 30 dB	> 40 dB
	≤ 25 ms	> (18) dB	> (24) dB
Double talk	> 25 ms	> (25) dB	> (34) dB
	≤ 25 ms	> (12) dB	> (18) dB
NOTE: Values in brackets are under study.			

Compliance shall be checked by the test described in annex A, subclause A.2.3.

5.4.2 Loudspeaking function

The required values for TCL and TCLw corresponding to one way transmission time, greater than 25 ms, shall result in a TCL greater than 30 dB and a TCLw greater than 40 dB.

When one way transmission time is less than (25) ms, the TCL shall be greater than (18) dB and the TCLw shall be greater than (24) dB.

NOTE 1: The values in brackets are under study.

The TCL and TCLw shall be tested with the volume control at its maximum setting. The results shall be corrected to RLR_{LST} + $SLR_{Handset}$ = 10 dB, according to the handset mode (I-ETS 300 245-2 [2]).

NOTE 2: The correction is given by the nominal values of RLR = 3 dB for loudspeaking and SLR = 7 dB relevant to low ambient noise. Further study is needed for noisy situations.

If there is a voice switching device for the enhancement of the TCLw, it shall be assumed that, in double talk, the sending path of the associated handset takes priority over the loudspeaking path.

Compliance shall be checked by the test described in annex A, subclause A.2.3

5.4.3 Call Progress Monitoring (CPM)

The voice transmission in the sending direction shall be disconnected from the B-channel during the use of the CPM function. TCLw shall be greater than 40 dB.

Compliance shall be checked by test described in annex A, subclause A.2.3.

5.5 Stability loss

The attenuation from the digital input to the digital output shall be, at any time, at least 6 dB for all frequencies in the range of 200 Hz to 4 kHz.

The test arrangements are defined in annex A, subclause A.1.4.2.

5.5.1 Hands free function

Whatever the signal processing implemented in the HFT, the instantaneous TCL, during any period of the process, shall be at least 6 dB.

For HFT designed in two or more pieces, the stability shall be evaluated in all the possible arrangements defined in annex A, subclause A.1.4.1.

Compliance shall be checked by the test described in annex A, subclause A.2.4.1.

5.5.2 Loudspeaking function

The stability of the loudspeaking telephony function shall be tested for an RLR value (measured in loudspeaking telephony mode) which is 10 dB higher (quieter) than at the maximum of the volume control.

Compliance shall be checked by the test described in annex A, subclause A.2.4.2.

5.6 Distortion

5.6.1 Harmonic distortion (sinusoidal signal)

5.6.1.1 Sending

The ratio of signal to harmonic distortion shall be above the following mask:

Table 4: Signal to harmonic distortion ratio, sending

Frequency	Ratio
315 Hz	26 dB
400 Hz	30,5 dB
1 kHz	30,5 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Compliance shall be checked by the test described in annex A, subclause A.2.5.1.1.

5.6.1.2 Receiving

The ratio of signal to harmonic distortion shall be above the following mask:

Table 5: Signal to harmonic distortion ratio, receiving

Frequency	Hands-Free Function	Loudspeaking function
315 Hz	26 dB	20 dB
400 Hz	26 dB	26 dB
500 Hz	30,5 dB	30,5 dB
1 kHz	30,5 dB	30,5 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Limits above 1 kHz are for further study.

Compliance shall be checked by the test described in annex A, subclause A.2.5.1.2.

5.6.2 Total distortion (pseudo-random noise signal)

The values are for further study.

5.6.2.1 Sending

Compliance shall be checked by the test described in annex A, subclause A.2.5.2.1.

5.6.2.2 Receiving

Compliance shall be checked by the test described in annex A, subclause A.2.5.2.2.

5.7 Out-of-band signals

5.7.1 Sending

With any signal defined in annex A, subclause A.2.6, above 4,6 kHz and up to 8 kHz, applied at the Hands-Free Reference Point (HFRP) at a level of - 28,7 dBPa, the level of any image frequency produced at the digital interface shall be below a level obtained for the broadband signal (annex A, subclause A.1.1.1), defined as the reference signal, by at least the amount (in dB) specified in table 6.

Table 6: Out-of-band requirement, sending

Frequency	Limit (minimum)
4,6 kHz	30 dB
8 kHz	40 dB

The limit at intermediate frequencies lies on a straight line drawn between the given values on a linear (dB)-logarithmic (frequency) scale.

Compliance shall be checked by the test described in annex A, subclause A.2.6.1.

5.7.2 Receiving

Any spurious out-of-band image signals in the frequency range 4,6 kHz to 8 kHz measured selectively at the point C shall be lower than the in-band level measured with a broadband reference signal, applied at a level of - 30 dBm0 . The minimum level of difference between the reference signal level and the out-of-band image signal level shall be as given in table 7.

Table 7: Out-of-band signals, receiving

Frequency	Limit (minimum)
4,6 kHz	35 dB
8 kHz	45 dB

The limit at intermediate frequencies lies on a straight line drawn between the given values on a linear (dB) - logarithmic (frequency) scale.

Compliance shall be checked by the test described in annex A, subclause A.2.6.2.

5.8 Noise

Three types of noise generated by the HFT or the LST have been identified:

- noise generated by the set in the "idle mode" (see subclauses 5.8.1.1 and 5.8.2.1);
- noise generated by the set when an activation signal is applied. As the noise is masked by the quantizing and harmonic distortion, the limits are described in subclause 5.6;
- noise generated by the set as soon as the activation signal is interrupted (see subclauses 5.8.1.2 and 5.8.2.2).

5.8.1 Sending

5.8.1.1 Idle mode

The noise produced by the set in the sending path shall not exceed - 64 dBm0p.

Compliance shall be checked by the test described in annex A, subclause A.2.7.1.1.

5.8.1.2 Noise during transition from active mode to idle mode

The mask for this noise requirement versus time is for further study.

Compliance shall be checked by the test described in annex A, subclause A.2.7.1.2.

5.8.2 Receiving

5.8.2.1 Idle mode

5.8.2.1.1 A-weighted

With the volume control set to the maximum, the noise level shall not exceed - 49 dBPa (A).

Compliance shall be checked by the test described in annex A, subclause A.2.7.2.1.

5.8.2.1.2 Third-octave band spectrum

With the volume control set to the maximum, the level in any 1/3 octave band, between 100 Hz and 10 kHz, shall not exceed a value of - 59 dBPa.

Compliance shall be checked by the test described in annex A, subclause A.2.7.2.1.

5.8.2.2 Noise during transition from active mode to idle mode

The mask for these noise requirements versus time is for further study.

Compliance shall be checked by the test described in annex A, subclause A.2.7.2.2.

5.9 Delay

The delay shall be less than 8 ms (5 ms for the telephone set to allow digital signal processing and 3 ms for the air path).

Measurements shall be performed on the two paths separately. The delay is the summation of these two values.

Compliance shall be checked by the test described in annex A, subclause A.2.8.

NOTE: When an Acoustic Echo Controller (AEC) is included in the set, the limit for delay is the sum of the 8 ms and of the AEC delay defined in subclause 5.11.

5.10 Switching characteristics

The definitions and figures can be found in ITU-T Recommendation P.34 [9], clause 5.

For pure voice switching device, the build-up time (T_R) shall be less than 15 ms. For other voice processing technologies implemented in the sets, T_R is for further study.

The switching time (T_S) shall be less than 150 ms.

Compliance shall be checked by the test described in annex A, subclause A.2.9.

NOTE: Switching characteristics on ambient noise are for further study.

5.11 Acoustic Echo Controllers characteristics

For further study.

NOTE: For the purposes of the HFT specifications, it seems necessary to take into account some parameters as defined in ITU-T Recommendation G.167. They are listed below:

- TCLwst - single talk. The weighted loss between the Rin and Sout network interfaces when the AEC is in normal operation, and when there is no signal coming from the local user;

- TCLwdt - double talk. The weighted loss between the Rin and Sout network interfaces when the AEC is in normal operation, and where the local user and the far-end user are active simultaneously;

- Received speech attenuation during double talk (Ardt). The received signal attenuation (at the Rout point) which is inserted by the AEC during double-talk events. The frequency response on the receive side during double talk is left for further study;

- Sent speech attenuation during double talk (Asdt). The sent signal attenuation (at the Sout point) which is inserted by the AEC during double-talk events. The frequency response on the sent side during double talk is for further study;

- Break-in time - single talk (Tonst). The time interval between the onset of the received signal (similarly the transmitted signal) and the instant when the attenuation on the receive path (similarly on the send path) reaches (3) dB. For this purpose, the other side is quiet;

- Break-in time - double talk (Tondt). The time interval between the onset of the received signal (similarly sent signal) and the instant when the attenuation on the receive path (similarly on the send path) reaches the value Ardt (similarly Asdt). For this purpose, the signal in the opposite direction of transmission is held at a specified level;

- Delay. The values correspond to the extra delay which can result from the AEC processing in the processing unit. The maximum delay permitted depends on the application (see subclause 4.6 of ITU-T Recommendation G.167).

5.12 Articulation

For further study.

Annex A (normative): Test specification

A.1 Test considerations

A.1.1 Test signals

The test signal levels specified in this annex refer to the active part of the signal.

In order to ensure that the test is representative of the normal operation, the test signal has two functions:

- terminal activation;
- providing the measurement stimulus without adversely affecting the activation.

It shall be checked that both functions are correctly achieved.

Appropriate types of test signal are:

- switched ON/OFF signals, as defined in annex A, subclauses A.1.1.1 and A.1.1.2, at a rate of 250 ms (± 5 ms) ON and 150 ms (± 5 ms) OFF;
- Composite Source Signal (CSS), described in annex B.

In the case of terminal equipment using technologies for which the test specifications in this annex are not suitable to prove conformance to this I-ETS, equivalent evaluation methods can be used, cfr. subclause A.1.5. The methods shall be documented by the supplier and evaluated by the test house.

All tests are to be carried out using only one type of test signal. The test signal shall be indicated in the test report.

The results on linear HFT are, in principle, equivalent for the two test signals.

For HFT incorporating adaptive Automatic Gain Control (AGC), AEC or other non-linear functions, the results may differ with the two signals.

CSS shall be used for equipment incorporating Adaptive Acoustic Echo Control functions and may be used when the switched signals do not activate properly the terminal for all tests described in this annex.

A.1.1.1 Broadband signal

The broadband signal shall be a gaussian pink noise, with a crest factor of 11 dB ± 1 dB.

The bandwidth of the broadband signal shall correspond to the 14 third octave bands from 200 Hz to 4 kHz.

The third-octave spectrum of electrically generated pink noise shall be equalized within ± 1 dB, while the acoustically generated shall be equalized at the Mouth Reference Point (MRP) within ± 3 dB.

The slope outside the bandwidth shall be at least 8 dB/third octave.

Broadband signals are used for testing sensitivity/frequency response, Loudness Ratings, TCL, TCLw, stability and switching characteristics.

A.1.1.2 Sinusoidal and narrow-band signals

a) Sinusoidal signals are used for testing Harmonic distortion and Delay;
b) Narrow-band signals are used for testing:

- total distortion. The test signal conforms with CCITT Recommendation O.131 [10];
- out-of-band signals.

NOTE: For the sending test, a special time multiplexing is implemented, as described in subclause A.2.6.1.

A.1.2 Test signal levels

A.1.2.1 Sending

Unless specified otherwise, the test signal level shall be -28,7 dBPa at the HFRP. The characteristics of the artificial mouth shall be according to ITU-T Recommendation P.51 [7]. There are two slightly different methods of calibrating the artificial mouth:

a) Method 1: using HFRP

The input signal from the artificial mouth is calibrated under free field conditions at the HFRP, such that the spectrum corresponds to annex A, subclause A.1.1 and the total level in the frequency range corresponding to the third octave bands from 200 Hz to 4 000 Hz is - 28,7 dBPa.

For calculating SLR the reference point is MRP. The sound pressure level at the MRP can then be calculated by adding 24 dB to the HFRP sound pressure [dBPa].

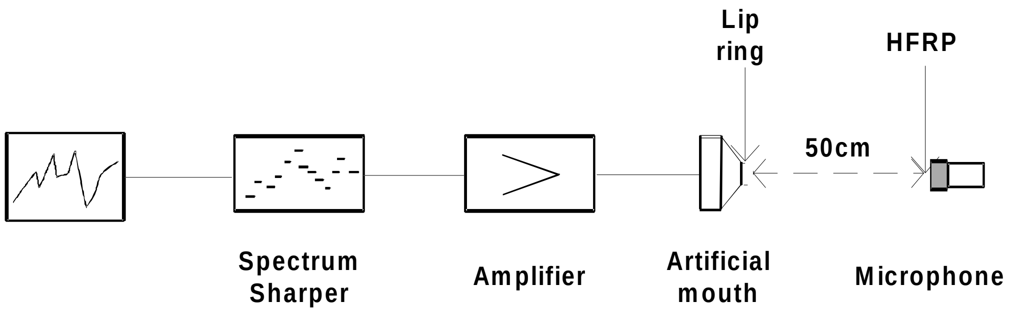


Figure A.1

b) Method 2: using MRP

The input signal from the artificial mouth is calibrated under free field conditions at the MRP, such that the spectrum corresponds to annex A, subclause A.1.1 and the total level in the frequency range corresponding to the third octave bands from 200 Hz to 4 000 Hz is - 4,7 dBPa.

Then the level is adjusted to - 28,7 dBPa at the HFRP. The spectrum is not altered. The actual level at the MRP (measured in third octaves) is used as reference for calculating SLR.

NOTE: The HFRP corresponds to the measurement point 11 of ITU-T Recommendation P.51 [7]. The transfer function between HFRP and MRP amounts to 24,0 dB + 3/- 4 dB.

A.1.2.2 Receiving

Unless specified otherwise, the applied test signal level at the digital input shall be - 30 dBm0, as far as the user-controlled receiving volume control is set at its maximum.

A.1.3 Test rooms

A.1.3.1 Anechoic room

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

Satisfactory free field conditions exist where errors, due to the departure from ideal conditions, do not exceed the values defined in table A.1, inside a sphere centred on point B (see figure 3 of ITU-T Recommendation P.34 [9]), with one metre radius, in absence of the table (see note 3 as a provisional method for controlling these conditions).

Table A.1

1/3 octave band center frequency (Hz)	Allowable departure (dB)
< 630	± 1,5 dB
800 to 5 000	± 1 dB
> 6 300	± 1,5 dB

The test for verification of the free field shall be performed with a wide band noise excitation and 1/3 octave analysis. The test signal level shall be - 20 dBPa.

Measurements are made along the seven axes which are numbered (1) to (7) in figure A.2, with the sound source placed at positions equivalent to B or C, as appropriate. Measurement points along each axis, taken from the front plan of the artificial mouth lip-ring are at the distances of 315 mm, 400 mm, 500 mm, 630 mm, 800 mm and 1 000 mm.

The broadband noise level shall not exceed - 70 dBPa(A). The octave band noise level shall not exceed the values specified in table A.2 below:

Table A.2: Noise level

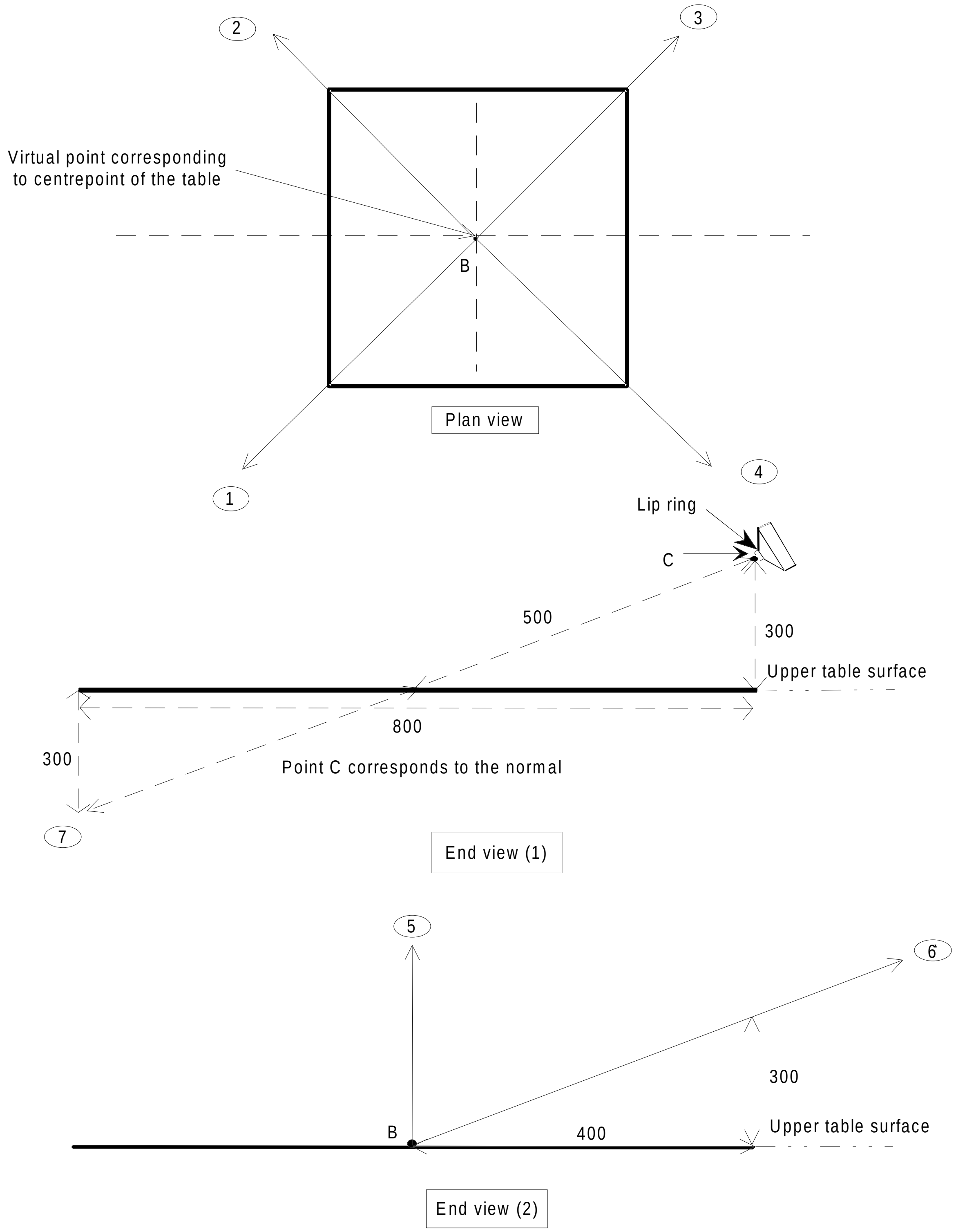
Centre frequency (Hz)	Octave band pressure level (dBPa)
63	- 45
125	- 60
250	- 65
500	- 65
1 k	- 65
2 k	- 65
4 k	- 65
8 k	- 65

NOTE 1: For TCL tests and echo cancelling characteristics, it could be useful to define a standardized test room. For further study.

NOTE 2: The ambient noise in test room for the adaptive gain control testing is under study.

NOTE 3: A room including the test arrangement fulfilling the following requirements probably meets the satisfactory conditions. Dimensions of the room: height ≥2,2 m; volume V ≥ 30 m^3. The table should be placed horizontally in the centre of the test room and there should be an inclination of ~ 30° between the table and the ceiling. The reverberation time T, measured at points B and C, should satisfy the following inequality:

$T(s) \leq 0{,}0033\ V\ (m^3)$; which is based on a calculation with the radius of 50 centimetres.



Dimensions in millimetres.

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

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

Figure A.2: Axes used in the determination of free field conditions for 1 m radius sphere

A.1.3.2 "Anechoic" tests in a reverberant field

Test methods permit the provision of "anechoic" tests in a reverberant field separating the direct field from the reflected one. The use of such methods is for further study.

A.1.4 Test arrangement

The test environment conditions (presence of a Head And Torso Simulator (HATS) (see ITU-T Recommendation P.58), variable acoustic conditions, ambient noise,...) are for further study.

A.1.4.1 Hands-free function

The HFT is placed on a table according to ITU-T Recommendation P.34 [9] (section 6.1: test table; section 6.2: test arrangements and figure 3 of ITU-T Recommendation P.34 [9]).

The artificial mouth axis and the microphone axis are coincident with the straight line drawn between point C and point B (see figure A.3).

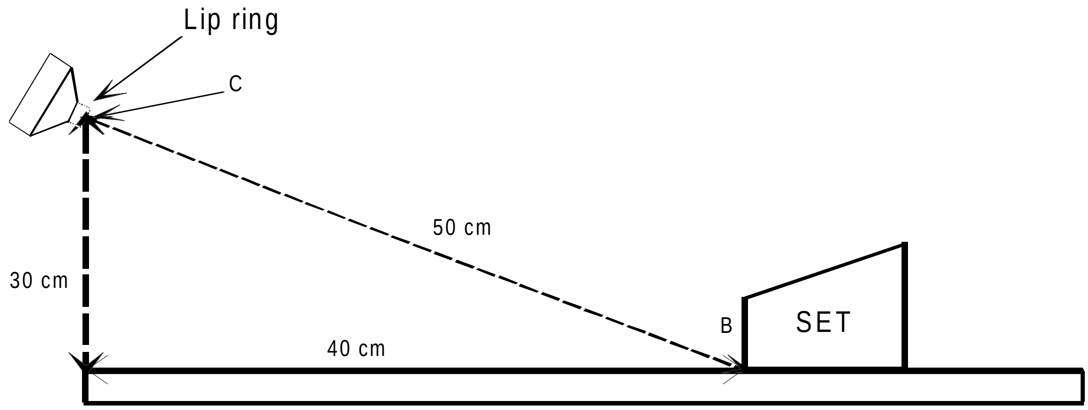


Figure A.3: Measurement configuration

For the test of switching characteristics, the artificial mouth shall be placed as described in figure A.3 and the microphone shall be placed at point D (as defined in figure A.4).

For stability control, the different pieces of the HFT (if the HFT is built in two or more pieces) shall be placed as close as possible to each other, but without modifying the normal usage of the HFT.

NOTE: If HFT is implemented in two or more pieces, care should be taken to ensure that the test arrangement does not modify the normal usage of the HFT. The case of special terminals (multifunctions) including handsfree function is for further study.

A.1.4.2 Loudspeaking function

The set is placed on the table according to CCITT Recommendation P.34 [9] (section 6.1: test table; section 6.2: test arrangement).

For TCL measurements the handset earphone "centre" shall be placed at point C with the microphone vertically below the earphone. The meaning of "centre" is the centre of the surface of the handset earphone which is placed normally against the ear. This surface is set at 90 degrees relative to the loudspeaker.

For stability measurements, the handset shall be placed as defined in I-ETS 300 245-2 [2] ("stability loss", annex A, subclause A.2.4.2). The set shall be placed symmetrically to the axis of the handset. The front side of the terminal is directed towards the corner formed by the three surfaces with its front edge at a distance of 1 m from this corner.

For the test of all characteristics, except TCL and stability loss, the handset shall be placed in the Loudness Rating Guard-ring Position (LRGP) on the test head. The centre of the artificial mouth lip-ring shall be placed at point D, as shown in figure A.4.

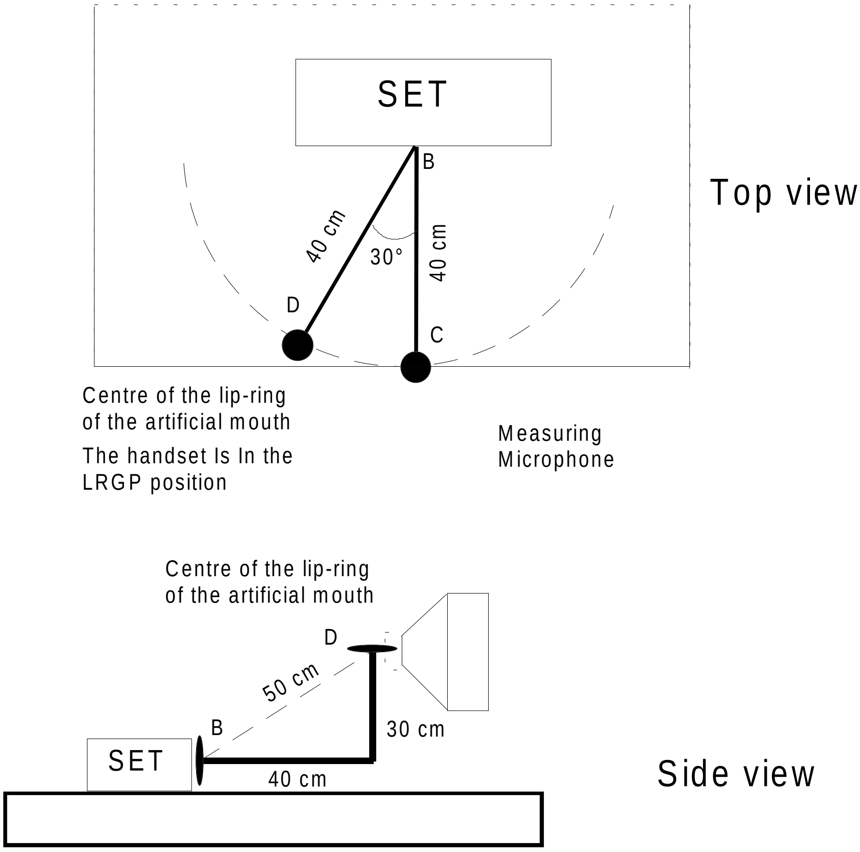


Figure A.4: Measurement position, LST (top and side views)

A.1.5 Alternative test methods

The requirements of this Part of the I-ETS are written on the basis of the test methods described in this annex. For some parameters, it is recognised that alternative test methods may exist. It shall be the responsibility of the test house to ensure that any alternative method is equivalent to that described in this annex.

A.2 Transmission requirements testing

A.2.1 Sensitivity/frequency response

The test signal is specified in subclause A.1.1.

Measurements shall be made of the 1/3 octave band levels as defined by the R10 series of preferred numbers in ISO 266 [11] for frequency bands from 200 Hz to 4 kHz inclusive.

NOTE: The frequency response masks in figures 1, 2 and 3 are defined on the basis of a pink noise signal, analysed in third octave bands.

A.2.1.1 Sending

The sensitivity for each 1/3 octave band is expressed as dB relative to 1 V (digital interface)/Pa (MRP). The sensitivity, as a function of the 1/3 octave band, is then plotted.

NOTE: For the calibration method 1 (using HFRP) the sending sensitivity for each third octave band is defined as:

$S_{mJ} = 20 \log V_s - 20 \log P_m$

where
V_s is the measured voltage across the line termination and P_m is the nominal sound pressure at the MRP (P_m = 0,58 Pa).

For the calibration method 2 (using MRP) the sending sensitivity for each third octave band is defined as:

$S_{mJ} = 20 \log V_s - 20 \log P_m + \text{Corr} - 24 \text{ dB}$

where
V_s is the measured voltage across the line termination; P_m is the applied sound pressure at the MRP and Corr is 20 log (P_{MRP}/P_{HFRP}) of the used artificial mouth (this value is given in the calibration chart of the artificial mouth. 24,0 dB is the ideal value).

A.2.1.2 Receiving

The sensitivity for each 1/3 octave band is expressed as dB relative to 1 Pa (measuring microphone position)/V (digital interface). The sensitivity as a function of the 1/3 octave band is then plotted.

A.2.2 Loudness ratings

A.2.2.1 Sending Loudness Rating (SLR)

a) The sending sensitivity shall be measured at each of the 14 frequency bands given in table 1 of ITU-T Recommendation P.79 [12], bands 4-17 (200 Hz - 4 000 Hz).

b) The sensitivity is expressed in terms of dB V/Pa and the SLR shall be calculated according to the formula 2-1 of ITU-T Recommendation P.79 [12], using the sending weighting factors from table 1 of ITU-T Recommendation P.79 [12] and according to section 3 of ITU-T Recommendation P.79 [12].

NOTE: For the calibration method 1 (using HFRP) the sending sensitivity for each third octave band is defined as:

$S_{mJ} = 20 \log V_s - 20 \log P_m$

where V_s is the measured voltage across the line termination and P_m is the nominal sound pressure at the MRP (P_m = 0,58 Pa).

For the calibration <u>method 2</u> (using MRP) the sending sensitivity for each third octave band is defined as:

$S_{mJ} = 20 \log V_s - 20 \log P_m + \text{Corr} - 24 \text{ dB}$

where V_s is the measured voltage across the line termination; P_m is the applied sound pressure at the MRP and Corr is 20 log (P_{MRP}/P_{HFRP}) of the used artificial mouth (this value is given in the calibration chart of the artificial mouth, 24,0 dB is the ideal value).

A.2.2.2 Receiving Loudness Rating (RLR)

a) The receiving sensitivity shall be measured at each of the 14 frequency bands given in table 1 of ITU-T Recommendation P.79 [12], bands 4 to 17 (200 Hz - 4 000 Hz).

b) The sensitivity is expressed in terms of dB Pa/V and the RLR(cal) shall be calculated according to the formula 2-1 of ITU-T Recommendation P.79 [12], using the receiving weighting factors from table 1 and according to section 3, of ITU-T Recommendation P.79 [12].
The RLR shall then be computed as RLR(cal) minus 14 dB according to CCITT Recommendation P.34 [9]), and without the L_E factor.

c) For the volume control range, an additional test level of -15 dBm0 shall be used.

A.2.3 Terminal Coupling Loss

The test signal is defined in subclause A.1.1. The provisional test signal level shall be - 15 dBm0. TCL shall be measured as attenuation from the digital input to the digital output, by the 14 third octave bands between 200 Hz and 4 kHz.

The TCLw (before correction) shall be calculated from CCITT Recommendation G.122 [6], with the following formula:

$$TCLw = -10 \ \log_{10} \ (\frac{1}{14}\sum_{i=1}^{14} Ai)$$

where Ai is the output/input power ratio at the i-th third octave band.

A.2.4 Stability loss

The stability loss shall be measured as an attenuation from the digital input to the digital output.

A.2.4.1 Hands-Free function

The test signal type and level and the test conditions are for further study.

A.2.4.2 Loudspeaking function

The test signal type and level and the test conditions are for further study.

A.2.5 Distortion

A.2.5.1 Harmonic distortion (sinusoidal signal)

The signal to harmonic distortion ratio shall be measured on each 1/3 octave centre frequency from 315 Hz to 1 kHz.

The test signal is a sinusoidal frequency (corresponding to the 1/3 octave frequencies between 315 Hz and 1 kHz) modulated ON/OFF at a rate defined in subclause A.1.1.

The sinusoidal signal shall be switched ON/OFF at the zero crossing.

The harmonics shall be measured selectively up to 3,15 kHz.

A.2.5.1.1 Sending

The sinusoidal signal level, calibrated at Hands-Free Reference Point (HFRP), shall be - 20 dBPa.

A.2.5.1.2 Receiving

The sinusoidal signal level shall be calibrated to - 20 dBm0.

The signal frequencies are limited at 1 kHz. Limits above 1 kHz are for further study.

A.2.5.2 Total distortion (pseudo-random noise signal)

The test signal shall be a band limited noise signal corresponding to CCITT Recommendation O.131 [10].

A.2.5.2.1 Sending

For further study.

A.2.5.2.2 Receiving

For further study.

A.2.6 Out-of-band signals

The activation signal level shall be the same as the reference signal level. The bandwidth of the test signal shall be 100 Hz.

A.2.6.1 Sending

For a correct activation of the HFT, the test signal according to annex A, subclause A.1.1, shall be used as the reference signal with a level according to annex A, subclause A.1.2. For the test, an out-of-band signal shall be provided as a frequency band signal centred on 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz respectively. The level of any image frequencies at the digital interface shall be measured.

The levels of these signals shall be as specified in subclause 5.7.1.

The complete test signal is constituted by t_1 ms of in-band signal (reference signal), t_2 ms of out-of-band signal and another time t_1 ms of in-band signal (reference signal).

The observation of the output signal on the first and second in-band signals permits control if the set is correctly activated during the out-of-band measurement. This measurement shall be performed during t_2 period:

- t_1 may be 250 ms;

- t_2 depends on the integration time of the analyser, less than 150 ms.

A.2.6.2 Receiving

For input narrow band signals centred on 500 Hz, 1 000 Hz, 2 000 Hz and 3 150 Hz, applied at the level of - 30 dBm0, the level of any out-of-band signals at frequencies up to 8 kHz shall be measured selectively at point C.

A.2.7 Noise

To ensure that the set is correctly stated for the sending direction and the receiving direction respectively, the test signal specified in annex A, subclause A.1.1 shall be applied with a level as specified in annex A, subclause A.1.2 for activation.

A.2.7.1 Sending

The noise level shall be measured in a quiet environment (ambient noise less than - 64 dBPaA) at the digital output with a measurement equipment including psophometric weighting according to CCITT Recommendation G.223 [13], table 4 and according to CCITT Recommendation O.41 [14] regarding dynamic requirements.

A.2.7.1.1 Idle mode

The idle mode noise shall be measured 500 ms after interrupting the activation signal.

A.2.7.1.2 Noise during the transition active mode to idle mode

The measurement consists in the evaluation of the noise generated by the set, from the instant (t_o) when the activation signal is interrupted, to 1 second after this instant t_o.

A.2.7.2 Receiving

A.2.7.2.1 Idle mode

The idle mode noise shall be measured 500 ms after interrupting the activation signal.

IEC 651 [15] shall be used to obtain the specification of a weighting.

A.2.7.2.2 Noise during transition from active mode to idle mode

The measurement consists of the evaluation of the noise generated by the set, from the time (t_o) when the activation signal is interrupted to 1 second after t_o.

A.2.8 Delay

The delay (D) in the send and receive direction shall be measured separately from MRP to digital interface (D_s) and from digital interface to measurement microphone (D_r).

Measurements shall be made with pairs of sine wave signals.

The nominal frequencies are 500 Hz, 630 Hz, 800 Hz, 1 kHz, 1,25 kHz, 1,6 kHz, 2 kHz and 2,5 kHz.

The delay is derived from the measurement of the phase shift between the sending signal on channel 1 (CH1) of the measurement equipment and the receiving signal on channel 2 (CH2) of this equipment. For each of the frequencies f_0 the phase shift shall be measured at the frequencies f_1 and f_2. "f_1" and "f_2" yield as follows: $f_1 = f_0 - 50$ Hz and $f_2 = f_0 + 50$ Hz.

NOTE: If the phase shift of f_2 and f_1 is greater than 180 degrees, then the frequency step should be reduced (e.g. 10 Hz).

The measurements shall be executed using the following steps:

1) output the sine wave test signal with the frequency f_1 on CH1;

2) measure the phase shift in degrees between CH1 and CH2 (p_1);

3) output the sine wave test signal with the frequency f_2 on CH1;

4) measure the phase shift in degrees between CH1 and CH2 (p_2);

5) compute the delay in milliseconds using the following formula:

$$D(f_0) = \frac{-1\,000 \quad x \quad (p_2 - p_1)}{360 \quad x \quad (f_2 - f_1)}$$

All the negative values of p_1 and p_2 for steps 2 and 4 correspond to a lagging of CH2 relative to CH1. Care shall be taken that no errors occur when the phase shift p passes 360° when switching from f_1 to f_2.

Finally, the average D of all values $D(f_0)$ for the different frequencies f_0 is calculated.

The delay introduced by the artificial mouth shall be measured by mounting a microphone at the MRP. The delay of all additional test equipment between the S/T interface and the digital input (CH2), respectively output (CH1), of the test equipment shall be determined. The values of these delays are needed for the correction of the measurement results.

The delay of the item under test is deducted from the formula:

$$D = D_s + D_r = D_{sm} + D_{rm} - D_e$$

where D_e is the delay of the test equipment;

D_{sm} is the measurement delay in the sending direction;

D_{rm} is the measurement delay in the receiving direction.

A.2.9 Switching characteristics

The user-controlled receiving volume control shall be set at its maximum.

The test signal shall be a pink noise switched ON/OFF, as defined in annex A, subclause A.1.1 (250 ms ON/150 ms OFF).

The measurements shall be made both in the sending and the receiving paths.

A.2.9.1 Test signal levels

For HFT testing, the sending test signal level shall be - 20 dBPa at the HFRP.

For LST testing, the sending test signal level shall be - 4,7 dBPa at the MRP.

The receiving test signal level shall be - 20 dBm0 at the digital interface.

A.2.9.2 Build-up time

a) Preliminary build-up time

The set shall be switched on, and after 15 seconds, T_R shall be measured on the first ON sequence signal.

Care shall be taken that, between the switching on of the set and the measurements, no noise appears in the room or on the line.

b) Build-up time, after an activation in the path under measurement

A continuous ON/OFF sequence shall be produced in a period of about 10 seconds in the path under measurements. After a 1 second pause, T_R shall be measured on the first ON sequence signal.

c) Build-up time, after activation in the opposite path

A continuous ON/OFF sequence shall be produced during about 10 seconds in the opposite path. After a 1 second pause, T_R shall be measured on the first ON sequence signal produced in the path under measurement.

A.2.9.3 Switching time

A continuous ON/OFF sequence shall be produced in the opposite path. At the end of the last ON sequence an ON sequence shall be started in the path under measurement, on which T_S shall be measured.

A.2.10 Echo cancelling characteristics

For further study.

A.2.11 Articulation

For further study.

Annex B (informative): Description of the Composite Source Signal (CSS)

B.1 General presentation of the CSS

When composing the CSS, the following three components were judged as being especially important:

- voiced signal to simulate voice properties;
- deterministic signal for measuring the transfer functions without statistical errors with constant power density spectrum of the excitation signal in the frequency domain to be measured;
- pause "signal" providing amplitude modulation.

The following features result:

1) short period of measurement;

2) feeding-in possibility of the test signal for the talking and listening direction at the same time (duplex operation).

The basic idea for using such a signal is to place the device under test in a well defined, reproducible state for the period of measurement and to secure that the transfer functions of the device do not change appreciably during the actual measurement (quasi-stationary). More precisely, the composite source signal (see figures B.1 to B.4) consists of the following components:

a) Voiced sound produced from the "artificial voice" signal according to ITU-T Recommendation P.50.

The voiced sound part of the CSS is the conditioning signal intended to activate possible speech detectors in voice-controlled systems. The reason the voiced sound has been chosen is that, presumably, all future hands-free telephones will quickly respond to a voiced sound. This signal is to activate a hands-free telephone for the direction of transmission to be measured. As the duration, beginning and end of the voiced sound are known exactly, this signal can also be used to measure the switching time for the direction of transmission under test. By means of the signal shape in the time domain the switching time and delay time of the entire system can be determined according to ITU-T Recommendation P.34 [9]. The duration of the signal amounts to 50 ms. Within this period any speech detector has recognised voice and has activated the system.

One practical example for voiced sound can be found in table B.1.

b) Pseudo-Noise (PN) signal.

The measurement signal is the Pseudo-Noise (PN) signal, with a crest factor of 11 dB ± 1 dB, presented after the voiced artificial speech sound. This signal has certain noise-like features. The magnitude of its Fourier transform is constant with frequency while the phase is changing. For measuring hands-free telephones, usually only the magnitude of the transfer function is of interest, the phase is not that important but can be determined as well.

The signal may be produced as follows:

- first, a complex spectrum is produced in the frequency domain according to the following equation:

$$H(k) = W(k) * e^{j * i_k * \pi} ; \quad k = -M/2, \ldots., M/2, \ without\ 0 ; i_k \ \{+1, 0\} random, i_k = -i_{-k} \qquad (1)$$

Index M is adjusted to the chosen FFT size (e.g. 2 048 points). The equation shows that the amount of the produced complex spectrum is constant for all frequencies if W(k) is chosen equal to a given constant value, e.g. 1 for all frequencies, whereas the phase may be + π or 0 for each frequency, corresponding to a random sequence. However, to produce a different weighting in the frequency domain, W(k) can easily be adjusted to produce different spectra for the duration of the PN-sequence. Then, this spectrum is transformed into the time domain by means of the inverse Fourier transform producing the following signal:

$$S(n) = \frac{1}{M} \sum_{k=-M/2, k \neq 0}^{M/2} H(k) * e^{j 2 \Pi * n * k / M} \quad ; n = -M/2, \ldots., M/2 \ without\ 0 \qquad (2)$$

Thus, a signal is produced which is limited in time (corresponding to the chosen length of the Fourier transform) and which is adjusted to the chosen FFT size correctly. If a longer time sequence is wanted, the signal can be cycled. This method permits time sequences of any length.

c) Pause.

The pause has two purposes. An initial pause before applying any measurement signal is necessary to put systems with time-variant transfer functions into a defined initial state. To this end, the pause should be as long as possible (> 1 s). If, however, the system is to be put into a constantly activated state (running speech like), the intermediate pauses should be shorter (about 100 ms) to provide suitable amplitude modulation to the composite signal.

The pause of the CS-sequence proposed is 150 ms.

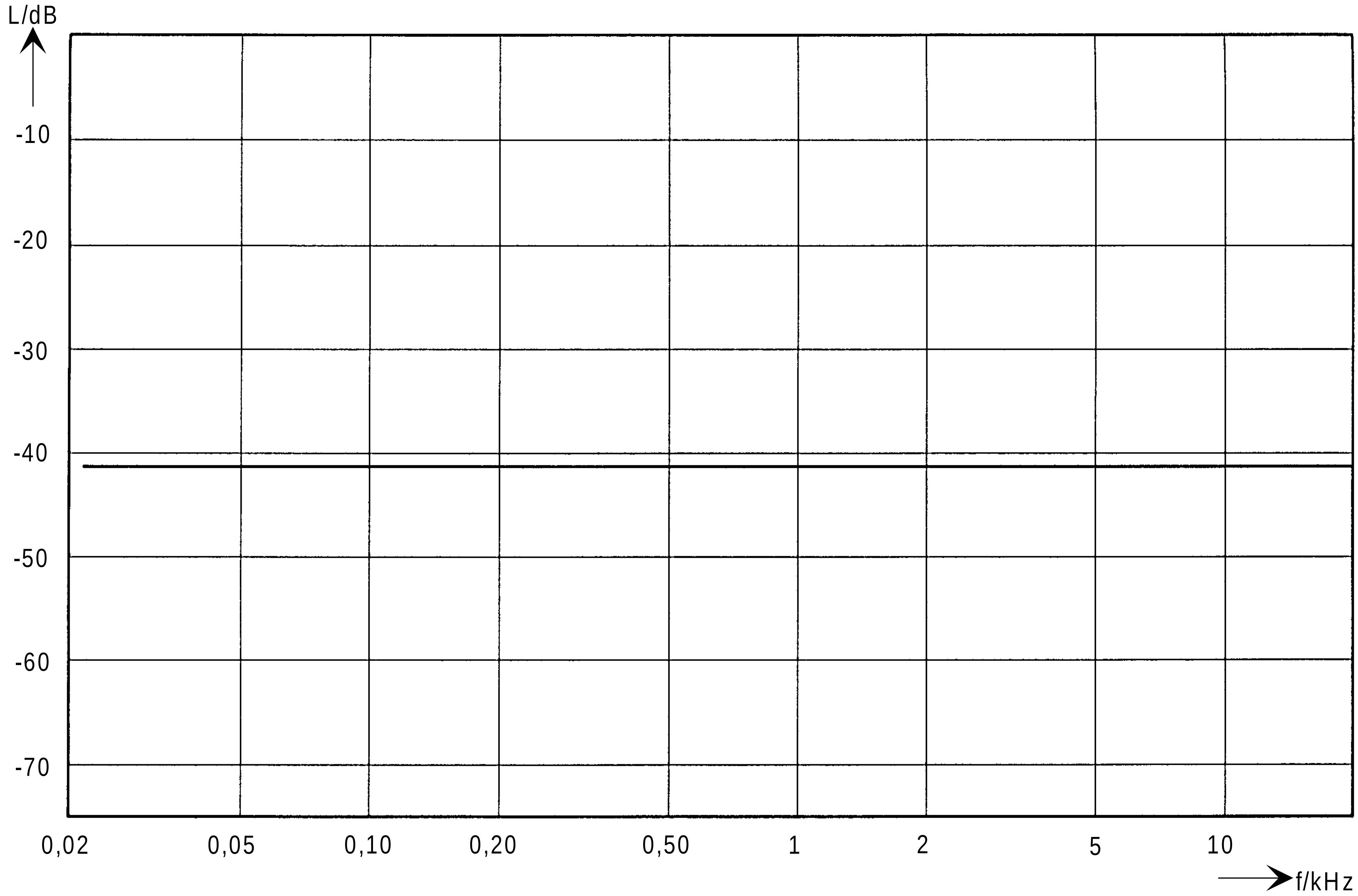


Figure B.1: Spectrum of the PN-sequence

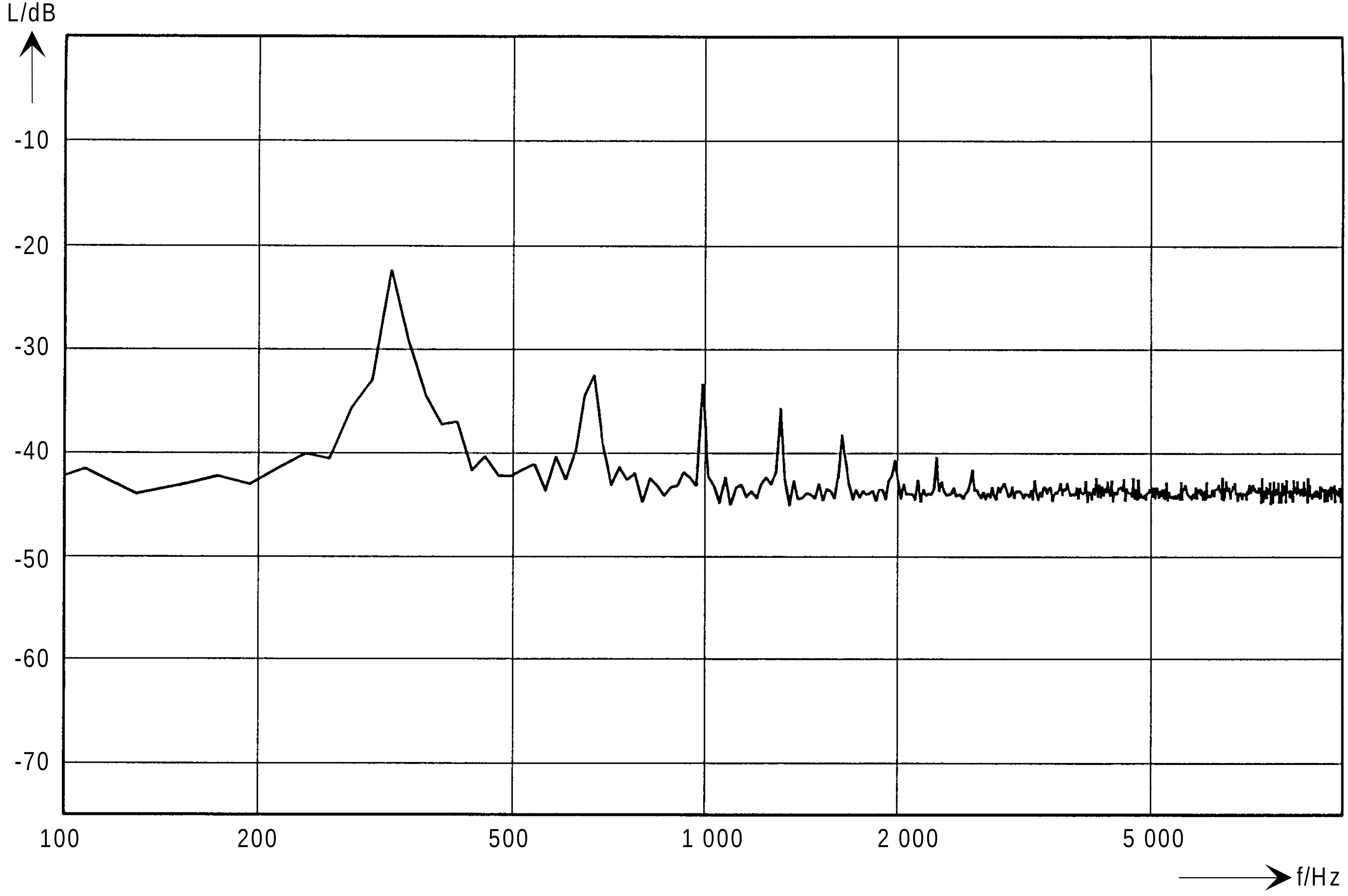


Figure B.2: Spectrum of the complete CSSI

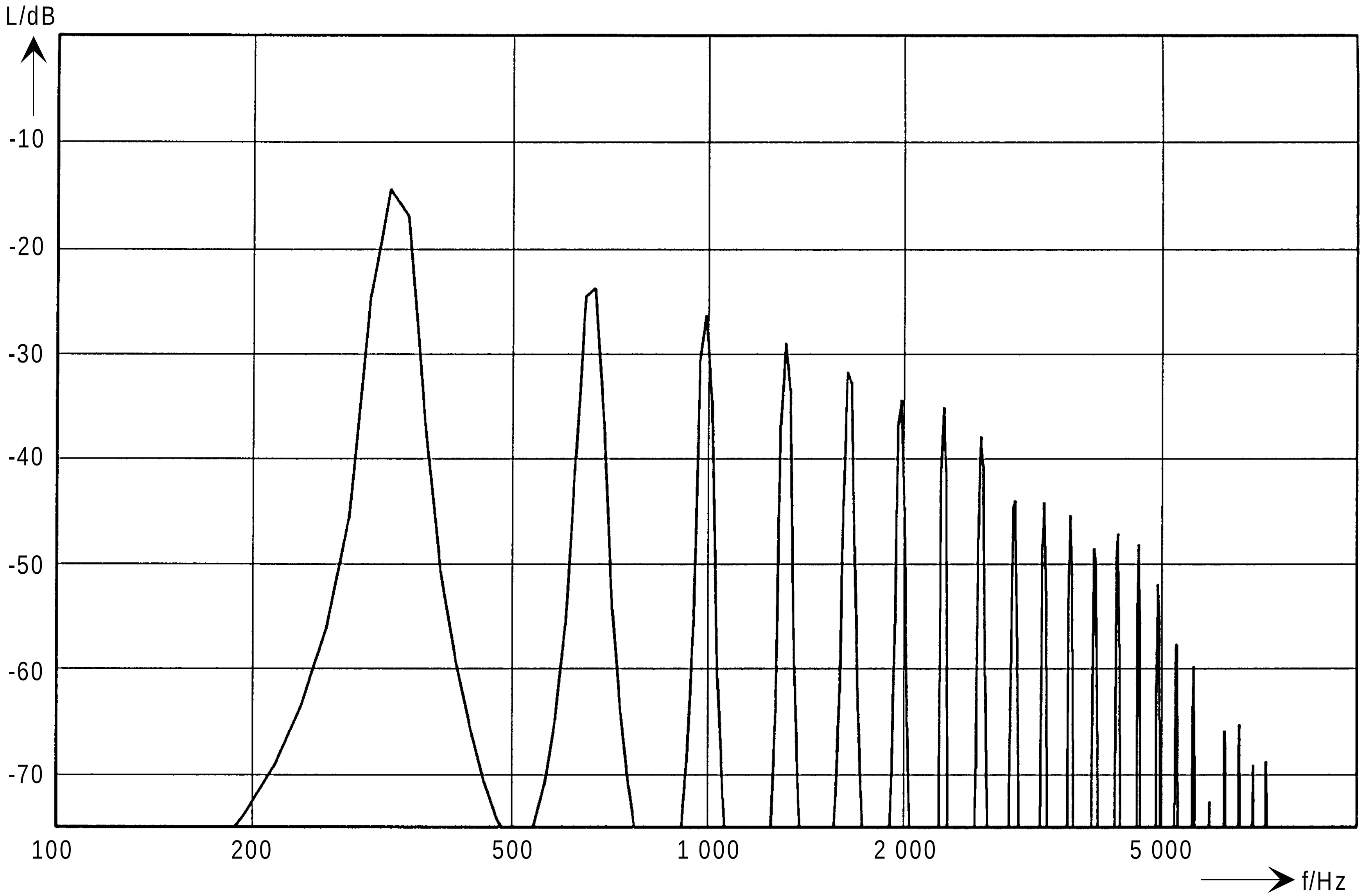


Figure B.3: Spectrum of voiced sound

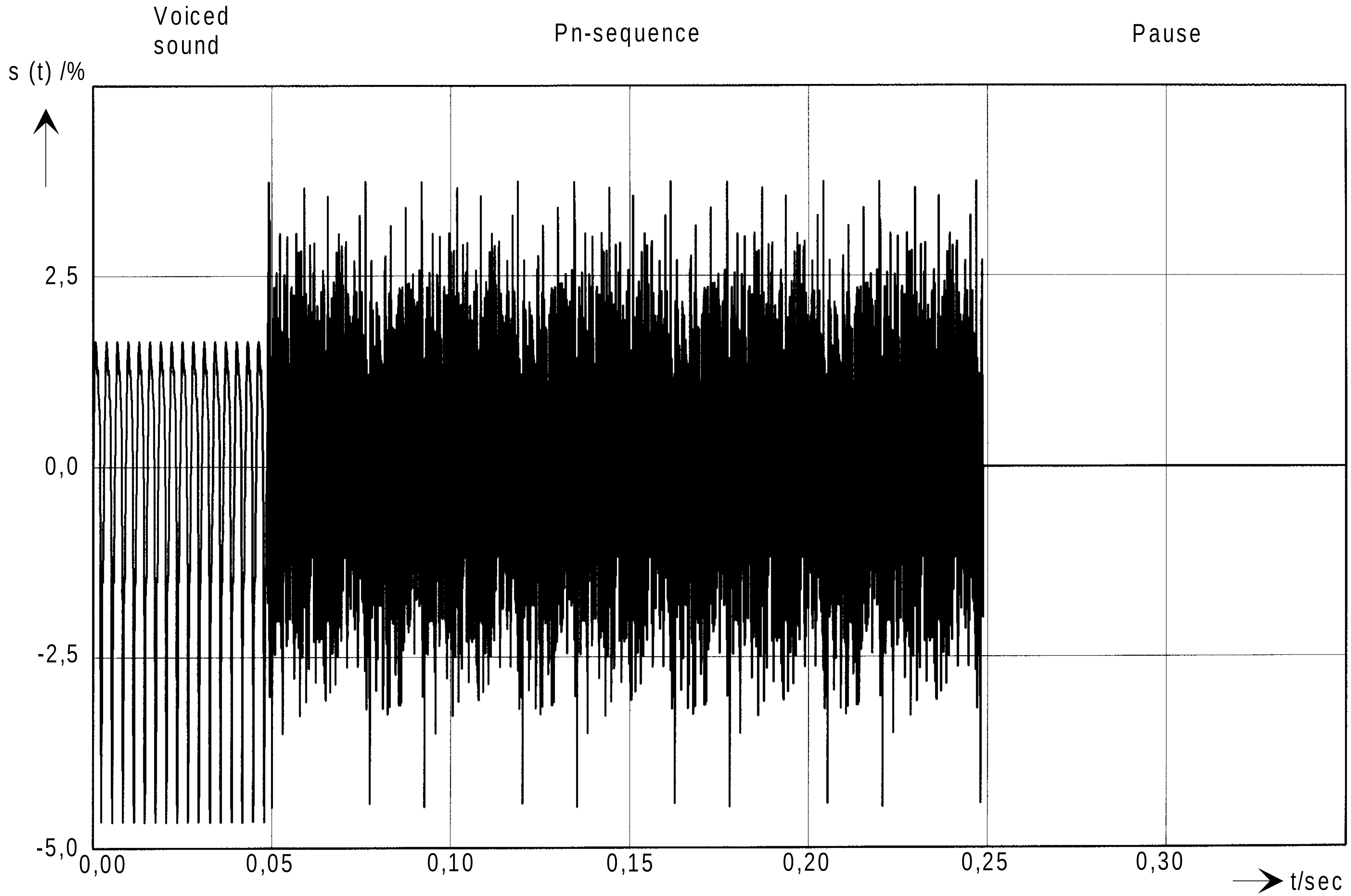


Figure B.4: Composite Source Signal (CSS)

B.2 Use of the CSS

B.2.1 Calculation and analysis

When using the CSS for measurements, the sequence of voiced sound, Pseudo-Noise signal and pause can be cycled. This means that, after the pause, the sequence starts again beginning with a voiced sound. Using this procedure, sequences of any length may be produced.

The following description is for a 44,1 kHz sampling rate. Other sampling rates may be used.

B.2.1.1 Principle of acoustical and electrical calibration - test signal levels

Having created a sequence as described above, this signal can be handled like a standard measurement signal, e.g. like the switched pink noise. The level calibration (acoustical and electrical) is done using the whole sequence including voiced sounds, PN-sequences and pauses. In principle, a standard RMS meter with a bandwidth of 20 kHz operating with "fast" averaging can be used. The preferred method, however, is to use FFT analysis for level calculations. The parameters for the FFT based calculation are:

- 44,1 kHz sampling rate;
- 2 048 points FFT length;
- rectangular windowing;
- no overlap;
- averaging over the whole (cycled) sequence, including voiced sounds, PN-sequences, pauses;
- calculation of the level from the power density spectrum derived by the FFT calculation (integration of the levels over all frequency components).

For acoustical calibration the standard configuration, as described in annex A, is used.

For electrical calibration the levels are calculated in the digital domain referring to dBm0 at the digital interface.

B.2.1.2 Analysis parameters

For the measurement of transfer functions in sending direction and in receiving direction and of Loudness Ratings, etc., the sequence constituted of voiced sound, PN-sequence and pause is cycled as well. The level of the complete sequence is adjusted in such a way that the overall level measured is according to the one specified as described above.

All measurements (analysis) are carried out only during the PN-sequence. For analysis of all transmission parameters in the frequency domain and Loudness Ratings, the measured and Fourier transformed signal should always be referred to the Fourier transformed input signal using the same analysis parameters. This input signal is measured at the reference point for measuring the sending characteristics. For the receiving direction, the measured and Fourier transformed signal is referred to the input signal fed in either at the digital interface or at the reference codec used for the measurement. The measurement can be done using either a two channel measurement system or using techniques where the analysed input signal (measurement signal measured at the reference point or the digital interface) can be stored. For the analysis the following parameters are used:

- 44,1 kHz sampling rate;
- 2 048 points FFT applied to the PN-sequence only;
- rectangular windowing;
- overlapping allowed between 0 % and 99,9 %. The same overlapping needs to be applied for measurement signal at the input of the test object (reference point or digital interface) and measured signal at the output of sending or receiving direction of the test object;
- referring the Fourier transformed signal measured either at the output of sending direction or receiving direction to the Fourier transformed signal at the corresponding excitation point (reference point, digital interface).

Some other measurements require sinusoidal signals or different noise signals to measure different parameters, e.g. distortion. In this case, the PN-sequence is replaced by the corresponding signal e.g. sinusoidal frequency or narrowband noise signal. The levels of the complete cycled composite source signals including the different types of measurement signal are calculated as described above. Measurements are carried out using the measurement signal included in the CSS and using the calculations as described in annex A.

B.2.2 Principles of testing applied to annex A

B.2.2.1 Measurements of frequency responses, TCL and Loudness Ratings, in single talk conditions

For the measurement of transfer functions in sending and receiving direction and for loudness ratings, etc., the sequence is cycled.

All measurements are carried out during the PN-sequence as described above. Loudness Ratings are calculated the same way by calculating the levels, derived from the Fourier transformation, in 1/3-octave bands (according to IEC 225), referring these to the corresponding levels derived from the Fourier transformed input signal in the same 1/3-octave band and calculating the loudness ratings by using the frequency bands as described in annex A, subclause A.2.2.

B.2.2.2 Distortion measurements

For distortion measurements the CSS is, in principle, the same as described in annex B, clause B.1. Instead of the PN-sequence, a sinusoidal frequency at the frequencies as described in annex A, subclause A.2.5 is inserted.

B.2.2.3 Out-of-band measurements

For out-of-band measurements the PN-sequence is substituted by the relevant test signal as described in annex A, subclause A.2.6. The activation is done by the voiced sound. The measurement is carried out during the period while the relevant test signal is applied.

B.2.2.4 Noise measurements

For activation of the set, the voiced sound described in annex B, clause B.1 is used.

B.2.2.5 Delay measurements

For this measurement, the PN-sequence is again substituted by the sinusoidal frequencies described in annex A, subclause A.2.8.

B.2.2.6 Switching characteristics

For measuring the switching characteristics the proposed noise sequence of 250 ms in annex A, subclause A.2.9 is substituted by the sequence of voiced sound and PN-sequence, as described in annex B, clause B.1.

B.2.2.7 Echo cancelling characteristics

For further study.

With reference to annex A, subclause A.1.1, the CSS described in annex B, clause B.1 can be used to measure these parameters.

B.3 Practical Composite Source Signal

B.3.1 Voiced signal to simulate voice properties

The duration of the signal amounts to 48,62 ms. Within this period any speech detector needs to have recognised voice and activated the system. The voiced sound can be described as a sequence of 16-bit words with a sampling rate of 44,1 kHz.

The following table of 134 words should be repeated 16 times to activate the system under test for a time of 48,62 ms.

Table B.1: Samples (linear decimal values, maximum load of 2^{15} = 32 768) of the part 1 (voice sound) of CSS (to be read in columns)

-76	2 098	3 116	2 930	2 392	1 824	1 306	-3 462	-7 492	-2 806	-626
112	2 244	3 158	2 866	2 410	1 772	1 170	-4 024	-6 414	-2 844	-456
298	2 360	3 180	2 808	2 430	1 742	9 68	-4 590	-5 334	-2 888	-298
472	2 456	3 180	2 764	2 444	1 750	702	-5 154	-4 428	-2 898	-130
628	2 538	3 168	2 728	2 460	1 760	394	-5 716	-3 772	-2 846	
776	2 626	3 146	2 686	2 472	1 762	76	-6 298	-3 360	-2 698	
916	2 730	3 132	2 632	2 452	1 736	-244	-6 912	-3 128	-2 460	
1 068	2 824	3 122	2 572	2 398	1 684	-594	-7 556	-3 002	-2 166	
1 234	2 904	3 108	2 496	2 300	1 624	-968	-8 194	-2 924	-1 846	
1 398	2 964	3 096	2 432	2 178	1 572	-1 384	-8 719	-2 870	-1 544	
1 572	2 996	3 076	2 382	2 068	1 516	-1 846	-8 998	-2 830	-1 274	
1 752	3 032	3 038	2 362	1 976	1 460	-2 356	-8 898	-2 800	-1 032	
1 932	3 072	2 992	2 368	1 892	1 390	-2 898	-8 378	-2 792	-818	

B.3.2 Pseudo noise signal

The parameters for the PN-sequence are:

- sampling rate: 44,1 kHz;
- 16-bit-word length;
- length of Fourier transformation: 2 048 points.

$$H(k) = \begin{cases} W(k) * e^{j*\Pi*i_k} & k = -928,....,+928\ without\ 0, i_k \{0,+1\}\ random \\ 0 & else \end{cases} \qquad (3)$$

According to formula (2) with H(k) as described in formula (3) above, the time signal is calculated by inverse Fourier-transformation. This sequence is repeated 4,307 times to achieve a length of 200 ms for the PN-measurement sequence.

According to the frequency resolution of 21,5 Hz (44,1 kHz / 2 048) there are 928 FFT-values in the frequency range between 0 kHz and 20 kHz. Each value W(k) is 152 680. It is calculated in such a way that the levels within a bandwidth of 0 kHz - 20 kHz are the same for the voiced sound and the PN-sequence.

B.3.3 Pause

The pause is used as described in annex B, clause B.1.

Annex C (informative): Bibliography

For the purposes of this Part of the I-ETS, the following references are provided for information:

1) ITU-T Recommendation G.167 (1993): "Acoustic echo controllers".

2) ITU-T Recommendation P.50 (1993): "Artificial voice".

3) ITU-T Recommendation P.58 (1993): "Head and torso simulator for telephonometry".

4) IEC 225 (1966): "Octave, half-octave and third-octave band filters intended for the analysis of sounds and vibrations".

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