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Technical Specification

Terrestrial Trunked Radio (TETRA);
Voice plus Data (V+D);
Part 16: Network Performance Metrics

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Contents

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Terrestrial Trunked Radio (TETRA).

The present document is part 16 of a multi-part deliverable covering the Voice plus Data (V+D), as identified below:

EN 300 392-1: "General network design";

EN 300 392-2: "Air Interface (AI)";

EN 300 392-3: "Interworking at the Inter-System Interface (ISI)";

ETS 300 392-4: "Gateways basic operation";

EN 300 392-5: "Peripheral Equipment Interface (PEI)";

EN 300 392-7: "Security";

EN 300 392-9: "General requirements for supplementary services";

EN 300 392-10: "Supplementary services stage 1";

EN 300 392-11: "Supplementary services stage 2";

EN 300 392-12: "Supplementary services stage 3";

ETS 300 392-13: "SDL model of the Air Interface (AI)";

ETS 300 392-14: "Protocol Implementation Conformance Statement (PICS) proforma specification";

TS 100 392-15: "TETRA frequency bands, duplex spacings and channel numbering";

TS 100 392-16: "Network Performance Metrics";

TR 100 392-17: "TETRA V+D and DMO specifications";

TS 100 392-18: "Air interface optimized applications".

NOTE: Part 10, part 13 (SDL) and part 14 (PICS) of this multi-part deliverable are in status "historical" and are not maintained.

Introduction

This TETRA series is intended to be an open standard that will support a multi-vendor market. In order to support this goal, it is necessary to have a common understanding of the parameters that affect a network's performance and how they can be measured. This is the scope of the present document. Further work may be carried out on values for some of these measured parameters, so that manufacturers and especially network operators can present a consistent quality of service to users of a network whilst supporting a multi-vendor environment.

1 Scope

The present document defines a series of network performance metrics that are applicable to TETRA networks, whose measurement and reporting makes it possible to know the impact of adding new terminals or new infrastructure to an existing TETRA network. Network performance parameters, inherent within a network, include those, which affect to the quality of an "end-to-end" connection as experienced by a subscriber. A network performance parameter may be considered as a function of the operation of the elements involved to form a connection, network load, network signalling and the processing required to realize a connection.

The present document contains voice quality test methods and values for full duplex calls as well as the transmission level plan for simplex calls; e.g. nominal sending level, sending level performance values and test methods.

Requirements on some measured values are outside the scope of the present document.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] ETSI EN 300 903: "Digital cellular telecommunications system (Phase 2+) (GSM); Transmission planning aspects of the speech service in the GSM Public Land Mobile Network (PLMN) system (GSM 03.50)".

[2] ETSI EN 300 392-2: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 2: Air Interface (AI)".

[3] ETSI EN 300 395 (series): "Terrestrial Trunked Radio (TETRA); Speech codec for full-rate traffic channel".

[4] ITU-T Recommendation G.100.1: "The use of the decibel and of relative levels in speechband telecommunications".

[5] ITU-T Recommendation G.111: "Loudness ratings (LRs) in an international connection".

[6] ITU-T Recommendation G.121: "Loudness ratings (LRs) of national systems".

[7] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[8] ITU-T Recommendation G.712: "Transmission performance characteristics of pulse code modulation channels".

[9] ITU-T Recommendation P.38: "Transmission characteristics of operator telephone systems (OTS)".

[10] ITU-T Recommendation P.50: "Artificial voices".

[11] ITU-T Recommendation P.51: "Artificial mouth".

[12] ITU-T Recommendation P.56: "Objective measurement of active speech level".

[13] ITU-T Recommendation P.57: "Artificial ears".

[14] ITU-T Recommendation P.58: "Head and torso simulator for telephonometry".

[15] ITU-T Recommendation P.64: "Determination of sensitivity/frequency characteristics of local telephone systems".

[16] ITU-T Recommendation P.79: "Calculation of loudness ratings for telephone sets".

[17] ITU-T Recommendation P.340: "Transmission characteristics and speech quality parameters of hands-free telephones".

[18] ITU-T Recommendation P.380: "Electro-acoustic measurements on headsets".

[19] ITU-T Recommendation P.501: "Test signals for use in telephonometry".

[20] ITU-T Recommendation P.581: "Use of head and torso simulator (HATS) for hands-free terminal testing".

[21] ITU-T Recommendation P.830: "Subjective performance assessment of telephone-band and wideband digital codecs".

[22] ISO 3: "Preferred numbers - Series of preferred numbers".

[23] ISO 3745: "Acoustics - Determination of sound power levels of noise sources using sound pressure - Precision methods for anechoic and hemi-anechoic rooms".

[24] IEC 61672-1: "Electroacoustics - Sound level meters - Part 1: Specifications".

[25] ISO 9614: "Acoustics - Determination of sound power levels of noise sources using sound intensity".

3 Definitions, symbols and abbreviations

3.1 Definitions

The definition for a specific network performance parameter or metric has been included in the annex applicable.

For the purposes of the present document, the terms and definitions given in EN 300 392-2 [2] and the following apply:

dBPa: sound pressure level relative to 1 Pascal expressed in dB

NOTE: 0 dBPa is equivalent to 94 dB SPL.

egress: elements within a network that comprise the output portion of an end-to-end connection between calling and called subscribers

end-to-end: scenario referred to a connection between the calling and called subscribers or applications (which may include more than one TETRA SwMI)

ingress: elements within a network that comprise the input portion of an end-to-end connection between calling and called subscribers

listener: subscriber who is currently receiving communication from the "talker"

network: all the elements required to provide the services available for the calling and, or, called subscriber including the users' apparatus as appropriate

NOTE: This definition of network is in contrast to the definition in the other parts of TETRA standards, where the word Network refers to the fixed part of the networks, also called SwMI without inclusion of radio terminals.

subscriber A: call originating user

NOTE: In other parts of TETRA standards "subscriber A" is also referred as "user A".

subscriber B: call receiving user

NOTE: In other parts of TETRA standards "subscriber B" is also referred as "user B".

talker: subscriber who is currently communicating with the "listener"

3.2 Symbols

For the purposes of the present document, the following symbols apply:

E	Egress
E_R	Egress R reference point
E_{R1}	Egress R_1 reference point
E_S	Egress S reference point
E_T	Egress T reference point
E_U	Egress U reference point
E_{U1}	Egress U_1 reference point
E_{Un}	Egress U_n reference point
E_V	Egress V reference point
E_{Vn}	Egress V_n reference point
E_W	Egress W reference point
I	Ingress
I_R	Ingress R reference point
I_{R1}	Ingress R_1 reference point
I_S	Ingress S reference point
I_T	Ingress T reference point
I_U	Ingress U reference point
I_{U1}	Ingress U_1 reference point
I_V	Ingress V reference point
I_{Vn}	Ingress V_n reference point
I_W	Ingress W reference point
ms	milliseconds

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ADC	Analogue to Digital Conversion
AGC	Automatic Gain Control
BS	Base Station
DAC	Digital to Analogue Conversion
DTS	Digital Test Sequence
DTX	Discontinuous Transmission
ERP	Ear Reference Point
GSM	Global System for Mobile communications
HATS	Head And Torso Simulator
IMP	Intermediate Monitoring Point
ISI	Inter-System Interface
MOS	Mean Opinion Score
MRP	Mouth Reference Point
MS	Mobile Station
OLR	Overall Loudness Rating
PLMN	Public Land Mobile Network
POI	Point Of Interconnection
PSTN	Public Services Telephone Network
QoS	Quality of Service

RLR	Receive Loudness Rating
SLR	Send Loudness Rating
SS	System Simulator
STMR	SideTone Masking Ratio
SwMI	Switching and Management Infrastructure
TCH	Traffic CHannel
TCL	Terminal Coupling Loss
TETRA	TErrestrial Trunked RAdio

4 Reference model for determination of a metric at an Intermediate Monitoring Point (IMP)

Figure 1 illustrates a model detailing Intermediate Monitoring Points (IMPs) where an intermediate network performance metric may be observed from.

Arrangements to monitor the appropriate information at an intermediate monitoring point, or points, are outside the scope of the present document.

The measurement of a metric may be a combination of the criterion detailed in the following clauses.

4.1 Intermediate Monitoring Point (IMP)

For the purposes of the present document Intermediate Monitoring Points (IMPs) shall be as defined in figure 1. It should be noted that some of the IMPs may be manufacturer specific, or non-existent in a particular network, and that several IMPs may be defined (from I_v to I_{vn} and from E_v to E_{vn}).

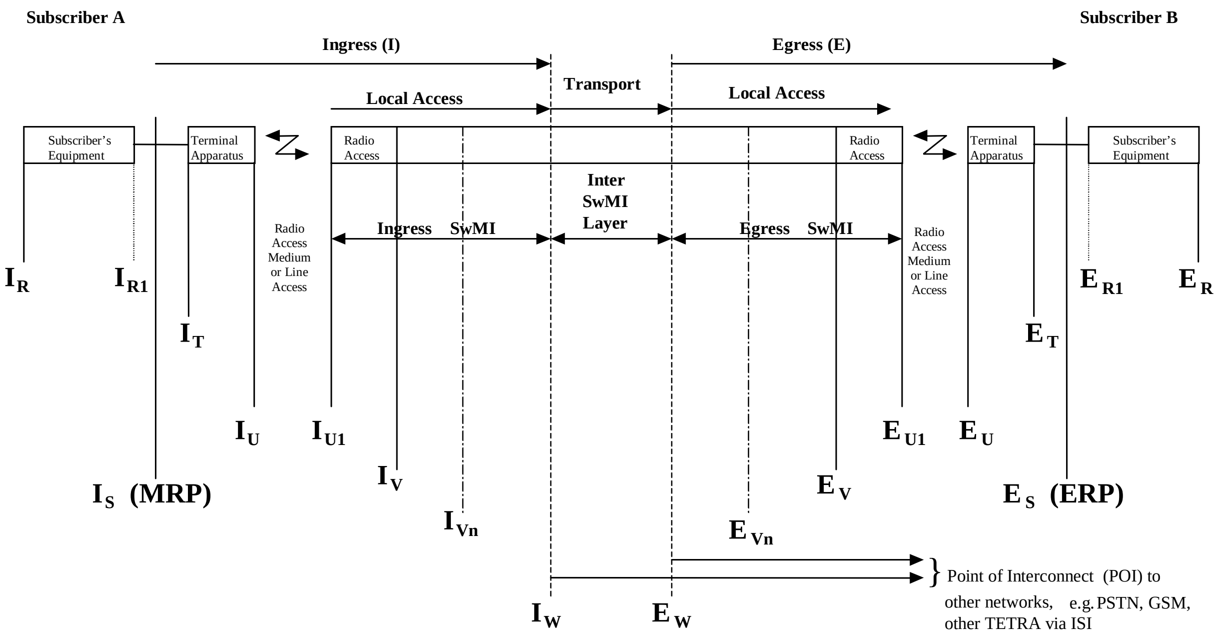


Figure 1: Intermediate Monitoring Point (IMP) model

4.2 Time domain model

Table 1 defines time instances to complement figure 1 to detailing the measurement of a network performance metric at IMPs in association with the time domain.

Table 1: Time domain instances

Time recorded at an IMP	Event	Remark
time t_a	Time when stimuli originating from subscriber A is observed at the given IMP.	The observed IMP may be any ingress or egress point.
time t_b	Time when stimuli originating from subscriber A is observed at the given IMP other than the IMP where time t_a was observed.	The observed IMP may be any ingress or egress point further towards subscriber B than IMP for t_a, so by default time t_a is less than time t_b.
time t_c	Time when network returns a valid response towards subscriber A due to the stimuli originating from subscriber A observed at the given IMP, (see note).	This time instance may be an intermediate or final response to the stimuli originating from subscriber A. Time t_b and time t_c has no pre-defined relationship.
time t_d	Time when network returns a valid response towards subscriber B due to the stimuli originating from subscriber A observed at the given IMP.	The observed IMP may be any ingress or egress point further towards subscriber B, so by default time t_a is less than time t_d.
time t_e	Time when stimuli originating from subscriber B is observed at the given IMP.	The observed IMP may be any ingress or egress point. When used in call set-up scenarios, then by default time t_d is less than time t_e. Time t_d may not have any relation to the measurement.
time t_f	Time when stimuli originating from subscriber B is observed at the given IMP other than the IMP where time t_e was observed.	The observed IMP may be any ingress or egress point further towards subscriber A, so by default time t_e is less than time t_f.
time t_g	Time when network returns a valid response towards subscriber B due to the stimuli originating from subscriber B observed at the given IMP, (see note).	This time instance may be an intermediate or final response to the stimuli originating from subscriber B. Time t_f and time t_g has no pre-defined relationship.
time t_h	Time when network returns a valid response towards subscriber A due to the stimuli originating from subscriber B observed at the given IMP.	The observed IMP may be any ingress or egress point further towards subscriber A other than IMP for t_e, so by default time t_e is less than time t_h.
time t_x	Time when network sends a first command.	The observed time t_x may be in relation of a call independent of observed time t_y in relation to another call.
time t_y	Time when network sends a second command.	The observed time t_y may be in relation of a call independent of observed time t_x in relation to another call.
NOTE: This table identifies only single observation time for a response back to the stimuli generating subscriber although even for that scenario there could be more than a single monitoring point.		

The time domain instances in table 1 are independent of possible interactions between subscriber A and network actions. Especially network may send a message observed at the IMP at time t_c without any stimuli originating from an action at subscriber B at time t_e.

NOTE: Multiple time instance may be observed due to a single stimuli e.g. t_a may be followed by one (t_c) in the direction towards subscriber A and another (t_d) in the direction towards subscriber (or subscribers) B.

As defined in table 1, a network performance metric observed from an IMP, or IMPs, may be determined as time differences e.g. from equations 1, 2 and 3:

Network performance metric time delay on subscriber A point of view = (time t_c) - (time t_a) **(1)**

An unidirectional network performance metric time delay = (time t_d) - (time t_a) **(2)**

Another unidirectional network performance metric time delay = (time t_h t_c) - (time t_e t_b) **(3)**

4.2.1 Measurements at a single IMP

When a network performance metric is measured at a single IMP, with the measurement in the direction towards subscriber B, times t_a and t_c or t_h, refer to table 1, shall be recorded at the same IMP for the purposes of calculation using e.g. equation 1 (e.g.: I_v within figure 1). When the measurement is in the directions towards subscriber A then e.g. times t_e and t_g are applicable.

4.2.2 Measurements between IMPs

When a network performance metric is to be measured from one IMP to another IMP, then almost any combination of times defined in the table 1 may be applicable and the times as appropriate shall be recorded at the appropriate two IMPs for the purposes of calculation (e.g.: between I_v and E_v within figure 1).

4.3 Traffic load

Traffic load may be considered to influence the result obtained when conducting measurements for a network performance metric. Network performance measurements may be considered for load levels of:

a) low traffic load;

b) medium traffic load; and

c) high traffic load.

NOTE: The traffic load definition may depend upon the service under measurement.

4.4 Network infrastructure

Network performance metrics may be considered in accordance with the infrastructure used to realize the connection serving subscribers A and B.

The geographical separation between subscribers A and B may influence the network performance metric result and measurement scenarios should be defined accordingly.

Measurement scenarios could be:

a) subscribers A and B served by the same TETRA Base Station (BS);

b) subscribers A and B are each served by TETRA Base Stations located at the effective extremities of the TETRA Network;

c) one of the subscribers is not served by the TETRA network.

In the scenario c) the network performance measurements may be performed at the Point of Interconnection (POI) between the TETRA network and the other network and be recorded for the TETRA network's portion.

Network performance measurement between monitoring points involving more than one network may be conducted end-to-end or between intermediate monitoring points, as appropriate.

4.5 TETRA services

A Network Performance Metric may be defined for:

a) Voice Services (Full-Duplex) involving a calling and called subscriber;

b) Voice Services (Half -Duplex) involving a calling and called subscriber;

c) Voice Services (Group Calls) involving more than two subscribers;

d) Data Services supporting Short Data Service messaging to and from a subscriber (including Status Messaging);

e) Data Services supporting Packet Data to and from a subscriber;

f) Data Services supporting Circuit Mode Data to and from a subscriber.

The definition of TETRA services is outside the scope of the present document, refer to EN 300 392-2 [2] for details.

When a new service is introduced to TETRA standard, the present document may need to be revised to cover it.

5 Factors affecting to the measurement results

The measurement results are dependent of many parameters, external as well as internal. Also the definition of the time an event has occurred has influence. The actual measurement arrangements should be recorded and results should be used carefully.

The measurements may be used as an aid to find difficulties in the system without actually identifying the reason or reasons. Operators may use the results to obtain a consistent grade of service in a multi-vendor TETRA network.

The identification of reason may require additional measurement equipment or measurement points.

Examples of parameters having influence on the measured values are:

- RF coverage;
- MCCH random access frame length;
- ACCH random access frame length;
- emergency call pre-emption;
- the number of intermediate entities (intra network signal routing);
- a traffic channel queue;
- subscriber access priority; and
- MS transmit permission.

Examples of definitions having influence on the measured values are:

- how the framing delay imposed by frame 18 is to be shared between the speech encoder and speech decoder, when determining the up-link voice delay of a terminal;
- whether a voice signal stimulus is considered as being detected at the air interface of the terminal, when transmission of the first block of two ACELP blocks is starting (start of slot), when both ACELP blocks have been transmitted (end of slot), or when both ACELP blocks have been sent (end of slot) plus the time represented by the displacement of the stimulus location from the start of the first block;
- whether message transmission time instance is at the start of the message transmission (first bit) or when the whole message is sent (last bit); and

- whether message reception time instance is at the reception of the message's first bit, at the reception of the message's last bit or at the completed decoding of the message (including total or partial re-transmissions due to propagation error and delivery to the message user).

One consistent manner to take into account message transmission and reception at the air interface is that:

- the message transmission instance is the time, when the transmission of the timeslot, which contains the message, starts (first bit); and
- the message reception instance is the time , when the message is completely received and delivered to the layer that is the user of the message (last bit plus needed lower layer processing time).

NOTE: Although the above definitions are nice on the air interface protocol point of view, their measurement may not be practicable in typical situations.

Although the present document identifies those factors their detailed mechanisms and how they should be taken into account are outside the scope of the present document.

6 Narrow-band full duplex transmission performance, loudness ratings, and sending level

6.1 Applicability

Clauses 6.2 to 6.3 contain performance requirements for terminals used to provide narrow-band full duplex calls, either as a stand-alone service, or as part of a multimedia service.

The value of SLR or sending level is strongly recommended, but it is recognized that where backward compatibility is an issue that these values need not be applied.

NOTE: It is acknowledged that in a multi-vendor network or where the TETRA Inter-System Interface (ISI) is involved that moves may be required towards this value by the operators and users of the network.

6.2 Overall loss/loudness ratings

6.2.1 General

An international telephony connection involving a TETRA network and the PSTN should meet the overall loudness rating (OLR) limits in ITU-T Recommendation G.111 [5]. The national parts of the connection should therefore meet the send and receive loudness rating (SLR, RLR) limits in ITU-T Recommendation G.121 [6].

OLR is defined as:

$$\text{OLR} = \text{SLR} + \text{Circuit Loss} + \text{RLR}$$

For the case where digital routings are used to connect the TETRA network to the international chain of circuits, the SLR and RLR of the national extension will be largely determined by the SLR and RLR of the TETRA network. The limits given below are consistent with the national extension limits and long-term objectives in ITU-T Recommendation G.121 [6].

The SLR and RLR values for the TETRA network apply up to the POI.

However, since the circuit loss of a TETRA SwMI is 0 dB, then the main determining factors are the characteristics of the terminal, including the analogue to digital conversion (ADC) and digital to analogue conversion (DAC). In practice, it is convenient to specify loudness ratings to the Air Interface. For the normal case, where the TETRA SwMI introduces no additional loss between the Air Interface and the POI, the loudness ratings to the PSTN boundary (POI) will be the same as the loudness ratings measured at the Air Interface. However, in some cases loss adjustment may be needed for interworking situations in individual countries.

The POI is a 0 dBr level reference point.

6.2.2 Connections with handset terminal

The nominal values of SLR to the 0 dBr level reference point should be:

SLR = 8 dB ± 3 dB;

Compliance shall be checked by the relevant tests described in clause E.7 of the present document.

6.2.3 Connections with desktop and vehicle-mounted hands-free terminal

The nominal values of SLR to the 0 dBr level reference point should be:

SLR = 13 dB ± 4 dB;

Compliance shall be checked by the relevant tests described in clause E.7 of the present document.

6.2.4 Connections with handheld hands-free MS not having an AGC function

The nominal values of SLR to the 0 dBr level reference point should be:

SLR = 13 dB ± 4 dB;

Compliance shall be checked by the relevant tests described in clause E.7 of the present document.

6.2.5 Connections with headset terminal

The nominal values of SLR to the 0 dBr level reference point should be:

SLR = 8 dB ± 3 dB;

Compliance shall be checked by the relevant tests described in clause E.7 of the present document.

6.3 Sending level of half duplex terminals with send AGC function

The sending level of speech in the 0 dBr level reference point of Half duplex terminals with send AGC function should be:

-20 dBm0 ± 4 dB for the range of use conditions specified by the manufacturer.

Compliance shall be checked by the relevant tests described in clause E.7.2.5 of the present document.

7 Transmission level plan for simplex calls

7.1 Scope

Clauses 7.2 to 7.7 define the transmission level plan for simplex calls in TETRA systems. The transmission level plan is described in terms of Nominal Sending Level, sending level performance requirements applicable for TETRA terminals and test methods.

The TETRA SwMI provides lossless transport of speech data between terminals and is therefore not affected by the clauses 7.2 to 7.7, refer to figure 2.

7.2 Transmission level plan for simplex calls

In simplex calls such as Group Call and Individual Call the acoustic interface may differ from those used for full duplex calls. Typically the talker holds the microphone of a handheld terminal in a location determined by him and influenced by his concurrent tasks, his personal preferences and to some degree the instructions in the user guide that comes with the terminal.

Therefore the acoustic interface often used for simplex calls has poor or no control of the distance from the talker's mouth to the microphone and so has both user dependent and time variant acoustic loss.

Unlike full duplex calls the listener receiving a simplex call cannot ask the talker to speak up, or to move closer to the microphone, if the receiving loudness is too low. Instead the listener must request a repetition of the message and critical seconds may be lost as result of this.

Therefore is use of a constant sending sensitivity not recommended for simplex calls.

To allow for a range of use distances and sending sensitivity values implemented dynamically during the call by means of an Automatic Gain Control (AGC) function, a Nominal Sending Level for speech is defined for simplex calls.

As consequence, TETRA systems have separate transmission level plans for full duplex and simplex calls.

However, the Nominal Sending Level for simplex transmission defined in the present annex ensures that the sending level of simplex terminals and the average sending level of full duplex terminals applying Sending Loudness Ratings are consistent.

7.3 Reference points and use conditions

7.3.1 0 dBr points of a simplex connection

The definition and location of 0 dBr points in terminals and system are common for simplex and full duplex connections. figure 2 describes the location of 0 dBr points of a simplex connection. The 0 dBr level reference point of a sending terminal is defined as the uniform PCM interface (UPCMI) of the TETRA speech encoder. The 0 dBr level reference point of a receiving terminal is defined as the UPCMI of the TETRA speech decoder. The loss of speech power from the UPCIM of the sending terminal to the UPCMI of the receiving terminal is considered being 0 dB. The transmission maximum in a 0dBr point is defined as +3,14 dBm0. See ITU-T recommendation G.100.1 [4] for further information on these topics.

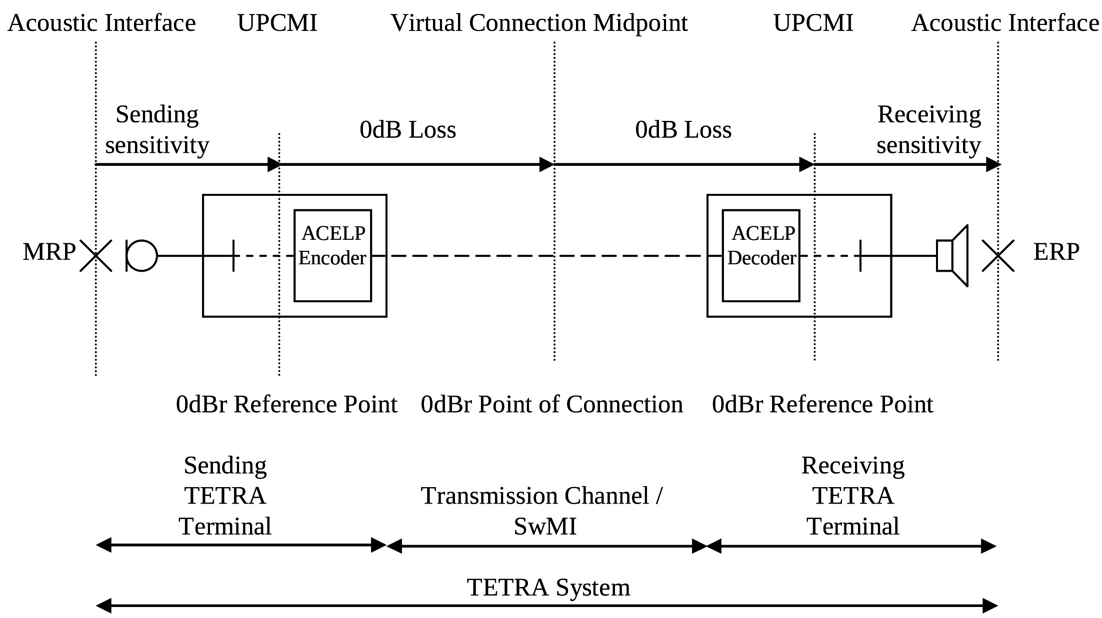


Figure 2: 0dBr points in a simplex connection

7.3.2 Characterization of use conditions for a terminal microphone

The manufacturer shall specify unambiguously the nominal set-up of the terminal equipment and the terminal microphone in particular and shall specify the nominal use condition associated with this set-up. The manufacturer shall specify whether the terminal has a send AGC function.

The nominal use condition is characterized by the nominal sound pressure of speech in the MRP and the nominal distance from the centre of the lip plane to the terminal microphone. This distance is equal to the length of the vector from the centre of the lip plane to the centre of the microphone surface or inlet hole (see ITU-T Recommendation P.58 [14] for information on MRP and lip plane).

For handset terminals, headset terminals, vehicle mounted terminals and desktop operated terminals the manufacturer shall specify the nominal set-up of the terminal so that the relevant test set-up from clause 7.5.1 to apply can be selected.

For handheld terminals, shoulder mounted terminals, and terminals with other types of user worn acoustic interfaces the test set-up with HATS is given by manufacturer's specification of the nominal set-up of the terminal equipment, with the position defined using the co-ordinate system in clause 7.5.1.6.

For terminals with send AGC function variation in use relative to the nominal use condition may be supported. The manufacturer should specify the two extremes of the range of use conditions supported. These extremes are defined as the combination of maximum distance and minimum sound pressure in the MRP respectively the combination of minimum distance and maximum sound pressure in the MRP. These combinations produce the lowest respectively the highest sound pressure at the microphone.

7.4 Nominal sending level and performance

7.4.1 Applicability

Clause 7.4 contains performance requirements for terminals used to transmit speech into the SwMI in simplex calls such as Group calls and Individual calls. The performance requirements apply for terminals covered by the following definition:

A TETRA terminal is defined as a self-contained or composite system boundary component having an acoustic interface and a TETRA speech codec.

The value of sending level is strongly recommended, but it is recognized that where backward compatibility is an issue that this value need not be applied.

NOTE: It is acknowledged that moves may be required towards this value by the operators and users of the network.

7.4.2 Nominal Sending Level

The Nominal Sending Level in the 0 dBr level reference point at the uniform PCM input of the TETRA speech codec should be -20 dBm0.

This value equates to -26,15 dBov (see ITU-T Recommendations P.830 [21] and G.100.1 [4]).

7.4.3 Sending level performance

7.4.3.1 Terminals without send AGC function

For simplex terminals with constant sending sensitivity, the sending level of speech in the 0 dBr level reference point at the uniform PCM input of the TETRA speech codec should be:

-20 dBm0 ± 4 dB for the nominal use condition specified by the terminal manufacturer.

The sending level of speech is defined as the active speech level according to ITU-T Recommendation P.56 [12] method B. Refer to clause 7.3.2 for characterization of the nominal use condition.

Compliance shall be verified by the relevant test described in clause 7.7 of the present document.

7.4.3.2 Terminals with send AGC function

For simplex terminals with variable sending sensitivity controlled by an AGC function, the sending level of speech in the 0 dBr reference point at the uniform PCM input of the TETRA speech codec should be:

-20 dBm0 ± 4 dB for the range of use conditions specified by the terminal manufacturer.

The sending level of speech is defined as the active speech level according to ITU-T Recommendation P.56 [12] method B. Refer to clause 7.3.2 for characterization of use conditions.

Compliance shall be verified by the relevant test described in clause 7.7 of the present document.

7.5 Test configurations

This clause describes the test setups for terminal acoustic testing.

7.5.1 Test setup for terminals

The general access to terminals is described in figure 3. The preferred acoustic access to terminals is the most realistic simulation of the "average" user. This shall be made by using HATS (head and torso simulator), with an appropriate ear simulation.

Appropriate mountings shall be used for handset terminals and handheld terminals. For terminals and microphones intended for mounting on the user's torso appropriate mountings and cloth simulation shall be applied on the torso of the HATS.

HATS is described in ITU-T Recommendation P.58 [14]. Appropriate artificial ears are described in ITU-T Recommendation P.57 [13] (type 3.3 and type 3.4 ear). A proper positioning of handsets in realistic conditions is found in ITU-T Recommendation P.64 [15], the test setups for various types of hands-free terminals can be found in ITU-T Recommendation P.581 [20].

The preferred way of testing is the connection of a terminal to a system simulator or to a system with a terminal simulator with exact defined settings and access points. The test responses are accessed either, electrically using a reference codec or using the direct signal processing approach.

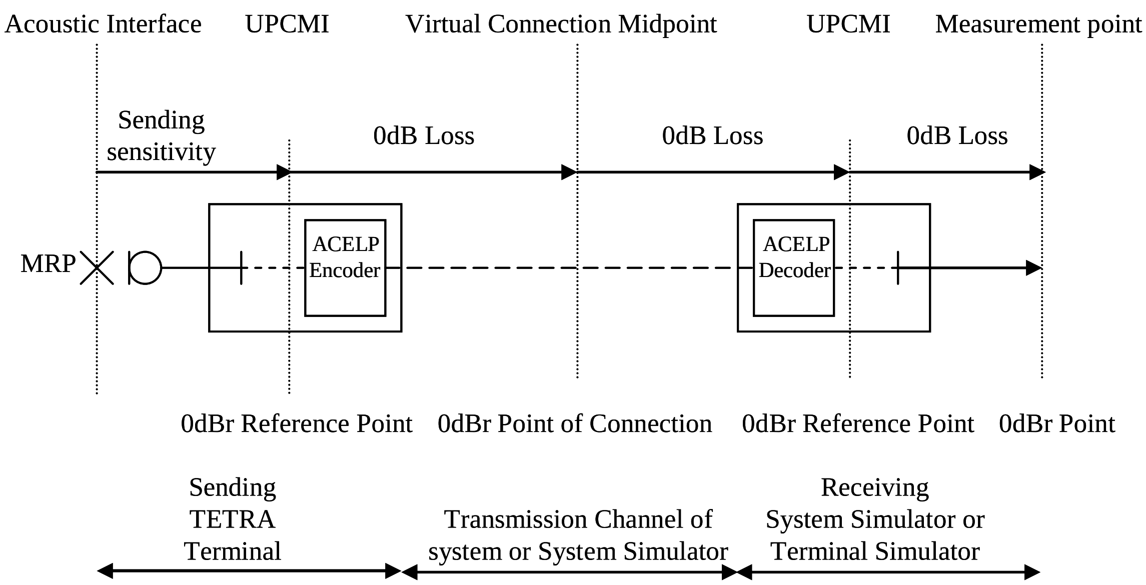


Figure 3: General access to terminals in a simplex connection

7.5.1.1 Set-up for handset terminals

The handset is mounted at the HATS equipped with type 3.3 or 3.4 ear simulators.

The handset is placed in the standardized position, HATS position, as described in ITU-T Recommendation P.64 [15]. This includes that the handset shall be in the ERP position (see ITU-T Recommendation P.64 [15]).

The artificial mouth shall conform to ITU-T Recommendation P.58 [14].

NOTE: The artificial ears of the HATS are not used for measurements.

7.5.1.2 Set-up for headset terminals

The headset is mounted at the HATS equipped with type 3.3 or type 3.4 ear simulators.

The position of the headset should be according to ITU-T Recommendation P.380 [18] and the force against the artificial ear shall be the same as applied in normal use.

For binaural headsets the HATS shall be equipped with a right ear simulator and a left ear simulator.

The type 3.3 ear simulator can be used for all types of headsets and earphones, whereas the type 3.4 ear simulator cannot be used with supra-concha headsets, supra-aural headsets and forward facing intra-concha headsets (see ITU-T Recommendation P.57 [13]).

For HATS with type 3.3 ear simulator, the soft pinna (hardness 35 degrees ± 6 degrees Shore-OO) shall be used (see ITU-T Recommendation P.57 [13]).

The artificial mouth shall conform to ITU-T Recommendation P.58 [14]. The artificial ear shall conform to ITU-T Recommendation P.57 [13] (type 3.3 or type 3.4).

NOTE: The artificial ears of the HATS are not used for measurements. The pinna simulators contribute to position the headset microphone as in normal use.

7.5.1.3 Set-up for vehicle mounted terminals

Vehicle mounted terminals may be measured either in a vehicle or in an anechoic room. For both of these two types of test environments, the setup will depend on whether HATS or a discrete artificial mouth is used as the acoustic test equipment.

For in-vehicle measurements, if HATS test equipment is used, the microphone should be positioned in the car as per ITU-T Recommendation P.581 [20]. The artificial mouth of the HATS shall comply with ITU-T Recommendation P.58 [14]. If in- vehicle measurements are made with a discrete artificial mouth, this should be positioned in the vehicle as per figure O.3. The discrete artificial mouth shall comply with ITU-T Recommendation P.51 [11]. A vehicle simulator may be used instead of an actual car. A standard vehicle simulator is described in TR 101 110 (see bibliography) Digital Cellular Telecommunications System (Phase 2+) Characterization test methods and quality assessment for hands-free mobiles.

The terminal equipment is mounted in the car as specified by the manufacturer.

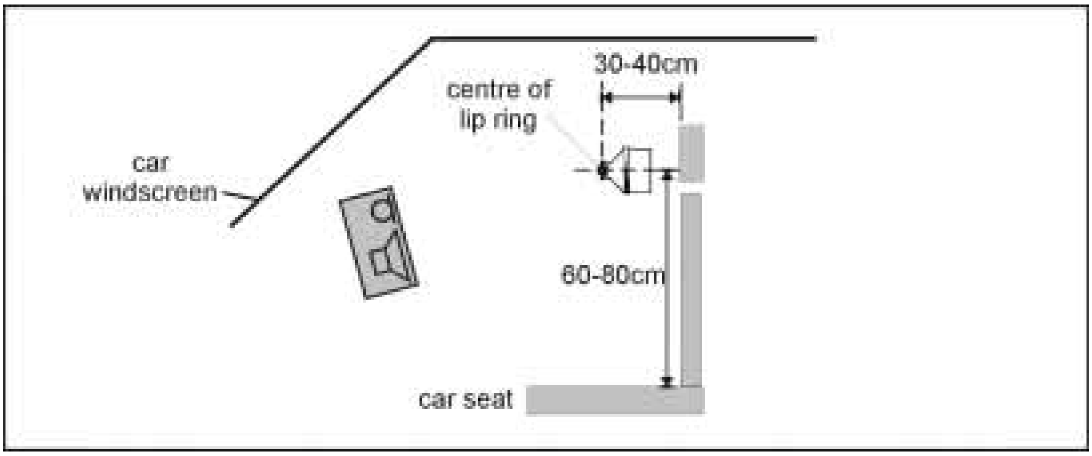


Figure 4: Test configuration for vehicle-mounted terminal, sending level, with discrete ITU-T Recommendation P.51 [11] artificial mouth

7.5.1.4 Set-up for desktop operated terminals

For HATS test equipment, setup for desktop terminals can be found in ITU-T Recommendation P.581 [20]. Measurement setup using a discrete ITU-T Recommendation P.51 [11] artificial mouth for desktop terminals can be found in ITU-T Recommendation P.340 [17].

The terminal equipment is set-up on the desk as specified by the manufacturer.

7.5.1.5 Position and calibration of HATS

The horizontal positioning of the HATS reference plane shall be guaranteed within ±2 degrees for testing desktop and vehicle mounted equipment.

For testing desktop and vehicle-mounted equipment, calibration of the artificial mouth in addition to that included in the test method descriptions is for further study.

7.5.1.6 Set-up for handheld terminals, shoulder-mounted terminals, and terminals with other sorts of user worn acoustic interfaces

The terminal is positioned in relation to the centre of the lips of the HATS (which is defined in ITU-T Recommendation P.64 [15]) by the terminal manufacturer for the use cases of the terminal as specified in clause 7.3.2.

The three-dimensional position of the terminal's microphone relative to the centre of the lips of the HATS shall be defined using the Cartesian co-ordinate system (x_m, y_m, z_m) described in annex E of ITU-T Recommendation P.64 [15].

The positions tested shall be recorded and stated with the test results.

7.5.2 Setup of the electrical interfaces

7.5.2.1 Codec approach and specification

Codec approach: In this approach, a codec is used to convert the companded digital input/output bit-stream of the system simulator to the equivalent analogue values. With this approach either a system with a terminal simulator providing access to a 0 dBr point or a system simulator, simulating the radio link to the terminal under controlled and error free conditions is required. The system simulator or terminal simulator shall be equipped with a high-quality codec whose characteristics are as close as possible to ideal.

Definition of 0 dBr level reference point:

D/A converter - a Digital Test Sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose rms value is 3,14 dB below the maximum full-load capacity of the codec shall generate 0 dBm across a 600 ohm load;

A/D converter - a 0 dBm signal generated from a 600 ohm source shall give the Digital Test Sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose RMS value is 3,14 dB below the maximum full-load capacity of the codec.

For testing a terminal a system simulator or terminal simulator shall use the ACELP speech codec as defined in EN 300 395 [3] series specifications. The transcoding from the output of the ACELP speech coding in the system simulator, or terminal simulator, to analogue signals shall be carried out using a transcoder to ITU-T Recommendation G.712 [8] (4-wire analogue) or using 16-bit uniform PCM with a PC soundcard having inputs and outputs calibrated to be 0 dBr points or by a transcoder to ITU-T Recommendation G.711 [7] (A-law PCM) for measurement to take place on a 30-channel E1 highway in the digital domain.

7.5.2.2 Direct digital processing approach

In this approach, the companded digital input/output bit-stream of the terminal connected through the radio link to the system simulator is operated upon directly. For the purposes of terminal acoustic testing, the direct digital processing shall use the default speech codec, namely the ACELP speech codec as defined in EN 300 395 [3] series specifications.

7.5.3 Accuracy of test equipment

Unless specified otherwise, the accuracy of measurements made by test equipment shall be better than defined in table 2.

Table 2: Required tolerances of test measurement equipment

Item	Accuracy
Electrical Signal Power	±0,2 dB for levels ≥ -50 dBm
Electrical Signal Power	±0,4 dB for levels < -50 dBm
Sound pressure	±0,7 dB
Time	±5 %
Frequency	±0,2 %

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Table 3: Required tolerances of test signals for measurements

Quantity	Accuracy
Sound pressure level at MRP	±1 dB for 200 Hz to 4 kHz ±3 dB for 100 Hz to 200 Hz and 4 kHz to 8 kHz
Electrical excitation levels	±0,4 dB (see note 1)
Frequency generation	±2 % (see note 2)
NOTE 1: Across the whole frequency range. NOTE 2: When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ±2 % on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where only the -2 % tolerance may be used.	

The measurements results shall be corrected for the measured deviations from the nominal level.

The sound level measurement equipment shall conform to IEC 61672-1 [24] type 1.

7.5.4 Test signals

Due to the coding of the speech signals, standard sinusoidal test signals are not applicable for terminal acoustic tests, appropriate test signals are defined in ITU-T Recommendation P.50 [10]. The test signal levels are referred to the average level of the test signal, averaged over the complete test sequence, unless specified otherwise.

NOTE: In case of testing with other types of speech signals such as real speech that may contain speech pauses, the sound level is referred to the active speech level of the test signal, measured according to ITU-T Recommendation P.56 [12] method B over the complete test sequence.

7.6 Test conditions

7.6.1 Environmental conditions

7.6.1.1 Handheld, handset, headset and shoulder mounted terminals

For handset terminals, handheld terminals, shoulder mounted terminals, headset terminals and terminals with other types of user worn acoustic interfaces the test room shall be practically free-field down to a lowest frequency of 300 Hz, the terminal microphone including the HATS shall lie totally within this free-field volume. This shall be met if deviations of the ideal free-field conditions are less than ±1 dB. Qualification of the test room may be performed using the method described in either ISO 3745 [23] annex A, or ITU-T Recommendation P.340 [17] clause 5.4.

Alternatively, a test room may be used which meets the following criterion:

- The relationship between the pressure at the MRP and that at 50 mm, 100 mm and 140 mm in front of the centre of the lip is within ±0,5 dB of that which exists in a known acoustic free-field (see ITU-T Recommendation P.58 [14]). If relevant for the set-up tested, the sound pressure at 500 mm in front of the lip shall be within ±0,5 dB of that which exists in a known acoustic free-field.
- The ambient noise level shall be less than -64 dBPa(A). However, this limit may be relaxed if it can be shown that the accuracy of the measurement is not impaired. The test report shall state the ambient noise level in dBPa(A) of the test environment.

7.6.1.2 Desktop terminals and vehicle terminals

Desktop terminals and vehicle terminals generally should be tested in their typical environment of application. Care must be taken, that e.g. noise levels are sufficiently low in order not to interfere with the measurements.

For desktop terminals the appropriate requirements shall be taken from ITU-T Recommendation P.340 [17].

The broadband noise level shall not exceed -70 dBPa(A). The octave band noise level shall not exceed the values specified in table 4.

Table 4: P.340 Noise level

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

7.6.2 System Simulator conditions

The system simulator should provide an error free radio connection to the terminal under test. The default speech codec, the ACELP speech codec shall be used. Discontinuous Transmission (DTX) (silence suppression) shall be disabled for the purposes of MS acoustic testing.

7.7 Sending level performance test methods

7.7.1 Applicability

The test methods of this clause 7.7 shall apply when testing sending level performance of TETRA terminals in simplex calls.

7.7.2 Sending level performance

7.7.2.1 General

Terminals without send AGC function support only the nominal use condition, whereas terminals with send AGC function support a range of use conditions including the nominal use condition. For both types of terminals the sending level performance shall be checked for the nominal use condition.

For terminals with send AGC function the sending level performance for the two extremes of the range of use conditions supported and should also be checked:

- The combination of maximum distance from the centre of the lip plane to the microphone and minimum sound pressure in the MRP producing the lowest sound pressure at the microphone.
- The combination of minimum distance from the centre of the lip plane to the microphone and maximum sound pressure in the MRP producing the highest sound pressure at the microphone.

7.7.2.2 Terminals without send AGC function

a) The test signal to be used for the measurements shall be 10 seconds of male artificial voice followed by 10 seconds of female artificial voice according to ITU-T Recommendation P.50 [10] with no pause between the two speech segments. The spectrum of the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be that specified by the manufacturer for the nominal use condition. The test signal level shall be measured as the wideband sound pressure level at the MRP.

b) Handset terminals and headset terminals are set-up according to clause 7.5.1.

   Handheld terminals, shoulder mounted terminals and terminals with other types of user worn acoustic interfaces are set-up in accordance with the manufacturer's specification. The distance between the centre of the lip plane of the artificial mouth and the microphone shall be according to the nominal use condition specified by the manufacturer.

   Vehicle mounted terminals and desktop terminals are set-up according to clause 7.5.1. The distance between the centre of the lip plane of the artificial mouth and the microphone shall be according to the nominal use condition specified by the manufacturer.

c) The test signal is applied to the terminal and the sending level measured for the entire duration of the test signal.

   The sending level in terms of dBm0 is measured at the 0 dBr point of the SS as the active speech level according to ITU-T Recommendation P.56 [12] method B.

   The test report shall state the nominal use conditions tested and the set-up used.

7.7.2.3 Terminals with send AGC function

a) The test signal to be used for the measurements shall be 10 seconds of male or female artificial voice according to ITU-T Recommendation P.50 [10]. The spectrum of the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be that specified by the manufacturer for the use condition under test. The test signal level shall be measured as the wideband sound pressure level at the MRP.

b) Handset terminals and headset terminals are set-up according to clause 7.5.1.

   Handheld terminals, shoulder mounted terminals and terminals with other types of user worn acoustic interfaces are set-up in accordance with the manufacturer's specification. The distance between the centre of the lip plane of the artificial mouth and the microphone shall be that specified by the manufacturer for the use condition under test.

   Vehicle mounted terminals and desktop terminals are set-up according to clause 7.5.1. The distance between the centre of the lip plane of the artificial mouth and the microphone shall be that specified by the manufacturer for the use condition under test.

c) The test signal is applied once to the terminal under test to allow the AGC to adapt the gain.

d) The test signal is again applied to the terminal and the sending level measured for the entire duration of the test signal.

The sending level in terms of dBm0 is measured at the 0 dBr point of the SS as the active speech level according to ITU-T Recommendation P.56 [12] method B.

The test report shall state the use conditions tested and the set-up used.

Annex A (informative): Subscriber A, connection establishment time

A.1 Scope

This annex defines the measurements and metrics applicable to TETRA networks to enable the effects associated with connection establishment time experienced by subscriber A to be manageable. Connection establishment time is an inherent quantity within a network and is experienced by subscribers when invoking connections to be conveyed by the network. The parameter's magnitude may be considered as a function of the operation of elements involved to form the connection, network load, network signalling and the processing required to realize a connection.

NOTE: Clause B.4 discusses interaction between subscriber A and subscriber B connections times.

A.2 Subscriber A, connection establishment time

Connection establishment time: The time span from the instant of the "last key press" from a subscriber's apparatus required to initiate a connection to the instant when a valid supervisory signal or connection confirmation, as appropriate, is returned by the network to that subscriber.

NOTE 1: A valid supervisory signal or connection confirmation may depend upon the type of call being set-up and conditions. For example: For hook signalling, the supervisory signal is the receipt of a "D-Alert" message. Normally a reception of "D-Call Proceeding" message should not be considered as a valid supervisory signal.

NOTE 2: Connection establishment time may be dependent upon the type of call. For Pre-emptive Priority Calls, SwMI actions may be used to minimize call establishment time.

NOTE 3: The connection establishment time as defined in this clause is the same as the time to the through connection as used in EN 300 392-2 [2], clause 14, only in specific situations.

The "end-to-end", Intermediate Monitoring Points (IMPs) and the time model as defined in figure 1, including their supportive clauses, are used in this annex.

A.3 Observation and reporting of connection establishment time

A.3.1 End-to-end connections between subscribers on the subscriber A point of view

To convey connection establishment information to the TETRA network from the calling subscriber A for a called subscriber B, signalling stimuli and response messages are involved. Connection establishment time may be determined for a connection invoked by subscriber A through monitoring the time difference between the instant a valid connection establishment stimuli message is issued (time $\mathbf{t_a}$) and the instant when a valid and appropriate connection establishment response message is received (time $\mathbf{t_c}$ or $\mathbf{t_h}$ as appropriate) at the same monitoring point detailed in figure 1 and table 1.

NOTE 1: Typically this measurement incorporates terminal apparatus (I_T) and may incorporate subscriber's equipment (I_R) processing times and the results are affected by implementation choices.

NOTE 2: End-to-end in this context means that the called subscriber's terminal apparatus has responded to the call set-up and that response has reached back to the calling subscriber.

NOTE 3: The end-to-end connection establishment time on both subscriber A and subscriber B point of view is further discussed in clause B.4.

A.3.2 Intermediate monitoring point in the direction towards subscriber B

Connection establishment time may be determined for a connection at an Intermediate Monitoring Point (IMP) in the direction towards Subscriber B through monitoring the time difference between the instant a valid connection establishment stimuli message is observed (time $\mathbf{t_a}$ at IMP) and the instant when a valid and appropriate connection establishment response message is recognized (time $\mathbf{t_c}$ at the same IMP) as illustrated in figure 1.

NOTE: This measurement method can remove effects of the subscriber equipment and terminal apparatus especially when the measurement is performed by special measurement equipment, for example, at the I_U reference point.

A.3.3 Measurements in-between intermediate monitoring points

Connection establishment time may be determined for a connection in-between Intermediate Monitoring Points (IMPs) by monitoring the time difference between the instant a valid connection establishment stimuli message is observed (time $\mathbf{t_a}$ at an IMP) and the instant when a valid and appropriate connection establishment response message is recognized (time $\mathbf{t_b}$ or $\mathbf{t_c}$ at another IMP) as illustrated in figure 1.

A.3.4 Examples of measurements

Example A, at a single IMP: where TETRA signalling, or similar, may be accessed referred to a "time-stamp", the connection establishment time at a single IMP may be calculated:

- **for voice or circuit mode data connections using hook signalling:** the time difference between the time instant when a "Set-up" message is detected and the time instant when an "Alerting" message is recognized, in this scenario the time instance are t_a and t_c;

- **for voice or circuit mode data connections using direct set-up signalling:** the time difference between time instant when a "Set-Up" message is detected and the time instant when a "Connect" message is recognized, in this scenario the time instances are t_a and t_h (direct call-setup response from subscriber B is generated by the MS not by the actual user).

NOTE 1: In some networks full duplex voice connections always use hook signalling and half duplex voice calls (group calls) use direct set-up signalling. That linkage is outside EN 300 392-2 [2], which supports any combination of hook/direct signalling and half duplex/full duplex voice calls for individual calls.

Example B, between IMPs: where TETRA signalling, or similar, may be accessed referred to a "time-stamp", the connection establishment time in-between two IMPs may be calculated:

- **for voice or circuit mode data connections for "a call originated from subscriber A":** the time difference between the time instant when a "Set-Up" message is detected at an IMP close to subscriber A and the time instant when the "Set-Up" message is recognized at a distant IMP, in this scenario the time instances are typically t_b and t_d;

- **for voice or circuit mode data connections using hook signalling for "a call received by subscriber B":** the time difference between the time instant when an "Alerting" message is detected at the distant IMP and the time instant when the "Alerting" message is recognized at the IMP close to the subscriber A, in this scenario the time instances are typically t_e and t_h;

NOTE 2: This scenario assumes that the SwMI implementation generates response towards subscriber A from a "U-Alert" message generated by the subscriber B MS.

- **for voice or circuit mode data connections using direct set-up signalling referred to "a call received by subscriber B":** the time difference between the time instant when a "Connect" message is detected at the distant IMP and the time instant when the "Connect" message is recognized at the IMP close to subscriber A, in this scenario the time instances are typically t_e and t_h.

NOTE 3: The above measurements are a division of example A scenarios into two time durations.

NOTE 4: Although these examples do involve some subscriber B (or subscriber B subscriber apparatus) actions the measurements are intended for subscriber A grade of service measurements. Refer to annex B for subscriber B related measurements.

NOTE 5: For actual measurements exact scenarios indicating which actual messages are used may be needed in order to get comparable results.

Annex B (informative): Subscriber B, connection establishment time

B.1 Scope

This annex defines the measurements and metrics applicable to TETRA networks to enable the effects associated with connection establishment time for the called subscriber B to be manageable. Subscriber B connection establishment time is an inherent quantity within a network and is experienced following subscriber B answering a call invoked by subscriber A. The parameter's magnitude may be considered as a function of the operation of elements involved to form the connection, network load, network signalling and the processing required to realize a connection.

Clause B.4 discusses interaction between subscriber A and subscriber B connection times.

B.2 Subscriber B connection establishment time

Subscriber B connection establishment time: the time span from the instant of the "last key press" from subscriber B's apparatus required to answer a call invoked by subscriber A to the instant when a valid connection confirmation is returned by the network to subscriber B.

NOTE: For direct call set-up, the "last key press" is considered to be the sending of the "U-Connect" message by subscriber B apparatus.

B.3 Observation and reporting of subscriber B connection establishment time

B.3.1 Subscriber B connection establishment time

To convey a connection establishment confirmation to the TETRA network from called subscriber B (following a connection establishment request from subscriber A), signalling stimuli and response messages are involved. Subscriber B connection establishment time may be determined for a connection invoked by subscriber A by monitoring the time difference between the instant a valid connection establishment acceptance message is issued (time $\mathbf{t_e}$ subscriber B "last key press") compared to the instant when a valid and appropriate connection establishment confirmation message is received (time $\mathbf{t_g}$) at the Intermediate Monitoring Point detailed in figure 1. Subscriber B connection establishment time is then calculated by equation:

Subscriber B connection establishment time = **(time $\mathbf{t_g}$) - (time $\mathbf{t_e}$).**

B.3.2 Intermediate monitoring point in the direction towards subscriber A

Subscriber B connection establishment time may be determined at an Intermediate Monitoring Point (IMP) towards subscriber A by monitoring the time difference from the instant that a valid connection establishment confirmation message is issued by subscriber B (time $\mathbf{t_f}$) compared to the instant when a valid connection establishment confirmation message is recognized (time $\mathbf{t_g}$) at the same IMP as illustrated in figure 1 and table 1. The subscriber B connection establishment time at that IMP can then be calculated using equation:

Subscriber B connection establishment time = **(time $\mathbf{t_g}$) - (time $\mathbf{t_f}$).**

B.3.3 Measurements in-between intermediate monitoring points

Subscriber B connection establishment time may be determined for a connection in-between Intermediate Monitoring Points (IMPs) in the direction towards subscriber A by monitoring the time difference between the instant a valid connection establishment confirmation message is observed (e.g. time $\mathbf{t_f}$ $\mathbf{t_b}$) at one IMP compared to the instant when a valid and appropriate connection establishment confirmation message is recognized (e.g. time $\mathbf{t_g}$) at another IMP illustrated in figure 1. The subscriber B connection establishment time at the in-between IMPs can then be calculated using equation:

Subscriber B connection establishment time **= (time $\mathbf{t_g}$) - (time $\mathbf{t_f}$).**

NOTE: This scenario is different than the one in clause B.3.2 although the same time instance name is used, refer to note in table 1.

B.3.4 Examples of measurements

Example A, at subscriber B: where TETRA signalling, or similar, may be accessed referred to a "time-stamp", the subscriber B connection establishment time may be calculated as the time difference between the instant subscriber B issues the last key press to answer the incoming call and the time instant when a "Connect Acknowledgement" message is recognized at subscriber B's apparatus, refer to clause B.3.1.

Example B, between subscriber B and an IMP in the direction towards subscriber A: where TETRA signalling, or similar, may be accessed referred to a "time-stamp", the subscriber B connection establishment time at an IMP may be calculated as the time difference between the time instant of the subscriber B issued "Connect" message to the incoming call and the time instant when the "Connect Acknowledgement" message is recognized at an IMP in the direction towards subscriber A, in this scenario time instances are typically t_e and t_g.

Example C, between two IMPs: where TETRA signalling, or similar, may be accessed referred to a "time-stamp", the subscriber B connection establishment time at two IMPs may be calculated as the time difference between the time instant of the subscriber B issued "Connect" message to the incoming call is detected at an IMP in the direction towards subscriber A and the time instant when a "Connect Acknowledgement" message is recognized at another IMP in the direction towards subscriber B, in this scenario time instances are typically t_h and t_g.

B.4 Interaction between subscriber A and subscriber B connections times

Through connection establishment time: the time span from the instant of the "last key press" from subscriber A's apparatus to the instant when a voice or circuit mode data path is established between subscribers A and B.

Connection times defined in annex A and in clauses B.1 to B.3.4 interact in the calculation of the through connection time and there is no single equation for the calculation in a general scenario.

In a network, time instances $\mathbf{t_g}$ and $\mathbf{t_h}$ can be arranged so that the voice or circuit mode data path is available before the traffic sending party is permitted to start transmission. That may affect the user perception of the call set-up time e.g. in the cases:

- the called subscriber B receives a "Connect Acknowledgement" message, but the calling subscriber A side has a traffic channel queue;
- the calling subscriber A receives a "Connect" message, but the called subscriber B side has a traffic channel queue and the called subscriber B will get the first permission to transmit.

NOTE: Total call set-up time on the subscribers' point of view may be defined from t_a to t_g or t_h as appropriate. In this case t_g or t_h are instances of the reception of the transmit allocations.

Annex C (informative): Disconnecting user initiated connection release time

C.1 Scope

This annex defines the measurements and metrics applicable to TETRA networks to enable the effects associated with connection release time to be manageable. Connection release time is an inherent quantity within a network and is experienced by subscribers when releasing connections through that network. The parameter's magnitude may be considered as a function of the operation of elements involved, network load, network signalling and the processing required to realize the connection release. The time duration for a network to return to its dormant state following a subscriber's connection release request may influence the network's total call attempt capacity.

C.2 Disconnecting user initiated connection release time

Disconnecting user initiated connection release time: the time span from the instant of the "last key press" (from a subscriber's apparatus) to release a connection to the instant when a valid release confirmation is returned by the network to the subscriber.

Total connection and resource release comprises also release of the other subscriber or subscribers in the call and it may be useful to extend the measurement to cover also the other subscribers.

C.3 Observation and reporting of connection release time

C.3.1 End-to-end connection release between subscribers

For an established connection between subscribers, connection release information is conveyed by signalling stimuli and response messages to/from the TETRA Network following a subscriber's "last key press" to release the connection. Connection release time, experienced by the subscriber who releases the connection, may be determined through monitoring the time difference between the instant a valid "key press" (or equivalent) is made to release the connection (e.g. time $\mathbf{t_a}$ or time $\mathbf{t_e}$) to the instant when a valid and appropriate connection release message is received (e.g. time $\mathbf{t_c}$ or time $\mathbf{t_g}$) at the monitoring point detailed in figure 1 (e.g.: I_T or I_R as appropriate in figure 1).

The end-to-end connection release includes also release of the other than the disconnection initiating subscriber release and from the network point of view the disconnection time instance is when the last valid and appropriate connection release message is received (any of the times $\mathbf{t_c}$, $\mathbf{t_d}$, $\mathbf{t_g}$ or $\mathbf{t_h}$) at the monitoring point related to the last subscriber in the released call.

C.3.2 Subscriber connection release at intermediate monitoring point

Connection release time may be determined at an Intermediate Monitoring Point (IMP) for the subscriber who releases the connection by monitoring the time difference between the instant a valid connection release stimuli message is observed at an IMP (time $\mathbf{t_a}$ or $\mathbf{t_e}$ as appropriate) and the instant when a valid and appropriate connection release response message is recognized at the same IMP (time $\mathbf{t_a}$ or $\mathbf{t_g}$ as appropriate) as illustrated in figure 1 (e.g.: I_U). The connection release request may be issued by either the "near" or "far" end subscribers.

Also in this case measurements of disconnection of all subscribers in the call may be appropriate, refer to clause C.3.1.

C.3.3 Measurements in-between intermediate monitoring points

Connection release time may be determined for a connection in-between Intermediate Monitoring Points (IMPs) by monitoring the time difference between the instant a valid connection release stimuli message is observed at an IMP (e.g. time $\mathbf{t_b}$) and the instant when a valid and appropriate connection release response message is recognized at another IMP (time $\mathbf{t_c}$) as illustrated in figure 1.

C.3.4 Examples of measurements

Example A, from a single IMP from the disconnecting subscriber point of view: where a TETRA, or similar, "disconnect" message may be accessed referred to a "time-stamp", the connection release time at one IMP may be calculated:

- **For voice and circuit mode data connections:** the time difference between the time instant when a "disconnect" message is detected compared to the time instant when a "release" message to the disconnecting subscriber is recognized, in this scenario time instances are e.g. t_a and t_c.

Example B, between IMPs: where a TETRA, or similar, "release" message may be accessed referred to a "time-stamp", the connection release time in-between IMPs may be calculated:

- **For voice and circuit mode data connections:** the time difference between the time instant when a user "release" message is detected at an IMP compared to the time instant when the "release" message is recognized at another IMP, in this scenario time instances are e.g. t_b and t_c.

Example C, total call release: where a TETRA, or similar, "release" message may be accessed referred to a "time-stamp", the connection release time at one or more IMPs may be calculated:

- **For voice and circuit mode data connections:** the time difference between the time instant when a "release" message is detected at an IMP compared to the last time instance when the or "release" message is recognized for the last subscriber in that call at the same or another IMP, in this scenario time instances are e.g. t_b and t_d.

Annex D (informative): One-way time delay

D.1 Scope

This annex defines the measurements and metrics applicable to TETRA networks to enable the effects associated with one-way time delay to be manageable. One-way time delay is an inherent quantity within a network and may be experienced by subscribers when communicating over established end-to-end connections within a network. The parameter's magnitude may be considered as a function of the elements involved to form the established connection, network load and the processing required to realize an established end-to-end connection. Excessive one-way time delay may be perceived by a subscriber as the hesitancy in the far-end subscriber responding during conversation or the perception of talker echo signals. For circuit mode data connections an excessive delay may also affect, for example, the performance of re-transmission protocols.

NOTE: Actually the user experiences the sum of one-way delays in both directions.

D.2 One-way time delay

One-way time delay: One-way time delay is the time taken by a signal applied at the input of an equipment to reach the output of that equipment, where the equipment may be an end-to-end connection.

D.3 Observation and reporting of time delay

D.3.1 End-to-end connections between subscribers

To convey information (e.g.: voice or circuit mode data) between subscribers connected by a network a time delay is experienced between the signal presented (by the "transmitter") and the signal received (by the "receiver"). One-way time delay may be determined for a connection between subscriber A and subscriber B by monitoring the time difference between the instant a defined voice or circuit mode data signal stimuli is applied (figure 1, e.g. time $\mathbf{t_a}$) and the instant when the same voice or circuit mode data signal stimuli is recognized at the far-end (figure 1 and table 1, time $\mathbf{t_d}$). The reference points are considered to be at user access points such as I_R, I_S or I_T and E_R, E_S or E_T.

For circuit mode data connections a predefined data pattern may be used as the signal stimuli.

NOTE: In the scenarios of this annex the subscriber A and subscriber B identifiers are used although they may not refer to the calling and called user of the call but to the source and destination subscribers in the actual measurement.

D.3.2 Measurements in-between intermediate monitoring points

One-way time delay may be determined for a connection in-between Intermediate Monitoring Points (IMPs) by monitoring the time difference between the instant a voice or circuit mode data signal stimuli is observed (time $\mathbf{t_a}$) at an IMP and the instant when the same voice or circuit mode data signal stimuli is recognized at another IMP (time $\mathbf{t_b}$) as illustrated in figure 1 and table 1.

The IMPs may also be the user access points such as I_R, I_S and I_T and E_R, E_S or E_T for direct one-way time delay measurement, refer to clause D.3.1.

D.3.3 Two-way time delay measurement

Two-way time delay may be determined for a connection at a single Intermediate Monitoring Point (IMP) by observing the time difference between the instant (time $\mathbf{t_a}$) a defined signal stimuli in the direction towards the far end and the instant (time $\mathbf{t_h}$), when the defined signal stimuli in the direction from the far end is recognized at the same IMP as illustrated in figure 1 and table 1. The measurement requires a loop back connection at the far end e.g. at E_R, E_S or E_T.

The two-way time delay measurement is possible only when the connection is a full duplex circuit.

D.3.4 Examples of measurements

Example A, at user access point: measurement of two-way time delay for a duplex voice or circuit mode data connection may be performed on an established end-to-end connection by observing the difference in time between the instant when a defined voice or circuit mode data signal stimuli is applied at a monitoring point (e.g.: I_S in figure 1) and the time instant when a recognizable version of the applied voice or circuit mode data signal stimuli is detected at the same monitoring point (e.g.: I_S in figure 1). Arrangements are required at the far-end of the connection to ensure that the received voice or circuit mode data signal stimuli are relayed via a "loop-back" (e.g.: looped at E_s in figure 1). An estimate of the one-way time delay can be calculated by dividing the result by two.

Example B, at an IMP: monitoring two-way time delay of a full duplex call at an Intermediate Monitoring Point (IMP in figure 1) may be performed using the difference in time between the instant when a defined voice or circuit mode data signal stimuli is observed at an Intermediate Monitoring Point (e.g.: I_v in figure 1) and the instant when a recognizable version of the applied signal stimuli is observed at the same Intermediate Monitoring Point (e.g.: I_v in figure 1). Arrangements are required at the far-end of the connection to ensure that the received voice or circuit mode data signal stimuli are relayed via a "loop-back" (e.g.: looped at IMP E_s in figure 1). In this scenario time instances are e.g. t_a and t_h.

Example C, measurements between IMPs: when measurements can be referred to "time-stamps", then monitoring one-way time delay between Intermediate Monitoring Points (IMP in figure 1) may be performed using the difference in time between the instant when a defined voice or circuit mode data signal stimuli is observed at an Intermediate Monitoring Point (e.g.: I_v in figure 1) and the instant when a recognizable version of the applied signal stimuli is observed at another Intermediate Monitoring Point (e.g.: I_w in figure 1). In this scenario time instances are e.g. t_a and t_d.

Annex E (informative): Voice quality

E.1 Scope

This annex defines the measurements and metrics applicable to TETRA networks to enable the effects associated with voice quality to be manageable. Subscribers' perception of "end-to-end" voice quality is influenced by all the losses comprising the connection between the subscribers, including acoustic, physical, electrical and coding losses inherent within the connection. Voice quality performance is typically specified and measured between the Mouth Reference Point (MRP), Ear Reference Point (ERP) and an Intermediate Monitoring Point (IMP) in terms of loudness ratings and Mean Opinion Score (MOS).

NOTE 1: The mean opinion score measurements are outside the scope of the present document.

NOTE 2: The voice quality measurements in the present document are proposed methods and are for further study.

E.2 Voice quality

Voice quality: measurement which takes account of all the acoustic and electrical losses that comprise a connection between subscribers enabling a representative assessment of the performance of the connection to be reportable.

NOTE: This voice quality measurement assumes a perfect connection without digital transmission errors. The voice quality due to the transmission bit errors and lost speech frames is outside the scope of the present document.

The "end-to-end" and Intermediate Monitoring Points (IMPs) defined in figure 1, including supportive clauses, are referred to within this annex.

For mobile networks EN 300 903 [1] provides guidance related to definition of OLR, SLR and RLR.

Subscribers' perception of Voice Quality associated with speech "echo" is presented in annex F.

E.3 Observation and reporting of voice quality

E.3.1 End-to-end connections between subscribers

For an established connection between subscribers, voice quality measurements may be made in terms of the Overall Loudness Rating (OLR), from the near-end MRP to the distant ERP (i.e.: between I_S and E_S) as detailed in figure 1.

E.3.2 Intermediate monitoring point measurement

For an established connection between subscribers, voice quality measurements may be made, in terms of the Send or Receive Loudness Rating (SLR and RLR), from the near-end MRP or ERP to/from a distant Intermediate Monitoring Point (IMP) as appropriate.

E.3.3 Measurements in-between intermediate monitoring points

Voice quality, in terms of transmission loss (decibels, dB), may be determined for a connection in-between electrical Intermediate Monitoring Points (IMP) through monitoring the loss. As TETRA is a digital communication networks the loss measurement is meaningful when at least one of the reference points is an analogue signal reference point such as I_S, E_T or POI as illustrated in figure 1.

NOTE: Loss measurements where signal is converted into a digital signal using a redundancy removing voice codec as in TETRA require a suitable measurement signal.

E.4 Examples of measurements

The following examples are applicable to full duplex and half-duplex calls, but excluding simplex calls.

Example A, connections with terminal handsets.

Example B, connections with handsfree terminals.

Example C, connections with a terminal supporting a headset. The methods given in ITU-T Recommendation P.38 [9] may be used in connection with measurements of SLR and RLR for such terminals.

Example D, connections with terminals in "speaker" mode of operation.

E.5 Test configurations

This clause E.5 describes the test setups for terminal acoustic testing.

E.5.1 Test setup for terminals

The general access to terminals is described in figure E.1. The preferred acoustic access to terminal is the most realistic simulation of the "average" user. This shall be made by using HATS (head and torso simulator), with an appropriate ear simulation and appropriate mountings for handset terminals in a realistic but reproducible way. Hands-free terminals shall use the HATS or free field microphone techniques in a realistic but reproducible way. HATS is described in ITU-T Recommendation P.58 [14]. Appropriate ears are described in ITU-T Recommendation P.57 [13] (type 3.3 and type 3.4 ear). A proper positioning of handsets in realistic conditions is defined in ITU-T Recommendation P.64 [15], the test setups for various types of hands-free terminals can be found in ITU-T Recommendation P.581 [20]. The artificial ears are described in ITU-T Recommendation P.57 [13].

The preferred way of testing is the connection of a terminal to the system simulator with exact defined settings and access points. The test sequences are fed in either, electrically using a reference codec or using the direct signal processing approach or acoustically using ITU-T specified devices.

E.5.1.1 Setup for handset terminals

The handset is mounted at the HATS equipped with type 3.3 or 3.4 ear simulators. The handset is placed in the standardized position, HATS position, as described in ITU-T Recommendation P.64 [15].

For HATS with type 3.3 ear simulator the soft pinna (hardness 35 degrees ± 6 degrees Shore-OO) shall be used (see ITU-T Recommendation P.57 [13].) The handset shall be in the ERP position (as defined in ITU-T Recommendation P.64 [15].) The handset positioner barrel thus set to the ERP position defines the actual application force of the handset against the ear simulator required for that particular handset.

For HATS with type 3.4 ear simulator the application force of the handset against the ear simulator shall be in the range specified in ITU-T Recommendation P.64 [15] or ITU-T Recommendation P.57 [13].

The artificial mouth shall conform to ITU-T Recommendation P.58 [14]. The artificial ear shall conform to ITU-T Recommendation P.57 [13], (type 3.3 or type 3.4).

E.5.1.2 Setup for headset terminals

The headset is mounted at the HATS equipped with type 3.3 or type 3.4 ear simulators.

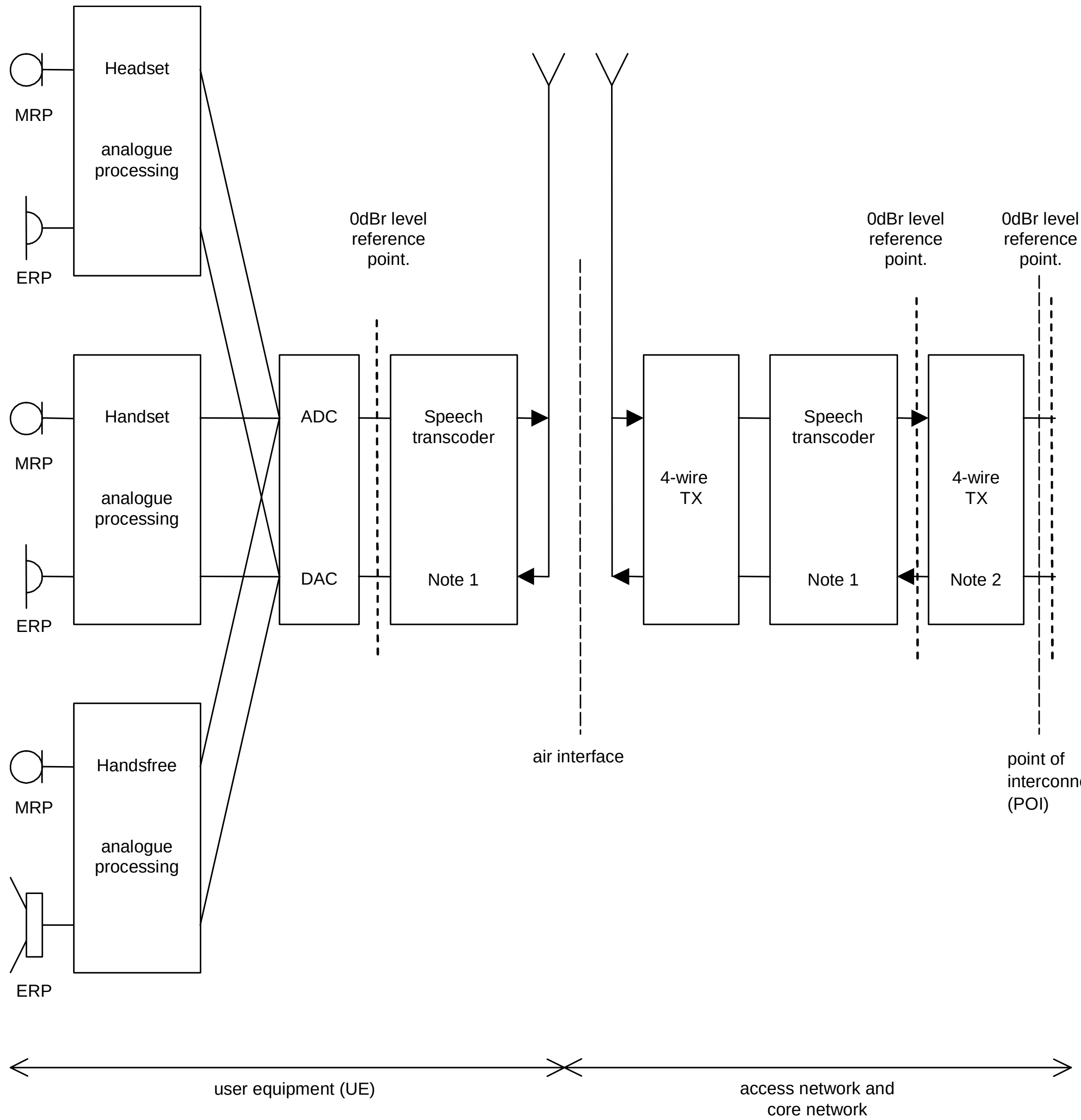
The position of the headset should be according to ITU-T Recommendation P.380 [18] and the force against the artificial ear shall be the same as applied in normal use.

For binaural headsets the HATS shall be equipped with a right ear simulator and a left ear simulator.

The type 3.3 ear simulator can be used for all types of headsets and earphones, whereas the type 3.4 ear simulator cannot be used with supra-concha headsets, supra-aural headsets and forward facing intra-concha headsets (see ITU-T Recommendation P.57 [13])

For HATS with type 3.3 ear simulator, the soft pinna (hardness 35 degrees ± 6 degrees Shore-OO) shall be used (see ITU-T Recommendation P.57 [13])

The artificial mouth shall conform to ITU-T Recommendation P.58 [14]. The artificial ear shall conform to ITU-T Recommendation P.57 [13] (type 3.3 or type 3.4)



NOTE 1: May include DTX functionality.
NOTE 2: Connection to PSTN should include electrical echo control (EEC).

Figure E.1: Interfaces for specification and testing of terminal narrow-band acoustic characteristics

E.5.1.3 Setup for hands-free terminals

E.5.1.3.1 Vehicle mounted hands-free

Vehicle mounted hands-free may be measured either in a vehicle or in an anechoic room. For both of these two types of test environments, the setup will depend on whether HATS or a discrete artificial mouth and discrete microphone are used as the acoustic test equipment.

For in-vehicle measurements, if HATS test equipment is used, it should be positioned in the car as per ITU-T Recommendation P.581 [20]. If in-vehicle measurements are made with a discrete microphone and discrete artificial mouth, they should be positioned in the car as per figure E.2 and figure E.3, respectively. The artificial mouth should comply with ITU-T Recommendation P.51 [11]. The microphone should be a pressure-field microphone complying with IEC 61672-1 [24]. The microphone should preferably be fitted with a random incidence corrector. A vehicle simulator may be used instead of an actual car. A standard vehicle simulator is described in TR 101 110 Digital Cellular Telecommunications System (Phase 2+) Characterization test methods and quality assessment for hands-free mobiles (see bibliography).

The hands-free equipment is mounted in the car as specified by the manufacturer.

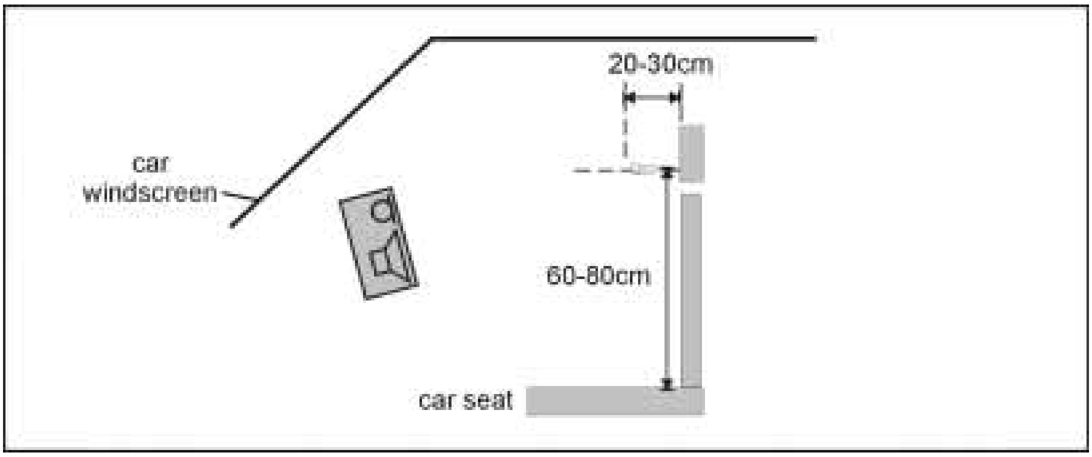


Figure E.2: Test Configuration for Vehicle mounted hands-free, receiving characteristics, with discrete measurement microphone

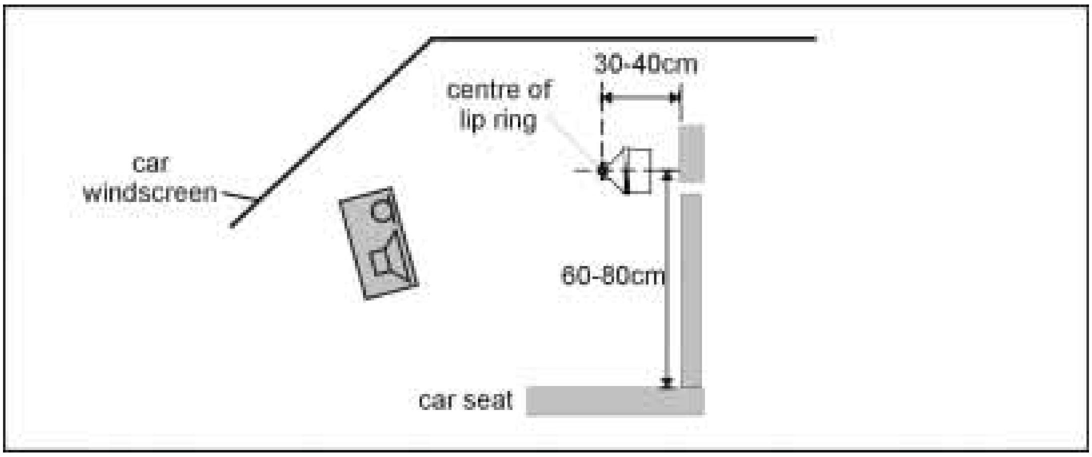


Figure E.3: Test Configuration for Vehicle mounted hands-free, sending characteristics, with discrete P. 51 artificial mouth

Specification testing of vehicle-mounted hands-free equipment in an anechoic room is for further study.

E.5.1.3.2 Handheld hands-free

Either HATS or a free-field microphone with a discrete ITU-T Recommendation P.51 [11] artificial mouth may be used to measure Hand-Held Hands-free type MS.

If HATS measurement equipment is used, it should be configured to the Hand-Held Hands-free MS according to figure E.4. The HATS should be positioned so that the HATS Reference Point is at a distance d_{HF} from the centre point of the visual display of the Mobile Station. The distance d_{HF} is specified by the manufacturer. A vertical angle θ_{HF} may be specified by the manufacturer.

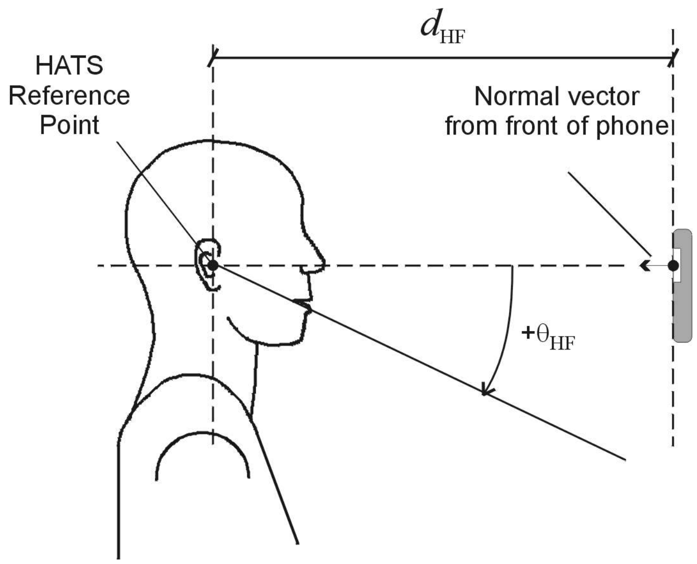


Figure E.4: Configuration of Hand-Held Hands-free MS relative to the HATS

If a free-field microphone with a discrete ITU-T Recommendation P.51 [11] mouth are used, they should be configured to the Hand-Held Hands-free MS as per figure E.5 for receiving measurements and figure E.6 for sending measurements. The measurement instrument should be located at a distance d_{HF} from the centre of the visual display of the Mobile Station. The distance d_{HF} is specified by the manufacturer.

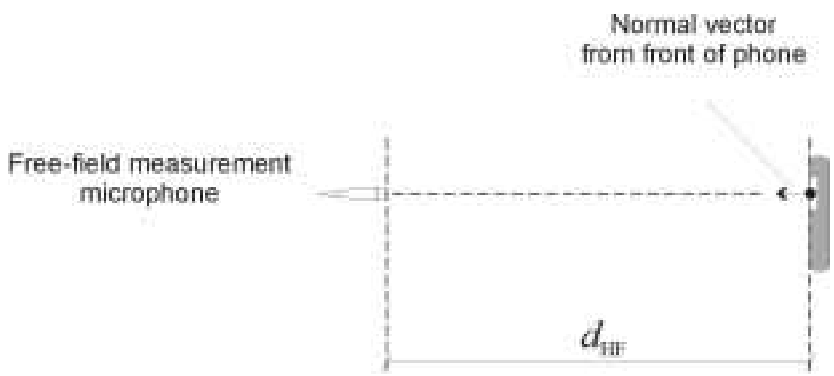


Figure E.5: Configuration of Hand-Held Hands-free MS, free-field microphone for receiving measurements

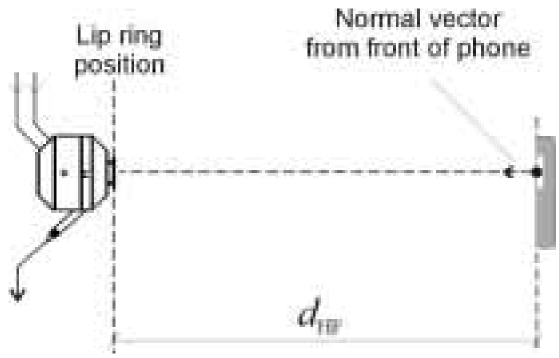


Figure E.6: Configuration of Hand-Held Hands-free MS, discrete P. 51 artificial mouth for sending measurements

E.5.1.3.3 Desktop operated hands-free

For HATS test equipment, definition of hands-free terminals and setup for desktop hands-free terminals can be found in ITU-T Recommendation P.581 [20]. Measurement setup using a free field microphone and a discrete ITU-T Recommendation P.51 [11] artificial mouth for desktop hands-free terminals can be found in ITU-T Recommendation P.340 [17].

E.5.1.4 Position and calibration of HATS

The horizontal positioning of the HATS reference plane shall be guaranteed within ±2 degrees for testing hands-free equipment.

The HATS shall be equipped with either Type 3.3 or 3.4 Artificial Ear. For hands-free measurements the HATS shall always be equipped with two artificial ears. The pinnas are specified in ITU-T Recommendation P.57 [13] for types 3.3 and 3.4 artificial ears. The pinna shall be positioned on HATS according to ITU-T Recommendation P.58 [14].

The exact calibration and equalization procedures as well as the combination of the two ear signals for the purpose of measurements can be found in ITU-T Recommendation P.581 [20]. For Handheld hands-free MS, the set-up corresponding to "portable hands-free" in ITU-T Recommendation P.581 [20] should be used.

E.5.1.5 Shoulder-mounted hands-free operation

Users are using this method of operation with either a remote audio accessory to the MS that extends its audio interfaces, such as a remote speaker microphone, or by wearing the MS on the shoulder.

The positioning of the MS or remote audio accessory on the shoulder of the HATS is for further definition.

E.5.2 Setup of the electrical interfaces

E.5.2.1 Codec approach and specification

Codec approach: In this approach, a codec is used to convert the companded digital input/output bit-stream of the system simulator to the equivalent analogue values. With this approach either a system providing access to a 0 dBr point or a system simulator, simulating the radio link to the terminal under controlled and error free conditions is required. The system simulator has to be equipped with a high-quality codec whose characteristics are as close as possible to ideal.

Definition of 0 dBr level reference point:

D/A converter: a Digital Test Sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose rms value is 3,14 dB below the maximum full-load capacity of the codec shall generate 0 dBm across a 600 ohm load;

A/D converter: a 0 dBm signal generated from a 600 ohm source shall give the digital test sequence (DTS) representing the codec equivalent of an analogue sinusoidal signal whose RMS value is 3,14 dB below the maximum full-load capacity of the codec.

Narrow band telephony testing

For testing a terminal supporting narrow-band telephony, the system simulator shall use the ACELP speech codec as defined in EN 300 395 [3] series specifications. The transcoding from the output of the ACELP speech coding in the system simulator, or reference transcoder of the system, to analogue signals shall be carried out using an transcoder to ITU-T Recommendation G.712 [8] (4-wire analogue) or using 16-bit uniform PCM with a PC soundcard having inputs and outputs calibrated to be 0 dBr points or by a transcoder to ITU-T Recommendation G.711 [7] (A-law PCM) for measurement to take place on an E1 highway in the digital domain.

E.5.2.2 Direct digital processing approach

In this approach, the companded digital input/output bit-stream of the terminal connected through the radio link to the system simulator is operated upon directly. For the purposes of terminal acoustic testing, the direct digital processing shall use the default speech codec, the ACELP speech codec as defined in EN 300 395 [3] series specifications.

Narrow band telephony testing

For testing a terminal supporting narrow-band telephony, the system simulator shall use the ACELP speech codec as defined in EN 300 395 [3] series specifications.

E.5.3 Accuracy of test equipment

Unless specified otherwise, the accuracy of measurements made by test equipment shall be better than defined in table E.1.

Table E.1: Measurement equipment accuracy

Item	Accuracy
Electrical Signal Power	±0,2 dB for levels ≥ -50 dBm
Electrical Signal Power	±0,4 dB for levels < -50 dBm
Sound pressure	±0,7 dB
Time	±5 %
Frequency	±0,2 %

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than defined in table E.2.

Table E.2: Signal accuracy

Quantity	Accuracy
Sound pressure level at MRP	±1 dB for 200 Hz to 4 kHz ±3 dB for 100 Hz to 200 Hz and 4 kHz to 8 kHz
Electrical excitation levels	±0,4 dB (see note 1)
Frequency generation	±2 % (see note 2)
NOTE 1: Across the whole frequency range. NOTE 2: When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ±2 % on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where only the -2 % tolerance may be used.	

The measurements results shall be corrected for the measured deviations from the nominal level.

The sound level measurement equipment shall conform to IEC 61672-1 [24] type 1.

E.5.4 Test signals

Due to the coding of the speech signals, standard sinusoidal test signals are not applicable for terminal acoustic tests, appropriate test signals (general description) are defined in ITU-T Recommendation P.50 [10] and P.501 [19]. Normative requirements for the use of test signals from ITU-T Recommendation P.501 [19] are for further study. More information can be found in the test procedures described below.

NOTE: As stated in clause E.5.2 for narrow-band telephony the ACELP speech codec shall be used.

For testing the narrow-band telephony service provided by a terminal the test signal used shall be band limited between 100 Hz and 4 kHz with a bandpass filter providing a minimum of 24 dB/octave. filter roll off, when feeding into the receiving direction.

The test signal levels are referred to the average level of the (band limited in receiving direction) test signal, averaged over the complete test sequence, unless specified otherwise.

E.6 Test conditions

E.6.1 Environmental conditions

E.6.1.1 Handset and headset terminals

The environmental conditions for testing handset and headset terminal is specified in clause E.6.1.1 as follows:

For handset and headset measurements the test room shall be practically free-field down to a lowest frequency of 300 Hz, the handset or the headset including the HATS shall lie totally within this free-field volume. This shall be met if deviations of the ideal free-field conditions are less than ±1 dB. Qualification of the test room may be performed using the method described in either ISO 3745 [23] annex A, or ITU-T Recommendation P.340 [17] clause 5.4.

Alternatively, a test room may be used which meets the following two criteria:

1) The relationship between the pressure at the mouth opening and that at 50 mm, 75 mm and 100 mm in front of the centre of the lip ring is within ±0,5 dB of that which exists in a known acoustic free-field.

2) The relationship between the pressure at the mouth opening and at the Ear canal Entrance Point (EEP) at both the left and right ears of the HATS does not differ by more than ±1 dB from that which exists in a known free-field.

The ambient noise level shall be less than -30 dBPa(A), for idle channel noise measurements the ambient noise level shall be less than -64 dBPa(A).

Echo measurements shall be conducted in realistic rooms with an ambient noise level less then -64 dBPa(A).

E.6.1.2 Hands-free terminals

Hands-free terminals generally should be tested in their typical environment of application. Care must be taken, that e.g. noise levels are sufficiently low in order not to interfere with the measurements.

For Desk-Top hands-free terminals the appropriate requirements shall be taken from ITU-T Recommendation P.340 [17].

The broadband noise level shall not exceed -70 dBPa(A). The octave band noise level shall not exceed the values specified in table E.3.

Table E.3: P.340 Noise level

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

Echo measurements shall be conducted in realistic rooms with an ambient noise level less then -70 dBPa(A).

E.6.2 System Simulator conditions

The system simulator should provide an error free radio connection to the MS under test. The default speech codec, the ACELP speech codec shall be used. Discontinuous Transmission (DTX) (silence suppression) shall be disabled for the purposes of MS acoustic testing.

E.7 Telephony transmission performance test methods

E.7.1 Applicability

The test methods in clause E.7 shall apply when testing a terminal which is used to provide narrow-band telephony.

The test methods for SLR and RLR are also applicable when a terminal is used in a full duplex call that is wholly within a TETRA network and does not use a POI or Gateway.

E.7.2 Overall loss/loudness ratings

E.7.2.1 General

The SLR and RLR values for the TETRA network apply up to the POI. However, the main determining factors are the characteristics of the terminal, including the analogue to digital conversion (ADC) and digital to analogue conversion (DAC). In practice, it is convenient to specify loudness ratings to the Air Interface. For the normal case, where the TETRA SwMI introduces no additional loss between the Air Interface and the POI, the loudness ratings to the PSTN boundary (POI) will be the same as the loudness ratings measured at the Air Interface.

The 0 dBr points of the TETRA network are shown in figure E.1.

These define that the circuit loss of a TETRA SwMI is 0 dB.

The overall loudness rating of full duplex connection in a TETRA network:

$$OLR = SLR + Circuit\ Loss + RLR$$

E.7.2.2 Connections with handset or headset terminal in full duplex calls

E.7.2.2.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is setup as described in clause E.5.1.

The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [16], equation (A-23b), over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 [16], table 1.

E.7.2.2.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The handset terminal is setup as described in clause E.5.1.

Handsets and monaural headsets: The receiving sensitivity referring to the ERP (see ITU-T Recommendation P.57 [13] table 2a for DRP to ERP correction) shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17. For the calculation, the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Binaural headsets: The output signal referred to the ERP (see ITU-T Recommendation P.57 [13] table 2a for DRP to ERP correction) of each artificial ear is power-averaged on the total time of analysis; the "right" and "left" signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave frequencies are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16] bands 4 to 17.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], equation (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [16].

d) No leakage correction shall be applied.

E.7.2.3 Connections with vehicle mounted and desk-top hands-free terminal

Vehicle mounted hands-free should be tested in the vehicle (for the totally integrated vehicle hands-free systems) or in a vehicle simulator, refer to TR 101 110 Digital Cellular Telecommunications System (Phase 2+) Characterization test methods and quality assessment for hands-free mobiles (see bibliography).

Free Field measurements for vehicle mounted hands-free are for further study.

E.7.2.3.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P. 50 [10]. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to -28,7 dBPa at the HFRP or the HATSHFRP (as defined in ITU-T Recommendation P.581 [20]) and the spectrum is not altered.

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

b) The hands-free terminal is setup as described in clause E 5. The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [16], equation (A-23b), over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 [16], table 1.

E.7.2.3.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in clause E.5. If HATS is used then it is freefield equalized as described in ITU-T Recommendation P.581 [20]. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the "right" and "left" signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17.

   For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], equation (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [16].

d) No leakage correction shall be applied. The hands-free correction as described in ITU-T Recommendation P.340 [17] shall be applied. To compute Receiving Loudness Rating (RLR) for hands-free terminal (see ITU-T Recommendation P.340 [17]), when using the combination of left and right ear signals from HATS the HFL_E has to be 8 dB, instead of 14 dB. For further information see ITU-T Recommendation P.581 [20].

E.7.2.4 Connections with handheld hands-free MS

E.7.2.4.1 Sending Loudness Rating (SLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P. 50 [10]. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to -28,7 dBPa at the HFRP or the HATSHFRP (as defined in ITU-T Recommendation P. 581 [20]) and the spectrum is not altered.

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

b) The hands-free MS is setup as described in clause E.5.1.3.2. The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [16], equation (A-23b), over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 [16], table 1.

E.7.2.4.2 Receiving Loudness Rating (RLR)

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free MS is setup as described in clause E.5.1.3.2. If HATS is used then it is freefield equalized as described in ITU-T Recommendation P.581 [20]. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the "right" and "left" signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17.

   For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [16], equation (A-23c), over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [16].

d) No leakage correction shall be applied. The hands-free correction as described in ITU-T Recommendation P.340 [17] shall be applied. To compute the Receiving Loudness Rating (RLR) for hands-free terminals (see ITU-T Recommendation P.340 [17]) when using the combination of left and right ear signals from HATS the HFL_E has to be 8 dB, instead of 14 dB. For further information see ITU-T Recommendation P.581 [20].

E.7.2.5 Connections with half-duplex terminal having send AGC function

Half duplex terminals or terminals with acoustic interface types which are capable only of half duplex operation in full duplex calls shall apply the SLR appropriate for the type acoustic interface or have sending level controlled by an AGC function.

The present clause applies solely for half duplex terminals with send AGC function.

Half duplex terminals with send AGC function may support a broader range of use conditions than do half duplex terminals that apply SLR for control of the sending level. The manufacturer or the half duplex terminals with send AGC shall specify the nominal use condition.

For Handheld hands-free terminals, the nominal use condition is characterized by the nominal sound pressure of speech in the MRP and the nominal position given by the distance d_{HF} according to figure E.4. A vertical angle θ_{HF} should be specified by the manufacturer.

For other types of terminal, the nominal use condition is characterized by the nominal sound pressure of speech in the MRP and the distance from the centre of the lip plane to the microphone. The manufacturer shall specify the use conditions of the terminal so that the relevant test set-ups from clause E.5.1 to apply can be selected.

The manufacturer should also specify the two extremes of the range of use conditions supported, and defined as the combination of maximum distance and minimum sound pressure in the MRP and the combination of minimum distance and maximum sound pressure in the MRP producing respectively the lowest and the highest sound pressure at the microphone.

E.7.2.5.1 Sending level for half duplex terminals with send AGC function

a) The test signal to be used for the measurements shall be 10 s of male or female artificial voice according to ITU-T Recommendation P.50 [10]. The spectrum of the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be that specified by the manufacturer for the use condition under test, e.g. the nominal use condition. The test signal shall be measured as the wideband sound pressure level at the MRP.

b) A Handheld hands-free MS is set-up as described in clause E.5.1.3.2 and in accordance with the manufacturer's specification of the use condition under test. The microphone of the handheld hands-free terminal shall be located in a point of free field condition. Other types of terminal are set-up in accordance with the manufacturer's specification of the use condition under test and the applicable set-up of clause E.5.1.

c) The test stimulus is applied once to the terminal under test to allow the AGC to adapt the gain.

d) The test stimulus is again applied to the terminal and the sending level measured for the entire duration of the stimulus signal.

   The sending level in terms of dBm0 is measured at the 0 dBr point of the SS as the active speech level according to ITU-T Recommendation P.56 [12] method B.

   The test report shall state and specify the use condition tested and the set-up used.

Annex F (informative): Echo performance

F.1 Scope

This annex defines the measurements and metrics applicable to TETRA networks to enable the effects associated with communication echo to be manageable. Listener and talker echoes are inherent quantities within a network and may be experienced by either subscriber A or subscriber B following the establishment of a connection. The parameter's magnitude may be considered as a function of the operation of elements involved to form the connection, including terminal apparatus.

This annex is only applicable to full duplex calls.

The measurements in the clauses F.4 to F.5 are proposed methods and are for further study.

NOTE: ITU-T recommendation G.131 has results of the tests of the user perception of connection quality as a function of echo and one-way delay.

F.2 Echo performance

Echo: the perception by the user of speech reverberation(s) within an established connection is referred to as echo. Talker echo refers to the perception by the talker of speech echo, while listener echo refers to the perception of echoes of the talker's speech by the listener.

The "end-to-end", Intermediate Monitoring Points (IMP) defined in figure 1, including their supportive clauses, are used in this annex.

F.3 Observation and reporting of echo performance

Guidance for echo performance observation and reporting is provided within EN 300 903 [1]. A TETRA network may be considered as a Public Land Mobile Network when referring to these documents.

For Terminal Coupling Loss (TCL), reference is made to EN 300 903 [1].

NOTE: The EN 300 903 [1] further refers to ITU-T Recommendations G.131 and G.165.

F.4 Measurement of Terminal Coupling Loss (TCL)

F.4.1 Acoustic echo control and TCL measurement in a handset or headset terminal

In the case of a handset terminal, the handset is suspended in free air in such a way that the inherent mechanical coupling of the handset is not affected. In the case of a headset terminal, the headset is mounted at the HATS, as described in clause E.5.

The testing shall be made under real use environmental conditions; a typical "office-type" room should be used. The ambient noise level shall be less than -64 dBPa(A). The attenuation from reference point input to reference point output shall be measured using the speech like test signal defined below.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [10] is altered.

Either a logarithmically spaced multi-sine or PN-sequence test signal shall be used.

The appropriateness of the multi-sine test signal for TETRA is for further study.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos(2\pi t \times f_{AM})\right] \times \cos(2\pi t \times f_{0i})\right]$$

where:

A = 0,5

f_{AM} = 4 Hz, μ_{AM} = 0,5

$f_{0\,i} = 250 \times 2^{(i/3)}$ Hz; i=1 to 11

CF = 14 dB ± 1 dB (10 dB + 4,26 dB due to 100 % AM modulation)

CF = Crest Factor = Peak to RMS ratio.

The training sequence level shall be -16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

NOTE: Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF = 3 dB. A test signal with a CF of maximum 15 dB can thus have a level of up to -8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

When using a PN-sequence, it should comply with ITU-T Recommendation P.501 [19] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. The test signal level is -3 dBm0. The low-crest factor is achieved by random-alternation of the phase between -180 degrees and +180 degrees.

The training sequence level shall be -16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

F.4.2 Acoustic echo control in a hands-free terminal

TCLw:

The hands-free is setup in a room where it is intended to be used, e.g. for an office type hands-free terminal a typical "office-type" room should be used; a vehicle-mounted hands-free terminal should be tested in a vehicle or vehicle simulator, as specified by the terminal manufacturer. [For reference on a suitable vehicle simulator see TR 101 110 "Digital Cellular Telecommunications System (Phase 2+)" (see bibliography). The ambient noise level shall be less than -70 dBPa(A). The attenuation from reference point input to reference point output shall be measured using a speech like test signal.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [10] is altered.

Either a logarithmically spaced multi-sine or PN-sequence test signal shall be used.

When using a logarithmically spaced multi-sine test signal, it is defined as:

$$s(t) = \sum_i \left[\left[A + \mu_{AM} \cos(2\pi t \times f_{AM})\right] \times \cos(2\pi t \times f_{0i})\right]$$

where:

A = 0,5

$f_{AM} = 4$ Hz, $\mu_{AM} = 0,5$

$f_{0\,i} = 250 \times 2^{(i/3)}$ Hz; i=1 to 11

CF = 14 dB ± 1 dB (10 dB + 4,26 dB due to 100 % AM modulation)

CF = Crest Factor = Peak to RMS ratio.

The training sequence level shall be -16 dBm0 in order not to overload the codec. The test signal level shall be -10 dBm0. The TCLw is calculated according to ITU-T Recommendation G.122, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s).

NOTE: Full scale of coder input signal corresponds to +3,14 dBm0 with sinusoidal signal, CF = 3dB. A test signal with a CF of maximum 15 dB can thus have a level of up to -8,86 dBm0 without overloading the codec. In order to get best dynamic range the signal amplitude should be as high as possible.

When using a PN-sequence, it should comply with ITU-T Recommendation P.501 [19] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. The test signal level is -3 dBm0. The low-crest factor is achieved by random-alternation of the phase between -180 degrees and +180 degrees.

The training sequence level shall be -16 dBm0 in order not to overload the codec. The TCLw is calculated according to ITU-T Recommendation G.122, clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band.

Care should be taken that the terminal under test considers the test signal as a speech-like signal.

F.5 Value for terminal coupling loss for control of talker echo

Users on communication systems have expressed their dislike of hearing their own speech as an echo (so called "Talker Echo") (see ITU-T recommendation G.131).

This dislike increases with increasing one-way transmission time, as shown in figure 1 in ITU-T Recommendation G.131.

Since TETRA has a large one-way transmission time, greater than 300 ms, then the value of talker echo loudness rating (TELR) needs to be large (56 dB or larger) for 1 % customer dissatisfaction.

$$TELR = SLR + TCLw + RLR$$

ITU-T recommendation P.310 clause 10 gives guidance about the need for Terminal Coupling Loss.

For TETRA terminals with a volume control, the TCLw value should be greater than 46 dB at maximum volume.

Annex G (informative): Channel re-assignment time

G.1 Scope

This annex defines the measurements and metrics applicable to TETRA networks to enable the effects associated with channel re-assignment time to be manageable. This parameter is applicable for TETRA systems where Traffic CHannel (TCH) resources are queued and may be experienced by subscribers when establishing connections to be conveyed by the network. The parameter's magnitude will impact the effective TCH capacity, call holding times, call attempt volumes and call establishment time.

Channel re-assignment time parameters will apply where normal subscriber connection establishment requests are in a queue for TCH resources.

G.2 Channel re-assignment time

Channel re-assignment time: the time span from the instant of a "Call Clear" to the instant when a traffic channel resource is successfully allocated by the network to another call.

NOTE 1: "Call Clear" is the sending of the appropriate message, e.g.: "D-Release", monitored e.g. at IMP I_U (in figure 1) or the reception of a "U-Disconnect" message monitored e.g. at I_{U1} (in figure 1).

NOTE 2: The definition of the channel re-assignment time implies that there is a queue for that traffic channel at the time of the "Call Clear" message.

The "end-to-end" and Intermediate Monitoring Points (IMP) and the times defined in figure 1, including supportive clauses, are used in this annex.

G.3 Observation and reporting of channel re-assignment time

Channel re-assignment time may be determined for a connection invoked by subscriber A through monitoring the time difference between the instant of the sending of a "Call Clear" message by the network for an ongoing call (time $\mathbf{t_x}$) at a air interface related IMP and the instant when a Traffic Channel (TCH) resource is allocated to support the new connection (time t_y) (time $\mathbf{t_c}$) at the same IMP detailed in figure 1.

NOTE: The channel re-assignment time as defined is independent of for what reason the call is released.

Annex H (informative): Mobility management success

For further study.

Annex I (informative): Packet data Quality of Service metrics

For further study.

Annex J (informative): Idle channel noise measurement

J.1 Idle channel noise (handset and headset terminal)

The measurements in clauses J.1.1 to J.1.2 are proposed methods and are for further study.

J.1.1 Sending

The terminal should be configured to the test equipment as described in clause E.5.1.

The environment shall comply with the conditions described in clause E.6.1 for idle channel noise measurement.

For testing narrow-band functionality, the Psophometric noise level at the output of the SS is measured. The psophometric filter is described in ITU-T Recommendation O.41.

A test signal may have to be intermittently applied to prevent "silent mode" or DTX operation of the MS. This is for further study.

J.1.2 Receiving

The terminal should be configured to the test equipment as described in clause E.5.1.

The environment shall comply with the conditions described in clause E.6.1.

A test signal may have to be intermittently applied to prevent "silent mode" operation of the MS. This is for further study.

For testing narrow-band or wideband functionality, the A-weighted level of the noise shall be measured at the ERP. The A-weighting filter is descried IEC 61672-1 [24].

Annex K (informative): Voice sensitivity/frequency characteristics measurement.

The measurements in clauses K1.1 to K.1.6 are proposed methods and are for further study.

K.1 Sensitivity/frequency characteristics

K.1.1 Handset or headset terminal sending

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

b) The handset or headset terminal is setup as described in clause E.5.1.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [22] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

c) The sensitivity is expressed in terms of dBV/Pa.

K.1.2 Handset or headset terminal receiving

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The handset or headset terminal is setup as described in clause E.5.1. The handset is mounted at the HATS / LRGP position (see ITU-T Recommendation P.64 [15]). The application force used to apply the handset against the artificial ear shall be within the range specified in ITU-T Recommendation P.64 [15].

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in IEC 3 for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V, referred to the ERP (see ITU-T Recommendation P.57 [13] table 2b for DRP to ERP correction.) For binaural headsets the frequency response is measured individually for each ear. Information about correction factors are available in ITU-T Recommendation P.57 [13].

K.1.3 Vehicle mounted and desk-top hands-free terminal sending

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to -28,7 dBPa at the HFRP or the HATSHFRP (as defined in ITU-T Recommendation P.581 [20]) and the spectrum is not altered.

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

b) The hands-free terminal is setup as described in clause E.5.1.

Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [22] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

K.1.4 Vehicle mounted and desk-top hands-free terminal receiving

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in clause 5. If the HATS is used then it is freefield equalized as described in ITU-T Recommendation P.581 [20]. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the "right" and "left" signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [22] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

K.1.5 Hand-Held hands-free MS sending

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence. The broadband signal level then is adjusted to -28,7 dBPa at the HFRP or the HATSHFRP (as defined in ITU-T Recommendation P. 581[20]) and the spectrum is not altered.

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

b) The hands-free terminal is setup as described in clause E.5.1.3.2. Measurements shall be made at one third-octave intervals as given by the R 40 series of preferred numbers in ISO 3 [22] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBV/Pa.

K.1.6 Hand-Held hands-free MS receiving

a) The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

b) The hands-free terminal is setup as described in clause E.5.1.3.2. If the HATS is used then it is freefield equalized as described in ITU-T Recommendation P.581 [20]. The equalized output signal of each artificial ear is power-averaged on the total time of analysis; the "right" and "left" signals are voltage-summed for each 1/3 octave band frequency band; these 1/3 octave band data are considered as the input signal to be used for calculations or measurements. Measurements shall be made at one third-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [22] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

c) The sensitivity is expressed in terms of dBPa/V.

Annex L (informative): Voice telephony sidetone measurements

The measurements in clauses L.1.1 to L.1.3 are proposed methods and are for further study.

L.1 Sidetone characteristics

L.1.1 Connections with Handset terminal

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset shall be positioned in the LRGP. The handset terminal is setup as described in clause E.5, with the following exception: The type 3.2 Low Leak artificial ear, according to ITU-T Recommendation P.57 [13] shall be used.

The possible use of type 3.2 High Leak, type 3.3, or type 3.4 artificial ears for measurement of the sidetone loss is for further study.

The sidetone path loss L_{meST} as expressed in dB shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17. The STMR (in dB) shall be calculated from the equation (B-4) of ITU-T Recommendation P.79 [16], using m = 0,225 and the weighting factors in table B.2 of ITU-T Recommendation P.79 [16].

L.1.2 Headset terminal

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The artificial ear type is for further study.

The sidetone path loss L_{meST} as expressed in dB shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17. The STMR (in dB) shall be calculated from the equation B-4 of ITU-T Recommendation P.79 [16], using m = 0,225 and the weighting factors in table B.2 of ITU-T Recommendation P.79 [16].

L.1.3 Hands-free terminal (all categories)

No requirement for other than echo control.

Annex M (informative): Voice telephony stability loss measurement

The measurements in this clause M.1 are proposed methods and are for further study.

M.1 Stability loss

Where a user controlled volume control is provided it is set to maximum.

Handset terminal: the handset is placed on a hard plane surface with the transducers facing the surface.

Headset terminal: for further study.

Hands-free terminal (all categories): no requirement other than echo loss.

A gain equivalent to the minimum stability margin is inserted in the loop between the go and return paths of the reference speech coder in the SS and any acoustic echo control is enabled.

A test signal according to ITU-T Recommendation O.131 is injected into the loop at the analogue or digital input of the reference speech codec of the SS and the stability is measured. The test signal has a level of -10 dBm0 and duration of 1 s.

No continuous audible oscillation shall be detected after the test signal is switched off.

Annex N (informative): Ambient noise rejection measurement

The measurements in clause N.1 are proposed methods and are for further study.

N.1 Ambient noise rejection

Handset and Headset terminal:

a) A 1/2 inch pressure microphone is calibrated using a known sound source and mounted at the MRP, without the HATS present. A frequency analyser is calibrated to enable the sound pressure levels at the microphone to be determined in 1/3 octave bands.

b) Flood the room in which the measurement is to be made with a band limited (100 Hz to 8 kHz) pink noise to within ±3 dB. The level at MRP shall be adjusted to 70 dB(A) (-24 dBPa(A)). The tolerance on this level is ±1 dB. The resulting sound spectrum is P_{rn} dBPa, measured in 1/3 octave bands.

   To ensure that the sound field is diffuse enough, the following apply:

   > The diffuse sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within ±3 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 3,15 kHz.

NOTE 1: The pressure intensity index, as defined in ISO 9614 [25], may prove to be a suitable method for assessing the diffuse field.

NOTE 2: Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers must be fed with non-coherent electrical signals to eliminate standing waves and other interference effects.

c) Position the HATS test head in the correct relative position to the MRP and mount the terminal handset or headset under test, according to clause E.5.1. Recalibrate the 1/3 octave frequency analyser using a known voltage source to facilitate the analysis of the voltage V_{rn}, where V_{rn} is the voltage at the audio output of the SS due to the noise spectrum input.

d) Set up a speech path between the terminal and the System Simulator (SS).

e) Determine, as a function of frequency, using the frequency analyser, in 1/3 octave bands (index j), the electrical output V_{jrn}, (expressed as dB relative 1 V) at the audio output of the SS for the applied acoustic pressure P_{jrn} (expressed as dB relative 1Pa) at the MRP. Since, the terminal sending sensitivity is not defined above 3,4 kHz the measurement shall be cut off at 3,4 kHz. For the bands below 315 Hz, the noise level shall be referenced to the speech level at 315 Hz to yield the DELSM.

   The room noise sensitivity is expressed as:

$$Sm_{jrn} = V_{jrn}\ (\text{dBV}) - P_{jrn}\ (\text{dBPa}).$$

   The MS ambient noise send sensitivity has now been determined.

f) The MS speech send sensitivity is now required. The required sensitivity is defined as the electrical output from the terminal, measured at the audio output of the SS, as a function of the free field sound pressure at the MRP of the artificial mouth.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [10]. The type of test signal used shall be stated in the test report. The 1/2 inch pressure microphone is calibrated using a known sound source. The frequency analyser is calibrated to measure in 1/3 octave bands. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The resulting sound spectrum is P_o dBPa, measured in 1/3 octave bands. The 1/3 octave frequency analyser should be re-calibrated, using a known voltage source, to facilitate the analysis of the voltage V_j. Where V_j is the voltage in each 1/3 octave band at the audio output of the SS due to the test signal input. Set up a speech path between the terminal and the SS. Determine the function of frequency, using the frequency analyser, and in 1/3 octave bands, the electrical output, V_j, (expressed as dB relative to 1 V), at the audio output of the SS for the applied acoustic pressure, P_{j0}, (expressed as dB relative to 1 Pa/V), at the MRP.

The speech sending sensitivity is expressed as:

$$Sm_{js} \text{ (dB)} = V_j \text{ (dBV)} - P_{jo} \text{ (dBPa) dB relative to 1 V/Pa.}$$

g) The difference of the room noise sensitivity and the speech sending sensitivity DELSM (Δ_{jSM}) in each 1/3 octave band for the terminal is determined as:

$$Sm_{jrn} - Sm_{js} \text{ (dB)}, \qquad \text{for } j = 1 \text{ to } 2,\ Sm_{js} = Sm_{3s}).$$

h) The Ambient noise rejection ANR is calculated as the single figure value according to the following formula, the ANR shall be ≥0 dB.

$$ANR = -\frac{4}{5}\sum_{i=1}^{13} \Delta_{jSM} \times 10^{-0,0175 W_{jsi}},$$

where:

j = The index of third octave bands centred at frequencies from 200 Hz to 3 150 Hz inclusive.

W_{jsi} = The sending weighting factors from ITU-T Recommendation P.79 [16], table 1 for the jth 1/3 octave band centre frequency.

Hands-free terminal (all categories including vehicle-mounted terminal):

Test method for hands-free operations is for further study.

Annex O (informative): Receiving loudness ratings

O.1 Applicability

The performance requirements in clauses O.2.1 to O.2.5 apply when terminal is used to provide narrow-band full duplex calls, either as a stand-alone service, or as part of a multimedia service.

The measurements in clause O.2 are proposed methods and are for further study.

O.2 Overall loss/loudness ratings

O.2.1 General

An international telephony connection involving a TETRA network and the PSTN should meet the overall loudness rating (OLR) limits in ITU-T Recommendation G.111 [5]. The national parts of the connection should therefore meet the send and receive loudness rating (SLR, RLR) limits in ITU-T Recommendation G.121 [6].

OLR is defined as:

$$OLR = SLR + \text{Circuit Loss} + RLR$$

For the case where digital routings are used to connect the TETRA network to the international chain of circuits, the SLR and RLR of the national extension will be largely determined by the SLR and RLR of the TETRA network. The limits given below are consistent with the national extension limits and long-term objectives in ITU-T Recommendation G.121 [6].

The SLR and RLR values for the TETRA network apply up to the POI.

However, since the circuit loss of a TETRA SwMI is 0 dB, then the main determining factors are the characteristics of the terminal, including the analogue to digital conversion (ADC) and digital to analogue conversion (DAC). In practice, it is convenient to specify loudness ratings to the Air Interface. For the normal case, where the TETRA SwMI introduces no additional loss between the Air Interface and the POI, the loudness ratings to the PSTN boundary (POI) will be the same as the loudness ratings measured at the Air Interface. However, in some cases loss adjustment may be needed for interworking situations in individual countries.

The POI is a 0 dBr level reference point.

O.2.2 Connections with handset terminal

The nominal values of RLR from the 0 dBr level reference point should be:

$$RLR = 2 \text{ db} \pm 3 \text{ dB}.$$

NOTE 1: The value of receiving level is strongly recommended for new work, but it is recognized that where backward compatibility is an issue that this value need not be applied.

NOTE 2: It is acknowledged that in a multi-vendor network or where the TETRA Inter-System-Interface (ISI) is involved that moves may be required towards this value by the operators and users of the network.

Where a user controlled receiving volume control is provided, the RLR shall meet the selected nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall not be less than (louder than) -13 dB.

With the volume control set to the minimum position the RLR shall not be greater than (quieter than) 18 dB.

Compliance should be checked by the relevant tests described in clause E.7 of the present document.

NOTE 3: The mechanical design of some terminal may make it impossible to seal the earpiece to the knife-edge of the ITU-T artificial ear. Minimal additional methods may be used to provide the seal provided that they do not affect the mounting position of the terminal with respect to the Mouth Reference Point and the Ear Reference Point.

O.2.3 Connections with desktop and vehicle-mounted hands-free terminal

The nominal values of RLR from the 0 dBr level reference point should be:

RLR = 2 ± 4 dB.

NOTE 1: The value of receiving level is strongly recommended for new work, but it is recognized that where backward compatibility is an issue that this value need not be applied.

NOTE 2: It is acknowledged that in a multi-vendor network or where the TETRA Inter-System-Interface (ISI) is involved that moves may be required towards this value by the operators and users of the network.

Compliance should be checked by the relevant tests described in clause E.7 of the present document.

Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control. It is recommended that a volume control giving at least 15 dB increase from the nominal RLR (louder) is provided for hands-free units intended to work in the vehicle environment. This is to allow for the increased noise volume in a moving vehicle.

O.2.4 Connections with handheld hands-free MS not having an AGC function

The nominal values of RLR from the 0 dBr level reference point should be:

RLR = 6 dB + 12 dB to 6 dB - 4 dB

NOTE 1: The value of sending level is strongly recommended for new work, but it is recognized that where backward compatibility is an issue that this value need not be applied.

NOTE 2: It is acknowledged that in a multi-vendor network or where the TETRA Inter-System-Interface (ISI) is involved that moves may be required towards this value by the operators and users of the network.

Compliance should be checked by the relevant tests described in clause E.7.

Where a user controlled volume control is provided, the RLR shall meet the nominal value at one setting of the control.

O.2.5 Connections with headset terminal

The nominal values of RLR from the 0 dBr level reference point should be:

RLR = 2 dB ± 3 dB with any volume control set to mid position.

NOTE 1: The value of sending level is strongly recommended for new work, but it is recognized that where backward compatibility is an issue that this value need not be applied.

NOTE 2: It is acknowledged that in a multi-vendor network or where the TETRA Inter-System-Interface (ISI) is involved that moves may be required towards this value by the operators and users of the network.

Where a user controlled receiving volume control is provided, the RLR shall meet the selected nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall not be less than (louder than) -13 dB.

With the volume control set to the minimum position the RLR shall not be greater than (quieter than) 18 dB.

Compliance should be checked by the relevant tests described in clause E.7.

Annex P (informative): Bibliography

- ITU-T Recommendation E.721: "Network grade of service parameters and target values for circuit-switched services in the evolving ISDN".
- ITU-T Recommendation I.352: "Network Performance Objectives for Connection Processing Delays in an ISDN".
- ITU-T Recommendation G.122: "Influence of national systems on stability and talker echo in international connections".
- ITU-T Recommendation G.131: "Control of talker echo".
- ITU-T Recommendation G.165: "Echo cancellers".
- ITU-T Recommendation O.41: "Psophometer for use on telephone-type circuits".
- ITU-T Recommendation O.131: "Quantizing distortion measuring equipment using a pseudo-random noise test signal".
- ITU-T recommendation P.310: "Transmission characteristics for telephone band (300-3400 Hz) digital telephones".
- ETSI EN 300 392-1: "Terrestrial Trunked Radio (TETRA); Voice plus Data (V+D); Part 1: General network design".
- ETSI TR 101 110: "Digital cellular telecommunications system (Phase 2+) (GSM); Characterisation, test methods and quality assessment for handsfree Mobile Stations (MSs) (GSM 03.58)".

History

Document history		
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