# ETSI TS 101 377-3-13 V1.1.1 (2001-03)

*Technical Specification*

## GEO-Mobile Radio Interface Specifications;
## Part 3: Network specifications;
## Sub-part 13: Transmission Planning Aspects of the Speech Service in the Public Satellite Mobile Network (PSMN) system;
## GMR-2 03.050

Reference

DTS/SES-002-03050

Keywords

GMR, GSM, GSO, interface, MES, mobile, MSS, network, public, radio, satellite, S-PCN, speech, transmission

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# Contents

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**IPRs:**

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| TS 101 377 V1.1.1 | Digital Voice Systems Inc | | US | US 5,715,365 | US |
| TS 101 377 V1.1.1 | Digital Voice Systems Inc | | US | US 5,754,974 | US |
| TS 101 377 V1.1.1 | Digital Voice Systems Inc | | US | US 5,226,084 | US |
| TS 101 377 V1.1.1 | Digital Voice Systems Inc | | US | US 5,701,390 | US |
| TS 101 377 V1.1.1 | Digital Voice Systems Inc | | US | US 5,826,222 | US |

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| TS 101 377 V1.1.1 | Ericsson Mobile Communication | Power Booster | GB | GB 2 251 768 | GB |
| TS 101 377 V1.1.1 | Ericsson Mobile Communication | Receiver Gain | GB | GB 2 233 846 | GB |
| TS 101 377 V1.1.1 | Ericsson Mobile Communication | Transmitter Power Control for Radio Telephone System | GB | GB 2 233 517 | GB |

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|---|---|---|---|---|---|
| TS 101 377 V1.1.1 | Hughes Network Systems | | US | Pending | US |

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|---|---|---|---|---|---|
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| TS 101 377 V1.1.1 | Lockheed Martin Global Telecommunic. Inc | Cellular Spacecraft TDMA Communications System with Call Interrupt Coding System for Maximizing Traffic ThroughputCellular Spacecraft TDMA Communications System with Call Interrupt Coding System for Maximizing Traffic Throughput | US | US 5,717,686 | US |
| TS 101 377 V1.1.1 | Lockheed Martin Global Telecommunic. Inc | Enhanced Access Burst for Random Access Channels in TDMA Mobile Satellite System | US | US 5,875,182 | |
| TS 101 377 V1.1.1 | Lockheed Martin Global Telecommunic. Inc | Spacecraft Cellular Communication System | US | US 5,974,314 | US |
| TS 101 377 V1.1.1 | Lockheed Martin Global Telecommunic. Inc | Spacecraft Cellular Communication System | US | US 5,974,315 | US |
| TS 101 377 V1.1.1 | Lockheed Martin Global Telecommunic. Inc | Spacecraft Cellular Communication System with Mutual Offset High-argin Forward Control Signals | US | US 6,072,985 | US |
| TS 101 377 V1.1.1 | Lockheed Martin Global Telecommunic. Inc | Spacecraft Cellular Communication System with Spot Beam Pairing for Reduced Updates | US | US 6,118,998 | US |

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# Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Satellite Earth Stations and Systems (SES).

The contents of the present document are subject to continuing work within TC-SES and may change following formal TC-SES approval. Should TC-SES modify the contents of the present document it will then be republished by ETSI with an identifying change of release date and an increase in version number as follows:

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where:

- the third digit (n) is incremented when editorial only changes have been incorporated in the specification;
- the second digit (m) is incremented for all other types of changes, i.e. technical enhancements, corrections, updates, etc.

The present document is part 3, sub-part 13 of a multi-part deliverable covering the GEO-Mobile Radio Interface Specifications, as identified below:

Part 1: "General specifications";

Part 2: "Service specifications";

**Part 3: "Network specifications";**

Sub-part 1: "Network Functions; GMR-2 03.001";

Sub-part 2: "Network Architecture; GMR-2 03.002";

Sub-part 3: "Numbering, Addressing and Identification; GMR-2 03.003";

Sub-part 4: "Restoration Procedures; GMR-2 03.007";

Sub-part 5: "Organization of Subscriber Data; GMR-2 03.008";

Sub-part 6: "Handover Procedures; GMR-2 03.009";

Sub-part 7: "Technical Realization of Short Message Service (SMES) Point-to-Point; GMR-2 03.040";

Sub-part 8: "Location Registration Procedures; GMR-2 03.012";

Sub-part 9: "Discontinuous Reception (DRX) in the GMR-2 System; GMR-2 03.013";

Sub-part 10: "Security Related Network Functions; GMR-2 03.020";

Sub-part 11: "Functions Related to Mobile Earth Station (MES) in idle Mode; GMR-2 03.022";

Sub-part 12: "Technical Realization of Facsimile Group 3 Transparent; GMR-2 03.045";

**Sub-part 13: "Transmission Planning Aspects of the Speech Service in the Public Satellite Mobile Network (PSMN) system; GMR-2 03.050";**

Sub-part 14: "Call Waiting (CW) and Call Hold (HOLD) Supplementary Services - Stage 2; GMR-2 03.083";

Sub-part 15: "Multiparty Supplementary Services; GMR-2 03.084";

Sub-part 16: "Technical Realization of Operator Determined Barring; GMR-2 03.015";

Sub-part 17: "Call Barring (CB) Supplementary Services - Stage 2; GMR-2 03.088";

Part 4: "Radio interface protocol specifications";

Part 5: "Radio interface physical layer specifications";

Part 6: "Speech coding specifications";

Part 7: "Terminal adaptor specifications".

# Introduction

GMR stands for GEO (Geostationary Earth Orbit) Mobile Radio interface, which is used for mobile satellite services (MSS) utilizing geostationary satellite(s). GMR is derived from the terrestrial digital cellular standard GSM and supports access to GSM core networks.

Due to the differences between terrestrial and satellite channels, some modifications to the GSM standard are necessary. Some GSM specifications are directly applicable, whereas others are applicable with modifications. Similarly, some GSM specifications do not apply, while some GMR specifications have no corresponding GSM specification.

Since GMR is derived from GSM, the organization of the GMR specifications closely follows that of GSM. The GMR numbers have been designed to correspond to the GSM numbering system. All GMR specifications are allocated a unique GMR number as follows:

GMR-n xx.zyy

where:

xx.0yy (z=0) is used for GMR specifications that have a corresponding GSM specification. In this case, the numbers xx and yy correspond to the GSM numbering scheme.

xx.2yy (z=2) is used for GMR specifications that do not correspond to a GSM specification. In this case, only the number xx corresponds to the GSM numbering scheme and the number yy is allocated by GMR.

n denotes the first (n=1) or second (n=2) family of GMR specifications.

A GMR system is defined by the combination of a family of GMR specifications and GSM specifications as follows:

- If a GMR specification exists it takes precedence over the corresponding GSM specification (if any). This precedence rule applies to any references in the corresponding GSM specifications.

NOTE: Any references to GSM specifications within the GMR specifications are not subject to this precedence rule. For example, a GMR specification may contain specific references to the corresponding GSM specification.

- If a GMR specification does not exist the corresponding GSM specification may or may not apply. The applicability of the GSM specifications is defined in GMR-n 01.201.

# 1 Scope

The present document is concerned with the transmission planning aspects pertaining to the speech service in the PSMN system. Due to technical and economic factors, there cannot be full compliance with the general characteristics of international telephone connections and circuits recommended by ITU-T.

The present document gives guidance as to the precautions, measures and minimum requirements needed for successful interworking of the PSMN with the national and international PSTN. The Recommendation identifies a number of routing and network configurations. The objective is to reach a quality as close as possible to ITU-T standards in order to safeguard the performance seen by PSTN customers.

# 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, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, subsequent revisions do apply.

[1] GMR-2 01.004 (ETSI TS 101 377-1-1): "GEO-Mobile Radio Interface Specifications; Part 1: General specifications; Sub-part 1: Abbreviations and Acronyms".

[2] GSM 03.04 (ETSI ETS 300 524): "European digital cellular telecommunication system (Phase 2); Signalling requirements relating to routing of calls to mobile subscribers" (V4.0.4).

[3] GSM 06.12 (ETSI ETS 300 580): "European digital cellular telecommunications system (Phase 2); Comfort noise aspect for full rate speech traffic channels" (V4.0.4).

[4] GSM 06.32 (ETSI ETS 300 580-6): "European digital cellular telecommunications system (Phase 2); Part 6: Voice Activity Detection (VAD) for full rate speech traffic channels" (V4.3.1).

[5] ETSI ETS 300 085 (1990): "Integrated Services Digital Network (ISDN); 3,1 kHz telephony teleservice; Attachment requirements for handset terminals (Candidate NET 33)".

[6] ITU-T Recommendation G.103 (1988): "Hypothetical reference connections".

[7] ITU-T Recommendation G.111 (1993): "Loudness ratings (LRs) in an international connections".

[8] ITU-T Recommendation G.113 (1996): "Transmission impairments".

[9] ITU-T Recommendation G.114 (1996): "One-way transmission time".

[10] ITU-T Recommendation G.121 (1993): "Loudness ratings (LRs) of national systems".

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

[12] ITU-T Recommendation G.131 (1996): "Control of talker echo".

[13] ITU-T Recommendation G.165 (1993): "Echo cancellers".

[14] ITU-T Recommendation G.223 (1984): "Assumptions for the calculation of noise on hypothetical reference circuits for telephony".

[15] ITU-T Recommendation G.703 (1991): "Physical/electrical characteristics of hierarchical digital interfaces".

[16] ITU-T Recommendation G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

[17] ITU-T Recommendation G.712 (1996): "Transmission performance characteristics of pulse code modulation channels".

[18] ITU-T Recommendation M.1020 (1993): "Characteristics of special quality international leased circuits with special bandwidth conditioning".

[19] ITU-T Recommendation M.1025 (1993): "Characteristics of special quality international leased circuits with basic bandwidth conditioning".

[20] ITU-T Recommendation M.1030 (1988): "Characteristics of ordinary quality international leased circuits forming part of private switched telephone networks".

[21] ITU-T Recommendation M.1040 (1988): "Characteristics of ordinary quality international leased circuits".

[22] ITU-T Recommendation O.132 (1988): "Quantizing distortion measuring equipment using a sinusoidal test signal".

[23] ITU-T Recommendation P.11 (1993): "Effect of transmission impairments".

[24] ITU-T Recommendation P.340 (1996): "Transmission characteristics of hands-free telephones".

[25] ITU-T Recommendation P.35 (1988): "Handset telephones".

[26] ITU-T Recommendation P.38 (1993): "Transmission characteristics of operator telephone systems (OTS)".

[27] ITU-T Recommendation P.50 (1993): "Artificial voices".

[28] ITU-T Recommendation P.51 (1996): "Artificial mouth".

[29] ITU-T Recommendation P.57 (1996): "Artificial ears".

[30] ITU-T Recommendation P.64 (1997): "Determination of sensitivity/frequency characteristics of local telephone systems".

[31] ITU-T Recommendation P.76 (1988): "Determination of loudness ratings; fundamental principles".

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

[33] ITU-T Recommendation E.180/Q.35 (1998): "Technical characteristics of tones for the telephone service".

[34] ITU-T Recommendation Q.551 (1996): "Transmission characteristics of digital exchanges".

[35] ISO 3 (1973): "Preferred numbers - series of preferred numbers".

# 3 Abbreviations

For the purposes of the present document, the abbreviations given in GMR-2 01.004 [1] and the following apply:

| | |
|---|---|
| ADC | Analogue to Digital Converter |
| ADPCM | Adaptive Differential Pulse Code Modulation |
| AEC | Acoustic Echo Control |
| DAC | Digital to Analogue Converter |
| DMR | Digital Mobile Radio |
| DSI | Digital Speech Interpolation |
| EEC | Electric Echo Control |
| EL | Echo Loss |
| ERP | Ear Reference Point |
| FDM | Frequency Division Multiplex |
| GSC | Gateway Station Controller |
| GTS | Gateway Transceiver Subsystem |

| | |
|---|---|
| GWS | Gateway Subsystem |
| LSTR | Listener Sidetone Rating |
| MRP | Mouth Reference Point |
| OLR | Overall Loudness Rating |
| PCM | Pulse Code Modulation |
| POI | Point of Interconnection (with PSTN) |
| RLR | Receiver Loudness Rating |
| SLR | Send Loudness Rating |
| STMR | Sidetone Masking Rating |
| UPCMI | 13-bit Uniform PCM Interface |

# 4 Introduction

Since the transmission quality and the conversational quality of the PSMN will in general be lower than the quality of the PSTN connection due to coding distortion, delay, etc, only some transmission aspects can be brought in line with ITU-T Recommendations. It is therefore necessary to improve the overall quality as much as possible by implementing proper routing and network configurations.

It should be recognized that the transmission plan for the PSMN cannot lead to major changes in the PSTN. However, it is important to use the improvements in the evolving PSTN (e.g. digitization, introduction of echo cancellers) in an effective way.

The transmission requirements are in the first place based on international connections. When the quality is sufficient for international connections, it can be assumed that the national connections will have the same or better quality.

In order to obtain a sufficient quality in the connection, it is preferable to have digital connectivity between the Gateway Subsystem (GWS) and the international exchange. The PSMN requirements are based on this assumption. When this situation cannot be provided, a lower quality must temporarily be accepted.

This Recommendation consists of two parts: one will deal with network configurations, the other with transmission performance.

The part about network configurations gives information about the reference connections, on which the transmission plan is based. Furthermore, some guidelines are presented for improvement of the transmission quality in the evolving (digital) PSTN.

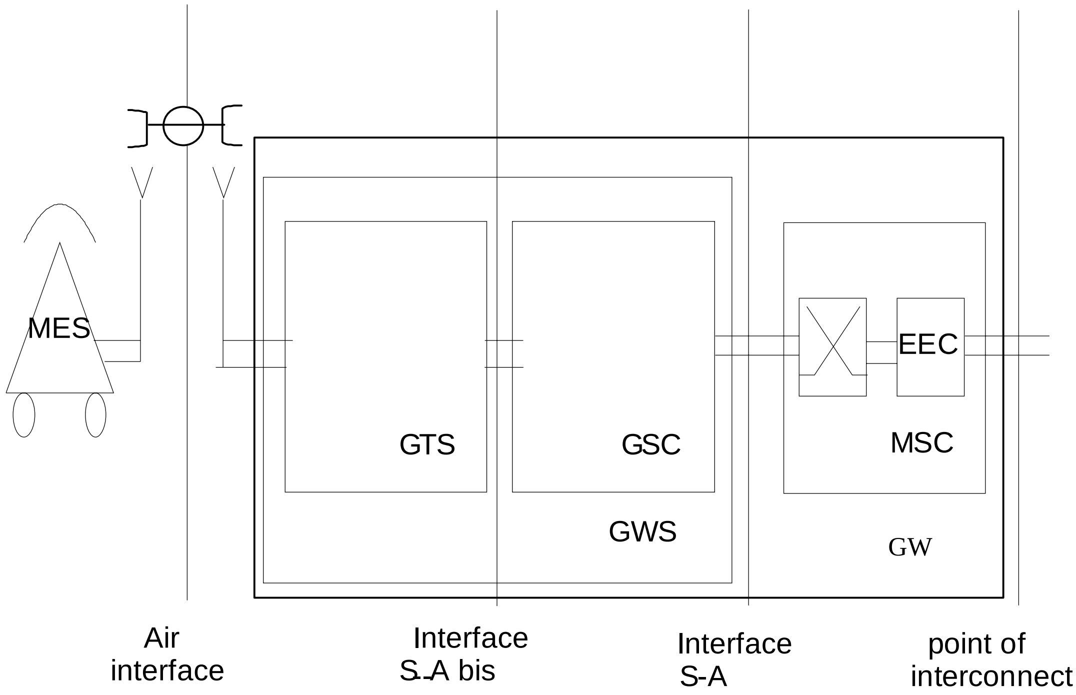
The part about transmission performance gives mainly characteristics of the transmission between MES acoustic interface (MRP/ERP) and the interface between the PSMN and the PSTN (POI). For transmission aspects where it is impossible to give overall characteristics, it is in some cases necessary to make recommendations for individual parts of the equipment.

Annex A considers the effects of the type of acoustic interfaces of the MES.

# 5 Network configurations

## 5.1 General

The basic configuration for the interworking with the PSTN is shown in figure 5.1-1.



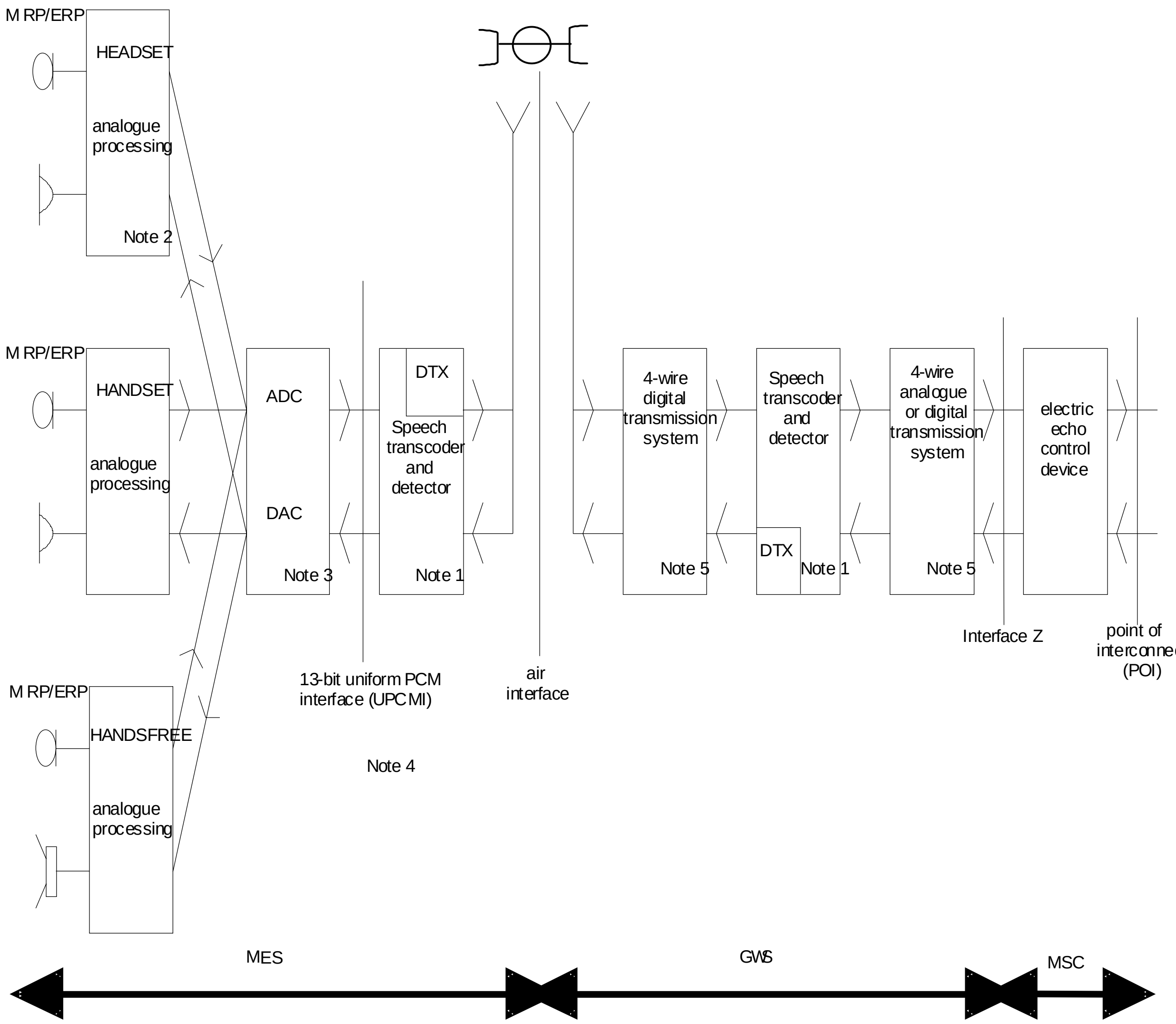

NOTE 1: For transmission planning purposes. Interfaces A and A bis are not required.
NOTE 2: The speech transcoder is at the GTS.

**Figure 5.1-1: Basic Configuration for Interworking with the PSTN**

## 5.2 Model of the PSMN

A more detailed model of the PSMN used for the consideration of transmission planning issues for speech is shown in figure 5.2-1. This model represents the main functions required and does not necessarily imply any particular physical realization. Routing of calls is given in Recommendation GSM 03.04 [2].

Any acoustic echo control is not specifically shown, as it will be provided by analogue processing of digital processing or a combination of both techniques.




NOTE 1: Speech detection is incorporated in the speech transcoder. Speech detection is needed to provide the function of DTX and, if required, acoustic echo control (see note 2).
NOTE 2: Acoustic echo control may not be provided in the case of the headset.
NOTE 3: Includes filtering.
NOTE 4: In simple talk and in double talk, when no centre clipper is used, and in double talk only if a centre clipper is used, the level of quantizing noise introduced by the speech transcoding will affect the level of residual acoustic echo when echo cancellation techniques are used for AEC.
NOTE 5: The transmission system need not be present.

**Figure 5.2-1: PSMN System Model Used for Consideration of Transmission Planning Issues**

# 5.3 Interfaces

The main interfaces identified within the GMR-2 Recommendations are shown in figure 5.2-1. For the purposes of this Recommendation, the Air Interface and the Point of Interconnect (POI) are identified along with two other interfaces, Interface Z and a 13-bit Uniform PCM Interface (UPCMI). These interfaces are needed to define the PSMN transmission characteristics and the overall system requirements.

The Air Interface is specified by GMR-2 05 series Recommendations and is required to achieve MES transportability. Analogue measurements can be made at this point by using the appropriate radio terminal equipment and speech transcoder. The losses and gains introduced by the test speech transcoder will need to be specified.

The POI with the PSTN will generally be at the 2 048 kbit/s level at an interface, in accordance with ITU-T Recommendation G.703 [15]. At the point, which is considered to have a relative level of 0 dBr, the analogue signals will be represented by 8-bit A-law, according to ITU-T Recommendation G.711 [16]. Analogue measurements may be made at this point using a standard send and receive side, as defined in ITU-T Recommendation G.712 [17].

Interface Z might be used in the case of direct MSC to MSC connections. Interface Z is of the same nature as the POI.

The UPCMI is introduced for design purposes in order to separate the speech transcoder impairments from the basic audio impairments of the MES.

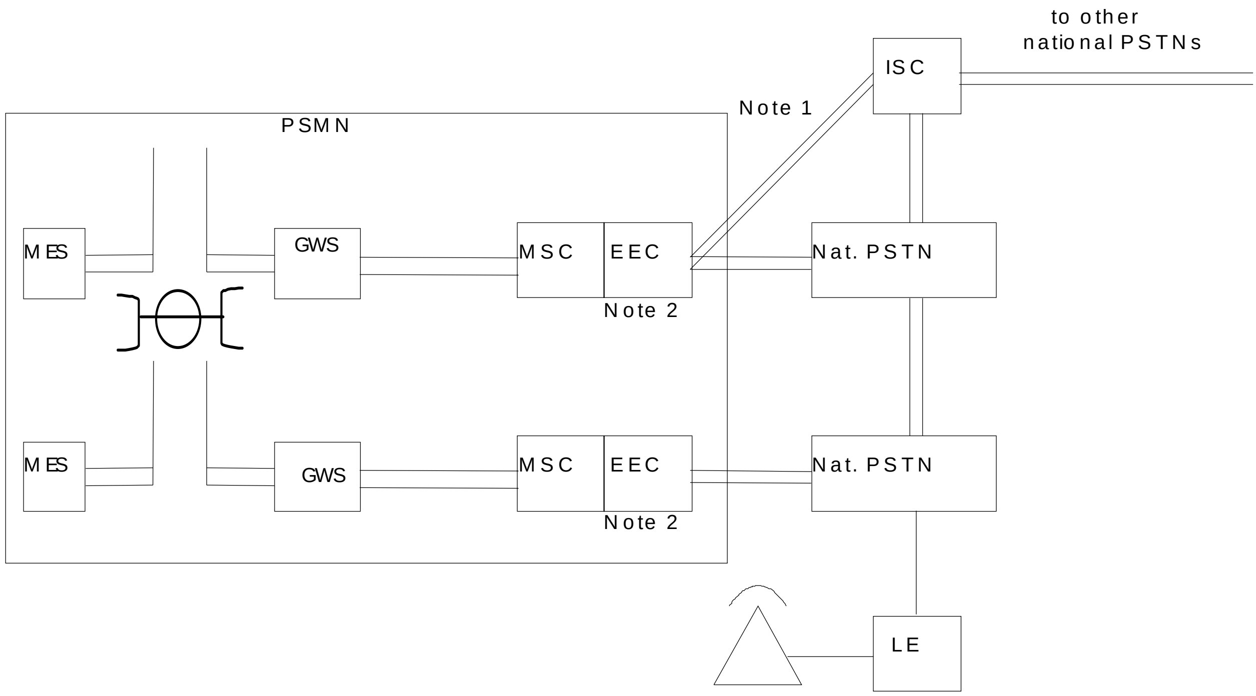
# 5.4 Configurations of Connections

## 5.4.1 General Configurations of Connections

Figure 5.4.1-1 shows a variety of configurations of connections. There are a number of PSTN features which should be avoided from such connections. These include:

- echo control devices in the international network. If present, and not disabled, these devices will be in tandem with PSMN echo cancellers and may introduce degradation;
- satellite routings. The delay inherent in the connections when added to the PSMN delay, may result in conversational difficulties. Double satellite links are likely to cause severe difficulties and special precautions should be taken to avoid this situation under call forwarding arrangements;
- digital speech interpolation systems (DSI). There is likely to be an adverse interaction between DSI and DTX;
- ADPCM. The distortion introduced by ADPCM on routes where PSTN echo control is not provided is likely to reduce the echo cancellation provided by the PSMN electric echo canceller;
- significant differences in clock rates on non-synchronized digital network components. The resulting phase roll and slips are likely to degrade the performance of the PSMN echo canceller;
- those analogue FDM routings which exhibit phase roll. Any phase roll due to the absence of synchronization between the carrier frequencies on the two directions of transmission is likely to degrade the performance of the PSMN echo canceller;
- tandem connections of sources of quantization distortion. The PSMN speech transcoder is estimated to be equivalent to 7 QDUs between uniform PCM interfaces (see ITU-T Recommendation G.113 [8]).

It is recognized that on some connections it may not be feasible to avoid these features, but in many cases, especially if taken into account at the planning stage, this should be possible.




NOTE 1: A direct link between MSC and ISC may be used in cases where Signalling System No. 7 is not provided in the PSTN or where a link via the PSTN would have excessive delay.

NOTE 2: An echo canceller should be provided at every POI to cancel any echo returning to the PSMN from the PSTN. This is necessary because the one-way echo path back to the MES greatly exceeds 25 ms (see ITU-T Recommendation G.131 [12]).

**Figure 5.4.1-1: PSMN to PSTN Interconnection Configurations**

### 5.4.2 Reference configurations to illustrate delay and echo control issues

Two basic reference configuration types shown in figures 5.4.2-1 and 5.4.2-2 are defined to illustrate delay and echo control issues. Intermediate echo control devices as shown in the figures are disabled by appropriate signalling between the MSC and ISC or MSC and MSC.

Reference configurations A (see figure 5.4.2-1) represent national or international connections where there is no echo control device in the PSTN. These reference configurations include re-routing configurations where the delay of the PSTN transmission path has not been extended.

Reference configurations B (see figure 5.4.2-2) represent national or international connections where echo control is provided in the PSTN. These reference configurations include re-routing configurations where the delay of the PSTN transmission path has been extended.

Direct routing and re-routing where the delay of the PSTN transmission path has not been extended and there is no echo control in the PSTN.

Normal configurations:

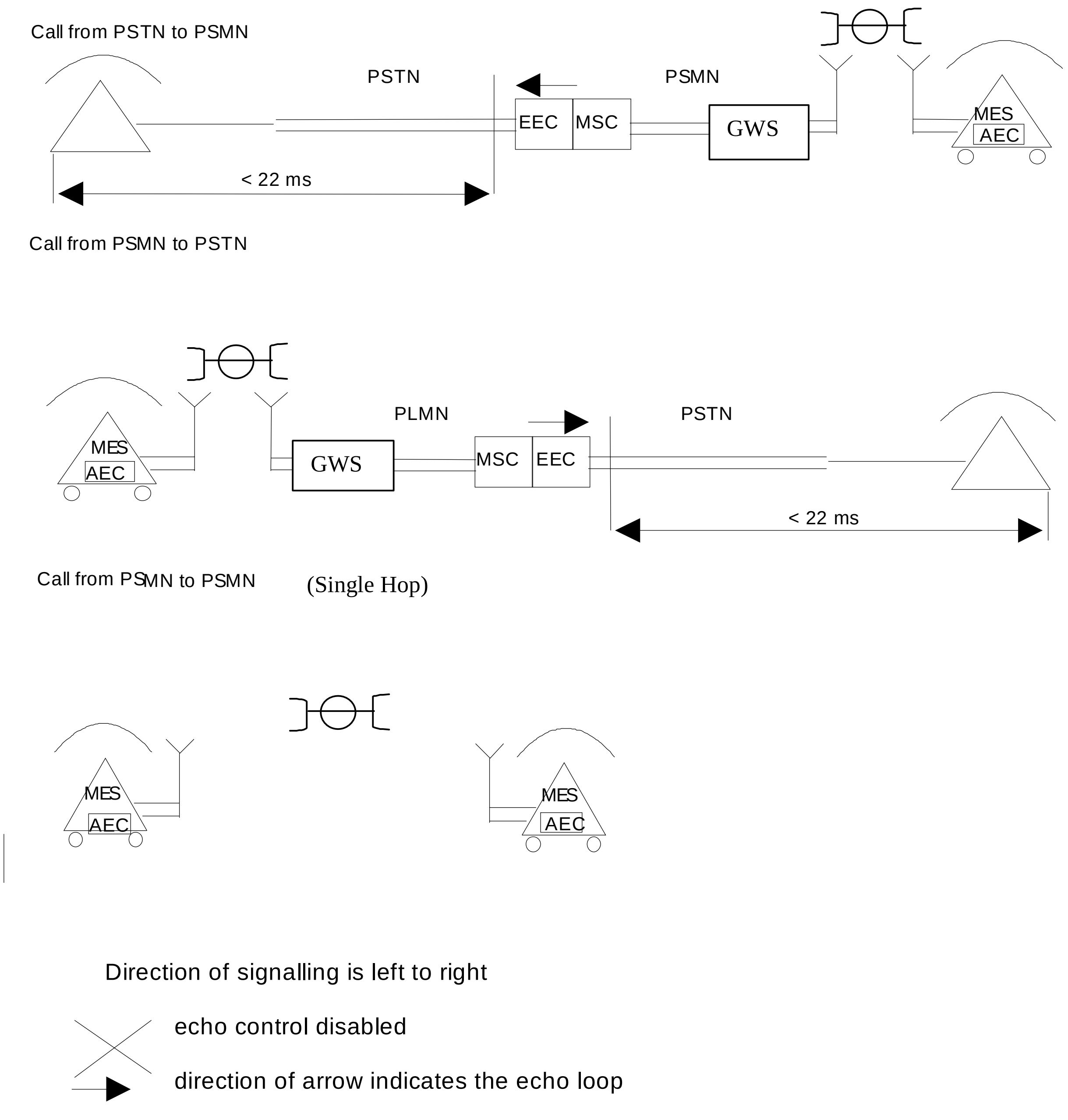
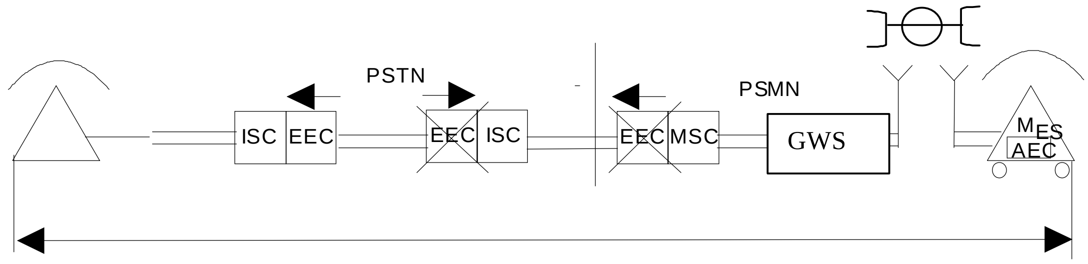



**Figure 5.4.2-1: Reference Configurations A**

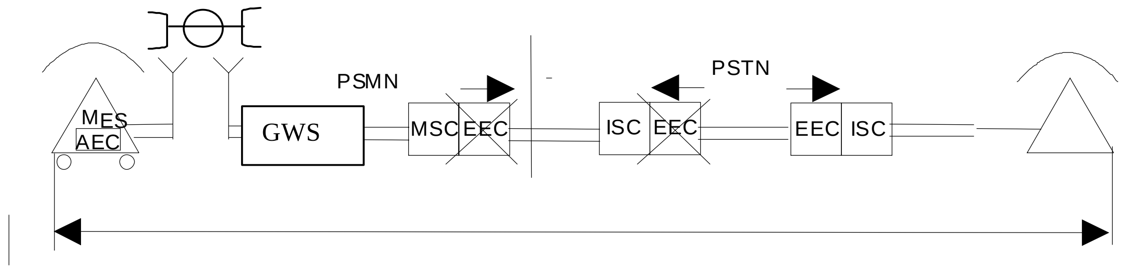
Direct routing and re-routings where the delay of the PSTN transmission path has been extended and there is echo control in the PSTN.

Normal Configurations:

Call from PSTN to PSMN




Call from PSMN to PSTN




Direction of signalling is left to right

echo control disabled

direction of arrow indicates the echo loop

**Figure 5.4.2-2: Reference Configurations B**

## 5.5 4-wire circuits in the PSMN

As shown in figure 5.2-1, the PSMN will usually contain transmission systems. Where present, they should provide 4-wire circuits.

In the case of digital circuits, which do not include any speech processing devices, the overall system requirements of the PSMN will not be affected by the presence of the link.

In the case of analogue links, the transmission characteristics (e.g. attenuation, attenuation distortion, noise) will affect the overall system requirements of the PSMN. ITU-T Recommendations M.1020 [18], M.1025 [19], M.1030 [20] and M.1040 [21] describe several transmission characteristics for leased circuits. In cases where the analogue link introduces loss, provision will have to be made at the interface to restore the loss.

# 6 Transmission performance

The overall transmission performance of connections in alternate conversation mode can be considered as a summation of the effects of:

- the audio part between the MRP/ERP and the UPCMI interface;
- the speech transcoder part including the effects of radio transmission, and speech processing between the UPCMI and the POI;
- the overall characteristics of the connection between POI and the other user.

There is not only a linear addition of these effects but there is also an influence from different parts of the connection on the performance of the speech transcoder and other speech processing devices.

Where possible, the transmission performance is specified between the MRP/ERP and the POI. Where this is not possible, the transmission aspects of the audio part mentioned above have been specified. The transmission aspects of the speech transcoder are specified in GSM 06 series Recommendations. In the following clauses, requirements are specified for the UPCMI, the Air Interface or the POI as appropriate.

The following clauses are applicable to handset MESs. In some places, reference is made to headset and handsfree MESs, but further study is needed to fully extend this Recommendation to these types of acoustic interface (see annex A).

The transmission requirements of the MES have been derived from the requirements of digital telephones stated in ETS 300 085 [5].

MESs will have to work in a variety of environments ranging from quiet office locations to very noisy environments as found in moving cars. In noisy conditions, different values for SLR, STMR and low frequency response may be required. These different values may be achieved by introducing some switchover function (manual or automatic). This point needs further study.

The overall transmission performance in full duplex conversation mode will also greatly depend on the performance of the echo control devices which may be included in the connection.

## 6.1 Overall Loss/Loudness ratings

The overall international connection involving PSMNs and the PSTN should meet the overall loudness rating (OLR) limits in ITU-T Recommendation G.111 [7]. The national parts of the connection should therefore meet the send and receive loudness rating (SLR, RLR) limits in ITU-T Recommendation G.121 [10].

For the case where digital routings are used to connect the PSMN 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 PSMN. The limits given below are consistent with the national extension limits and long-term objectives in ITU-T Recommendation G.121 [10].

The SLR and RLR values for the PSMN apply up to the POI. However, the main determining factors are the characteristics of the MES, including the analogue to digital conversion (ADC) and digital to analogue conversion (DAC). Hence, in practice, it will be convenient to specify loudness ratings to the Air Interface. For the normal case, where the PSMN 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.

These values are directly applicable to the case of an MES operating in a conventional non-mobile noise environment. Studies have shown that under the PSMN noise environment, speech levels are likely to be higher. Hence, in order to avoid clipping in the speech transcoder, the value of SLR may need to be increased.

NOTE: Measurement of SLR and RLR using sinusoidal test frequencies may not be sufficiently accurate because of the adaptive characteristics of the PSMN full-rate speech transcoder. A possible method is to use the artificial voice described in ITU-T Recommendation P.50 [27] to measure send and receive sensitivities. A method used by one administration uses the artificial voice to measure the loudness rating according to the Zwicker algorithm.

### 6.1.1 Connections with handset MESs

The nominal values of SLR/RLR to the POI shall be:

- SLR = 8 ± 3 dB;
- RLR = 2 ± 3 dB.

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 shall be checked by the tests described in clauses B.1.1 and B.1.2.

NOTE: The mechanical design of some MESs 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 MES with respect to the Mouth Reference Point and the Ear Reference Point.

### 6.1.2 Connections with hands-free MESs using loudspeakers

The SLR and RLR should be measured and computed using the methods given in ITU-T Recommendation P.340 [24] clause 6 with an artificial voice satisfying ITU-T Recommendation P.50 [27].

The values of SLR/RLR to/from the POI should be:

- SLR = 13 ± 3 dB (see note 1);
- RLR = 2 ± 3 dB with the volume control set to the mid position.

A receive volume control should be provided with a range of between ±7,5 dB and ±15 dB.

The use of values towards the most sensitive end of the range may result in problems with amplifying crosstalk from other channels (see note 2).

NOTE 1: This value takes into account the ITU-T Recommendation P.340 [24]; the SLR of a handsfree telephone should be 5 dB higher than the corresponding value for a handset instrument. The tolerance of ±3 dB is provisional.

NOTE 2: This procedure assumes no automatic gain control in the mobile terminal. The use of such techniques is not recommended for mobile applications.

NOTE 3: Further work is required to develop a practical test method using ITU-T Recommendation P.50 [27].

### 6.1.3 Connections with headset MESs

The SLR and RLR should be measured and computed using methods given in ITU-T Recommendation P.38 [26]. This Recommendation currently gives a measuring technique for supra-aural earphone and insert-type receivers. Study is continuing on other types of earpieces in ITU-T SGXII.

The nominal values of SLR/RLR to/from the POI should be:

- SLR = 8 ± 3 dB;
- RLR = 2 ± 3 dB with any volume control set to mid position.

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.

## 6.2 Stability Loss

The stability loss presented to the PSTN by the PSMN at the POI should meet the principles of the requirements in clauses 2 and 3 of ITU-T Recommendation G.122 [11]. These requirements will be met if the attenuation between the digital input and digital output at the POI is at least 6 dB at all frequencies in the range 200 Hz to 4 kHz under the worst-case acoustic conditions at the MES (any acoustic echo control should be enabled).

For the normal case of digital connection between the Air Interface and the POI, the stability requirement can be applied at the Air Interface. The worst-case acoustic conditions will be as follows (with any volume control set to maximum):

**Handset MES:** the handset lying on, and the transducers facing, a hard surface with the earpiece uncapped.

**Handsfree MES:** a representative worst-case position of microphone and loudspeaker (for further study).

**Headset MES:** for further study.

NOTE: The test procedure will need to take into account the switching effects of echo control and DTX.

## 6.3 Delay

### 6.3.1 General

A significant propagation time between the two ends of a connection causes difficulties in conversation over the connection. This arises from two causes. Firstly, the signal is reflected back from the distant end causing an echo to the talker (this is considered in clause 6.4). Secondly, even if ideal echo control were achieved, the delay between a user talking and receiving a reply from the user at the distant end of the connection could cause conversational difficulty.

PSMNs will be connected to the PSTN at a point where present planning rules allow for a delay of less than 12 ms (see ITU-T Recommendation G.114 [9] clause 2.2a). The delay within the PSMN will greatly exceed this. If unacceptable circuit delays are not to be experienced by users, action will have to be taken when planning routes or during call set-up.

### 6.3.2 Sources of delay

#### 6.3.2.1 Elements of the PSMN that cause delay

The delay of the PSMN is made up of the following elements:

- speech transcoding delay;
- radio channel coding delay;
- PSMN network delay (i.e. fixed elements such as multiplexing, geostationary satellite propagation, switching, echo control).

#### 6.3.2.2 Elements of the PSTN that cause delay

ITU-T recommendation G.114 [9] identifies various elements present in some PSTN connections which cause delay. These include:

- coaxial, radio and optical fibre terrestrial transmission systems;
- geostationary satellites;
- digital speech interpolators;
- digital exchanges (see also ITU-T recommendation Q.551 [34]);
- echo cancellers.

## 6.3.3 Effects of delay

Some recent studies have suggested that under ideal conditions, i.e.:

- effective control of all echoes without clipping by the use of good echo cancellers;
- low background noise leading to an absence of perceptible noise contrast;
- low distortion of transmitted signals;
- ideal loudness ratings;
- users can tolerate a circuit delay well in excess of 400 ms (currently the maximum delay recommended in ITU-T Recommendation G.114 [9]). Other studies indicate that the difficulty caused by circuit delay increases when impairments, such as imperfect echo control caused by echo suppressers, clipping and noise contrast, are present.

However, the mobile environment is very harsh, with high background noise levels and distortion from the speech transcoder. In particular, the use of acoustic echo suppression could give rise to severe speech clipping and noise contrast. Also the operation of the voice switching used with DTX will give impairments similar to those caused by echo suppression. All subjective tests performed with echo suppressers indicate that, because of the increased effect of clipping with increased delay, the difficulty experienced by users increases rapidly with delay. According to curve 2 of figure A.1 of ITU-T Recommendation G.114 [9], the percentage of users experiencing difficulties with echo suppressors reaches 20 % with a delay of 150 ms rising to 40 % with a delay of 300 ms.
ITU-T Recommendation G.114 [9] annex A details the test conditions under which this curve was derived and it concludes that connections with more than 300 ms can only be used by very disciplined users who are aware of the problems involved in such a connection. However, recent work has indicated that delays of up to 500 ms can be used satisfactorily, provided that effective echo cancellation is incorporated in the link.

## 6.3.4 Allocation of delay to the PSMN

### 6.3.4.1 Allocation of delay to the PSMN when using a Basic Rate system

#### 6.3.4.1.1 Normal Mode Speech

Taking account of Recommendations on the separate factors described in clause 6.3.2.1, the maximum one-way delay in the PSMN between the MRP/ERP and the Point of Interconnection (see figure 5.2-1) will be 439 ms.

#### 6.3.4.1.2 Robust Mode Speech

The speech quality associated with the basic rate - robust mode system is the same as that of the basic rate - normal mode system (considering both the speech transcoder and the radio sub-system). In order to achieve the same overall transmission quality, the maximum one-way delay in the PSMN between the MRP/ERP and the Point of Interconnection (see figure 5.2-1) should be maintained at 439 ms.

### 6.3.4.2 Allocation of delay to the PSMN in case of "Single Hop" Mobile-to-Mobile Calls

The speech quality associated with the "Single Hop" Mobile-to-Mobile system is the same as that of the basic rate - normal mode system (considering both the speech transcoder and the radio sub-system). In order to achieve the same overall transmission quality, the maximum one-way delay in the PSMN between the MRP/ERP of one Mobile caller to the ERP/MRP of the second Mobile caller should be maintained at 474,4 ms, in both the basic rate - normal mode and basic rate - robust mode systems.

### 6.3.5 Delay of various network configurations

#### 6.3.5.1 National and international connections with no echo control in the PSTN (reference configurations A)

Reference configurations A (see figure 5.4.2-1) contain no echo control in the PSTN because present planning rules require the use of echo control devices only when the PSTN delay between two fixed PSTN users exceeds 25 ms. This leads to a maximum PSTN delay of 22 ms from the point of interconnection to the PSMN (see clause 6.4.2).

#### 6.3.5.2 National and international connections with echo control in the PSTN (reference configurations B)

Reference configurations B (see figure 5.4.2-2) contain echo control in the PSTN because present planning rules require their use when the PSTN delay between PSTN users exceeds 25 ms. However, action may have to be taken by administrations when planning routes or at call set-up to limit the maximum delay.

Clause 6.3.3 describes how the impairments from the harsh mobile environment when coupled with delay can give rise to difficulty. If very good cancellation of both electrical and acoustic echo can be achieved and there are no sources of speech clipping or noise contrast either in the PSMN or the PSTN part of the connection, the circuit delay should be kept below 500 ms.

PSTN routings resulting in extended path lengths and delays are likely to cause severe degradation to the quality of the connection and may result in significant difficulty. This is particularly the case when the connection contains one or maybe more satellite links.

These connections should be avoided in network planning and, if this is not possible, then the facilities of Signalling System No. 7 should be used to control the routing of the call at call set-up to minimize the effects.

### 6.3.6 Delay related requirements on the MES

#### 6.3.6.1 Basic Rate MES

##### 6.3.6.1.1 Normal Mode Speech

In accordance with the outline of transmission delay in various GMR-2 system elements contained in GMR-2 03.050, the transmit-receive delay in the MES shall not exceed 157,6 ms as defined in annex C.

##### 6.3.6.1.2 Robust Mode Speech

In accordance with the outline of transmission delay in various GMR-2 system elements contained in GMR-2 03.050, the transmit-receive delay in the MES shall not exceed 157,6 ms as defined in annex C.

## 6.4 Echo

### 6.4.1 General

There are two main sources of echo:

- acoustic echo caused by the acoustic path between receive and transmit transducers;
- electrical echo caused by coupling between the transmit and receive directions of transmission. The primary source of this form of echo is a two-to-four wire converter.

Electrical echo can be eliminated by the use of end-to-end four-wire transmission. Acoustic echo will be generated in all telephone instruments with the exception of carefully designed headsets.

In general, electrical echo is characterized by a short reverberation time and low dispersion while acoustic echo is likely to have a longer reverberation time and greater dispersion. The case of the acoustic echo may be further complicated by the time variant nature of acoustic echo which may be more severe in the mobile environment.

Curves showing the tolerance to echo, taking account of the relationship between the delay and the level of the echo, are given in ITU-T Recommendation G.131 [12] figure 2. In practice, it has been found that for any connection with a delay of greater than 25 ms, some form of echo control will be required to reduce the level of the echo (ITU-T Recommendation G.131 [12] Rule M).

With the expected maximum one-way delay in the PSMN, acoustic echo control will be required in the MES to reduce the echo returned to the distant end and electrical echo control will be required at the POI to reduce the echo returned to the PSMN user from the PSTN. The design of these echo control devices should be such as to provide operation in full duplex mode (as opposed to alternate mode).

The echo loss (EL) presented by the PSMN at the POI should be at least 46 dB during single talk. This value takes into account the fact that a MES is likely to be used in a wide range of noise environments. This requirement should be met for both handset and handportable MESs. The requirement for handsfree MESs is for further study. The test method is defined in clause B.9.

## 6.4.2 Electrical echo control in the PSMN (Reference configurations A)

The electrical echo control device at the interface with the PSTN should meet the requirements given in ITU-T Recommendation G.165 [13], but with an end delay of 60 ms. This refers to $t_d$ in clause 3.2 of ITU-T Recommendation G.165 [13]. The 60 ms is calculated as follows. ITU-T Recommendation G.131 [12] states that the maximum length of connection which need not have echo control has a mean one-way propagation time of 25 ms. However, this figure is the sum of the delays of the international connection and the maximum national delays at each end of the connection. Since the interconnection of the PSMN to the PSTN is unlikely to be at a point where the PSTN delay is > 22 ms, and the dispersion may be up to 8 ms, the maximum expected end delay which the echo canceller in the MSC should expect is:

(22 + 8) x 2 = 60 ms (see figure 6.4.2-1)

Certain countries on the geographical limits of a continent may need to increase this limit as there may be a proportion of connections which do not comply with ITU-T Recommendation G.131 [12] having a mean one-way delay of greater than 25 ms and yet are not provided with echo control.

## 6.4.3 Acoustic echo control in the PSMN

Acoustic echo control provided in the MES should provide an EL of 46 dB at the POI (see clause 6.4.1) over the likely range of acoustic end delays. If acoustic echo control is provided by voice switching, comfort noise should be injected. This comfort noise shall operate in the same way to that used in Discontinuous Transmission system (DTX). Effectively, the acoustic echo loss is provided by the MES as the GMR network is zero loss from the air interface to the POI. Hence, the 46 dB requirement shall be applied to the MES.

### 6.4.3.1 Acoustic echo control in a handsfree MES

The telephone transmission parameters for handsfree MES are for further study. However, the basic requirement of 46 dB echo loss should apply to handsfree terminals, also. If acoustic echo control is provided using some form of echo cancellation technique, the cancellation algorithm should be designed to cope with the expected reverberation and dispersion. In the case of the handsfree MES, this reverberation and dispersion may be time variant. The expected values of dispersion are under study.

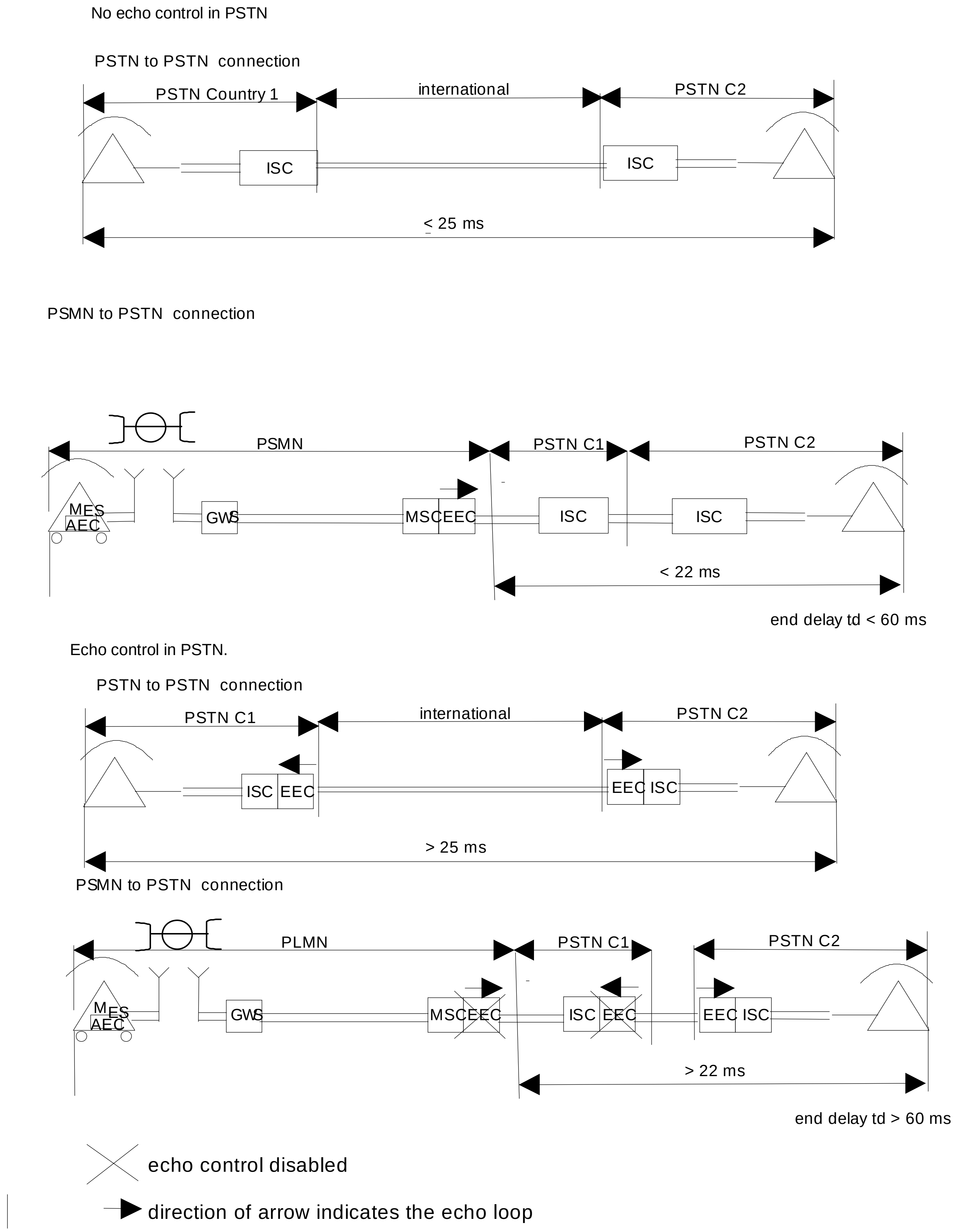



**Figure 6.4.2-1: End Delay Requirements for PSMN EEC**

#### 6.4.3.2 Acoustic echo control in a handset MES

The echo loss requirement for the handset MES shall be 46 dB. Careful acoustic design of the handset body and selection of the mouth and earpiece transducers may facilitate the required acoustic echo loss without the need for active echo control techniques. However, should echo cancellation be employed the echo canceller should be capable of dealing with the variations in handset positions when in normal use. The implication of this is under study.

#### 6.4.3.3 Acoustic echo control in a headset MES

The echo loss requirement for a headset shall be 46 dB. Due to the obstacle effect of the head in this type of terminal, careful design might mean that no active echo control is necessary.

### 6.4.4 Interaction between tandem echo control devices (reference configurations B)

On long international routes or routes containing a satellite path, network echo control devices will be present in accordance with ITU-T Recommendation G.131 [12], Rule M. These devices will be echo suppressers or echo cancellers generally with centre clippers. The tandem connection of such devices can lead to increased clipping and, if echo suppressors are used, additional loss. It is recommended that signalling or routing means be used to avoid the tandem connections of echo control devices whenever possible (see figures 5.4.2-2 and 6.4.2-1).

## 6.5 Clipping

### 6.5.1 General

The loss of the start or the end of a speech burst is known as clipping, the main cause of which is voice switching controlled by voice activity detection. Voice switching occurs in devices within the network or within terminal devices. The following devices employ voice switching:

- echo suppressors. These are generally located at an ISC at either end of a long international connection or connections using satellites;
- echo cancellers with centre clippers. These are located as for the echo suppressors above. In addition, it is recommended that they be used in the MSC at the interface with the PSTN. Clipping in these devices arises from the action of the centre clipper only;
- digital speech interpolators (DSI). These devices are used in circuit multiplication equipment which are often employed on international connections;
- discontinuous transmission (DTX) devices. These are located in the PSMN;
- loudspeaking telephones. These are used in the PSTN and in the PSMN. It should be noted that regulations in certain countries prohibit the use of handheld MESs by drivers of moving vehicles.

### 6.5.2 Properties of voice switches in the PSMN

Recommendation GSM 06.32 [4] specifies the requirements for the voice activity detector used for DTX and the total clipping allowed in the MES. Any voice switching used for acoustic echo control should not exceed these limits. Information on recommended characteristics of handsfree telephones is given in ITU-T Recommendation P.340 [24], clause 5.

### 6.5.3 Problems of tandem voice switching

The effect of tandem voice switches, which are not under one common control, will be an increase in clipping. Moreover, under conditions of high or rapidly changing ambient noise, false detection of speech is likely to occur in the voice activity detectors in DSI equipment or network echo control devices. These devices are generally designed for constant and low levels of noise.

In order to minimize clipping, the following action should be taken:

- intermediate tandem voice switching devices in the network should be either disabled by signalling means or avoided by routing means;
- the voice switching for the MES for acoustic control and for DTX should be under one common control.

However, it should be noted that, in many cases, it will not be possible to exclude DSI equipment or loudspeaking telephones from the connection.

## 6.6 Idle channel noise

### 6.6.1 Sending

The maximum noise level produced by the apparatus at the UPCMI under silent conditions in the sending direction shall not exceed -64 dBm0p.

NOTE 1: This level includes the eventual noise contribution of an acoustic echo canceller under the condition that no signal is received.

NOTE 2 This figure applies to the wideband noise signal. It is recommended that the level of single frequency disturbances should be 10 dB lower (ITU-T Recommendation P.11 [23]).

Compliance shall be checked by the test described in clause B.2.1.

### 6.6.2 Receiving

The maximum (acoustic) noise level at the handset MES when no signal (O-level) is received from the speech transcoder shall be as follows:

if no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user-controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured in the artificial ear contributed by the receiving equipment alone shall not exceed -57 dBPa(A) when driven by a PCM signal corresponding to the decoder output value number 1;

where a volume control is provided, the measured noise shall also not exceed -54 dBPa(A) at the maximum setting of the volume control.

NOTE: In a connection with the PSTN, noise conditions as described in ITU-T Recommendation G.103 [6] can be expected at the input (POI) of the PSMN. The characteristics of this noise may be influenced by the speech transcoding process (for further study).

Compliance shall be checked by the test described in clause B.2.2.

# 6.7 Noise contrast

## 6.7.1 General

On any PSMN call there is likely to be continuous background noise which is present regardless of whether the users are talking or not. There may also be one or more voice-operated devices; these effectively break the circuit when there is no speech on it.

Noise contrast problems are caused by the background noise being interrupted when the circuit is broken so that the user listening on the circuit hears the background noise being continually switched on and off. This is particularly disturbing for a user talking to a PSMN user in a moving vehicle because the background noise being modulated in this way is at a very high level. In this situation, it has been found that speech intelligibility can be impaired.

The main sources of background noise are:

- background acoustic noise picked up by the microphone. For a loudspeaking telephone in a moving vehicle the speech/noise ratio can be as low as 0 dB;
- idle channel noise. This includes noise generated in the transmission system (thermal noise and crosstalk) the switching system and in speech transcoders.

## 6.7.2 Elements of a PSMN which can cause noise contrast impairment

The following elements can cause noise contrast impairments:

- the acoustic echo control device in the MES. A moving vehicle presents a very difficult environment for an echo canceller, so an echo suppressor is likely to be used (possibly in conjunction with an echo canceller). Echo suppressors contain voice-operated switches;
- DTX. The transmitter switching will cause a PSTN user talking to a PSMN user to hear modulation of the mobile background noise. It will also cause the PSMN user to hear modulation of the PSTN noise. The PSTN noise will vary from connection to connection and should decrease in the future with increasing network digitalization;
- the electric echo control devices protecting the PSMN user against echo returned from the PSTN. The centre clipper in this echo canceller will cause some noise modulation.

## 6.7.3 Reduction of noise contrast

A reduction in noise contrast:

- reduces conversational difficulty, particularly for long conversations;
- allows a greater tolerance on the matching of the level and spectrum of the comfort noise to the ambient noise.

NOTE: Preliminary tests in vehicles indicate that, in a constant noise environment with a hands-free MES and a signal-to-noise ratio of approximately 10 dB, a maximum level mismatch of 2 dB can be tolerated. The comfort noise spectrum was a reconstruction of the averaged medium term ambient noise spectrum.

### 6.7.3.1 Reduction of noise contrast by limiting the noise received by the microphone

The characteristics of the ambient noise (spectrum and level) depend on the environment in which the MES is used. As a microphone is characterized by its sensitivity and directivity, only part of this noise will enter the microphone.

A general principle for reducing noise contrast is to maximize the signal-to-noise ratio at the microphone input. This can be achieved by simultaneously increasing directivity, reducing sensitivity, and placing the microphone close to the mouth of the talker. Consequently, the implementation of the acoustic terminal will significantly affect the dynamic range of the noise contrast.

#### 6.7.3.1.1 Headset MES

In the case of a headset and if DTX is disabled, then noise contrast will not be present since acoustic echo control (with centre clipping) is not required. If DTX is enabled, then only a small amount of noise contrast might result since the microphone would be close to the talker's mouth and would follow the movement of the talker's head, thus fulfilling the general principle described above. In the worst case, the headset is likely to give a minimum of 15 dB signal-to-noise ratio. (This value is for further study).

#### 6.7.3.1.2 Handset MES

In the case of a handset, and if DTX is disabled, then noise contrast will not be present if optimized echo cancelling techniques (without residual echo clipping) are used to control the acoustic echo (providing 46 dB EL). If DTX is enabled or acoustic echo control with centre clipping is used, then only a small amount of noise contrast might result since the microphone would be close to the talker's mouth and would follow the movement of the talker's head, thus fulfilling the general principle described above. In the worst case, the handset is likely to give a minimum of 15 dB signal-to-noise ratio. (This value is for further study).

#### 6.7.3.1.3 Handsfree MES

In the case of a handsfree telephone and even if DTX is disabled, noise contrast will be introduced unless 46 dB EL can be provided without the use of centre clipping. This is unlikely to be achievable. As the microphone is distant from the talker's mouth, and as the talker may be moving during the conversation, the sensitivity of the microphone has to be high and directivity low. This could result in a worse case signal-to-noise ratio of 0 dB. (This value is for further study).

The following is given as interim guidelines. In the case of a vehicle mounted handsfree MES, the characteristics of the microphone should be such as to limit the change in speech level to 5 dB for all positions of the talker while sitting.

### 6.7.3.2 Reduction of noise contrast by insertion of comfort noise

GSM 06.12 [3] specifies comfort noise to be used both for acoustic echo control with centre clipping and DTX.

## 6.7.4 Consequence of the introduction of high comfort noise levels on other voice-operated devices

Two problems associated with other voice switching devices (e.g. DSI) may result from the introduction of high levels of comfort noise:

- the high comfort noise level may be interpreted as a voice signal;
- if the high level of comfort noise is detected as noise, then another source of comfort noise at a different level may be introduced downstream, thus increasing the noise contrast.

# 6.8 Sensitivity/frequency characteristics

## 6.8.1 Headset and Handset MESs

### 6.8.1.1 Sending

The sensitivity/frequency characteristics from MRP to the UPCMI shall be as follows:

- the sending sensitivity - frequency response (from MRP to digital interface) shall be within a mask which can be drawn between the points given in table 1. The mask is drawn with straight lines between the breaking points in table 1 on a logarithmic (frequency) - linear (dB sensitivity) scale.

**Table 1: Sending sensitivity/frequency mask**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | -12 | - |
| 200 | 0 | - |
| 300 | 0 | -12 |
| 1 000 | 0 | -6 |
| 2 000 | 4 | -6 |
| 3 000 | 4 | -6 |
| 3 400 | 4 | -9 |
| 4 000 | 0 | - |
| NOTE: All sensitivity values are dB on an arbitrary scale. | | |

Compliance shall be checked by the test described in clause B.3.1.

#### 6.8.1.2 Receiving

The sensitivity/frequency characteristics from the UPCMI to the ERP shall be as follows:

The receiving sensitivity - frequency response (from the digital interface to the ERP) shall be within the mask which can be drawn with straight lines between the breaking points in table 2 on a logarithmic (frequency) - linear (dB sensitivity) scale.

**Table 2: Receiving sensitivity/frequency mask**

| Frequency (Hz) | Upper limit | Lower limit |
|---|---|---|
| 100 | -12 | - |
| 200 | 0 | - |
| 300 | 2 | -7 |
| 500 | (note 2) | -5 |
| 1 000 | 0 | -5 |
| 3 000 | 2 | -5 |
| 3 400 | 2 | -10 |
| 4 000 | 2 | - |
| NOTE 1: All sensitivity values are dB on an arbitrary scale.<br>NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale. | | |

Compliance shall be checked by the test described in clause B.3.2.

### 6.8.2 Handsfree MES

Recommended sensitivity/frequency characteristics curves for handsfree terminals are given in clause 4 of ITU-T Recommendation P.340 [24].

## 6.9 Distortion

### 6.9.1 Sending

The sending part between MRP and the UPCMI shall meet the following distortion requirements:

NOTE: Digital signal processing other than the transcoder itself is included in this requirement (e.g. echo cancelling).

The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 3 of ITU-T Recommendation G.223 [14]) shall be above the limits given in table 3 unless the sound pressure at MRP exceeds +10 dBPa.

**Table 3: Limits for signal-to-total distortion ratio**

| SENDING level dB relative to ARL | Sending Ratio (dB) |
|---|---|
| -35 | 17,5 |
| -30 | 22,5 |
| -20 | 30,7 |
| -10 | 33,3 |
| 0 | 33,7 |
| +7 | 31,7 |
| +10 | 25,5 |

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

Compliance of the sending distortion shall be checked by the test described in clause B.4.1.

The gain variation relative to the gain for ARL shall remain within the limits given in table 4. For intermediate levels, the same limits for gain variation apply.

**Table 4: Variation of gain with input level, sending**

| SENDING level dB relative to ARL | SENDING level dB relative to ARL | Lower limit (dB) |
|---|---|---|
| 13 | 0,5 | -0,5 |
| 0 | 0,5 | -0,5 |
| -20 | 0,5 | -0,5 |
| -20 | 0,5 | -2 |
| -30 | 0,5 | -2 |
| -30 | 1 | - |
| -40 | 1 | - |
| -40 | 2 | - |
| -45 | 2 | - |

Compliance of the gain variation in the sending direction shall be checked by the test described in clause B.5.1.

## 6.9.2 Receiving

The receiving part between the UPCMI and ERP shall meet the requirements in this clause at the nominal setting of the volume control:

The ratio of signal-to-total distortion power measured with the proper noise weighting (see ITU-T Recommendation G.223 [14], table 4) shall be above the limits given in table 5 when the sound pressure at ERP is up to +10 dBPa. For sound pressures exceeding +10 dBPa at the ERP there is no distortion requirement.

**Table 5: Limits for signal-to-total distortion ratio**

| RECEIVING level at the digital interface (dBm0) | Receiving Ratio (dB) |
|---|---|
| -45 | 17,5 |
| -40 | 22,5 |
| -30 | 30,5 |
| -20 | 33,0 |
| -10 | 33,5 |
| -3 | 31,2 |
| 0 | 25,5 |

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

Compliance of the receiving distortion shall be checked by the test described in clause B.4.2.

The gain variation relative to the gain at an input level of -10 dBm0 shall be within the limits given in table 6. For intermediate levels, the same limits for gain variation apply. The requirement is valid for sound pressures up to +10 dBPa. For sound pressures exceeding +10 dBPa at the ERP there is no lower limit requirement.

**Table 6: Variation of gain with input level, receiving**

| RECEIVING level at the digital interface (dBm0) | Upper limit (dB) | Lower limit (dB) |
|---|---|---|
| +3 | 0,5 | -0,5 |
| -10 | 0,5 | -0,5 |
| -40 | 0,5 | -0,5 |
| -40 | 1 | -1 |
| -50 | 1 | -1 |
| -50 | 2 | -2 |

Compliance of the gain variation in the receiving direction shall be checked by the test described in clause B.5.2.

# 6.10 Sidetone

## 6.10.1 Sidetone loss

A sidetone requirement is appropriate for MESs using handsets and headsets. There are separate requirements for listener sidetone (LSTR) and talker sidetone (STMR). The listener sidetone performance is considered as the major parameter affecting the user perception of the system. Though talker sidetone is important to give the user some comfort in using the equipment. A value for D is recommended which is the discrimination of the sending section of speech over noise.

The value of the Listener Sidetone Rating (LSTR) shall not be less than 15 dB. Where a user-controlled receiving volume control is provided, the LSTR shall meet the requirement given above at the setting where the RLR is equal to the nominal value.

Compliance of the LSTR requirement shall be checked by the test described in clause B.6.2.

The nominal value of the Sidetone Masking Rating (STMR) shall be 13 dB ± 5 dB. Where a user-controlled receiving volume control is provided, the STMR shall meet the requirement given above at the setting where the RLR is equal to the nominal value.

Compliance of STMR requirement shall be checked by the test described in clause B.6.1.

It is recommended that the STMR is independent of the volume control.

It is recommended, but not required, that the sending path is designed so that the value of D, defined in ITU-T recommendation G.111 [7], should not be less than 2 dB, where LSTR = STMR + D.

## 6.10.2 Sidetone distortion

The third harmonic distortion generated by the terminal equipment shall not be greater than 10 %.

Compliance shall be checked by the test described in clause B.7.

# 6.11 Out-of-band signals

## 6.11.1 Discrimination against out-of-band input signals

When out-of-band signals are applied at the MRP, a range of frequencies will be transmitted to the UPCMI. For these signals, the following requirements shall apply.

With any sine-wave signal above 4,6 kHz and up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (-4,7 dBPa at MRP) by at least the amount (in dB) specified in table 7.

**Table 7: Discrimination levels**

| Applied sine-wave frequency | Limit (minimum) (note) |
|---|---|
| 4,6 kHz<br>8 kHz | 30 dB<br>40 dB |
| NOTE: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale. | |

Compliance shall be checked by the test described in clause B.8.1.

### 6.11.2 Spurious out-of-band signals

The level of out-of-band signals at the ERP shall meet the following requirements when the relevant input signals are simulated at the UPCMI.

With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3,4 kHz and at a level of 0 dBm applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4,6 to 8 kHz measured selectively in the artificial ear shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 8.

**Table 8: Discrimination levels**

| Image Signal frequency | Equivalent Input Signal Level (note) |
|---|---|
| 4,6 kHz<br>8 kHz | -35 dBm0<br>-45 dBm0 |
| NOTE: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale. | |

Compliance shall be checked by the test described in clause B.8.2.

## 6.12 Requirements for information tones

The PSMN should be capable of transmitting information tones generated by the PSTN in the range 300 Hz to 1 800 Hz conforming to ITU-T Recommendation E.180/Q.35 [33], Reference 3.

## 6.13 Crosstalk

### 6.13.1 Near and far end crosstalk

The near end or far end crosstalk ratio between two complete PSMN connections should not be less than 65 dB.

### 6.13.2 Go/return crosstalk

The crosstalk ratio between the go and return channels of a single PSMN connection should not be less than 55 dB. This is to avoid nullifying the effect of the electrical echo canceller at the MSC. The requirement applies for an acoustic input signal at the MRP with a measurement being made at the UPCMI in the opposite direction of transmission.

# Annex A (informative): Considerations on the Acoustic Interface of the Mobile Station

## A.1 Handsfree MES

The handsfree MES will almost certainly require the use of non-linear processing for the acoustic echo control, the extraction of the speech from high levels of ambient noise. The implementation of these functions may well cause degradation to the overall transmission quality and cause difficulty to the distant end user especially during duplex conversation.

## A.2 Handset MES

The handset MES, depending on the detailed implementation, might not require the use of non-linear processing for the acoustic echo control. Also, the position of the microphone should give a significantly improved signal-to-noise ratio, compared with a handsfree MES especially in noisy environments. This is likely to result in significantly improved transmission quality compared with the handsfree MES and easier duplex conversation.

## A.3 Headset MES

The headset MES is likely to be the simplest, since with careful design, it might not require acoustic echo control. As with the handset case, the signal-to-noise ratio should be significantly improved compared with the handsfree MES especially in noisy environments. Consequently, the headset MES is likely to give the best transmission quality and easiest duplex conversation.

## A.4 Inter-reaction with DTX

Because of the improved signal-to-noise ratio, both the headset MES and the handset MES are likely to give better transmission quality when DTX is enabled than that of handsfree MES.

# Annex B (normative): Transmission requirements testing

When an artificial ear is required, the ITU-T Recommendation P.57 [29] Type 1 artificial ear shall be used.

If requested by the terminal supplier, the ITU-T Recommendation P.57 [29] Type 3.2 artificial ear shall be used for all tests. In this case the following apply:

- the low leakage option of Type 3.2 artificial ear shall be adopted;
- the force against the ear shall be as specified in ITU-T Recommendation P.57 [29];
- sound pressure measurements shall be referred to the ERP as specified in ITU-T Recommendation P.57 [29];
- no leakage correction shall be made in the calculation of RLR (i.e. $L_E = 0$).

The test report shall indicate the type of artificial ear used for the receive tests.

# B.1 Loudness ratings

## B.1.1 Sending Loudness Rating (SLR)

a) The sending sensitivity shall be measured at each of the 14 frequencies given in ITU-T Recommendation P.79 [32], table 2, bands 4 to 17.

b) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [32], formula 4.19b, over bands 4 to 17, and using the sending weighting factors from ITU-T Recommendation P.79 [32], table 2, adjusted according to table 3 of this Recommendation.

## B.1.2 Receiving Loudness Rating (RLR)

a) The receiving sensitivity shall be measured at each of the 14 frequencies listed in ITU-T Recommendation P.79 [32], table 2, bands 4 to 17.

b) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [32], formula 4.19c, over bands 4 to 17, using the receiving weighting factors from table 2 of this Recommendation, adjusted according to table 3 of this Recommendation.

c) The artificial ear sensitivity shall be corrected using the real ear correction of ITU-T Recommendation P.79 [32], table 4.

NOTE: The value of real ear correction of ITU-T Recommendation P.79 [32], table 4 were derived for one type of handset conforming to the shape defined in ITU-T Recommendation P.35 [25].

These values are used in the present document because there is no measurement method agreed for the real ear correction. If a method of measurement is agreed, it is intended to change the present document to use the values appropriate to each handset.

# B.2 Idle Channel Noise

## B.2.1 Sending

With the handset mounted at LRGP and the earpiece sealed to the knife-edge of the artificial ear in a quiet environment (ambient noise less than 30 dBA), the noise level at the digital output is measured with apparatus including psophometric weighting according to ITU-T Recommendation G.223 [14], table 4.

NOTE: The ambient noise criterion should be met if the ambient noise does not exceed NR20.

## B.2.2 Receiving

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A signal corresponding to decoder output value number 1 shall be applied at the digital interface. The level of the noise shall be measured in the artificial ear.

The ambient noise for this measurement shall not exceed 30 dBA.

# B.3 Sensitivity/frequency Characteristics

## B.3.1 Sending

a) The handset is mounted in the LRGP (see annex A of ITU-T Recommendation P.76 [31]). The earpiece is sealed to the knife-edge of an artificial ear.

b) A pure tone signal with a sound level of -4,7 dBPa (in accordance with ITU-T Recommendation P.64 [30] shall be applied at the MRP as described in ITU-T Recommendation P.64 [30], using an artificial mouth conforming to ITU-T Recommendation P.51 [28].

c) A digital measuring instrument, or high quality digital decoder followed by an analogue level measuring set, shall be connected at the interface.

d) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [35] for frequencies from 100 Hz to 4 kHz inclusive.

At each frequency, the output level for a sound pressure of -4,7 dBPa shall be measured.

## B.3.2 Receiving

a) The handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

b) A digital signal generator shall be connected at the digital interface delivering a signal equivalent to a pure tone level of -16 dBm0, see ITU-T Recommendation P.64 [30].

c) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [35] for frequencies from 100 Hz to 4 kHz inclusive.

At each frequency, the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

# B.4 Distortion

## B.4.1 Sending

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz is applied at the MRP.

The level of this signal is adjusted until the output of the terminal is -10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels: -35, -30, -25, -20, -15, -10, -5, 0, 5, 10 dB relative to ARL.

The ratio of the signal to total distortion power of the digital signal output shall be measured with the psophometric noise weighting (see ITU-T Recommendations G.712 [17] and O.132 [22]).

## B.4.2 Receiving

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A digitally simulated sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz shall be applied at the digital interface at the following levels: -45, -40, - 35, -30, -25, -20, -15, -10, -5, 0 dBm0.

The ratio of the signal-to-total distortion power shall be measured with the psophometric noise weighting in the artificial ear (see ITU-T Recommendations G.712 [17] and O.132 [22]).

# B.5 Variation of gain with input level

## B.5.1 Sending

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz is applied at the MRP. The level of this signal is adjusted until the output of the terminal is -10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels: -45, -40, -35, -30, -25, -20, -15, -10, -5, 0, 5, 10, 13 dB relative to ARL.

The variation of gain relative to the gain for ARL is measured.

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

## B.5.2 Receiving

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A digitally simulated sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz shall be applied at the digital interface at the following levels: -50, -45, -40, -35, -30, -25, -20, -15, -10, -5, 0, 3 dBm0.

The variation of gain relative to the gain at an input level of -10 dBm0 shall be measured in the artificial ear.

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

# B.6 Sidetone

For the tests described in clauses B.6.1 and B.6.2, the digital input of the terminal shall be driven by a PCM signal corresponding to decoder value number 1.

## B.6.1 Talker sidetone (STMR)

a) The handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A pure tone signal of - 4,7 dBPa shall be applied at the mouth reference point. For each frequency given in ITU-T Recommendation P.79 [32], table 2, bands 4 to 17, the sound pressure in the artificial ear shall be measured.

b) The sidetone path loss LmeST as expressed in dB and the STMR (in dB) shall be calculated from ITU-T Recommendation P.79 [32], formula 8-4 , using the weighting factors of column (3) in table 6 of this Recommendation (unsealed), and values of LE in accordance with ITU-T Recommendation P.79 [32], table 4.

## B.6.2 Listener sidetone (LSTR)

a) The sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +4 dB/-2 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20).

b) A calibrated half-inch microphone is mounted at MRP. The sound field is measured in one-third octave bands. The spectrum shall be "Pink noise" as described in ITU-T recommendation P.64 [30], annex B to within ±1 dB and the level shall be adjusted to 70 dBA (-24 dBPa(A)). The tolerance on this level is ±1 dB.

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

d) Measurements are made in one-third octave bands for the 14 bands centred at 200 Hz to 4 kHz (bands 4 to 17). For each band the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

e) The listener sidetone path loss is expressed in dB and the LSTR shall be calculated from the ITU-T Recommendation P.79 [32], formula 8-4, using the weighting factors in column (3) in table 6 of the Recommendation, and the values of LE; in accordance with table 4 of the Recommendation.

# B.7 Sidetone distortion

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear. An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range of 315 Hz to 1 kHz is connected to the artificial ear.

A pure-tone signal of -4,7 dBPa is applied at the mouth reference point at frequencies of 315 Hz, 500 Hz, and 1 kHz. For each frequency, the third harmonic distortion shall be measured in the artificial ear.

# B.8 Out-of-band signals

## B.8.1 Discrimination against out-of-band input signal

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

For input signals at frequencies of 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz at the level specified in clause 3.11.1, the level of any image frequencies at the digital interface shall be measured.

### B.8.2 Spurious out-of-band signals

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

For input signals at the frequencies 500, 1 000, 2 000, and 3 150 Hz applied at the level specified in clause 3.11.2, the level of spurious out-of-band image signals at frequencies of up to 8 kHz shall be measured selectively in the artificial ear.

## B.9 Acoustic echo loss

The MES is mounted at the Loudness Rating Guarding Position (LRGP), (see ITU-T Recommendation P.76 [31], annex 1), with the earpiece sealed to the knife-edge of the artificial ear, conforming to ITU-T Recommendation P.57 [29].

A call is set up between the MES and the SS.

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

An implementation of the ITU-T Recommendation P.50 [27] artificial speech shall be connected to the analogue or digital input of the reference speech encoder of the SS. This implementation could be a real time algorithm producing the artificial speech or a pre-recorded tape of the artificial speech. Both "male" and "female" artificial speech is required.

A ten second segment of the "male" artificial speech is applied to the analogue or digital input of the reference speech encoder of the SS. The third octave power of the input signal shall be measured. The echo loss signal is not measured at this stage as the first ten-second segment is used to allow any acoustic echo cancellation devices within the MES to adapt to the echo path.

Immediately afterwards a second ten second segment of the "male" artificial speech is applied to the analogue or digital input of the reference speech encoder of the SS. The third octave power of the echo signal is measured at the digital output of the reference speech decoder of the SS.

The difference between the third octave input power and the third octave output power is entered into the ITU-T Recommendation G.122 [11] TCL algorithm and the acoustic echo loss calculated.

The test shall be repeated with the "female" artificial speech and the results of both "male" and "female" averaged to give the final result.

# Annex C (normative): MES delay requirement definition

## C.1 Basic rate MES delay requirement definition

The symbol definitions for the calculations in this annex are:

Trftx: The time required for transmission of a TCH radio interface frame over the air interface due to the interleaving and de-interleaving (given in table 4/2) (system dependent) and multiplexing.

Ttransc: The speech encoder processing time, from input of the last PCM sample to output of the final encoded bit (implementation dependent).

Tsample: The duration of the segment of PCM speech operated on by the speech transcoder.

Tencode: The time required for the channel encoder to perform channel encoding (implementation dependent).

Trxproc: The time required after reception over the radio interface to perform equalization, channel decoding and SID-frame detection (implementation dependent).

Tproc: The time required after reception of the first coded sample to process the speech encoded data for the basic-rate speech decoder and to produce the first PCM output sample (implementation dependent).

Ta/d Delay in the analogue to digital converter in the downlink.

Tmargin An allowance for system entities that are implementation dependant.

Td/a Delay in the digital to analogue converter in the uplink.

Tpcm The duration of a segment of PCM speech for the downlink processing delay.

The basic rate MES speech delay in the uplink direction is the delay between an acoustic event at the MRP to the last bit of the corresponding speech frame at the antenna connector and shall not exceed:

MES uplink delay = Ta/d + Tmargin + Tsample + Ttransc + Tencode + Trftx

= 0,0 ms+ 10,0 ms+ 60,0 ms+ 17,0 ms+ 1,6 ms+ 40,0 ms

= 128,6 ms.

The basic rate MES speech delay in the downlink direction is the delay between the first bit of a speech frame at the antenna connector and the last acoustic event at the ERP corresponding to that speech frame and shall not exceed:

MES downlink delay = Tpcm + Trftx + Trxproc + Tproc + Tmargin + Td/a

= 12 ms+ 0,0 ms+ 3,4 ms+ 3,6 ms+ 10,0 ms+ 0,0 ms

= 29,0 ms.

The round trip delay shall therefore not exceed:

128,6 ms + 29,0 ms = 157,6 ms.

# Annex D (informative): PSMN delay requirement definition

## D.1 Spacecraft delay definition

The symbol definitions for the calculations in this annex are:

Tforward: The maximum one-way delay through the channelizer filter in the forward case.

Treturn: The maximum one-way delay through the channelizer filter in the return case.

Tmobile: The maximum one-way delay through the channelizer filter for the "Single Hop" mobile-to-mobile case (including buffering for TDMA processing)

Tpropagation: The maximum one-way delay for free space propagation from the Earth to the satellite back to Earth.

The Spacecraft delay between the MES and the PSTN POI shall not exceed:

Spacecraft delay = Treturn + Tpropagation

= 0,2 ms+ 266,8 ms

= 267,0 ms.

The Spacecraft delay between two MES shall not exceed:

Spacecraft delay = Tmobile + Tpropagation

= 50,0 ms+ 266,8 ms

= 316,8 ms.

## D.2 Gateway delay definition

The delay through the Gateway on the Transmit side shall not exceed:

- GW Transmit delay = 143,0 ms.

The delay through the Gateway on the Receive side shall not exceed:

- GW Receive delay = 43,0 ms.

The round trip delay shall therefore not exceed:

- 143,0 ms + 43,0 ms = 186,0 ms.

# D.3 Total delay definition

The maximum one-way delay from the MES to the PSTN POI shall not exceed:

- Mobile-to-PSTN delay = 128,6 ms + 267,0 ms + 43,0 ms = 438,6 ms.

The maximum one-way delay from the PSTN POI to the MES shall not exceed:

- PSTN-to-Mobile delay = 143,0 ms + 267 ms + 29,0 ms = 439,0 ms.

The maximum one-way delay between two MES shall not exceed:

- Mobile-to-Mobile delay = 128,6 ms + 316,8 ms + 29,0 ms = 474,4 ms.

# History

| Document history | | |
|---|---|---|
| V1.1.1 | March 2001 | Publication |
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