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TRANSMISSION SYSTEMS AND MEDIA

TRANSMISSION PLAN ASPECTS OF SPECIAL CIRCUITS AND CONNECTIONS USING THE INTERNATIONAL TELEPHONE CONNECTION NETWORK

TRANSMISSION PLANNING ASPECTS OF THE SPEECH SERVICE IN DIGITAL PUBLIC LAND MOBILE NETWORKS

ITU-T Recommendation G.173

(Previously "CCITT Recommendation")

FOREWORD

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

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

ITU-T Recommendation G.173 was prepared by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

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 Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

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

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

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TRANSMISSION PLANNING ASPECTS OF THE SPEECH SERVICE IN DIGITAL PUBLIC LAND MOBILE NETWORKS

(Helsinki, 1993)

1 Introduction

The purpose of this Recommendation is to describe the special transmission planning aspects of digital public land mobile networks (PLMN). This Recommendation is limited to PLMN implemented with terrestrial-based facilities only. Mobile networks that include satellite links are not covered by this Recommendation, but are under separate consideration. In addition, this Recommendation provides guidelines and advice to Administrations and recognized network operators as to the kind of precautions, measures and minimum requirements which are needed for a successful interworking of digital public land mobile networks with the national and international PSTN and ISDN. Transmission planning aspects of analogue public land mobile networks are not covered in this Recommendation.

The performance objectives of digital public land mobile networks are to reach a quality as close as possible to CCITT Recommendations. Due to the technical and economical factors of such mobile networks a full compliance with the general characteristics of international telephone connections and circuits recommended by CCITT is not guaranteed on all connections.

The interworking between a public land mobile network and the PSTN/ISDN, especially the signalling system between both networks is described in Recommendation Q.70.

A summary of transmission requirements for digital public land mobile systems is presented below. Annex A provides a more complete discussion of definitions and factors which may prevent full compliance with CCITT Recommendations. Annex B gives examples of PLMN and ISDN/PSTN interconnections and configurations, Supplement No. 32 (to this Recommendation) provides information on characteristics of specific systems.

2 Transmission performance

A model of a connection between a PLMN and the PSTN/ISDN is shown in Figure 1. A more detailed model of PLMN and the relevant interfaces are described in Annex A.

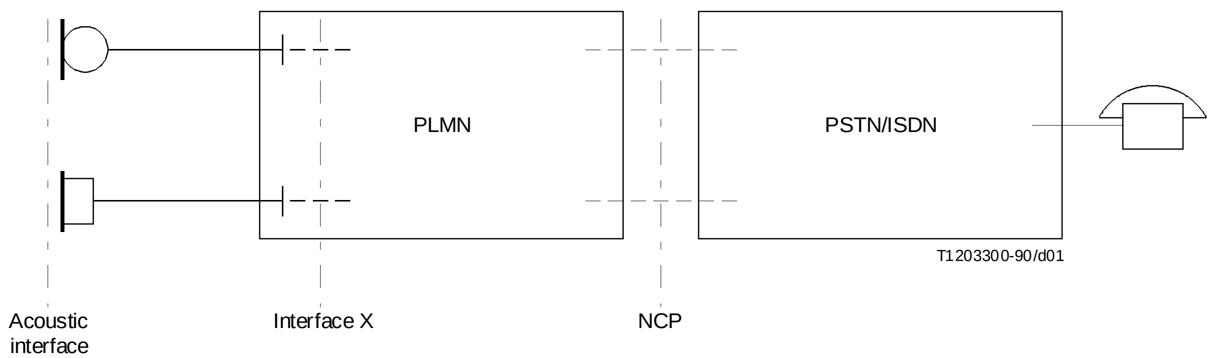


FIGURE 1/G.173

Connection between a PLMN and PSTN/ISDN

The overall transmission performance of a connection from the PLMN to the PSTN/ISDN can be considered as a summation of the effects of:

- the audio part between the acoustic interface and interface X;
- the speech transcoder part including the effects of radio transmission and speech processing between interface X and the network connection point (NCP), see A.1.5;
- the overall characteristics of the connection between the NCP 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.

However, in several cases the main determining factors are the characteristics of the audio part when the additional call path in the PLMN is digital.

Where possible, the transmission performance is specified between the acoustic interface and interface X. The transmission aspects of the whole digital mobile system depend on the technical design and may be different from network operator to network operator. To control the end-to-end transmission quality recommended by CCITT the requirements given below should be met by the PLMN.

2.1 Loudness ratings

The recommended values for the loudness ratings are:

	Short-term objectives	Long-term objectives
SLR	7 dB to 9 dB	8 dB
RLR	1 dB to 3 dB	2 dB

These values are directly applicable to the case of a mobile station operating in a conventional non-mobile noise environment. Studies have shown that in a PLMN noise environment, speech levels are likely to be higher. Hence, in order to avoid clipping in the speech transcoder it may be necessary to increase the value of the SLR.

The loudness rating values should be measured between the acoustic interface and the interface X. In the usual case where the PLMN introduces no additional loss or attenuation distortion between interface X and the NCP, the loudness ratings up to the PSTN/ISDN will be the same as loudness ratings measured at interface X. For more details see A.3.1.

2.2 Stability

The stability requirements will be met if the attenuation between the digital input and digital output of the PLMN at the NCP is at least 6 dB. For more details see A.3.2.

2.3 Propagation time

The following values are given for the mean one-way propagation time of the PLMN:

recommended value < 40 ms (see Note).

NOTE – Technical and practical considerations do not permit the recommended value to be achieved at this time. Existing PLMNs should provide acceptable service on many connections but compliance with Recommendations G.114 and G.131 may not be assured on all connections (see A.3.3).

2.4 Echo control

Due to the high propagation time, echo control is necessary within the PLMN. For reducing the PSTN/ISDN network echo, an echo control device is needed at the NCP. The PLMN echo loss at the NCP shall be greater than 45 dB. If there is no additional loss in the PLMN, the terminal coupling loss of the MS has to fulfil the echo requirements.

For discussion related to connections involving PLMN echo control devices and PSTN/ISDN echo control devices, see A.3.4. Subclause A.3.4 also included discussion on PLMN echo loss objectives for the evolving PSTN/ISDN.

Technical and practical considerations do not permit the recommended value to be achieved at this time. According to Recommendation P.31 the weighted terminal coupling loss (TCL_W) of a digital telephone should be greater than 40 dB.

The characteristics of the PLMN, including the PLMN canceller used to cancel PSTN/ISDN network echoes, must not compromise the operation of network based G.165 cancellers.

2.5 Quantizing distortion

For the quantizing distortion of the PLMN in error free conditions between the acoustic interface and the NCP, the following values are given:

recommended value < 4 qdu (see Note 1).

The qdu of PLMN under realistic bit-error-rate conditions (conditions to be determined):

recommended value < 7 qdu (see Note 1).

NOTES

1 Technical and practical considerations do not permit the recommended values to be achieved at this time by all systems. Existing PLMNs should provide acceptable service on many connections but compliance with Recommendation G.113 may not be assured on all connections (see A.3.5).

2 It should be noted that in some cases such as low bit rate (≤ 8 kbit/s) non-waveform codecs, the concept of quantizing distortion units (qdu) may not apply.

2.6 Clipping

For clipping no specific requirement is given. For more information see A.3.6.

2.7 Noise contrast

For noise contrast no specific requirement is given. For more information see A.3.7.

2.8 Other requirements

The PLMN should be designed to be in accordance with relevant CCITT Recommendations concerning:

- sidetone;
- noise;
- sensitivity/frequency characteristics;
- distortion;
- out-of-band signals;
- go/return crosstalk.

The transmission requirements of handset MS should be derived from the requirements of digital telephones stated in Recommendation P.31. For headset telephones, information can be found in Recommendation P.38 and for handsfree telephones using loudspeakers, see Recommendation P.34.

Annex A

Definitions and detailed transmission aspects

(This annex forms an integral part of this Recommendation)

A.1 Definitions

For the purpose of this Recommendation, the following definitions apply:

A.1.1 base station (BS): A base station (BS) is a radio transmitter/receiver station in a PLMN which provides the radio transmission path to the mobile station. Several base stations are connected via leased lines to a mobile services switching centre.

A.1.2 digital mobile system: The basic configuration of a digital mobile system is shown in Figure A.1. A digital mobile system consists of the mobile station, radio transmission path, base station, leased line and the mobile services switching centre up to the network connection point (NCP).

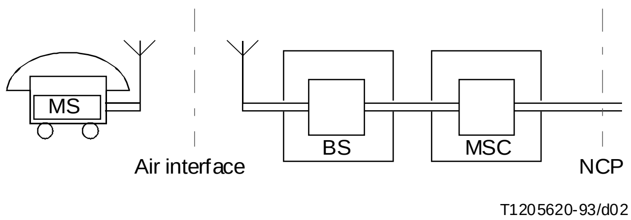


FIGURE A.1/G.173

Basic configuration of a digital mobile system

A.1.3 mobile services switching centre (MSC): The MSC performs all necessary signalling and transmission functions in order to establish connections to and from mobile stations. A mobile station is registered at one MSC which functions as its home-MSC for charging and billing purposes.

In order to obtain radio coverage in a given geographical area, a number of base stations are normally required, i.e. each MSC is interconnected with several base stations. In addition, several MSCs may be required in order to cover the area of a PLMN.

A.1.4 mobile station (MS): A mobile station (MS) is transportable terminal equipment providing different services to the customer in a PLMN. It provides access via a base station to the PSTN/ISDN and to other mobile stations.

A.1.5 network connection point (NCP): A network connection point is a point at which, in general, two independent networks are interconnected. In the context of this Recommendation, NCP refers to the connection point between a PLMN and the PSTN/ISDN.

A.1.6 public land mobile network (PLMN): A public land mobile network may be defined as a number of mobile services switching centre areas within a common numbering plan and a common routing plan.

With respect to their functions, the PLMNs may be regarded as independent communications entities, even though different PLMNs may be interconnected through the PSTN/ISDN for the forwarding of calls or network information. The MSCs of a PLMN can be interconnected similarly to allow interaction.

The use of the PLMN concept is illustrated in Figure A.2 where various PLMNs and their interfaces to the fixed networks are shown. It should be noted that a PLMN may have several interfaces with the fixed network (e.g. one for each MSC). Interworking between two PLMNs may be performed via an international switching centre.

Figure A.2 also shows the information paths between a PSTN/ISDN and a PLMN and between two different PLMNs. The solid lines indicate a possible physical path between the PLMNs through the PSTN/ISDN. The dotted line indicates that for some interactions an end-to-end information path may exist between the PLMNs.

The PLMN is connected via an NCP to the PSTN/ISDN. If there are two mobile service suppliers in the same country, they can be connected through the same PSTN/ISDN.

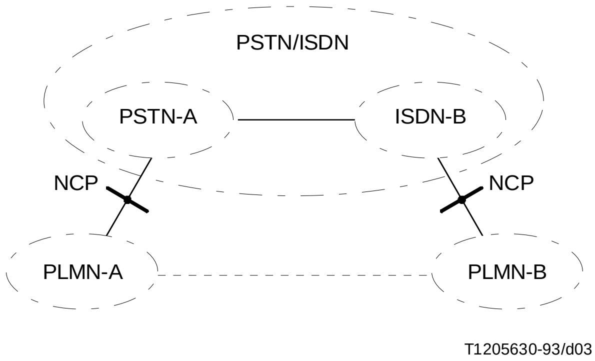


FIGURE A.2/G.173

The use of the PLMN concept for country A and country B

A.2 Detailed model of the digital land mobile system

A.2.1 General

A more detailed model of the digital land mobile system used for the consideration of transmission planning issues for the speech service is shown in Figure A.3. This model is an example of the relevant functions and does not necessarily imply any particular physical realization.

In a digital mobile system three mobile stations with different acoustic interfaces have to be considered:

– handset MS;

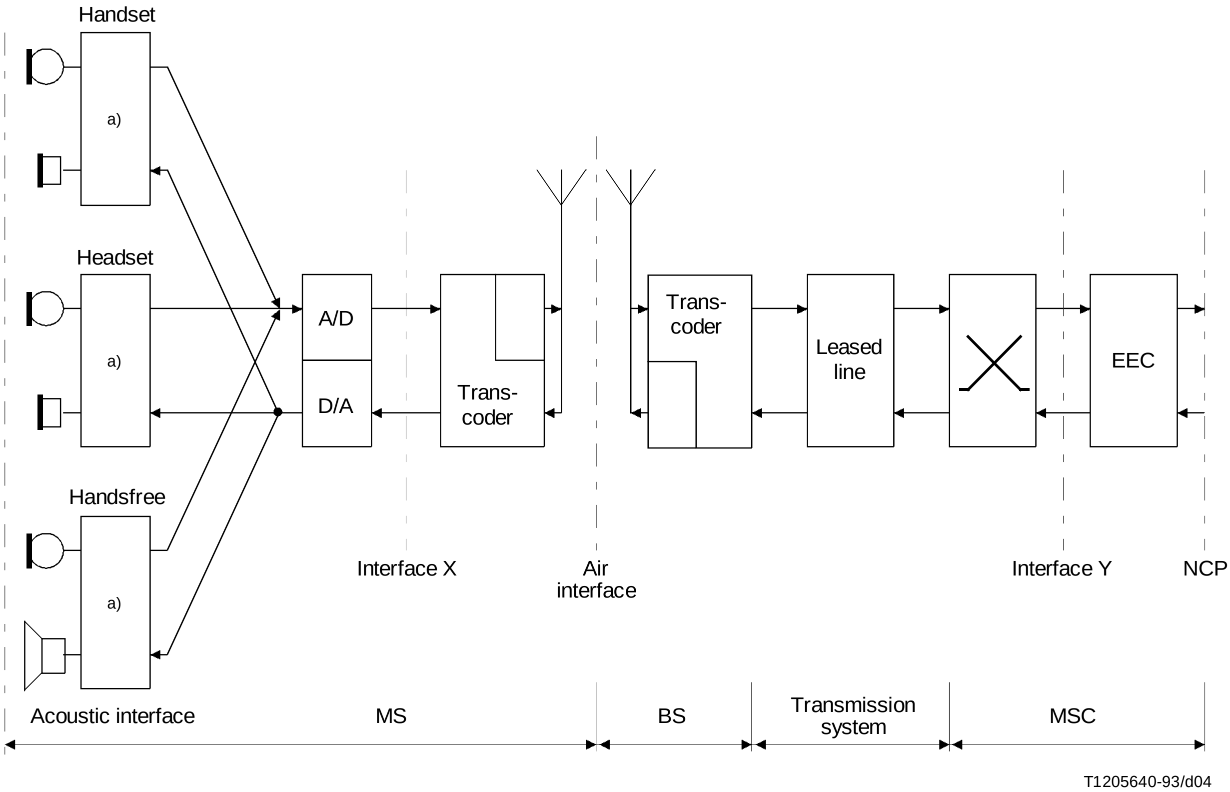
– headset MS;

– handsfree MS using loudspeakers.

Several coding methods with low rate encoding (LRE) have been proposed to achieve spectrum efficiency on the radio path. In addition to the LRE, it is also possible to include digital speech interpolation via the radio path.

For transmission planning aspects the transmission system (leased line) between the BS and the MSC have to be taken into account.

Because of speech coding and/or other channel processing techniques, the propagation time is of such magnitude that echo control devices may be required at both ends of the digital mobile system. Any acoustic echo control in the MSs will be provided by analogue processing or digital processing or a combination of both techniques. The electric echo control device at the NCP should be under the control of the MSC so that the electric echo control device can be enabled or disabled for calls to and from the PSTN/ISDN.



EEC Electrical echo cancellation

a) Possible acoustic echo control, see A.3.4.

FIGURE A.3/G.173

Model of the digital mobile system used for consideration of transmission planning issues

A.2.2 Interfaces

The main interfaces identified within the digital mobile system are shown in Figure A.1 and Figure A.3. For the purpose of this Recommendation, the air interface and the network connection point (NCP) are identified along with three other interfaces, the acoustic interface, interface X and interface Y. These interfaces are needed to define the digital mobile system transmission characteristics.

The acoustic interface for handset MS is described by the mouth reference point (MRP) and the ear reference point (ERP) and is used for measuring the audio part of the handset MS. The acoustic interface of all other types of MS is for further study.

Interface X is introduced for design purposes in order to separate the speech transcoder impairments from the basic audio impairments of the MS. At that point, which is considered to have a relative level of 0 dBr, the analogue signals will be represented by 8 bit A-law or μ-law according to Recommendation G.711. A 13/14 bit linear PCM signal could also represent the analogue signals.

The air interface is specified by the PLMN network operator and is required to achieve MS transportability. Analogue measurements can be performed at this point by using the appropriate radio terminal equipment and speech transcoder.

Interface Y might be used in the case of direct MSC-to-MSC connections within the PLMN. The interface will generally be at the 2048 kbit/s or 1544 kbit/s level in accordance with Recommendation G.703. At that point, which is considered to have a relative level of 0 dBr, the analogue signals will be represented by 8 bit A-law or μ-law according to Recommendation G.711. Analogue measurements may be conducted at this point using the standard send and receive sides as defined in Recommendation G.714.

The network connection point (NCP) is the connection point between the PLMN and the PSTN/ISDN. The NCP has the same electrical characteristics as interface Y.

A.2.3 Transmission system in the digital mobile system

As shown in Figure A.3, the digital mobile system will usually contain transmission systems providing 4-wire circuits.

If the PLMN includes digital leased circuits which do not include any speech processing devices, the overall requirements of the digital mobile system will not be affected by the presence of the circuit with the exception of propagation time.

If the PLMN includes analogue leased circuits, the transmission characteristics (e.g. attenuation, attenuation distortion, noise, propagation time) will affect the overall requirements of the digital mobile system. Recommendations M.1020, M.1025, M.1030 and M.1040 describe several transmission characteristics for leased lines. In cases where the analogue lines introduce loss, provision will have to be made to compensate this loss.

A.3 Detailed transmission performance

A.3.1 Loudness ratings

For planning purposes, loudness ratings (LRs) are defined by objective methods as described in Recommendations P.64, P.65 and P.79. The values given in terms of LR in Recommendations G.111 and G.121 should provide an adequate loudness of speech in international telephone connections. The maximum value of the overall loudness rating (OLR) is 29 dB and the optimum OLR value is 10 dB.

Recommendation G.121 defines the following nominal send loudness rating (SLR) and receive loudness rating (RLR) values of the national system for the long-term:

SLR = +7 dB to +9 dB;

RLR = +1 dB to +3 dB.

Where digital routings are used to connect the PLMN to the international chain of circuits, the SLR and RLR of the national system will be largely determined by the SLR and RLR of the PLMN.

The SLR and RLR values for the PLMN apply up to the NCP. However, the main determining factors are the characteristics of the MS. Hence, in practice, it will be convenient to specify loudness ratings up to interface X. In the usual case where the PLMN introduces no additional loss or attenuation distortion between interface X and the NCP, the loudness ratings up to the PSTN/ISDN will be the same as the loudness ratings measured at interface X. However, in some cases loss adjustment may be needed for interworking situations in individual PLMNs.

These values are directly applicable to the case of an MS operating in a conventional non-mobile noise environment. Studies have shown that in a PLMN noise environment, speech levels are likely to be higher. Hence, in order to avoid clipping in the speech transcoder it may be necessary to increase the value of the SLR.

The network planning guidance given in this Recommendation is based on the use of handset MS. However, this mode of network planning also allows headset and handsfree MS to be connected.

The SLR and RLR should be measured and computed using the methods given in the relevant CCITT Recommendations.

A.3.2 Stability

The stability loss presented to the PSTN/ISDN by the PLMN at the NCP should meet the principal requirements of clauses 2 and 3/G.122. These requirements will be met if the attenuation between the digital input and digital output at the NCP is at least 6 dB at all frequencies in the range of 200 Hz to 4000 Hz under the worst-case acoustic conditions at the MS (any acoustic echo control should be enabled).

In the case of a digital connection between interface X and the NCP, the stability requirements can be applied to interface X. The worst-case acoustic conditions will be as follows (with any volume control set to maximum):

- handset MS — the handset is lying on, and the transducers are facing, a hard surface;
- headset MS — for further study;
- handsfree MS — a representative worst-case position of microphone and loudspeaker (for further study).

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

A.3.3 Propagation time

It is necessary in a telephone connection to limit the propagation time between the two users. Too long a propagation time causes difficulties in conversation over the connection. This can be attributed to two reasons. Firstly, the signal is reflected from the distant end causing an echo to be returned to the talker. Secondly, even if ideal echo control has been 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.

Recommendation G.114 provides guidance on the mean one-way propagation time in telephone connections.

Recommendation G.131 contains a "practical rule" (Rule M), according to which some form of echo control is required to reduce the level of the echo if the one-way propagation time is above 25 ms. This rule was developed in conventional arrangements of electrical echos for two-to-four wire converters.

As a network performance specification, the following limitations apply to a mean one-way propagation time in cases where echo sources exist and appropriate echo control devices, such as echo suppressors and echo cancellers, are used:

i) 25 to 300 ms, acceptable;

ii) 300 to 400 ms, acceptable with:
- effective control of all echos without clipping by use of good echo cancellers;
- low background noise leading to an absence of perceptible noise contrast;
- low distortion of transmitted signals;
- ideal loudness rating;

iii) above 400 ms, unacceptable.

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 control could give rise to severe speech clipping and noise contrast. Also, the operation of voice switching will lead to impairments similar to those caused by echo suppression.

The propagation time depends on the following elements of the PLMN:

- speech transcoding;
- radio channel coding;
- PLMN network (e.g. fixed elements such as transmission systems, digital exchanges, echo control devices).

The following values are given for the mean one-way propagation time of the PLMN:

recommended value < 40 ms (see Note).

NOTE – Technical and practical considerations do not permit the recommended values to be achieved at this time. Existing PLMNs should provide acceptable service on many connections but compliance with Recommendations G.114 and G.131 may not be assured on all connections.

The propagation time shall be calculated in accordance with Recommendation G.114 taking into account the various elements present in the connections.

In Annex B some reference connections are given to illustrate propagation time and echo control issues.

A.3.4 Echo control

Talker echo is an undesirable phenomenon that may be observed in a telephone connection if the signal propagation time combined with echo sources is significant. There are two main sources of echo:

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

Electrical echo can be eliminated by the use of end-to-end four-wire transmission. Acoustic echo will be generated in all telephone sets 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. Cases involving acoustic echo may be even more complex in view of the fact that acoustic echo varies with time.

Echo planning guidelines have been provided within Recommendation G.131. Fundamentally, the planning objective is for not more than 1% of users who perceive talker echo at a level that causes dissatisfaction with the telephone call.

To reduce the level of electrical talker echo, there are various ways of inserting echo control devices (ECDs) in connections having a one-way propagation time of more than 25 ms.

Acoustic echo cancellation is under further study.

Recommendation Q.522 calls for ECDs to be "under the control of" an exchange. The same procedure is necessary for the MSC. For the MSC this will, as a rule, imply that it is most convenient to use digital ECDs which will actually form part of the controlling MSC. Furthermore, the future need to distinguish between different ISDN services ought to be considered.

Various ECD designs are described in the G.160-Series Recommendations. It is generally agreed that the echo canceller provides the best performance in all circumstances and therefore this is the preferred type.

The loudness of an echo path may be expressed by means of the "Talker Echo Loudness Rating (TELR)". The TELR is the addition of the SLR, RLR and the weighted echo loss L_e (see Recommendation G.122) in the echo loop. It is normal practice to use the minimum nominal SLR and RLR values when calculating the TELR for a connection in order to achieve the worst-case TELR value as a result.

The curves of Figure 2/G.131 indicate the minimum value of the TELR that must be introduced into the echo path if no ECD is included. The TELR is shown as a function of the mean one-way propagation time. The 1% curves applicable to fully digital connections call for the following TELR values:

TELR = 55 dB for connections having 400 ms mean one-way propagation time.

If there is a digital call path in the PLMN as well as in the PSTN/ISDN and if the sum of the minimum nominal LR values (SLR + RLR = 10 dB) of the telephone set is considered, then the echo loss at any echo source should correspond to $L_e = 45$ dB.

However, where the sum of the minimum nominal LR values (SLR + RLR) for the evolving PSTN is less than 10 dB, then correspondingly higher values of L_e will be required to meet the G.131 objective. For example if SLR + RLR = 6 dB, then the echo loss should correspond to be $L_e = 49$ dB.

Because of speech coding and/or other channel processing techniques, the propagation time is of such magnitude that echo control devices may be required at both ends of the digital mobile system. Acoustic echo control will be provided in the MSS and electric echo control at the NCP. Figure A.4 illustrates two different cases:

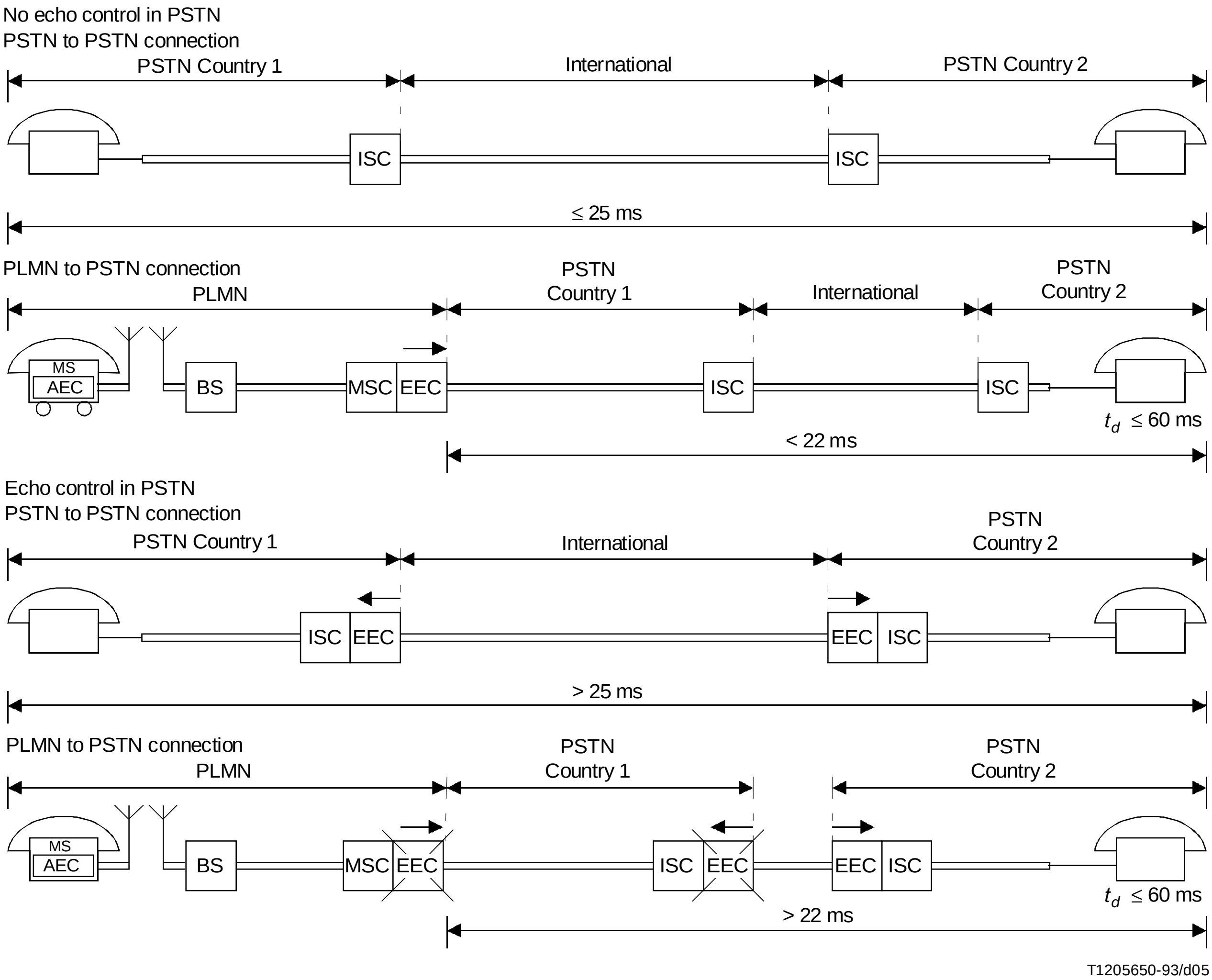
a) no echo control in the PSTN/ISDN;

b) echo control in the PSTN/ISDN.

Connections should be designed to comprise the minimum practical number of ECDs.

The rules, in Recommendation G.131 for PSTN/ISDN connections with echo control devices, do not give sufficient information about the requirement to disabling the intermediate echo control device. For the time being, in this Recommendation all echo control devices which may cause an impairment, and thus may control the overall performance when tandemed are identified in Figure A.4. Depending on the design of echo cancellers, intermediate echo cancellers may not need to be disabled. For echo suppressors, it seems to be necessary. This is under study.

Currently in most cases, there are no signalling facilities to disable intermediate echo control devices in the PSTN/ISDN.



This echo control device should be disabled unless it can operate in a tandem configuration with negligible echo performance degradation

Direction of arrow indicates the echo loop

FIGURE A.4/G.173

Echo control between PLMN and PSTN/ISDN

The electrical echo control device within the PLMN at the NCP should meet the requirements of Recommendation G.165 but with an echo path delay of t_d = 60 ms. The 60 ms is calculated as follows. Recommendation G.131 states that the maximum length of connections which need no echo control has a mean one-way propagation time of 25 ms. However, this value is the sum of the delays of the international connection and the maximum national delays at each end of the connection. The interconnection of the PLMN with the PSTN/ISDN is unlikely to be at an NCP where the PSTN/ISDN delay is greater than 22 ms. Taking account that the dispersion may be up to 8 ms, the maximum echo path delay which the echo canceller in the MSC should expect is

$$t_d = (22 \text{ ms} + 8 \text{ ms}) \times 2 = 60 \text{ ms}.$$

Certain countries situated near the edge of a continent may need to increase this value as there may be a number of connections which do not comply with Recommendation G.131 having a mean one-way propagation time of more than 25 ms and yet are not provided with echo control.

Terminal coupling loss provided in the MS should provide an echo loss of 45 dB at interface X over the likely range of acoustic end delays. If acoustic echo control is provided by voice switching, comfort noise should be injected. 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 a handset MS, a careful design may allow the use of echo cancellation techniques without non-linear processing. The implications of this are under study.

In the case of a headset MS, a careful design can mean that no echo control is necessary.

In the case of a handsfree MS, the reverberation and dispersion may be time variant. The expected values of dispersion are under study.

In Annex B some reference connections are given to illustrate propagation time and echo control issues.

A.3.5 Quantizing distortion

The incorporation of digital processes in international telephone connections, particularly during the mixed analogue/digital period, can result in an appreciable accumulation of transmission impairments. It is, therefore, necessary to ensure that this accumulation does not reach a point where it can seriously degrade overall transmission quality.

From the point of view of quantizing distortion, Recommendation G.113 recommends that no more than 14 units of quantizing distortion should be introduced in an international telephone connection.

In principle, the number of units for other digital processes are determined by comparison with an 8-bit PCM codec pair such that the distortion of the digital process being evaluated is assigned n qdu if it is equivalent to an 8-bit PCM processes in tandem.

Detailed information about planning values for quantizing distortion can be found in Recommendation G.113. The given values are only valid for speech transmission.

The number of units of transmission impairment in an international telephone connection should not exceed:

$$5 + 4 + 5 = 14 \text{ units.}$$

Under the above rule each of the two national systems of an international telephone connection are permitted to introduce a maximum of 5 units of transmission impairment and the international chain up to 4 units.

For the quantizing distortion of the PLMN in error free conditions between the acoustic interface and the NCP, the following values are given:

recommended value < 4 qdu (see Note).

The qdu of PLMN under realistic bit-error-rate conditions (conditions to be determined):

recommended value < 7 qdu (see Note).

NOTE – Technical and practical considerations do not permit the recommended values to be achieved at this time by all systems. Existing PLMNs should provide acceptable service on many connections but compliance with Recommendation G.113 may not be assured on all connections.

Speech quality is to be specified not only on the error-free condition, but also in the channel conditions corresponding to the radio zone edge, where the average bit error rate (BER) may reach over 1% due to Rayleigh fading.

Such bad channel conditions usually cause speech signal distortion larger than that caused by quantization. It is necessary to set requirements on the high bit error rate conditions at the zone edge.

A.3.6 Clipping

The loss of the start or the end of 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;
- echo cancellers with centre clippers;
- digital speech interpolation;
- acoustic echo control in telephone sets.

The effect of tandem voice switches which are not under a 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 actions could be taken:

- intermediate tandem voice switching devices in the PSTN/ISDN should be either disabled by means of signalling means or avoided by means of routing;
- the voice switching for acoustic echo control in the MS and DSI in the PLMN should be under a common control.

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

A.3.7 Noise contrast

Continuous background noise is likely to be present in the case of all PLMN calls no matter whether users are talking or not. There may also be one or more voice-operated devices; these effectively break the circuit provided there is no speech on it.

Noise contrast problems are caused by background noise being interrupted when the circuit is broken, so that the user listening on the circuit hears the background noise being continuously switched on and off. This is particularly disturbing for a user talking to a PLMN user in a moving vehicle because the background noise 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 noise are:

- acoustic background noise picked up by the microphone;
- idle channel noise.

The following elements of a PLMN can cause noise contrast impairments:

- the acoustic echo control device in the MS;
- DSI within the PLMN;
- the electric echo control devices protecting the PLMN user against echos returned from the PSTN/ISDN.

The characteristics of the ambient noise (spectrum and level) depend on the environment in which the MS 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 interface of the terminal will significantly affect the dynamic range of the noise contrast.

The noise contrast can also be reduced by insertion of comfort noise to be used for the acoustic echo control with centre clipper in the MS and for voice switching within the PLMN.

Two problems associated with other voice switching devices (e.g. DSI equipment within the PSTN/ISDN) 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 and thus increase the noise contrast.

References

[1] CCITT Contribution COM XII-4, *Transmission Planning; Aspects of the Speech Service in the GSM PLMN System (Norway)*, Geneva, 1989.

Annex B

Examples of interconnections

(This annex forms an integral part of this Recommendation)

Figure B.1 shows the PLMN to PSTN/ISDN interconnection configuration. The PLMN is interconnected with the PSTN/ISDN via a gateway switching centre (GSC).

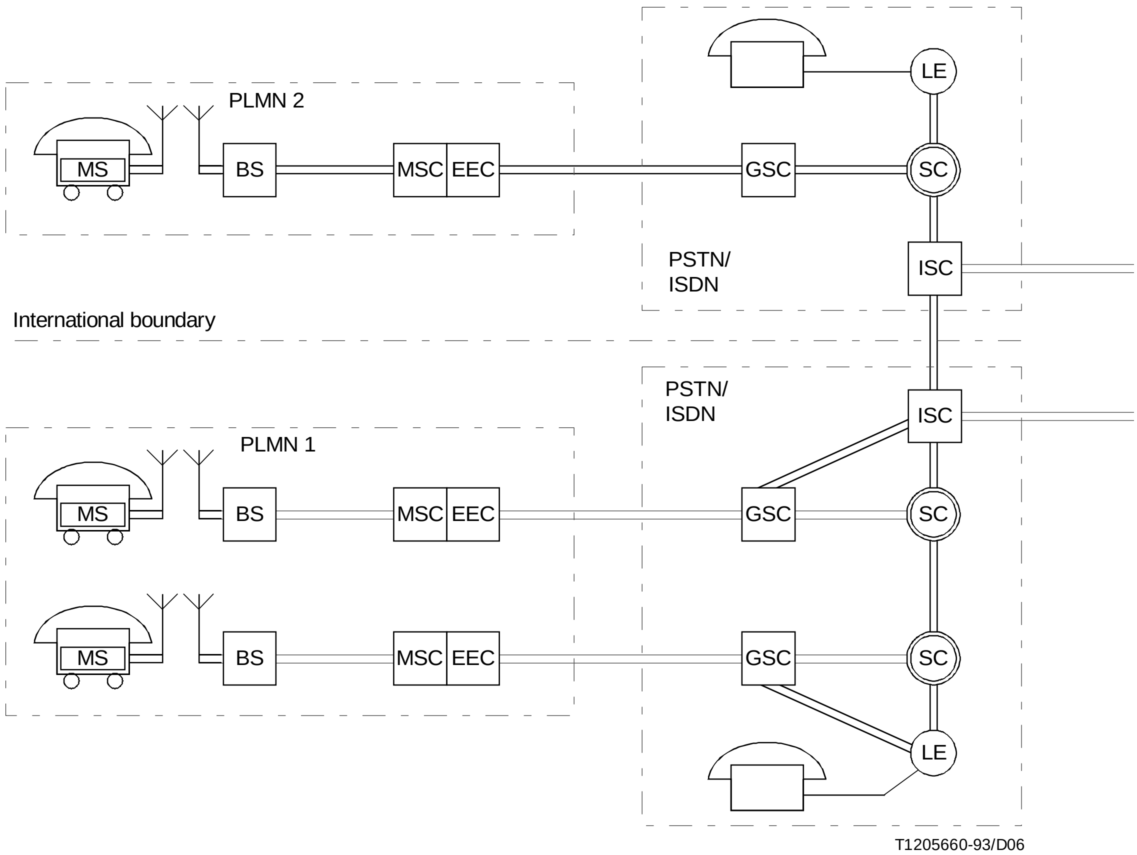


FIGURE B.1/G.173

PLMN to PSTN/ISDN interconnection configuration

In the case of PLMN connections to or from the PSTN/ISDN it may be impossible to achieve full compliance with other CCITT Recommendations if the PSTN/ISDN includes the following features:

– *intermediate echo control devices in the international network* – (If present, and not disabled, these devices will be in tandem with PLMN echo cancellers and may cause degradation.)

– *satellite routings* – (The delay inherent in the connections, when added to the PLMN delay, may result in conversational difficulties. Double satellite links are likely to cause severe difficulties and may be used, if necessary, to provide service.)

– *digital speech interpolation systems* – (An adverse interaction between voice switching systems in the PLMN is likely to occur.)

– *ADPCM* – (The distortion introduced by ADPCM on routes where PSTN/ISDN echo control is not provided is likely to reduce the echo cancellation performed by the PLMN 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 PLMN echo canceller.)

– *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 PLMN echo canceller.)

– *tandem connections of sources of quantizing distortion* – (The PLMN speech transcoder is equivalent up to 5 qdu’s between linear PCM interfaces.)

It is recognized that in some connections it may not be feasible to avoid these features without changes in other Recommendations, but in many cases, especially if taken into account at the planning stage, it may be possible to reduce the probability of occurrence by agreements between administrations on particular connections.

B.1 Connections between MSs

Figure B.2 shows the different types of connections between two PLMN users.

In the first case the MSCs are connected directly. From the transmission point of view, it is therefore recommended to use interface Y for such connections to avoid electrical echo cancellation.

If the PLMN users are connected via the PSTN/ISDN, the two electrical echo cancellers should be disabled so that the transmission quality is not reduced.

For PLMN users in different countries the third connection via the PSTN/ISDN is valid. If there are echo control devices on the international part, they should be also disabled together with the two electrical echo cancellers within the PLMN.

B.2 Connections from the PLMN to PSTN/ISDN

Figure B.3 shows the different types of connections between PLMN users and PSTN/ISDN users.

For connections from the PLMN to the PSTN/ISDN two different cases with echo control or no echo control in the PSTN/ISDN have to be considered (see also Figure A.4). In the case of long routing in the PLMN, the MSCs should be connected directly via interface Y.

B.3 Connections from the PSTN/ISDN to PLMN

Figure B.4 and Figure B.5 show the different types of connections between PSTN/ISDN users and PLMN users.

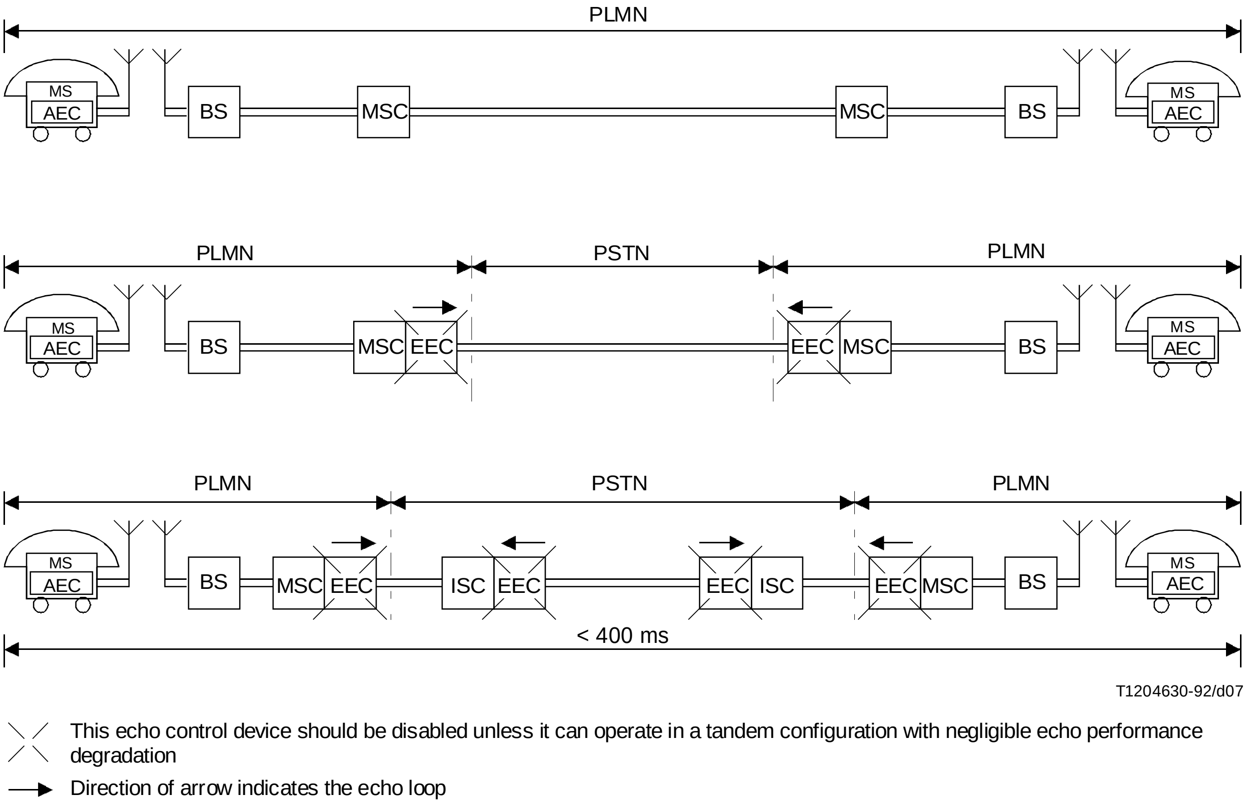


FIGURE B.2/G.173

Connections between PLMN users

For connections from the PSTN/ISDN to the PLMN two different cases with echo control or no echo control in the PSTN/ISDN have to be considered (see also Figure A. 4). In the case of long routing in the PLMN, the MSCs should be connected directly via interface Y.

Figure B.5 shows connections with re-routing where the overall propagation time of the call path has been extended beyond transmission planning limits.

The routings are likely to cause severe degradation to the quality of the connection and may result in significant difficulty, particularly when the connection contains a number of equipments having different impairments.

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

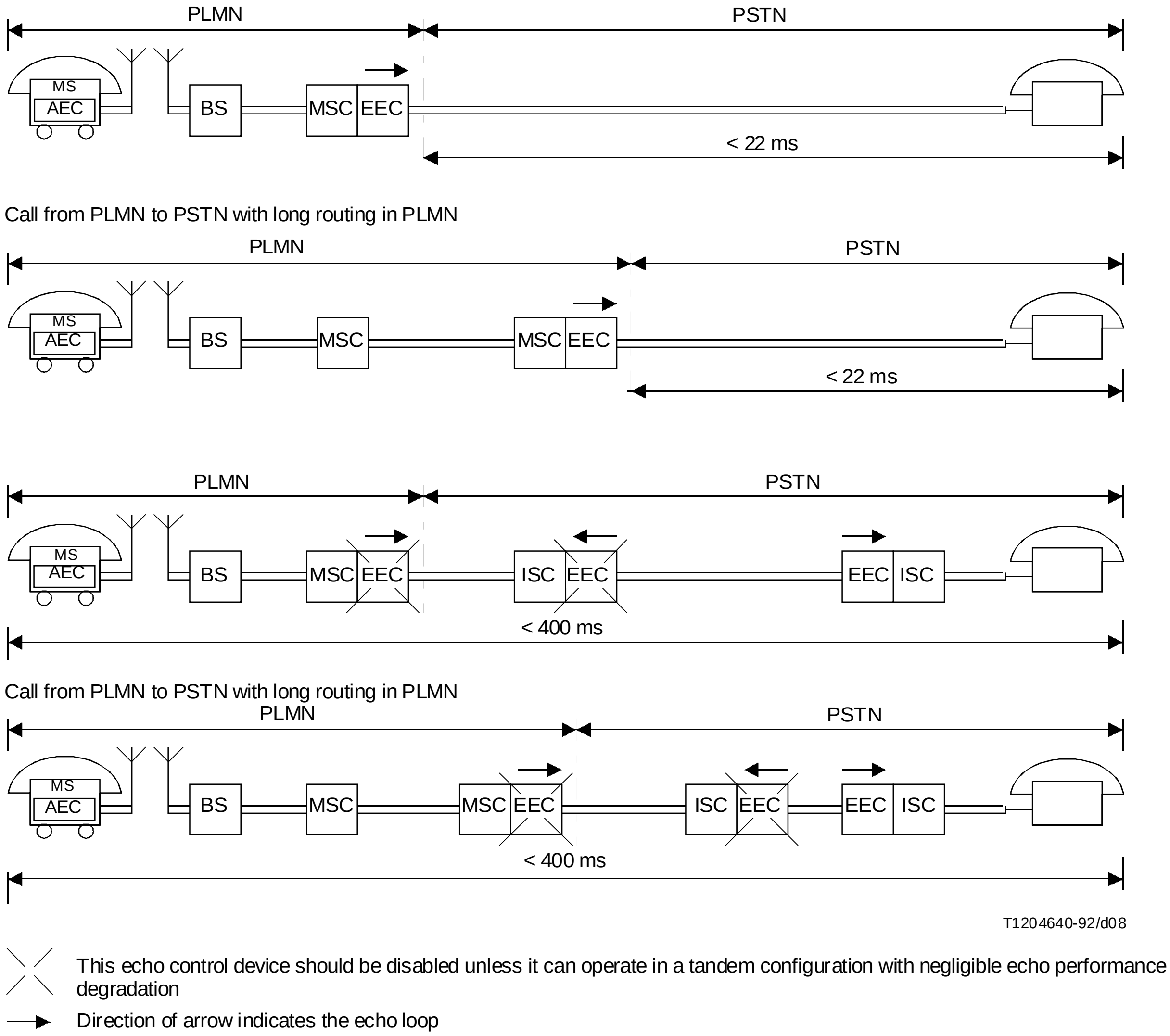


FIGURE B.3/G.173

Connections between PLMN user and PSTN/ISDN user

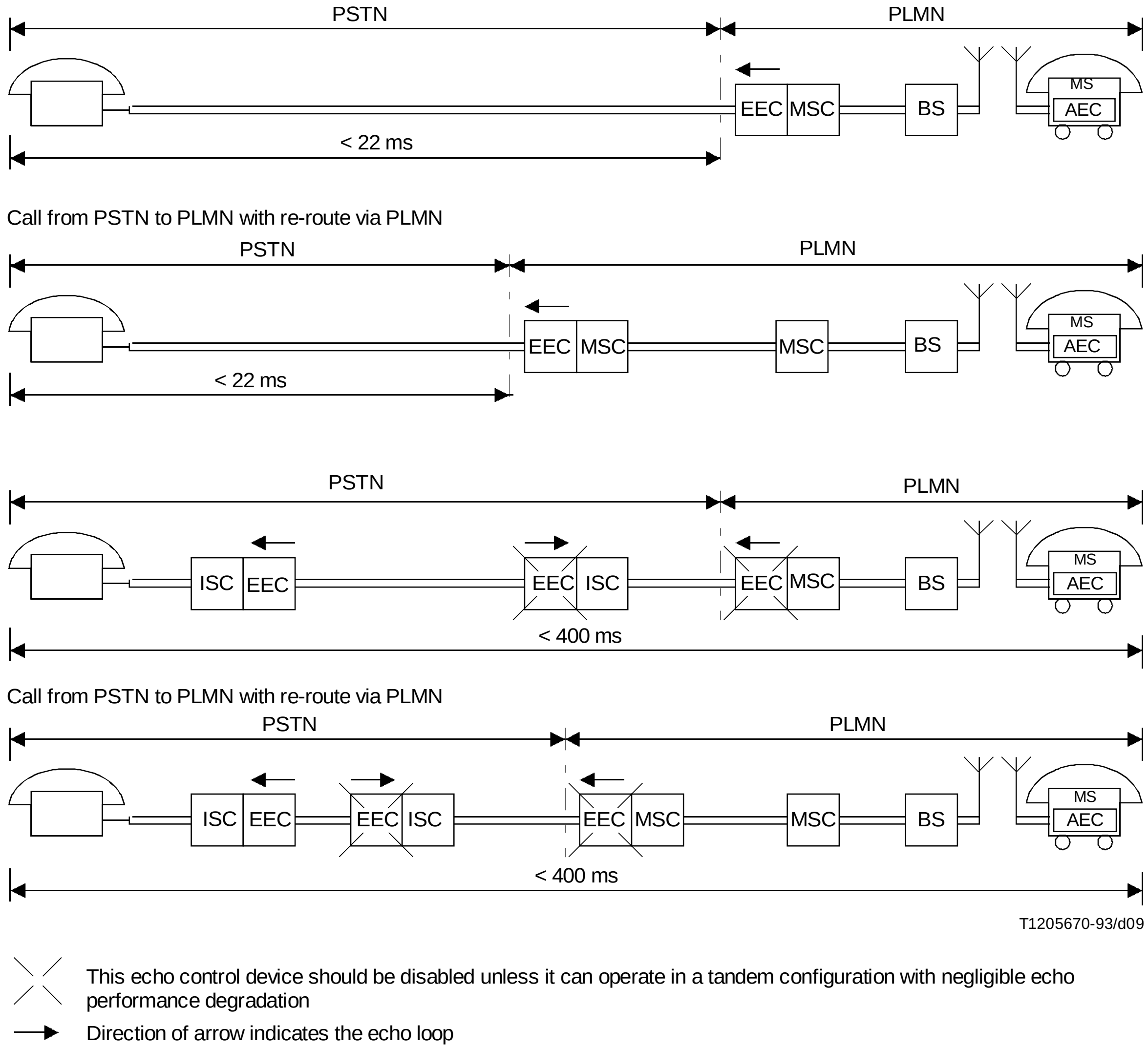


FIGURE B.4/G.173

Connections between PSTN/ISDN user and PLMN user

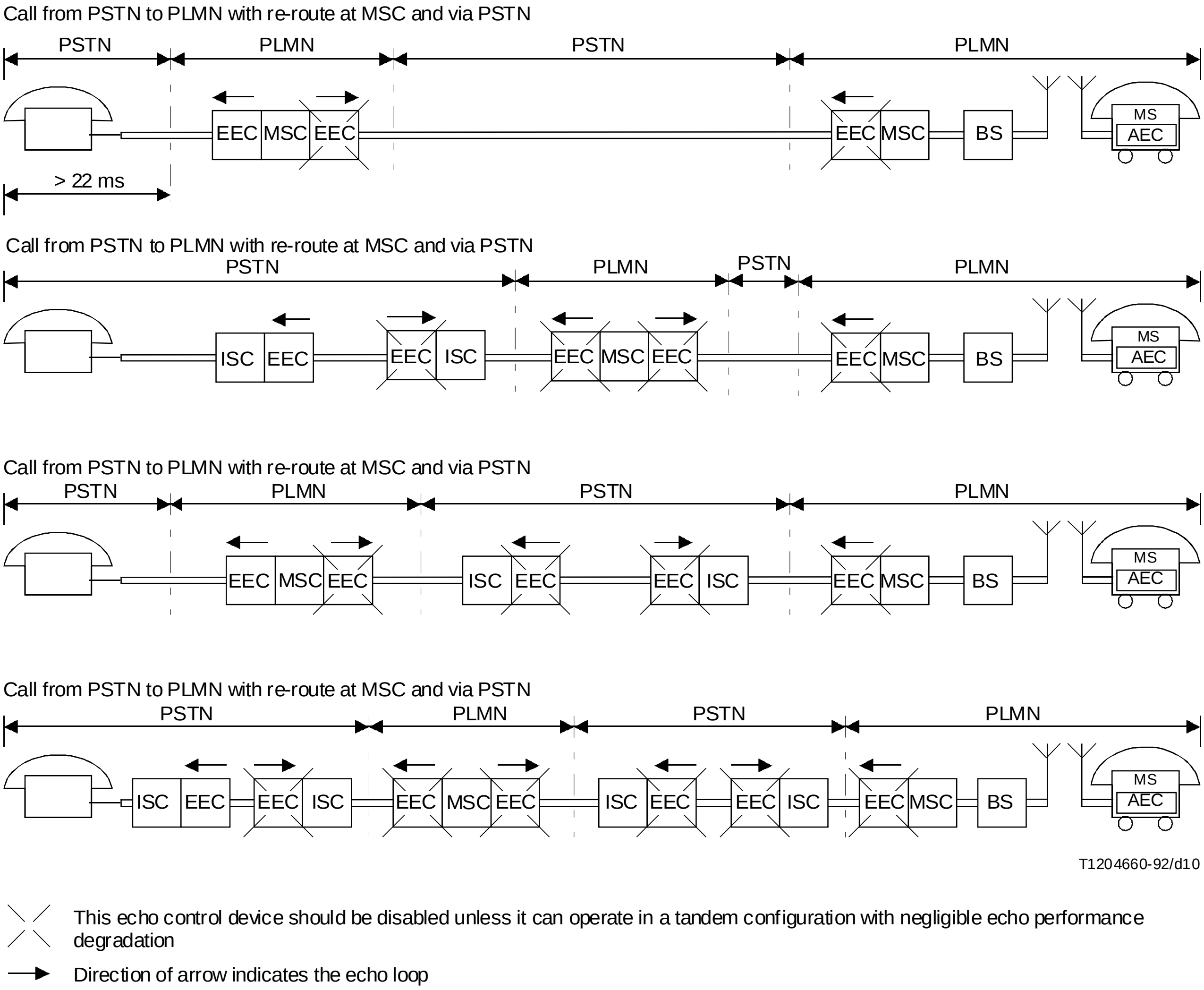


FIGURE B.5/G.173

Connections between PSTN/ISDN user and PLMN user by means of re-routing