ND1701:2006/03

Recommended Standard for the UK National Transmission Plan for Public Networks

Issue 5

This document did not receive unanimous agreement within NICC, with some parties unable to agree with the numerical value assigned to the *The national network delay objective* in section 7.3.1.2.1. However, it was agreed by NICC to publish this document as it represents at the time of publication the best consensus views of the majority of the NICC membership in the face of sustained objection by 2 organisations.

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Issue and amendment record

Issue status	Amendment	Description	Date
1		Originated by The Network Performance Design Standards Group of PNO-IG	20/07/94
2	Editorials		05/08/94
3 (draft a) (draft b) (final draft)	Editorials “ “	To align with current ITU-T recommendations and ETSI Standards “ “	31/3/99 17/5/99 10/6/99
4 (draft a) 4 (draft b)	New reference diagram and appendix on new technologies Editorials Revised Section 7	Changes recommended by the End-to_end Task Group of PNO-IG To align with current ITU-T recommendations and ETSI Standards Changes agreed by NICC drafting group	26/7/02 8/10/04
5 (draft a) 5 (draft b)	New section 11 on new technologies Section on new technologies moved to section 8 and amended	Incorporation of draft Annex B material into body of the document Changes suggested during End-to-End QoS Task Group teleconference	18/11/05 23/11/05

		(22/11/05)	
5 (draft c)	Section 8 further amended. Note added to 11.2. Internal references updated.	Changes suggested during End-to-End QoS Task Group meeting (29/11/05)	01/12/05
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5 (draft e)	Sections 8.5 and 10.5 added. 7.3.1.2 split into sub-clauses. Futher Editorial improvements made.	Changes suggested during End-to-End QoS Task Group meeting (12/01/06)	13/01/06
5 (draft f)	Minor editorial changes and corrections.	Changes suggested during End-to-End QoS Task Group final revew.	31/1/06
5	Four minor editorial corrections.	Approved by TSG membership.	14/3/06

1. Scope, purpose, application

1.1 Scope

This standard provides transmission recommendations for connections (including ISDN and VPN/Centrex) carried on the public switched telecommunications network (PSTN) that are all digital from Network Terminating Point (NTP) to NTP with digital or analogue access lines from appropriate end-user terminals or private networks. These recommendations are based on assumptions about certain characteristics of the end-user terminal equipment and private networks, in particular compliance with the ETSI terminal equipment requirements set out in TBR8, TBR21, EG201 121, EN301 437 and TBR38 (see the note in section 4.2).

It is intended that these recommendations be implemented in an evolutionary manner by Operators as required by applicable services, and not to existing networks, architectures or services.

The standard specifies recommended design values for network loss and other key transmission parameters. The standard also provides echo control planning and connection delay guidelines. To ensure satisfactory NTP-NTP performance to end-users, recommendations are given for the appropriate range in values for these parameters at the access interfaces to end-user terminals and interconnecting networks.

Sections 7 and 8 of this document include guidelines which should be taken into account by operators planning to introduce new technologies, such as IP and ATM, into the UK PSTN. They are not intended to be applicable to all voice over IP (VoIP) applications.

1.2 Purpose

This standard provides planning rules and guidance for the evolution of networks to handle all-digital services (such as those provided by new switch and transport technologies, and data networks supporting voice), while maintaining compatibility with the current analogue and digital networks, thereby continuing high quality service to end users by the telecommunications industry. An additional purpose is to ensure correct and easy interconnection of United Kingdom Public Telecommunications Operators, as well as connections with terminal equipment and private networks for which compatible standards have been developed.

1.3 Application

This standard applies to voice and data services established over fixed and mobile public networks including any connected private networks that themselves may contain VPN/Centrex services. The standard applies to connections with either analogue or digital access. Although there are many different types of access, they are classified for the purposes of this standard, in generic transmission classes illustrated in the new connection elements diagram:

- Copper Access
- Fixed Radio Access
- xDSL Access (e.g. ADSL, VDSL)
- GSM Access
- Cable Modem Access
- UMTS Access

Either analogue or digital access connections may provide access to private networks.

2. Referenced and related standards and publications

This standard is intended to be used with the following standards:

Table 1: ITU-T standards

Recommendation	Description
ITU-T Recommendation V.2	Power Levels for Data transmission over telephone lines. Blue book, Volume VIII-Fascicle VIII.1 (1988)
ITU-T Recommendation G.111	Loudness Ratings (LRs) in an International Connection. (03/93)
ITU-T Recommendation G.113	Transmission Impairments. (02/96)
ITU-T Recommendation G.114	One-way Transmission Time. (02/96)
ITU-T Recommendation G.121	Loudness Ratings (LRs) of National Systems. (03/93)
ITU-T Recommendation G.122	Influence of National Systems on Stability and Talker Echo in International Connections. (03/93)
ITU-T Recommendation G.126	Listener Echo in Telephone Networks. (03/93)
ITU-T Recommendation G.131	Control of Talker Echo. (08/96)
ITU-T Recommendation G.165	Echo Cancellers. (03/93)
ITU-T Recommendation G.168	Digital Network Echo Cancellers
ITU-T Recommendation G.171	Transmission Plan aspects of Privately Operated Networks. Blue book, Volume III-Fascicle III.1. (1988)
ITU-T Recommendation G.711	Pulse Code Modulation (PCM) of Voice Frequencies. Blue Book, Volume III - Fascicle III.4, Melbourne 1988.
ITU-T Recommendation G.726	40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM). (12/90)
ITU-T Recommendation G. 727	5-, 4-, 3- and 2- bits Sample Embedded Adaptive Differential Pulse Code Modulation (ADPCM) (05/94)
ITU-T Recommendation G. 728	Coding of speech at 16 kbit/s using Low-Delay Code Excited Linear Prediction
ITU-T Recommendation G. 728 Annex G	16 kbit/s Fixed Point Specification
ITU-T Recommendation G.803	Architecture of transport networks based on the SDH
ITU-T Recommendation G.810	Definitions and Terminology for Synchronisation Networks. (08/96)
ITU-T Recommendation G.811	Timing Characteristics of Primary Reference Clocks.(09/97)

Recommendation	Description
ITU-T Recommendation G.812	Timing Requirements of Slave Clocks Suitable for use as Node Clocks in Synchronisation Networks.(06/98)
ITU-T Recommendation G.813	Timing characteristics of SDH equipment slave clocks (SEC).(8/96)
ITU-T Recommendation G.821	Error Performance of an International Digital Connection Operating at a Bit Rate below the Primary Rate and Forming Part of an ISDN. (08/96)
ITU-T Recommendation G.822	Controlled Slip Rate Objectives on an International Digital Connection. Blue Book, Volume III - Fascicle III.5.(11/88)
ITU-T Recommendation G.823	The Control of Jitter and Wander within Digital Networks which are based on the 2048 kbit/s Hierarchy. (03/93)
ITU-T Recommendation G.825	The control of jitter and wander within digital networks which are based on the SDH. (03/00)
ITU-T Recommendation G.826	Error Performance Parameters and Objectives for International Constant Bit Rate Paths at or above Primary Rate. (08/96)
ITU-T Recommendation G.921	Digital Sections based on the 2048kbit/s Hierarchy. Blue Book ,Volume III- Fascicle III.5. (11/88)
ITU-T Recommendation Q.551	Transmission Characteristics of Digital Exchanges. (11/96)

Table 2: ETSI standards

Standard	Description
EN 301 797	Harmonized EN for CT2 cordless telephone equipment cordless telephone equipment covering essential requirements under article 3.2 of the R&TTE directive
EN 300 001 Edition 4 January 1997	Attachment to Public Switched Telephone Network (PSTN); General technical requirements of equipment connected to an analogue subscriber interface in the PSTN (candidate NET4)
ETS 300 085 December 1990	Integrated Services Digital Network (ISDN); 3.1kHz telephony teleservice. Attachments requirements for handset terminals (Candidate NET33).
EN 300 175-8 January 2002	Digital European Cordless Telecommunications (DECT) Common interface (CI) Part 8: Speech coding and transmission.
ETS 300 283 April 1994	Business Telecommunications (BT); Planning of loudness rating and echo values for private networks digitally connected to the public network
ES 200 677 March 1998	Public Switched Telephone Network (PSTN); Requirements for Handset Telephony
EG 201 050 V1.2.2 February 1999	Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network
EG 202 086 February 1999	Speech Processing, Transmission and Quality Aspects (STQ); Objectives and Principles for the Transmission Performance of Multiple Interconnected Networks that aim to provide "traditional quality" Telephony Services
EG 201 121 February 2000	A guide to the application of TBR 21

Standard	Description
ETS 300 462	Transmission and multiplexing (TM); Generic Requirements for Synchronisation Networks. Part1 – Definitions and terminology for synchronisation networks. (May 1998) Part2 – Synchronisation network architecture (August 1999) Part3 – Control of jitter and wander in transport networks.(May 1998) Part4 – Timing characteristics of slave clocks suitable for synchronisation supply to SDH and PDH equipment. (June 1998) Part5 – Timing characteristics of slave clocks suitable for operation in SDH equipment.(May 1998) Part6 – Timing characteristics of primary reference clocks. (June 1998) Part7 - Timing characteristics of slave clocks suitable for synchronisation supply to equipment in local node applications.
TBR 8 October 1998	Integrated Services Digital Network(ISDN); Telephony 3.1kHz teleservice. Attachments requirements for handset terminals.
TBR 21 January 1998	Terminal Equipment (TE); Attachment requirements for pan-European approval for connection to the analogue Public Switched Telephone Networks (PSTNs) of TE (excluding TE supporting the voice telephony service) in which network addressing, if provided, is by means of Dual Tone Multi Frequency (DTMF) signalling.
EN 301 437 June 1999	Terminal Equipment (TE); Attachment requirements for pan-European approval for connection to the analogue Public Switched Telephone Networks (PSTNs) of TE supporting the voice telephony service in which network addressing, if provided, is by means of Dual Tone Multi Frequency (DTMF) signalling
TBR 38 May 1998	Public Switched Telephone Network (PSTN); Attachment requirements for a terminal equipment incorporating an analogue handset function capable of supporting the justified case service when connected to the analogue interface of the PSTN in Europe

Standard	Description
GTS 03.05 January 1995	European digital cellular telecommunication system (phase 1); Technical performance objectives (GSM 03.05)
GTS GSM 03.50 September 1996	Digital cellular telecommunication system (phase 2+); Transmission planning aspects of the speech service in the GSM Public Land Mobile Network system. (GSM 03.50)

Table 3: UK NICC Code of Practice

Document	Description
NCOP040 (NICC TG 23)	A Voluntary Code of Practice for the Design of Private Telecommunication Networks
PNO-IG/NNTP (NICC study group 29)	National network timing plan

3. Definitions, abbreviations, acronyms and symbols

ADSL Asynchronous Digital Subscriber Line.
BIP Bit Interleaved Parity.
CATV Cable television.
CPE Customer Premises Equipment.
CRC Cyclic Redundancy Check.
CT2 Cordless Telephony (system 2).
DECT Digital European Cordless Telephony.
ETS European Telecommunication Standard.
ETSI European Telecommunications Standards Institute.
GPS Global Positioning System.
GSM European Digital Cellular Telecommunication System, Global system for mobile communications.
GSTN Global Switched Telephony Network.
HRX Hypothetical Reference Connection.
ISC International Switching Centre.
ISDN Integrated Services Digital Network.
ISP Internet Service Provider.
ITU-T International Telecommunications Union - Telecommunications Standardisation Section.
LD-CELP Low Delay Code Excited Linear Prediction.
NTP Network Termination Point.
OLR Overall Loudness Rating.
PBN Packet (or Cell) Based Network.
PBX Private Branch Exchange.

PCM	Pulse Code Modulation.
PDH	Plesiochronous Digital Hierarchy.
ppm	Parts per million.
PRC	Primary Reference Clock.
PSTN	Public Switched Telephone Network .
qdu	Quantising Distortion Unit.
RLR	Receive Loudness Rating.
SDH	Synchronous Digital Hierarchy.
SLR	Send Loudness Rating.
STM-N	Synchronous Transport Module (level N).
TACS	Total Access Communications System.
TU12	Tributary Unit (first level, second rate).
UMTS	Universal Mobile Telecommunications System.
xDSL	Digital Subscriber Line, where ‘x’ represents different bandwidths structures.
VDSL	Very High Speed Digital Subscriber Line.

4. Responsibilities and general criteria

4.1 Responsibilities

The primary responsibility for determining compliance with these recommendations lies with the operator selected by the customer to carry the call. For example, if a customer dials a number preceded by the access code for, say, 'Commsco', then Commsco is responsible for the quality of the call. It is the responsibility of Commsco to negotiate appropriate interconnect agreements with other operators to ensure adequate network performance for its customers.

4.2 General criteria

For typical PSTN voice connections over digital networks, voice transmission quality depends primarily on received level, delay, echo and quantising noise. Talker echo is a particular source of voice transmission degradation as a result of increased delays in digital networks. For significant delays, control of talker echo is achieved by the addition of echo cancellers.

Moreover in the case of voice band data transmission, quality is particularly sensitive to quantising noise, and saturation of the encoder by modem signal levels will result in higher error rates. Consequently, modem signal levels should be controlled by the voice band data terminal to ensure proper levels at the encoder.

Perceived call quality is also affected by the overall transmission delay, where excessive delay can make conversation difficult. This is in addition to echo constraints.

In the case of a voice connection with 4-wire (where separate channels are used for each direction of transmission) digital access to either a 4-wire end-user terminal or a digital private network, echo can only occur as a result of acoustic coupling or crosstalk within the terminal or handset. The digital terminal design requirements will therefore need to provide sufficient talker echo control to preclude the need for network-based echo control devices.

The performance of terminal equipment is an important factor in end-to-end quality. This is particularly true for a fully digital environment, where the characteristics of the end-user terminal essentially determine the overall performance.

The key transmission planning characteristics of end user terminal equipment are send/receive loudness ratings, quantising distortion and (for 4 wire equipment) delay and echo loss. Current European requirements for these parameters are included in a number of documents.

Examples include:

ISDN (Telephony)	(ETS 300 085)
CT2	(EN 301 797)
DECT	(I-ETS 300 175-8)
GSM/PCN	(GTS 03.05)
Private Networks	(ETS 300 283)
Private Networks	NCOP040
PSTN Voice Access	(Draft EN 301 437)
PSTN Telephones	(ES 200 677)

Network Operators should use the relevant ETSI requirements or quidelines to become aware of likely terminal equipment characteristics. **Note:** There are no longer any mandatory type approval requirements for CPE that have to be met before equipment can be placed on the market and put into service. Nevertheless, it is necessary to make some assumptions that CPE will behave in a similar manner to that specified in voluntary standards in order to make recommendations on suitable network performance limits.

The following principles were used in developing this standard:

(1) For end-user services accessing operator networks through digital access, no loss is inserted in the network. Therefore, when both access connections are digital, a digital bit stream applied at the users terminal will be transmitted without modification to the far end users terminal.

(2) Digital bit streams are transported between switches without modification except where such modification occurs intentionally as a function of a service.

(3) For digital access, it is assumed that loss or level control is incorporated as part of the end-user voice terminal, voice-band data terminal, PBX or private network, and hence is not required in the public network.

(4) For analogue access lines, it is desirable to insert network loss, where required, as near to the end-user terminal as possible.

5. Reference connections

5.1 Generic

The Reference Connection Elements diagram (figure 5.1) shows the building blocks that can be used to construct an end to end network. Annex A gives some examples of how these connection elements can be chained together to describe end-to-end connections. The body of this document describes planning guidelines for:

- a "traditional" PSTN/ISDN/GSM network built of copper, fixed radio and GSM access connection elements, circuit-switched core network connection elements and the international GSTN;
- the emulation of the PSTN service using new, packet-based technologies.

Sections 7 and 8 include guidelines that should be carefully considered when incorporating the following new technologies into the UK PSTN:
- xDSL access
- cable modem access
- UMTS access
- packet- or cell-based core networks
- hybrid packet- or cell-based/PSTN core networks

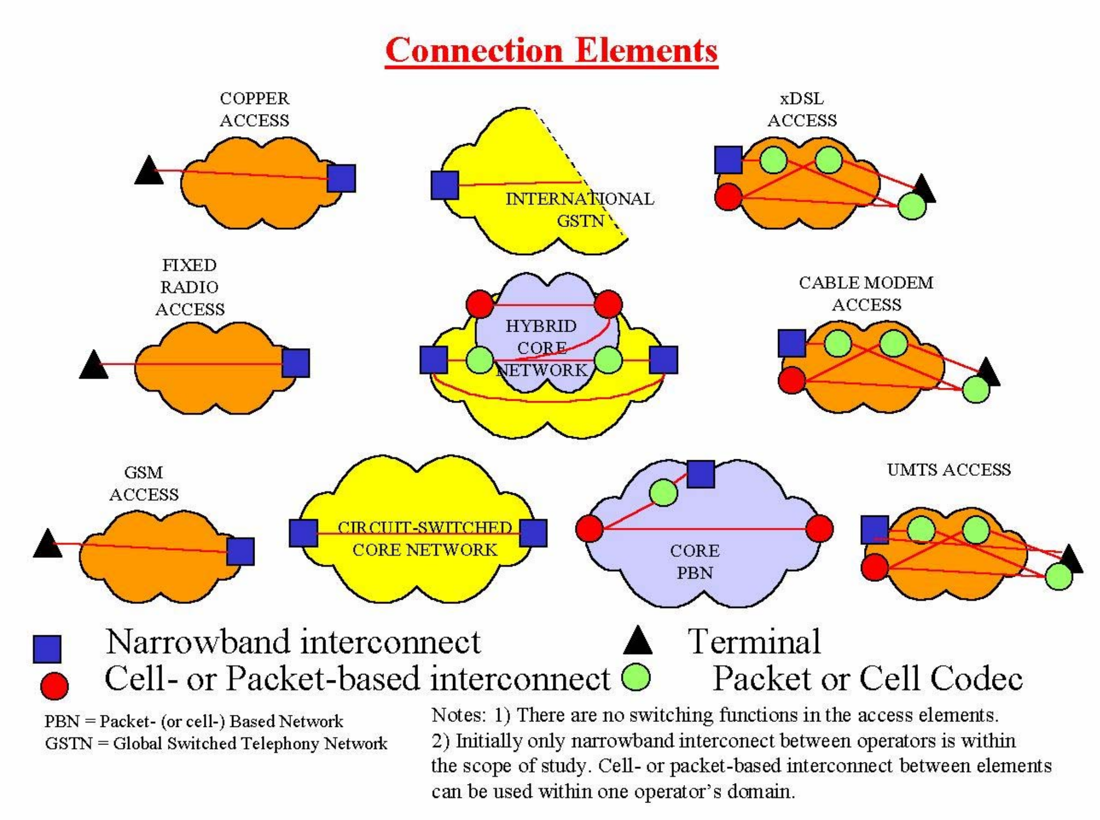


Figure 5.1: Reference Connection Elements

5.2 VPN

An increasing configuration is the use of Virtual Private Networks (VPN) to access the trunk network. From a performance viewpoint this can be described in terms of an addition to the generic reference model combining allowances for a Private Network and the conventional local access.

A VPN connection reference model is illustrated in Figure 5.2. This is applicable also to CATV and ISP.

Conventional private network access to Core Network.

VPN access to Trunk Network

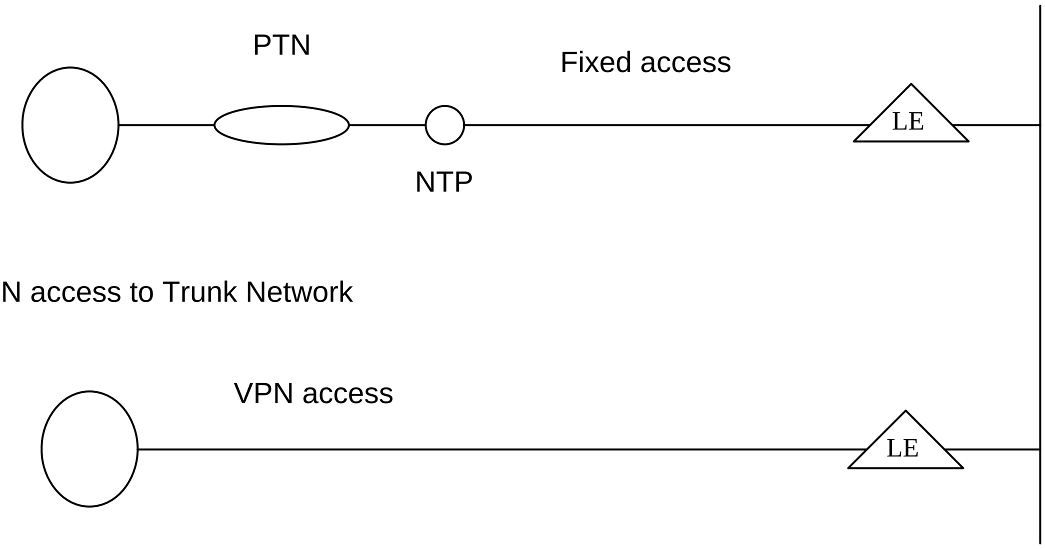


Figure 5.2: VPN Network performance design standards reference model

6. Transmission levels

6.1 General issues

Transmission levels on multi-operator connections are of particular importance as they impact on a number of performance areas:

- end to end transmission loss as seen by customers e.g. the loudness of telephony connections.
- signal levels sent to line
- level of talker-echo on telephony connections.
- stability margins on connections
- performance of echo control devices.

This section concentrates on the recommended values for transmission loss, stability and signal level sent to line. Echo control issues are covered in Section 7.

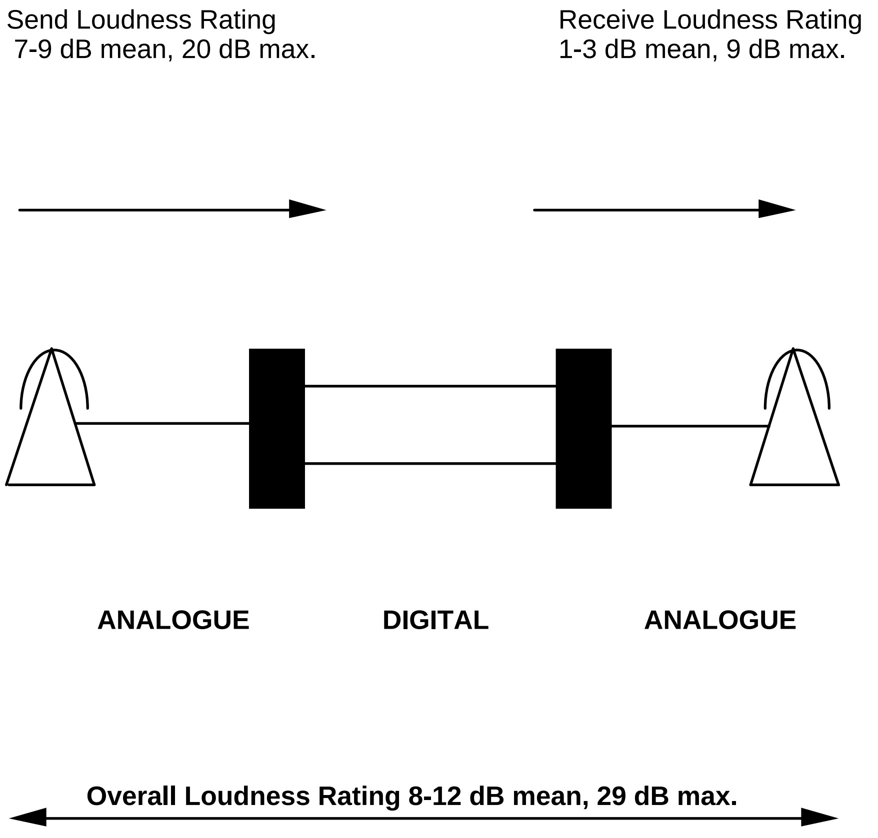
6.2 Telephony connections

Traditionally end to end transmission losses for telephony connections have been specified in terms of Overall Loudness Rating (OLR) which take into account performance of telephone instruments and hence represent the acoustic to acoustic performance as seen by customers. ITU-T recommendations in the G-series include maximum long term and short term objectives for values of OLR.

International network planning has required an apportionment of the end to end OLR values into allowances for national extensions and the international transit network. The international apportionment principles can also be applied to the current UK environment where end to end performance may be determined by the networks of several operators. As the inter-exchange networks of UK operators are likely to use digital transmission and switching (thus adding zero loss to the overall connection), the following simplified model can be used for transmission loss planning. OLR can be sub-divided into SLR (Send Loudness Rating) and RLR (Receive Loudness Rating) where:

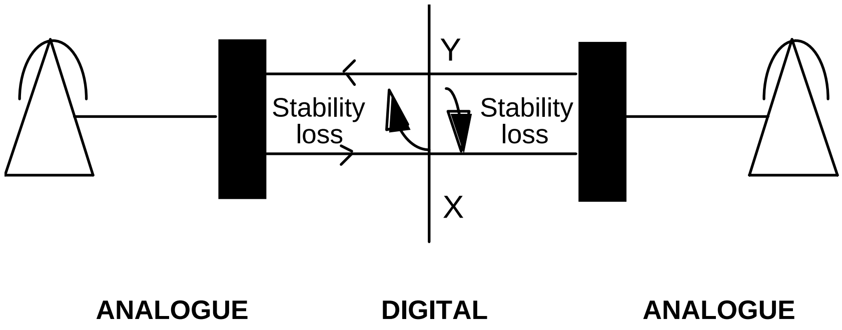
OLR=SLR + RLR dB

SLR and RLR represent the transmission losses between the acoustic input/output points and the digital bit stream of the switched network of the two directions of transmission. Recommended values for SLR and RLR are also included in the G-series Recommendations as shown in Figure 6.1.



LOUDNESS RATINGS -RECOMMENDED OBJECTIVES AS DEFINED IN G.111.

Figure 6.1: Loudness ratings



Stability loss (X-Y and Y-X) ≥ 6dB

Figure 6.2: X Stability loss

6.3 UK Loudness rating values (OLR, SLR, RLR)
Achieving the OLR objectives for multi-operator calls within the UK requires agreement of SLR and RLR objectives. It is recommended that switched connections within the UK should meet ITU-T recommended values for SLR and RLR as shown above. Meeting these objectives will also ensure that UK international access connections are consistent with the international transmission loss plan.

Demonstration of compliance with these objectives will require the responsible network operator (see sub-section 4.1) to be aware of the contribution to SLR/RLR from the customer equipment as well as other operators' networks. Customer equipment requirements (see ETSI documents) are listed in Section 4 (see also the note in section 4.2).

It should be noted that cellular network operators may employ digital level control in the mobile switched network to adequately control SLR and RLR values.

6.4 Voice-band data connections
Compliance with the above recommendations for telephony is likely to result in acceptable values of transmission loss for most voice-band data services.

6.5 Signal levels sent to line
Signal levels on voice services will normally be adequately controlled by limiting the minimum value of SLR as outlined in Recommendation G.111.

Recommended maximum power levels for non-voice signals (e.g. data transmission) are given in ITU-T recommendation V.2 (power levels for data transmission over telephone lines) and TBR 21.

6.6 Stability

The stability performance requirements for multi-operator networks are aimed at ensuring that the end-end connection does not enter a state of oscillation during the various stages of a call (set-up, clear-down, and the information transfer phase).

Stability is determined by a number of network factors such as transmission loss, impedance matching and design of handsets. However the overall requirement can be expressed in terms of a 'stability loss' specified at the boundary between 2 networks. The principle of this measurement is outlined in ITU-T Recommendation G.122 (Influence of national systems on stability talker-echo and listener echo in international connections).

In the UK environment the stability loss presented by networks should meet the principles of the requirements in Sections 2 and 3 of Recommendation G.122. These requirements will be met if the attenuation between the input and output ports at a network boundary is at least 6 dB at all frequencies in the range 200 Hz - 4 kHz for all connections, (see Figure 6.2).

Demonstration of compliance with these objectives will require the network operator responsible for quality (see sub-section 4.1) to be aware of the stability loss contribution from customer equipment and other operators' networks. Recent changes to the approval standards for terminal equipment mean that there is a risk that on very short local lines the stability loss could be a low as 8 dB (assuming a 6 dB contribution from the network).

7. Delay and echo

7.1 General

The introduction of digital switching and transmission systems into the public network (including cell-based and packet-based technologies) has caused transmission delay and echo control to become increasingly significant issues. There are two main reasons for this:

1. Digital systems generally have higher inherent delay than analogue systems. This means that delay and echo can be experienced on trunks where it was not previously a problem.

2. Digital networks generally have a lower transmission loss than analogue networks. This can result in a higher level of echo being returned to the talker.

The recommended approach to minimise delay and echo effects is to specify limits for end-to-end delay and echo path loss that will ensure that delay and echo are not problems. However, some national calls within the UK will inevitably exceed the delay limits, and echo control will be required for these calls. The deployment of new switch and transport technology will require increasing use of echo control. Echo control can be achieved either by the use of echo cancellers or a high echo path loss.

Longer delays, even when echo is effectively controlled, need to be minimised as transmission delays themselves are a source of quality of service degradation.

7.2 Connections without echo control

For voice applications on UK connections not employing echo control, the NTP - NTP one-way delay should be less than 15ms for at least 95% (see Note 1) of calls. Assuming that customer network delays at each end of the link do not exceed 5ms (as recommended in ETS 300 283, EG 201 050 and EG 202 086), then the end-user to end-user delay should normally be less than 25ms. This aligns with ITU-T Recommendation G.131 that sets 25ms as the limit above which active echo control will be required. It should be noted that the introduction of cell-based or packet-based technology (such as VoIP) into customer networks could result in delays exceeding 5ms in the CPE.

For the small proportion of calls that exceed the 15ms limit, an absolute limit of 25ms is recommended. Users may consider such calls to be unacceptable, since customer network delays could raise the end-user to end-user delay to more than 30ms. The delays on international access connections need to be considered as they will have an impact on echo canceller deployment in the international network. The overall objective is for the NTP to ISC delay to be less than 9ms for at least 95% of calls, with an absolute limit of 12ms.

In situations where calls have to be re-routed around failed sections of the network, it is acceptable for the proportion of calls meeting recommended delay limits to fall below 95%. However, the absolute limits given above should not be exceeded.

The delay limits given above are based upon the assumption that transmission losses are set as described in Section 6 and that the echo path loss is at least 15dB. Since the implementation of the RTTE Directive in April 2000, individual CPE product performance approval standards are no longer applicable. Hence this means that the echo path loss value specified in the approval standards for non-handset voice terminal equipment cannot be relied upon for all types of CPE. In the case of private networks (made up of a number of CPE products), such a problem may not arise and consultation with the private network operator is recommended.

There is a risk that on very short local lines the echo path loss for these devices could be a low as 10 dB (assuming a 6 dB contribution from the network). A digital interconnection between two digital networks should not introduce any gain or loss, so the transmission level of a call crossing between the two networks will be determined at the periphery of the access network (for digital access, the transmission levels will be set by the CPE). The use of in-line transmission loss (i.e. loss that is encountered by the legitimate signal as well as the echo) is not recommended as a means of controlling echo.

Note 1: Compliance with the 95% objectives should be able to be demonstrated by the operator responsible for the quality of the call. In order to minimise the risk of some customers always encountering unacceptable performance, the objectives should be applied on a catchment area basis. Using the 'Commsco' example (see 4.1) 95% of the Commsco's national calls originating from a particular catchment area served by the “Commsco” switch should have NTP-NTP delays of less than 15ms.

7.3 Connections with echo control

Where echo control is provided it is generally true that the performance of the connection for interactive traffic (either voice or data) is degraded as delay increases. It should be an objective when planning connections with echo control to minimise the delay.

7.3.1 National connections with echo control

Connections within the UK requiring echo control fall into three categories:

1. Connections over the fixed network with delays sufficiently long to potentially cause annoying talker echo and delay.
2. Connections between mobile digital networks or wireless access networks and the fixed network.
3. Connections between digital mobile network users or wireless access networks users. (Echo control is by acoustic loss in the case of connections between digital mobile network users).

With connections over the fixed network within the UK it is recommended that 95% of national connections with echo cancellation should meet a 125ms one way delay limit and that no connections should exceed 150ms one-way delay.

Similarly with connections between digital mobile networks or wireless networks and the fixed network within the UK it is recommended that 95% of call connections should meet a 125ms one-way delay limit and that no connection should exceed the 150ms one way delay limit.

With connections between mobiles on digital networks or users of wireless access networks within the UK it is recommended that 95% of calls have a one-way delay of less than 215ms and that no connection should exceed 230ms one-way delay.

See **Note 1** above and **Note 3** below on compliance with 95% objectives.

Note 2: These values will need to be reviewed for the case of national satellite connections.

7.3.1.1 Special considerations for non-TDM networks

The following one-way delay objectives are recommended for the component parts of UK connections for 100% of call cases. There may be access technologies other than mobile and wireless which offer advantages to customers but which add more than the delay allocated to an access portion. It may be necessary to add specific delay objectives for such technologies in the future. The delays presented below are a single figure for each network element. It is recognised that some processes are unidirectional and some configurations lead to different delays in one direction compared to another. However, the customer perception of delay is based the interaction between the two callers i.e. it is based on the round trip delay. So although a unidirectional analysis of delay can be made, it may be possible to show in some cases that only an evaluation of the average of the two directions is necessary.

- Fixed Access (between the UNI and the switched network serving the access): Traditional copper pair 3ms.
- Digital Mobile or Fixed Wireless Access (between the point of A/D conversion and the switched network serving the access): 104ms for GSM access and fixed wireless access; 111ms for 3G access. This value excludes any delay associated with processing in the terminal for handsfree operation.
- Originating Switched Network (between ingress from originating access network and egress to interconnection with subsequent switched network operator): The objective

for an Originating Switched Network is equal to the national network delay objective (see 7.3.1.2).

- Transit Switched Network (between interconnection from preceding switched network operator and interconnection with subsequent switched network operator): The objective for a transit Switched Network is equal to the national network delay objective (see 7.3.1.2).
- Terminating Switched Network (between ingress from interconnection from preceding switched network operator and egress to terminating access network): The objective for a Terminating Switched Network is equal to the national network delay objective (see 7.3.1.2).
- Originating/Terminating Switched Network (between ingress from originating access and egress to terminating access): The objective for an Originating/Terminating Switched Network is equal to the national network delay objective (see 7.3.1.2).

These recommended delay objectives apply to each operator whose equipment forms part of the end-to-end call path. The simplest call path will consist of two access components interconnected by an Originating/Terminating Switched Network.

These recommended delay objectives do not include signal propagation time (which is almost negligible for calls within the UK) nor do they include an allowance for the following functions which introduce a small amount of delay but which should occur only once in the call path:

- Analogue-Digital conversion and Digital-Analogue conversion
- Echo cancellation

The overall effect across all networks of propagation delay, A/D and D/A conversion and echo cancellation should be less than 10ms delay for all UK calls. Each network in a call path shall introduce no more than 1ms of propagation delay for each 100km (direct, point-to-point distance between ingress and egress) the call is carried across their infrastructure. The permissible propagation delay includes any network re-routeing (e.g. for number portability) but excludes the effects of any customer actions, such as call forwarding from the original intended destination of the call.

In the case where an operator manages both the access and the originating or terminating switched network components of a connection, their delay objectives may, if necessary, be concatenated and divided between access and switching, bearing in mind the need to keep delay low. Where access technologies with intrinsically low delays are used operators are encouraged to refrain from using the allowance for access components to increase the delay in their switching component.

7.3.1.2 The national network delay objective

7.3.1.2.1 Networks interconnected using TDM

Operators should endeavour to keep delay to a minimum. For networks which are interconnected using TDM the interim maximum one-way delay across a packet-switched network should not exceed 35ms (excluding propagation delay) however network providers are advised to engineer for a target one-way delay value of 25ms or less in order to limit the degradation of PSTN voice quality.

It is recognised that the interim maximum one-way delay value will result in perceptible degradation for certain UK call scenarios. Therefore the interim maximum one-way delay will be reviewed within 2 years with the expectation of reducing the interim maximum one-way delay value to the target one-way delay value for TDM based interconnect and also defining a IP based interconnect delay objective avoiding many of the delay intensive functions and consequent degradations.

It is recognised that some packet-based networks may have already been deployed and that some operators may have already committed themselves to procure equipment which cannot achieve the delay target. The delay target cannot be retrospectively applied to such networks but in the interest of maintaining PSTN quality it is recommended that they move towards this figure as soon as possible.

Operators should be aware that in the event of excessive delay causing QoS problems to customers the primary responsibility for resolving the delay problem will lie with any operators which have deployed networks with delays significantly above the target delay. Networks unable to resolve internal network delay issues would be expected to migrate to packet based interconnects or aim to mitigate problems by adopting alternative network interconnect routeing or topology.

Note 3: The introduction of islands of packet-based technologies into the UK PSTN may have an effect on perceived QoS. As these technologies become more widespread the probability of a call undergoing multiple packetisations increases and the accumulated impact on QoS, in terms of delay and speech quality, may be perceived by some users. It is possible that calls between a very limited number of certain number pairs may encounter reduced quality during the earlier stages of PSTN evolution towards IP. Calls involving complex routings are the most likely to be affected. Network operators should discuss the potential impact of any planned introduction of new technologies, particularly IP, with interconnected operators in order to minimise any potential adverse effect on call quality. Section 8 describes a set of general planning guidelines that should be carefully considered when incorporating new technologies into the UK PSTN.

7.3.1.2.2 Networks interconnected using IP

Operators should endeavour to keep delay to a minimum. Using IP interconnects for IP networks reduces the possibility of having multiple delay-intensive packetisation functions. The IP interconnect may also reduce the number of dejitter buffer instances, although as packet delay variation is thought to be additive this requires that the remaining dejitter buffers are larger and there may not be a reduction in the overall delay.

As the packetisation and dejitter functions are directional functions, any recommendations will need to allocate separate objectives to the two directions of the call for a given connection element.

By considering the delay associated with specific functions (e.g. packetisation, switching and packet dejittering) it should be possible to allocate objectives for different scenarios of originating, terminating and transit cores with both IP and TDM interconnect at the same time.

Specific delay objectives for networks interconnected using IP will be added to a future issue of this document.

7.3.1.2.3 Networks interconnected using other technologies

Delay objectives for networks interconnected using technologies other that TDM and IP are currently outside the scope of this document.

7.3.2 International connections with echo control

International connections originating or terminating within the UK requiring echo control fall into two categories:

1. Fixed network originated connections.
2. Mobile network or wireless access network originated connections.

The maximum one way delay for the international connection will depend on the nature of the destination network e.g. fixed, digital mobile or wireless access and the transmission media used e.g. satellite or cable.

With international connections using echo control it is recommended that connections with a one way delay exceeding 400ms should only be provided under exceptional circumstances such as back-up circuits provided via satellite to maintain service in the event of cable circuit congestion or failure. In addition some destinations may only be reached via satellite routeings, opening up the possibility for long delay calls exceeding ITU-T G.114 guidelines.

The UK components of international connections should follow the guidance given above regarding delay.

7.4 Echo Control

Echo cancellation should be employed in any network that uses packet- or cell-based technologies as the delay introduced by such technologies is likely to push the end-to-end delay above 25ms.

Echo cancellers compliant with the requirements of ITU-T Recommendation G.168 should be employed.

The presence of an echo canceller in a call path should be signalled to adjacent networks in accordance with ITU-T Recommendation Q.115. Ideally all echo cancellers should be disabled with the exception of the two closest to the two potential sources of echo. Annex B illustrates a number of scenarios in which echo is required and shows which echo cancellers should be disabled by signalling. Note: Existing signalling systems do not enable the presence of echo cancellers in private networks to be signalled to public networks. If G.168 compliant echo cancellers are used in private networks then the potential presence of two echo cancellers working in the same direction of a call path will not degrade speech quality.

In general, the tail capacity must be at least 64ms to prevent any potential failure to cancel echo in some routeing scenarios. As an exception to this, the tail capacity can be smaller when the echo path is known to be less than 64ms (e.g. when the echo canceller is cancelling acoustic echo in a terminal).

Circuits with echo suppressors can be connected with circuits equipped with echo cancellers without additional performance degradation caused by the canceller; however, the overall performance will be limited by that provided by the poorer performing device.

In some cases echo cancellers conforming to G.165 may have already been deployed. The interaction of G.165 and G.168 echo cancellers may need to be the subject of further study**.**

Note that much new detail regarding the use of echo cancellers is provided in ITU-T Recommendation G.108.2.

8. Planning guidelines for the incorporation of new technologies into the UK PSTN

Careful consideration should be given to the effect on QoS (notably delay) when planning the introduction of new technologies into the PSTN. The guidelines given in this section should be taken into account in order to minimise the impact on QoS of the introduction of any new technologies.

This document only addresses the performance limits of public networks (i.e. the performance between a pair of Network Termination Points). The existence of private networks cannot be ignored in network planning because they will have an effect on end to end impairments. Historically the NTP-NTP delay performance of the PSTN was below 15ms and private network operators have been able to use this low delay to allow the implementation of low bit rate codecs and IP technology and still achieve “acceptable” quality for calls from the private network to the PSTN. As IP technology is deployed in the PSTN, private network operators should bear in mind that the delay in the PSTN will increase. The limits for PSTN delay set out in sections 7.3.1 and 7.3.1.2 of the current document should be taken into consideration when planning access and private networks.

Note: The ITU-T’s E-model (as contained in Recommendation G.107 and G.108) provides transmission planning guidance based on the perceived effect of impairments, including those of delay and codecs.

Note: In the context of section 8 the term “jitter” refers to IP packet jitter (also called “packet delay variation”). This should not be confused with jitter in the sense of the timing of digital signals, as discussed in section 9.

8.1 General Rules

Rule 1. In order that the number of "packetisations", and therefore delay, can be kept to a minimum, networks carrying voice over IP technology should ideally be interconnected using IP. *Note: -The availability of TDM and IP or other broadband interconnects within the UK is driven by other considerations as well as performance. Timescales for the implementation or withdrawal of interconnect products is well beyond the scope of this document.*

Rule 2. Care should be taken to ensure that the performance of PSTN traffic is not adversely affected by other traffic types. The network should deliver a very low latency and jitter and preserve PSTN grade of service (GoS) even under high or rapidly fluctuating load and/or mix of other traffic types.

8.2 Delay and Packet Jitter Rules

Rule 1. Transmission delay should be minimised. See section 7.3 for detailed guidance.

Rule 2. IP packet jitter should be minimised. *Note: The specification of packet jitter objectives is for further study.Operators should be aware of the contribution that jitter in* *packet streams entering the public network from a private network makes to the performance of the overall end-to-end connection.*

Rule 3. IP jitter buffer delay should be minimised. *Note: There is a trade-off between this parameter and packet loss, with shorter jitter buffer delays potentially resulting in higher packet loss. This should be borne in mind when optimising these two parameters.*

Rule 4. A de-jitter function is required at the point of transition from a packetised voice bit stream to a synchronous voice bit stream. For calls within the UK, de-jitter functions should not be applied on a per network element basis or at the boundary between two IP-based networks interconnected with IP. *Note: There may also be a need use a de-jitter buffer at points where the codec type or speech sample size are different to those specified in 8.3 Rule 1 and Rule 2. ITU-T Rec. Y.1541 allows for up to 50ms of jitter end-to-end in the most stringent QoS Classes. Jitter of this magnitude may be present on calls entering the UK.*

Rule 5. In order to help minimise delay, adaptive jitter buffers are recommended. For calls involving fax , dial-up modem and ISDN clearmode the jitter buffer should be fixed at an optimum value on a per connection basis so as to avoid data loss during dynamic changes of the jitter buffer size. For ISDN clearmode the fixed jitter buffer size should be established on call set-up (i.e. before any user data is sent). For fax and other modem calls the call will start as a voice call with an adaptive jitter buffer, the jitter buffer should then train to a fixed setting.

8.3 Codec Rules

Rule 1**.** The G.711 A-law codec, without silence suppression, shall be used as the default interoperability option for interconnect purposes because it has the lowest impairment value and therefore it allows more delay for a given voice quality level. *Note: Connections can be established using other codec pairs, either by bilateral interconnect agreement or end-to-end per-call negotiation. Similarly other codec pairs can be employed within one operator's network or in access networks. In order to guarantee adequate QoS codecs other than G.711 should only be employed in situations where there is certainty over the routing of calls. Operators should be aware of the potential impact that codecs other than G.711 can have on perceived speech quality.*

Rule 2. The speech frame size and the number of speech frames per packet should be minimized when using IP technology. In order that the effects of packetisation on delay can be kept to a minimum, for voice carried over IP technology a speech frame size of 10ms for G.711 should be supported for UK NGN-interconnect. – *Note: Speech frame sizes of other than 10ms may be more appropriate for certain codec types. Operators should be aware of the potential impact that the additional delay caused by speech frame sizes of more than 10ms can have on perceived speech quality*

Rule 3. Operators should be aware that silence suppression may adversely affect the perceived quality of a call and should not be used except in situations where there is certainty over the routing of calls.

Rule 4. The following add delay and or coding impairment and should be avoided where possible:

- multiple packetisations *Note: The benefit of avoiding multiple packetisations is only gained if networks use the same speech frame size and number of speech frames per packet. See Rule 2 above.*

- transcoding

- asynchronous tandeming[1] -

8.4 Packet Loss Rules

Rule 1. IP packet loss for voice traffic should essentially be zero. Each operator in the UK should to aim achieve a packet loss ratio of better than 0.01% - this loss in the UK leg of an international connection should enable the end-to-end loss given in QoS Classes 0 and 1 in ITU-T Recommendation Y.1541 to be met. In the worst case UK scenario this would not exceed 0.12%. Appendix I of ITU-T Recommendation G.113 gives some guidance on the effect of packet loss on speech quality as estimated by the G.107 E-model.). *Note: There is a trade-off between packet loss and jitter buffer delay, with shorter jitter buffer delays potentially resulting in higher packet loss. This should be borne in mind when optimising these two parameters.*

Rule 2. The support of ISDN clearmode on in an IP-based network requires more stringent packet loss objectives than the support of voice traffic. Objectives for end-to-end packet loss for support of clearmode will be added to a future issue of this document.. The allocation of this objective to individual operator domains is also for further study. *Note: The most stringent packet loss requirements in Y.1541 are for the provisional QoS Classes 6 and 7, which have an end-to-end packet loss objective of* $1x10^{-5}$.

Rule 3. When using IP, packet loss concealment with G.711 should be considered. G.711 Appendix 1 PLC should be used, unless the packet loss ratio of a connection segment is low enough not to require PLC at the endpoint of that segment. For example, the packet loss rate required to support packet transmission of clear 64kbit/s ISDN channels may be very low, such that a connection which uses the same network for transmission of packet voice may not require PLC. The potential quality improvement from PLC should be balanced against the delay penalty that PLC may introduce. A delay of 3.75ms is characteristic of Appendix 1/G.711 PLC. Some proprietary PLCs (integrated with the jitter buffer) claim zero added delay. With TDM interconnect, the IP-TDM gateway at the network egress is responsible for applying PLC as necessary, and the consequential delay would use some of that network's delay budget. Following the introduction of IP interconnect, the IP-TDM gateway terminating the multi-operator IP connection segment would be responsible for PLC. It is possible that another operator's preceding network might introduce a level of packet loss which makes PLC necessary in a very low loss network which terminates a multi-operator IP connection segment.

8.5 Post Dial Delay

The introduction of new signalling methods associated with IP technology into the PSTN will mean that calls are processed in a different way to that associated with Signalling System No.7. From a service perspective it is important that any new signalling methods do not have

[1] The term asynchronous tandeming originally referred to a series connection of speech coders that requires digital to analog conversion followed by re-sampling and re-encoding. Today it also refers to cases where the speech samples must be reconstructed and then re-encoded by the next codec.

an adverse effect on customer perception of connection processing performance parameters such as post dial delay (PDD).

There are no current UK standards for PDD, but in order to preserve the customer experience of PDD it may be necessary to define some design objectives in a future edition of this document.

Any design objectives should take account of different call types (e.g. simple fixed-to-fixed PSTN calls, fixed-to-mobile calls and calls involving Intelligent Networks (IN)) and different network loads. It may be necessary to define a mean and/or a percentile-based objective (e.g. an objective for 95% of calls).

In the meantime, operators choosing to deploy IP-based technologies into the PSTN should ensure that they do not have an adverse impact on connection processing performance.

9. Jitter and wander

9.1 Terms and definitions

Jitter: Is the short term variations of the significant instants of a digital signal from their ideal positions in time.

Wander: Is the long term variations of the significant instants of a digital signal from their ideal positions in time.

9.2 Jitter and wander in digital networks

Jitter and Wander are inherent characteristics of digital networks.

Jitter and Wander from various sources can accumulate, therefore, limits are required for the maximum jitter and wander that can appear at network interfaces and at the input and output of network equipment. Limits are also required for the jitter and wander generated by equipment and the jitter gain imposed by equipment.

Network limits for the maximum output jitter and wander at hierarchical interfaces are described in ITU-T Recommendations G.823 (PDH) and G.825 (SDH).

Jitter limits for digital sections may be found in ITU-T Recommendation G.921

9.3 Jitter limits appropriate to digital equipments

The jitter performance for individual digital equipments can be divided into the following three categories.

9.3.1 Jitter and wander tolerance of digital input ports

All digital input ports shall be able to accommodate levels of jitter up to the maximum network limit as defined in ITU-T Recommendation G.823 (PDH) and G.825 (SDH).

9.3.2 Maximum output jitter in the absence of input jitter

The actual limits of output jitter in the absence of input jitter for any valid signal condition may vary depending on the type of equipment. However in all cases the limits shall never exceed the maximum permitted network limits described in G.823 (PDH) and G.825 (SDH).

9.3.3 Jitter transfer characteristics

The jitter transfer functions shall not exceed the limits detailed in ITU-T Recommendations G.823 (PDH) and G.825 (SDH).

10. Network synchronisation

10.1 General

In the Plesiochronous Digital Hierarchy (PDH) synchronisation is necessary to control slips occurring between elements which switch or cross-connect traffic below the 2Mbit/s level of the PDH hierarchy. In the Synchronous Digital Hierarchy , this is extended to include all levels of the hierarchy including elements that provide add/drop functions (ADMs).

Slip is the deletion or repetition of a primary rate (2Mbit/s) frame (125 μs) which occurs in an elastic buffer due to a difference between the write-in (remote) and read-out (local) clocks where the remote and local clocks are derived from separately timed networks.

The period between slips (slip rate) depends upon the accuracy of the clocks used, for example with 2 Caesium based (atomic) clocks to ITU-T G811 each with an accuracy of ± 1 part in 10" the worst case slip rate will be once every 72.3 days. In the SDH this is manifested as pointer movement . Pointers are used to indicate the position of tributaries within an SDH frame. Adjustment occurs in bytes or multiples of bytes and reflects the difference between write-in and read-out clocks (remote and local as in the PDH case).

Because of the potentially large swings in byte movement it is recommended that the SDH should not be used to carry timing at the VC-12 level. A preferred method is to carry timing at the STM-N rate.

Most operators running digital networks use Primary Reference Clocks (PRCs) based upon the ITU-T recommendation G.811 and have built synchronisation networks which ensure that all network clocks are traceable to the PRC.

The main ITU-T recommendations applicable to synchronisation matters are:

- G.803: Architecture of transport networks based on the SDH.
- G.810: Definitions and terminology for synchronisation networks.
- G.811: Timing characteristics of primary reference clocks.
- G.812: Timing requirements of slave clocks suitable for use as node clocks in synchronisation networks
- G.813 Timing characteristics of SDH equipment slave clocks (SEC).
- G.822: Controlled slip rate objectives on an international digital connection.

The main ETSI standards that are applicable to synchronisation matters are covered in ETS 300 462 part 1 through to part 7.

ETS 300 462 – 1: Definitions and terminology for synchronisation networks
ETS 300 462 - 2: Synchronisation network architecture
ETS 300 462 - 3: The control of jitter and wander in synchronisation networks
ETS 300 462 - 4 :Timing characteristics of slave clocks suitable for synchronisation supply to Synchronous Digital Hierarchy (SDH) and Plesiochronous Digital Hierarchy (PDH) equipment
ETS 300 462 - 5: Timing characteristics of slave clocks suitable for operation in Synchronous Digital Hierarchy (SDH) equipment
ETS 300 462 - 6:Timing characteristics of primary reference clocks
ETS 300 462 - 7:Timing characteristics of slave clocks suitable for synchronisation supply to equipment in local node applications

Further guidance on synchronisation may also be found in recommendation ITU-T G.803 (Architecture of transport networks based on the SDH).

A UK National Network Timing Plan for implementation of synchronisation within the UK is described in PNO-IG/NNTP.

10.2 The synchronisation hierarchy

Many established operators deploy their own Primary reference clocks (PRCs) based on the ITU-T G811 standard.

Slave clocks (i.e. those clocks which are traceable to the PRC) are defined by the ITU-T in recommendation G812. Six different types of clocks are specified within G.812, i.e. Type 1 through to Type 6.

Specifically Type 1, 5 and 6 are used in the 2,048Mbit/s network hierarchy. Type 1 is equivalent to the ETSI ETS 300 462-4 Synchronisation Supply Unit (SSU) as used in the SDH synchronisation hierarchy. This clock level is also acceptable as for use in PSTN network applications and may either be integrated within another part of equipment such as an SDH cross-connect or it may be deployed as a piece of Standalone Synchronisation Equipment (SASE) which will time multiple equipments.

Type 5 and Type 6 clocks are equivalent to the older clocks that are more traditionally found in the PSTN / PDH network hierarchy at the transit and local levels.

The transit clock has a superior holdover stability compared to the local clock and is used mainly at international switch centres. The local clock is used in other switch centres and peripheral sites. The relative performance of these clocks is described in Table 10.1.

The SDH equiment clock (SEC) is described in G.813 and under normal operation it should be locked and traceable to the network's G.811 PRC. It is the lowest order of clock in the SDH network.

The lowest level of clock classification is the basic internal clock of the PDH mux equipment which has an accuracy of $\pm$50 ppm at 2.048Mbit/s.

Table 10.1: Long term performance of clocks

	Accuracy	Holdover Stability	Offset	Days to Slip Minimum	Slip /day Maximum
Primary ref clock	$1x10^{-11}$			70.00 days	
Transit node clock G.812 Type 5 Clock		$1x10^{-9}$/d	$5x10^{-10}$	1.44 days	0.69 slips
Local node Clock G.812 Type 6 Clock		$2x10^{-8}$/d	$1x10^{-8}$	0.07 days	13.82 slips

G.812 Type 5 and 6 clocks have been used in Table 10.1 because they are equivalent to the older clocks that are more traditionally found in PSTN / PDH network hierarchies at the transit and local levels.

10.3 Availability of synchronisation references

As a result of the synchronisation hierarchy i.e. the traceability of all clock sources to a PRC complying with ITU-T G811, it is possible to derive timing references over traffic carrying connections by means of bilateral agreement between operators. Alternatively an operator can provide his own PRC. This could take the form of a standard Caesium based system complying with ITU-T G811 or alternatively an off air system using LORAN C or GPS (Global Positioning System) with a secure backup such as a rubidium. In the case of the latter, a performance of 2 orders of magnitude better than ITU-T G811 can be achieved.

10.4 Slip-rate performance

10.4.1 Access

The allocation of slip-rate performance for an access connection can be identified as the local portion in Table 2 G.822 of the ITU-T Recommendations (see Table 9.2).

Table 10.2: Slip rate objective for the access network (Local Loop of HRX)

Performance Category	Mean Slip Rate	Proportion of the total available time in 1 year

(a)	Not more than 3 slips in 24 hours	99.56%
(b)	Between 2 slips per 24 Hours and 30 slip per hour	0.4%
(c)	Worse than 30 slips per hour	0.04%

Category (a) corresponds to normal operation (it should be noted that the nominal slip performance should not exceed 1 slip in 72.3 days between 2 PRC-controlled networks due to plesiochronous operation alone).

Category (b) performance corresponds to slip-rate degradation due to a G.812 Type 5 clock in free-run mode, although many of these clocks would maintain category (a) slip rates in free-run for a considerable period.

Category (c) slip rates may be due to the lowest level clock (internal clock, e.g. G.812 Type 6) in free-run mode.

10.4.2 Trunk

The allocation of slip-rate performance for a Trunk connection can be identified as the national transit portion in Table G.822 of the ITU recommendations (see Table 9.3).

Table 10.3: Slip rate objectives for the Trunk Network (Transmit Portion of HRX)

Performance Category	Mean Slip Rate	Proportion of the total available time in 1 year
(a)	Not more than 3 Slips in 10 days.	99.934%
(b)	Between 3 slips in 10 Days and 30 slips per Hour.	Note more than 0.06%
(c)	Worse than 30 slips.	Less than 0.006%

The categories (a), (b) and (c) are as for Table 10.2.

10.5 Synchronisation for networks interconnected using IP

There is a need to maintain the accuracy of current synchronisation solutions while there are still TDM elements in the UK PSTN. It is recommended that current synchronisation solutions for the UK PSTN/ISDN are extended to any packet gateways which may generate RTP media streams which could traverse an IP interconnect, as these solutions are known to achieve sufficiently good synchronisation. These solutions can also be used to support the measurement of IP packet delay variation incoming across an IP interconnect where such a measurement is required. Such strict synchronisation may not be required in an all IP network, however, there may be continued benefit in the provision of highly accurate clocks.

11. Other parameters

11.1 Error performance

11.1.1 General issues

Error performance of digital transport networks is of key importance as it determines the end to end performance of both end to end digital services and analogue services supported over a digital core network. In addition the transport network supports other network and customer functions such as the transfer of signalling information.

There are 2 principal ITU-T Recommendations covering error performance issues:

- G.821: Error performance of an international digital connection operating at a bit-rate below the primary rate and forming part of an ISDN.
- G.826: Error performance parameters and objectives for international constant bit-rate digital paths at or above the primary rate.

The relationship between the 2 recommendations is important as it is likely that both will be required to adequately cover error performance in the evolving UK networks.

- G.821: addresses design objectives and is applicable to Nx64kbit/s circuit switched connections (where 1<N<32) assuming there is negligible contribution from switches or multiplexers; in addition it does not include the error bursts arising from controlled slips. G.821 specifies objectives during available time in terms of errored seconds and severely errored seconds. It recognises that error measurements will generally be made on paths operating at higher bit rates (where G.826 applies) rather than end to end on 64 kbit/s connections.
- G.826: gives performance objectives for transport paths (at primary rate or above) that support the circuit switched connections. The parameters are specified in a block-based form to be compatible with the inherent error detection codes (e.g. Cyclic Redundancy Check(CRC) or Bit Interleaved Parity(BIP)) used in transport paths.

The ITU-T objective was that there should be basic compatibility between G.821 and G.826 such that transport paths meeting the objectives of G.826 are highly likely to be able to support end to end 64 kbit/s connections that meet G.821 requirements. This may not be true in all practical cases.

11.1.2 End to end design objectives

An end to end design performance objective for 64 kbit/s switched connections should initially be based on G.821 with appropriate scaling for the UK (the G.821 limits refer to a 27,500 km hypothetical reference connection).

The end to end limits for a 27,500 km reference connection as given in G.821 are as follows:

> Fewer than 0.2 % of 1 second intervals to have a bit error ratio worse than $1\text{-}10^{-3}$ (severely errored seconds)
> Fewer than 8% of 1 second intervals to have any errors (equivalent to 92 % error-free seconds).

G.821 recommends apportionment on the basis of block allowances assuming that the access part of a network (the first 1250 km) and the central 25,000 km receive separate allowances. Connections within the UK will generally be less than 2500 km in length hence it is proposed that the end-end UK limit should be no greater than the G.821 recommended allowance for the two access parts i.e.:

The following limits are recommended:

Fewer than 0.12% of 1 second intervals to have a bit error ratio worse than $1\text{-}10^{-3}$
Fewer than 4.8% of 1 second intervals to have any errors.

It is important that the relevant notes and explanations in G.821 are taken into account when designing to these objectives. For example the above limits only apply during available time and need to be averaged over suitably long time periods. e.g. 1 month.

It should also be noted that the limits apply to 64 kbit/s switched connections. They are not relevant to sub 64 kbit/s paths , for example the mobile digital links carrying compressed voice traffic.

11.1.3 Network segment objectives

Performance objectives for network segments should be set using the error performance events and parameters defined in G.826 and will apply to paths at or above the primary rate.

Paths within the UK should meet the objectives given for the national portion of the international reference connection. Essentially these consist of a block allowance plus a distance related allowance for each segment at the various transmission rates of between 1.5 Mbit/s and 160 Mbit/s.

The detailed parameter definitions and objectives are not reproduced in this document as it is important to take into account all the issues detailed in G.826 when developing a set of objectives.

Note this document does not cover performance for higher layer encapsulation such as ATM although ITU-T Recommendations (e.g. I.356 – ATM Cell Transfer Performance) exist. At the present time as these are normally transported via primary rate/SDH hierarchical bearers, compliance with G.821/826 should ensure satisfactory performance of these higher layers between networks.

11.2 Quantising distortion

(Editor's Note:
The contents of section 11.2 will be revised by E2E QoS TG at a later stage, and added as an up-issue to this document. Quantisation distortion is too limited to be a valuable speech quality assessment tool in modern networks. It is intended that revisions to this section will be based on the more rounded approach to impairments given in the ITU-T's E-model (G.107).)

11.2.1 Background

Quantising Distortion is introduced during the analogue to digital conversion process where the analogue input is sampled and encoded into a finite set of values. On decoding it is not possible to recover exactly the original analogue signal and the resulting difference is called quantising noise or quantising distortion. Quantising noise can also be introduced by digital recoding process (e.g., code conversion from 64 kbit/s to 32 kbit/s) where additional distortion may be introduced.

Quantising distortion is generally in terms of Quantising Distortion Units (qdu) where 1 qdu is equivalent to the distortion introduced by a single analogue-digital - analogue encoding cycle using 8bit coding to ITU-T Recommendation G.711 (i.e. standard PCM encoding). The number of qdus introduced by other coding systems can be estimated using subjective techniques. Example values are given in Table 10.1.

Table 11.1: Examples of qdu allocations

Digital Process	Quantising Distortion Units
8 - bit PCM codec pair (G.711)	1
32 kbit/s ADPCM (G.726 or G.727) (PCM -ADPCM - PCM)	2.5
GSM Mobile Radio (each end)	7
16 kbit/s LDCELP (G.728) (analogue- PCM - LDCELP - PCM - analogue)	3.5
8-7-8 bit transcoding (A or u law)	3

Sources: ITU-T Recommendation G.113 and ETSI GSM 03.50

The relevant international recommendation is ITU-T G.113. (Transmission Impairments) which gives qdu limits for the overall international connection with sub divisions into the national and international portions. Broadly these are as follows:

End to end: 14 qdu (with short term relaxation to 18 qdu)

National extension (to subscriber) : 5 qdu (with short term relaxation to 7 qdu)

International segment: 4 qdu

The notes to G.113 are significant when planning qdu allocations, e.g. the ability to assign half the indicated values to the send or receive parts.

11.2.2 Performance objectives

The objectives should follow the G.113 guidance but note the following:

- UK switched digital connections between local exchanges are generally expected to provide transparent 64 kbit/s paths without use of speech compression or other processing techniques.
- Access connections from the customer up to the local exchange (including private networks) may utilise speech compression techniques.
- Mobile radio access systems will use speech compression techniques.
- The G.113 long term limits should be applied to the fixed network connection

portion shown on the Reference Connection while the "short term" relaxations are appropriate for the mobile network. The recommended limits are shown in Figure 10.1.

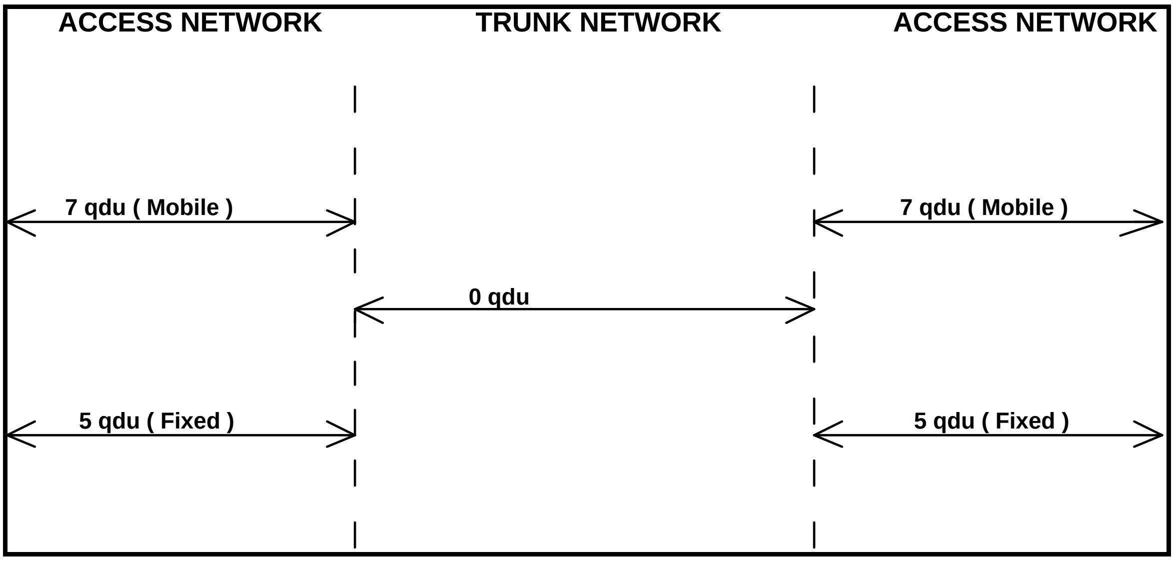


Figure 10.1: Reference model qdu limits

10.3 Noise

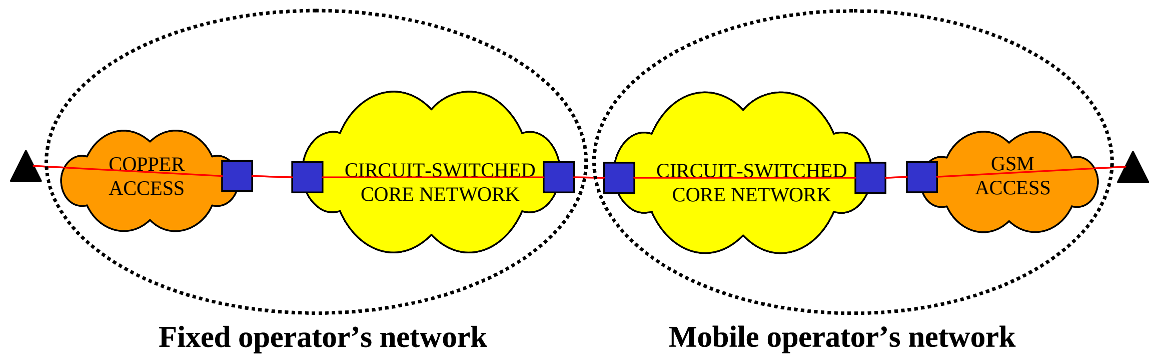
Noise can have serious impact on the end to end performance perceived by users if levels are not adequately controlled. However with modern networks using digital transmission and switching, potential analogue noise sources are normally limited to the analogue access sections and the analogue to digital conversion function**.** Recommended limits for these sources are given in the relevant ITU-T Recommendations e.g. the Q.500 series recommendations include noise limits for digital local exchanges.

Errors on digital transport systems will also be perceived as noise sources by telephony customers . The error performance recommendations above for a 64 kbit/s path will provide acceptable voice performance for the majority of customers. The effects of errors on mobile radio links supporting compressed voice is for further study.

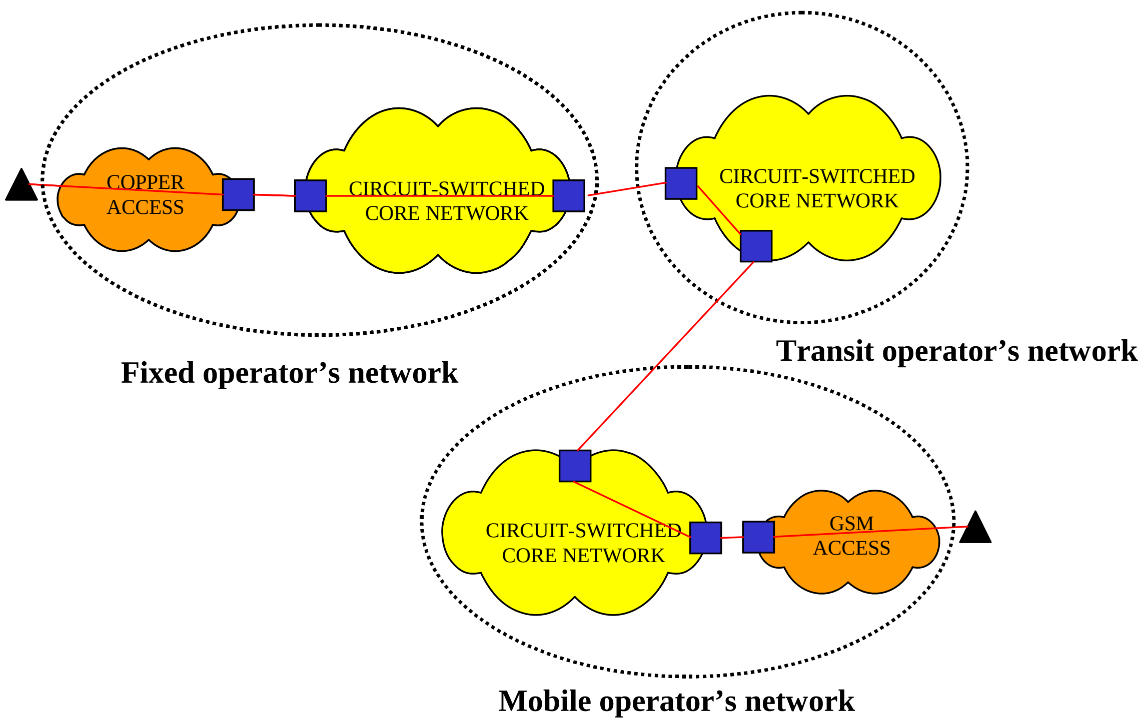
10.4 Crosstalk

Potential analogue crosstalk sources in modern networks are normally limited to the analogue access sections and the analogue to digital conversion function. Recommended crosstalk limits for exchanges are given in the ITU-T Q.500 series Recommendations.

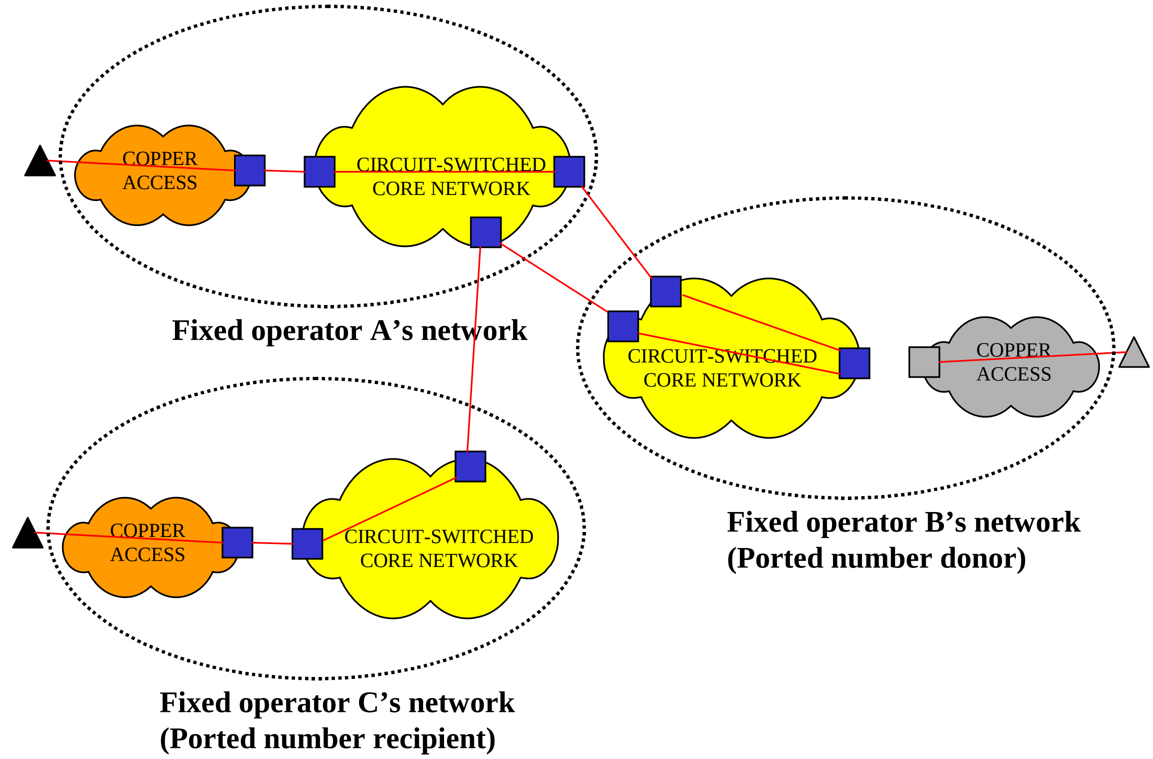
Annex A – Examples of describing end-to-end connections using connection elements



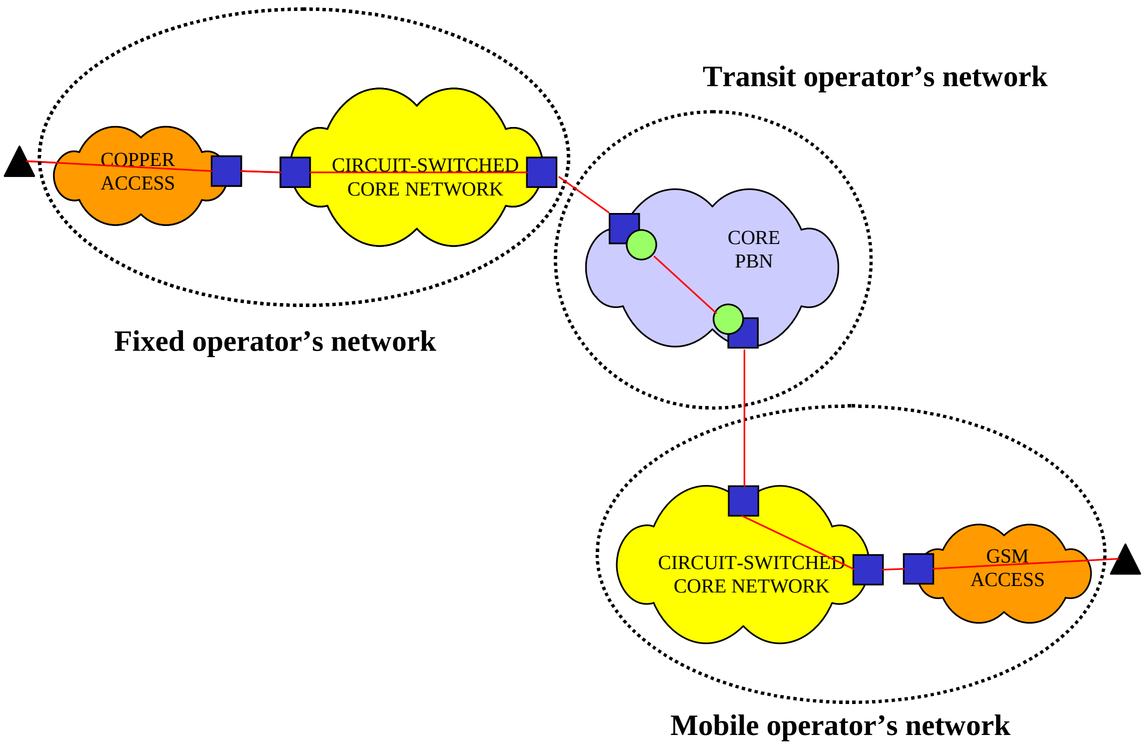
Example 1: Traditional PSTN to GSM with direct interconnect



Example 2: Traditional PSTN to GSM without direct interconnect



Example 3: Traditional PSTN to ported PSTN number (no direct interconnect between B and C)



Example 4: Traditional PSTN to GSM customer via a packet-based transit

Annex B – Example echo cancellation scenarios

Diagram showing location of echo cancellers for call involving Mobile

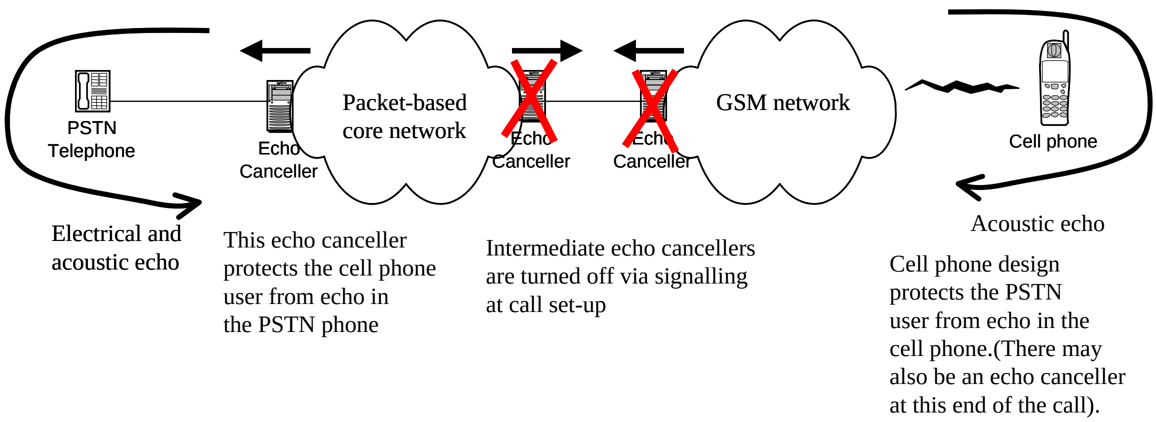


Diagram showing location of echo cancellers for Fixed Networks

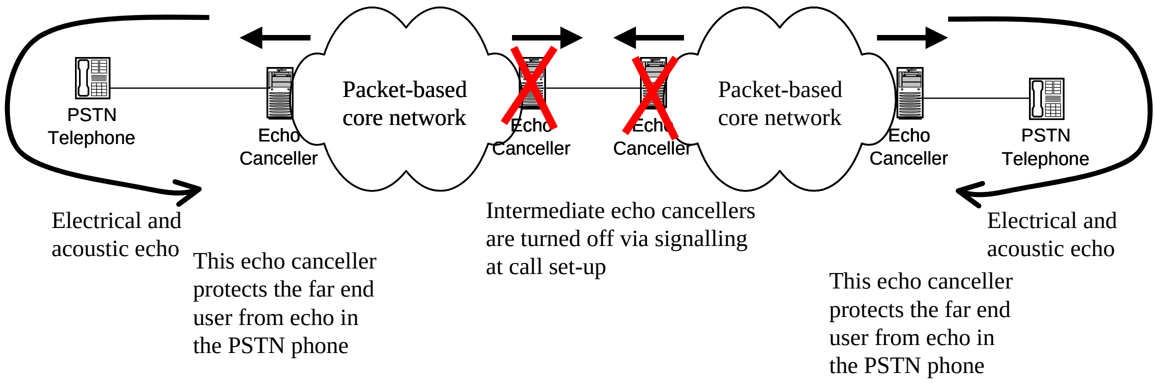


Diagram showing location of echo cancellers for Fixed Networks

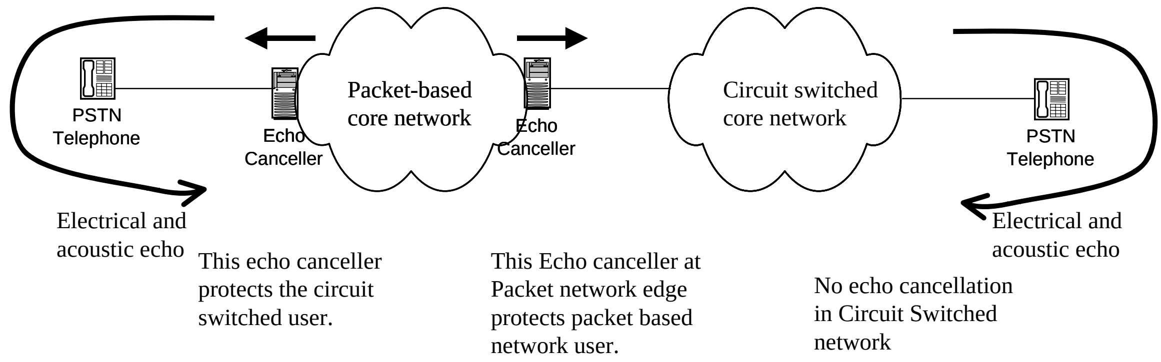
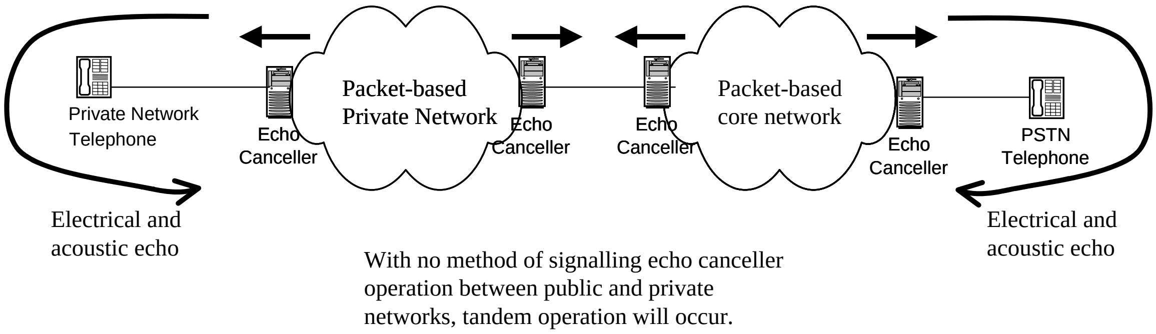


Diagram showing location of echo cancellers for calls with Private Packet based networks.



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